TUESDAY MORNING, 27 JUNE 2017 BALLROOM B, 8:00 A.M. TO 9:00 A.M.

Session 3aIDa

Interdisciplinary: Keynote Lecture

Keynote Introduction—8:00

Invited Paper

8:05

3aIDa1. Hearing as an extreme sport: Underwater ears, infra to ultrasonic, and surface to the abyss. Darlene R. Ketten (Otology and Laryngology, Harvard Med. School, Boston Univ. and Harvard Med. School, Boston, MA 6845, dketten@whoi.edu)

It has been argued that "hearing" evolved in aquatic animals. Auditory precursors exist in fossil Agnatha and cephalopod statolithic organs, but how/when/why did a dedicated acoustic receptor, the true primordial ear, first appear? Did hearing arise linearly or independently, in parallel, in and out of the water? Modern aquatic species have an extraordinary range of auditory systems, from simple pressure receptors to complex biosonar systems. What drives this breadth of "hearing"? Vertebrate ears reflect selective pressures. While vision, touch, taste, and olfaction are important, only hearing is ubiquitous. Even natural mutes, like goldfish and sea turtles, listen. Ears capture passive and active sound cues. Auditory structures, honed by habit and habitat, delimit species abilities to detect, analyze, and act on survival cues. Cochleae, from shrews to bats to wolves to whales, evolved from the essential papilla of stem reptiles, elongating, coiling, increasing in complexity that enhanced frequency discrimination, but with heads tuned to the physics of sound in their media. Air-water parallels evolved: ultrasonic echolocators and massive infrasonic specialists. The ear then is a window into the evolutionary push-pull driven by three tasks that shaped the several thousand elements packed into every auditory system: feed, breed, and survive another day.

TUESDAY MORNING, 27 JUNE 2017 ROOM 206, 9:15 A.M. TO 12:20 P.M.

Session 3aAAa

Architectural Acoustics: Retrospect on the Works of Bertram Kinzey I

Gary W. Siebein, Cochair
Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Chair's Introduction—9:15

Invited Papers

9:20

3aAAa1. Professor Bertram Y, Kinzey, Jr., mentor, scholar, practitioner, artist, and colleague. Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Professor Bertram Y. Kinzey, Jr., had significant impacts on the shape of architectural acoustics today through his teaching, professional practice, research, and service. He influenced several generations of students at Virginia Tech and the University of Florida where he developed innovative, laboratory-based courses to teach architects about acoustics and other environmental technology subjects. At Florida, he initiated graduate coursework to allow students to specialize in Environmental Technology and later architectural acoustics as part of the professional Master of Architecture curriculum, with a thesis in acoustics in addition. His teaching was based on ideas expressed in his landmark textbook, *Environmental Technologies in Architecture* that was published as a primer in the art and science of sustainable design in 1963, over 50 years ahead of its time. His students entered the profession with ideas of sustainability, integrating serious, graduate level research into their design work, grounded by the experiences that Bert brought to the classroom from his practice where he consulted on most of the major buildings built in Florida from the 1960s until his retirement many years later. The acoustical richness of the buildings that he worked on sounds clearly in all of the cities and towns where he worked!

9:40

3aAAa2. Contrapuntal design: The influence of Bertram Y. Kinzey, Jr., on architectural pedagogy at the University of Florida. Martin A. Gold (Architecture, Univ. of Florida, 231 ARCH, Gainesville, FL 32611-5702, mgold@ufl.edu)

Bertram Kinzey introduced modes of analysis, conceptualization, and exploration, as a dedicated and productive researcher, author, teacher, architect, consultant, and musician that were transformative and continue to reverberate internationally through the work of his students. Bert's insightful theoretical and practical relationships between music, acoustics, and architecture were drawn from his reflective approach to teaching and refined through active practice. Perhaps foundational to his educational contribution, is his interest and talent as an organist and organ builder leading to his use of "contrapuntal" to describe the interaction of music and architecture—new notes overlapping those lingering in the space and time of reverberance. Overlapping, independent, yet harmonizing elements might also describe Bert's approach to architectural design as a theoretical construct; an integrative process; and as a measure of great architecture. Contrapuntal thinking is well known in literature and music but is just emerging in architecture. Bert introduced this important idea—among many others at the University of Florida—through his writings and tenure as a UF professor. This paper discusses Bert's 'contrapuntal thinking' and the subsequent influence on architectural education through a recent graduate studio project for a new arts park and concert hall for the community of Sarasota, Florida.

10:00

3aAAa3. A living example—An attitude towards sustainable living. Weihwa Chiang (Vice President Office, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd. Section 4, Taipei 106, Taiwan, whch@mail.ntust.edu.tw)

A sustainable built environment aims for minimizing environmental impact while maintaining healthy living. The Acoustics and Environmental Technology programs started by Professor Bertram Y. Kinzey, Jr., at the University of Florida were established with architecture design integration in mind. The programs and Professor Kinzey's influence were not limited to the books and theories but also to the lives of his students. His remarkable work, Bailey Hall at Broward Community College, the University Auditorium renovation at UF and classroom are valuable memories to students who sat in them daily for decades. His passive design signature house in Gainesville is another unforgettable inspiration for students. It was where he successfully applied sustainable design concepts and theories in reality in the early 1960s. These are evidence of how he inspires students by his life as a solid practitioner as well as a modern "Renaissance man." Despite his retirement from UF, his influence has been passing on and spread around the world continually. His pass-through guidance has been followed in teaching, research, design and planning projects with a holistic approach. It not only peruses technical excellence with thoughtful social-economic concerns but heavily relies on "Kinzeian" humanity and attitude for sustainable living and architecture.

10:20

3aAAa4. The shape of sound. Hyun Paek and Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com)

The work of Bertram Y. Kinzey, Jr., has influenced generations of architecture students and acousticians to visualize the sounds that become the acoustic signature of a performance or rehearsal space. His research and practice has shaped spaces we hear and listen in. Through the shaping of space, music and speech are enhanced and optimized by the addition of each curve, angle, planes, that become the unique character of the space. The sounds created on stages of theaters, concert halls, worship platforms, or lecterns are reflected, directed, and propagated to enhance the experience of the listener. Striving to achieve the perfect impulse response, we have learned from Bertram Kinzey that the acoustician's work is not unlike the work of a sculptor, an interior designer, or an architect. Each surface and each texture on the surface has a purpose and a meaning to its existence.

10:40–11:00 Break

11:00

3aAAa5. Bert Kinzey, teacher, and collaborator in design. William W. Brame (606 NE First St., Gainesville, FL 32601, b.brame@brameheck.com)

As both his student and his client, I have known Bert Kinzey since 1971. In 1973 I started my career in Gainesville, and recently retired as the senior partner of the 106 year old firm known as Brame Heck Architects, Inc. From my earliest days up until 2004 when Bert moved away to Virginia, I had the pleasure of retaining Bert to serve as our architectural acoustics consultant on a wide variety of projects. These projects included music rooms for educational facilities, conference centers, worship centers, and multi-purpose facilities that required excellent acoustics regardless of what was taking place in the space. Ever the teacher, he would explain his recommendations and solutions to each situation in such a way that I (and our clients) understood both the acoustical theory and the resulting physical manifestations that were being incorporated into the facility. We would routinely seek Bert's expertise for assuring that our new facilities were properly designed to isolate mechanical or environmental sounds. And, if a client had an existing facility that needed to remedy such noises, he would perform field testing and explore various options to solve those problems. Bert is a true gentleman and master of his profession.

11:20

3aAAa6. Studying with Professor Bertram Y. Kinzey, Jr., in the University of Florida, School of Architecture Environmental Technologies Option and its impact on my professional career. Bruno E. Ramos (BEA Architects Inc., 3075 NW South River Dr., Miami, FL 33142, ber@beai.com)

Professor Kinzey was instrumental in the careers of many professionals. He challenged students daily, pushed us to think "outside the box" and considered the integration of sustainable design principles in buildings long before LEED ever existed. I recall him saying "the old tests are in the library, you may go and review them... as it's my job to test your knowledge in different ways." He encouraged his students to gain the knowledge and experience to derive the answers. The scale models we built to measure and study acoustics and lighting within spaces also influenced my professional career. We applied those same concepts in marine architecture using a "Kinzey type" solution by building a scale model to determine the requirements and force needed to haul a large barge out of shallow water. Lastly, I am personally grateful to Professor Kinzey for stimulating the pursuit of excellence in his students which helped me to pass the A.R.E. and LEED licensing, at a very young age, in my first attempt. Professor Kinzey has had a lifetime of significant work in Acoustics and has influenced thousands of students in a profound manner.

11:40

3aAAa7. An appreciation of the educational achievements of Bert Kinzey. Edward G. Clautice (Global Buildings, Jacobs Eng., 1100 North Glebe Rd., Arlington, VA 22201, edward.clautice@jacobs.com)

The aim of this paper is to describe the achievements of Bertram Kinzey in the field of acoustics within architectural education. Bert Kinzey's ability to bridge engineering and architecture is of particular note to the personal experience of the author. With Bert's personal efforts that the University of Florida was able to begin accepting students into the program who did not have a traditional undergraduate degree in architecture. As the first student to matriculate into the Graduate School of Architecture at the University of Florida from this category the author owes a great deal of his personal career to Bert Kinzey. Bert lead the Environmental Technologies option at the School, which brought the art of music, the art of architecture and the engineering of applied technology into a combination which has enriched the profession(*). The study of concert hall architecture and acoustics has benefited greatly from Bert's leadership where acoustical modeling and auralization can stand beside traditional visual techniques in the field of architectural modeling.

12:00

3aAAa8. Spreading acoustics to architecture programs. Sang Bum Park (School of Architecture and Eng. Technol., Florida A and M Univ., 1938 South Martin Luther King Jr. Blvd., Tallahassee, FL 32307, sang.park@famu.edu)

Environmental Technology is one of the important core courses in an architecture program. It provides architectural students with basic physical principles in thermodynamics, acoustics, lighting, and indoor air quality and architectural design factors that affect those environmental qualities. This course is the only channel for the architectural students to be exposed to acoustical education and research, unless there are master's degree programs associated with acoustics at the school. Professor Bertram Kinzey, Jr., was the first one who planned and started the environmental technology and architectural acoustics program in the School of Architecture at UF. Since then, UF has been contributing as a producer of acoustical educators and researchers. The author taught discussion sessions of Environmental Technology as a graduate teaching assistant between 2008 and 2012 at UF, while he conducted acoustical research for his doctoral degree. He is now spreading the acoustical education to the School of Architecture and Engineering Technology at FAMU where previously there was no faculty or acoustical researchers with acoustical backgrounds through courses such as Environmental Systems in Architecture and Architectural Acoustics. He also provides the design studios with acoustical workshops and helps thesis students whose design priority lies in environmentally sustainable buildings.

Session 3aAAb

Architectural Acoustics: Room Acoustics Design for Improved Behavior, Comfort, and Performance II

Nicola Prodi, Cochair
Dept. of Engineering, University of Ferrara, via Saragat 1, Ferrara 44122, Italy

Kenneth P. Roy, Cochair
Building Products Technology Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603

Invited Papers

9:20

3aAAb1. S&N-S Light: An anthropic noise control device to reduce the noise level in densely occupied spaces encouraging personal control of voice. Sonja Di Blasio, Giulia Calosso, Giuseppina E. Puglisi, Giuseppe Vannelli (Dept. of Energy, Politecnico di Torino, Corso DC degli Abruzzi, 24, Torino 10129, Italy, sonja.diblasio@polito.it), Simone Corbellini (Dept. of Electronics and Telecommunications, Politecnico di Torino, Turin, Italy), Louena Shtrepi, Marco C. Masoero (Dept. of Energy, Politecnico di Torino, Torino, Italy), and Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Turin, Italy)

Recently, in many fields related to environmental quality, such as thermal and visual quality, the tendency is to customize the comfort according to the user's needs. A tailored comfort zone is planned in public spaces, in which occupants can set their own comfort level with passive or active systems. In this context, the reduction of noise due to anthropic sources can be seen as a priority. In densely occupied spaces, such as classrooms, workplaces, restaurants, and outdoor spaces, the noise due to users chatting has a detrimental effect upon performance, health, and environmental quality. This study reports applications of S&N-S Light, Speech & Noise Stop-Light, a patented smart phonometric device with a warning light activation of exceeding of predetermined anthropic sound levels limits, which encourages personal voice control through visual feedback. The light activation, with green, yellow, and red color, is based on an adaptive algorithm that accounts for pre-defined statistical noise levels; therefore, accidental noise levels can be filtered. The device has been used in classrooms, restaurants, and urban squares. Results indicate a decrease of noise levels with S&N-S Light, especially when the occupants received training about the device benefits.

9:40

3aAAb2. Designing triangular diffusers for architectural acoustics. Trevor J. Cox (Acoust. Res. Ctr., The Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, t.j.cox@salford.ac.uk) and Peter D'Antonio (Chesapeake Acoust. Res. Inst. LLC, Bowie, MD)

Pyramids and wedges can be used to change how sound is reflected in concert halls and other performance spaces. Simple geometric acoustic models can explain the reflection behavior when the wavelength of sound is small compared to the dimensions of the faces. Depending on the angle between adjacent surfaces, considerable dispersion, moderate diffusion, or specular reflection can result. There will be a bandwidth where geometric models are inaccurate because diffraction will be significant. Consequently, this study uses 2D Boundary Element Methods (BEMs) to overcome the wavelength limit. It is assumed that the understanding from 2D triangles can be generalized to 3D surfaces such as pyramids. A numerical optimization process is used to design arrays of triangles, examining the effect of depth, asymmetry, and periodicity. The performance for shallow and deep surfaces will be presented for different incident sound fields.

Contributed Papers

10:00

3aAAb3. The research of ceramic as sound absorption material in underground space. Hui Li and Xiang Yan (Acoust. Lab of Architecture School, Tsinghua Univ., Main Bldg., Rm. 104, Beijing, Beijing 100084, China, lihuisylvia@aliyun.com)

Beijing-Zhangjiakou High Speed Rail will be completed at the end of 2019. Badaling Station, the only underground station in this line, is the deepest high speed rail station in China with a depth of 102 m. The platform and transforming hall are both narrow spaces, which means acoustic treatment is necessary for the sake of speech articulation. Taking fireproofing, waterproofing, long lasting, culture and visual effect into consideration, ceramic is the most and only appropriate material being used in the underground station. This article will introduce the whole process of design and scale model test for the ceramic Helmholtz resonant cavity.

10:20–10:40 Break

10:40

3aAAb4. Enhancement of bass frequency absorption in fabric-based absorbers. Jonas Schira (Sales Manager Acoust., Gerriets GmbH, Im Kirchenhürstle 5-7, Umkirch 79224, Germany, jschira@gerriets.com)

Variable Acoustics has become an important topic in the acoustic design of multi-purpose venues but also in classical concert halls and opera houses. Varying the reverberation time in the middle and high frequencies can easily be

achieved by using fabric-based absorbers like curtains or roll banners. When using fabric-based absorbers, the proportionally low absorption capability in the bass frequencies below 400 Hz can be challenging when planning a multi-purpose facility. A full range variable system would provide a great tool to consultants and architects. Double layer roll banner systems with a highly absorbing fabric are mostly installed freely hanging in front of the wall. Research shows a significant enhancement of the bass frequency absorption if the fabric is installed in an enclosed housing. This paper will examine the fundamental problem of low bass absorption capability but will also show the measurement data and technical solution for the described problem.

11:00

3aAAb5. Noise insulation of a curtain wall for natural ventilation. Jean-Philippe Migneron, Andre Potvin, and Jean-Gabriel Migneron (School of Architecture, Univ. Laval, 1, cote de la Fabrique, PQ City, QC G1K 7P4, Canada, jean-philippe.migneron.1@ulaval.ca)

The growing interest in natural or hybrid ventilation systems brings a challenge for good integration of openings in building façades. In a noisy environment, there is a major limitation for the use of direct openings in common building envelopes. As a part of a research project dedicated to this problem, it is possible to evaluate the impact of a curtain wall that could be added to get a double skin façade. Experimental measurements made in laboratory conditions lead to the estimation of usual noise reduction and sound transmission class. The airflow at constant differential pressure was assessed as a function of the aperture and compared to sound insulation. Analyzing those parameters together give useful information for the design of passive ventilation with a significant airflow when acoustic performance is an important issue. This paper aims to detail performances of common curtain wall assemblies.

11:20

3aAAb6. Design and optimization of the sound diffusers using radial basis functions-based shapes and genetic algorithms. Ricardo Patraquim, Luis Godinho , and Paulo Amado Mendes (ISISE, Dept. Civil Eng., Univ. of Coimbra, DEC/FCTUC - Rua Luis Reis Santos, Coimbra 3030-788, Portugal, ricardo.patraquim@gmail.com)

Sound diffusers are a common technical solution used in the last four decades for conditioning performance rooms with greater acoustic requirements. A significant number of the acoustic diffusers commercially available are based on the phase grating diffusers or Schroeder-type diffusers. However, in some particular cases, the visual appearance of the acoustic conditioning of the room with QRDs is considered by architects to be unaesthetic or visually unattractive in modern spaces, and thus, other geometrical forms of the diffusive surfaces or elements need to be customized and explored. The optimization of the diffusers' design has been a topic of intense research in the last years. In this paper, the authors propose an alternative technique to define new shapes of sound diffusion configurations, based on the use of radial basis functions (RBF). In addition, to allow the definition of optimal surface shapes for a given frequency band, a genetic algorithm is used. The diffusion coefficient is computed within the optimization procedure using the Kirchoff integral equation. A set of application results are presented and discussed and experimental evaluation of the diffusers is performed in a simplified semi-anechoic room (to evaluate diffusivity) according to ISO 17497-2 in order to compare the numerical results.

11:40

3aAAb7. Rezonator effects of student cabinets used in classes. Filiz B. Kocyigit (Architecture, Atilim Univ., incek, Ankara 06560, Turkey, filizbk@gmail.com)

Achieving acoustic comfort in high school is of great importance for increasing the quality of education. Due to the high number of students in classrooms, canteens, cafeterias, and corridors, the voices of young girls and young men are intense. Apart from that, HVAC systems, lighting fixtures, announcement systems, and electronic devices increase the background sound level and students communicate with each other with a higher sound pressure level. The conditions of use of the training areas require hygiene and materials resistant to vandalism. This necessitates the use of hard and smooth material, which increases the RT of the interior spaces. Different methods are being sought for ensuring the value of RT required to increase D50, C80, and S/N level in student-student-teacher communication in classrooms. For this purpose, different schools were observed in different sizes, covered with different materials and interior materials, and RT, EDT, D50, and C80 measurements, Lmax, Lmin, and Leq measurements were made. The studies show that circular and rod-shaped openings with ventilation purpose in the student cupboards used in the space show swallowing characteristics at low frequencies and they are working as a rezonator. In this study, samples from different classrooms and factors affecting indoor sound quality were evaluated.

Session 3aAAc

Architectural Acoustics: Acoustic Regulations and Classification of New and Retrofitted Buildings II

Birgit Rasmussen, Cochair
SBi, Danish Building Research Institute, Aalborg University Copenhagen, A.C. Meyers Vænge 15, Copenhagen SV 2450, Denmark

Jorge Patricio, Cochair
LNEC, Av. do Brasil, 101, Lisbon 1700-066, Portugal

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

Contributed Papers

9:20

3aAAc1. Determining "reasonable" levels of sound insulation in domestic properties for use in building regulations. Richard G. Mackenzie, Nicola Robertson (RMP Acoust., Edinburgh Napier Univ., 42 Colinton Rd., Edinburgh, Scotland EH10 5BT, United Kingdom, ri.mackenzie@napier.ac.uk), and Sean Smith (Inst. for Sustainable Construction, Edinburgh Napier Univ., Edinburgh, United Kingdom)

The Scottish Building Regulations, similar to other countries regulations, provide standards for the protection of occupants health. Minimum sound insulation standards are provided to control noise passing through walls and floors from neighboring properties. In Scotland, the minimum standard should provide adequate protection for a "**reasonable**" person from "**normal**" living activities. This paper presents the findings of a research study undertaken by Edinburgh Napier University, to assess the level of sound insulation that test subjects would deem to be "**reasonable**" for a range of noise sources passing through a variety of construction types. The study undertaken within the Universities auralization suite assessed the responses involving over 100 participants subjected to common domestic noise sources passing through separating structures. The participants were given the ability to adjust the source noise level for each source type to determine the level of noise from their neighbor they would consider "**reasonable**" to tolerate. The participants determinations were correlated to the equivalent sound insulation for each source and construction assessed. The paper will present the setup in the test suite, and the results of the study will be presented in a range of acoustic parameters, $D_{nT,w}$, STC, R'_w and be assessed against a range of countries current standards.

9:40

3aAAc2. A strategy for sustainable acoustic classification scheme of dwellings. Miomir Mijic (School of Elec. Eng., Univ. of Belgrade, Bulevar Kralja Aleksandra 73, Belgrade 11000, Serbia, emijic@etf.rs), Aleksandar Milenkovic, Danica Boljevic (Acoust. Lab., Institut IMS, Belgrade, Serbia), and Dragana Sumarac Pavlovic (School of Elec. Eng., Univ. of Belgrade, Belgrade, Serbia)

The new legislation in Serbia for sound insulation in buildings, currently in preparation, should introduce for the first time a national implementation of dwellings' acoustic classification. There are some structural systems and building typology that are common in existing housing stock. Some of them are caused by seismic demands and some by architects' routines. All that introduced realistically accessible range of apparent sound reduction index values at different positions in existing buildings. To achieve a sustainable classification scheme in prepared new legislation the analysis of sound insulation in existing buildings were performed and implications of possible boundary values between different classes on the housing stock acoustic score were analyzed. Based on such approach, the suggestions concerned with the classification scheme and class limits are presented.

Invited Papers

10:00

3aAAc3. Acoustic regulations and classification of various types of buildings in the Nordic countries. Steindor Gudmundsson (Verkis Consulting Engineers Ltd., Ofanleiti 2, Reykjavik IS-103, Iceland, stgu@verkis.is)

It is relatively well known that in the Nordic Countries, there are national classification standards for dwellings, and one of the classes (class C) is referred to in the building regulations as minimum acoustic quality. In Norway, Iceland, and Sweden, there are also national acoustic classification standards for different other types of buildings, and in Norway and Iceland, class C is referred to as minimum acoustic quality in the building regulations for these buildings as well as for dwellings. In Sweden, the acoustic quality for these buildings is recommended, but not mandatory. In the paper, the different types of premises for work included in the standards are discussed with examples of some of the acoustic demands. The regulated buildings include schools, kindergardens, hospitals, and nursing institutions. Hotels and offices are also included, and also the minimum sound absorption and maximum noise levels in restaurants, cultural, and sports buildings and many other different premises for work. Sometimes it is decided not to use only the minimum demands, defined by class C, but to use the better quality defined by class B (or even class A).

10:20–10:40 Break

10:40

3aAAc4. Experience obtained during the ten years of the sound classification practical use for regulation building acoustics in Lithuania. Aleksandras Jagniatinskis, Boris Fiks, and Marius Mickaitis (Vilnius Gediminas Tech. Univ., Linkmenu 28, Vilnius 08217, Lithuania, akustika@vgtu.lt)

Advantages of application the sound classification schemes were foreseen in the two applications. In the legal regulation, the scheme expresses in the users friendly and easy understandable form protection against noise requirements for the buildings. The scheme is also a tool for designer to advice criteria of the suitable acoustic comfort in premises and also to label buildings according acoustic quality. Classification scheme implemented in Lithuania comprise five acoustic comfort classes—A, B, C, D, and E. Acoustical requirements expressed by the C sound class limit values correspond to the at least acceptable acoustic comfort level. The lowest (worst) sound class is E and comprise limit values corresponding to the acoustical comfort level in old buildings erected under sound insulation requirements existed before. By this reason, the step in limit values between different classes cannot be permanent and depends from the changes in acoustical demands during the time. For enforcement, legal requirements expressed through mandatory sound class C for new dwellings and E for renovated buildings from 2007 pre-completing testing become mandatory. In this approach, more important becomes a guideline for verification of compliance with an acoustic class.

11:00

3aAAc5. Revision of the Swedish sound classification scheme for premises. Krister Larsson (Bldg. Technology/Sound & Vibrations, RISE Res. Institutes of Sweden, Box 857, Boras SE-50115, Sweden, krister.larsson@sp.se)

The Swedish building code puts requirements on noise protection in new buildings. For residential buildings, minimum requirements are given directly in the building code, and demands for better acoustic quality are given in a sound classing scheme according to the Swedish Standard SS 25267, which has been revised in 2015. For premises, such as offices, schools, or hotels, the building code does not contain specific quantified minimum requirements on the acoustic properties, but refer instead to the sound classing standard SS 25268 for tabulated values. Sound class C corresponds to the minimum requirements for new buildings, and demands for better acoustic quality are given according to class B or class A. The sound classing standard for premises, SS 25268, is currently under revision, and during the last years, the needs for changes and updates have been collected in cooperation with stakeholders and experts. The paper presents the motivations for the revision and the status of the work. Ideas for major revisions are being discussed such as suggestions for improved room acoustic requirements in schools, as well as sound insulation and acoustic comfort in open plan offices.

11:20

3aAAc6. A pilot study on acoustic regulations for office buildings—Comparison between selected countries in Europe. Birgit Rasmussen (SBi, Danish Bldg. Res. Inst., Aalborg Univ. Copenhagen, A.C. Meyers Vænge 15, Copenhagen SV 2450, Denmark, bir@sbi.aau.dk)

Acoustic regulations or guidelines for office buildings are found in several countries in Europe. The main reason is to ensure satisfactory acoustic working conditions for the various tasks and activities taking place in the many different kinds of rooms in such buildings. Examples of room types are offices, meeting rooms, open-plan offices, corridors, reception areas, dining areas, all with different acoustic needs. Some countries specify a few acoustic limit values only, while others define several different criteria, guidelines only, or a combination of requirements and guidelines. As a pilot study, comparison between requirements in selected countries in Europe has been carried out. The findings show a diversity of limit values for acoustic requirements. The paper includes examples of requirements for reverberation time, airborne and impact sound insulation, noise from traffic and from service equipment. Examples of guidelines will also be presented. The discrepancies between countries are being discussed, and some priorities for adjusting acoustic regulations will be given. In addition to a set of regulations or guidelines, some countries have office buildings included in national acoustic classification standards with different acoustic quality levels. The paper will indicate examples of such classification criteria for comparison with acoustic regulations.

Contributed Paper

11:40

3aAAc7. Acoustic behavior of facades: Acoustic isolation versus air permeability. Diogo M. Ferreira (Engenharia Cvil, Faculdade de Ciências e Tecnologia da Universidade de Lisboa FCT-UNL, Rua do Casal, n°29 1°B, Cacém 2735-354, Portugal, diogom.fer88@gmail.com)

Today, the acoustic comfort of dwellings is revealed as a very important factor in the context of overall comfort of its inhabitants. For this comfort, it contributes very importantly the sound insulation that the facades of buildings can provide. Since the facades comprise of an opaque portion and a translucent portion, the latter will be more relevant to the sound insulation that the facade element provides, which is influenced by the window itself, as by the openings and air permeability associated with. In order to evaluate the influence of the possible differences of acoustic insulation on a residential façade over time, an experimental study was developed in LNEC. The experimental work was based on a set of acoustic and air permeability tests considering several opening areas, between 0,5 cm^2 and 250 cm^2, in a given test window. The results obtained allow to evaluate in which direction the sound insulation versus air permeability can parameterize the performance of the facades of buildings, and what is its relationship to the well-being of residents. For the 0.5 and 1 cm^2 areas of aperture, no acoustic or air permeability differences were found in relation to the reference values, assuming that this scenario does not cause any significant variation in a housing façade. For the last two areas studied (200 and 250 cm^2), in acoustic and air permeability terms, it is concluded that, for these areas, the scenario is similar to that of an open window due to the high loss of sound insulation and the low air permeability.

Session 3aAB

Animal Bioacoustics: Comparative Bioacoustics: Session in Honor of Robert Dooling I

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

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Chair's Introduction—9:15

Invited Papers

9:20

3aAB1. Perceptual perseverance in a passerine with permanent papillar impairment. Amanda Lauer (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu) and Robert Dooling (Psych., Univ. of Maryland, College Park, MD)

The Belgian Waterslager canary is unique for its loud, low-pitched song which is accompanied by a hereditary pathology involving missing and damaged hair cells in the basal end of the papilla. These birds were been bred for hundreds of years for loud, low-pitched song. Breeders likely selected for high frequency hearing loss. In spite hair cell regeneration, the papillae in these birds never approaches that of normal-hearing canaries. Auditory nerve and brainstem responses are also diminished, and auditory brainstem nuclei show reduced cell size. And these birds show a suite of psychoacoustic deficits consistent with impaired active processing as seen in humans with hearing loss. It is rather remarkable, then, that Belgian Waterslagers are able to learn, discriminate, and produce complex, species-specific sounds with such impaired frequency selectivity and phase processing. This feat, in the presence of severe peripheral auditory damage, underscores the importance of temporal information in the avian auditory perception and vocal learning. The obvious genetic basis of this pathology places the Belgian Waterslager canary in another unique position of being the only nonhuman organism which must navigate through vocal development and vocal learning in the face of an inherited developmental peripheral auditory pathology.

9:40

3aAB2. Cormorant audiograms under water and in air. Ole N. Larsen (Biology, Univ. of Southern Denmark, Campusvej 55, Odernse M 5230, Denmark, onl@biology.sdu.dk), Jakob Christensen-Dalsgaard, Alyssa Maxwell, Kirstin A. Hansen, and Magnus Wahlberg (Biology, Univ. of Southern Denmark, Odense M, Denmark)

Little is known about underwater hearing abilities of diving birds. To help fill this gap we measured audiograms of cormorants (*Phalacrocorax carbo*) using two different methods. Wild-caught cormorant fledglings were anesthetized and their auditory brainstem responses (ABR) to clicks and tone bursts were measured; first in an anechoic box in air and then in a large water-filled tank with their head and neck submerged. In addition, audiograms were obtained from two adult cormorants using the psychophysical method of constant stimuli both in air and underwater. The shape of audiograms obtained from in-air ABR-recordings was similar to that reported for birds of similar size. The highest sensitivity in air was found at about 2 kHz, while the most sensitive response was at about 1 kHz underwater. In general, the audiograms obtained from psychophysical measurements were similar to the ABR audiograms but had much higher sensitivity than the ABR audiograms. The results from both methods suggest that cormorants have rather poor in-air hearing compared to similar-sized birds. Their underwater hearing sensitivity, however, is higher than what would have been expected for purely air-adapted ears, and it is likely that cormorants use their underwater hearing abilities during foraging dives.

10:00

3aAB3. Studies of rhythmic synchronization in avian vocal learners using operant conditioning methods. Yoshimasa Seki (Aichi Univ., 1-1 Machihata-machi, Toyohashi 4418018, Japan, yoshimasa.seki@gmail.com) and Kazuo Okanoya (The Univ. of Tokyo, Meguro-ku, Japan)

Researchers have argued that non-human animals exhibit a sense of rhythm. Budgerigars and Bengalese finches were trained to peck a key iteratively in response to metronomic stimuli using an operant conditioning method. Peck timing in Budgerigars was distributed around the stimulus onset of the metronome, suggesting the birds synchronized their body movement to the rhythm of the metronome. However, peck timing in finches appeared to correspond to an estimated reaction time, suggesting that peck responses were a mere reaction to the metronome stimuli. Next, budgerigars were trained to peck two keys alternatively without any metronomic stimuli, so that pecking of the keys was self-paced. Metronomic sounds were created to match the intervals of the self-paced pecking. Additional metronomic stimuli were created to be 10% faster, 10% slower, and 20% slower than the original self-paced metronome stimuli for each bird.

3a TUE. AM

These sped up or slowed down metronomic sounds were played back in the background of the self-paced pecking task. In this experiment, rhythmic synchronization was not observed; however, one bird exhibited shorter pecking intervals for the faster metronome and longer intervals for the slower metronome, suggesting that the metronomic stimuli influenced peck timing in this bird in the absence of training.

10:20

3aAB4. Elaborate network of avian intracranial air-filled cavities and its potential role in hearing. Kenneth K. Jensen (Starkey Hearing Technologies, 8901 Rockville Pike, Bethesda, MD 20889, kenneth.kragh.jensen@gmail.com), Jakob Christensen-Dalsgaard (Inst. of Biology, Univ. of Southern Denmark, Odense M, Denmark), and Ole N. Larsen (Inst. of Biology, Univ. of Southern Denmark, Odernse M, Denmark)

Many avian species possess an intracranial air-filled passage, directly connecting the medial surfaces of the tympanic membranes, called the interaural canal. It is known to greatly improve directional hearing by passive acoustics in small animals where the external interaural delay is too minute to allow temporal neural coding. For long, the avian interaural canal was assumed to be a simple cylindrical cavity. Contrary to this, we discovered through CT scans and other techniques that many birds (e.g., zebra finches and pigeons) do in fact have a rather elaborate system of interconnected air-filled cavities throughout the entire skull. The cavities communicate directly or indirectly with the tympanic membranes. How does this network affect the directional hearing in birds? On one hand, it may simply be an adaptation to flight and play little or no role in hearing. On the other hand, theoretical considerations suggest that the directional response may be optimized through frequency dependent “tuning” of attenuation and phase shift through the interaural canal. In this talk, we will first present the anatomy, then present some preliminary directional responses from zebra finch ears, and finally discuss future directions and considerations for what may be the functional interaural canal in birds.

10:40

3aAB5. Psychophysical basis for finite-state song syntax in Bengalese finches. Kazuo Okanoya (Life Sci., The Univ. of Tokyo, 3-8-1 Komaba, Meguro-ku 153-8902, Japan, cokanoya@mail.ecc.u-tokyo.ac.jp)

Bengalese finches have been domesticated for 250 years in Japan from wild white-rumped munias originally imported from China. Bengalese finches were domesticated for parental behavior and white color morphs, but not for songs. Nevertheless, Bengalese finches sing complex songs: 2-5 song notes were chunked and chunks are organized into finite-state syntax. Song duration indicates physical fitness of the bird and song complexity stimulates female reproductive behavior. We examined sequential expertizes in Bengalese finches using behavioral procedures. In a click detection task, birds were trained to peck when they heard a short click embedded in a chunk or between chunks. Reaction times were longer in the former cases. In a flash-song interruption task, song termination occurred more often when the flash was given between chunks. These data are suggestive of the perceptual and motor reality of chunk structures. In a serial reaction time task, birds where trained to peck horizontally arranged keys in certain order. Male birds learned the task better than females, suggesting that song motor control capacity may be utilized other motor domains as well. However, abstract rule learning by auditory discrimination was not possible, suggesting that the sequential expertizes maybe limited on the motor domain.

11:00

3aAB6. Mouse psychoacoustics: Not just re-Dooling the bird psychoacoustics research. Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu)

The clever artist Willie Nelson once said “the early bird gets the worm but the second mouse gets the cheese.” While Willie was not likely referring to animal psychoacoustics, the quote easily applies to the historical trajectory of the field. For years, birds served as a primary model for human hearing. Many species were already domesticated. They were known to be vocal learners. The hearing abilities of numerous species of birds were similar to humans. Finally, birds could be quickly trained using operant conditioning procedures and positive reinforcement. Robert Dooling was a pioneer of these techniques and results from his laboratory were instrumental for birds being used as models for human auditory processing for decades. With the development of the mouse genome in 2002, however, many researchers turned their focus instead towards these small mammals as research models. Genetically engineered strains of mice mimicking human disorders could be easily developed and studied, but unfortunately, the basic hearing and communication abilities of mice were largely unknown, limiting their utility as models. Successful but numerous adaptations of Dooling’s behavioral methods have been recently made to measure auditory acuity in mice to measure the perception of both simple stimuli and complex vocalizations.

11:20

3aAB7. Vocal production, auditory perception, and signal active space in an open habitat specialist, the Grasshopper Sparrow. Bernard Lohr (Dept. of Biological Sci., Univ. of Maryland Baltimore County, 1000 Hilltop Circle, Baltimore, MD 21250, blohr@umbc.edu)

Grasshopper Sparrows are specialists in open grassland habitats and face acoustic challenges normally associated with that habitat type. They produce several types of calls and two distinct types of song, all of which are high-pitched for songbirds (6—10 kHz). The primary territorial song, also known as the “buzz” song, consists of 3 or 4 brief introductory notes followed by a high-pitched, rapidly modulated trill. The function of the secondary song type, or “warble” song, remains unknown, but data from autonomous recording units demonstrates a correlation with pairing status and breeding cycle timing. Operant discrimination tests with Grasshopper Sparrows show a broader audiogram and extended high frequency auditory limit when compared with other small songbirds, suggesting that these birds, and potentially related species as well, have evolved a wider spectral range of auditory sensitivity in this habitat type. Auditory detection and discrimination thresholds were used to explore the active space and communication distances of this species’ vocalizations using a habitat bioacoustics model incorporating its normal territorial behavior. Results suggest that songs can be detected up to three territories away from the singer, but that birds may have difficulty discriminating between different conspecific songs more than two territories away.

Contributed Papers

11:40

3aAB8. Saliency of temporal fine structure in zebra finch vocalizations. Nora H. Prior, Edward Smith, Gregory F. Ball, and Robert Dooling (Biology Psych., Univ. of Maryland, 4094 Campus Dr., College Park, MD 20742, nhprior@umd.edu)

Previous work has shown that birds. in general, and zebra finches. in particular, have remarkable sensitivity to temporal fine structure (TFS). While spectral, envelope, and TFS cues are present in vocalizations, TFS has been largely ignored since sonographic, and not time waveform, analyses have been the mainstay in bioacoustics. However, birds' impressive sensitivity to TFS raises the question of whether behaviorally relevant information is carried within the TFS. Indeed, zebra finches have at least 10 call types that both males and females use in sophisticated ways to coordinate activities. Furthermore, zebra finch vocalizations are typically composed of harmonic stacks, rich in TFS. Here, we isolated and described patterns in the TFS between and within individuals for different call types and constructed test stimuli from these patterns of TFS for psychoacoustic experiments. Demonstrating TFS sensitivity within natural stimuli would argue for the increasing salience of TFS for real-life communication in birds. [Work supported by a NIDCD T32 DC000046-16 to NHP.]

12:00

3aAB9. Micro-scale habitat use of humpback whales around Maui Nui, Hawaii. Anke Kügler (Hawaii Inst. of Marine Biology, Univ. of Hawaii Manoa, 2525 Correa Rd. HIG 132, Honolulu, HI 96822, akuegler@hawaii.edu) and Marc Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii Manoa, Kaneohe, HI)

Each winter, thousands of North Pacific humpback whales (*Megaptera novaeangliae*) migrate from their high latitude feeding grounds in Alaska to mate and calve in the shallow tropical waters around the Main Hawaiian Islands. Previous studies on humpback whales in Hawaii have focused on the whales' acoustic behavior and their general distribution within the islands, but little is known about small-scale habitat preferences. Off the island of Maui, anecdotal reports from commercial operators and researchers tell of clusters of whales within the breeding area. However, to our knowledge, no studies have been conducted to examine the phenomenon of micro-scale aggregations. A pilot study using passive acoustic monitoring with Ecological Acoustic Recorders (EARs) was conducted from January through early March 2016 at three sites off Maui, using male singers as a proxy for relative whale abundance. Root-mean-square sound pressure levels (SPLs) were calculated to compare low frequency acoustic energy (0-1 kHz) between the different sites. Preliminary results indicate that singers alternate between the two farthest sites. Further, different diel patterns in song activity where observed among the sites. These results suggest at least some degree of variable spatial and temporal habitat use and that further monitoring is warranted.

TUESDAY MORNING, 27 JUNE 2017 ROOM 310, 9:15 A.M. TO 12:20 P.M.

Session 3aAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Measurements of Sediment Transport and Near-Bottom Structures I

James Lynch, Cochair
Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole, MA 02543

Peter D. Thorne, Cochair
Marine Physics and Ocean Climate, National Oceanography Centre, National Oceanography Centre, Joesph Proudman Building, 6 Brownlow Street, Liverpool L3 5DA, United Kingdom

Chair's Introduction—9:15

Invited Papers

9:20

3aAO1. Perspectives of ongoing acoustic developments for measuring sediment dynamics. Peter D. Thorne (Joseph Proudman Bldg., National Oceanogr. Ctr., 6 Brownlow St., Liverpool, Merseyside L3 5DA, United Kingdom, pdt@noc.ac.uk) and David Hurther (Lab. of Geophysical and Industrial Flows (LEGI), CNRS UMR 5519, Grenoble, France)

Sediment entrainment, transport, and deposition over bedforms can be highly dynamic with strong spatially temporal variability. To probe the multi-scaled processes of sediment transport, there has been continuing developments of instrumentation to obtain high resolution measurements of near-bed sediment dynamics. Such observations are used for both the development and assessment of process based sediment transport modeling. Here, results are reported from studies on developing high resolution acoustic instruments, deployed to make co-located observations of bedforms, the near-bed and suspended concentration fields, and the horizontal and vertical

components of intra-wave and turbulent flows. To evaluate the instruments, a series of bottom boundary layer measurements were collected over sandy sediments under differing conditions. The acoustic instruments under examination consisted of a Bedform and Suspended Sediment Imager, BASSI, a three dimensional acoustic ripple profiler, 3D-ARP, and high resolution Acoustic Concentration and Velocity Profilers, HR-ACVP. The results obtained from the instruments are used to illustrate ongoing developments in acoustics and its expanding capability for studying the dynamics of near-bed sediment transport processes.

9:40

3aAO2. Acoustic measurements of sediment transport. From pulse-coherent Doppler to an autonomous bathymetry vessel. Peter Traykovski (Appl. Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, ptraykovski@whoi.edu)

In this talk, we present recent advances in sediment transport research using high frequency acoustic sensors. The topics covered will range from development and use of a multifrequency pulse-coherent Doppler system to measuring bathymetry with unmanned autonomous surface vessel (ASV) equipped with a state-of-the-art swath bathymetry sensor. The pulse coherent Doppler was able to measure velocity profiles through the wave boundary layer in a high concentration fluid mud layer off the coast of Louisiana. At the end of energetic wave events, as the forcing decreased, the acoustic measurements revealed the settling of the mud layer with transition from turbulent to laminar flow, and a four order of magnitude increase of the viscosity of the mud and water fluid. This increase was inferred from the vertical structure of the wave boundary layer and offers a unique in-situ non-invasive approach to measuring the viscosity of flows with complex rheology. The ASV measurements allow O(10 cm) resolution bathymetric measurements in energetic tidal environments, where navigation of manned vessels on accurate track lines suitable for repeat bathymetry measurements is difficult, and are providing new insights into the dynamics of tidal inlets.

10:00

3aAO3. Measuring sediments: Backscatter optics, laser diffraction, acoustics, and combined optics and acoustics. Yogesh C. Agrawal (Sequoia Sci., Inc., 2700 Richards Rd., Ste. 107, Bellevue, WA 98005, yogi.agrawal@sequoiasci.com)

Over the last 3-plus decades, point measurements of sediment concentrations in water have mostly been made using optical backscatter. Physics tells us that this signal correlates with particle area concentration so that while measuring volume concentrations, suspended load (sand) can be swamped by wash load (fines) and remain unseen. Over 2 decades ago, we introduced laser diffraction to the marine-aquatic environment, delivering concentration, size distribution (and settling velocity spectrum when equipped with a setline tube), with the characteristic of uniform sensitivity to a 200:1 range of grain sizes, excluding flocs which have variable fractal dimensions. Most recently, we introduced a high-frequency 8 MHz acoustic backscatter point-measurement system with the attraction of enhanced sensitivity to suspended load, *i.e.,* opposite of backscatter optics, a higher limit to sediment concentrations, and greater tolerance to fouling. Very recently, a small conceptual jump was made to combine backscatter optics and 8MHz backscatter acoustics into a sensor with *nearly* uniform sensitivity to sizes over a 1—500 micron size range. I will review each briefly and illustrate how well these methods work in nature. Finally, I will offer thoughts on the information content of multi-frequency acoustics.

10:20–10:40 Break

10:40

3aAO4. Relating acoustic and optical measurements to particle and floc concentrations in the bottom boundary layer. Christopher R. Sherwood (US Geological Survey, 384 Woods Hole Rd., Woods Hole, MA 02543, csherwood@usgs.gov)

Profiles of optical and acoustic properties were measured by moving instruments vertically in the bottom boundary layer, between the bottom and about 2 m above the sea floor, at a sandy inner shelf site 12 m deep. Profiles were performed every two hours for 36 days, spanning a range of wave and current conditions. Acoustic instruments on the profiling arm included a three-frequency acoustic backscatter profiler and a 1.5-MHz acoustic Doppler velocimeter. Optical instruments on the arm measured backscatter, attenuation, and absorption. Stationary instruments on the main tripod measured waves, currents, Reynolds stress, and vertical temperature/salinity gradients. The acoustic backscatter measurements were coupled with optical measurements and recent models for the backscattering and absorption by flocs to provide time series of particle-concentration and size profiles. Remarkable changes in particle sizes, concentrations, and inferred densities and settling velocities were observed. These observations demonstrate the value of the traditional Rouse profile assumptions often used in suspended-sediment transport calculation, but also reveal the rich temporal and spatial complexity of the near-bottom particle field.

11:00

3aAO5. Acoustic scattering from flocculating suspensions. Sarah Bass, Erin V. King, Andrew J. Manning (School of Biological and Marine Sci., Univ. of Plymouth, Drake Circus, Plymouth PL4 8AA, United Kingdom, sbass@plymouth.ac.uk), and Peter D. Thorne (National Oceanogr. Ctr., Liverpool, Merseyside, United Kingdom)

Acoustic backscatter from sediment suspensions in the marine environment has been limited in application by the lack of understanding of how sound scatters from flocculating particles. To support theoretical development of sound scattering, combined measurements of high frequency acoustic backscatter and particle population characteristics are presented over a range of flocculating suspensions, from natural *in-situ* muddy suspensions to laboratory controlled pure clay flocs. Field measurements of cohesive suspended sediments were made in the meso-tidal Tamar Estuary, Devon, UK over several tidal cycles during spring tides. Controlled laboratory experiments were conducted using oscillating grid turbulence to suspend kaolin, oxidized, and natural marine sediments sieved below 63 microns. In both field and laboratory cases distributions of floc size and settling velocity were acquired using video techniques (from which effective density was derived) and acoustic backscatter measured over frequencies of 1-4 MHz. Particle measurements were complimented by pumped suspension samples later analyzed for mass and organic content. Measured scattering properties from the various sediments are compared against each other and with predictions from a hybrid elastic-fluid sphere model. Initial results suggest a good agreement with the model for both *in-situ* field suspensions and oxidized natural sediments from the laboratory.

11:20

3aAO6. Doppler sonar measurements of bedload transport: Evaluations using computer models and lab trials. Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of NF, Chemistry-Phys. Bldg., St. John's, NF A1B 3X7, Canada, zedel@mun.ca), Alex E. Hay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), Greg Wilson (CEOAS, Oregon State Univ., Corvallis, OR), and Jenna Hare (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Studies have demonstrated that bottom velocity measurements from Doppler sonar systems are proportional to bedload transport rates. Given the complexity of acoustic backscatter and sidelobe interactions near the bottom boundary, the exact source of this signal is not obvious. We explore this measurement using a computer simulation and a series of laboratory trials. The system that we have developed for these studies, MFDop, is a multi-frequency (1.2-2.2 MHz), bistatic Doppler sonar that provides 3-component velocity profiles over a ~ 30 cm profile with ~5 mm resolution at a rate of 50 profiles/sec. Model simulations show that side-lobe contamination biases the velocity measurements above the bottom but that effect is reduced in the bedload layer itself. We report on tests of our system in field conditions at the St. Anthony Falls Laboratory (SAFL). The SAFL facility provides a 1.8 m deep, 2.75 m wide flume tank. In our trials, we used a 1 m depth flow of about 1 m/s over a mobile bed of sand with median grain size d50 = 0.4 mm. We find agreement between transport estimates determined using the MFDop, the sediment trap system integrated in the SAFL flume, and estimates based on bedform migrations.

11:40

3aAO7. Turbulent particle flux measurement with pulse-coherent Doppler sonar. Alex E. Hay and Kacie Conrad (Oceanogr., Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H4R2, Canada, alex.hay@dal.ca)

The accuracy of velocity estimates from pulse-coherent acoustic Doppler systems is related to the magnitude, R, of the ensemble-averaged complex pulse-pair correlation. The closer R is to unity, the more accurate the estimate. The accuracy of particle concentration estimates is also related to the value of R but, in contrast, high accuracy requires R to be less than unity, i.e., the amplitudes from individual pulses in an ensemble should be independent, in order to beat down Rayleigh statistics associated with configuration noise. Consequently, when estimating the turbulent component of the particle flux, i.e., the product of the particle concentration and velocity fluctuations, a tradeoff arises between these conflicting requirements for velocity and concentration accuracy. This tradeoff is investigated through a statistical model of sound scattering from particles embedded in idealized turbulence, and laboratory experiments in which particle velocities in turbulent flow are measured both acoustically, via pulse-coherent sonar, and optically, using particle imaging velocimetry.

Contributed Paper

12:00

3aAO8. Sediment transport studies—A brief retrospective look. James Lynch, Peter Traykovski, and Arthur Newhall (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole, MA 02543, jlynch@whoi.edu)

In this paper, we present a short retrospective look at the evolution of acoustical, optical, and related measurements of sediment transport, bottom stress, and bottom bedforms over the past quarter century, using our programs at the Woods Hole Oceanographic Institution as a "representative sample." Results from the High Energy Benthic Boundary Layer Experiment, the Sediment Transport on Shelves and Slopes experiment, and the Strata Formation on Shelves and Slopes experiment will be shown. Emphasis will be given to both technological and scientific advances. [Work supported by ONR.]

Session 3aBAa

Biomedical Acoustics: Advances in Shock Wave Lithotripsy I

Robin Cleveland, Cochair
Engineering Science, Inst. Biomedical Engineering, University of Oxford, Old Road Campus Research Building, Oxford OX3 7DQ, United Kingdom

Adam D. Maxwell, Cochair
University of Washington, 1013 NE 40th St., Seattle, WA 98105

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Pennsylvania State University, Penn State, 201E Applied Sciences Building, University Park, PA 16802

Chair's Introduction—9:15

Invited Papers

9:20

3aBAa1. Ultrasound, shock waves, and phonons. Rainer Pecha (RP Acoust. e.K., Friedhofstrasse 27, Leutenbach 71397, Germany, rainer.pecha@rp-acoustics.de)

On Dec. 10, 2016, the brilliant experimental physicist Prof. Wolfgang Eisenmenger passed away completely unexpectedly. His extensive work had influence on many different fields of acoustics. This presentation gives an overview on his remarkable and inspiring life and research.

9:40

3aBAa2. Stone formation. James C. Williams (Anatomy and Cell Biology, Indiana Univ. School of Medicine, 635 Barnhill Dr., MS-5055, Indianapolis, IN 46202, jwillia3@iupui.edu) and James E. Lingeman (Urology, Indiana Univ. School of Medicine, Indianapolis, IN)

How stones are retained within the kidney while small in size is still not fully understood. In this talk, we will show two examples of how stones are retained during early growth: One is growth on Randall's (interstitial) plaque, and the other is growth on mineral that has formed as a luminal plug in a terminal collecting duct. These two mechanisms of stone retention during early growth have distinctive morphologic features that can be seen by methods that show the microscopic structure of the stones. Stones growing on Randall's plaque display a mineralized region (composed of apatite) that is typically not large in size (less than 0.5 mm across) but which usually shows luminal spaces, which are signs of its origin in the connective tissue of the papilla. Stones growing on ductal plugs also show attachment to a piece of apatite, but the apatite regions are typically larger (often >1 mm long and >0.5 mm wide), and they are solid, without spaces running through them. Still other stone formers exhibit neither of these known mechanisms of stone retention, and we propose urinary stasis as a third possible way that stones are retained within the kidney. We propose that knowing the mechanisms of stone retention during early stone formation should allow for better treatment of stone diseases.

10:00

3aBAa3. New insights into the mechanisms and process of stone fragmentation in shock wave lithotripsy. Pei Zhong (Mech. Eng. and Mater. Sci., Duke Univ., 101 Sci. Dr., Durham, NC 27708, pzhong@duke.edu)

Stone fragmentation in shock wave lithotripsy (SWL) is the consequence of dynamic fatigue produced by stress waves and cavitation. Stress waves [longitudinal (or P), transverse (or S), and surface acoustic waves (SAW)] and associated tensile and shear stresses are the primary driving forces to create fracture, initially from pre-existing (or intrinsic) flaws inside the stone. In contrast, cavitation produces pitting on the stone surface, and consequently, introducing new (or extrinsic) flaws to weaken the stone structure during SWL. Stress waves and cavitation act synergistically to produce effective and successful stone comminution in SWL, with cavitation serving as catalysts to enhance the efficiency of stress waves-driven stone fracture. In this talk, contemporary understanding about the mechanisms and process of stone fragmentation in SWL will be summarized, using a heuristic model which incorporates two important lithotripter field parameters (i.e., pressure and dose) that can critically influence the treatment outcome. The effects of stone size, geometry, composition on the transient stress field produced inside the stone, and the potential role of SAW in crack initiation and propagation will be discussed to provide physical insights into improvements in lithotripsy device design and treatment strategy.

10:20

3aBAa4. Quantification of the shielding of kidney stones by bubble clouds during burst wave lithotripsy. Kazuki Maeda, Tim Colonius (California Inst. of Technol., Pasadena, CA), Wayne Kreider, Adam D. Maxwell, and Michael Bailey (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu)

Bubble clouds can shield kidney stones from insonification, and limit stone breakage during burst-wave lithotripsy (BWL), a recently proposed technique that uses focused ultrasound pulses with an amplitude of O(1-10) MPa and frequency of O(0.1) MHz. We use numerical simulations to quantify the magnitude of such shielding. In the simulations, we solve for the radial evolution of Lagrangian bubbles coupled to a compressible fluid using volume-averaging techniques. The resulting equations are discretized on an Eulerian grid. In particular, we quantify the reduction in acoustic energy flux incident on a rigid, plane wall that models the stone surface. We consider a burst wave with an amplitude of 6 MPa and a bubble cloud of diameter O(1) mm. The size distribution of nuclei, the number density of bubbles, and the distance of the cloud from the wall are varied, We show that a cloud containing O(10) bubbles with a diameter of O(10) um can reduce the total energy flux by more than 50%, largely independent of distribution of nuclei. Finally, we compare the simulation results with high-speed images and hydrophone measurements of bubble clouds from companion experiments. [Work supported by NIH 2P01-DK043881.]

10:40

3aBAa5. Acoustic removal of cavitation nuclei to enhance stone comminution in shockwave lithotripsy. Timothy L. Hall, Hedieh Alavi Tamaddoni (Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109, hallt@umich.edu), Alexander P. Duryea (Histosonics, Inc., Ann Arbor, MI), and William W. Roberts (Univ. of Michigan, Ann Arbor, MI)

Cavitation bubbles are formed by shockwaves as part of the normal SWL procedure and can assist in fragmentation when they collapse against a stone. However, following collapse, the bubble cloud leaves behind a large population of residual micron sized bubble "nuclei" that can interfere with subsequent shockwaves. This often manifests as more efficient fragmentation at lower shockwave repetition rates where there is sufficient time for nuclei to dissolve. This study will show how the application of low amplitude, unfocused ultrasound bursts can be used to stimulate bubbles to coalescence or dispersion from the shockwave path by the primary and secondary Bjerknes forces. Applying these bursts in between shockwaves reduces the bubble nuclei shielding effect allowing more energy to reach the stone and increasing efficacy. Our results will show this technique is effective at reducing the number of shocks required for stone comminution on a clinical electromagnetic lithotripter with a simple supplemental transducer to generate the low amplitude field.

Contributed Papers

11:00

3aBAa6. Passive acoustic mapping of cavitation during shock wave lithotripsy. Kya Shoar, Erasmia Lyka, Constantin Coussios, and Robin Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, kya.shoar@magd.ox.ac.uk)

Passive acoustic mapping (PAM) has previously been used to localize inertial cavitation during high intensity focused ultrasound. Here, this technique has been applied to shock wave lithotripsy (SWL), a non-invasive procedure whereby kidney stones are fragmented. Conventional diagnostic ultrasound probes were used to detect acoustic emissions during SWL. Signals consisted of reverberation sound from the incident shock wave followed, several hundred microseconds later, by emissions from cavitation collapses. Time-gating was used to isolate the cavitation signals, which were then processed using PAM to create spatial maps of the cavitation activity. Experiments in water indicated the spatial resolution was an ellipsoidal volume 5mm long by 1mm wide. Experiments were carried out in ex vivo pig kidneys and it was observed that cavitation was initiated in the region of the focus but moved laterally by up to 10mm and during treatment exhibited a general migration towards the source. These results suggest that PAM can be used as a tool to map the location of cavitation during SWL and has the potential to differentiate cavitation in tissue (that could contribute to injury) from cavitation near the stone which affects comminution. [Work supported in part by NIH through P01-DK43881.]

11:20

3aBAa7. Interaction between lithotripsy-induced surface acoustic waves and pre-existing cracks. Ying Zhang, Chen Yang, Defei Liao, and Pei Zhong (Mech. Eng. and Material Sci., Duke Univ., Hudson 229, Durham, NC 27708, zhang.ying@duke.edu)

The interaction between pre-existing cracks and surface acoustic waves (SAW) in lithotripsy is investigated. Surface acoustic waves are generated at a water-glass interface by an incident shock wave produced by the spark discharge of a nano pulse lithotripsy (NPL) device or an electromagnetic shock wave lithotripsy (SWL) source. Evidence of SAW, including leaky Rayleigh wave and Scholte wave, will be presented based on photoelastic imaging and numerical stimulations using COMSOL. A clear correlation between SAW and the location of the maximum tensile stress produced on the glass boundary has been identified, which can lead to ring-like fractures on a flat glass surface exposed to NPL-generated spherically divergent shock waves. To simulate cavitation-induced surface pitting in SWL, pre-existing crack will be introduced on the glass surface by microindentation using a Vickers or Knoop indenter. The interaction of SAW with the pre-existing cracks will be examined to characterize crack extension and branching as a function of their location and orientation to the incident shock wave.

11:40

3aBAa8. Improving environmental and stone factors toward a more realistic *in vitro* lithotripsy model. Justin Ahn (Urology, Univ. of Washington School of Medicine, Seattle, WA), Wayne Kreider, Christopher Hunter, Theresa Zwaschka, Michael Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Mathew Sorensen, Jonathan Harper, and Adam D. Maxwell (Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu)

To improve *in vitro* lithotripsy models, we investigated the effects of multiple experimental variables on stone fragmentation. We performed timed burst wave lithotripsy (BWL) and shock wave lithotripsy (SWL) exposures in a water tank with the following variable parameters: water gas content (60, 30, and 15% O_2), temperature (20 and 37°C), stone holder degree of enclosure (open wire basket, polyvinyl chloride (PVC) open-ended gel, and similar material anatomically accurate artificial kidney), and stone type (Begostone at 2 mixture ratios, calcite, calcium oxalate monohydrate (COM), and uric acid). At least 3 stones were treated for each condition, with fragmentation defined as percent stone mass <2 mm. Begostone (2:1 powder:water ratio) treated with BWL at 20°C vs. 37°C showed 75±13% vs. 62±6% breakage, respectively. Using the same stone type, gas concentrations of 60, 30, and 15% O_2 showed breakage of 23±4%, 55±8%, and 82±16%, respectively. More enclosed kidney phantoms showed decreasing lithotripsy efficacy of 94±11%, 64±21%, and 13±2% breakage in basket, PVC, and anatomic phantoms, respectively. 2:1 Begostone most

3a TUE. AM

closely mimicked COM stone breakage. SWL exposures produced similar trends. This work indicates the importance of controlling multiple variables during *in vitro* lithotripsy experiments. [Work supported by NIH P01 DK043881 and K01 DK104854.]

12:00

3aBAa9. Multiphase fluid-solid coupled analysis of shock-bubble-stone interaction in shock wave lithotripsy. Kevin G. Wang and Shunxiang Cao (Dept. of Aerosp. and Ocean Eng., Virginia Tech, Rm. 332, Randolph Hall, 460 Old Turner St., Blacksburg, VA 24060, kevinwgy@vt.edu)

A novel multiphase CFD-CSD coupled computational framework is applied to investigate the interaction of a kidney stone immersed in liquid with a lithotripsy shock wave (LSW) and a gas bubble near the stone. The main objective is to elucidate the effects of a bubble in the shock path to the elastic and fracture behaviors of the stone. The computational framework couples a finite volume two-phase computational fluid dynamics (CFD) solver with a finite element (FE) computational solid dynamics (CSD) solver. The stone surface is represented as a dynamic embedded boundary in the CFD solver. The evolution of the bubble surface is captured by solving the level set equation. The interface conditions are enforced through the construction and solution of local fluid-solid and two-fluid Riemann problems. The results of shock-bubble-stone simulations suggest that the dynamic response of a bubble to LSW varies dramatically depending on its initial size. Bubbles with an initial radius smaller than a threshold collapse within 1 μs after the passage of LSW; whereas larger bubbles do not. Moreover, this study suggests that a non-collapsing bubble imposes a negative effect on stone fracture while a collapsing bubble may promote fracture on the proximal surface of the stone.

TUESDAY MORNING, 27 JUNE 2017 — ROOM 312, 9:15 A.M. TO 12:20 P.M.

Session 3aBAb

Biomedical Acoustics: Partial Differential Equation Constrained and Heuristic Inverse Methods in Elastography I

Mahdi Bayat, Cochair
Biomedical and Physiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Wilkins Aquino, Cochair
Civil and Environmental Engineering, Duke University, Hudson Hall, Durham, NC 27708

Chair's Introduction—9:15

Invited Papers

9:20

3aBAb1. Variational formulations for elastic inversion from full-field wave data. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu)

Ultrasound elastography uses propagating P-waves to measure the deformation of soft elastic solids, including soft tissues. In some cases, the deformation of interest is quasistatic while in others it is a propagating S-wave. From the measured deformation field, it is of interest to infer the distribution of tissue rigidity and/or shear wave speed distribution within the imaged region. We review several variational formulations that have been proposed to solve this inverse problem, and their respective mathematical properties. The primary focus will be on variational formulations that lead to direct solution of inverse problem. These include the virtual fields method (and the related method of Romano, Shirron, and Bucaro), the direct error in constitutive equation formulation, and the adjoint-weighted variational formulation. We will consider consistency with the original boundary value problem, stability of the variational formulation, and required continuity of both data functions and the unknown modulus distributions. For completeness, we will briefly review three cost functions related to iterative solution of the inverse problem: energy error, error in the constitutive equation (ECE), and stabilized output least squares.

9:40

3aBAb2. A comparative study of full-wave inversion and local time-of-flight approaches in elastography. Wilkins Aquino (Civil and Environ. Eng., Duke Univ., Hudson Hall, Durham, NC 27708, wa20@duke.edu), Mahdi Bayat (Physiol. and Biomedical Eng., Mayo Colllege of Medicine, Rochester, MN), Olalekan A. Babaniyi (Civil and Environ. Eng., Duke Univ., Durham, NC), and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Colllege of Medicine, Rochester, MN)

Most of the current techniques for elastography rely on a pointwise measurement of the induced shear waves. A common assumption in these methods is plane wave propagation in an unbounded domain and local homogeneity of the medium. Because of these simplifying assumptions, complex wave patterns, boundary conditions, and interfaces can present significant challenges to these methods. On the other hand, general nonlinear optimization approaches with PDE constraints relax the underlying assumptions of planar waves and

unbounded domains and, hence, can handle very general conditions. However, this generality usually leads to a higher computationally expense as compared to time-of-flight methods. Therefore, the added computational expense and higher complexity of these optimization approaches needs to be justified in the context of elastography. In this work, we present a recent study that compares shear modulus reconstructions obtained with a PDE-constrained optimization approach with a conventional shear wave elastography method using synthetic data. We show that our optimization approach produced significantly more consistent and accurate results than a conventional SWE method. Moreover, we show that material distributions that lead to strong wave diffraction present serious challenges to time-of-flight local approaches, while they can be handled naturally with optimization-based approaches. [Acknowledgment: This research was supported by NIH Grant R01CA174723.]

10:00

3aBAb3. A machine learning alternative to model-based elastography. Cameron Hoerigd (BioEng., Univ. of Ilinois at Urbana-Champaign, 1270 Digital Comput. Lab., MC-278, Urbana, IL 61801, hoerig2@illinois.edu), Jamshid Ghaboussi (Civil and Environ. Eng., Univ. of Ilinois at Urbana-Champaign, Urbana, IL), and Michael F. Insana (BioEng., Univ. of Illnois at Urbana-Champaign, Urbana, IL)

Model-based elastography methods suffer severe limitations in imaging the complex mechanical behavior of real biological tissues. We adapted the Autoprogressive Method (AutoP) to address these limitations by approaching the inverse problem with machine learning tools. AutoP combines finite element analysis (FEA) and artificial neural networks (ANNs) with force and displacement measurements to develop soft-computational models of mechanical behavior. Unlike model-based elastography methods, only measurement data inform the material properties learned by the ANNs. Because this machine learning approach foregoes the initial model assumption, AutoP can be applied to anisotropic, time-varying, and nonlinear media common in biomedical imaging applications. We first implemented AutoP to characterize linear-elastic gelatin phantoms and ex vivo rabbit kidneys to demonstrate the potential for medical imaging. Those models required an estimate of the interior geometry via segmentation of the B-mode images. In our current work, the capabilities of AutoP are extended by developing a novel ANN architecture. Incorporating spatial information as part of the input to a pair of ANNs working in tandem allows the models to learn the spatially varying mechanical behavior, thus precluding the segmentation requirement. We will demonstrate this new approach to elasticity imaging by presenting elastograms generated by trained ANN material models.

10:20–10:40 Break

10:40

3aBAb4. Breast ultrasound elastography using inverse finite element elasticity reconstruction. Abbas Samani, Seyed R. Mousavi (Western Univ., Dept. of Medical Biophys., Medical Sci. Bldg., London, ON N6A 5C1, Canada, asamani@uwo.ca), Hassan Rivaz (Concordia Univ., Montreal, QC, Canada), Ali Sadeghi-Naini, and Gregory Czarnota (Sunnybrook Res. Inst., Toronto, ON, Canada)

Breast cancer is the most common cancer in women worldwide. Its early detection is paramount for its successful treatment outcome. Among imaging techniques developed for breast cancer diagnosis, elastography has shown good promise. In this presentation, a breast ultrasound elastography method will be described, and its application in breast cancer patients will be demonstrated. The method follows the quasi-static elastography approach where the breast is stimulated using regular ultrasound transducer. RF data are utilized within a dynamic programming minimization algorithm for tissue motion tracking, leading to 2D (axial + lateral) strain field. This field is processed within a novel inverse finite-element reconstruction framework to reconstruct the breast Young's modulus distribution. The framework uses Hooke's law to obtain the Young's modulus distribution. It is iterative where the stress distribution is updated using finite element method at the end of each reconstruction iteration. To ensure convergence, the Young's modulus was averaged within 5x5 finite element windows. A phantom study mimicking breast cancer was performed to validate the developed system which demonstrated high (93%) accuracy of Young's modulus reconstruction. The method was then applied to breast cancer patients where elastography images reconstructed using the proposed method showed its effectiveness in clinical setting. The only hardware required in the system is an ultrasound scanner. As such, it is a promising candidate for clinical cancer diagnosis.

11:00

3aBAb5. Shear modulus is a good surrogate for total tissue pressure: Preliminary studies with a xenograft pancreatic cancer tumor model. Marvin M. Doyley, Hexuan Wang (Elec. and Comput. Eng., Univ. of Rochester, 333 Hopeman Eng. Bldg., Rochester, NY 14627, m.doyley@rochester.edu), Michael Nieskoski, and Brian Pogue (Thayer School of Eng., Darmouth College, Hanover, NH)

Pancreatic ductal adenocarcinoma (PDA) is a common and lethal disease, with a 5-year survival rate of less than 6%. The absence of a functional vasculature and the build-up of dense stromal regions impede drug delivery that prevents the disease from being eradicated, even when surgery is combined with chemotherapy. We hypothesize that real-time measurement of total tissue pressure, as related to cancer cell growth and drug delivery, will enable translational research into patient-specific therapies. Since no imaging methods can measure tissue pressure *in vivo*, we investigated the feasibility of using elastography to provide a surrogate measure of total tissue pressure. Specifically, we performed studies on orthotopically and subcutaneously grown xenograft tumors (n = 20) to assess how the shear modulus of naturally occurring AsPc-1 pancreatic tumors varies with stromal density. The results of this investigation revealed that there is 6 kPa difference between the shear modulus of orthotopically and subcutaneously grown tumors. A strong correlation was observed between the shear modulus of the extracellular matrix and tissue pressure measured with a pressure probe. We also observed good correlation between shear modulus and collagen density. These preliminary results demonstrate that elastography is a good imaging surrogate biomarker of total tissue pressure.

11:20

3aBAb6. Elastic reconstruction of shear modulus and stress distribution for assessing abdominal aorta aneurysmal rupture risk. Doran Mix, Michael Stoner, and Michael S. Richards (Surgery, Univ. of Rochester Medical Ctr., Univ. of Rochester Med. Ctr., 601 Elmwood Ave., Rochester, NY 14642, michael.richards@rochester.edu)

The necessity of surgical intervention of abdominal aorta aneurysms is based on a risk-reduction paradigm primarily relying on trans-abdominal ultrasound (US) measurements of the maximum diameter of an AAA. However, AAA diameter is only a rough estimate of rupture potential and elastographic estimates of material property changes and stresses within aortic tissue may be a better predictor. This work presents an elastic imaging technique to match model predicted displacement fields to those measured using clinical US. A linear elastic finite-element model is used and is assumed to be undergoing a quasi-static, plane strain deformation. This technique uses a regularization scheme to incorporate geometric segmentation information, as a penalty or soft prior, to counter the inherent ill-posedness of the inverse problem. In addition, displacement fields are measured and accumulated over the entire cardiac cycle and incorporated simultaneously in the reconstruction technique to improve the signal to noise of the recovered modulus distribution. Model predicted strain fields and modulus distributions are used to predict the relative stress induced over the cardiac cycle. Results of validation studies comparing modulus and stress fields performed using finite-element simulations of 3D and time dependent geometries, tissue-mimicking phantom simulations, and initial clinical results will be presented.

11:40–12:20 Panel Discussion

TUESDAY MORNING, 27 JUNE 2017 — ROOM 205, 9:20 A.M. TO 11:40 A.M.

Session 3aEA

Engineering Acoustics and Physical Acoustics: Microelectromechanicalsystems (MEMS) Acoustic Sensors I

Vahid Naderyan, Cochair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Kheirollah Sepahvand, Cochair
Mechanical, Technical University of Munich, Boltzmannstraße 15, Garching bei Munich 85748, Germany

Robert D. White, Cochair
Mechanical Engineering, Tufts University, 200 College Ave., Medford, MA 02155

Invited Papers

9:20

3aEA1. Acoustic performance of MEMS microphones. Past, present, and future. Michael Pedersen (Novusonic Corp., P.O. Box 183, Ashton, MD 20861, info@novusonic.com)

An overview will be presented of the current state of commercial MEMS microphone technology with focus on acoustic performance metrics such as noise and sound pressure limits, bandwidth, and low frequency behavior. Important limitations for MEMS element and electronic performance will be discussed in the context of the various transducer technologies currently being pursued or offered. The performance of MEMS microphones, since their commercial introduction in the early 2000s, has been strongly driven by the mobile phone handset application, which continues to be the most important market segment by volume. Substantial improvements in performance have been realized since the inception, to meet opposing demands of better acoustic performance ***and*** lower power consumption/smaller size. A brief summary of the development trajectory and possible future directions will be given. With the advent of new important applications, such as home automation, and a general movement in design towards digital interfaces, other blends of microphone performance requirements are emerging. A discussion of such requirements and their possible impact on acoustic MEMS design will be provided.

9:40

3aEA2. When good mics go bad. Martin D. Ring (Consumer Electronics ProDC Eng., Bose Corp., the Mountain, M.S.; 271-E, Framingham, MA 01701-9168, ring@bose.com)

Billions of MEMs microphones are fabricated each year and the vast majority of them are used for voice pickup in Consumer Electronics (CE) devices where anomalous behavior or failure can cause little more than annoyance. Some millions of these microphones are destined for more sophisticated and/or critical applications where malfunction can lead to unpleasant and possibly dangerous situations

for both man and machine. This talk will discuss some of the environmental stimuli that our lives provide and our observations of MEMS microphone thermal variation, pressure overload characteristics, and lack of mechanical robustness to dust, gases, and fluids. In some cases, capacitive and piezoelectric sensors will be compared.

10:00

3aEA3. Microphone and microphone array characterization utilizing the plane wave tube method. Tung Shen Chew, Arthur Zhao, and Robert Littrell (Vesper, 77 Summer St., Boston, MA 02110, rlittrell@vespermems.com)

Microelectromechanical Systems (MEMS) Microphone arrays are becoming ubiquitous in consumer electronics. Large and expensive anechoic chambers are commonly used to characterize these arrays. Individual MEMS microphones, on the other hand, are typically tested using one of three methods: a free field calibration in an anechoic chamber, a pressure field calibration in a pressure chamber, or a pressure field calibration in a plane wave tube (PWT). In this work, we present a PWT system for testing a single microphone as well as a second PWT system for testing an array of four MEMS microphones. Both systems utilize a 3D printed portion of the tube that is designed to minimize reflections and standing waves while allowing the sound pressure to reach a calibrated instrumentation microphone and the MEMS microphone(s) under test. With these PWT systems, we characterize individual microphones up to 30 kHz and microphone arrays up to 3 kHz. Further, the array test system is used to measure the polar pattern of the microphone array at several frequencies and measure the impact of microphone mismatch on array performance. This PWT test methodology is a size and cost effective way to characterize MEMS microphone arrays.

10:20–10:40 Break

10:40

3aEA4. Thermal boundary layer limitations on the performance of micromachined microphones. Michael Kuntzman, Janice LoPresti, Yu Du, Wade Conklin, Dave Schafer, Sung Lee, and Peter Loeppert (Knowles Corp., 1151 Maplewood Dr., Itasca, IL 60143, michael.kuntzman@knowles.com)

The extent to which thermal boundary layer effects limit the performance of micromachined microphones is examined. A lumped element network model for a micromachined microphone is presented which includes a ladder network in parallel with the adiabatic back volume compliance to account for the transition of the enclosure from adiabatic to isothermal conditions when the thermal boundary layer becomes large compared to the enclosure dimensions. The thermal correction to the cavity impedance contains a resistive component which contributes thermal-acoustic noise to the system. The model results are compared to measurements taken from commercially available microphone units with various back volume sizes, and the simulated relative noise power contribution of each acoustic noise source is calculated. The impedance of the back volume, including the thermal correction factor, is compared to the adiabatic compliance and the impedance derived from thermoacoustic finite element analysis. It is shown that the noise due to the thermal component of the back volume impedance becomes significant in microphones with small back volumes and effectively sets an upper bound on the signal-to-noise ratio of a microphone of given package dimensions.

Contributed Papers

11:00

3aEA5. Calibration and characterization of MEMS microphones. Andrea Prato, Alessandro Schiavi (INRIM, Strada Delle Cacce 91, Torino 10135, Italy, a.prato@inrim.it), Irene Buraioli (Politecnico di Torino, Torino, Italy), Davide Lena (STMicroelectronics, Torino, Italy), and Danilo Demarchi (Politecnico di Torino, Torino, Italy)

In recent years, the increase in the number of smartphones led to a remarkable demand for low-cost microphones. This was met by the rapid development of MEMS (MicroElectroMechanical Systems) microphones, whose technology is becoming a promising perspective for future noise measurements based on new acoustic sensor networks. Nevertheless, current Standards do not provide proper calibration and test procedures for these microphones. In this work, calibration standard procedures have been adapted to characterize condenser MEMS microphones by comparison technique with laboratory standard microphones. Microphone parameters (sensitivity, frequency response, linearity, directivity, stability, and dynamic range) and changes of sensitivity with temperature (from -10 °C to + 50 °C) and humidity (from 25% to 90%) have been evaluated in a hemi-anechoic room and in an environmental chamber, respectively. These procedures open up the possibility to provide a robust and metrological characterization of MEMS microphones for noise measurements.

11:20

3aEA6. Characterization of the vibration response of miniature microphones by subtraction. Jonathan D. Walsh, Quang T. Su (Eng., Binghamton Univ., 4400 Vestal Parkway East, Binghamton, NY 13902, jwalsh3@binghamton.edu), and Daniel M. Warren (Knowles Electronics, Itasca, IL)

Presented is a test methodology for characterizing the vibration sensitivity of miniature microphones for hearing aids. A common method for obtaining the vibration sensitivity of a system is to use an electrodynamic shaker to deliver a calibrated vibration input and measure the corresponding output. When the system under test is a microphone, it is difficult to obtain the vibration response since the measured output will also be due to coherent sound created by the vibration delivery system. The method models the microphone as a system with two inputs, vibration and sound, with one electronic output. Using frequency-domain signal processing, this method extracts the vibration response from a shaker-driven signal by subtracting a synthesized acoustic response signal. When compared to vibration measurements under vacuum, the vibration responses from the two methods generally agree. The vibration response estimates produced using this algorithm are more accurate than vacuum chamber data due to the loading and stiffening effects caused by the presence of air, as would occur under standard operating conditions. This test method allows the rapid acquisition of microphone vibration responses by eliminating the need for a vacuum chamber, or carefully designed acoustic baffling.

Session 3aIDb

Interdisciplinary, and Education in Acoustics and Student Council: Graduate Programs in Acoustics Poster Session

Dominique A. Bouavichith, Cochair
New York University, 33 Washington Sq. W, 1115, New York, NY 10011

Brent O. Reichman, Cochair
Brigham Young University, 453 E 1980 N, #B, Provo, UT 84604

Michaela Warnecke, Cochair
Psychological and Brain Sciences, Johns Hopkins University, 3400 N Charles St, Dept Psychological & Brain Sciences, Baltimore, MD 21218

All posters will be on display from 9:20 a.m. to 12:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 9:20 a.m. to 10:50 a.m. and authors of even-numbered papers will be at their posters from 10:50 a.m. to 12:20 p.m.

Invited Papers

3aIDb1. University at Buffalo, SUNY: Variety of graduate programs in acoustics. Anastasiya Kobrina (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

University at Buffalo, SUNY has an outstanding reputation, due to its commitment to research and knowledgeable faculty. UB is known for its diversity in auditory research spanning from the psychophysics of hearing in humans and animals to the neurophysiological mechanisms of hearing. This unique variety leads to collaborations spanning departments and laboratories. The Cognitive Psychology doctoral program is aimed at training students for research-oriented careers. Graduate students are exposed to a variety of laboratories and courses in order to develop collaborations and build research skills. In addition, the Psychology Department holds regular colloquia on various general topics in psychology, and the Cognitive Psychology and Behavioral Neuroscience areas both hold weekly "brownbag" seminars, often related to topics in auditory processing and communication. Biological Sciences Department, the Communication Disorders Department, as well as the Center for Hearing and Deafness, make UB a perfect oasis of auditory research. Thus, the University at Buffalo is an ideal fit for training in hearing research and for facilitating collaborations.

3aIDb2. The Physical Acoustics Research Program at the University of Louisiana at Lafayette. Andi Petculescu (Univ. of Louisiana at Lafayette, 240 Hebrard Blvd., Lafayette, LA 70503, andi@louisiana.edu)

The Department of Physics at UL Lafayette has a strong history of acoustics research. Recently, the program has expanded considerably as a result of renewed interest in acoustics-related research. Tied into the current rethinking of the Department's multiple physics tracks, the Physical Acoustics Research Program offers ample opportunities for students—both undergraduate and graduate—to be involved in wide-spectrum research in acoustics. This program, unique in Louisiana, involves acoustic sensing of alien environments, atmospheric and underwater acoustics, ultrasonics in the solid state, ultrasonic materials characterization and structural health monitoring, seismology and geodynamics. In parallel to research projects, the faculty offer a variety of acoustics-related courses on topics such as Matlab- and Python-based computational acoustics, solid state acoustics, experimental techniques in acoustics, room acoustics, as well as machine learning techniques for wave propagation.

3aIDb3. The Graduate Program in Acoustics at the Pennsylvania State University. Victor Sparrow and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

The Graduate Program in Acoustics at Penn State is the only program in the U.S. offering the Ph.D. degree in acoustics as well as M.S. and M.Eng. degrees in acoustics. An interdisciplinary program with faculty from a variety of academic disciplines, the Graduate Program in Acoustics is administratively aligned with the College of Engineering and closely affiliates with the Applied Research Laboratory. Research areas include structural acoustics, nonlinear acoustics, architectural acoustics, signal processing, aeroacoustics, biomedical ultrasound, transducers, computational acoustics, noise and vibration control, psychoacoustics, and underwater acoustics. Course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, flow induced noise, spatial sound and 3D audio, marine bioacoustics, and acoustics of musical instruments. Penn State Acoustics graduates serve widely throughout military and government labs, academic institutions, consulting firms and industry. This poster describes faculty research areas, laboratory facilities, student demographics, successful graduates, and recent enrollment and employment trends.

3aIDb4. Graduate training opportunities in the hearing sciences at the University of Louisville. Pavel Zahorik, Jill E. Preminger, and Christian Stilp (Dept. of Otolaryngol. and Communicative Disord. & Dept. of Psychol. and Brain Sci., Univ. of Louisville, University of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

The University of Louisville currently offers two branches of training opportunities for students interested in pursuing graduate training in the hearing sciences: A Ph.D. degree in experimental psychology with concentration in hearing science, and a clinical doctorate in audiology (Au.D.). The Ph.D. degree program offers mentored research training in areas such as psychoacoustics, speech perception, spatial hearing, multisensory perception, and language development. The program guarantees students four years of funding (tuition plus stipend). The Au.D. program is a 4-year program designed to provide students with the academic and clinical background necessary to enter audiologic practice. Both programs are affiliated with the Heuser Hearing Institute, which, along with the University of Louisville, provides laboratory facilities and clinical populations for both research and training. An accelerated Au.D./Ph.D. training program that integrates key components of both programs for training of students interested in clinically based research is under development. Additional information is available at http://louisville.edu/medicine/degrees/audiology and http://louisville.edu/psychology/graduate/experimental.

3aIDb5. Graduate studies in acoustics at the University of Notre Dame. Thomas Corke (Univ. of Notre Dame, Notre Dame, IN) and Christopher Jasinski (Univ. of Notre Dame, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com)

The University of Notre Dame department of Aerospace and Mechanical Engineering is conducting cutting edge research in aeroacoustics, structural vibration, and wind turbine noise. Expanding facilities are housed at two buildings of the Hessert Laboratory for Aerospace Engineering and include two 25 kW wind turbines, a Mach 0.6 wind tunnel, and an anechoic wind tunnel. Several faculty members conduct research related to acoustics and multiple graduate level courses are offered in general acoustics and aeroacoustics. This poster presentation will give an overview of the current research activities, laboratory facilities, and graduate students and faculty involved at Notre Dame's Hessert Laboratory for Aerospace Engineering.

3aIDb6. Graduate research opportunities in acoustics at the University of Michigan, Ann Arbor. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

The University of Michigan is host to a wide array of acoustics research which encompasses many of the core Technical Committees of the ASA. Within the Department of Mechanical Engineering work is being done to advance the field of remote sensing and underwater acoustics, to better understand the physics of the cochlea in human hearing, and even to design safer football helmets. Within the University of Michigan Medical School, faculty and graduate students are constantly advancing techniques for diagnostic and therapeutic ultrasound procedures. In the Department of Naval Architecture and Marine Engineering, computational methods are being used to predict sound signatures and structural loading of complex sea vessels. Researchers in the Linguistics Department are using acoustic, perceptual, and articulatory methods to analyze human speech. And while these are only a sample of the projects taking place at Michigan, new opportunities for acoustics research and collaboration open up each semester. Combined with a rich course catalog, first-rate facilities, and prospects for publication, these opportunities prepare Michigan graduate students for careers in industry and academia alike. Go Blue!

3aIDb7. Graduate programs in Hearing and Speech Sciences at Vanderbilt University. G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

The Department of Hearing and Speech Sciences at Vanderbilt University is home to several graduate programs in the areas of Psychological and Physiological Acoustics and Speech Communication. Programs include the PhD in Audiology, Speech-Language Pathology, and Hearing or Speech Science, Doctor of Audiology (Au.D.), and Master's programs in Speech-Language Pathology and Education of the Deaf. The department is closely affiliated with Vanderbilt University's Graduate Program in Neurobiology. Several unique aspects of the research and training environment in the department provide exceptional opportunities for students interested in studying the basic science as well as clinical-translational aspects of auditory function and speech communication in complex environments. These include anechoic and reverberation chambers capable of multichannel presentation, the Dan Maddox Hearing Aid Laboratory, and close connections to active Audiology, Speech-Pathology, Voice, and Otolaryngology clinics. Students interested in the neuroscience of communication utilize laboratories for auditory and multisensory neurophysiology and neuroanatomy, human electrophysiology and neuroimaging housed within the department and at the neighboring Vanderbilt University Institute for Imaging Science. Finally, department faculty and students engage in numerous engineering and industrial collaborations, which benefit from our home within Vanderbilt University and setting in Music City, Nashville Tennessee.

3aIDb8. Graduate Education in Acoustics at The Catholic University of America. Joseph F. Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), Shane Guan (Office of Protected Resources Permits, Conservation and Education Div., National Marine Fisheries Service, Silver Spring, MD), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

The Catholic University of America (CUA) has a graduate program with a long history in acoustics dating back to the early 1930s. The acoustics program moved to the School of Engineering in the 1960s when there were strong needs in underwater acoustic studies to meet U.S. Naval applications. The end of the Cold War was concurrent with a decline in the CUA's acoustics education that persisted into the 1990s. However, renewed interests in acoustical engineering, acoustic metamaterial, and environmental acoustic research has revived the acoustics research and education programs at the CUA in recent years. Currently, a variety of graduate level acoustic courses are being offered in the CUA's Mechanical Engineering Department. Students can pursue a master's degree or Ph.D. degree with research in acoustics or vibrations. The courses in the program include a two-course sequence in fundamentals in acoustics, and more focused courses in ocean acoustics, atmospheric acoustics, acoustic metrology, marine bioacoustics, nonlinear vibration, acoustic imaging, and acoustic metamaterials. In addition, CUA offers masters and Ph.D. programs to students who are interested in the field of acoustic research.

3aIDb9. Graduate acoustics education in the Cockrell School of Engineering at The University of Texas at Austin. Michael R. Haberman (Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292), Neal A. Hall (Elec. and Comp. Eng., The Univ. of Texas at Austin, Austin, TX), Mark F. Hamilton (Mech. Eng., Univ. of Texas at Austin, Austin, TX), Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, Austin, TX, pswilson@mail.utexas.edu)

While graduate study in acoustics takes place in several colleges and schools at The University of Texas at Austin (UT Austin), including Communication, Fine Arts, Geosciences, and Natural Sciences, this poster focuses on the acoustics program in Engineering. The core of this program resides in the Departments of Mechanical Engineering (ME) and Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and eight graduate acoustics courses are cross-listed in ME and ECE. Instructors for these courses include staff at Applied Research Laboratories at UT Austin, where many of the graduate students have research assistantships. The undergraduate course, taught every fall, begins with basic physical acoustics and proceeds to draw examples from different areas of engineering acoustics. Three of the graduate courses are taught every year: a two-course sequence on physical acoustics, and a transducers course. The remaining five graduate acoustics courses, taught in alternate years, are on nonlinear acoustics, underwater acoustics, ultrasonics, architectural acoustics, and wave phenomena. An acoustics seminar is held most Fridays during the long semesters, averaging over ten per semester since 1984. The ME and ECE departments both offer Ph.D. degree qualifying exams in acoustics.

3aIDb10. Graduate Acoustics at the University of New Hampshire. Anthony P. Lyons (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824), Jennifer L. Miksis-Olds (Univ. of New Hampshire, Durham, NC), and Thomas C. Weber (Univ. of New Hampshire, Durham, NH, tom.weber@unh.edu)

The University of New Hampshire (UNH) offers several opportunities for graduate students interested in studying acoustics and its application. Faculty mentors who are expert in acoustic methods and technologies reside in a range of programs and departments that are largely focused on the use of acoustics in the marine environment, including biological science, earth science, mechanical engineering, natural resources and earth systems, ocean engineering, and oceanography. UNH faculty mentors who specialize in acoustics are active in the Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics technical committees. Recent studies by faculty and students focusing on fundamental acoustic problems, such as those that would cause a graduate student to be a regular attendee of meetings of the Acoustical Society of America, have come largely from mechanical engineering, ocean engineering, and the newly formed School of Marine Sciences and Ocean Engineering. Graduate students in these programs of study have the opportunity for formal classroom training in the fundamentals of acoustics, vibrations, and advanced topics in ocean acoustics as they pursue their graduate training.

3aIDb11. Graduate Acoustics at Brigham Young University. Scott D. Sommerfeldt, Jonathan Blotter, Timothy W. Leishman, Scott L. Thomson, Kent L. Gee, Brian E. Anderson, Tracianne B. Neilsen, and William Strong (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

Graduate studies in acoustics at Brigham Young University prepare students for jobs in industry, research, and academia by complementing in-depth coursework with publishable research. Graduate-level coursework provides students with a solid foundation in core acoustics principles and practices. A new acoustical measurements lab course provides a strong foundation in experimental techniques and writing technical memoranda. Labs across the curriculum include calibration, directivity, scattering, absorption, Doppler vibrometry, lumped-element mechanical systems, equivalent circuit modeling, arrays, filters, room acoustics measurements, active noise control, and near-field acoustical holography. Recent thesis and dissertation topics include active noise control, directivity of acoustic sources, room acoustics, radiation and directivity of musical instruments, energy-based acoustics, time reversal, nondestructive evaluation, flow-based acoustics, voice production, aeroacoustics, sound propagation modeling, nonlinear propagation, and high-amplitude noise analyses. Recently, the BYU acoustics program has added two faculty members and increased the number of graduate students, who are expected to develop their communication skills, present their research at professional meetings, and publish in peer-reviewed acoustics journals. Graduate students also often serve as peer mentors to undergraduate students on related projects and may participate in field experiments to gain additional experience.

3aIDb12. Distance Education Master of Engineering in Acoustics from Penn State. Daniel A. Russell and Victor Sparrow (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

The Graduate Program in Acoustics at Penn State provides online access to graduate level courses leading to the M.Eng. degree in Acoustics. Lectures are broadcast live via Adobe Connect to students scattered around the world, while archived recordings allow working students to access lectures at their convenience. Students earn the M.Eng. in Acoustics degree by completing 30 credits of coursework (six required courses and four electives) and writing a capstone paper. Since 1987, more than 135 distance education students have completed the M.Eng. degree in Acoustics. Many other students take individual courses as non-degree students. Courses offered online include elements of acoustics and vibration, elements of waves in fluids, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, aerodynamic noise, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, flow induced noise, spatial sound and 3D audio, marine bioacoustics, and acoustics of musical instruments. This poster describes the distance education experience leading to the M.Eng. degree in Acoustics from Penn State and showcases student demographics, capstone paper topics, enrollment statistics and trends, and the success of our graduates.

3aIDb13. Biomedical research at the image-guided ultrasound therapeutics laboratories. Christy K. Holland (Dept. of Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng. Program, Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu), T. Douglas Mast (Dept. of Biomedical, Chemical and Environ. Eng., Univ. of Cincinnati, Cincinnati, OH), Kevin J. Haworth (Dept. of Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), and Todd A. Abruzzo (Dept. of Radiology, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH)

The Image-guided Ultrasound Therapeutic Laboratories are located at the University of Cincinnati in the Heart, Lung, and Vascular Institute, a key component of efforts to align the UC College of Medicine and UC Health research, education, and clinical programs. These extramurally funded laboratories, directed by Prof. Christy Holland, are comprised of graduate and undergraduate students, post-doctoral fellows, principal investigators, and physician-scientists with backgrounds in physics, chemistry, and biomedical engineering, and clinical and scientific collaborators in fields including cardiology and neurosurgery. Prof. Holland's research focuses on biomedical ultrasound including sonothrombolysis, ultrasound-mediated drug and bioactive gas delivery, development of echogenic liposomes, early detection of cardiovascular diseases, and ultrasound-image guided tissue ablation. Prof. Todd Abruzzo, an experienced neurointerventional radiologist, directs preclinical porcine sonothrombolysis studies. The Biomedical Ultrasonics and Cavitation Laboratory, directed by Prof. Kevin Haworth, employs ultrasound-triggered phase-shift emulsions for image-guided treatment of cardiovascular disease, especially thrombotic disease. Imaging algorithms incorporate both passive and active cavitation detection. The Biomedical Acoustics Laboratory, directed by Prof. T. Douglas Mast, employs ultrasound for monitoring thermal therapy, ablation of cancer and vascular targets, transdermal drug delivery, and noninvasive measurement of tissue deformation.

3aIDb14. Acoustics research and graduate studies within the College of Engineering at the University of Nebraska—Lincoln. Lily M. Wang, Erica E. Ryherd (Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu), Joseph A. Turner (Univ. of Nebraska - Lincoln, Lincoln, NE), and Jinying Zhu (Univ. of Nebraska - Lincoln, Omaha, NE)

The University of Nebraska—Lincoln (UNL) offers opportunities to study and conduct research in acoustics within a number of our graduate engineering degree programs, including (1) Architectural Engineering (AE) within the Durham School of Architectural Engineering and Construction, (2) Civil Engineering (CIVE) and (3) Mechanical and Materials Engineering (MME). Dr. Lily Wang and Dr. Erica Ryherd (faculty in the Durham School, based at UNL's Scott Campus in Omaha) are active in architectural acoustics and noise. More information on the 'Nebraska Acoustics Group' within the Durham School may be found online at http://nebraskaacousticsgroup.org/. Dr. Jinying Zhu (faculty in CIVE, also based on UNL's Scott Campus in Omaha) focuses in structural acoustics, using ultrasonic waves for non-destructive evaluation of concrete structures and material. Dr. Joseph Turner (faculty in MME, based at UNL's City Campus in Lincoln) studies ultrasound propagation through complex media for quantitative characterization of materials/microstructure (http://quisp.unl.edu). UNL additionally hosts an active student chapter of the Acoustical Society of America, the first to be founded in 2004. The poster will describe the graduate-level acoustics courses and lab facilities at UNL, as well as the research interests and achievements of our faculty, graduates, and students.

3aIDb15. Acoustics-related graduate programs at the University of Minnesota. Kelly L. Whiteford (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu), Peggy B. Nelson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Hubert H. Lim (Biomedical Eng., Univ. of Minnesota, Minneapolis, MN), Mark Bee (Ecology, Evolution, and Behavior, Univ. of Minnesota, St. Paul, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

The University of Minnesota offers a wide variety of graduate programs related to acoustics, primarily in the areas of Speech Communication, Psychological and Physiological Acoustics, and Animal Bioacoustics. Degree programs include Psychology (Ph.D.), Speech-Language-Hearing Sciences (M.A., Au.D., and Ph.D.), Biomedical Engineering (M.S. and Ph.D.), Ecology, Evolution, and Behavior (Ph.D.), and Neuroscience (Ph.D.). Faculty across departments have a shared interest in understanding how the ear and brain work together to process sound and in developing new technologies and approaches for improving hearing disorders. The university offers a number of resources for pursuing research related to these topics. The Center for Applied and Translational Sensory Science (CATSS) provides opportunities for utilizing interdisciplinary collaborations to better understand sensory-related impairments, including hearing loss and low vision. Within CATSS is the Multi-Sensory Perception Lab, which houses shared equipment, including eye trackers, and electroencephalography. The Center for Magnetic Resonance Research houses several ultrahigh field magnets, while the Center for Neural Engineering and affiliated faculty labs also house multiple neuromodulation and neurorecording devices to interact with and monitor neural activity in humans and animals. Students and faculty gather monthly for the Acoustic Communication Seminar, where labs alternate presenting their research findings and identify new collaborative research directions.

3aIDb16. Acoustics education opportunities at UMass Dartmouth. David A. Brown (ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net), John R. Buck, Karen Payton, Paul J. Gendron, and Antonio Costa (ECE, Univ. of Massachusetts Dartmouth, North Dartmouth, MA)

The University of Massachusetts Dartmouth has a long tradition of research and course offerings in Acoustics and Signal Processing within the Department of Electrical and Computer Engineering dating back to the 1960s. The department has four full-time faculty with funded research programs in acoustics related areas as well as unique research/calibration facilities including a large underwater acoustic test facility and three fully autonomous underwater vehicles. UMass Dartmouth offers B.S., M.S., and Ph.D. degrees in Electrical Engineering with research opportunities and course offerings in fundamentals of acoustics, underwater acoustics, electro-acoustic transducers, medical ultrasonics, signal processing, speech processing, communications, and detection theory. The department works closely with the Center for Innovation and Entrepreneurship (CIE) and many local companies and government research centers. The poster will highlight course offerings and research opportunities. http://www.umassd.edu/engineering/ece/.

3aIDb17. Acoustics and ocean engineering at the University of Rhode Island. Lora J. Van Uffelen, Gopu R. Potty, and James H Miller (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu)

Acoustics is one of the primary areas of emphasis in the Ocean Engineering Department at the University of Rhode Island, one of the oldest Ocean Engineering programs in the United States. The program offers Bachelors, Masters (thesis and non-thesis options), and Ph.D. degrees in Ocean Engineering. These programs are based at Narragansett Bay, providing access to a living laboratory for student learning. Some key facilities of the program are the 100-foot-long wave tank, acoustics tank, and R/V Endeavor, a UNOLS oceanographic research vessel operated by the University of Rhode Island. At the graduate level, students are actively involved in research focused in areas such as acoustical oceanography, propagation modeling, geoacoustic inversion, marine mammal acoustics, ocean acoustic instrumentation, and transducers. An overview of classroom learning and ongoing research will be provided, along with information regarding the requirements of entry into the program.

3aIDb18. Graduate Studies in Acoustics at Northwestern University. Jennifer Cole, Matthew Goldrick, and Ann Bradlow (Dept. of Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, jennifer.cole1@northwestern.edu)

Northwestern University has a vibrant and highly interdisciplinary community of acousticians. Of the 13 ASA technical areas, 3 have strong representation at Northwestern: Speech Communication, Psychological and Physiological Acoustics, and Musical Acoustics. Sound-related work is conducted across a wide range of departments including Linguistics (in the Weinberg College of Arts and Sciences), Communication Sciences & Disorders, and Radio/Television/Film (both in the School of Communication), Electrical Engineering & Computer Science (in the McCormick School of Engineering), Music Theory & Cognition (in the Bienen School of Music), and Otolaryngology (in the Feinberg School of Medicine). In addition, *The Knowles Hearing Center* involves researchers and labs across the university dedicated to the prevention, diagnosis, and treatment of hearing disorders. Specific acoustics research topics across the university range from speech perception and production across the lifespan and across languages, dialect and socio-indexical properties of speech, sound design, machine perception of music and audio, musical communication, the impact of long-term musical experience on auditory encoding and representation, auditory perceptual learning, and the cellular, molecular, and genetic bases of hearing function. We invite you to visit our poster to learn more about the "sonic boom" at Northwestern University!

3aIDb19. Graduate research and education in architectural acoustics at Rensselaer Polytechnic Institute. Ning Xiang, Jonas Braasch, and Todd Brooks (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

The Graduate Program in Architectural Acoustics has been constantly advanced from its inception in 1998 with an ambitious mission of educating future experts and leaders in architectural acoustics, due to the rapid pace of change in the fields of architectural-, physical-, and psycho-acoustics. Recent years the program's pedagogy using "STEM" (science, technology, engineering, and mathematics) methods has been proven to be effective and productive, including intensive, integrative hands-on experimental components that integrate architectural acoustics theory and practice. The graduate program has recruited graduate students from a variety of disciplines including individuals with B.S., B.Arch. or B.A. degrees in Mathematics, Physics, Engineering, Architecture, Electronic Media, Sound Recording, Music and related fields. Graduate students under this pedagogy and research environment have been succeed in the rapidly changing field. RPI's Graduate Program in Architectural Acoustics has since graduated more than 120 graduates with both M.S. and Ph.D. degrees. Under the guidance of the faculty members they have also actively contributed to the program's research in architectural acoustics, communication acoustics, psychoacoustics, signal processing in acoustics, as well as our scientific exploration at the intersection of cutting edge research and traditional architecture/music culture. This paper illuminates the evolution and growth of the graduate program.

3aIDb20. Graduate studies in Acoustics and Noise Control in the School of Mechanical Engineering at Purdue University. Patricia Davies, J. S. Bolton, and Kai M. Li (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, daviesp@purdue.edu)

The acoustics community at Purdue University will be described with special emphasis on the graduate program in Mechanical Engineering. Around 30 Purdue faculty study aspects of acoustics and closely related disciplines and so there are many classes to choose from as graduate students structure their plans of study to complement their research activities and to broaden their understanding of acoustics. In Mechanical Engineering, the primary emphasis is on understanding noise generation, noise propagation, and the impact of noise on people, as well as development of noise control strategies, experimental techniques, and noise impact prediction tools. The noise control research is conducted at the Ray W. Herrick Laboratories, which houses several large acoustics chambers that are designed to facilitate testing of a wide array mechanical systems, reflecting the Laboratories' long history of industry-relevant research. Complementing the noise control research, Purdue has vibrations, dynamics, and electromechanical systems research programs and is home to a collaborative group of engineering and psychology professors who study human perception and its integration into engineering design. There are also very strong ties between ME acoustics faculty and faculty in Biomedical Engineering and Speech Language and Hearing Sciences.

Session 3aMU

Musical Acoustics: Session in Honor of Thomas D. Rossing

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

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

D. Murray Campbell, Cochair
School of Physics and Astronomy, University of Edinburgh, James Clerk Maxwell Building, Mayfield Road, Edinburgh EH9 3JZ, United Kingdom

Chair's Introduction—9:15

Invited Papers

9:20

3aMU1. Six decades of inspiration: Thomas D. Rossing, internationally renowned musical acoustician, writer, educator, and friend. D. Murray Campbell (Acoust. and Audio Group, Univ. of Edinburgh, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh EH9 3FD, United Kingdom, d.m.campbell@ed.ac.uk)

Tom Rossing occupies a very special place in the international community of researchers in musical acoustics. Generations of undergraduate and postgraduate students have been directed to *The Physics of Musical Instruments* by Fletcher and Rossing as the most helpful and authoritative textbook in this interdisciplinary field. Specialists in the study of percussion and stringed instruments have enjoyed and profited from the textbooks which he has written and edited on these topics, and from the many research papers which he has published. He has traveled widely, researching and teaching not only in the United States but also in Australia, England, France, Germany, the Netherlands, Scotland, South Korea, and Sweden. He has been a stalwart supporter of the series of International Symposia in Musical Acoustics, and has made an outstanding contribution to the promotion of education in acoustics. Musical acousticians worldwide owe Tom a great debt of gratitude for his collaboration, inspiration, and friendship.

9:40

3aMU2. Professional and personal interactions with Tom Rossing. William J. Strong (Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, strongw.byu@gmail.com)

This talk was motivated by time spent in Australia at the University of New England in Armidale, NSW, during the latter part of 1980. My wife and I and our three youngest sons spent five months there while I carried out research on the flute with Neville Fletcher and Ron Silk. As you might suspect from the title of the talk, Tom was also at the University of New England and our times there overlapped. Tom was solicitous of our sons which enriched their Australian experience. Tom's research was concerned with acoustical aspects of percussion instruments. Though discussions of our respective research projects was limited, our shared experience in Australia led to our continuing interaction during the ensuing years. A major part of the talk will consider interactions with Tom at other times and in other places.

10:00

3aMU3. "Mode studies in musical instruments," a journey with Tom. Uwe J. Hansen (Phys., Utah Valley Univ., 64 Heritage Dr., Terre Haute, Indiana 47803-2374, uwe.hansen@indstate.edu)

For me, that journey began when Tom agreed to have me work with him in 1984. While Tom was busy at the Minneapolis ASA meeting, I learned about holographic interferometry at Dick Peterson's laboratory at Bethel College, while mode mapping a guitar, for which Tom had designed a support rack which isolated the front and back plates, enabling us to record their principle resonances. Immediately following that experience, we went on a whirlwind tour, meeting with some of the "Greats" in musical acoustics. Starting with Carleen Hutchins, we then met with Norman Pickering in Southampton, did some modal analysis guitar studies at the Steinway laboratories with William Y. Strong, and later visited with Gaby Weinreich. We concluded the tour with a visit to Gila Eba,'s guitar building studio. In the course of studying two-tone Chinese bells, we used judicious mirror placements to observe the bell modes in three dimensions. Eventually, the wet plates were replaced by Karl Stetson's computer based device, enabling us to study many instruments, such as Caribbean Steel-Pans much more efficiently. All these experiences led to world-wide opportunities, and a life-long, cherished friendship.

10:20

3aMU4. Tom Rossing's influence on our understanding of the acoustics of wind instrument mechanisms. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

During his outstanding career, Tom Rossing studied the acoustics of every major class of musical instruments. During my first trip to the United States, to participate as a student in the ASA Columbus 1999 meeting, I was given the opportunity to measure my free-reed pipe in Tom Rossing's lab using his novel laser vibrometer. Like many others, he soon also became my role model for his broad knowledge, scientific depth, and his sheer pragmatism (that enabled him to write so many groundbreaking books). In particular, our generation greatly benefits from his gift for accurately describing the key fundamentals of complex acoustic phenomena based on a traditional understanding of physics and the use of groundbreaking measurement techniques such as laser vibrometry and TV holography. Tom Rossing never separated the hard work required to understand the underlying mechanisms of musical instruments from the cultural importance and delight of listening to and performing music. And so, he reminded everybody, right after 9/11 during the ISMA 2001 conference, that music brings people and cultures together and that our research is needed right now more than ever.

10:40–11:00 Break

11:00

3aMU5. Sound and shape of *pyeongyoung*, stone chime and *pyeonjong*, bell chime. Junehee Yoo (Phys. Education, Seoul National Univ., Kwanak-ro1, Kwanak-gu, Seoul 151-742, South Korea, yoo@snu.ac.kr)

My main research with Tom is about Korean traditional stone chimes, pyeongyoung and bell chimes, pyeonjong which have been allocated as a set of instruments in Korean traditional court music. The vibrational mode frequencies and the frequency ratios of the modes in modern pyeongyoung and pyeonjong have been studied. The modal shapes of stones and bells were mapped by TV holography, by scanning with an accelerometer and animated by STAR. The vibrational mode frequencies and mode shapes of ancient stone chimes are analyzed and their dependence on stone shapes had been studied by using finite element methods. The dependence of mode shapes and frequencies on vertex angle and base curvature suggests that the geometries used in late Chinese bianqing and Korean pyeongyeong may have been selected to give the best sound. Based on the research with Tom, I could extend the study to reconstruct the whangjongeum or scale in Korean traditional instruments. My Korean group have measured frequencies of historical 261 pyeongyeong stones and 236 pyeonjong bells mainly from the 14th to 19th centuries. The frequencies and the frequency ratios of the modes were analyzed by the era of building them.

11:20

3aMU6. Musical acoustics and science education: What I learned from Thomas D. Rossing. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

My professional journey into teaching and working on musical acoustics projects owes a great amount to the mentorship I was fortunate to receive from Thomas D. Rossing. I had the great privilege to meet Dr. Rossing as an undergraduate when my advisor arranged for us to make some measurements in Rossing's acoustics lab at Northern Illinois University. I expressed interest in attending NIU to study with him and was thrilled to realize that dream when I started (and completed) my Ph. D. program under Dr. Rossing's supervision. I frequently think about the many ways in which I learned what was important for student learning about science, especially acoustics, from my time as Dr. Rossing's teaching assistant. Throughout the years of knowing him, he has continuously been an inspiration to my way of thinking about how I teach and how I mentor my students. His pursuit of knowledge and love of learning have been instilled in my teaching and in the musical acoustics work that I do with students. In this talk I will highlight the various ways in which my career and life have been enriched by the example that Thomas Rossing has set.

11:40

3aMU7. The Rossing factor: How I benefited from being his student. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

From 1998 to 1991, I had the privilege of pursing a master's degree with Dr. Thomas D. Rossing, exploring a thesis on the nonlinear behavior of piano hammers at Northern Illinois University. I later earned a Ph.D. degree in Acoustics from Penn State, but my experience as Rossing's student was instrumental in helping me develop a diverse set of research skills and interests which have to be proved extremely beneficial throughout my academic career. Dr. Rossing involved me in several side projects (like optical holographic interferometry, experimental modal analysis, mode scanning), that were extracurricular to my thesis research but which led to several publications on a variety of acoustics topics, both while his student and later on my own as a physics and acoustics faculty member. The experimental skills acquired in Rossing's acoustics laboratory generated the inspiration for many of my own classroom demonstrations and research projects with my own students. This talk will summarize the many ways that Dr. Rossing's example as a teacher, researcher, and author have had a significant influence on my own academic career and success.

12:00–12:20 Panel Discussion

Session 3aNSa

Noise: Mechanical System Noise

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

Shiu-Keung Tang, Cochair
Department of Building Services Engineering, The Hong Kong Polytechnic University, Hong Kong, Hong Kong

Chair's Introduction—9:15

Invited Paper

9:20

3aNSa1. Results of a laboratory round robin for ASTM International Standard E477-13 for duct silencers. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

ASTM International initiated a laboratory round robin in November 2014 to support the development of a precision and bias statement for the recently revised standard E477-13. Two different silencer designs were constructed for testing by 5 participating laboratories. This paper will discuss the revised test method, the test results, and identify future modifications to the standard that should be looked at in the future.

Contributed Papers

9:40

3aNSa2. Machinery noise in a commercial building. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

A commercial building was designed for installation of offices, open public commercial sites, and exhibits, with multiple fixed small and large stores and wide corridors, to include small temporal exhibits or small booths for commercial or information purposes. Noise generated from all of the building services machinery, such as temperature control and hygiene systems, was an issue, so ventilating systems, water pumps, garbage disposal, and the like had to be installed in a convenient way in order to avoid intruding noise, either to workers or customers. The building is a large solid concrete structure where every effort had to be made in order to properly insulate all the machinery producing noise and vibrations in order to provide all the necessary services, and at the same time, adequate acoustics comfort.

10:00

3aNSa3. Rolling noise modeling in buildings. Fabien Chevillotte, François-Xavier Bécot, and Luc Jaouen (Matelys, 7 rue des Maraîchers, Bât B, VAULX-EN-VELIN 69120, France, fabien.chevillotte@matelys.com)

New buildings in urban areas are divided in commercial and living surfaces. This usage has revealed critical disturbances due to the noise of the trolleys delivering at time where the buildings are mostly occupied, e.g., early times in the morning. Rolling trolleys indeed generate low frequency vibrations (below 100 Hz) which propagate easily in the entire building structure and in upper storeys. This work presents an original model for rolling noise in buildings. The developed model is able to account for the ground surface roughness as well as the rolling wheel asperity profile. It also enables to consider the mechanical impedance of the ground including some possible flooring noise treatment. It is shown that the model is able to correctly reproduce the measured level of vibrations and measured noise levels. It is also proved to accurately predict the sensitivity to different types of rolling noise and floorings having various properties, based on a single layer or a multi-layer construction.

10:20

3aNSa4. The inlets and outlets of ducted silencer selection. Thomas Kaytt and Alana DeLoach (Vibro-Acoust. Consultants, 490 Post St., Ste. 1427, San Francisco, CA 94102, tom@va-consult.com)

The passive in-duct attenuator is an established, but often misused, staple of the modern HVAC design toolbox. All too often, a quick silencer selection without thought to the system at large can cause havoc with a noise sensitive application. How often have you seen a "typical" silencer scheduled throughout a project without regard to the specifics of the rooms being treated? This paper will review various types of silencers as well as location & selection methods to balance noise control needs against airflow capacities, air quality, and space restrictions. Case studies will be presented to illustrate some of the classic silencer design and installation errors.

10:40

3aNSa5. Noise control for a public transport interchange—A Hong Kong experience. Shiu-Keung Tang (Bldg. Services Eng. Dept., Hong Kong Polytechnic Univ., Hong Kong, Hong Kong, shiu-keung.tang@polyu.edu.hk)

Recently, a large public transport interchange (PTI) is proposed to be built near to a new public housing estate in a relatively remote area of Hong Kong. The aim is to provide convenient transport to the residents in the housing estate. In the design stage of the whole development, it is found that the idling buses inside the PTI are likely to create noise problem as the PTI is designed to be built in open air so as not to block the views of the

residents. There are also already nearby shopping malls and markets. In the view of the aesthetics, the PTI will be built using mainly glass panels and is designed to be ventilated by natural wind. There will be openings on the roofs of the PTI as heated air exhaust. In order to reduce the noise transmission towards the nearby housing blocks, the interior of the PTI and the openings are lined with sound absorption. A ray-tracing simulation was carried out. With the appropriate opening orientation sand amount of sound absorption, the noise levels at the building façades in concern can be kept under 48 dBA.

11:00

3aNSa6. Pool equipment mechanical noise impact. Walid Tikriti (Acoustonica, LLC, 33 Pond Ave., Ste. 201, Brookline, MA 02445, wtikriti@acoustonica.com)

The paper discusses noise impact from mechanical systems in residential buildings. The project presented discusses noise impact related to pool equipment mechanical systems. Sound readings were taken and recorded with different settings. Sound analysis and noise mitigation solutions will be discussed. Project located in Paradise Island, Nassau, Bahamas.

TUESDAY MORNING, 27 JUNE 2017 ROOM 202, 9:15 A.M. TO 12:00 NOON

Session 3aNSb

Noise, Education in Acoustics, ASA Committee on Standards, and Psychological and Physiological Acoustics: Using Acoustic Standards in Education

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Lawrence L. Feth, Cochair
Speech and Hearing Science, Ohio State University, 110 Pressey Hall, 1070 Carmack Road, Columbus, OH 43210

Massimo Garai, Cochair
DIN, University of Bologna, Viale Risorgimento 2, Bologna 40136, Italy

Chair's Introduction—9:15

Invited Papers

9:20

3aNSb1. Incorporating measurement standards in an advanced acoustics laboratory course. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

In an advanced acoustics laboratory course at Brigham Young University, students are introduced to ANSI measurement standards in the context of sound power. They are introduced to the anatomy of a typical acoustics standard and then plan and carry out sound power measurements of an electric leaf blower using both reverberation chamber and sound intensity methods. The students are required to write a technical memorandum describing (a) the blower's radiated sound power levels over an appropriate frequency range, as obtained with the two methods; (b) setup documentation and deviations from the standards' recommended practices; and (c) how any deviations might have contributed to discrepancies between the sound power levels obtained with the two methods. In this talk, a description of the experience from the faculty and student perspectives is given, along with plans for future improvements.

9:40

3aNSb2. German DIN 18041 Acoustic Quality in Rooms. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg 26121, Germany, nocke@akustikbuero-oldenburg.de)

DIN 18041 was first published in 1968 and at that time summarized a lot of knowledge in the field of room acoustics on the design of everyday life rooms such as class rooms, lecture halls, conference rooms, etc. DIN 18041 was first revised in 2004; a second revision was undertaken from October 2013 to mid 2015 to commit the room acoustic requirements for the implementation of the inclusion in the field of hearing and to take into account trends in the modern architecture. In addition to these technical and social aspects DIN 18041 of 2016 with the new title "Acoustic quality in rooms — requirements, recommendations and instructions for planning" gives clarifications and additions as well as deletions compared to the edition of 2004. These changes are presented and discussed. The revision of DIN 18041 provides clear and unambiguous guidelines described as requirements and recommendations for everyday rooms where the mutual listening and understanding but also finding of quietness is of significant importance. The use of standards in education for education facilities will be discussed.

10:00

3aNSb3. The role of acoustic standards in the improvement of educational spaces in Italy. Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Turin, Italy), Dario D'Orazio, and Massimo Garai (DIN, Univ. of Bologna, Viale Risorgimento 2, Bologna 40136, Italy, massimo.garai@unibo.it)

A good acoustics is one of the main requirements for indoor educational spaces, where information is transmitted from a teacher to students mainly by oral communication. Often obligatory regulations do not take into account this aspect as expected, but technical standards can provide a more informed and sound reference. In Italy, a new standard is under development at UNI (the national standardization body) to provide a technical framework for the design and use of teaching rooms in schools and a guideline has been released by AIA (Italian Acoustical Association) to provide a comprehensive, easy-to-read guide to authorities and stakeholders on the acoustical design of schools. The underlying principles and methods will be presented and discussed, as well as the way they are taught in University courses in order to raise awareness on these topics in future engineers and architects.

10:20–10:40 Break

10:40

3aNSb4. Incorporating standards into an instrumentation class. Lawrence L. Feth (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, feth.1@osu.edu)

First-year AuD students at Ohio State take *Acoustics and Instrumentation* in their first semester in the program. The course is offered simultaneously with courses in psychoacoustics and anatomy and physiology of the auditory system, as well as the first course in assessment covering behavioral testing. It is essentially a prerequisite for the two-course sequence on hearing aids, and later assessment courses covering middle ear measurements and ABR, as well as courses on cochlear implants and hearing conservation. Students enrolled in the course generally do not have a strong background in math and physics, but they are competent in algebra and trigonometry. One goal of the course is to develop a conceptual understanding of the basic acoustical and electronics principles underlying electroacoustic measurements. included with in those goals is the introduction to electroacoustic standards and to the role they play in the practice of audiology. To that end, the on-line instruction on standards offered by ANSI and the content of a recent two volume issue of *Seminars in Hearing* are incorporated into the "conventional" materials on acoustics and electronics. One unique feature of the course is the use of essay exams to test underlying concepts usually taught by "plug-and-chug" drills.

11:00

3aNSb5. Acoustic standards in audiology education. Peggy B. Nelson and Robert S. Schlauch (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu)

Standards are essential for the practice of audiology. Microphones, sound level meters, and all calibration equipment and procedures depend on effective standards. Equipment and methods for assessing hearing function, and for the fitting and evaluation of sensory aids for hearing loss all require the development and refinement of good standards. The involvement of audiologists in standards development is essential for maintaining high quality professional service. Graduate students in audiology are introduced to standards from both American National Standards Institute (ANSI) and International Standards Organization (ISO) during their graduate education at the University of Minnesota. Particular areas include calibration, audiometry, hearing aids, cochlear implants, and noise measurement and exposure. Methods for incorporating standards into graduate education will be discussed.

11:20

3aNSb6. Acoustic standards for education of hearing conservation and industrial hygiene professionals. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The practice of hearing conservation and industrial hygiene requires that workers' noise exposures and hearing be evaluated. The National Institute for Occupational Safety and Health has supported the development of new standards to assess hearing, hearing protector effectiveness and examined methods to best assess noise exposures to better understand the risk of workers developing noise induced hearing loss. This talk will consider noise measurement using both American National Standards Institute (ANSI) and International Standards Organization (ISO) acoustic standards. The paper will compare and contrast the hearing protector evaluation and rating standards for ANSI and ISO. The potential for using applications with mobile devices will be discussed.

Contributed Paper

11:40

3aNSb7. Uncertainty of ANSI/ASA S12.42 Hearing Protection Device Impulsive Measurements. Cameron J. Fackler, Elliott H. Berger, and Michael E. Stergar (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com)

ANSI/ASA S12.42 was extended in 2010 to include methods for measuring the performance of hearing protection devices (HPDs) in impulsive noise conditions. The standard specifies the instrumentation, methods, and data analysis required to measure impulse peak insertion loss (IPIL). IPIL is defined as the amount by which an HPD reduces the effective peak level of an impulsive sound. To characterize HPDs whose attenuation may be level-dependent, IPIL is measured at several impulse peak sound pressure levels, typically in the range of 130-170 dB. Factors contributing to the uncertainty of IPIL measurements include repeatability in the generation of test impulses, variability of the HPD samples under test and their repeated fitting to the test fixture, and spectral properties of the impulse source and the HPD's attenuation. To help inform future revisions to the S12.42 standard, we quantify IPIL measurement uncertainty for a variety of HPDs tested with two different impulsive sound sources. End users of HPDs should be educated about the uncertainty inherent in IPIL assessments of HPD performance.

Session 3aNSc

Noise, Physical Acoustics, ASA Committee on Standards, and Structural Acoustics and Vibration: Aircraft Noise and Measurements (Poster Session)

Victor Sparrow, Chair
Grad. Program in Acoustics, Penn State, 201 Applied Science Bldg., University Park, PA 16802

All posters will be on display from 10:20 a.m. to 11:40 a.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 10:20 a.m. to 11:00 a.m. and authors of even-numbered papers will be at their posters from 11:00 a.m. to 11:40 a.m.

Contributed Papers

3aNSc1. Potential changes in aircraft noise sound quality due to continuous descent approaches. Abhishek K. Sahai (Aircraft Noise and Climate Effects (ANCE), Delft Univ. of Technol., Kluyverweg 1, Delft, Zuid-Holland 2629HS, Netherlands, a.k.sahai@tudelft.nl), Miguel Yael Pereda Albarran (Inst. of Aerosp. Systems (ILR), RWTH Aachen Univ., Aachen, Northrhine-Westphalia, Germany), and Mirjam Snellen (Aircraft Noise and Climate Effects (ANCE), Delft Univ. of Technol., Delft, Netherlands)

This paper presents an analysis of how flying Continuous Descent Approaches (CDAs) can affect the quality of sounds that aircraft produce in airport vicinities. It is well known that CDAs present potential benefits in terms of community noise impact with reductions in excess of 5 dBA in peak noise levels. It is however unclear if these reductions in A-weighted level, which is a poor predictor of perceived annoyance, also correspond to an improvement in the quality of the aircraft sounds that reach the residents on the ground. A real comparison can only be made by comparing the sounds an aircraft produces while flying a CDA with a standard approach procedure. A short-range and a long-range aircraft are simulated to fly a standard approach procedure and a CDA with 3, 4, and 5 degree glideslope angle. The noise produced over both approach procedures is then auralized at representative ground locations, and the sounds are analyzed for changes in sound quality. Quantifying the changes in the aircraft sounds in terms of sound quality metrics provides much clearer information regarding how the sound the residents hear has changed, and if the CDAs actually result in an improved sound quality and hence lower annoyance.

3aNSc2. Reduced aerodynamic drag concepts for acoustic liners. Christopher Jasinski (Univ. of Notre Dame, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com) and Thomas Corke (Univ. of Notre Dame, Notre Dame, IN)

The objective of this paper is to describe the development of reduced aerodynamic drag concepts for acoustic liners in turbofan engine nacelles. Conventional acoustic liners help aircraft to achieve U.S. governmental noise regulations, however they are responsible for a measurable increase in the total aerodynamic drag of the aircraft. As regulations on commercial aircraft noise become more strict, additional surfaces may be covered with acoustic liner, heightening the need for the understanding and reduction of aerodynamic drag caused by liners. A linear force balance has been designed at the Mach 0.6 Wind Tunnel at the University of Notre Dame to evaluate the aerodynamic and acoustic characteristics of both conventional and 3D-printed liner samples. Through multi-parameter characterization, reduced drag concepts for future liners have been created. This paper will discuss the experimental work done to develop these reduced drag concepts, outline the future work to be done, and discuss the potential acoustic coupling effect on aerodynamic drag.

3aNSc3. Methodology for designing aircraft having optimal sound signatures. Abhishek K. Sahai, Tom van Hemelen, and Dick G. Simons (Aircraft Noise and Climate Effects (ANCE), Delft Univ. of Technol., Kluyverweg 1, Delft, Zuid-Holland 2629HS, Netherlands, a.k.sahai@tudelft.nl)

This paper presents a methodology with which aircraft designs can be modified such that they produce optimal sound signatures on the ground. With optimal sound it is implied in this case sounds that are perceived as less annoying by residents living near airport vicinities. A novel design and assessment chain has been developed which combines the aircraft design process with an auralization and sound quality assessment capability. It is demonstrated how different commercial aircraft can be designed, their sounds auralized at representative locations in airport vicinities and subsequently assessed for sound quality. As sound quality is closely related to the perceived annoyance, it is expected that designs with improved sound quality would also be perceived as less annoying. By providing a feedback to the design optimizer in terms of one of the sound quality metrics or a suitable combination thereof, the designs of aircraft can be altered to produce potentially less annoying sounds. The paper will focus on three current aircraft and will demonstrate the application of the novel design chain to auralize and alter their sounds toward improved sound quality. The presented methodology can also be extended to unconventional aircraft configurations and propulsion concepts, for optimizing future aircraft sounds.

3aNSc4. Practical consideration for measuring airborne ultrasound measurement. Isaac Harwell and Arno S. Bommer (CSTI Acoust., 16155 Park Row, Ste. 150, Houston, TX 77084, isaac@cstiacoustics.com)

When attempting to make meaningful measurements, airborne ultrasound in the 20 kHz to 100 kHz frequency range presents several significant challenges. These include source and sensor directivity, low signal-to-noise ratios, the inaudible nature of such sounds, and the lack of widespread literature on the subject. For each of these challenges, the practical impacts versus conventional acoustic measurements are identified and discussed. Solutions and suggestions are presented to allow reliable and repeatable measurements when presented with these challenges, including critical information to have prior to measurement, an assortment of techniques which may be employed when performing these measurements, and a brief overview of useful post-processing techniques. The concepts presented in this paper are illustrated using ultrasound measurements made in a vivarium.

Session 3aPA

Physical Acoustics and Noise: Eco-acoustics: Acoustic Applications for Green Technologies and Environmental Impact Measurements

JohnPaul R. Abbott, Cochair
Department of Physics and Astronomy, National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Dr., Room 1044, Oxford, MS 38677

André Fiebig, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Chair's Introduction—9:15

Invited Papers

9:20

3aPA1. Elastic energy harvesting: Materials and applications. Josh R. Gladden (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

As societies wean themselves off fossil fuel based energy sources, an "all of the above" approach will be required to satisfy expanding energy needs. This necessitates a renewed creativity from the scientific and engineering communities. Various ambient energy sources hold potential to supply power in particular applications. Solar and wind are of course well known examples, but vibrational, or elastic, energy should not be overlooked. A key component in harnessing any ambient energy source is the transduction mechanism to convert the energy from its original form into electrical. In this talk, I will explore available energy densities in a number of common scenarios, novel energy conversion materials, and discuss some niche applications.

9:40

3aPA2. Sound labels for classifying environment-friendly products—Progress and challenges. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

People are permanently exposed to noise caused by numerous products. The environmental awareness increases, the harmful effect of noise on humans is well acknowledged, and at the same time the desire for acoustic comfort rises. Thus, different emission-related labels are introduced as a reference for consumers informing about relevant product emissions. In fact, surveys show that product sound is already one of the top product features regarding the purchase decision. Thus, it seems that fewer emissions of everyday products are beneficial for all—consumers as well as manufacturers, and finally public health. However, most of the current sound labels use only simple noise level indicators and are only optional leading to an insignificant impact on purchase decisions and reducing the benefit of acoustically-friendly products on our acoustic environment. Moreover, several existing sound labels consider only simple minimum specifications and neglect sound quality related aspects at all. The paper provides an overview of the current status of sound labeling focusing on the European market. Moreover, case studies are presented illustrating massive differences in the sound quality of products on the market within the same product category. The limitations of current sound labeling approaches and initiatives are discussed in detail.

10:00

3aPA3. Environmentally friendly parametric alarm for alerting marine mammals of approaching vessels. Edmund R. Gerstein (Charles E. Schmidt College of Sci., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com) and Laura A. Gerstein (Leviathan Legacy Inc., Boca Raton, FL)

Marine mammals are vulnerable to boat, barge, and ship collisions. Although more commonly identified and reported in busy coastal areas, collisions are not restricted to shipping lanes or shallow water environments. A common denominator is that they all occur near the surface. Here the acoustical laws of reflection and propagation significantly limit the ability of marine mammals to hear and locate the sounds of approaching vessels. Acoustic measurements from controlled ship passages through vertical hydrophone arrays demonstrate the confluence of factors that poses auditory detection challenges for both whales and manatees. A highly directional, environmentally-friendly, low intensity underwater parametric alarm has been developed to mitigate these challenges & safely alert marine mammals of approaching vessels. The efficacy has been demonstrated with wild manatees. Ninety-five percent of manatees during alarm-on trials elicited avoidance reactions while only 5% of manatees during alarm-off trials elicited any change in behavior. The mean distance at which manatees reacted to boat approaches during alarm-on trials was also 20 m compared to only 6 m for alarm-off trials ($F= 218.4$ $df = 1$, $p< 0.01$). Counter-intuitive to speed reduction laws, slow vessels can be more difficult for marine mammals to detect and locate. The low intensity, directional alarm assures animals can detect & locate vessels at distances sufficient to avoid injury. [Funded by DOD Legacy Natural Resource Management Program, USFWS Permit MA063561-4.]

3a TUE. AM

10:20

3aPA4. Photoacoustic spectroscopy accurately measures atmospheric $^{12}CO_2$ and $^{13}CO_2$ concentrations and optical properties of carbonaceous aerosols. Keith A. Gillis (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov), Christopher Zangmeister, Zachary Reed, and James Radney (Chemical Sci. Div., National Inst. of Standards and Technol., Gaithersburg, MD)

Understanding of today's climate and predictions of future climate require accurate input data to model the energy balance between the sun's irradiance and Earth's atmosphere, oceans, land, and surface ice. An important driver of climate change is the absorption and scattering of sunlight by carbon-based aerosols (soot, smoke, etc) that have widely-varying, source-dependent, and history-dependent optical properties. We use a resonant photoacoustic spectrometer (PAS) to measure the optical absorption cross-section of various carbonaceous aerosols that we generate and characterize *in situ*. The photoacoustic signal is directly proportional to the energy absorbed by the particles. When combined with simultaneous measurements of the total extinction using cavity ring-down spectroscopy, we obtain the particles' wavelength-dependent albedo (fraction of incident light scattered). Another important driver of climate change is atmospheric carbon dioxide, a greenhouse gas. With the remarkable linearity, sensitivity, and resolution of our PAS resonator, we measure the individual concentrations of $^{12}CO_2$ and $^{13}CO_2$ in atmospheric samples to determine the isotopic ratio $^{13}C/^{12}C$, which gives a clue to its origin. A temperature-controlled portable PAS system continuously monitors the concentration of atmospheric $^{12}CO_2$ on a NIST rooftop.

10:40

3aPA5. Using correlated noise of turbulent flow in a long-wavelength acoustic flowmeter to measure the average flow speed. John Paul R. Abbott (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., Rm. 1044, Oxford, MS 38677, johnpaul.abbott@gmail.com), Keith A. Gillis, Michael Moldover (National Inst. of Standards and Technol., Gaithersburg, MD), and Lee Gorny (Mountain View, CA)

Current methods to measure the flow of gaseous emissions from coal-burning power plant smokestacks have uncertainties of 5% to 20%, which is unsuitable if a carbon pricing program is implemented. As part of its Greenhouse Gas and Climate Science Measurements Program, the Fluid Metrology Group at the National Institute of Standards and Technology (NIST) is investigating methods to reduce the uncertainty of flow measurements from smokestacks. In particular, NIST's scale model long-wavelength acoustic flowmeter (LWAF) uses low-frequency plane waves to measure the average axial flow speed, V, of turbulent fluid flow in a duct with an uncertainty of 1%. To apply this technology to smokestacks, we are investigating cross-correlations of low-frequency flow noise. The spectral density of the measured flow noise is consistent with fluctuations smaller than the duct diameter D for $f >> V/D$. The amplitude and width of the correlation peak for broadband flow noise is shown to be dependent on V, and a model of this dependence is forthcoming. Our current work, developing this model, includes filtering the broadband data to determine phase shifts, modeling the effects of the radiation impedance, and examining effects on the broadband flow noise spectra. The results of these investigations are presented.

11:00

3aPA6. Challenges in practical operational testing of a nuclear powered thermoacoustic sensor. James A. Smith (Nuclear Sci. and Technol., Idaho Natioanl Lab., P.O. Box 1625, Idaho Falls, ID 83415, James.Smith@INL.Gov), Steven L. Garrett (Penn State, State College, PA), Andrew Bascom (Penn State, Idaho Falls, PA), Brenden Heidrich (Nuclear Sci. and Technol., Idaho Natioanl Lab., Idaho Falls, ID), and Michael Heibel (Westinghouse, Church Hill, PA)

The world's first nuclear powered, wireless power and temperature sensor was demonstrated in Penn State's Breazeale Nuclear Reactor during the last week of September 2015. The sensor consisted of a thermoacoustic heat engine powered by nuclear fission designed to acoustically telemeter temperature and neutron flux information. The acoustic frequency of operation and the amplitude of the acoustic signal were proportional to the temperature and the reactor power respectively. The proof-of-concept tests were conducted in the research reactor twice daily over five days. Sensor performance was as expected with the exception that the amplitude of the acoustic signal diminished after each test. In this paper we will present our "wet sock" theory that a seal weld isolating the thermal insulation from the reactor coolant at the hot end of the thermoacoustic sensor failed early in the testing. This allowed water to be drawn in each time the thermoacoustic sensor cooled down reducing the efficiency of the insulation and therefore the sensor output. Thermometric data will be presented that supports our hypothesis. The result of this testing validates the resilience of the thermoacoustic sensor to adverse conditions present in the core of a nuclear reactor even when degraded. Lessons learned in the initial testing will be carried forward to the planned Advanced Test Reactor experiments in 2017.

Contributed Papers

11:20

3aPA7. Vibro-acoustic imaging development for microstructure characterization and metrology. James A. Smith, Eric D. Larsen, and Larry D. Zuck (Nuclear Sci. and Technol., Idaho Natioanl Lab., P.O. Box 1625, Idaho Falls, ID 83415, James.Smith@INL.Gov)

Vibro-acoustics (VA) is being developed into a portable scanning infrastructure for a novel material characterization technique focused on nuclear applications to characterize fuel, cladding materials, and structures at the Idaho National Laboratory (INL). The proposed VA technology is based on ultrasound and acoustic waves; however, it provides information beyond what is available from the traditional ultrasound techniques and can expand the knowledge on nuclear material characterization and microstructure evolution. VA is a three-dimensional (3D) imaging modality based on ultrasound-stimulated acoustic emission. VA uses the force caused by the beating of two frequencies to generate an acoustic emission signal. Due to absorption or reflection, the energy density in the object at an acoustic focal point changes to produce a "radiation force." This force locally vibrates the object, which results in an acoustic field that depends on the characteristics of the object at that point. This acoustic field is detected for every point in the object; the resulting data is used to make an image of the object's mechanical properties. This paper will focus on the development of a portable scanning system to image *in situ* the microstructure of materials used in nuclear applications.

11:40

3aPA8. Acoustic measurement infrastructure to enable the enhancement of nuclear reactor efficiency and safety. Vivek agarwal and James A. Smith (Nuclear Sci. and Technol., Idaho Natioanl Lab., P.O. Box 1625, Idaho Falls, ID 83415, James.Smith@INL.Gov)

Nuclear research reactors are used to test efficiency and safety of new technology, such as material or fuel samples under prototypic commercial conditions, operation of reactor prototypes, safety studies, assessment of neutron parameters, etc. Many experiments performed in research reactors require *in-situ* measurements to monitor the progress and performance of conducted tests. The core of any nuclear reactor presents a particularly harsh environment for sensors and instrumentations. The reactor core also imposes challenging constraints on signal transmission from inside the reactor core to outside of the reactor vessel. In this paper, an acoustic measurement infrastructure (AMI) installed at the Advanced Test Reactor (ATR), located at Idaho National Laboratory, is presented. The AMI consists of ATR in-pile structural components, coolant, acoustic receivers, primary coolant pumps, a data-acquisition system, and signal processing algorithms. Intrinsic and cyclic acoustic signals generated by the operation of the primary coolant pumps are collected using acoustic receivers and processed. The characteristics of the intrinsic signal can indicate the process state of the ATR during operation (i.e., real-time measurement). The innovation of AMI can be extended to collect information on other phenomena, such as fuel motion, individual fuel rod vibration, loose parts, thermal expansion, and flow blockage, that occur inside an operating nuclear reactor.

12:00

3aPA9. Optimization of acoustic absorption by green walls made of foliage and substrate. Emmanuel Attal, Nicolas Côté (ISEN, IEMN UMR CNRS 8520, Lille, France), Takafumi Shimizu (Daiwa House Industry, Central Res. Lab., Nara City, Japan), and Bertrand Dubus (ISEN, IEMN UMR CNRS 8520, 41 boulevard Vauban, Lille cedex 59046, France, bertrand.dubus@isen.fr)

Green walls may absorb sound and contribute to noise reduction in urban areas. Recent experiments demonstrate that a foliage layer placed above a substrate layer may lead to a significant increase of acoustic absorption coefficient in a broad frequency range. However, the physical origin of this improvement remains unclear. In this work, measurements are carried out in an impedance tube on foliage, substrate, and foliage/substrate samples using the three-microphone two-load method. Acoustic absorption coefficient and surface specific impedance are measured in rigid backing condition between 100 and 1000 Hz. Effective speed of sound and characteristic impedance are also experimentally determined for foliage and substrate. For foliage/substrate samples, a good agreement is obtained between measured acoustic absorption coefficients and calculated ones using the effective properties of foliage and substrate layer and matrix manipulations. Analysis of results reveals that absorption coefficient spectrum is mainly explained by the thickness resonances of the sample and by the impedance matching between air and substrate provided by the foliage.

TUESDAY MORNING, 27 JUNE 2017 ROOM 311, 9:15 A.M. TO 12:20 P.M.

Session 3aPPa

Psychological and Physiological Acoustics: Auditory Cognition and Scene Analysis in Complex Environments

Barbara Shinn-Cunningham, Cochair
Boston University, 677 Beacon St., Boston, MA

Janina Fels, Cochair
Institute of Technical Acoustics, RWTH Aachen University, Neustr. 50, Aachen 52074, Germany

Volker Hohmann, Cochair
Medical Physics, Universität Oldenburg, Postfach, Oldenburg 26111, Germany

Chair's Introduction—9:15

Invited Papers

9:20

3aPPa1. Examining auditory selective attention in reverberant environments. Josefa Oberem (Medical Acoust. Group, Inst. of Tech. Acoust., RWTH Aachen Univ., RWTH Aachen, Kopernikusstrasse 5, Aachen 52074, Germany, job@akustik.rwth-aachen.de), Julia Seibold, Iring Koch (Cognit. and Experimental Psych., Inst. for Psych., RWTH Aachen Univ., Aachen, Germany), and Janina Fels (Medical Acoust. Group, Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

Using a well-established binaural-listening paradigm, the ability to intentionally switch auditory selective attention was examined under anechoic, low reverberation (0.8 s) and high reverberation (1.75 s) conditions. Twenty-four young, normal-hearing subjects were tested in a within-subject design to analyze influences of the reverberation times. Spoken word pairs by two speakers were presented

simultaneously to subjects from two of eight azimuth positions. The stimuli were word pairs that consisted of a single number word (i.e., 1 to 9) followed by either the German direction "UP" or "DOWN." Guided by a visual cue prior to auditory stimulus onset indicating the position of the target speaker, subjects were asked to identify whether the target number was numerically smaller or greater than five and to categorize the direction of the second word. Switch costs (i.e., reaction time differences between a position switch of the target relative to a position repetition) were larger for the high reverberation condition. Furthermore, the error rates were highly dependent on reverberation times and interacted with the congruency effect (i.e., stimuli spoken by target and distractor may evoke the same answer (congruent) or different answers (incongruent)), indicating larger congruency effects in higher reverberation.

9:40

3aPPa2. Measuring auditory spatial perception in realistic environments. Virginia Best (Dept. Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu), Jorg M. Buchholz, and Tobias Weller (National Acoust. Labs., Macquarie University, NSW, Australia)

While much is known about how well listeners can locate single sound sources under ideal conditions, it remains unclear how this ability relates to the more complex task of spatially analyzing realistic acoustic environments. There are many challenges in measuring spatial perception in realistic environments, including generating simulations that offer a level of experimental control, dealing with the presence of energetic and informational masking, and designing meaningful behavioral tasks. In this work we explored a new method to measure spatial perception in one realistic environment. A large reverberant room was simulated using a loudspeaker array in an anechoic chamber. Within this room, 96 different "scenes" were generated, comprising 1-6 concurrent talkers seated at different tables. Listeners were presented with 45-sec samples of each scene, and were required to count, locate, and identify the gender of all talkers, using a touchscreen interface. Young listeners with normal hearing were able to reliably analyze scenes with up to four simultaneous talkers, while older listeners with hearing loss demonstrated errors even with two talkers at a time. Localization accuracy for detected talkers, as measured by this approach, was sensitive both to the complexity of the scene and to the listener's degree of hearing loss.

10:00

3aPPa3. What leads to audio-visual object formation and when is it helpful? Ross K. Maddox (Biomedical Eng. and Neurosci., Univ. of Rochester, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195, ross.maddox@rochester.edu)

An important aspect of parsing complicated auditory scenes is forming perceptual objects that correspond to individual sound sources (e.g., a friend speaking). Once formed, an object can be selected from the acoustic mixture and attended. Auditory cognition can be greatly enhanced by a concomitant visual stimulus (e.g., the speaking friend's mouth movements), but the mechanism or mechanisms underlying that benefit are not well understood. Previous studies have treated auditory and visual stimuli as separate sources of information and found that they are optimally combined. However, we have shown that auditory selective attention can be improved by a visual stimulus that offers no information at all. Our experiments suggest that these benefits are derived from the visual stimulus being bound to the target auditory stimulus into a single cross-modal object through temporal coherence of those stimuli's features. In this talk we will discuss a model of auditory-visual object formation that allows an uninformative but coherent visual stimulus to aid listening, along with data that suggest performance benefits may be dependent on the similarity of the target and the masker.

10:20

3aPPa4. Electrophysiological markers of auditory perceptual awareness and release from informational masking. Andrew R. Dykstra, Marnie E. Shaw, and Alexander Gutschalk (Dept. of Neurology, Ruprecht-Karls-Universität Heidelberg, Im Neuenheimer Feld 400, MEG Lab, Heidelberg 69120, Germany, andrew.dykstra@med.uni-heidelberg.de)

In complex acoustic environments, even suprathreshold sounds that are faithfully represented in the ascending auditory pathway sometimes go unperceived, a phenomenon termed informational masking. Little is known regarding the large-scale brain dynamics giving rise to conscious perception under informational masking, particularly outside auditory cortex. To examine this question, we combined simultaneous M/EEG with trial-by-trial perceptual reports and anatomically constrained distributed source estimates. Listeners reported the moment at which they became aware of spectrally isolated and otherwise suprathreshold tone streams rendered sometimes inaudible by random multitone masker "clouds." While all targets elicited early responses in auditory cortex, later auditory-cortex activity (peaking between 150 and 200 ms) was only observed for targets that were detected. A robust P3-like response with distributed sources was observed for the second detected target (immediately preceding listeners' reports), and was greatly diminished or absent for prior and subsequent targets. The results highlight late, distributed aspects of neuronal activity associated with task-related post-perceptual processing (i.e., task relevance), but argue against this activity underlying conscious perception, *per se*.

10:40–11:00 Break

11:00

3aPPa5. The role of self-motion processing in impairments of the spatial perception of auditory scenes. W. Owen Brimijoin, Graham Naylor, and Andrew McLaren (Inst. of Hearing Res. - Scottish Section, Medical Res. Council/Chief Scientist Office, MRC/CSO Fl. 3, New Lister Bldg., 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, owen.brimijoin@nottingham.ac.uk)

Natural auditory scenes typically include some motion. When this motion is the result of a moving *listener*, as is often the case, an accurate spatial percept requires that the listener be able to: 1) determine sound source location over time, 2) determine the extent and characteristics of his/her own motion, and 3) integrate these two pieces of information together. We assembled a battery of tests to evaluate these three criteria in normal, hearing-impaired, and balance-impaired listeners. The battery consists of a dynamic visual acuity test to estimate the listener's self-motion processing, a measure of the minimum audible movement angle to determine source-motion acuity, and an adapted version of a previously published dynamic front/back illusion to examine the ability to combine self and sound-source motion. We found that listeners with potential vestibular impairment were more likely to have larger differences between source-motion acuity and self-motion integration acuity. This finding and the observed inter-subject variability together underscore the sensitive nature

of the ongoing comparison between one's own motion and the motion of the acoustic world. [Work supported by the MRC (Grant No. U135097131) and the Chief Scientist Office of the Scottish Government.]

11:20

3aPPa6. Evaluation of scene analysis using real and simulated acoustic mixtures: Lessons learnt from the CHiME speech recognition challenges. Jon P. Barker (Comput. Sci., Univ. of Sheffield, Regent Court, 211 Portobello, Sheffield S1 4DP, United Kingdom, j.p.barker@sheffield.ac.uk)

Computational auditory scene analysis is increasingly presented in the literature as a set of auditory-inspired techniques for estimating "Ideal Binary Masks" (IBM), i.e., time-frequency domain segregations of the attended source and the acoustic background based on a local signal-to-noise ratio objective (Wang and Brown, 2006). This talk argues that although IBMs may be a useful stand-in when evaluating signal-processing systems, they can provide a misleading perspective when considering models of auditory cognition. First, there is no evidence that human cognition computes or requires an explicit binary mask representation (ideal or otherwise). Second, evaluation of an IBM requires artificially-mixed acoustic scenes in order to provide access to the ground truth mask. It is possible that systems that work well on artificially mixed acoustic scenes will fail to generalize to real data. The danger in predicting real performance from results obtained on artificial mixtures is seen in an analysis of systems submitted to the recent CHiME distant microphone speech recognition challenges which evaluates on both types of data (http://spandh.dcs.shef.ac.uk/chime). It is argued that rather than presume specific internal representations, auditory scene analysis systems can be best evaluated by direct comparison of human and machine percepts, e.g., in the case of a speech recognition task, comparison of human and machine transcriptions at a phonetic level.

11:40

3aPPa7. Modeling speech localization, identification, and word recognition in a multi-talker setting. Angela Josupeit, Joanna Luberadzka, and Volker Hohmann (Medizinische Physik and Cluster of Excellence Hearing4all, Univ. of Oldenburg, Medizinische Physik, Fakultaet VI, Universitaet Oldenburg, Oldenburg 26111, Germany, angela.josupeit@uni-oldenburg.de)

In many everyday situations, listeners are confronted with complex acoustic scenes. Despite the complexity of these scenes, they are able to follow and understand one particular talker. This contribution presents auditory models that aim to solve speech-related tasks in multi-talker settings. The main characteristics of the models are: (1) restriction to salient auditory features ("glimpses"); (2) usage of periodicity, periodic energy, and binaural features; and (3) template-based classification methods using clean speech models. Further classification approaches using state-space models will be discussed. The model performance is evaluated on the basis of human psychoacoustic data [e.g., Brungart and Simpson, Perception & Psychophysics, 2007, 69(1), 79-91; Schoenmaker and van de Par, Physiology, Psychoacoustics and Cognition in Normal and Impaired Hearing, 2016, 73-81]. The model results were mostly found to be similar to the subject results. This suggests that sparse glimpses of periodicity-related monaural and binaural auditory features provide sufficient information about a complex auditory scene involving multiple talkers. Furthermore, it can be concluded that the usage of clean speech models is sufficient to decode speech information from the glimpses derived from a complex scene, i.e., computationally complex models of sound source superposition are not required for decoding a speech stream.

12:00–12:20 Panel Discussion

TUESDAY MORNING, 27 JUNE 2017 ROOM 304, 10:00 A.M. TO 12:20 P.M.

Session 3aPPb

Psychological and Physiological Acoustics: Environmental Auditory Experience

Andrzej Miskiewicz, Chair
Department of Sound Engineering, Fryderyk Chopin University of Music, Okolnik 2, Warsaw 00-368, Poland

Contributed Papers

10:00

3aPPb1. Annoyance due to railway noise and vibrations: A comparison of two methods of collecting annoyance scores. Philéas Maigrot (Univ. Lyon, LGCB and LVA, INSA-Lyon/LVA, 25 bis Ave. Jean Capelle, Villeurbanne 69621, France, phileas.maigrot@insa-lyon.fr), Catherine Marquis-Favre (Univ. Lyon, ENTPE, Laboratoire Génie Civil et Bâtiment, Vaulx-en-Velin, France), and Etienne Parizet (Univ. Lyon, INSA-Lyon, Laboratoire Vibrations Acoustique, Villeurbanne, France)

An experiment has been conducted in order to determine if the method of collecting partial and overall annoyance scores—during separated sessions or during the same session—has an influence on the participants' answers. The experiment used controlled noise and vibration stimuli corresponding to a train pass-by, recorded inside a house in the vicinity of a railway track. 32 participants attended 4 sessions A, B, C, and D during each of which they were presented with 16 combinations of noise and vibrations. They had to evaluate partial annoyance due to noise in the presence of vibration (**session A**), partial annoyance due to vibrations in the presence of noise (**session B**) or overall annoyance (**session C**). Lastly, they were asked to rate partial and overall annoyances in a same session (**session D**). Results show that partial and overall annoyance scores, simultaneously collected during session D, were quite similar to the ones respectively collected

during dedicated sessions. Furthermore, this method is convenient as a reduced number of stimuli is presented to each participant.

10:20

3aPPb2. Annoyance response and improvement of Zwicker's psychoacoustic annoyance model aiming at tonal noises. Guoqing Di (Inst. of Environmental Pollution & Control Technol., Zhejiang Univ., Yuhangtang Rd. 866, Hangzhou, Zhejiang Province 310058, China, dgq@zju.edu.cn)

Zwicker's psychoacoustic annoyance model can be used to estimate the relative degree of noise annoyance. However, this model cannot be well applied to compare the annoyance degrees of tonal noises and atonal noises. In order to improve its estimation effect on tonal noises, 3 groups of noise samples were selected randomly, i.e., 27 low-frequency tonal noise samples induced by a 1000 kV transformer with A-weighted equivalent sound pressure levels ranging from 41.2 dBA to 73.0 dBA; 30 low-, mid-, or high-frequency tonal/atonal noise samples with loudness levels ranging from 60 phon to 80 phon; and 60 other noise samples with A-weighted equivalent sound pressure levels ranging from 40.7 dBA to 75.0 dBA. Laboratory listening tests were conducted on the above 3 sample groups respectively via an 11-point numerical scale. The Zwicker's psychoacoustic annoyance model was improved by taking tonality into account, and introducing the evaluation result of the first noise sample group (1000 kV transformer noise samples) to determine the coefficients in the model. The applicability of the improved model was examined by the evaluation results of the other two groups as well as the data in a previous research on annoyance of 220 kV/500 kV transformer noises. Results show that the improved model can estimate the relative annoyance degrees caused by various types of tonal / atonal noises much more accurately.

10:40–11:00 Break

11:00

3aPPb3. Clang, chitter, crunch: Perceptual organisation of onomatopoeia. Oliver C. Bones, William J. Davies, and Trevor J. Cox (Computing, Sci. and Eng., Univ. of Salford, 119 Newton Bldg., Acoust. Res. Ctr., Salford, Greater Manchester M5 4WT, United Kingdom, o.c.bones@salford.ac.uk)

A method has been developed that utilizes a sound-sorting and labeling procedure, with correspondence analysis of participant-generated descriptive terms, to elicit perceptual categories of sound. Unlike many other methods for identifying perceptual categories, this approach allows for the interpretation of participant categorization without the researcher prescribing descriptive terms. The work has allowed robust sound taxonomies to be created, which give insight into categorical auditory processing of everyday sounds by humans. Work on common audio search terms has highlighted that onomatopoeia are an important group that has been largely overlooked in quotidian sound studies. These are words for which the meaning of the word maps onto the sound of the utterance, and are an example of sound symbolism where there is a non-arbitrary link between the form and the meaning of word. Early analysis of the data suggests that people do draw on sound symbolism to carry out the categorization, but that in addition they also draw similarities between the inferred sound sources, such as organic versus non-organic.

11:20

3aPPb4. Perception of environmental sounds: Recognition-detection gaps. Andrzej Miskiewicz, Teresa Rosciszewska, and Jacek Majer (Dept. of Sound Eng., Fryderyk Chopin Univ. of Music, Okolnik 2, Warsaw 00-368, Poland, misk@chopin.edu.pl)

The study was conducted to assess the detection and recognition thresholds for 16 selected environmental sounds and determine the sound pressure level difference between those thresholds, called the recognition-detection gap (RDG). The sounds were recorded with a dummy head and played back through headphones. Recognition and detection thresholds were measured for two groups of listeners—musicians and non-musicians, in two conditions: in quiet and in the presence of multitalker masking noise added to the signal. The results demonstrate that RDG considerably varies, depending on the acoustic characteristics the sound, from about 2 to as much as 20 dB for sounds that are particularly difficult to recognize. The difficulty with which a listener recognizes the sounds, assessed on the basis of RDG, is also reflected by the steepness of psychometric functions for recognition. The present findings do not support a working hypothesis in which it was assumed that musical training results in an enhancement of the ability to recognize sounds at levels close to detection threshold, when not all their acoustic signatures are clearly audible, and manifests itself in smaller RDGs.

11:40

3aPPb5. The auditory experience of infants born prematurely. Brian B. Monson (Dept. of Pediatric Newborn Medicine, Brigham and Women's Hospital, Harvard Med. School, 75 Francis St., Boston, MA 02115, bmonson@research.bwh.harvard.edu)

The third trimester of gestation is a time of rapid auditory and brain development, and auditory experience during this period affects brain development. For example, newborns exhibit auditory learning and memory for acoustic stimuli frequently heard *in utero*. Preterm infants in the neonatal intensive care unit (NICU) during the third-trimester-equivalent period have a vastly different auditory experience than their fetal peers *in utero*, but the acoustic differences between these two environments are not well defined. The goal of quantifying these differences is to better understand and even predict their impact on auditory processing deficits exhibited by preterm infants later in childhood. Here I will present data describing the acoustic environment of the NICU, including noise sources, alarms, and sound levels measured in a NICU in a Boston hospital. One striking finding is that, in stark contrast to the intrauterine environment, periods of silence in the NICU are abundant. The consequences of this atypical perinatal acoustic exposure on auditory and brain development are unknown.

12:00

3aPPb6. Psychoacoustic sonification for tracked medical instrument guidance. Tim Ziemer (Spatial Cognition Ctr. (BSCC), Univ. of Bremen, Neue Rabenstr. 13, Hamburg 20354, Germany, tim.ziemer@uni-hamburg.de) and David Black (Inst. for Medical Image Computing, Fraunhofer MEVIS, Bremen, Germany)

In image-guided surgery, displays show a tracked instrument relative to a patient's anatomy. This helps the surgeon to follow a predefined path with a scalpel or to avoid risk structures. A psychoacoustically motivated sonification design is presented to help assist surgeons in navigating a tracked instrument to a target location in two-dimensional space. This is achieved by mapping spatial dimensions to audio parameters that affect the magnitude of different perceptual sound qualities. Horizontal distance and direction are mapped to glissando speed and direction of a Shepard tone. The vertical dimension is divided into two regions. Below the target, the vertical distance controls the LFO speed of an amplitude modulation to create a regular beating well below the threshold of roughness sensation. Above the target elevation, the vertical deflection controls the depth of frequency modulation to gradually increase the number and amplitudes of sidebands, affecting perceived noisiness and roughness. This redundancy is necessary because the magnitudes of each single sound quality are only differentiable with little confidence. In a preliminary study, non-surgeons successfully identified a target field out of 16 possible fields in 41% of all trials. The correct cardinal direction was identified in 84%. Based on findings and further psychoacoustic considerations, the mapping range is optimized and an implementation of an additional depth dimension is discussed.

Session 3aSAa

Structural Acoustics and Vibration: Energy Methods in Acoustics and Vibration I

Donald B. Bliss, Cochair
Mechanical Engineering, Duke University, 148B Hudson Hall, Durham, NC 27705

Linda P. Franzoni, Cochair
Dept. of Mech. Eng. and Materials Sci., Duke Univ., Box 90271, Durham, NC 27708-0271

Otto von Estorff, Cochair
Institute of Modelling and Computation, Hamburg University of Technology, Hamburg 21073, Germany

Chair's Introduction—9:15

Invited Papers

9:20

3aSAa1. Modeling high frequency broadband acoustic fields inside rectangular and cylindrical enclosures using an energy-intensity boundary element method. Donald B. Bliss, David Raudales (Mech. Eng., Duke Univ., 148B Hudson Hall, Durham, NC 27705, dbb@duke.edu), Krista A. Michalis (Naval Surface Warfare Ctr. Carderock Div., West Bethesda, MD), and Linda P. Franzoni (Mech. Eng., Duke Univ., Durham, NC)

Accurate prediction of high frequency broadband acoustic fields inside enclosures with curved boundaries typically requires time consuming frequency-by-frequency numerical techniques. A quick solution is developed for general enclosure shapes using an energy-intensity boundary element method, previously tested for rectangular geometries and now extended to cylinders with flat or spherical endcaps. Derived from first principles, which are reviewed, the approach uses uncorrelated spreading energy-intensity boundary sources to directly solve for the steady-state mean-square pressures. The enclosure boundary is discretized into radiating panels that account for interior energy transfer and satisfy prescribed reflection and absorption boundary conditions. Half-space orthogonal spherical harmonics serve as basis functions to emulate both diffuse and specular reflections. Panel interactions are calculated by an energy-accurate quadrature technique and studied for curved boundaries. Computationally intensive benchmark solutions are developed for verification. A fully correlated Helmholtz solution is derived for the cylindrical enclosure using a novel internal scattering approach to calculate high frequency pressure, and numerically integrated over a third-octave band. Comparisons between the fully correlated solution and the energy method for enclosures of various aspect ratios reveal excellent agreement. Simulations are also presented contrasting interesting behavioral differences between diffuse and specular reflection fields.

9:40

3aSAa2. Radiosity and radiative transfer in sound and vibration. Alain Le Bot (CNRS - Ecole centrale de Lyon, Ecole centrale de Lyon 36, av. Guy de Collongue, Ecully 69134, France, alain.le-bot@ec-lyon.fr)

At high frequencies in acoustics, the most popular method is ray-tracing and its variants including radiosity. In structural vibrations, the most known method is rather statistical energy analysis [1]. Both methods may be derived from a unique approach based on radiative transfer equations analogous to radiative exchanges of energy in thermics [2]. In this study, we present an overview of the radiative transfer equations in sound and vibration. We first show that radiosity is equivalent to ray-tracing with Lambertian reflection. In steady-state condition, radiosity is strictly equivalent to the view factor method in thermics. But in transient condition, radiosity provides an elegant solution to predict reverberation beyond the validity of Sabine's law. The theory is also well suited for structural rays in built-up structures. The radiative transfer equations account for reflection and transmission at interfaces of structural components. Sound radiation may also be described in the limit of high frequencies by this approach. It is also shown that a radiative transfer equation may include diffraction in a simple way. Simple and multiple diffraction by corners can be predicted. For all these phenomena, some numerical examples are presented to illustrate the relevance of the approach. [1] Foundation of statistical energy analysis in vibroacoustics, A. Le Bot. Oxford University Press, Oxford UK, 2015. [2] High frequency vibroacoustics: a radiative transfer equation and radiosity based approach, A. Le Bot, E. Reboul, Wave Motion, 2014.

10:00

3aSAa3. A regression-based energy method for predicting structural vibration and interior noise. Shung H. Sung (SHS Consulting, 4178 Drexel Dr., Troy, MI 48098, ssung@asme.org) and Donald J. Nefske (DJN Consulting, Troy, MI)

A regression-based energy method is developed for predicting the structural vibration and interior noise for prescribed loads applied to a structural-acoustic enclosure subject to differences in the structural or acoustic design. The formulation is based on the energy transfer functions that relate the applied load energy to the structural or acoustic response energy. The energy transfer functions are determined from a statistical regression analysis of the measured or predicted multiple responses that result from the differences in the structural or acoustic design. The applied load energy is determined analytically or experimentally for prescribed loading conditions. The energy method can then be used to estimate the mean-value and variation of the structural or acoustic response for different structural or acoustic designs and various prescribed loading inputs. A simple tube-mass-spring-damper system terminated with absorption material with variation is presented as an example. The practical application of the method to estimate the interior noise in an automotive vehicle for road and aerodynamic loads at different speeds is then presented. Comparisons of the predicted versus measured mean-value and variation of the sound pressure response show reasonable agreement. The methodology is generally applicable for rapidly estimating the structural or acoustic response for different designs and various loading conditions.

10:20

3aSAa4. Predicting the variance of the frequency-averaged energetic response in hybrid finite element—Statistical energy analysis. Edwin Reynders (KU Leuven, Kasteelpark Arenberg 40, Leuven 3001, Belgium, Edwin.Reynders@bwk.kuleuven.be) and Robin S. Langley (Univ. of Cambridge, Cambridge, United Kingdom)

In this contribution, the hybrid finite element-statistical energy analysis method is extended such that not only the mean and the ensemble variance of the harmonic system response can be computed, but also the ensemble variance of the frequency band-averaged system response. The computed variance represents the uncertainty that is due to the assumption of a diffuse field in components of the hybrid system. The developments start with a cross frequency generalization of the diffuse field reciprocity relationship between the total energy in a diffuse field and the cross spectrum of the external loading. By making extensive use of this generalization in a first-order perturbation analysis, explicit expressions are derived for the variance of the vibrational energies in the diffuse components and for the variance of the cross spectrum of the response of the deterministic components. These expressions are extensively validated against Monte Carlo analyses of systems consisting of connected plates, in which diffuse fields are simulated by randomly distributing small concentrated masses, acting as wave scatterers, across the diffuse components.

10:40

3aSAa5. High frequency analysis of a point-coupled parallel plate system. Dean R. Culver and Earl Dowell (Duke Univ., 12 Prestwick Pl., Durham, NC 27705, culver.dean@gmail.com)

The RMS response of various points in a system comprised of two parallel plates coupled at a point undergoing high frequency, broadband transverse point excitation of one component is considered. Through this prototypical example, Asymptotic Modal Analysis (AMA) is extended to two coupled continuous dynamical systems. It is shown that different points on the plates respond with different RMS magnitudes depending on their spatial relationship to the excitation or coupling points in the system. The ability of AMA to accurately compute the RMS response of these points (namely the excitation point, the coupling points, and the hot lines through the excitation or coupling points) in the system is shown. The behavior of three representative prototypical configurations of the parallel plate system: two similar plates (in both geometry and modal density), two plates with similar modal density but different geometry, and two plates with similar geometry but different modal density. After examining the error between reduced modal methods (such as AMA) to Classical Modal Analysis (CMA), it is determined that these several methods are valid for each of these scenarios. The data from the various methods will also be useful in evaluating the accuracy of other methods including SEA.

11:00

3aSAa6. General thermodynamics of vibrating systems. Antonio Carcaterra (Dept. of Mech. and Aerosp. Eng., Sapienza, Univ. of Rome, Via dei Velfra, Tarquinia, VT 01016, Italy, carcaterra.antonio@gmail.com)

This paper introduces a general view of a generalized thermodynamic theory for vibrating systems, with special emphasis to vibration and acoustics applications. One of the basis, is the temperature concept for Hamiltonian systems to describe the energy flow between two coupled sub-systems. As a result, a general and strict method to approach the energy analysis of linear and nonlinear systems, with potential applications both in theoretical mechanics as well as in engineering vibroacoustics and Statistical Energy Analysis is disclosed. The opportunity of a strict mathematical foundation to this important physical and engineering problem, is provided by the introduction of the Khinchin's entropy. The analysis shows that, under (i) linearity, (ii) weak coupling, and (iii) close-to-equilibrium conditions, a Fourier-like heat transmission law is obtained, where the thermodynamic temperature in proportional to the modal energy of the system, that is the ratio of its total energy and the number of its degrees of freedom. Generalized results both for large shocks and for nonlinear systems are indeed derived in closed form for weak anharmonic potentials, showing in this case that the temperature depends on a series of integer and fractional powers of the system's modal energy. At the end, a generalized statistical energy analysis of nonlinear systems is outlined.

11:20

3aSAa7. A boundary element based method for accurate prediction of the surface pressure cross-spectral density matrix. Jerry W. Rouse (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 North Dryden St., MS 463, Hampton, VA 23681, jerry.w.rouse@nasa.gov)

Accurate prediction of the surface pressure cross-spectral density matrix is necessary to predict the dynamic response of a structure loaded by a diffuse acoustic field. The cross-spectral density matrix describes the frequency dependence of the correlation between the surface pressure at all pairs of points on the structure. Most often the cross-spectral density matrix is obtained from either a uniform distribution of incident plane waves or direct application of the diffuse field spatial cross-correlation function. While the method of plane waves is relatively more accurate, especially at low frequencies, the necessary distribution of incidence angles and ensemble size can be problematic. This talk shall present a boundary element based methodology for determining the surface pressure cross-spectral density for any structure and frequency, including the effects of scattering and shielding. The method involves a power spectral density formulation of the boundary element method and takes advantage of the underlying foundations in potential theory. The method can be generalized beyond diffuse fields and can be applied to structures having a known surface impedance.

Contributed Papers

11:40

3aSAa8. Absorption scaling theory applied to an energy-intensity boundary element method for prediction of broadband acoustic fields in enclosures. David Raudales and Donald B. Bliss (Mech. Eng., Duke Univ., Edmund T. Pratt Jr. School of Eng. Box 90300 Hudson Hall, Durham, NC 27708, david.raudales@duke.edu)

Insight into the acoustic energy distribution in enclosures is gained by applying Absorption Scaling Theory. Broadband high-frequency acoustic fields within 3D rectangular enclosures are modeled with an energy-intensity boundary element method (EIBEM) that replaces the enclosure boundary with a distribution of broadband uncorrelated directional intensity sources to simulate either diffuse or specular reflection. Assuming a highly reflective enclosure, the boundary panel strengths are expanded in a power series with the spatially-averaged absorption as a small parameter. For diffuse reflection fields, where the theory is well developed, a matrix formulation is derived for each of the expansion coefficients that must be solved in sequential order. The leading order term in the expansion is inversely proportional to the scaling parameter and estimates the average level inside the enclosure, the next term gives spatial variation independent of average absorption level, while higher order terms account for the spatial variation of energy due to the distribution of absorption and the location of sources. For a highly reflective enclosure, only a few terms are needed to accurately predict the mean-square pressures. Similar behavior is demonstrated for specular reflection using numerical simulations. Applications include theoretical and empirical enclosure design and assessment, and solving the inverse problem.

12:00

3aSAa9. Sound radiation efficiency of unbaffled plates using an edge source integral equation. Kiran C. Sahu, U. Peter Svensson (Dept. of Electron. Systems, NTNU, Norway, Acoust. Res. Ctr., Norwegian Univ. of Sci. and Technol., Trondheim NO - 7491, Norway, kiran.sahu@ntnu.no), and Sara R. Martín (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

The accurate prediction of sound radiation from unbaffled vibrating plates remains a challenging problem. Finite structures make the sound waves diffract off and around the edges, an effect which is particularly strong at low frequencies. This phenomenon can be modeled with an edge source integral equation (ESIE) [A. Asheim, U. P. Svensson, J. Acoust. Soc. Am. 133 (2013) 3681-3691]. The modeling is based on a separation of the radiated sound into a geometrical-acoustics (GA) term, which equals the infinite-baffle solution, and diffraction of first- and higher orders. Expressions for the GA term and first-order diffraction are available explicitly, whereas higher-order diffraction is calculated through the solution of an integral equation. We present a method with a combination of time- and frequency-domain modeling which gives particularly efficient modeling. In this study, the sound radiation efficiency is targeted so only the sound field at the plate is computed. Thereby the numerically challenging receiver positions of the ESIE method at visibility zone boundaries are avoided. The results of the present study are compared with published results [A. Putra, D.J. Thompson, Applied Acoustics 71 (2010) 1113-1125] and close agreement is found, for a number of vibration modes.

Session 3aSAb

Structural Acoustics and Vibration, Physical Acoustics and Engineering Acoustics: Acoustic Metamaterials III

Christina J. Naify, Chair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Invited Papers

10:40

3aSAb1. Effects of visco-thermal losses in metamaterials slabs based on rigid building units. Vicente Cutanda Heniquez (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, vcuhe@elektro.dtu.dk), Victor Manuel Garcia-Chocano, and Jose Sanchez-Dehesa (Wave Phenomena Group, Universitat Politècnica de València, Valencia, Valencia, Spain)

Potential applications of negative-index acoustic metamaterials are strongly limited by absorptive effects of different origin. In this context, we present an investigation of the visco-thermal effects on the acoustic properties of double-negative metamaterials based on specifically designed rigid units with subwavelength dimensions. It is shown that visco-thermal losses dissipate about 70% of the acoustic energy associated to the excitation of monopolar and dipolar resonances, leading to the suppression of negative refractive index. Our numerical simulations based on the Boundary Element Method (BEM) are in excellent agreement with recent experimental data showing the quenching of the double-negative transmission peak. The BEM numerical model, which has been specifically adapted to this purpose, has also been validated against an equivalent Finite Element Method model. We also present the results and discuss the differences of visco-thermal effects on monopolar resonances leading to negative bulk modulus metamaterials, and Fabry-Perot resonances in metamaterial slabs.

11:00

3aSAb2. Non-reciprocal sound propagation in zero-index metamaterials. Li Quan, Dimitrios Sounas, and Andrea Alu (The Univ. of Texas at Austin, 1616 Guadalupe St. UTA 7.215, Austin, TX 78701, alu@mail.utexas.edu)

Moving media have recently attracted attention for their ability to break reciprocity without magnetic materials. By spinning air in an acoustic cavity, it was recently shown that it is possible to realize an acoustic circulator [R. Fleury, D. Sounas, A. Alù, Science 343, 516 (2014)], with applications for sonars and medical imaging devices. Here we show that the non-relativistic Fresnel-Fizeau effect at the basis of these mechanisms can be boosted in zero-index acoustic metamaterials, due to their large phase velocity. This is a different scenario than resonant structures, where the Fresnel-Fizeau effect is boosted by the effectively large wave-matter interaction distance, even for large intrinsic refractive index for the moving medium. Our results open a new venue to use zero-index metamaterials, and can become practically important in the realization of non-reciprocal acoustic imaging systems with built-in isolation and protection from reflections.

Contributed Papers

11:20

3aSAb3. Nonlinear metamaterial from piecewise discontinuous acoustic properties. Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, alexey.titovich@navy.mil)

Numerous metamaterials have recently been designed which exhibit either negative effective density and/or bulk modulus leading to extraordinary wave bearing capabilities. This research increases the design space by considering displacement-dependent properties which are discontinuous over a period of oscillation. Such nonlinear behavior has been studied in dynamical systems with intermittent contact. This work applies those analytical results together with numerical methods to provide insight into the harmonic and subharmonic generation in acoustic metamaterials with properties differing from compression to rarefaction. Stability is analyzed providing criteria for the onset of chaotic behavior.

11:40

3aSAb4. Nonlinearity based wave redirection for acoustic metamaterials. Saliou Telly (Mech. Eng., Univ. of Maryland College Park, 14359 Long Channel Dr., Germantown, MD 20874, stelly@umd.edu) and Balakumar Balachandran (Mech. Eng., Univ. of Maryland College Park, College Park, MD)

Advances in metamaterials have revealed novel opportunities for controlling wave propagation paths for various applications not realizable with conventional materials. Some prominent examples are schemes for electromagnetic and acoustic cloaking and focusing devices. In the classical approach to the formulations of these devices, one exploits a change of physical coordinates to achieve a desired wave behavior within a finite space. Such a change can be interpreted as a transformation of material properties when the field equations of interest are invariant to coordinate transformations. With regard to acoustics, this approach is constrained to fluid-like metamaterials amenable to the propagation of longitudinal waves only. Complications arise with solid materials because of their inherent ability to sustain both longitudinal and transverse waves, which refract differently in linear isotropic materials because of dissimilar propagation speeds.

In this work, the authors explore wave redirection mechanisms that may take advantage of nonlinear wave propagation phenomena in a solid metamaterial. Starting from the classical nonlinear Murnaghan model, a hyperelastic material is formulated to realize coupling between shear and compressional modes that could lead to a more suitable refractive behavior for acoustic wave redirection in a solid metamaterial. The formulated model is studied using analytical and numerical tools.

12:00

3aSAb5. Development of the underwater acoustic prism. Katherine F. Woolfe, Jeffrey S. Rogers, Matthew D. Guild, Charles Rohde (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, District of Columbia, katherine.woolfe@gmail.com), Christina J. Naify (Jet Propulsion Lab., Pasadena, CA), and Gregory Orris (Naval Res. Lab., Washington, District of Columbia)

The acoustic prism (i.e., leaky wave antenna) has been experimentally demonstrated in air as a way to steer an emitted beam using only a single broadband acoustic source. The prism relies on a leaky, dispersive waveguide to provide a unique radiation angle for each narrowband frequency projected by the acoustic source. In air, the leakage occurs through a series of periodically spaced shunts in the waveguide. This study examines an acoustic prism design that is capable of operating underwater, where leakage occurs through the waveguide wall itself due to the much lower impedance contrast of the waveguide material in water to that in air. This results in a geometrically simpler design in the underwater case. However, shear wave effects must be considered in the design of the underwater acoustic prism. The waveguide wall is constructed out of a composite material to have a high impedance but a low shear modulus, which are both necessary conditions to decrease sidelobes in the radiated pressure field. Numerical results indicate that the acoustic prism design is capable of scanning a range of frequencies from broadside to forward endfire. Experimental realization of the underwater acoustic prism is also discussed. [Work sponsored by ONR.]

TUESDAY MORNING, 27 JUNE 2017 BALLROOM A, 9:20 A.M. TO 12:20 P.M.

Session 3aSC

Speech Communication: Prosody (Poster Session)

Steven M. Lulich, Chair

Speech and Hearing Sciences, Indiana University, 4789 N White River Drive, Bloomington, IN 47404

All posters will be on display from 9:20 a.m. to 12:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 9:20 a.m. to 10:50 a.m. and authors of even-numbered papers will be at their posters from 10:50 a.m. to 12:20 p.m.

Contributed Papers

3aSC1. Reading aloud: Acoustic differences between prose and poetry. Filip Nenadic and Benjamin V. Tucker (Dept. of Linguist, Univ. of AB, 1617 8515 112 St. NW, Edmonton, AB T6G 1K7, Canada, nenadic@ualberta.ca)

Research on silent reading has shown that text genre influences the way texts are read, including differences between prose and poetry (e.g., Zwaan, 1994; Hanauer, 1998). There is little data examining whether text layout (prose vs. poetry) affects the way it is read aloud by non-expert readers, and, if yes, how do readers express those differences acoustically. Native speakers of Serbian (N = 28) and English (N = 37) read aloud twenty short texts in their native language. Stimuli were original texts that were acceptable as both prose and poetry, written by young published authors. Each text was formatted in four layouts (prose left aligned and justified, a single stanza and verses in multiple stanzas). Each participant saw each text in only one of these layouts. Separate mixed effects logistic regression analyses were performed for each language, testing whether prose vs. poetry layouts influenced the silent period and utterance duration, pitch, and intensity values of the productions. Differences and similarities between reading prose and poetry and between Serbian and English participants are discussed.

3aSC2. Focus effects on acoustic cues to sibilant place of articulation. Yung-hsiang Shawn Chang (Dept. of English, National Taipei Univ. of Technol., Zhongxiao E. Rd., Sec. 3, No. 1, Taipei 106, Taiwan, shawnchang@ntut.edu.tw)

Prosodically driven contrast enhancement has been reported for vowels and consonantal voicing, but limited evidence of such prosodic strengthening effects has been found for consonantal place of articulation (e.g., Cole *et al.* 2007, Silbert & de Jong 2008). Chang and Shih (2015) extended similar investigation to the Mandarin alveolar-retroflex contrast. They had participants disambiguate an alveolar or retroflex *syllable* with phonologically unrelated syllables (e.g., contrasting /sa/ with /bo/) and found no focus enhancement of the alveolar-retroflex contrast. The current study investigated whether employing a smaller contrastive focus domain (i.e., disambiguating contrastive *sibilants*, such as contrasting /sa/ with /ʂa/) would give rise to sibilant hyperarticulation. Map tasks with stimuli that were vowel context-balanced and lexical frequency-controlled were used for elicitation of focused and unfocused productions of Mandarin alveolars and retroflexes. Results showed that contrastive focus results in adjustments of non-contrastive properties (i.e., longer syllable and frication duration, as well as higher frication amplitude) without enhancing the feature-defining dimension (i.e., a greater acoustic distance between alveolar and retroflex sibilants). Along with evidence from English /s/ and /ʃ/ in Silbert & de Jong (2008), it is suggested that the place contrast of coronal sibilants is less subject to cue-enhancing hyperarticulation.

3aSC3. Prosodic characteristics of speech directed to adults and to infants with and without hearing impairment. Laura Dilley, Elizabeth Wieland, Evamarie Burnham (Michigan State Univ., Dept. of Communicative Sci. and, East Lansing, MI 48824, ldilley@msu.edu), Yuanyuan Wang, Derek Houston (The Ohio State Univ., Columbus, OH), Maria V. Kondaurova (Univ. of Louisville, Louisville, KY), and Tonya Bergeson (Indiana Univ. School of Medicine, Indianapolis, IN)

Infant-direct (ID) and adult-directed (AD) speech are distinguished via multiple acoustic-prosodic characteristics, but it is unclear how these differences map onto linguistic constructs, including pitch accents, prominences, and phrasal boundaries, or how a child's hearing impairment affects

3a TUE. AM

caregiver prosody. In two studies, trained analysts coded prosody in corpora of mothers reading to their children (ID condition) or another adult (AD condition). In Study 1, 48 mothers read a storybook to their infants aged 3, 9, 13, or 20 months or an experimenter. In Study 2, 11 mothers read a storybook to their child with a cochlear implant at 3 months post-implantation or to an experimenter; each hearing-impaired child was paired two normal-hearing dyads based on the hearing-impaired child's chronological age and amount of hearing experience. ID speech contained a greater density of pitch accents and prominences than AD speech. There was no difference in distributions of phrase boundaries across speech styles, and hearing status did not mediate effects of speech style on prosody. Results suggest that acoustic differences distinguishing ID and AD speech map onto combined phonological structural and gradient paralinguistic characteristics and contribute to understanding effects of child hearing loss on caregiver input. [Work supported by NIH Grant 5R01DC008581-07.]

3aSC4. The perception of speech rate in non-manipulated, reversed and spectrally rotated speech reveals a subordinate role of amplitude envelope information. Volker Dellwo and Sandra Schwab (Phonet. Lab., Universitaet Zurich, Plattenstrasse 54, Phonet. Lab., Zurich 8005, Switzerland, volker.dellwo@uzh.ch)

Previous research suggests that the broad-band amplitude envelope (ENV) of speech is crucial for the perception of speech rate and timing. The present experiment tested this claim using non-manipulated and spectrally rotated speech (rotated around 2.5 kHz) with a bandwidth of 5 kHz which both contain identical ENV and reversed speech in which the temporal organisation of ENV is distorted. 44 listeners of Swiss German rated perceived speech tempo on a continuous scale reaching from "rather slow" to "rather fast" in 48 stimuli (4 sentences $\times$ 4 speakers $\times$ 3 signal conditions). Results revealed a significant effect of signal condition. Both reversed and spectrally rotated speech were perceived as significantly faster than clear speech but there was no difference between spectrally rotated and reversed speech. Results were consistent for all sentences and speakers. Results suggest that the intelligibility of the signal plays a higher role in the perception of speech rate than the presence of the ENV.

3aSC5. Spectro-temporal cues for perceptual recovery of reduced syllables from continuous, casual speech. Laura Dilley, Meisam K. Arjmandi, and Zachary Ireland (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Function words may be highly reduced, with little to no discontinuity marking their onsets to cue their segmentation from continuous speech. The present study investigated whether reduced function words lacking onset discontinuities have residual timing cues that could be used for word segmentation. Participants (n = 51) briefly viewed sentences and spoke them from memory to elicit casual speech. They were randomly assigned to either a "function-word present" condition (n = 29) in which experimental items contained a critical function word expected to frequently blend spectrally with context, or a "function-word absent" set (n = 22) with phonetically matched items lacking the critical word. Acoustic analyses confirmed that in "function-word present" sentences, critical words lacked detectable onset discontinuities 60% of the time. Critically, in the "function-word present" condition, portions of speech containing critical function words were longer, both in terms of absolute duration and normalized for context speech rate, compared with matched portions in the "function-word absent" condition, even when the former were highly reduced and lacked onset discontinuities. These findings suggest that relative duration cues provide substantial information which may be used by listeners for segmentation of highly reduced syllables from continuous speech. [Work supported by NSF Grant BCS-1431063.]

3aSC6. A model of Mandarin Chinese question intonation. Edward Flemming (Linguist & Philosophy, MIT, 77 Massachusetts Ave., 32-D808, Cambridge, MA 02139, flemming@mit.edu)

Echo questions in Mandarin Chinese provide an interesting case study of the interaction between lexical tone and intonation because it appears that echo questions are distinguished from declaratives by modifying the F0 trajectory of the sentence-final lexical tone in ways that depend on the identity of that tone. The most general effect is that the offset of the final tone is higher in a question than in a declarative rendition of the same sentence. If the final tone is high or falling, F0 is raised throughout the tone, but if the final tone if low or rising, the F0 minimum is not raised in questions. We propose a quantitative analysis according to which question intonation consists of a high boundary tone realized simultaneously with the offset of the final lexical tone. Compromise between the boundary tone and targets for the lexical tone results in raising of the offset of the tone. Additional effects result from interactions with other targets pertaining to the shape of the lexical tone. For example, simply raising the offset of a falling tone would result in failure to realize a sufficiently steep fall, so the onset of the fall is raised as well.

3aSC7. Prosodic cues to psychosis risk. Emily Cibelli, Jennifer Cole (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, emily.cibelli@northwestern.edu), Vijay Mittal (Psych., Northwestern Univ., Evanston, IL), and Matthew Goldrick (Linguist, Northwestern Univ., Evanston, IL)

Schizophrenia is known to impact prosody, often described as "flat affect." Individuals with schizophrenia show reduced F0 variability relative to neurotypical individuals (Rapcan *et al.* 2010). However, the speech of adolescents at high risk for psychosis in the prodromal (pre-diagnosis) stage has not been investigated, leaving open the question of whether speech prosody might be an early signal of symptoms associated with psychotic disorders. To investigate this issue, the speech of 18 ultra high-risk (UHR) youth (ages 16-21) was compared to 18 age- and gender-matched controls. F0 (pre-processed for smoothing and error correction) was extracted from 10-minute segments of speech recorded during clinical interviews. Using LDA classification, F0 summary statistics (mean and variability) separated male UHR and control speakers (69% accuracy) but not female speakers (42% accuracy), consistent with gender differences in psychosis onset during adolescence (Ochoa *et al.*, 2012). Linear models of symptoms measured by the Structural Interview for Prodromal Syndromes (Miller, 2003) found that F0 mean and variability predicted negative symptoms in UHR speakers when controlling for age and gender. These results suggest that prosodic markers documented in individuals with psychosis may also be present in prodromal populations, pointing to speech as a potential biomarker for psychosis risk.

3aSC8. Loudness trumps pitch in politeness judgments: Evidence from Korean. Kaori Idemaru, Lucien Brown (East Asian Lang. and Literatures, Univ. of Oregon, Eugene, OR 97403, idemaru@uoregon.edu), Bodo Winter (Univ. of Birmingham, Merced, California), and Grace E. Oh (Konkuk Univ., Seoul, South Korea)

Politeness is a vital aspect of everyday life that is receiving increased attention in sociophonetic research. The current study investigated how deferential and intimate stances, examples of politeness-related expressions, are conveyed by phonetic cues in Korean. Previously, we found that Korean listeners can distinguish these stances based on speech acoustics alone. The current study manipulated fundamental frequency (F0) and intensity of spoken Korean utterances to investigate the specific role of these cues in politeness judgments. Across three experiments with a total of 63 Korean listeners, we found that intensity reliably influenced politeness judgments, but F0 did not. An examination of individual differences revealed that *all* listeners interpreted deferential stances to be associated with low intensity: quiet utterances were perceived as deferential. On the other hand, the interpretation of F0 varied across listeners: some perceived high-pitched utterances as deferential and others perceived low-pitched utterances as deferential. These results present a challenge to the Frequency Code as a universal principle underlying politeness phenomena. The results also indicate that perception does not perfectly mirror production in politeness expressions in Korean, since previous production studies have reliably found low pitch to be associated with deferential stances.

3aSC9. Uptalk in northern Irish English. Anna Jespersen (Dept. of English, Aarhus Univ., Vesterport 7, 105, Aarhus C 8000, Denmark, anna.jespersen@cc.au.dk)

This paper presents work in progress on the phonetic realization of uptalk rises in Northern Irish English, a variety which is well-known for another type of rising intonation, the rise-plateau(-slump) (Cruttenden 1997; Grabe 2002; Ladd 2008). However, a recent pilot study has shown that the steeper and more steadily rising uptalk rise, which is mainly associated with American and Antipodean Englishes, is now found not only in Southern British English (cf. Arvaniti and Atkins 2016), but also in Northern Ireland. For this study, 6 female speakers were recorded while taking part in a Map Task and approximately 3 minutes of speech was examined. Intonational rises were labeled using the IViE guidelines (see Grabe *et al.* 2000; Grabe 2002) and f0 measurements were taken at 10 intervals between the low starting points and peaks of each rise. Rises were then assigned to either the rise-plateau or uptalk categories according to the phonological label assigned and the steepness, height and steadiness of the rise. This study thus provides confirmation that Northern Irish English speakers really use uptalk rises, and acoustic evidence of how these differ from the variety's traditional rise-plateaux.

3aSC10. Segmental intonation in tonal and non-tonal languages. Maida Percival and Kaz Bamba (Linguist, Univ. of Toronto, 100 St. George St., 4th Fl., Toronto, ON M5S 3G3, Canada, maida.percival@mail.utoronto.ca)

The nature of edge intonational contours as well as acoustics of fricatives have generally been independently discussed in the literature (Hughes & Halle 1956; Ladd 1996; Gussenhoven 2004 inter alia). Voiceless consonants were traditionally conceived as irrelevant to the study of utterance-level intonation and thought merely to interrupt pitch contours (Bolinger 1964). However, Niebuhr (2012) proposes that the two domains interact, reporting that German fricatives exhibit relatively higher centre of gravity (CoG) and higher acoustic energy in the context of rising intonation. This phenomenon, known as segmental intonation, has been found in some languages (Polish, Zygis *et al.* 2014; Dutch, Heeren 2015), but remains controversial in others (English, Niebuhr p.c.). We test this hypothesis by replicating the reading task in Niebuhr's (2012) study for English and also extending to a tonal language, Cantonese, in which F0 is used grammatically to distinguish words, in addition to intonation. Preliminary results from 10 speakers suggest that segmental intonation in the form of higher CoG and intensity exists in English, but not in Cantonese. With additional data recently collected, we hope to confirm these findings and contribute to determining whether the segmental intonation is an epiphenomenon of speech production in general or not.

3aSC11. Recognition of emotional prosody in Mandarin: Evidence from a synthetic speech paradigm. Cecilia L. Pak and William F. Katz (Commun. Sci. and Disord., Univ. of Texas at Dallas, 800 West Campbell Rd., Richardson, TX 75080, sxl083020@utdallas.edu)

Pitch-dominant information is reduced in the spectrally-impoverished signal transmitted by cochlear implants (CIs), leading to potential difficulties in perceiving voice emotion. However, this evidence comes from non-tonal languages such as English, in which pitch information is not required for lexical meaning. In order to better understand how hearing impaired (HI) speakers of a tone language with cochlear implants (CIs) process emotional prosody, an experiment was conducted with healthy normal hearing (NH) Mandarin-speaking adults listening to synthetic stimuli designed to resemble CI input. Listeners heard short sentences from a read-speech database produced by professional actors. Stimuli were selected to express four emotions ("angry," "happy," "sad," and "neutral"), under four conditions which varied the lexical tones of Mandarin. Listeners heard natural speech and three noise-vocoded speech conditions (4-, 8-, and 16-spectral channels) and made a four-alternative, forced-choice decision about the basic emotion underlying each sentence. Preliminary results indicate more accurate emotional prosody recognition for natural speech than for synthesized speech, with greater accuracy for higher channel stimuli than lower channel stimuli. The findings also suggest NH Mandarin-speaking listeners show lower overall vocal emotional prosody accuracy compared with previous studies of non-tonal languages (e.g., English).

3aSC12. Proposal of description for an intonation pattern: The simulacrum of neutral intonation. Marcus Vinicius Moreira Martins and Waldemar Ferreira-Netto (Letras Clássicas e Vernáculas, Univ. of São Paulo, Av. Professor Luciano Gualberto, 403 - Departamento de Letras Clássicas e Vernáculas, São Paulo, São Paulo 05508-900, Brazil, marcusvmmartins@gmail.com)

The aim of this study is describe the intonation pattern Simulacrum of Neutral Intonation (SNI), defined as the monotone speech produced by the speakers with a prevalence of rhythm and a flat speech melody. As first step, 71 recordings were separated by gender of the subjects and their emotional states: 9 anger male and female speech, 10 neutral male speech, 11 neutral female speech, 11 sad male speech, 9 sad female speech, 7 SNI female speech, and 5 SNI male speech. The 71 samples were analyzed in terms of 44 acoustical parameters established a priori and separated by clustered using the parameters selected by Principal Component Analysis. The four final parameters of this analysis were as follows: (i) lower F0 value, (ii) lower dispersion of positive variations of Focus/Emphasis, (ii) lower mean interval between units of bearing of intonation, and (iv) lower median of these same intervals. These values shows a possible characterization of intonational speech register, called Simulacrum of Neutral Intonation. The SNI record is similar to other speech registers, as those used by speakers with psychic disorders or in stressful situations. These values confirms some findings, as seen in Ragin *et al.* (1989) and Thomas *et al.* (1990).

3aSC13. Word length modulates the effect of emotional prosody. Seung Kyung Kim (Aix Marseille Univ, CNRS, LPL, 5 Ave. Pasteur, Aix en Provence 13100, France, kim.seungkyung@gmail.com)

Previous work has shown that emotional prosody, independent of the lexical carrier, activates words associated with the emotional information (Kim, 2015; Kim & Sumner, submitted). For example, hearing a non-emotional word (*pineapple*) uttered with angry prosody facilitates recognition of angry-associated words (*mad*). Building on this finding, the current study delves into the nature of the affective priming between emotional prosody and emotional words and tests if word length modulates affective priming. Word length is an important dimension in lexical processing, as longer words are shown to produce stronger lexical activation than shorter words (Pitt & Samuel, 2006). I hypothesize that social information shows a stronger effect in spoken word processing when lexical activation is weaker. Then we should find stronger affective priming with shorter words than longer words. This hypothesis was tested with a cross-modal priming experiment. The visual targets were 12 angry related words (e.g., *mad, upset*). The targets were preceded by two-, three-, or four-syllable non-emotional primes (e.g., *atom, envelope, aluminum)* spoken with angry prosody. Listeners recognized angry words faster after hearing angry prosody than after hearing neutral prosody when the prime words were short (2 syllables) but not when the prime words were longer (3-4 syllables). The current results provide evidence that social effects in word recognition are modulated by the strength of lexical activation.

3aSC14. A cross-linguistic study of speech modulation spectra. Léo Varnet (Laboratoire des Systèmes Perceptifs, Institut d'Étude de la Cognition, ENS Paris, Laboratoire des Systèmes Perceptifs, 29 rue d'Ulm, Paris 75005, France, leo.varnet@ens.fr), Maria C. Ortiz-Barajas, Ramón Guevara Erra, Judit Gervain (Laboratoire Psychologie de la Percept., Université Paris-Descartes, CNRS, Paris, France), and Christian Lorenzi (Laboratoire des Systèmes Perceptifs, Institut d'Étude de la Cognition, ENS Paris, Paris, Ile de France, France)

Languages show systematic variation in their sound patterns and grammars. Accordingly, they have been classified into typological categories such as stress-timed vs. syllable-timed on the basis of their rhythms, Head-Complement vs. Complement-Head on the basis of their basic word order, or tonal vs. non-tonal on the basis of the presence/absence of lexical tones. To date, it has remained incompletely understood how these linguistic properties are reflected in the acoustic characteristics of speech in different languages. In the present study, the amplitude-modulation (AM) and frequency-modulation (FM) spectra of 1862 utterances produced by 44 speakers in 12 languages were analyzed. Overall, the spectra were similar across languages. However, a perceptually based representation of the AM

spectrum revealed significant differences between languages. The maximum value of this spectrum distinguished between HC non-tonal, CH non-tonal, and tonal languages, while the exact frequency of this maximum value differed between stress-timed and syllable-timed languages. Furthermore, when normalized, the f0-modulation spectra of tonal and non-tonal languages also differed. These findings reveal that broad linguistic categories are reflected in differences in temporal modulation features of different languages. This has important implications for theories of language processing and acquisition.

3aSC15. Effects of dialect and syllable onset on Serbian tone timing. Robin P. Karlin (Linguist, Cornell Univ., 103 W Yates St., Ithaca, NY 14850, karlin.robin@gmail.com)

In recent years, the c-center effect has been posited as the main coordinative structure for lexical tone. This has two predictions: (1) the timing of tone gestures is affected by syllable onset complexity, and (2) tone gestures will start relatively late compared to the first onset consonant. However, little work has investigated these predictions, thus far focusing on languages with few word-initial clusters. In this acoustic study, I examine two dialects of Serbian, a pitch-accent language that allows word-initial sonorant-sonorant clusters such as /ml/. In both dialects, the H(igh) tone excursion starts later in syllables with complex onsets, in accordance with prediction 1 and is suggestive of a c-center structure. The dialect comparison addresses the second prediction, as Valjevo Serbian has early peaks, and Belgrade Serbian has late peaks. The results show that the early peaks in Valjevo Serbian are due to a combination of both earlier H onset and shorter pitch excursion, which suggests that at least one of the dialects is not using a c-center structure. Based on these results, a model of tonal representation is proposed that has implications for the possible coordinative structures of lexical tone.

3aSC16. Imitation of F0 timing in Mandarin disyllabic sequences. Hao Yi (Dept. of Linguist, Cornell Univ., 307 Columbus Ave., APT 9, Syracuse, NY 13210, hy433@cornell.edu) and Sam Tilsen (Dept. of Linguist, Cornell Univ., Ithaca, NY)

This study investigates the control of relative timing between tones and segments in Mandarin Chinese. Thirty native Mandarin speakers participated in an experiment in which they imitated the variation in a disyllabic, bi-tonal sequence—Tone2 + Tone2 (rising + rising). The stimuli vary parametrically in the relative timing of F0 turning points with respect to the segment boundary. The variation occurs either within the first syllable or between the two syllables. The results show that within the first syllable, speakers did not imitate the variation in the relative timing patterns. However, across syllable boundaries, such parametric variation leads to more faithful imitations in terms of the relative timing of F0 turning points. Therefore, native Mandarin speakers are more sensitive to variation in the relative timing patterns across syllable boundaries than within the first syllable. This shows that the control over the relative timing between F0 gestures and articulatory gestures within the first syllable is more stable than across syllable boundaries. We argue this is because final lengthening of the second syllable can provide an additional co-selection set with which the Low tone gesture can be associated.

3aSC17. Listener adaptation to lexical stress misplacement. Maho Morimoto (Linguist, Univ. of California, Santa Cruz, 1156 High St., Santa Cruz, CA 95064, mamorimo@ucsc.edu)

Speech including unfamiliar accents can result in decreased processing efficiency. However, listeners are able to overcome the difficulty of processing accented speech with adequate exposure, through the process of perceptual adaptation (Norris, McQueen & Cutler 2003; Bradlow & Bent 2008, among others). The current study addresses the role of word-level prosodic information in listener adaptation to accented speech. Specifically, it investigates adaptation to lexical stress misplacement in English, and examines how it compares with adaptation to segmental mismatches and accentedness at the whole utterance level. 91 native speakers of English were exposed to English words in isolation with canonical and non-canonical stress position, while performing a speeded cross-modal matching task. Results suggest that adaptation to lexical stress misplacement is largely comparable to adaptation to non-canonical productions at the segmental or utterance level in terms of speed and generalizability across lexical items. Results also indicate that adaptation to lexical stress misplacement is generalizable across talkers to some extent.

3aSC18. Decoding linguistically-relevant pitch patterns from frequency-following responses using hidden Markov models. Fernando Llanos, Zilong Xie, and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas at Austin, 1801 Rio Grande St., 103, Austin, TX 78701, fllanos@utexas.edu)

Pitch encoding is often studied with frequency-following response (FFR), a scalp-recorded potential reflecting phase-locked activity from auditory subcortical ensembles. Prior work using FFR have shown that long-term language experience modulates subcortical encoding of linguistically-relevant pitch patterns. These studies typically rely on FFRs averaging across thousands of repetitions, due to low signal-to-noise ratio of single-trial FFRs. Here, we evaluated the extent to which hidden Markov models (HMMs), *with fewer numbers of trials*, can be used to quantify pitch encoding as well as capture language experience-dependent plasticity in pitch encoding. FFRs were recorded from fourteen Mandarin Chinese and fourteen American English passively listening to four Mandarin tones (1000 trials per tone). HMMs were used to recognize FFRs to each tone in individual participants. Specifically, HMMs were trained and tested across FFR sets of different sizes, ranging from 50 to 500 trials. Results showed that HMMs were able to recognize tones from FFRs, above chance, across all training sizes and languages. Interestingly, HMMs picked up language differences (Chinese $>$ English) at very small sizes for training (e.g., 200) and testing (e.g., 100). These findings highlight the potential benefits of using HMMs to reduce experimental time and efforts in FFR data collection. [Project funded by NIH NoR01DC013315 834 (B.C.).]

3aSC19. Effects of consonant voicing on vocalic segment duration across resonants and prosodic boundaries. D. H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu)

Most languages, and especially English, reduce the duration of vocalic segments before voiceless obstruents relative to voiced ones. Previous studies examined this effect for final singleton consonants in tauto- and hetero-syllabic contexts within a word. Here, monosyllabic words are examined for the effect both across resonants (e.g., "code"/"coat" vs. "cold"/"colt") and across word boundaries and stress conditions (e.g., "no CODE"/"NO code" vs. "no GOAD"/"NO goad"). Preliminary results showed a typical effect for singleton stops, with vocalic segment reduced 33.6% for the voiceless stop. Vocalic segments were somewhat reduced with an intervening resonant (26.4%) but the resonance itself was more reduced (57.3%); vocalic segment and resonance were approximately the same duration as the vocalic segments alone in singleton syllables. The vocalic segment of "no" decreased slightly with following voiceless stop (8.2% when unstressed, 10.2% when stressed); this was smaller than other studies' effect across syllable boundaries within words. The results indicate that the word and syllable structure impose timing effects of different magnitudes on both vowels and resonants due to voicing of an adjacent stop. It is surprising that the effect occurs across word boundaries and further that stress makes little difference. [Work supported by NIH grant DC-002717.]

Session 3aSP

Signal Processing in Acoustics, Engineering Acoustics, and Architectural Acoustics: Signal Processing for Directional Sensors III

Kainam T. Wong, Chair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, DE 605, Hung Hom KLN, Hong Kong

Contributed Papers

9:20

3aSP1. Numerical study on the effect of various parameters on beamforming performance and the estimation of particle velocity using a circular array. Sea-Moon Kim and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., 32 Yuseong-daero 1312beon-gil, Yuseong-gu, Daejeon 34103, South Korea, smkim@kriso.re.kr)

Numerous studies on beamforming techniques have been done for the estimation of source direction with an array. The conventional beamforming (delay and sum), MVDR, and MUSIC are some examples. Recently, the frequency difference beamforming was also introduced for applications to high frequency ranges or the elimination of the aliasing effect. This talk compares performance of the beamforming techniques including frequency difference beamforming using a circular array. The effects of the various parameters, such as number of sensors, array radius, frequency range, and SNR, on the beamforming performance are studied. The error analysis for the estimation of particle velocity is also discussed. [This work was financially supported by the research project PES9020 funded by KRISO.]

9:40

3aSP2. Modal beamforming for circular acoustic vector sensor arrays. Berke M. Gur (Mechatronics Eng., Bahcesehir Univ., Ciragan Cad. Osmanpasa Mektebi Sok., No: 4-6 Besiktas D-527, Istanbul 34349, Turkey, berke.gur@eng.bau.edu.tr)

Vector sensors are directional receivers that measure the vectorial particle velocity associated with an acoustic wave rather than the scalar pressure. Therefore, arrays of vector sensors possess some desirable directional properties compared to conventional arrays of pressure sensors. In this paper, a modal beamformer for circular arrays of 1-D acoustic vectors sensors are presented. Arrays of both radially and circumferentially oriented vector sensors are considered. It is shown that the highly directional modes of the acoustic velocity field can be extracted from the sensor measurements using the spatial Fourier transform. These modes are weighted and combined to form narrow steerable beams. The highest order of mode that can be extracted is limited by the number of vector sensors utilized in the array. Theoretical analysis and numerical simulations indicate that the proposed modal beamformer attains the same directivity performance as that of circular pressure sensor array beamformers but outperforms them in terms of white noise gain. In addition, it uses half the number of sensors to achieve the same directivity performance of a circular vector sensor array modal beamformer reported previously in the literature. The proposed method is suitable for in-air and underwater low frequencies array processing applications.

Invited Papers

10:00

3aSP3. Influence of beat phenomenon on direction of arrival estimation based on a single vector hydrophone. Hongning Liu, Yi Zheng, Yufeng Mao, Yanting Yu, and Xiaodong Gong (Shandong Acad. of Sci. Inst. of Oceanographic Instrumentation, 28 Zhejiang Rd., Shinan District, Qingdao, China, maoyf_sdioi@163.com)

The vector hydrophone can measure sound pressure and orthogonal components of particle velocity co-locately in space and simultaneously in time. DOA (direction of arrival) estimation based on a single vector hydrophone has attracted extensive attention. Beat phenomenon can cause confusion in the DOA estimation results. In this paper, the impact mechanism of beat phenomenon to DOA estimation by cross spectrum method and average acoustic intensity method is explored through theoretical study and simulation. The result shows that beat phenomenon causes the change of sound pressure and vibration velocity amplitude with time and leads to DOA estimation results of these two methods are instability eventually. First, DOA is estimated by the average sound intensity method, if the temporal resolution is smaller than the minimum envelope period, the DOA estimation will change over time; Second, DOA is estimated by the cross spectral method, if the FFT frequency resolution is greater than frequency difference, it is unable to distinguish the frequency of these two signals. Subsequent operations will also be confused by two signal energy, causing error estimates. In the end, the solutions and application examples are proposed. The conclusion could be applied to improve the DOA estimation performance of a single vector hydrophone.

10:20–10:40 Break

10:40

3aSP4. Temporal-spatial distributions for a repertoire of fin whale vocalizations from directional sensing with a coherent hydrophone array. Heriberto A. Garcia (Elec. and Comput. Eng., Northeastern Univ., 151 Mystic St., Apt 35, Arlington, MA 02474, garcia.he@husky.neu.edu), WEI HUANG (Elec. and Comput. Eng., Northeastern Univ., Malden, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

The ability to monitor and differentiate vocalizations from a given marine mammal species can be challenging with single sensor measurements when there are multiple marine mammal species vocalizing in close proximity and when the vocalizations have not been observed or documented previously. Here we employ a large-aperture coherent hydrophone array system with directional sensing to detect, localize, and classify a repertoire of fin whale vocalizations using the passive ocean acoustic waveguide remote sensing (POAWRS) technique. The fin whale vocalizations are comprised of their characteristic 20 Hz centered pulses, interspersed by 130 Hz centered upsweep calls, and other vocalizations with frequencies ranging between 40 and 80 Hz. The directional sensing ability of POAWRS is essential for associating various call types to fin whales after long term tracking of the vocalization bearing-time trajectories and localizations over multiple diel cycles. Here, we quantify the relative diel occurrence of the three distinct fin vocalization types and apply the results to infer their behaviors as a function of the observation region.

11:00

3aSP5. Instantaneous wide-area passive acoustic monitoring of surface ships and submerged vehicles using a coherent hydrophone array. Wei Huang (Elec. and Comput. Eng., Northeastern Univ., 500 Broadway, Apt. 3157, Malden, MA 02148, huang.wei1@husky.neu.edu), Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), and Heriberto A. Garcia (Elec. and Comput. Eng., Northeastern Univ., Arlington, MA)

The ability to monitor surface ships and other ocean vehicles continuously over instantaneous wide areas is essential for a wide range of applications including defense and ocean environmental assessment. Here, we employ a large-aperture coherent hydrophone array system to detect, localize, and classify several surface ships and other ocean vehicles from their sounds radiated underwater using the passive ocean acoustic waveguide remote sensing (POAWRS) technique. The approach is calibrated for four distinct research and fisheries survey vessels with accurately known locations obtained from global positioning systems (GPS). Acoustic signals passively recorded on the coherent hydrophone array are first beamformed for their azimuthal dependencies. The sounds radiated by ocean vehicles are automatically detected using a threshold detector from the beamformed spectrograms. The bearing versus time trajectories of sequences of detections are used to localize the ocean vehicles by employing the moving array triangulation technique. The sounds radiated by the ships are dominated by distinct tonals and cyclostationary signals in the 50 to 2000 Hz frequency range. The temporal-spectral characteristics of these signal can be used to classify each ship. Our analysis indicates the ocean vehicles can be instantaneously monitored and tracked over wide areas spanning more than 300 km in diameter.

Contributed Paper

11:20

3aSP6. The MITRE undersea sounding experiment 2016. Nicholas A. Rotker, Ballard J. Blair, Caitlyn N. Marcoux, Alexander J. Tooke (The MITRE Corp., MS S118, 202 Burlington Rd., Bedford, MA 01730, nrotker@mitre.org), Ilya A. Udovydchenkov (The MITRE Corp., Woods Hole, MA), Bindu Chandna, Melissa G. Meyer (The MITRE Corp., Bedford, MA), and Harold T. Vincent (Ocean Eng. Dept., Univ. of Rhode Island, Narragansett, RI)

This talk describes the MITRE Undersea sounding experiment (MUSE16) conducted in Narragansett Bay from September 12-23, 2016, where acoustic communication, localization waveforms, and signal processing techniques were explored. This experiment utilized newly developed acoustic buoys which were designed and built by the University of Rhode Island (URI) Ocean Engineering Dept. in collaboration with the MITRE Corporation. The buoys use Global Positioning Satellites (GPS) for localization and time synchronization and are capable of both transmitting and receiving acoustic data in the range of 8-18 kHz. The buoys were designed to further research in the areas of acoustic communications, channel modeling, and continuous active sonar (CAS). For the communication and channel modeling experimentation, modulated M-sequences of various sequence length were transmitted to explore channel characterization and communication enhancements. For the CAS experimentation, Linear Frequency Modulated (LFM) chirps of various bandwidths and center frequencies were explored as well as utilization of several underwater targets. A description of the prototype buoys including hardware, software, experimental setup, types of data collected, as well as some initial results will be discussed.

Invited Paper

11:40

3aSP7. Direction-finding techniques for a small-aperture hydrophone array. Martin Gassmann, Sean M. Wiggins, and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9152 Regents Rd., Apt. L, La Jolla, CA 92037, mgassmann@ucsd.edu)

A volumetric array of hydrophones was coupled to a long-term, autonomous acoustic recorder and deployed ~130 km offshore of Southern California to the seafloor (~1300 m depth) to track continuous-wave (CW) and transient underwater sound sources. The array was composed of four, wide-band, omnidirectional hydrophones closely-spaced ~1 m apart. Sampling was continuous at 100 KSamples/s for each hydrophone over a period of more than two months. To track CW sound sources, conventional and adaptive beamforming techniques were implemented. The array's beam pattern characteristics as a function of frequency (10—1000 Hz) were investigated by simulating the arrival of plane waves from directions of interest. Beamforming techniques were opportunistically applied to nearby (<5 km) transiting commercial ships with automated identification system (AIS) transmitted locations. Discrepancies were within the physical dimension of the transiting ships. Source levels for each of the transiting ships were estimated at the ships' various aspects to

characterize the directionality of underwater ship noise. In addition, a cross-correlation based direction-finding technique for transient sounds was developed and implemented for short duration (<1 ms), frequency-modulated echolocation clicks emitted by deep-diving beaked whales.

Contributed Paper

12:00

3aSP8. Design of a microphone array for near-field conferencing applications. Pieter Thomas (Res. group WAVES, Ghent Univ., Technologiepark-Zwijnaarde 15, Gent B-9052, Belgium, pieter.thomas@ugent.be), Reinout Verburgh, Michael Catrysse (Televic Conference, Kachtem, Belgium), and Dick Botteldooren (Res. group WAVES, Ghent Univ., Ghent, Belgium)

Microphone arrays are becoming increasingly popular for conferencing applications and near-field speech recording. In this work, a 16-element cylindrical microphone array is designed for beamforming toward a nearby speaker, while reducing the influence of competing talkers. A two-stage approach is used to obtain the desired array directivity pattern, optimizing both microphone locations and filter weights. The positions of the microphones are optimized by using a hybrid optimization technique, taking into account the influence of the nearby acoustic environment (array shape and conferencing desk). FIR filter coefficients for each microphone are derived from a regularized least-squares (LSQR) solution, combined with null-steering. An implementation of the array is made with digital MEMS microphones and the performance of the design is evaluated experimentally and compared with a classically used goose-neck microphone.

TUESDAY MORNING, 27 JUNE 2017 — ROOM 309, 9:20 A.M. TO 12:00 NOON

Session 3aUWa

Underwater Acoustics, Acoustical Oceanography, Engineering Acoustics, and Signal Processing in Acoustics: A Century of Sonar I

Michael A. Ainslie, Cochair
Underwater Tech. Dept., TNO, P.O. Box 96864, The Hague 2509JG, Netherlands

Kevin D. Heaney, Cochair
OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039

Invited Papers

9:20

3aUWa1. From Canton to the Curie brothers: The dawn of sonar. Michael A. Ainslie (Acoust. and Sonar, TNO, P.O. Box 96864, The Hague 2509JG, Netherlands, michael.ainslie@tno.nl) and Willem D. Hackmann (Oxford, United Kingdom)

Toward the end of the First World War, Paul Langevin and Robert Boyles developed the first underwater echo-ranging systems capable of detecting and localizing submarines, with important consequences for the outcome of the Second World War. Their invention involved the use of quartz transducers for initial generation of sound and reception of the echo, and vacuum tubes for amplification of the (much weakened) received signal. The distance to the object responsible for the echo could be deduced from the two-way travel time and known speed of sound. The advances in scientific knowledge leading to these technologies are traced from the discoveries of the compressibility of water (1762, John Canton), thermionic emission (1853, Edmond Becquerel), and piezoelectricity (1880, Jacques and Pierre Curie).

9:40

3aUWa2. Sonar science and technology in Russia in the 20th century. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Russian underwater acoustics traces its roots to the 19th century empirical sound propagation studies on the *Sea Devil* submarine and theoretical predictions of guided propagation in shallow water. During the World War II, the acute needs to save lives and contribute to the war effort led to rapid expansion of acoustic research and development, especially in mine countermeasures. The growth continued in the post-war years until the USSR collapse and was inspired by the opportunities for naval and civilian applications, which had been opened up by the discovery of the SOFAR channel and deeper understanding of the ocean physics. This paper will briefly review some milestones of underwater acoustic research and development in Russia, from Mikhail Lomonosov to Leonid Brekhovskikh and from Admiral Makarov's current velocity probe to sonars on early nuclear-powered submarines. Applications of the sonar to improve navigation and characterize the underwater environment will be emphasized.

10:00

3aUWa3. Fessenden and Boyle: Two Canadian Sonar Pioneers. Harold M. Merklinger (Defence R & D Canada – Atlantic (retired), Dartmouth, NS, Canada) and Dale D. Ellis (Oceanogr., Dalhousie Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com)

Reginald Fessenden developed voice-modulated radio in the early 20th century using special alternators and microphones capable of handling kilowatts of power. When the *Titanic* struck the iceberg in 1912, Fessenden, then in Boston, developed an audio-frequency acoustic transmitter-receiver, which he used to detect an iceberg at two miles. When the "Great War" began, UK interests turned to detecting submarines. Ernest Rutherford, on behalf of the Bureau of Investigation and Research (BIR), invited former students A. B. Wood, and R. W. Boyle, to join the anti-submarine effort. Fessenden was also consulted. The team detected submarines using Fessenden's device but experienced difficulty determining target bearing. After consulting Paul Langevin, Boyle agreed that quartz transducers operating at ultrasonic frequencies should provide a solution. Boyle then developed the UK Type 112 "ASDIC" which was being fitted to Royal Navy warships just as the War ended. Boyle returned to the University of Alberta after the war and continued work in ultrasonics. In 1929 he was appointed Director of Physics at Canada's National Research Council. There he established an Acoustics Section, and during World War II started a scientific activity in Halifax, NS, which later became the Naval Research Establishment.

10:20

3aUWa4. Sonar technology and underwater warfare from World War I to the launch of USS *Nautilus* in 1954, the first nuclear submarine. Willem D. Hackmann (Linacre College, Oxford OX1 3JA, United Kingdom, magicwdh@aol.com)

Sonar research began in the First World War to curb the U-boat menace. Radical methods had to be devised both organizationally and technically to combat this new form of warfare. Civilian scientists were drafted into naval research. A new field of science was created: underwater acoustics for passive listening, and ultrasonic echo-ranging for active detection of submarines. If the war had lasted only a few more months, prototype Asidic equipment (the Royal Navy's name for sonar) would have gone operational. During the interwar years the American and Royal Navies developed their sonar "searchlight" systems, while before 1935 the German Navy concentrated on sophisticated passive listening arrays (incorporating American WWI research) to protect their capital ships from enemy submarines. Searchlight sonar technology evolved sharply in WWII. The nuclear submarine in 1954 required a complete rethink of the sonar scanning techniques developed over the previous 40 years. This lecture is based on full access to naval documents and interviews with scientists involved in this work in Britain and the USA for my book *Seek and Strike* (HMSO, 1984).

10:40–11:00 Break

11:00

3aUWa5. German Navy sonar development during the two world wars and interwar years. Willem D. Hackmann (Linacre College, Oxford OX1 3JA, United Kingdom, magicwdh@aol.com)

In the German Navy, as in the UK and America, the development of underwater acoustical detection was shaped by institutional, political, and technical constraints and in response to tactical events. German hydrophone development in WWI was less advanced than that of the Allies, who focussed on combating an all-out U-boat war. The German Navy did not develop the variety of hydrophones developed by the Allies in particular the towed hydrophone array of the American Navy, which nevertheless inspired their passive sonar arrays known as "Gruppenhorchgerät" (GHG), German for "group listening device" in the interwar years, developed for the long-range protection of their capital ships, until 1935 with the signing of the Anglo German Naval Agreement when they commenced their submarine building program. For strategic reasons, Germany continued to concentrate on pro-submarine research so that at the start of WWII the German Navy had developed sophisticated GHG systems available which they improved further into the "Balkon" system of the later high-speed U-boats, Type XXI and XXVI. Their operational echo-ranging Asdic equipment was tactically inferior to that of the Royal Navy, and some interesting prototypes in development did not reach active service.

11:20

3aUWa6. Profiles in sonar: Reflections on F. V. Hunt. Anthony I. Eller (OASIS, Inc., 1927 Byrd Rd., Vienna, VA 22182, ellera@oasislex.com) and Robert W. Pyle (11 Holworthy Pl, Cambridge, MA)

Frederick Vinton Hunt was one of the towering figures who helped to shepherd the development of SONAR, an acronym for Sound Navigation and Ranging, from its infancy to near maturity. We two authors of this paper had the great advantage and pleasure of having Ted Hunt as a mentor during the latter part of his career and the early stages of our own. In this presentation we outline briefly the history of SONARs development as a means of placing Hunt's contributions in context. This would include largely his time at Harvard during the R&D phase of SONAR's growth, followed by his role during the 25 years following WWII as educator. Ideas are discussed related to what features of his mentorship are responsible for creating the next generation of leaders in science and technology.

11:40

3aUWa7. Overview of Chinese underwater acoustics. Lianghao Guo and Zhenglin Li (State Key Lab. of Acoust., Inst. of Acoust., CAS, China, No. 21 North 4th Ring Rd., Haidian District, Beijing, Beijing 100190, China, glh2002@mail.ioa.ac.cn)

French physicist Paul Langevin was the pioneer of active sonar [Ainslie, <Principles of Sonar Performance Modeling>, p.10, Springer, 2010]. His graduate student and later collaborator, Dezhao Wang, founded the first Chinese underwater acoustics laboratory at the Chinese Academy of Sciences (CAS) in 1958. Shie Yang founded the first department of underwater acoustics engineering of China at Harbin Engineering University in 1959. In accordance with the development strategy of "from the near to the distant; from the shallow to the deep," Chinese underwater acoustics research has been active in shallow-water acoustics, and has increased its efforts in deep sea acoustics in recent years. The Chinese underwater acoustics community has nurtured nine of CAS/CAE(Chinese Academy of

Engineering) academicians and several world outstanding scholars in shallow-water acoustics. Under the support from the CAS and US ONR, two comprehensive China-US joint sea-going experiments were conducted; many international conferences in ocean acoustics were held in China. This paper briefly introduces the history of Chinese underwater acoustics research and overviews some underwater acoustics progresses that have publicly been released in China, including basic research in ocean acoustics, signal processing, and underwater acoustic transducers. [Work supported by the National Natural Science Foundation of China under Grant Nos. 61571436, 11434012, and 41561144006.]

TUESDAY MORNING, 27 JUNE 2017 ROOM 306, 9:20 A.M. TO 12:20 P.M.

Session 3aUWb

Underwater Acoustics: Sound Propagation and Scattering in Three-Dimensional Environments III

Ying-Tsong Lin, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Frédéric Sturm, Cochair
Acoustics, LMFA, Centre Acoustique, Ecole Centrale de Lyon, 36, Avenue Guy de Collongue, Ecully 69134, France

Invited Paper

9:20

3aUWb1. Sound speed profile measurement uncertainties due to rough interfaces: A parametric study using the Langston-Kirchhoff model. Samuel Pinson (SHOM, Appl. Sci. Bldg., Rm. 202a, State College, Pennsylvania 16802, samuelpinson@yahoo.fr)

In the context of sound-speed profile measurement by the image source method, interface roughnesses are responsible for result uncertainties. The image source method models the reflected wave from a layered media as a collection of images which are the mirror reflections of the real source over the interfaces. From image source positions, one can deduce the sound-speed profile. Interface roughnesses might blur these image sources and reduce the accuracy of their localization. Using the Langston-Kirchhoff model of a 3D layered media with rough interfaces, it is possible to perform a parametric study of these uncertainties as a function of roughness parameters. In the aim of performing roughness parameter inversion, theoretical uncertainties are calculated and compared with estimated uncertainties from numerical experiments.

Contributed Papers

9:40

3aUWb2. Three-dimensional acoustic scattering in highly geometrically constrained environments. Irena Lucifredi (SOFAR Acoust., 44 Garfield Ave. #2, Woburn, MA 01801, euler001@yahoo.com) and Raymond J. Nagem (Boston Univ., Boston, MA)

For active sonar systems, transmission Loss (TL) and reverberation level (RL) are key parameters derived from the acoustic fields predicted by models and used to assess sonar performance. The existing propagation and scattering models may be appropriate for applications in the deep ocean or open littorals, but sonar operators are increasingly being asked to perform tasks including navigation or detection in significantly more confined waterways such as rivers or ports. Physics-based models are generally not available for predicting the acoustic field in such highly geometrically constrained and dynamic 3D environments often characterized by highly variable azimuthal boundaries such as irregularly shaped port geometries, piers, and breakwaters that may also have high large tidally driven depth variations over short time periods. They also may be populated with large scattering objects such as deep draft vessels and mooring dolphins. A virtual source research model capable of predicting the three-dimensional field, including propagation, scattering, and reverberation in such complex underwater environments is presented here. A complex, variable geometry, harbor scenario containing a large scattering object such as a vessel hull has been modeled in order represent the resultant acoustic field and to investigate scattering mechanisms present. [Work supported by ONR.]

10:00

3aUWb3. Scattering from objects buried in sandy sediments using a fully scattered field finite element approach. Anthony L. Bonomo and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The finite element method is utilized to study the scattering response from objects buried in a homogeneous sand half-space. An approach is adopted where the total field is separated into contributions due to the incident, reflected, transmitted fields in the absence of the buried object and the scattered field due to the presence of the buried object. Since the incident, reflected, and transmitted fields can be determined analytically for the case of a flat water-sediment interface, only the scattered field needs to be solved for numerically. This approach results in much faster run times while reducing spurious reflections from the boundaries of the computational domain. The buried object and the sediment half-space can be treated as fluid, viscoelastic, or poroelastic. [Work supported by ONR, Ocean Acoustics.]

10:20

3aUWb4. Application of smoothed finite element method to acoustic scattering from underwater elastic objects. Yingbin Chai (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei, China), Zhixiong Gong (Dept. of Phys. and Astronomy, Washington State Univ., Webster Physical Sci. 754, Pullman, WA 99164-2814, zhixiong.gong@wsu.edu), Wei Li (Hubei Key Lab. of Naval Architecture and Ocean Eng. HydroDynam., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei, China), and Tianyun Li (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei, China)

In this work, the smoothed finite element method (S-FEM) is employed to solve the acoustic scattering from underwater elastic objects. The S-FEM, which can be regarded as a combination of the standard finite element method (FEM) and the gradient smoothing technique (GST) from the meshless methods, was initially proposed for solid mechanics problems and has been demonstrated to possess several superior properties. In the S-FEM, the smoothed gradient fields are acquired by performing the GST over the obtained smoothing domains. Due to the proper softening effects provided by the gradient smoothing operations, the original "overly-stiff" FEM model is softened and the present S-FEM possesses a relatively appropriate stiffness of the continuous system. Therefore, the quality of the numerical results can be significantly improved. The numerical results from several typical numerical examples demonstrate that the S-FEM is quite effective to handle acoustic scattering from underwater elastic objects and can provide more accurate numerical results than the standard FEM.

10:40

3aUWb5. Kirchhoff approximation for scattering from solid spheres at flat interfaces: Improved description at large grazing angles. Aaron M. Gunderson and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, aaron.gunderson01@gmail.com)

The Kirchhoff approximation has previously been used to model scattering from partially exposed elastic spheres breaking through a flat interface, with variable exposure level, grazing angle, and frequency [J. Acoust. Soc. Am. 136, 2087 (2014)]. The limits of the Kirchhoff integral are determined by the boundaries of illumination on the sphere for each scattering path. Recent adaptations to the methods of boundary determination within the Kirchhoff approximation have yielded faster numerical integration algorithms and higher similarity of results when compared to experimental scattering data and the exact solution at half exposure [J. Acoust. Soc. Am. 140, 3582-3592 (2016)]. Additional steps have been taken to account for the partial blocking of the interface by the partially exposed sphere, through inclusion of a new correction term. This correction is largest at high grazing angles, low frequencies, and low target exposures, and drastically improves the Kirchhoff approximation within these limits. [Work supported by ONR.]

11:00

3aUWb6. Bistatic scattering from underwater elastic spheres and cylinders: Interference and resonance phenomena. Aaron M. Gunderson (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, aaron.gunderson01@gmail.com), Aubrey L. Espana (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA)

Scattering by underwater elastic spheres and cylinders is considered over a full range of scattering angles. Three models are presented in the frequency domain: an exact solution for scattering from an elastic sphere, a finite cylinder approximate solution, and a finite element simulation for the finite cylinder. Close agreement between the two cylinder models speaks to the strength of the finite cylinder approximate solution. Within each model, the dependence of the scattering on frequency and angle is discussed. Resonance structure is highlighted, and families of ridges and valleys in the data can be described by interference models between near-side and far-side Rayleigh paths and the specular reflection. A thorough understanding of bistatic scattering from elastic targets is helpful for data acquisition from moving sources (such as an AUV) or acoustic arrays, as well as studying monostatic scattering from targets at an interface, where the interface may direct bistatic paths to the receiver. [Work supported by ONR.]

11:20

3aUWb7. Modal analysis of split-step Fourier parabolic equation solutions in the presence of rough surface scattering. Mustafa Aslan (Turkish Naval Acad., Postane Mah. Rauf Orbay Cad. No:290, Istanbul 34940, Turkey, maslan@dho.edu.tr) and Kevin B. Smith (Naval Postgrad. School, Monterey, CA)

Determining accurate solutions of acoustic propagation in the presence of rough surface scattering is of significant interest to the research community. As such, there are various approaches to model the effects of rough surface scattering. Typically, the models used assume an idealized pressure release condition at the sea surface boundary. This boundary condition can easily be accommodated through a variety of modeling techniques for flat surfaces, but it is necessary to introduce additional complex methods in numerical models for rough surfaces. Parabolic equation (PE) models utilizing split-step Fourier (SSF) algorithms have been employed to treat the rough surface displacements. These include methods such as the field transformational technique (FFT), and direct modeling of the physical water/air interface discontinuity. Previous work highlighted phase errors in SSF-based models with large density discontinuities at the interfaces, which were minimized by employing a hybrid split-step Fourier/finite-difference approach. However, such phase errors were largely absent in the presence of rough surface scattering. In this work, the PE solutions are decomposed into normal modes in order to determine which modes dominate the phase error in the presence of flat surfaces, and to confirm that these modes are highly scattered in the presence of rough surfaces.

11:40

3aUWb8. Creating synthetic aperture images using experimental and simulated backscattering of solid elastic cubes. Viktor Bollen, Timothy Daniel, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu)

To study aspect and material dependent backscattering of objects, we used cubes as their geometry has distinct exposure points for different backscattering mechanisms. We insonified two solid cubes, one brass and the other steel, in underwater tank scale experiments, measuring their backscattering using circular, linear, and cylindrical synthetic aperture sonar. The right angles of the edges allow for enhanced backscattering by Rayleigh waves, which couple in at material-specific angles, since the waves are retroreflected by the cube's edges [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. 105, 700-710 (1998)]. In concert with the experiments, we model the backscattering from our cubes using Kirchhoff-Integration based simulations. The simulation isolates the specular responses, simplifying the complex responses from the cube's geometry and allowing us to identify specific effects, such as splitting from the corner reflections of a cube when its top ridge is tilted. Using Fourier based back-projection algorithms, we reconstructed target images from both the experimental and simulation results. Three dimensional images were also recreated from the cylindrical data. Using a combination of experimental and simulation results, we identified aspect dependent mechanisms. [Work supported by ONR.]

12:00

3aUWb9. Effect of nonlinear internal wave on monostatic reverberation in the shallow water region with underwater sound channel. Jungyong Park (Seoul National Univ., Bldg. 36, Rm. 212, Seoul National University, 1, Gwanak-ro, Gwanak-gu, Seoul 151 - 744, South Korea, ioflizard@snu.ac.kr), Youngmin Choo (Defense System Eng., Sejong Univ., Seoul, South Korea), Woojae Seong, and Sangkyum An (Seoul National Univ., Seoul, South Korea)

The effect of nonlinear internal wave on reverberation is investigated for the shallow water region having an underwater sound channel, which was observed during SAVEX15 experiment. When source is located near the channel axis, the reverberation from rough bottom is insignificant because of trapped modes not interacting with the bottom. However, when a nonlinear internal wave is present, the reverberation level increases since trapped modes transfer to bottom interacting modes due to sound fluctuation by the nonlinear internal wave. This trend can be explained by the coupled mode equation [(2014). J. Acoust. Soc. Am. **135**, 610-625]. To theoretically describe the increase of bottom reverberation, we simplify the situation

where two modes are present; one is a trapped mode and the other is bottom interacting mode. In the situation, the trapped mode transfers to the bottom interacting mode after its encounter with the nonlinear internal wave. The bottom reverberation has different patterns and levels before and after the internal wave, and thus, it highly depends on the distance between source/receiver and internal wave.

TUESDAY MORNING, 26 JUNE 2017 EXHIBIT HALL D, 9:00 A.M. TO 12:00 P.M.

Exhibit

The instrument and equipment exhibit is located near the registration area in Exhibit Hall D.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Exhibit hours are Sunday, 25 June, 5:30 p.m. to 7:00 p.m., Monday, 26 June, 9:00 a.m. to 5:00 p.m., and Tuesday, 27 June, 9:00 a.m. to 12:00 noon.

Coffee breaks on Monday and Tuesday mornings, as well as an afternoon break on Monday, will be held in the exhibit area.

TUESDAY AFTERNOON, 27 JUNE 2017 ROOM 206, 1:15 P.M. TO 3:20 P.M.

Session 3pAAa

Architectural Acoustics: Retrospect on the Works of Bertram Kinzey II

Gary W. Siebein, Cochair
Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Chair's Introduction—1:15

Invited Papers

1:20

3pAAa1. Standing on the shoulders of an ET giant. Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd., Sec. 4, Taipei 10607, Taiwan, akustx@mail.ntust.edu.tw)

Professor Bertram Y. Kinzey, Jr., is the main author of the Environmental Technologies in Architecture. This valuable book was first published in 1951 and has been intellectually stimulating for Architecture students at UF and around the world. It was astonishing to read through the Preface of the book. It reveals this giant's farsighted provision on how an architect should take account of complex environmental control systems and balance them with the physiological and psychological needs of the occupants during the design process. These environmental control systems include thermal, atmosphere, and environmental control, acoustics, sanitation, lighting, electrical machinery, and power distribution, coincide with current "hot" sustainable design topics. In particular, he expresses the idea of seamless integration between the architect and his engineering consultants through entire life cycle of a project. This concept, serves as the heart of the BIM development and its implementation process. Following his insight, the current architecture design curriculum and sustainable research areas at NTUST has worked toward this holistic integration mode. As said by Newton, "If I have seen further it is by standing on the shoulders of giants"!

1:40

3pAAa2. Benefiting from Bertram Kinzey's legacy. Gary Madaras (Acoust., ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

In 1991, when I rolled into Gainesville, FL, to begin doctoral studies at the University of Florida's Department of Architecture, I did not know who Bertram Kinzey was. I did not realize that he was actually the reason that I, and a first wave of doctoral students, could study architectural acoustics in the United States. I entered an academic environment that had a solid and broad foundation in architectural acoustics ~ scale models, instrumentation, custom acquisition software, room acoustic measurement and subjective survey databases and faculty relationships throughout the university. This platform was part of Bertram Kinzey's legacy. I will review how I, and many others, benefited from it during our studies at the University of Florida and since then in our careers.

2:00

3pAAa3. Subjective listening tests: Perception and preference of simulated sound fields. Michael Ermann, Andrew Hulva (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), Tanner Upthegrove (Inst. for Creativity Arts and Technol., Virginia Tech, Blacksburg, VA), Randall J. Rehfuss (Architecture + Design, Virginia Tech, Dubllin, VA), Walter Haim, Aaron Kanapesky, Trey Mcmillon, Caroyln Park, Alexander Reardon, Jeffrey Rynes, Sam Ye (Architecture + Design, Virginia Tech, Blacksburg, VA), and Charles Nichols (Music, Virginia Tech, Blacksburg, VA)

Concert hall sound fields were simulated by architecture students and anechoic recordings were convolved to create auralizations in those simulated performance spaces. Then an architectural feature was altered digitally and a second track was auralized. College music students were recruited, tested for hearing loss, and brought to a low-reverberance room with a spatial sound array of 28 mounted speakers. They were asked to identify which of the two simulated tracks they prefer. We compared simulated performance spaces: (1) with four tiers of balconies vs with one tier of balcony; (2) with an over-stage canopy vs without a canopy; (3) with separate balcony boxes vs with a continuous balcony not fragmented by box walls; and (4) with a higher scattering coefficient vs a lower scattering coefficient. Those in the audience will be invited to judge preference between the tracks for themselves. The study will be framed by the extraordinary career arc of Bert Kinzey who engaged architecture students in the study of architectural acoustics at both Virginia Tech and at the University of Florida.

2:20

3pAAa4. Generating the spatial forms of auditoriums based on distributed sentience. Ganapathy Mahalingam (Architecture and Landscape Architecture, North Dakota State Univ., Dept. 2352, PO Box 6050, Fargo, ND 58108-6050, Ganapathy.Mahalingam@ndsu.edu)

The generation of the spatial form of an auditorium based on acoustical parameters related to Reverberance, Loudness, Clarity, Lateral Energy, and Balance was defined in the early 90s using a concept called "acoustic sculpting." The central engine in the generation of the spatial form of the auditorium was the locus of the direct and reflected paths of sound from a source to a receiver. This defined an elliptical volume with two focii (source and receiver). In the initial implementation of "acoustic sculpting" in design systems for auditoria, this elliptical locus was used to generate the spatial form of the auditorium with just one source-receiver pair. In this paper, the initial concept of "acoustical sculpting" is extended, with loci generated by multiple source-receiver pairs. The elliptical locus is shown to address the key acoustical parameters. The generation of the spatial form of an auditorium from multiple elliptical loci, which define the distributed sentience of the audience, is proposed using Boolean operations. The generation of the spatial form of the auditorium proceeds from multiple desires for performance parameters at specific spatial locations, and is a desire-driven design. This will enable design solutions such as preferred seating and a tunable auditorium.

2:40

3pAAa5. Bertram Kinzey, Jr.: Teaching across generations. Keely Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com)

Bertram Kinzey, Jr., is an educator, author, acoustician, organ builder, architect, and mentor to many in the world of architecture and architectural acoustics. This paper looks at several meaningful personal accounts experienced by the author growing up. This paper examines several of the many people he directly impacted though his professorship, research and practice, and how they have in turn impacted others. It also provides examples of a second generation of people who have benefited from those he directly impacted and how they have gone on to promulgate Bert's ideas throughout our field and the world. Mr. Kinzey has influenced an ever-expanding web of multiple generations of architects and architectural acousticians around the world.

3:00

3pAAa6. Other testimonials for the life and work of Professor Bertram Y. Kinzey, Jr. Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

This session invites testimonials from people in the audience or in the form of letters from those who cannot attend to the life, work, and influences of Professor Bertram Y. Kinzey, Jr.

Session 3pAAb

Architectural Acoustics: Architectural Acoustics and Audio: Even Better Than the Real Thing I

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Wolfgang Ahnert, Cochair
Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany

Chair's Introduction—1:15

Invited Paper

1:20

3pAAb1. Unusual architectural spaces and the challenges they present to musicians and performance artists. Thomas J. Plsek (Brass/Liberal Arts, Berklee College of Music, MS 1140 Brass, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu) and Joanne G. Rice (Mobius, Quincy, MA)

For almost 15 years, trombonist Tom Plsek and performance artist Joanne Rice have been exploring the notion of sound and visual performance in spaces not conducive to any normal practices. This presentation will focus on the acoustics of two very different spaces, the abandoned quarries in Quincy, Massachusetts, and the 808 Gallery at Boston University, a 16,000 square foot rectangular space with tall ceilings, and how practices were developed to create performances under what most would consider very challenging environments.

Contributed Paper

1:40

3pAAb2. Reverberation time analysis for nonrectangular rooms using the Monte Carlo method. Giora Rosenhouse (Acoust., SWANTECH Ltd., 9 Kidron St., Haifa 3446310, Israel, giora@swantech.co.il)

Reverberation time (RT) of a nonrectangular room involves its mean free path (MFP). Statistics of sound ray consecutive collisions of the enveloping room surfaces solve the MFP value. The classical halls RT formulae are based on physical models and assumptions, including discrepancies. Thus, we use here the Monte Carlo Method (MCM) to calculate MFP and the probabilities of sound rays colliding with their surrounding surfaces and their RT. The method presented here includes statistical analysis, sensitivity to changes of different parameters, validation and interpretation. It calculates averages of a series of lengths of individual paths for each ray emanating from a sound source and its direction cosines, chosen at random from a uniform distribution and considering range by means of pseudo random generator (PRG). The process recurs for a fixed number of rays and a computer run for an average of one block and its MFP. Each run involves an initial PRG seed. The analysis of a fixed number of paths of one ray is a result of deterministic algorithm and thus its values are dependent. Therefore, its averages constitute a population that follows the large numbers theorem, which does not necessarily follow the central limit theorem.

Invited Papers

2:00

3pAAb3. Rapid impedance tube determination of fabric transparency. Peter D'Antonio (Chesapeake Acoust. Res. Inst. LLC, 15618 Everglade Ln, #106, Bowie, MD 20716, dr.peter.dantonio@gmail.com) and Trevor J. Cox (Acoust. Eng., Univ. of Salford, Salford, United Kingdom)

Acousticians are continually being asked to verify fabric transparency for applications with absorptive and diffusive surfaces, as well as in sound reinforcement. Standard reverberation chamber methods can be used, but require large fabric and fiberglass samples. A quick and simple impedance tube method has been developed requiring only a 160 mm x 160 mm sample. Two measurements are made. One with an anechoic wedge termination and another with a rigid termination in which the fabric is applied to a 50 mm 6pcf fiberglass substrate. Four microphones are placed at a quarter and three quarters of the square tube's width and height and the signal is summed.

This unique microphone placement minimizes the 1st, 2nd, and 3rd harmonics resulting in a fourfold increase in the upper frequency limit. Impulse response measurements are made at three distances from the sample to calculate the reflection factor, impedance and normal incidence absorption coefficient from 63 Hz to 4000 Hz in one measurement. The small fabric sample is large enough to minimize non-homogeneous effects. A study of fabrics with a wide range of transparencies reveals how both acoustically transparent and backed fabrics can be used as sonic equalizers depending on the application.

2:20

3pAAb4. Comparison of methods used in design simulation and in emulation of electrical and acoustic systems for audio. Douglas Rollow (Res. and Innovation, Sennheiser, 550 15th St, Ste. 37, San Francisco, CA 94103, tad.rollow@sennheiser.com)

Contemporary audio recording and reproduction systems offer exceptional signal quality in terms of bandwidth, resolution, and linearity. Simulations used in the design and analysis of acoustic spaces, electroacoustic transducers, and electronic components are effective in predicting specific metrics of performance, but end-to-end system simulation would be exceedingly difficult even over extended run time. In media production workflows, emulation is used to create perceptually useful renderings of these same systems, with the real-time digital signal path providing emulation of acoustic environments coupled with transducers, and of nonlinear euphonic elements. These algorithms serve some of the same function as those in the design tools, and the present work uses the opportunity to compare engineering goals, economic constraints, and performance requirements.

2:40

3pAAb5. The room they want to hear: Subjective preference of reverberation in vocal recordings. Yuri Lysoivanov (Recording Arts, Tribeca Flashpoint College, 28 N. Clark St. Ste. 500, Chicago, IL 60602, yuri.lysoivanov@tribecaflashpoint.edu)

Audio professionals in the digital era have hundreds of reverberation options at their disposal, from the uber-realistic to the dynamic and creative. Often the choice of reverb is made with a set of heuristics—based on the engineer's experience and opinion for what achieves the best musical and dramatic effect. In this presentation, we turn to the listener and explore their preferences of reverberation on vocal performances. Through a series of experiments, we evaluate the aesthetic preferences of the casual music consumer and compare them to those of the experienced audio engineer. In addition, we examine factors that may predispose listeners to prefer certain reverberation characteristics over others.

3:00

3pAAb6. Sound reinforcement for a divisible auditorium. Deb Britton, Kevin Hodsgon (K2 Audio, 5777 Central Ave., Ste. 225, Boulder, CO 80301, deb@k2audio.com), and Gain Foster (K2 Audio, Woodford, VA)

As part of an audiovisual renovation to a conference center, one particular space, an auditorium that can be divided into four separate spaces, posed a particular sound reinforcement challenge. This paper describes the architecture of the space, its various uses, and presents our approach to providing flexible, yet high-quality sound reinforcement.

Session 3pAAc

Architectural Acoustics: Robust Heavy and Lightweight Constructions for New-Build and Retrofit Buildings

Matthew V. Golden, Cochair
Pliteq, 616 4th Street, NE, Washington, DC 20002

Stefan Schoenwald, Cochair
Laboratory for Acoustics/Noise Control, Empa Swiss Federal Laboratories for Materials Science and Technology, Überlandstrasse 129, Dübendorf 8606, Switzerland

Lieven De Geetere, Cochair
Division Acoustics, Belgian Building Research Institute, Lombardstraat 42, Brussel 1000, Belgium

Invited Papers

1:20

3pAAc1. Predicting sound radiation and sound transmission in orthotropic cross-laminated timber panels. Andrea Santoni, Paolo Bonfiglio, Patrizio Fausti (Eng. Dept., Univ. of Ferrara, via Saragat, 1, Ferrara, FE 44122, Italy, andrea.santoni@unife.it), and Stefan Schoenwald (Lab. for Acoust. and Noise Control, Empa Swiss Federal Labs. for Material Sci. and Technol., Dübendorf, Switzerland)

In the last decades, new materials and new technologies which satisfy sustainability and energy efficiency demands have been developed for the building construction market. Lightweight structures are becoming increasingly popular, but it has been proved that they cannot provide satisfactory sound insulation. Therefore a proper acoustic treatment needs to be specifically designed, considering both airborne and structure-borne sound sources. Cross-laminated timber (CLT) elements, for example, have had great success in the last twenty years, both in Europe and North America. CLT plates, due to their peculiar sub-structure, exhibit an orthotropic behavior; they have different stiffness properties along their two principle directions. This paper investigates prediction models for orthotropic plates designed to evaluate sound radiation due to mechanical excitation, and sound transmission due to acoustic excitation. Particular attention is paid to the influence on sound radiation of non-resonant vibration, or near-field vibration, in the case of mechanical excitation. The purpose of these simplified models is to be an efficient tool for acousticians, architects and engineers, helpful in the design process for new buildings and retrofitting of existing ones. The validation of numerical results with experimental data are reported. The applicability of the models and their limitation are finally discussed.

1:40

3pAAc2. Advanced methods to determine sound power radiated from planar structures. Stefan Schoenwald (Lab. for Acoustics/Noise Control, Empa Swiss Federal Labs. for Mater. Sci. and Technol., Überlandstrasse 129, Dübendorf 8606, Switzerland, stefan.schoenwald@empa.ch), Sven Vallely (Univ. of New South Wales, Sydney, NSW, Australia), and Hans-Martin Tröbs (Lab. for Acoustics/Noise Control, Empa Swiss Federal Labs. for Mater. Sci. and Technol., Dübendorf, Switzerland)

In building acoustics, the most fundamental aim is to determine the sound power radiated by building elements. In this paper, two methods that are more sophisticated than the conventional measurement of sound pressure in a receiving room are presented and discussed. Both methods, the Discrete Calculation Method and the Integral Transform Method, require only the surface velocity measured in a grid on the radiating surface as input data. Thus the sound power is univocally associated to the considered element. The first assumes a series of radiating piston sources on the surface that move with same velocity and phase relationship as the structure. The second uses spatial Fourier Transformations to determine the radiated sound power in the wavenumber domain, analogous to Nearfield Acoustic Holography. The Integral Transform Method additionally obtains the angle dependent flexural wavenumbers of the structure, which is essential for the analysis of the element dynamics, and as input data for the prediction of sound radiation efficiency and transmission loss of orthotropic building elements, for example, Cross-Laminated-Timber elements. Both methods were applied in some exemplary cases; based on their performance and results, conclusions are drawn on the benefits of both methods.

2:00

3pAAc3. Analysis of impact sound insulation properties of wooden joist and cross laminated timber (CLT) floor constructions. Anders Homb (SINTEF Bldg. & Infrastructure, Høgskoleringen 7B, Trondheim 7465, Norway, anders.homb@sintef.no)

During the last years, there has been an increased interest of developing lightweight constructions with improved sound insulation properties compared to previous, well-known solutions. There have been a progress on prediction tools as well as new solutions, for instance, from the WoodWisdom-Net project "Silent Timber Build." Previous research establish the fact that including the frequency

range below 100 Hz is crucial to get satisfying correlation between perceived and measured impact sound insulation quantities. Research work at SINTEF Building & Infrastructure concerning lightweight floor constructions include therefore frequencies down to at least 50 Hz. The paper will present analysis of a number of laboratory measurement results from joist floor constructions and CLT floor constructions. The paper will focus on the most challenging parameter, the impact sound insulation including the spectrum adaptation term $C_{I,50-2500}$. Analysis will include parameters such as the mass of the floors, the height of the floors, and of course structural connections and properties of the resilient layers involved. Improved low frequency properties introduce of course more mass and or stiffness of the floor, but the result also show that optimization of the contributing components and material properties is necessary for the development of robust and environmental friendly solutions.

2:20

3pAAc4. New acoustic solutions for cross laminated timber based buildings. Lieven De Geetere, Bart Ingelaere, and Arne Dijckmans (Div. Acoust., Belgian Bldg. Res. Inst., Lombardstraat 42, Brussel 1000, Belgium, lieven.de.geetere@bbri.be)

Cross laminated timber is becoming a popular building material for the construction of multifamily dwellings, offices, hotels, etc. The acoustic challenges are important: in contrast to light weight timber frame constructions, it is far more sensitive to flanking transmission and its equally relatively light weight and orthotropic character in comparison with traditional heavy stone requires special solutions for the direct airborne and impact sound insulation. The paper presents a new, special, and patented building system that almost annihilates flanking transmission and will detail optimized low frequency sound insulation solutions for floors.

2:40

3pAAc5. Comparison of sound isolation performance of floor/ceiling assemblies across structural systems. Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

The recent building boom in greater Boston has given rise to multifamily buildings utilizing a wide range of structural systems, including cast-in-place concrete, pre-cast concrete, steel, heavy timber, and various wood frame structural systems. Both laboratory and field measurements indicate that typical floor/ceiling assemblies utilized with each structural system have certain characteristic acoustical attributes. Although these assemblies perform differently, virtually all of these structural systems have been utilized in buildings advertising "luxury" residential living. This presentation compares recently obtained sound isolation data among the various structural types. Specific acoustical design challenges and constraints (such as budget, available floor-to-floor height, and others) are identified, and guidelines for achieving sound isolation commensurate with a "luxury" standard are outlined for each of the structural systems examined.

3:00

3pAAc6. Achieving high sound isolation between rooms with stone wool ceilings and plenum barriers when the ceiling grid runs continuously over partial height demising walls. Gary Madaras (Acoust., ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com) and Andrew Heuer (Acoust., NGC Testing Services, Buffalo, NY)

The Optimized Acoustics Research Program is a multi-year investigation being conducted by ROCKFON, ROXUL, and NGC Testing Services into efficient and economical means of constructing interior architecture that complies with the higher sound absorption and higher sound isolation criteria in standards, guidelines, and building rating systems. Prior program updates have included the negative effects of noise flanking paths through ceiling light fixtures and air distribution devices (Inter-Noise 2015) and optimizing the combination of absorptive, stone wool, ceilings and lightweight plenum barriers to achieve sound transmission class (STC) ratings of 40, 45, and 50 without full height demising walls (Noise-Con 2016). This update addresses a worst-case scenario, when the suspended ceiling grid runs continuously over partial height demising walls. Building the walls to the underside of the ceiling grid avoids the need to reconstruct the ceiling if the walls are relocated in the future. However, this approach has historically not complied with the standards and results in poor acoustic performance. The current research shows how to install standard 16 mm (5/8 in.) thick, stone wool, modular, acoustic ceilings and 38 mm (1.5 in.) thick stone wool plenum barriers to achieve ratings up to STC 52.

Session 3pAB

Animal Bioacoustics: Comparative Bioacoustics: Session in Honor of Robert Dooling II

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

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Contributed Papers

1:20

3pAB1. Environmental vocalization adaptation: Animals compensating for restricted visibility and mobility. David Browning (Browning Biotech, 139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter Herstein (Browning Biotech, Westerly, RI)

To survive, animals must always be aware of their surroundings and they adopt different communication strategies to adapt to changing environmental conditions. This paper describes some vocalization adaptations. Horses in open pasture usually communicate visually but when placed in separate barn stalls, with restricted visibility and mobility, must rely more on their vocalizations, referred to as barntalk. In addition they rapidly learn to recognize sounds from unseen objects of interest such as a feed cart or their owner. In some cases animals utilize a new vocalization, Asiatic wild dogs (Dholes) have developed a short whistle to keep contact among their hunting pack when visibility is reduced by underbrush, while not interfering significantly with listening for their prey. Nowhere is there more vocalization adaptation than in a jungle, with limited visibility, difficult mobility, and the added complication of many other competing sounds. As a result Sumartran rhinos have developed a far more complex vocalization than their African relatives on the open plains. An interesting variety of jungle vocalization strategies have developed, from the Tapirs sweeping whistles to the toned howls of New Guinea singing dogs.

1:40

3pAB2. Laboratory mice can behaviorally discriminate between natural and synthetic ultrasonic vocalizations. Anastasiya Kobrina, Laurel A. Screven (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu), Elena J. Mahrt (Biology, Washington State Univ., Pullman, WA), Micheal L. Dent (Psych., SUNY Univ. at Buffalo, Buffalo, NY), and Christine Portfors (Biology, Washington State Univ., Vancouver, WA)

Mice produce spectrotemporally complex ultrasonic vocalizations (USVs), thought to be important for social interactions such as mating. Previous research has established that mice are capable of detecting and discriminating natural, synthetic, and altered USVs using behavioral methodologies. The current study examined whether mice are capable of discriminating natural USVs from their synthetic USV analogs. Discrimination performance was tested in five adult mice using operant conditioning procedures with positive reinforcement. Mice were trained to nose poke to one hole during a repeating natural or synthetic USV, which would begin a trial. Subjects were required to poke to a second hole when they discriminated a change in the repeating background after a variable interval. The target stimuli were natural and synthetic versions of the same USVs, as well as other USVs. Mice can discriminate between some natural USVs and their synthetic renditions but not others. Discrimination performance across all stimuli was correlated with spectrotemporal similarity. Mice utilized duration, bandwidth, and peak frequency differently for natural and synthetic USV discrimination. These results contribute to our understanding of the ways USVs may be used for acoustic communication in mice. [This work was supported by NIDCD grant R01-DC012302 to MLD.]

2:00

3pAB3. Relative salience of syllable order versus syllable fine structure in Zebra Finch song. Shelby Lawson (Psych., Univ. of Maryland, 112 Beverley Ave., Edgewater, MD 21037, sllawson@smcm.edu), Adam Fishbein (NACS, Univ. of Maryland, College Park, MD), Nora H. Prior (Biology, Univ. of Maryland, College Park, MD), Bernard Lohr (Univ. of Maryland Baltimore County, Baltimore, MD), Gregory F. Ball, and Robert Dooling (Psych., Univ. of Maryland, College Park, MD)

Zebra finches have become a popular model for the investigation of the motor and perceptual mechanisms underlying vocal learning. These birds are closed-ended learners that have a brief sensitive period for song learning, after which a new song cannot be learned. This song is a single, highly stereotyped, invariant, sequence of 3-8 harmonic syllables, termed a motif, which is repeated several times throughout the song bout. Here, using an operant conditioning discrimination task, we confirm earlier results that these birds find syllable reversals (i.e., changes in temporal fine structure) highly salient. By contrast, finches find syllable ordering in these natural motifs much less salient. The ability to extract and learn sound patterns is a common feature of animal bioacoustic systems including, of course, human speech and language learning. We know from both single unit recordings in the auditory forebrain as well as operant studies, that these birds encode the syllable ordering in song-like stimuli and can learn to discriminate differences in syllable ordering. The large difference in salience between these two sets of acoustic features in these natural motifs—syllable structure versus syllable ordering—raises significant questions about how these natural sounds are processed in the CNS.

2:20

3pAB4. Social experience influences ultrasonic vocalization perception in laboratory mice. Laurel A. Screven and Micheal L. Dent (Psych., Univ. at Buffalo, B29 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu)

Mice emit ultrasonic vocalizations (USVs) which vary in spectrotemporal parameters (e.g., frequency, amplitude, and duration) in a multitude of social situations. USVs are often assumed to possess speech-like characteristics, although it has not yet been established that mice are using USVs for communication purposes. Previous studies have shown changes in auditory cortex activity in maternal females to pup calls, but it is currently unknown how previous social experience with other mice throughout development affects perception of adult vocalizations. To test the effect of socialization, we used an operant conditioning task to determine if discrimination of USVs was negatively impacted by chronic social isolation compared to mice that were group housed throughout their lifespan. Mice discriminated between twelve USVs of three different categories. Mice that had been

socially isolated since weaning showed deficits in discrimination of some USVs. Additionally, socially isolated mice required more training and testing, and more trials to complete the task than socially experienced mice. These results indicate that experiencing USVs during social interactions affects how the mice perceive their vocalizations, suggesting that these vocalizations could have context-specific meaning that is learned through hearing USVs within the appropriate social context.

2:40

3pAB5. How canaries listen to their song. Adam Fishbein (Neurosci. & Cognit. Sci., Univ. of Maryland, 4123 Biology-Psych. Bldg., College Park, MD 20742, afishbei@umd.edu), Shelby Lawson (Psych., Univ. of Maryland, Edgewater, MD), Gregory F. Ball, and Robert Dooling (Psych., Univ. of Maryland, College Park, MD)

Canaries have become important models for the study of vocal learning. The male produces long song bouts up to a minute long consisting of various syllables, each repeated in flexibly sequenced phrases. Little is known about how the birds listen to song though behavioral observations clearly show that female canaries are more sexually responsive to a special song element—so called "sexy" syllables—which are distinguished by a high syllable repetition rate, wide-bandwidth, and multiple notes. Here, operant conditioning and psychophysical techniques were used to determine the discriminability of variation in syllable and phrase morphology. Results show that canaries can discriminate the subtle differences among syllables and phrases using spectral, envelope, and temporal fine structure cues but they are no better than budgerigars used as controls. There was also evidence that perception of sexy syllables is distinctive for canaries. On the whole, while canaries can hear the fine details of the acoustic structure of their song, the evidence suggests that they listen at a more synthetic rather than analytic level. These results give clues to how the canary perceptual system may be shaped to process male song and make judgments about mate quality.

TUESDAY AFTERNOON, 27 JUNE 2017 ROOM 310, 1:20 P.M. TO 3:40 P.M.

Session 3pAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Measurements of Sediment Transport and Near-Bottom Structures II

James Lynch, Cochair

Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543

Peter D. Thorne, Cochair

Marine Physics and Ocean Climate, National Oceanography Centre, National Oceanography Centre, Joesph Proudman Building, 6 Brownlow Street, Liverpool L3 5DA, United Kingdom

Contributed Papers

1:20

3pAO1. Combining echo and hydrodynamic measurements for estimating non uniform sand transport under waves and currents. Rodrigo Mosquera and Francisco Pedocchi (IMFIA, Universidad de la República - Facultad de Ingeniería, Julio Herrera y Reissig 565, Montevideo, Montevideo 11300, Uruguay, rmosquer@fing.edu.uy)

We present recently obtained measurements in the Atlantic coast of Uruguay, with a 1 MHz ADCP (Sentinel V20, Teledyne RDI, USA). The ADCP was deployed at 16 m depth for 4 months using a light weight structure deployed from a small boat. During the deployment, 3.6 m significant high, 12 s peak period waves, and 0.8 m/s currents were recorded. The vertical distribution of suspended sediment in this environment is controlled by the turbulence level induced by both currents and waves. Assuming a representative sediment size and adapting the Rouse-Vanoni profile showed to give results that did not agree with the concentration profiles inverted from the recorded echo profiles. Therefore, a method that combined variations of the sediment size distribution with the turbulence level was developed showing very good agreement with the information from the echo profile. In the developed method, the turbulent sediment diffusivity was computed from the mean current and wave measurements. However, this showed limitations for conditions with strong waves and weak currents. As an alternative, direct measurements of the turbulent fluctuations from the instantaneous beam velocity recordings were used to compute the turbulent diffusivity. In addition, different turbulent diffusivity distributions for combined current-wave conditions were evaluated.

1:40

3pAO2. Sound speed and attenuation in water-saturated granular materials at MHz frequencies. Jenna Hare and Alex E. Hay (Oceanogr., Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H4R2, Canada, jenna.hare@dal.ca)

Sound speed and attenuation measurements are reported for water-saturated granular materials (natural sand and glass beads) at frequencies of 1.0 to 1.8 MHz. Median grain diameters were 0.219 to 0.497 mm, corresponding to kd>1, i.e., the scattering regime. The measurements were made for different thicknesses of sediment resting on a reflective surface using a monostatic geometry. The attenuation estimates compare well with previously reported experimental results and to the predictions of multiple scattering theory. The sound speed estimates exhibit the negative dispersion predicted by theory, but compared to previous measurements are biased low by 100 m/s to 200 m/s. It is argued that this bias is due to microbubbles in concentrations of O(10) ppm by volume.

2:00

3pAO3. Ambient noise from turbidity currents. Matthew Hatcher, Alex E. Hay (Oceanogr. Dept., Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H 4R2, Canada, mhatcher@dal.ca), and John E. Hughes Clarke (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Sediment mass transport from the Squamish River delta into the adjacent fjord (Howe Sound, British Columbia) is dominated by discrete turbidity current events which have incised semi-permanent channels on the delta front and out onto the prodelta. Acoustic data were collected in the spring of 2013, including both active and passive systems. Data from the active sonars are used to determine flow speed, flow thickness and suspended sediment concentration. The noise generated by these discrete turbidity currents is broadband (10 to 200 kHz) and, based on the sediment grain size and flow speeds, is shown to be due to sediment-generated noise, most likely at the base of the flows.

2:20

3pAO4. Suspended sediment flux statistics in unidirectional flow, using a pulse-coherent acoustic Doppler profiler. Greg Wilson (Oregon State Univ., 104 CEOAS Administration Bldg., Corvallis, OR 97331-5503, wilsongr@coas.oregonstate.edu) and Alex E. Hay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

An experiment was conducted in the St. Anthony Falls Lab main channel flume, involving sand sediment (d50 = 0.4 mm) in unidirectional flow. The flow conditions were 1 m/s flow rate and 1 m depth, and the bed state consisted of quasi-linear sand dunes with ~1 m wavelength and 10-20 cm height. A pulse-coherent acoustic Doppler instrument (MFDop) measured high-resolution near-bed vertical profiles of velocity and backscatter amplitude at various positions spanning the dune profile. These measurements are used to obtain probability distributions of instantaneous suspended sediment concentration and flux, which are compared to predictions from two stochastic theories. While the stochastic theories were previously developed and validated for low transport rates over a flat bed, the acoustic measurements enabled measurements at a much higher transport rate; despite this, the new measurements remain in remarkably good agreement with the theories.

2:40

3pAO5. Transient reflection from mud during molecular diffusion of salt. Gabriel R. Venegas and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., Univ. Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu)

Harbor basins and estuarine environments experience drastic salinity fluctuations in the water near the water-sediment interface, which can significantly affect how sound interacts with a low-velocity mud bottom. This presents challenges in applications including mine detection, port protection and shallow water sonar. In a previous investigation of this system, a mud sample that was saturated with fresh water was instantaneously exposed to salt water. Laboratory measurements of plane wave reflection from the water-mud interface were obtained using a time-gated broadband impulse as time evolved. Results suggested molecular diffusion of salt into the sample had altered the reflected pulse's amplitude and caused a depth-dependent impedance profile in the mud. The sediment was discretized into layers much thinner than a wavelength and a multi-layer steady-state model was used to predict the reflection from the diffusing mud. As the effective diffusion length reached a critical value, predictions of the steady-state model began to deviate from the measurements, indicating that a transient solution was required. A model of the transient reflection of a finite-length pulse from a half space with an arbitrary impedance will be presented and model results compared with laboratory measurements. [Work supported by ONR.]

3:00

3pAO6. Influence of sand/silt particle size distributions on compressional wave attenuation in marine mud sediments. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Boston Univ., East Sandwich, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

High porosity marine mud from different sites typically contains different amounts of clay, sand, and silt particles, along with other material. A recent talk [Pierce *et al.*, ASA Honolulu, 5aAO1 (2016)] explored a mechanism for why sand and silt particles in suspension can provide the dominant contributions to the frequency dependence of compressional wave attenuation. The card-house structure of the clay is critical in supporting the particles and keeping them separated. Example calculations for spherical particles of the same size showed physically reasonable attenuation behavior at low and high frequencies. This presentation considers extensions of the approach, particularly accounting for distributions of particle sizes, and emphasizes comparisons of attenuation predictions with available field data. Using reasonable assumptions about the clay volume and the sand and silt distributions, it is possible to estimate the numbers of sand and silt particles, and consequently the attenuation, from the sediment porosity. Such results can be compared with measurements of attenuation frequency dependence for measured porosities, in order to validate or refine the model for the mechanism. Of specific interest is determination of the frequency bands over which the attenuation increases nearly linearly with frequency, as is often estimated or assumed. [Work supported by ONR.]

3:20

3pAO7. Role of clay particle electrostatics and the dielectric permittivity of suspended silt particles in sound attenuation in mud. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Sound attenuation in marine mud sediments is partly caused by viscous dissipation of acoustically induced flow past suspended silt particles. Clay particles in the surrounding lattice carry electrostatic charges, causing high porosity, so one asks why silt particles do not settle because of gravity to the bottom of the mud layer. Explanation of the suspension and the associated attenuation of sound proceeds from consideration of a quartz sphere immersed in mud. The somewhat-random electric field created by the clay particles causes an electric dipole moment to arise in the sphere because of its dielectric permittivity. This is proportional to the electric field and varies with position, and the result is an electrostatic force on the sphere, the force being proportional to the gradient of the electric field. In equilibrium, this force is balanced by a gravity force. There is a natural spring constant associated with deviations from equilibrium, and the resulting dynamical model is a fixed-mass sphere subjected to a spring force, the force of gravity, and the viscosity-associated force caused by the motion of the surrounding fluid and the no-slip condition at the sphere's surface. Paper quantitatively discusses the model's implications on the suspension theory of sound attenuation. Results suggest approximate validity for representative acoustic frequencies of model where sphere has same density as water. [Work supported by ONR.]

Session 3pBAa

Biomedical Acoustics: Advances in Shock Wave Lithotripsy II

Robin Cleveland, Cochair
Engineering Science, University of Oxford, Inst. Biomedical Engineering, Old Road Campus Research Building, Oxford OX3 7DQ, United Kingdom

Adam D. Maxwell, Cochair
University of Washington, Seattle, WA 98105

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Pennsylvania State University, Penn State, 201E Applied Sciences Building, University Park, PA 16802

Contributed Papers

1:20

3pBAa1. Assessing the effect of lithotripter focal width on the fracture potential of stones in shockwave lithotripsy. Shunxiang Cao (Dept. of Aerosp. and Ocean Eng., Virginia Tech, Rm. 332, Randolph Hall, 460 Old Turner St., Blacksburg, VA 24060, csxtovt@vt.edu), Ying Zhang, Defei Liao, Pei Zhong (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), and Kevin G. Wang (Dept. of Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA)

This talk presents a combination of computational and experimental study on the effect of lithotripter focal width on the fracture potential of stones treated at various distances from the lithotripter focus. Two representative lithotripter fields are considered: (1) the original Siemens Modularis with a focal width of 7.4 mm and (2) a modified version with a larger focal width of 11.0 mm with comparable acoustic pulse energy of 40 mJ. The interaction of these two lithotripter fields with spherical and cylindrical model stones located at 0 to 12 mm from the shock wave axis is investigated. Specifically, a three-dimensional CFD (Computational Fluid Dynamics)—CSD (Computational Solid Dynamics) coupled computational framework is used to simulate the propagation of stress waves, as well as the initiation and propagation of fractures. The two-scale Tuler-Butcher fracture criterion will be employed: it will be calibrated experimentally, then applied to assessing stress-induced stone damage. An element deletion method will be applied to simulate fracture. Characteristic changes in wave focusing and interference in relation to the buildup of the maximum tensile stress inside the stone will be presented. The physical mechanism(s) responsible for the different fracture patterns observed at different off-axis distances will be discussed.

1:40

3pBAa2. A multi-element HIFU array system: Characterization and testing for manipulation of kidney stones. Mohamed A. Ghanem (Aeronautics and Astronautics Eng., Univ. of Washington, 123529 4th Ave. ne, Seattle, WA 98125, mghanem@uw.edu), Michael Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Bryan Cunitz, Wayne Kreider, Christopher Hunter (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle WA and Phys. Faculty, Moscow State Univ., Seattle, Washington), and Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle WA and Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Most progress in acoustic trapping and levitation has been achieved with the use of multiple sound sources, standing waves, and low density or very small objects. Repositioning kidney stones in the human body is an important clinical application where acoustic levitation could be employed; however, it requires manipulation of larger and heavier objects. The goal of this work was to calibrate the acoustic output of a 1.5-MHz, 256-element array designed in our laboratory for HIFU research, which is also capable of generating vortex beams to manipulate mm-sized objects in water and to move them in any direction without moving the source. Electrical circuits were developed for matching each element of the array to an output channel of a Verasonics ultrasound engine to allow for efficient power transfer to the transducer. Acoustic holography was used to calibrate and equalize outputs across all channels. Manipulation of artificial kidney stone targets made of plastic, glass, or cement (2—8 mm) and comparable in size to or larger than the wavelength in water by electronic steering of the vortex beam lateral to the acoustic axis was demonstrated. [Work supported by NIH P01 DK043881, K01 DK104854, R01 EB007643, and NSBRI through NASA NCC 9-58.]

2:00

3pBAa3. Finite element optimization of circular element used in lithotripsy and histotripsy piezoelectric transducers. Gilles Thomas, Jean-Yves Chapelon, and Cyril Lafon (U1032, INSERM, 151, cours Albert Thomas, LYON 69424, France, gilles.thomas@inserm.fr)

Multi-element piezoelectric transducers used in shock wave generators for disrupting kidney stones (lithotripsy) or ablating soft tissues (histotripsy) are emitting high amplitude burst waves of one to ten pulses. Thus, most of the signal is sent while the piezoelectric transducers vibrate in a transient state. This work aims at optimizing the design of these elements. A transient finite element model of a piezoelectric circular element is presented. It includes the piezoelectric disk, the epoxy and plastic casing, the surrounding water and the RLC discharge circuit. The model was then used for parametric optimization of the electrical components and the front and back layers. It has been validated by comparing the numerical and experimental results of one 400 kHz, 37.3 mm diameter LT02 element (EDAP-TMS). The surface pressure field as measured by a fiberoptic hydrophone was in good agreement with the simulation. Elements with parameters resulting of optimization from different objective functions, which depend of the application desired, were built and validated with experimental results. The model can be effectively used for the rapid and optimal design of piezoelectric circular elements working in a transient state. [Work supported by an industrial grant from EDAP-TMS.]

2:20

3pBAa4. Investigation of stone damage patterns and mechanisms in nano pulse lithotripsy. Chen Yang, Ying Zhang, and Pei Zhong (Duke Univ., PO BOX 90300, 144 Hudson Hall, Durham, NC 27708, cy71@duke.edu)

Nano Pulse Lithotripter (NPL) utilizes spark discharges of ~30-ns duration, released at the tip of a flexible probe under endoscopic guidance, to break up kidney stones. Different damage patterns have been observed using BegoStone samples, including crater formation underneath the probe tip, crack development from the distal wall, and radial and ring-shape crack initiation in the proximal surface of the stone. Multiple mechanisms have been proposed for stone disintegration in NPL: dielectric breakdown near the probe tip, shockwave induced by the spark discharge, and asymmetric collapse of bubbles. Experiments have been performed to correlate the proposed mechanisms with the damage patterns observed. Comparison between micro-CT images of the damage initiation sites and COMSOL simulation of the stress field in the stone indicates that the observed cracks are most likely to be produced by the locally intensified tensile stresses produced by surface acoustic waves (SAW) generated by the incidence of the spark-generated, spherically divergent shockwave on the proximal surface of the stone, and their interactions with bulk acoustic waves (P or S) upon reflection at stone boundaries. Dielectric breakdown may contribute to crater formation. However, the contribution of cavitation to stone fragmentation in NPL appears to be minimal.

2:40

3pBAa5. The acoustic field of a spherical self-focusing Eisenmenger electromagnetic shockwave source (EMSE). Abtin Jamshidi Rad and Friedrich Ueberle (Life Sci., HAW Hamburg, Ulmenliet 20, Hamburg 21033, Germany, Abtin.Rad@HAW-Hamburg.de)

Shockwave sources are used in extracorporeal lithotripsy, pain therapy and a wide range of other medical applications. Typical lithotripter source pulses mostly achieve positive pressure amplitudes of ca. 35 to 100 MPa and tensile pressures amplitudes up to -20 MPa. Already 1962, Eisenmenger designed a plane electromagnetic source, which was further developed into a spherically shaped, self-focusing electromagnetic source (ca. 1991). We have one of his prototype sources (Membrane diameter 120 mm, focal distance 200 mm), which was measured according to the standard IEC61846. Focus and field measurements were done using a single-spot fiberoptic hydrophone and compared to a multi-spot optical hydrophone. Very low variations of the acoustic output were found (for the peak positive pressure and for the energy). Notably, in contrast to many other shockwave sources, the spherical EMSE provides steep shockwaves (10...20 ns risetime) in the focus at comparably low pressures (33...36 MPa), even at lower energy settings. Peak negative pressures were in the range of $<$ -10 MPa. Focus and field measurements show the interesting properties of the spherical self-focusing EMSE, also in comparison to stronger focusing setups.

3:00

3pBAa6. Nonlinear elasticity imaging with dual frequency ultrasound. Johannes Kvam (Dept. of Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Maridalsveien 221, Oslo 0467, Norway, johannes.kvam@gmail.com), Stian Solberg, Ole Martin Brende, Ola Finneng Myhre, Alfonso Rodriguez-Molares (Dept. of Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Jørgen Kongsro (Norsvin SA, Oslo, Norway), and Bjørn A.J. Angelsen (Dept. of Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

Elasticity imaging has shown to enhance diagnostic capabilities of disease as tissue elasticity is often associated with pathological condition. Methods such as Acoustic Radiation Force Impulse imaging and Supersonic Shear Wave imaging have shown good results at shallow depths. Second-order ultrasound field (SURF) imaging is a dual band imaging technique that utilizes a low frequency (LF) pulse to manipulate the nonlinear elasticity of the medium observed by a co-propagating high frequency pulse (HF). The manipulation of the material parameters causes the HF to experience a change in propagation velocity which depends on the manipulation pressure and nonlinear elasticity of the medium. The change in propagation velocity causes an accumulative delay or advancement compared to a single HF pulse, called the nonlinear propagation delay (NPD). By transmitting multiple pulse complexes with different LF polarities the technique has proven capabilities for suppression of multiple scattering noise. By observing local variations in the development of NPD, it is possible to estimate the nonlinear elasticity variation of the medium. Simulations with k-Wave toolbox have shown the methods feasibility, also at greater depths. *In-vitro* and *in-vivo* experiments have also yielded promising results.

Session 3pBAb

Biomedical Acoustics: Partial Differential Equation Constrained and Heuristic Inverse Methods in Elastography II

Mahdi Bayat, Cochair
Biomedical and Physiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Wilkins Aquino, Cochair
Civil and Environmental Engineering, Duke University, Hudson Hall, Durham, NC 27708

Contributed Papers

1:40

3pBAb1. Shear wave phase velocity dispersion estimation in viscoelastic materials using the Multiple Signal Classification method. Matthew W. Urban (Dept. of Radiology, Mayo Clinic College of Medicine and Sci., 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Piotr Kijanka (Dept. of Robotics and Mechatronics, AGH - Univ. of Sci. and Technol., Krakow, Poland), Bo Qiang (The Nielsen Co., Oldsmar, FL), Pengfei Song (Dept. of Radiology, Mayo Clinic College of Medicine and Sci., Rochester, MN), Carolina Amador (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine and Sci., Rochester, MN), and Shigao Chen (Dept. of Radiology, Mayo Clinic College of Medicine and Sci., Rochester, MN)

Shear wave elastography (SWE) is clinically used for the measurement of soft tissue mechanical properties. Most SWE methods assume that the tissue is elastic, but soft tissues are inherently viscoelastic. The viscoelasticity can be characterized by examining phase velocity dispersion. Methods to extract the phase velocities from the spatiotemporal data, v(x,t), involve using a two-dimensional Fourier transform. The Fourier representation, V(k,f), is searched for peaks that correspond to the phase velocities. We present a method that uses the Multiple Signal Classification (MUSIC) method to provide robust estimation of the phase velocity dispersion curves. We compared results from the MUSIC method with the current approach of searching for peaks in the V(k,f) representation. We tested this method on digital phantom data created using finite element models (FEMs) in viscoelastic media excited by a simulated acoustic radiation force push from a curved linear array. We evaluated the algorithm with different levels of added noise. Additionally, we tested the methods on data acquired in viscoelastic phantoms with a Verasonics system. The MUSIC algorithm provided dispersion estimation with lower errors than the conventional peak search strategy. This method can be used for evaluation of wave velocity dispersion in viscoelastic tissues and guided wave propagation. [This work was supported in part by grant R01DK092255.]

2:00

3pBAb2. Shear waves in pressurized poroelastic media. Navid Nazari (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, navidn@bu.edu) and Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA)

Shear wave elastography measures shear wave speed in soft tissues for diagnostic purposes. In Rotemberg *et al.* [J. Biomech. 46(11), (2013), pp. 1875-1881] and Rotemberg *et al.* [Phys. Med. Biol. 57(2), (2012), pp. 329-341], shear wave speed measurements were shown to depend on prestrain, but not necessarily prestress, in a perfused canine liver. We model this phenomenon by examining incremental waves in a pressurized poroelastic medium with incompressible phases. For a poroelastic material with strain energy function W, which due to pressurization undergoes a volume expansion of Δ, we find the following general expression for the shear wave speed: $c^2/{c_0}^2=(W_1(\Delta)+W_2(\Delta)\Delta^{2/3})/\Delta^{2/3}(W_1(1)+W_2(1))$. Here, c_0 is the shear wave speed in the unpressurized material, $W_j=\partial W/\partial I_j$, and I_j is an invariant of the Cauchy-Green strain tensor. We also find important restrictions on the form of the strain energy function, which are typically not satisfied by strain energy functions commonly assumed for soft tissues.

2:20

3pBAb3. Magnetoelastic waves in a soft electrically conducting solid in a strong magnetic field. Daniel Gendin and Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, digendin@bu.edu)

Shear wave motion of a soft, electrically conducting solid in the presence of a strong magnetic field excites eddy currents in the solid. These, in turn, give rise to Lorentz forces that resist the wave motion. We derive a mathematical model for linear elastic wave propagation in a soft electrically conducting solid in the presence of a strong magnetic field. The model reduces to an effective anisotropic dissipation term resembling an anisotropic viscous foundation. We consider the application to magnetic resonance elastography, which uses strong magnetic fields to measure shear wave speed in soft tissues for diagnostic purposes. We find that for typical values of magnetic field, mass density, and electrical conductivity of soft tissues, eddy current dissipation is negligible. For materials with higher conductivity (e.g., metals), the effect can be stronger.

2:40

3pBAb4. Shear elasticity and shear viscosity imaging in viscoelastic phantoms. Yiqun Yang (Michigan State Univ., 2152 Eng. Bldg., 428 S Shaw Ln, East Lansing, MI 48824, yangyiqu@msu.edu), Pengfei Song, Shigao Chen, Matthew W. Urban (Dept. of Radiology, Mayo Clinic College of Medicine and Sci., Rochester, MN), and Robert J. McGough (Michigan State Univ., East Lansing, MI)

Shear wave elasticity imaging determines the mechanical parameters of soft tissue by analyzing measured shear waves induced by an acoustic radiation force. Currently, the widely used time-of-flight method calculates the correlation between shear waveforms at adjacent lateral observation points to estimate the shear elasticity value. Although this method provides accurate estimates of the shear elasticity in purely elastic media, our experience suggests that this approach overestimates the shear elasticity values in viscoelastic media because the effects of diffraction, attenuation, and dispersion are not taken into account. To address this problem, we have developed an approach that directly accounts for all of these effects when estimating the shear elasticity. This new approach simulates shear waveforms using a Green's function-based approach with a Voigt model, while the shear elasticity and viscosity values are estimated using an optimization-based approach by comparing measured shear waveforms with simulated shear

waveforms in the time-domain. This operation is then performed on a point-by-point basis to generate images. The results indicate that there is good agreement between the simulated and measured shear velocity waveforms, and that this approach yields improved images of the shear elasticity and shear viscosity. [Work supported, in part, by NIH Grant R01DK092255.]

3:00

3pBAb5. An ultrasound surface wave elastography technique for noninvasive measurement of surface lung tissue. Xiaoming Zhang, Thomas Osborn, Boran Zhou, Brian Bartholmai, James F. Greenleaf, and Sanjay Kalra (Mayo Clinic, 200 1st ST SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Ultrasonography is not widely used in clinic for lung assessment because ultrasound cannot image deep lung tissue. In this abstract, we present a novel technique, lung ultrasound surface wave elastography (LUSWE), for measuring the elastic properties of superficial lung tissue. In LUSWE, a small, local, and 0.1 second harmonic vibration is generated on the chest of a subject. The speed of surface wave on the lung is measured by using an ultrasound probe. We are evaluating LUSWE for assessing patients with interstitial lung disease (ILD). LUSWE may be useful for assessing ILD because most ILD patients have typical fibrotic scars in the peripheral and subpleural regions of the lung. In a large clinical study of ILD patients, we measure both lungs through six intercostal spaces for patients and controls. The surface wave speed is measured at 100 Hz, 150 Hz, and 200 Hz. In an example, the surface wave speed is 1.88 ± 0.11 m/s and 3.3 ± 0.37 m/s, respectively, for a healthy subject and an ILD patient at 100 Hz and in the same intercostal space. LUSWE may compliment the clinical standard high-resolution computed tomography for assessing ILD. LUSWE may be useful for assessing other lung disorders.

TUESDAY AFTERNOON, 27 JUNE 2017 ROOM 205, 1:20 P.M. TO 3:20 P.M.

Session 3pEA

Engineering Acoustics and Physical Acoustics: Microelectromechanicalsystems (MEMS) Acoustic Sensors II

Vahid Naderyan, Cochair
Physics/National Center for Physical Acoustics, University of Mississippi, NCPA, 1 Coliseum Drive, University, MS 38677

Kheirollah Sepahvand, Cochair
Mechanical, Technical University of Munich, Boltzmannstraße 15, Garching bei Munich 85748, Germany

Robert D. White, Cochair
Mechanical Engineering, Tufts University, 200 College Ave., Medford, MA 02155

3p TUE. PM

Invited Papers

1:20

3pEA1. MEMS-Based acoustic sensors for fluid mechanics and aeroacoustics. Mark Sheplak (Dept. Elec. and Comput. Eng., Univ. of Florida, 216 Larsen Hall, PO Box 116200, Gainesville, FL 32611-6200, sheplak@ufl.edu)

This talk presents the development of several microelectromechanical systems (MEMS)-based acoustic transducer technologies for fluid mechanics and aeroacoustics applications. Specifically, this presentation will focus on several aluminum nitride piezoelectric MEMS dynamic pressure sensors and microphones. These devices offer the promise of reducing cost, improving performance, and increasing mounting flexibility over existing conventional microphone technologies. Specifically, a transducer with no external power requirement has a key advantage for a large-channel count, widespread deployment. The modeling and design aspects of these devices are reviewed. First, the electroacoustic transduction is predicted via piezoelectric composite plate theory. Lumped element models are then synthesized to describe the dynamic characteristics of the transducer diaphragm and the cavity/vent structure. Constrained nonlinear design optimization using a sequential quadratic programming scheme is then performed to determine the design parameters. Representative results for several applications will then be presented. Finally, unresolved technical issues are summarized for future sensor development.

1:40

3pEA2. MEMS resonant acoustic wake-up sensor. Jonathan Bernstein, Mirela Bancu, Daniel Reilly, Marc Weinberg, Doug Gauthier, and Richard Elliot (Draper Lab., Draper Lab., 555 Technol. Square, Cambridge, MA 02139, jbernstein@draper.com)

Draper has built zero-power MEMS wake-up sensors for DARPA's N-ZERO Program. This program aims to enable unattended sensor arrays that last for years, limited only by battery discharge rates. For some targets, characteristic frequency signatures allow detection using narrow band passive acoustic resonators. The MEMS sensors use ambient acoustic inputs to actuate a wake-up electrical relay.

Resonant sensors from 30 Hz to 180 Hz have been fabricated. The device rotates in response to an acoustic input, thereby avoiding large displacements due to gravity which would occur with a linear actuator. An adjustable acoustic cavity designed as part of the package is used to tune the resonant frequency to match a particular target. FEA Modeling was performed to achieve desired spring constants and resonant frequency. A rotary-acoustic lumped element equivalent circuit model was used to analyze the effect of the cavity and leakage resistances on the device performance. We will show finished MEMS devices and acoustic test data. [This research was developed with funding from the Defense Advanced Research Projects Agency (DARPA).]

2:00

3pEA3. Field-effect transistor-based transduction and acoustic receiving transducers. Wonkyu Moon, Min Sung (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 405, POSTECH, San31, Hyoja-dong, Nam-gu, Pohang, Kyungbuk 790784, South Korea, wkmoon@postech.ac.kr), Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), Pohang, Gyungbuk, South Korea), and Junsoo Kim (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), Pohang, Gyeongsangbuk-do, South Korea)

Microphones and hydrophones are representative acoustic receiving transducers. To properly receive sound waves, a receiver must be smaller than the wavelength of the target sound. The target wave characteristics do not impose any lower limits on the size of microphones. When the performance of a smaller microphone or hydrophone will be satisfactory, users generally choose a smaller device since smaller receivers are easier to install and use. However, miniaturized microphones are less sensitive at low frequencies and conventional infrasound detectors are considerably larger than those for higher frequency sounds. These trends in receiver size can be explained by considering the transduction characteristics of microphones and hydrophones. We describe two transduction mechanisms based on field-effect transistors (FET) and use them to develop new microphones and hydrophones. We used theoretical analysis and experiments to show that the sensitivity and frequency response functions of FET-based microphones and hydrophones are size-independent. These results suggest that more sensitive micro-machined microphones and hydrophones, with better frequency response functions, may be available for use in the near future. [Work supported by the Institute of Civil Military Technology Cooperation Center (16-CM-SS-18).]

2:20

3pEA4. Sensing sound with a nanoscale fiber. Ronald Miles and Jian Zhou (Mech. Eng., SUNY Binghamton, Dept. of Mech. Eng., Vestal, NY 13850, miles@binghamton.edu)

A feasibility study is presented of the use of a thin fiber to detect sound. The fiber is assumed to be supported on each end, with a deflection in response to a sound wave that propagates in the direction perpendicular to its long axis. The driving force on the fiber is the result of viscous forces in the oscillating air, which are well-known to be very important in determining the flow-induced motion of small structures. A simplified analytical model of the fiber's response is presented where it is argued that for fibers that are sufficiently thin, elastic and inertial effects become strongly dominated by viscous forces in the fluid. As a result, the fiber's motion becomes a very close approximation of that of the acoustic flow in its vicinity. Electrodynamic transduction of the fiber's motion provides a means of sensing sound with remarkable accuracy if the fiber diameter is taken to be measurably below one micron.

Contributed Papers

2:40

3pEA5. Characterization of a virtual array based on MEMS microphones for the analysis of acoustic sources. Alberto Izquierdo, Juan J. Villacorta (TeleCommun. Eng. School. Signal Theory Dept., Univ. of Valladolid, Valladolid, Spain), Lara del-Val (Industrial Eng. School. Mech. Eng. Dept., Univ. of Valladolid, Paseo del Cauce, Valladolid, Spain, lvalpue@eii.uva.es), and Luis Suarez (Superior Tech. School, Civil Eng. Dept., Univ. of Burgos, Burgos, Spain)

Using arrays with digital MEMS microphones and FPGA-based acquisition/processing systems allows to build systems with hundreds of sensors at a reduced cost. This work analyzes the performance of a virtual array with 6400 MEMS (80×80) microphones. The system is composed by a 2D positioning system that places a physical array of 64 microphones (8×8) in a grid with 8 x 8 positions, obtaining a spatial aperture of 2 x 2 meters. The measured beampattern is compared with the theoretical one for several frequencies and pointing angles. The beampattern of the physical array is also estimated for each one of the 64 positions used by the positioning system. Also, the measured beampattern and the focusing capacity are analyzed, since beamforming algorithms assume spherical wave due to the large dimensions of the array. Finally, frequency and spatial responses for a set of different acoustic sources are obtained showing angular resolutions of the order of tenths of degree.

3:00

3pEA6. Capacitive micromachined ultrasound transducers for acoustic anemometry on Mars. Gardy K. Ligonde, Daniela A. Torres, Eric F. Abboud, Wang-Kyung Sung, James Vlahakis (Mech. Eng., Tufts Univ., 200 College Ave., Anderson 204, Medford, MA 02155, gardy.ligonde@tufts.edu), Donald J. Banfield (Astronomy, Cornell Univ., Ithaca, NY), and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA)

An acoustic anemometer is under development for measuring wind speeds on Mars. Acoustic anemometry allows simultaneous measurement of wind speed and the speed of sound by measuring the acoustic time of flight in the forward and reverse directions. Acoustic anemometry avoids some sources of measurement error that plague other techniques for measuring winds in planetary atmospheres, such as hot wire measurements, or laser based tracking of scattered light from dust. The particular focus of this paper is the ultrasound transducers needed for the instrument. Capacitive micromachined ultrasound transducers (CMUT) fabricated at Tufts University have previously been described for atmospheric pressure operation (ASA Fall Meeting 2012). In this work, the transducers have been modified and tested under low pressure conditions similar to the atmospheric conditions expected on Mars (4.5 Torr). We describe a comparison between the modeled and measured transducer frequency response. The CMUT resonant frequency decreased from 204 kHz at 760 Torr to 116 kHz at 1 Torr. This is predicted by the models. The quality factor increased with decreasing pressure as expected, and is accurately modeled above 50 Torr. However, at pressures below 50 Torr, unmodeled damping mechanisms dominate acoustic losses, and a purely acoustic model underpredicts damping.

Session 3pMU

Musical Acoustics: General Topics in Musical Acoustics I (Poster Session)

Thomas Moore, Chair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

All posters will be on display from 1:20 p.m. to 3:40 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:20 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 3:40 p.m.

Contributed Papers

3pMU1. A perceptually orientated approach for automatic classification of timbre content of orchestral excerpts. Aurelien Antoine and Eduardo Miranda (Interdisciplinary Ctr. for Comput. Music Res. (ICCMR), Univ. of Plymouth, The House Bldg. - Rm. 304, Plymouth PL4 8AA, United Kingdom, aurelien.antoine@postgrad.plymouth.ac.uk)

In this paper, we report on the development of a perceptually orientated and automatic classification system of timbre content within orchestral audio samples. Here, we have decided to investigate polyphonic timbre, a phenomenon emerging from the mixture of instruments playing simultaneously. Moreover, we are focusing on the perception of the entire orchestral sound, and not individual instrumental sound. For accessibility to non-Acoustics experts, we chose to use verbal descriptors of timbre quality, such as brightness and roughness, to represent the timbral content of the samples. We based our acoustic analysis on the existing research into the perception and description of timbre. However, with a lack of agreed metrics, we had to establish a comparative scale for each timbre attribute implemented in our system, which is based on an analysis of audio recordings, in order to identify the dominant timbral attribute. To improve the classification accuracy, the system continually calibrates this scale as new audio files are analyzed. Preliminary analysis of our results shows a correlation between the system's classification and human perception, which is promising for further developments, such as standardizing metrics for perceived responses of timbral attributes or implementing systems for music production tasks.

3pMU2. A model for the observed decorrelation of partials in the overtone spectra of bowed stringed instruments. Sarah R. Smith and Mark F. Bocko (Univ. of Rochester, 405 Comput. Studies Bldg., P.O. Box 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

It has been shown that the overtone frequencies in the spectra of bowed stringed instruments played with vibrato exhibit less than perfect pairwise correlations. However, these results are inconsistent with the mechanism of performing vibrato by changing the length of the string. Since modulating the string length affects the frequencies of all string modes proportionately, it is curious that the overtones exhibit less than perfect correlations. The observed decorrelations, therefore, may be attributed to the filtering of the string's vibrations by the mechanical-acoustic resonant modes of the instrument body. The exact frequency deviations depend upon the frequencies of instrument's resonant modes in relation to the string's overtone frequencies and the width and rate of the vibrato. By modelling the instrument body as a sum of resonant modes driven by a frequency modulated saw tooth wave, we develop an analytical model relating the observed frequency deviations to the modal properties of the instrument. The effect of a single resonant mode on the instantaneous frequency trajectories is found analytically and informs numerical simulations of instruments with multiple modes. The simulated results compare well with data from recorded violin tones.

3pMU3. Characterization of free field radiation in brass instrument horns under swept sine excitation using a linear microphone array. Amaya López-Carromero (School of Phys. and Astronomy, Univ. of Edinburgh, 1608, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh, Scotland/Midlothian EH9 3FD, United Kingdom, s1374028@sms.ed.ac.uk), Jonathan A. Kemp (Univ. of St Andrews, St Andrews, Fife, United Kingdom), and D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, Edinburgh, United Kingdom)

A linear array with 23 microphones is used to scan a planar section of the sound field radiated into an anechoic environment by a range of brass instruments excited by a sinusoidal sine sweep. The planar section contains the symmetry axis of the bell and covers a rectangular area of 0.9 by 0.6 metres, starting at the plane of the bell and extending away from it along the longest side. The linear microphone array is perpendicular to the symmetry axis and is stepped along the axis. The resulting matrix of signals is processed to separate the linear and non-linear parts of the response; the three-dimensional pressure distribution in the sound field is deduced on the assumption that the field is cylindrically symmetric. This data then allows the visualization and further analysis of the frequency dependence of radiated wave fronts in brass instrument bells. Comparison is drawn between the observations and the predictions of several popular radiation models.

3pMU4. Non-invasive measurement of acoustic coupling between the clarinet bore and its player's vocal tract. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

As acoustically coupled resonators, the bore of a clarinet or other woodwind instrument and the vocal tract of the player interact in ways that affect timbre and pitch. Pitch in particular is strongly dependent on vocal tract acoustics when the bore-tract coupling is strong, such as when a tract impedance maximum is close in frequency and amplitude to a bore impedance maximum. Direct investigation of bore-tract coupling requires invasive measurement of bore and tract input acoustic pressures (or impedances), and one particular technique makes us of the ratio of these pressures (in the frequency domain, Pt/Pb) at harmonics of the reed vibration fundamental frequency. A non-invasive, model-based approach to investigating bore-tract coupling has been developed, which depends on a free-field microphone recording the sound produced by the instrument (P1), and an accelerometer placed against the skin of the neck recording skin vibrations (P2) related to intra-tract acoustic pressures. An additional, model-based calibration step is required. The ratio of these two signals (in the frequency domain, P2/P1) following calibration is qualitatively similar to the ratio Pt/Pb, and approaches quantitative identity as the model-based calibration step improves.

3pMU5. Construction of a finite element model of the Japanese *koto*. Angela K. Coaldrake (Music, Univ. of Adelaide, Elder Conservatorium of Music, University of Adelaide, Adelaide, SA 5005, Australia, kimi.coaldrake@adelaide.edu.au)

This paper presents the steps in developing a finite element model of the Japanese *koto* (13-stringed zither) in Comsol Multiphysics® v5.2a and some of the issues encountered. As the instrument is 1.8 meters in length and hand crafted there are many internal irregular shapes. Early attempts at creating a geometry were unsatisfactory. To address this issue a CT scan with 2400 cross-sections was used to measure the internal details. A mesh was created from the scan using Simpleware® software. The result was a mesh with 430,000 elements for the instrument alone, placed in a sphere of air resulting in over 7 million degrees of freedom. This new model therefore has required the use of high performance computing to produce a second of acoustic output. The issue of the physical properties of paulownia, the less well-characterized highly anisotropic wood used to construct the *koto*, has proven more intractable. Scanning electron microscopy, frequency response and acoustic camera studies of the original instrument provided important insights into paulownia in particular and developing the model in general. A number of studies have been undertaken to validate the model including comparing it with the original instrument. Further studies of the acoustics of the *koto* are in progress.

3pMU6. Simplified exponential horns. Jean-Pierre Dalmont, Carole Kameni, and Philippe Béquin (Laboratoire d'Acoustique de l'Université du Maine, Ave. Olivier Messiaen, Le Mans 72085, France, jean-pierre.dalmont@univ-lemans.fr)

The exponential horn is known as the shape realizing the best matching between a source and the external field for frequencies higher than its cut-off frequency. In practice, the horn being of finite length the effective cut-off is significantly higher and resonances appear as waves are reflected at the end of the horn. So, the response of the horn is far from being flat. Then, a question arises: in what extend the shape of an exponential horn has to be strictly respected in order to keep its main acoustical properties. The present paper intend to answer this question. First, a review of the different ways to calculate the input impedance of a horn and their accuracy is made. Comparison with input impedance measurements show that plane wave approximation is often sufficient even when the horn is strongly bended. Second, some criteria are proposed to characterize horns ability to match with the external field. These criteria are finally used to compare the performances of some simplified geometries of horns to that of strictly exponential horns.

3pMU7. Numerical study of nonlinear distortion of resonant states in acoustic tubes. Roberto Velasco-Segura and Pablo L. Rendon (Laboratorio de Acústica y Vibraciones, Centro de Ciencias Aplicadas y Desarrollo Tecnológico, Universidad Nacional Autónoma de México, Circuito Exterior S/N, C. U., Delegación Coyoacán, Mexico City 04510, Mexico, roberto.velasco@ccadet.unam.mx)

A numerical study of nonlinear acoustic propagation inside tubes is presented. Thermoviscous attenuation is included, giving rise to wall losses associated with the boundary layer. The full-wave simulation is performed in the time domain, over a 2D spatial domain assuming axial symmetry, and it is based on a previously validated open source code, using Finite Volume Method implemented in GPU (FiVoNAGI) [Velasco & Rendón, A finite volume approach for the simulation of nonlinear dissipative acoustic wave propagation, 2015]. One intended application is the identification of resonance frequency shifts in the nonlinear regime in brass musical instruments as a function of bore profile and amplitude of the driving stimulus. To gain insight on the nonlinear processes taking place inside the tube, visualizations are presented, differentiating spectral components and traveling waves in both directions.

3pMU8. Use of H1-H2 to quantify formant tuning for notes and portions of the vibrato cycle in the second passaggio of a professional female singer. Richard C. Lissemore (Speech-Language-Hearing Sci., Graduate Center/The City Univ. of New York, 499 Fort Washington Ave., Apt. #4F, New York, NY 10033, rlissemore@gradcenter.cuny.edu), Christine H. Shadle (Haskins Labs., New Haven, CT), Kevin Roon, and D. H. Whalen (Speech-Language-Hearing Sci., Graduate Center/The City Univ. of New York, New York, NY)

Singing voice pedagogy emphasizes that an acoustic change occurs in standard, classical, techniqued soprano voice between the musical notes D5♮ (587 Hz) and F5♮ (698 Hz). For low vowels, this involves a transition from tuning of the second resonance to the second harmonic (F2/H2) to tuning of the first resonance to the fundamental (F1/F0). In this single-subject study, we quantified the acoustics of this transition as the amplitude difference between the first and second harmonics (H1-H2). Results showed a clear and substantial change from negative to positive H1-H2 values at a pivot point between E5♭ and E5♮, implying the resonance tuning. Non-techniqued singing, with the same singer, showed no such change. F0 fluctuation (vibrato) of $\pm$90 cent at the pivot point resulted in positive H1-H2 values at vibrato maxima and negative ones at vibrato minima. Additionally, H1-H2 values were consistently higher at vibrato maxima than minima throughout the transition area. Potential explanations for the latter result are: (i) vocal tract resonances are located just above the sung F0, or (ii) the vibrato cycle is accompanied by an articulatory change, possibly laryngeal movement. This illustrates the intricacies of formant tuning and suggests future possibilities for numerical assessment of vocal technique.

3pMU9. Velocity analysis of the vacuum-driven clarinet reed. Carter K. Richard and Whitney L. Coyle (Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, crichard@rollins.edu)

A vacuumed artificial mouth has been assembled and tested to measure reed velocity for a Bb clarinet along the width of the reed. Reed velocity measurements may be useful for better estimation of parameters in physical models, such as the relevant surface area of the vibrating reed. Use of a vacuum system instead of a pressurized mouth chamber allows straightforward observation and manipulation of the mouthpiece apparatus. Point measurements of reed velocity were obtained via a laser-Doppler vibrometer directed at the reed surface when artificially blown. Simultaneous high-speed exposures were recorded to visualize reed motion. Preliminary results indicate that the velocity amplitude of any torsional motion in the reed is negligible compared to an asymmetric reed velocity, likely caused by natural limitations of the clarinet ligature. Velocity measurements also indicate that the reed may sometimes rebound against the mouthpiece in its oscillatory period. High-speed exposures support this conclusion by visualizing the reed deformation as it collides with the mouthpiece. This "rebound" deformation may contribute flow into the clarinet system. Further work will expand this measurement technique for a full grid along the surface of the reed, with various ligature mounts, and will seek to verify experimental measurement with analytic models.

3pMU10. Stepwise regimes in elephant trumpet calls: Similarities with brass instrument behavior. Joël Gilbert (Laboratoire d'Acoustoique de l'Université du Maine - CNRS, Ave. Olivier Messiaen, LE MANS 72085, France, joel.gilbert@univ-lemans.fr), Angela Stoeger (Dept. of Cognit. Biology, Univ. of Vienna, Vienna, Austria), Benjamin Charlton (School of Biology & Environment Sci., Univ. College Dublin, Dublin, Ireland), and David Reby (School of Psych., Univ. of Sussex, Brighton, United Kingdom)

Trumpet calls are very loud voiced signals given by highly aroused elephants, and appear to be produced by a forceful expulsion of air through the trunk. Beyond their characteristic "brassy quality" previously attributed to shockwave formation, some trumpet calls are also characterized by stepwise fundamental frequency increase and decrease. Here we used spectral analysis to investigate the frequency composition of trumpet calls from one Asian and one African elephant. We found that the frequency interval between the steps were consistent with resonances expected in the exceptionally long elephant vocal tract. Such stepwise regimes are commonly observed in brass instruments as self-sustained oscillations transiently align on the bore's resonance frequencies during arpeggios. We suggest that this production mechanism may constitute a rare example of source-filter interaction (where the filter properties affect the behavior of the source) in the vocal system of a terrestrial mammal. These preliminary observations also emphasize how the generalization of musical acoustic models can provide useful insight into the production of animal vocal signals.

3pMU11. Embrace for impact: Formant reconstruction in sudden pitch raise while singing. Chi-Yang Long (Graduate Inst. of Foreign Literatures and Linguist, National Chiao Tung Univ., 4F., No.20, Ln. 185, Sec. 2, Jinshan S. Rd., Da'an Dist., Taipei 10644, Taiwan, garylung710@gmail.com) and Yuwen Lai (Graduate Inst. of Foreign Literatures and Linguist, National Chiao Tung Univ., Hsinchu, Taiwan)

Professional singers are trained to maintain vocal configuration by suppressing laryngeal elevation when singing high notes. The present study investigates the organization of vocal filter in sudden pitch raise condition by examining the corresponding acoustic correlates. More specifically, we are interested in whether this phenomenon may have differential effect on vowels with different height. In the experiment, high and low vowels were embedded in a nonsense Mandarin carrier sentence "kuai-lai-kuai-lai___yi-po" with C4E4G4E4___D4D4 pitch contour. Thirty amateur singers were recorded singing the sentence in four melodic conditions with the pitch intervals between the preceding syllable /lai/ to target word manipulated. The conditions are Micro (E4 to D4), Macro (E4 to D5), all High (every word sung in D5) and Null (only the last three words sung in D5-D5D5 contour). The results show that high pitch singing (Macro, High, and Null) indeed induces formant reconstruction when compared to Micro and the effect is markedly stronger in Macro. Furthermore, high vowel are more susceptible than low vowel and undergoes greater degree of formant reconstruction. The results provide acoustic grounding for the possible interplay between diction and pitch contour.

3pMU12. A study on assessing clarity in pronunciation of soprano singers. Uk-Jin Song and myungjin bae (Dept. Information and TeleCommun. Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul 156743, South Korea, imduj@ssu.ac.kr)

The voice of soprano singers reaches at the highest notes among female vocalists. Soprano singers usually have low clarity with respect to pronunciation because their jaw joints and mouth shapes tend to stay rigid in order to maintain high notes for a long time. Five formants found in human voice are differently shaped depending on his or her different physical structures. Especially, high clarity in pronunciation has a distinct formant shape from F1 through F5, because the jaw joint and the mouth shape have a large influence on F4 and F5 appearing in the high frequency range. This paper comparatively analyzes the vocal voices of four different Korean soprano singers concerning clarity in pronunciation. The results of acoustic analysis of these four singers shows that the singer A and the singer D show from F1 ~ up to F3 above 2 kHz but F4 and F5 do not appear in this range, and the singer C had somewhat inconsistency with her formant characteristics as a whole. In the singer B, formants from F1 through F5 are distinctly shown even above 2 kHz. The study concludes that the singer B has the best clarity in pronunciation.

3pMU13. On a sound analysis of Korean Trot singer Nam in-su's voice. Bong Young Kim, Ik-Soo Ann, and Myungjin Bae (Sori Sound Eng. Lab, Soongsil Univ., 1212 Hyungnam Eng. Building 369 Sangdo-Ro, Dongjak_Gu, Seoul, Seoul 06978, South Korea, bykim8@ssu.ac.kr)

The word "'trot" means "walk quickly" and the fox-trot, which refers to the performance rhythm of social dance and it is now being used as a word relating to musical genre and performance. The late singer Nam In-su was Korean who performed in the early stage of Korean trots, and he had sung about 1,000 songs for about 20 years since his debut in 1938. His representative songs are "Sorrowful Serenade" (1938), "Vanish Away, the 38th Parallel (the Ceasefire Line of the Korean War)" (1949) and "Parting at Busan Station" (1953). His songs consoled Korean people against the sorrow of a lost and colonized home country and healed the pain generated from the division between South and North Korea and the loss due to the Korean war. This paper analyzes the acoustic components of his vocal voice that induce the softness and liveliness. His voice crosses over three octaves, and even when he sings quickly, his high-pitched sound spreads wide. The sound connection between measures is natural and smooth. In addition, deep vibrations appear at all frequency bandwidths, and pronunciations of lyrics are accurate, so his voice sounds very lively. This study will be helpful in understanding the value of popular singers' vocal voice supported with acoustic analysis.

3pMU14. On characteristics of Trot singer Bae Ho's singing voice. Sang Bum Park and Myungjin Bae (Soongsil Univ., Sando-ro 369, Dongjak-gu, Seoul 06978, South Korea, sbpark8510@naver.com)

In Korea, the trot began shaping its own style in the 1960s. Then, later in the 1970s, it became more specialized having four-fourths beat rhythm of fox-trot, and the Korean style has been finally established with a strong beat and a unique chopping technique. Although there are many trot singers in Korea, the singer Bae Ho was very special in using the heavy bass accompaniment which was later frequently used by other pop singers. The singing voice of Bae Ho gives us the feeling of special softness and appealingness. In addition, his vocalization features the addition of a deep vibe to the song, so that the listeners feel sympathetic and comfortable. This study compares the singing voice of Bae Ho to that of his mimic singers by examining acoustic characteristics of their voices including amplitude, frequency and duration. The acoustic analysis proved that the singing voice of Bae Ho is clearer and has longer duration of vibration in the bass section than that of mimic singers. Bae Ho has a natural talent in expressing the bass part as well as in the midrange part without changing the tone of his voice, while the mimic singers reveal many unnatural connections between measures.

3pMU15. Comparison of classification of musical genre obtained by subjective tests and decision algorithms. Aleksandra Dorochowicz (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Poland), Agata Majdańczuk (Audio Acoust. Lab., Gdansk Univ. of Technol., Gdansk, Poland), Piotr Hoffmann (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Poland), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The aim of the study is to conduct subjective tests on audio excerpt assignment to music genre and to carry out automatic classification of musical genres with the use of decision algorithms. First, the musicology background of classifying music into styles and genres is discussed. Then, an online survey is created to perform subjective tests with a group of listeners, whose task is assigning audio samples to selected music genres. Next, a set of music descriptors is proposed and all music excerpts are parametrized. For checking parameter redundancy the Principal Component Analysis (PCA) is performed. The created database containing feature vectors is then utilized for automatic music genre classification. Two classifiers, namely: Belief Networks and SMO (*Sequential Minimal Optimization Algorithm*) are employed for the purpose of music genre classification. The last step of this study is to compare the efficiency of the listeners classification with the automatic music genre classification system designed by the authors. Conducted tests show to what extent listeners' assignment and the automatic classification results agree. It is also observed that very known performers are often rated without problems. Contrarily, songs of less known artists are more difficult to assign to the given genre.

3pMU16. Robust Hidden Markov Models for limited training data for birdsong phrase classification. Kantapon Kaewtip, Abeer Alwan (Elec. Eng., UCLA, 623 1/2 Kelton Ave., Los Angeles, CA 90024, jomjkk@gmail.com), and Charles Taylor (Dept. of Ecology and Evolutionary Biology, UCLA, Los Angeles, CA)

Hidden Markov Models (HMMs) have been studied and used extensively in speech and birdsong recognition but they are not robust to limited training data and noise. This work present a novel method to training GMM-HMMs with extremely limited data—and possibly noisy—by sharing HMM components and generating more training samples that cover the variation of the models. We propose an efficient state-tying algorithm that takes advantage of unique characteristics of birdsongs. Specifically, the algorithm groups HMM states based on the spectral envelopes and fundamental frequencies, and the state parameters are estimated according to the group assignments. For noise-robustness, prominent time-frequency regions (time-frequency ranges expected

3p TUE. PM

to contain high energy for a particular HMM state) are used to compute the state emitting probability. In Cassin's Vireo phrase classification using 75 phrase types, the results show that the proposed state-tying algorithm significantly outperforms both traditional state-tying algorithms and baseline HMMs in most training conditions (using 1, 2, 4, 8, and 16 samples). Factors such as number of training data, number of shared components, and level of background noise are also studied in this work.

3pMU17. Real-time localization and tracking of musical instruments with spherical microphone arrays. Jonathan Mathews and Jonas Braasch (Architectural Sci. - Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, mathej4@rpi.edu)

This research describes a method for dynamic beamforming with a particle filter to localize and track musical instruments in a real-time context. Using a spherical harmonic framework, spherical microphone arrays are able to decompose three-dimensional sound fields into their basic components, which enables detailed analysis and efficient spatial filtering algorithms. In recent years, methods for determining relative source positions around an array using steerable beams have been studied. By creating multiple weighting functions based on spherical harmonic components, many beams can be generated simultaneously and can be used to dynamically track instruments via an iterative process.

3pMU18. Acoustical characteristics of Chinese musical instrument bamboo flute. Linhui Peng (Ocean Technol., Ocean Univ. of China, 238 Songling Rd., Information College, Qingdao, Shangdong 266100, China, penglh@ouc.edu.cn) and Tao Geng (Music performance, School of Music of South China Normal Univ., Guangzhou, Guangdong, China)

There are few reports on acoustical study about Chinese music instruments, which is an area worthy of researching. Nowadays there are more and more new and comprehensive ways that can be used to analyze acoustical characteristics and quality of musical instruments, such as experimental modal analysis, finite element software. It is said that music is a kind of a world language. However, the music is an expression of a culture, which is expressed by the musical instrument with its specific acoustical property and character. Therefore, it is necessary to research the acoustical characteristics that can express Chinese culture in the acoustical music instrument study. Bamboo flute is one of the most important musical instruments in any Chinese music ensemble or Chines orchestra. Acoustical structure and characteristics of bamboo flute tone are researched. Meanwhile, acoustical characteristic features for some main playing technique of bamboo flute are also researched. Then, the identified characteristics of bamboo flute tone related with Chinese music and culture are analyzed.

3pMU19. Intonation detection in a melodic context. Gabriella Marrone (Commun. Disord., Stockton Univ., 101 Vera King Farris Dr., Galloway, NJ 08205, marroneg@go.stockton.edu) and Neil L. Aaronson (Natural Sci. and Mathematics, Stockton Univ., Galloway, NJ)

Listeners with a wide range of formal and informal musical experience were asked to listen to an eight-tone diatonic C Major scale, generated using a piano sample library, in which one of four notes (D4, F4, A4, or C5) would be mistuned in 13 different mounts between -32¢ and +32¢. Listeners were told which note might be mistuned and were simply asked to indicate whether the scale was in-tune or not. Each listener was exposed to each degree of mistuning ten times. The frequency with which they said a scale was in-tune as a function of the degree of mistuning was plotted for each note and listener, to which a three-parameter pseudo-normal distribution (mean, standard deviation, height) was fitted. The standard deviation indicated the sensitivity of the listener to intonation in each case (large deviation implied low sensitivity to intonation). Listeners were then ranked based on their musical background, training, and experience. The effect of musical training on intonation sensitivity was a significant factor ($p<0.001$). There was also a significant effect of the particular note on the sensitivity of listeners, with the intonation of A4 being most difficult to detect across listeners ($p<0.001$).

3pMU20. Exploring some questions on occlusion effect in the human auditory system when a musician or singer's external ear canal is blocked. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Sometimes some musicians and singers prefer to use their palm or other objects to cover or occlude at least one ear during performance. This practice may be helpful to enhance their self monitoring of the sound production using the occlusion effect. The basic occlusion effect in the human auditory system has been explored and reported in the literature by several individuals. According to those reports, the musician or singer can hear his or her own voice or musical instrument significantly louder when the ear canal is blocked at the outer end. Many clinicians routinely utilize detectability of vibrations from a tuning fork when placed on the mastoid process and the ear canal is occluded. This is popularly known as the Bing effect. These empirical data suggest existence of an efficient acoustic connectivity from the vocal tract to the ear canal for healthy individuals. Therefore, this study will explore quantified effects of the classic Bing effect for normal healthy musicians and singers across the audio spectrum.

Session 3pNSa

Noise: Implications of Community Tolerance Level Analysis for Prediction of Community Reaction to Environmental Noise

Sanford Fidell, Cochair
Fidell Associates, Inc., 23139 Erwin St., Woodland Hills, CA 91367

Truls Gjestland, Cochair
Acoustics, SINTEF Digital, SINTEF, Trondheim N 7465, Norway

Chair's Introduction—1:15

Invited Papers

1:20

3pNSa1. Community tolerance level as a paradigmatic shift in development of dosage-response relationships. Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St., Woodland Hills, CA 91367, sf@fidellassociates.com)

Dosage-response functions relating the prevalence of a consequential degree of annoyance in communities to cumulative exposure to transportation noise have typically been derived by correlational methods. Such non-causal analyses fit a curve to a set of field observations of annoyance prevalence rates in multiple communities, typically by means of univariate logistic regression. The resulting curve passes through the centroid of a cloud of data points, but provides no insight into the great variability among communities in annoyance prevalence rates at similar noise exposure levels. Community Tolerance Level analysis partitions annoyance into acoustic and non-acoustic components. This approach is a normative (that is, causal and prescriptive) one that fits data sets to an *a priori* function in order to estimate a second parameter. The second parameter is the deviation from the assumed (effective loudness) growth function, expressed in decibel-denominated units as the level of cumulative noise exposure at which half of a community is predicted to be highly annoyed by transportation noise. CTL analyses of community response to transportation noise thus permit direct quantification not only of the growth of annoyance with noise exposure, but also of the aggregate effect of all non-acoustic influences on annoyance prevalence rates.

1:40

3pNSa2. Estimating community tolerance for wind turbine noise annoyance. Stephen Keith and David Michaud (Health Canada, Canadian Federal Government, 775 Brookfield Rd., Ottawa, ON J9H7J9, Canada, david.michaud@hc-sc.gc.ca)

A-weighted WTN contributed less than 10% to the strength of the multiple regression model developed for wind turbine noise (WTN) annoyance in Health Canada's Community Noise and Health Study (CNHS). Improvements required consideration of non-LAeq variables unknown or even inapplicable beyond the CNHS. To facilitate cross-study comparisons, an analysis of WTN annoyance was conducted based on the community tolerance level (CTL) model. The rate of increase in WTN annoyance was effectively estimated using a loudness function, as shown in Eq. (1). $\%HA = 100 \exp(-(1/[10((DNL-CTL+5.306)/10)]^{0.3}))$. (1) By convention, CTL is the DNL from Eq. (1) where 50% of the community would be highly annoyed. The CTL for WTN annoyance from field studies published to date ranges from 57.1 to 64.6 DNL (mean = 62, $\sigma = 3$). CTL values developed by others for transportation noise sources suggest that, on average, communities are between 11 dB and 26 dB less tolerant of WTN than of other sources, depending on the source. Confidence in these results should increase as future studies in this area produce additional estimates for the relationship between WTN level and the prevalence of high annoyance. The CTL analytical methods, assumptions, strengths, and limitations are presented.

2:00

3pNSa3. Multilevel modeling and regression as applied to community noise annoyance surveys. D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, david.k.wilson90.civ@mail.mil), Nicole M. Wayant (Geospatial Res. Lab., U.S. Army Engineer Res. and Development Ctr., Alexandria, VA), Edward T. Nykaza (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), Chris L. Pettit (Aerosp. Eng., U.S. Naval Acad., Annapolis, MD), and Chandler M. Armstrong (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Tolerance to noise varies between communities, and between individuals comprising the communities. A multilevel modeling approach is useful for capturing such individual- and community-level variations. Here, we consider a model in which the community-level variations are sampled explicitly with community noise surveys, while the individual-level variations are sampled only in a statistical sense (i.e., hidden). The community-level variations are specifically quantified by the community tolerance level (CTL) [Fidell *et al.*, J. Acoust. Soc. Am. **130**(2), 791-806 (2011)]. Simulations based on the multilevel model indicate that the community- and individual-level variations have distinct statistical signatures, both of which are evident in noise annoyance surveys involving transportation noise.

The annoyance curve for a previously unsurveyed community depends on the mean and variance of the CTL, and the sum of the hidden variances in noise tolerance and exposure among individuals in the community. Regression analyses of transportation noise annoyance using a multilevel, generalized linear model (GLM) enable noise tolerances and their variations at the two model levels to be distinguished and quantified.

2:20

3pNSa4. CTL—A useful tool for inter-survey comparisons. Truls Gjestland and Femke B. Gelderblom (Acoust., SINTEF Digital, SINTEF, Trondheim N 7465, Norway, truls.gjestland@sintef.no)

The concept of *Community Tolerance Level* was presented in 2011 and in subsequent analyses the CTL has proven to be a useful tool for comparing results from different surveys on noise annoyance. The CTL provides a robust single-number description of the prevalence of annoyance in a community noise survey. It has been shown that accurate predictions of the prevalence of highly annoyed residents can be achieved by relatively few survey interviews as the selection of respondents does not have to cover the whole range of exposure. A re-analysis of existing studies on aircraft noise annoyance has shown that there has been virtually no change in people's response to this type of noise over the past 50 years, but there is a distinct difference in the response depending on the rate of which the noise situation has been changing. A similar analysis of results on road traffic noise from surveys conducted over a period of more than 50 years, will be presented.

2:40

3pNSa5. Observations of non-zero asymptotes for high annoyance at low sound exposure levels. Richard Horonjeff (RDH Acoust., 81 Liberty Square Rd. #20-B, Boxborough, MA 01719, rhoronjeff@comcast.net)

Social survey data relating the prevalence of a high annoyance to transportation noise typically indicate a decrease in annoyance rate with decreasing exposure. At very low exposure levels, the expectation is that the percentage of highly annoyed respondents will asymptotically approach a value of zero. In many studies this has in fact been the case. In some studies, however, the lower asymptote does not appear to be zero, however, but rather some small, non-zero value. Such non-zero asymptotic values are sometimes nearly constant over an exposure range of as much as 10 decibels. In other words, the observed prevalence of noise-induced annoyance appears to be independent of sound level over an extended range, before beginning to depend on noise dose at higher levels. Several examples of this phenomenon are presented and discussed.

3:00

3pNSa6. Applying community tolerance level to high-energy impulse sounds. Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

Fidell, Schultz and Green [JASA **84**, pp. 2109-2113 (1988)] and Green and Fidell [JASA **89**, pp. 234-243 (1991)] suggested a systematic approach to distinguishing loudness-driven from community-specific contributions to the prevalence of transportation noise-induced annoyance. Schomer [JASA **86**, pp. 835-836 (1989)] showed how the approach could be adapted to predicting community response to high amplitude impulsive noise, simply by changing the exponent of the dosage-response function. The current presentation demonstrates in a more accessible fashion that the basic approach identified in the 1989 paper can be applied to predicting community response to high amplitude impulsive noise. The approach builds on an application by Schomer *et al*. [JASA **131**, pp. 2772-2786 (2012)] of the CTL approach to road and rail traffic noise. The latter paper explains the "Community Tolerance Level" method from the perspective of an observer watching a process, rather than as a participant in the process. These principles are used in this paper to reinforce the conclusions of the 1989 analysis by Schomer.

Session 3pNSb

Noise, Physical Acoustics, ASA Committee on Standards, and Structural Acoustics and Vibration: Sonic Boom Noise V: Turbulence, Predictions, and Measurements

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

Victor Sparrow, Cochair
Grad. Program in Acoustics, Penn State, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—1:15

Invited Papers

1:20

3pNSb1. Sonic booms in atmospheric turbulence ground measurements in a hot desert climate. Edward A. Haering (Res. AeroDynam., NASA Armstrong, M.S. 2228, PO Box 273, Edwards, CA 93523, edward.a.haering@nasa.gov)

The Sonic Booms in Atmospheric Turbulence (SonicBAT) Project flew a series of 20 F-18 flights with 69 supersonic passes at Edwards Air Force Base in July 2016 to quantify the effect of atmospheric turbulence on sonic booms. Most of the passes were at a pressure altitude of 32,000 feet and a Mach number of 1.4, yielding a nominal sonic boom overpressure of 1.6 pounds per square foot. Atmospheric sensors such as GPSsonde balloons, Sonic Detection and Ranging (SODAR) acoustic sounders, and ultrasonic anemometers were used to characterize the turbulence state of the atmosphere for each flight. Spiked signatures in excess of 7 pounds per square foot were measured at some locations, as well as rounded sonic-boom signature with levels much lower than nominal. This presentation will quantify the range of overpressures and Perceived Level of the sonic boom as a function of turbulence parameters, and also present the spatial variation of these quantities over the array. Comparisons with historical data will also be shown. The NASA Armstrong Research Center's team is made up of KBR Wyle, Gulfstream, Boeing, Pennsylvania State University, Lockheed Martin, Eagle Aeronautics, and Laboratoire de Mécanique des Fluides et d'Acoustique (LMFA in France).

1:40

3pNSb2. Statistics of supersonic signatures propagated through simulated atmospheric turbulence. Trevor A. Stout and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, tastout6@gmail.com)

The issue of annoyance caused by acoustical signatures from supersonic aircraft may be quantitatively predicted through propagation simulations. In particular, a model of energy focusing or defocussing due to turbulence in the atmospheric boundary layer is necessary to predict the real variations found whenever measurements are taken at multiple locations or at multiple times. The present algorithm solves the Khoklov-Zobolotskaya-Kuznetzov (KZK) equation and attempts to account for all important atmospheric effects such as refraction, thermoviscous absorption, relaxation processes, and nonlinearity, as well as turbulence effects due to temperature and vector wind variations. The turbulence model uses von Karman spectra, with vertically-varying length scales to better model the full extent of the boundary layer. The presentation will discuss some early results and relevant statistics, as well as the feasibility of comparison to recent overflight tests in NASA's SonicBAT program. [Work supported by NASA via a subcontract from KBRwyle.]

2:00

3pNSb3. Removing distortions from ground sonic boom measurements for certification of over land flight. John M. Morgenstern (Lockheed Martin Aeronautics, 1011 Lockheed Way, Palmdale, CA 93599, john.morgenstern@lmco.com)

Ground measurements of sonic booms contain distortions acquired during their propagation through the atmosphere that greatly changes their loudness. Quiet shaped boom vehicles are being proposed for acceptable over land flight. Because of distortion loudness variations, many ground measurements would be needed to develop accurate statistics of loudness. And highly turbulent atmospheres increase loudness variations and affect average loudness, so a limited range of atmospheric conditions would be required for a certification procedure of acceptable loudness. More flights and limited conditions would substantially increase vehicle certification cost. A measurement de-turbulencing technique is shown to greatly reduce atmospheric distortions. It improves loudness accuracy to (hope to quantify) from a single pass over a line of at least 25 microphones. The method combines a de-turbulencing technique developed by Plotkin, with a spatial averaging technique used in wind tunnel measurements of sonic boom, and with additional analytical methods being developed. The de-turbulencing and spatial averaging has been applied to F-5SSBD measurements, resulting in a consistent loudness 2 PLdB quieter than analytical predictions due to greater rounding of the averaged turbulence distortions.

2:20

3pNSb4. Prediction of various sonic boom signatures observed in large scale experiment. Masashi Kanamori, Takashi Takahashi, and Yoshikazu Makino (Inst. of Aeronautical Technol., Japan Aerosp. Exploration Agency, 7-44-1, Jindaijihigashi-machi, Chofu, Tokyo 182-8522, Japan, kanamori.masashi@jaxa.jp)

Results of predicting sonic boom signatures observed in D-SEND#2 flight test are presented in this study. D-SEND#2 flight test was held in northern Sweden in 2015 in order to demonstrate JAXA's low-boom design concept. The flight test provides three kinds of waveforms: conventional N wave, diffracted U-shaped waveform, and the low-boom waveform. In this study, the method of predicting one or more waveforms above will be introduced.

2:40

3pNSb5. Evaluation of finite difference approximations of absorption and dispersion implemented in sonic boom propagation model equations. Matthew T. Collmar (Gulfstream Aerosp. Corp., 32 Innovation Dr., Savannah, GA 31408, matthew.collmar@gulfstream.com) and Joseph A. Salamone (Gulfstream Aerosp. Corp., Tybee Island, GA)

Recently developed one-dimensional far-field sonic boom propagation model equations are variants of an augmented Burgers equation. The Gulfstream Aerospace Corporation far-field sonic boom propagation tool, GACBoom, employs a numerical approach that consists of operator splitting in the time-domain. The mechanisms in the equation include geometric spreading, atmospheric stratification, nonlinearity, absorption, and dispersion. For propagation of sonic booms in the atmosphere, the predominant dissipative mechanisms are thermoviscous attenuation and molecular relaxation due to diatomic oxygen and nitrogen, the latter of which also contribute to dispersion. One method for utilizing predictions with these mechanisms is to employ a finite difference approximation to the dissipative and dispersive terms in the sonic boom propagation model equation. This paper uses a discrete Fourier analysis to investigate both spatial and temporal discretization ramifications for the dissipative and dispersive components included in the model equations and compares the results against analytic formulations found in the literature. Concluding remarks include numerical considerations when accounting for the influence of absorption and dispersion in sonic boom propagation.

3:00

3pNSb6. The use of optical methods for measuring irregular reflection of weak acoustic shock pulses from a solid surface in air. Maria M. Karzova, Petr V. Yuldashev (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Didier Dragna (LMFA - UMR CNR 5509, Univ. Lyon, Ecole Centrale de Lyon, Ecully, France), Sébastien Ollivier (LMFA - UMR CNR 5509, Universite Lyon 1, Ecole Centrale de Lyon, Ecully, France), Vera Khokhlova (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (LMFA - UMR CNR 5509, Univ. Lyon, Ecole Centrale de Lyon, Ecully, France)

Irregular reflection ofacoustic weak shockwaves (acoustic Mach numbers less than 0.01) is known as the von Neumann paradox and could not be described by the three-shock theory. In this work, nonlinear reflection regimes were studied experimentally using spark-generated spherically divergent *N*-waves reflecting from a plane rigid surface. Two optical methods were used in the measurements: a Schlieren system to visualize reflection patterns and a Mach-Zehnder interferometer to reconstruct pressure waveforms. The reconstruction was performed by applying the inverse Abel transform to the phase of the signal measured by the interferometer. The Mach stem formation was observed close to the surface as a result of collision of the incident and reflected front shocks of the *N*-wave and further away from the surface where the reflected front shock interacted with the incident rear shock. It was shown that irregular reflection occurred in a dynamic way and the length of the Mach stem increased while the N-wave propagated along the surface. In addition, reflection patterns were analyzed for several rough plane surfaces with different roughness size. The height of the Mash stem was found shorter for surfaces with larger dimension of the roughness.

Session 3pNSc

Noise: Effects of Noise and Perception (Poster Session)

William J. Murphy, Chair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

All posters will be on display from 1:20 p.m. to 3:40 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:20 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 3:40 p.m.

Contributed Papers

3pNSc1. What is a safe noise exposure level for the public? Daniel Fink (The Quiet Coalition, The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

What is a safe noise exposure level for the public? This question should underlie basic and applied research on the effects of sound on humans, and regulatory efforts to control public noise exposure. The safe public noise exposure level cannot be an unadjusted occupational standard. Unlike other occupational exposures, noise exposure continues outside the workplace. Occupational limits must be adjusted for increased exposure time, from 8 to 24 hours daily, 240 workdays to 365 days annually, and from 40 work years to the entire lifespan. Recommended safe noise exposure levels depend on which adverse noise effect is being considered. To prevent hearing loss, the U.S. Environmental Protection Agency (EPA) adjusted the U.S. occupational recommended exposure level of 85 A-weighted decibels for additional exposure time to calculate a 70 decibel time weighted average (TWA) exposure level. EPA did not adjust for lifespan years so the correct safe exposure level is likely lower. The World Health Organization (WHO) also recommends 70 decibels to prevent hearing loss. EPA and WHO determined that non-auditory health impacts of noise occur at 55 decibels TWA, with annoyance starting at 45 decibels. These are the safe noise exposure levels for the public.

3pNSc2. Noise exposure in berthing rooms of Naval ships. Shakti K. Davis, Christopher Smalt, and Paul Calamia (BioEng. Systems and Technologies, MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, shakti@ll.mit.edu)

Noise on aircraft carriers is known to exceed hazardous noise levels as jets launch and land on the flight deck and loud machinery operates below deck. Crew members often reach their daily noise allowance while performing work duties but the conditions for auditory recovery onboard are not well understood. To address this gap, we assisted the Navy in recording 24h persistent noise measurements in several berthing rooms on the USS Nimitz (CVN-68). During flight operations, the 8h time-weighted average (TWA) noise levels in these below-deck living spaces ranged between 75 and 81 dBA. While the levels fall below the Department of Defense TWA limit of 85 dBA, these conditions may not support auditory recovery from temporary threshold shifts that occurred during work hours. Another potential noise hazard in these rooms is impulse noise from flight deck catapults and arresting wires, with peak levels as high as 143 dB. In this presentation, we describe an analysis of the 24h noise exposure from aircraft-carrier berthing rooms including steady-state and impulse noise exposure metrics defined in MIL-STD-1474E. [Work supported by the ONR under Air Force Contract No. FA8721-05-C-0002.]

3pNSc3. Development of laboratory-based test methods for the comparative evaluation of performance of advanced hearing protection devices. Theodore F. Argo, Brian Zadler, Cortney LeNeave, and Jennifer Jerding (Appl. Res. Assoc., Inc., 7921 Shaffer Parkway, Littleton, CO 80127, targo@ara.com)

Standard acoustic test methods may not fully capture the performance characteristics of advanced passive and active hearing protection devices. Development of new laboratory-based test methods with the ability to discriminate between the performance characteristics of these devices, such as level-dependent or nonlinear effects, without the use of human subjects was undertaken. Measurements of hearing protection device performance with respect to signal quality, sound localization, self-noise, and impulse response were performed. Signal quality and sound localization were both tested using a compact 3D positioning apparatus which gave us directivity information without the use of human subjects or a large hemispherical test fixture. A sound isolation box was used not only for self-noise but also for impulse response measurements. Results of the impulse testing were compared to freefield shock tube results to ensure consistency among methods. This new evaluation methodology, when performed on an array of advanced hearing protection devices, can provide supplemental or alternative performance data for the relative comparison of devices under test. The results allow for distinction between devices based on preferred characteristics, for example, low self-noise and good signal quality favored over better impulse protection. [Research supported by US Army Natick Soldier Research, Development, & Engineering Center.]

3pNSc4. Evaluation of a calculation method of noise exposure from communication headsets. Hilmi R. Dajani (School of Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, ON K1N6N5, Canada, hdajani@site.uottawa.ca), Flora G. Nassrallah (Hearing Res. Lab., Univ. of Ottawa, Ottawa, ON, Canada), Caroline Chabot (Audiology/SLP Program, Univ. of Ottawa, Ottawa, ON, Canada), Nicolas N. Ellaham (Hearing Res. Lab., Univ. of Ottawa, Ottawa, ON, Canada), and Christian Giguère (Audiology/SLP Program, Univ. of Ottawa, Ottawa, ON, Canada)

Specialized standardized methods for the measurement of noise exposure from communication headsets or sound sources close to the ear include the Microphone in Real Ear and manikin techniques, as specified in ISO 11904-1/2. The 2013 version of Canadian standard Z107.56 introduced a simpler calculation method to increase accessibility to communication headset exposure assessments for the widest range of stakeholders in hearing loss prevention. The calculation method only requires general and widely accessible sound measurement equipment and basic computational steps that account for the main determinants of exposure such as the background noise around the user, the sound attenuation of the communication headset, and the expected communication duration and effective listening signal-to-noise ratio. This paper reviews recent research on the effects of the spectral and temporal characteristics of the background noise and the headset

3p TUE. PM

configuration on the speech listening level. Results indicate that the listening level is largely insensitive to spectral and temporal variations in the background noise and that A-weighted noise level is a good predictor of listening level once headset attenuation is taken into account. It is also found that one-sided headsets increase exposure by 6-7 dB compared to two-sided headsets due to binaural summation.

3pNSc5. On-body and in-ear noise exposure monitoring. Christopher Smalt, Shakti K. Davis, Paul Calamia, Joe Lacirignola (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, Christopher.Smalt@ll.mit.edu), Olha Townsend (Brown Univ., Lexington, MA), Christine Weston, and Paula Collins (MIT Lincoln Lab., Lexington, MA)

Accurately estimating noise dosage can be challenging for personnel who move through multiple noise environments. Dose estimates can also be confounded by the requirement to wear hearing protection in some areas, but not others. One concept for improved dose estimates under these conditions is to capture noise in-the-ear and on-body simultaneously. An additional benefit of this setup is that hearing protection fit can be assessed in real time. To evaluate this dual-microphone approach, we prototyped a noise dosimetry device for military environments where loud impulse noise such as weapons fire drives the need for high dynamic range and a high sampling rate. In this presentation, we describe a system where the in-ear microphone is acoustically coupled to a disposable foam or flange hearing protection eartip. Initial laboratory tests with a shock tube and acoustic test fixture (ATF) show more than 30 dB noise reduction between the on-body and in-ear microphones. Furthermore, in-ear levels are consistent with eardrum measurements in the ATF. One concern with body-worn dosimeters is their susceptibility to shock artifacts from microphone motion or handling. To address this issue, we also investigate co-locating an accelerometer with the on-body microphone to help remove shock artifacts. [Work supported by the Office of Naval Research.]

3pNSc6. An assessment of the permissible exposure limit for industrial complex noise exposure. Wei Qiu (SUNY Plattsburgh, 101 BRd. St., Plattsburgh, NY 12901, wei.qiu@plattsburgh.edu), Meibian Zhang, and Jianmin Jiang (Zhejiang Provincial Ctr. for Disease Control and Protection, Hangzhou, Zhejiang, China)

An 8-hr time-weighted average exposure of 85 dBA was adopted as the permissible exposure limit (PEL) by most current international standards for exposure to noise. Using the definition of material hearing impairment (MHI) as the average of hearing threshold levels for both ears exceeded 25 dB at 1, 2, 3, and 4 kHz, NIOSH estimated that the excess risk was 8% for workers exposed to an average noise level of 85 dBA over a 40-year working lifetime. However, this 85 dBA PEL was based on the data that was acquired from workers exposed to steady state (Gaussian) noises. In this study a database of 648 workers (age 36.9 yrs +/- 7.8, exposure duration 12.8 yrs +/- 8.0) exposed to non-Gaussian noises were used to assess the 85 dBA PEL. Among them 222 subjects exposed to noise at level of 80-84 dBA and 426 subjects at 85-89 dBA. Although the average duration of exposure from non-Gaussian database was much less than 40 years, the prevalence of MHI was 16.7% at level of 80-84 dBA and 31.5% at level of 84-89 dBA. The results show that 85 dBA PEL may not well protect hearing for workers exposed to non-Gaussian complex noise exposure.

3pNSc7. Real-time estimation of the passive attenuation for otoacoustic emission measurement probes. Vincent Nadon (ÉTS, 6298 rue d'Aragon, Montreal, QC H4E 3B1, Canada, vincent.nadon@etsmtl.ca) and Jeremie Voix (ÉTS, Montréal, QC, Canada)

Despite hearing loss prevention programs in place at work, noise-induced hearing loss (NIHL) remains the first reported occupational disease. To improve the detection of over-exposure to noise, an initial proof-of-concept for field monitoring of inner-ear health using otoacoustic emissions (OAE) was developed and successfully validated in laboratory conditions. However, in real-life situations, the unsupervised placement of the OAE probes remains a challenge: proper fit of the probe in the ear canal is required to ensure adequate hearing protection of the wearer, to improve the signal-to-noise ratio of OAE measurements and to maintain the proper calibration of stimuli signals in the case of the Distortion Product OAE (DPOAE) approach used. Algorithms were recently developed to quantify, in real-time, from the OAE probe microphones and receivers signals, the passive attenuation of the OAE probes. The validation of this approach was conducted, in laboratory conditions, on five human subjects exposed to industrial and pink noise recordings at realistic levels.

3pNSc8. Face the music: Classical music students and the sound of performance. Stephen Dance (School of the Built Environment and Architecture, London South Bank Univ., Borough Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk)

Since the implementation and enforcement of the European Union Physical Agents Directive (Noise) the Acoustics Group has collaborated with the Royal Academy of Music creating the noise team formed from administrators, scientists, and senior management. Our challenge was to allow these highly talented artists to practice, rehearse, and perform safely during their time at the Royal Academy of Music. This ten year project involved more than 3000 musicians measuring sound exposure of each instrument group and the hearing acuity of every student, as well as hearing surveillance of a sample of graduates. At each occurrence, the students were questioned as to their aural environment. The paper will focus upon the hearing acuity of undergraduates after studying music for a period of four years.

3pNSc9. Quiet rim driven ventilation fan design. Mark P. Hurtado, Daniel Wu, and Ricardo Burdisso (Virginia Polytechnic Inst. and State Univ. (Virginia Tech), 506 Broce Dr., Apt. 16, Blacksburg, VA 24060, phmark15@vt.edu)

Auxiliary ventilation fans are commonly used as a form of temperature and humidity control of working environments. However, they often generate high noise levels that cause permanent hearing impairment from prolonged exposure. Dominant noise sources from ventilation fans such as noise due to the aerodynamic interaction of the fan with the struts to support the hub motor and the blade-duct tip gap can be eliminated using a rim driven fan. However, tones from blade steady loading and thickness and the broadband airfoil self-noise of the fan are not eliminated. This noise is proportional to the 4-6th power of the tip speed. Hence, the approach here is to design a ring driven ventilation fan that reduces the fan tip speed while optimizing the blade profile to retain aerodynamic performance. The fan design has been 3D printed and implemented using a motor mounting system to validate the predicted performance. The motor mounting system integrates the fan design, a bell mouth shaped duct and a brushless DC motor. The experiments show good agreement with the predicted mechanical power, axial flow, and radiated noise over a range of speeds. Results suggest that a ring driven fan reduces noise while maintaining aerodynamic performance.

3pNSc10. Smart-phone application to collect ambient noise field-data for users, audiologists, and researchers. Roger M. Logan (12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

Could / should a smart-phone application be developed to measure and analyze social ambient noise that would: provide results to the user, store analyses locally for use by audiologists, and upload data for scientific research?

3pNSc11. The importance of attentive listening. Andrew S. Harris. (Andrew S. Harris, Inc., 26 Tappan St., Manchester, MA 01944, asharrisinc@gmail.com)

We all live in a world full of sounds. During my life, I have been aware of the sounds around me. Until I was about 23 years old, I experienced many noise environments and tried to remember how they sounded, but did not have any knowledge of acoustics. My knowledge of acoustics began when I took Bob Newman's architectural acoustics courses at MIT. It continued with work at BBN. Bob used examples of sounds, to illustrate his teaching. Bob stressed the importance of listening carefully and remember what you heard. If you need to analyze noise environments, it vitally important to listen very carefully and to remember the sounds. Always participate in noise measurements if you are responsible for reporting the noise and

recommending noise reduction treatments. During the 55 years since I began studying acoustics, there have been many changes in the issues we face. Focusing on the importance of careful listening, this paper will consider major sources of changes in the outdoor noise environment, i.e., increased numbers and kinds of sources; increased levels of activity; and increased desire for quiet.

3pNSc12. Social benefit analysis of reduced noise from electrical city buses in Gothenburg. Krister Larsson (Bldg. Technology/Sound & Vibrations, RISE Res. Institutes of Sweden, Box 857, Boras SE-50115, Sweden, krister.larsson@sp.se) and Maria Holmes (Environmental agency, City of Gothenburg, Gothenburg, Sweden)

The city of Gothenburg is the second largest city in Sweden and has an ambitious traffic strategy to increase the share of public transport substantially until 2035. At the same time the city is growing due to urbanization and the densification of the city leads to an anticipated growth in city bus traffic. However, noise from buses might lead to negative consequences for the citizens and electrical buses could be a way to reduce noise and emissions from the public transport system. In this study a comparison of noise levels and social costs of bus types with different powertrains are presented. Initially, noise emissions from three bus types were measured on a test track. The propulsion noise was extracted and coefficients for the Nord2000 Road prediction model were adapted. The Nord2000 model was used to calculate façade noise levels in the city center, as well as in a smaller focus area. The predicted noise levels were used to calculate health effects according to DALYs, as well as social costs according to ASEK. In addition, indoor maximum noise levels were calculated for typical façade cases based on the measurements. The results show that the largest benefits from electrical buses are obtained during acceleration, for example, at bus stops, and for maximum levels indoors. However, these situations are not taken appropriately into account in current social cost models.

3pNSc13. Numerical prediction of the broadband sound attenuation of a commercial earmuff: Impact of the cushion modeling. Kévin Carillo (Mech., École de Technologie Supérieure, 1100, rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, kevin.carillo.1@ens.etsmtl.ca), Franck C. Sgard (IRSST, Montreal, QC, Canada), and Olivier Doutres (Mech., École de Technologie Supérieure, Montréal, QC, Canada)

Passive earmuffs are commonly used when the sound level cannot be reduced at the source. They are mainly characterized by their sound attenuation that can be either measured or simulated. In this work, the sound attenuation of a commercial earmuff is calculated using a finite element model from 100 Hz to 5 kHz. Emphasis is put on the foam-filled cushion which is the trickiest component to model because of its physical complexity. This multiphasic cushion is modeled in a simplified way as an equivalent solid, either isotropic or transverse isotropic in order to take into account the added transverse stiffness due to the bulging of the cushion polymeric sheath. The accuracy of these models is investigated by comparison with measurements. The insertion loss (IL) predicted with the isotropic cushion model is highly underestimated between 500 Hz and 2.5 kHz due to the presence of an unrealistic mode of transverse deformation. It is found that (1) neglecting the acoustic excitation on the cushion's flanks of the isotropic model or (2) using the transverse isotropic cushion model significantly improves the simulated IL.

3pNSc14. Evaluation of noise exposures and hearing loss at a hammer forge company. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Scott E. Brueck, Judith Eisenberg (Hazard Evaluation and Tech. Assistance Branch, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH), Edward L. Zechmann (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH), and Edward F. Krieg (Statistics Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

The NIOSH health hazard evaluation program evaluated employees' exposures to high level continuous and impact noise at a hammer forge company. Personal dosimetry data were collected from 38 employees and noise exposure recordings were collected during two visits to the facility. Extensive audiometric records were reviewed and trends for hearing loss, threshold shifts and risk of hearing loss were assessed. The effectiveness of hearing protection devices for hammer forging was evaluated with an acoustic test fixture. A longitudinal analysis was conducted on the audiometric data set that included 4750 audiograms for 483 employees for the years 1981 to 2006. The analysis of the audiometric history for the employees showed that 82% had experienced a NIOSH-defined hearing threshold shift and 63% had experienced an OSHA-defined standard threshold shift. The mean number of years from a normal baseline audiogram to a threshold shift was about 5 years for a NIOSH threshold shift and was about 9 years for an OSHA threshold shift. Overall hearing levels among employees worsened with age and length of employment. The NIOSH audiometric test criteria in addition to OSHA threshold shift criteria to assess threshold shifts could provide an opportunity for early intervention to prevent future hearing loss.

Session 3pPAa

Physical Acoustics: Chains, Grains, and Origami Nonlinear Metamaterials

Julian D. Maynard, Cochair
Physics, Penn State University, 104 Davey Lab, Box 231, University Park, PA 16802

Vincent Tournat, Cochair
LAUM, CNRS, Université du Maine, Av. O. Messiaen, Le Mans 72085, France

Chair's Introduction—1:15

Invited Papers

1:20

3pPAa1. Amplitude-dependent shock wave propagation in three-dimensional microscale granular crystals. Morgan Hiraiwa and Nicholas Boechler (Dept. of Mech. Eng., Univ. of Washington, Mech. Eng. Bldg., Box 352600, Seattle, WA 98195, boechler@uw.edu)

Ordered arrays of elastic spherical particles in contact, often referred to as granular crystals, have been used as a model system for understanding the dynamics of granular media and explored as a type of designer material for acoustic wave tailoring applications. Due to the Hertzian interactions between the particles, granular crystals composed of macroscale particles have been shown to support strongly nonlinear phenomena including shock and solitary waves. In this presentation, we will describe recent progress in our studies of laser-generated shock wave propagation in self-assembled three-dimensional microscale granular crystals, where adhesive forces between the particles play a major role. Specific features studied include the dependence of the shock wave velocities and absorption on excitation amplitude. Dynamic failure processes such as crater formation and spallation are also explored. The experimental measurements are compared with reduced-dimension discrete element model simulations. New understanding of strongly nonlinear phenomena in microgranular media has potential applications to areas including energetic material dynamics, shock mitigation, and ultrasonic wave tailoring.

1:40

3pPAa2. Acoustics in disordered granular materials: From the particle scale to force networks. Karen Daniels (Dept. of Phys., North Carolina State Univ., Box 8202, Raleigh, NC 27695, kdaniel@ncsu.edu)

Granular materials are inherently heterogeneous, and continuum models of properties such as the shear modulus and sound speed often fail to quantitatively capture their dynamics. One likely reason, particularly in soft materials, is the transmission of forces via a chain-like network of strong forces. I will present several methods of characterizing sound transmission within such materials, combining both high-speed imaging and particle-scale piezoelectric measurements. In experiments on compressed granular materials, we observe that the amplitude of propagating sound is on average largest within particles experiencing the largest forces, due to the increased particle contact area. In addition, we find that the particle-scale density of vibrational modes exhibits systematic changes associated with the amount of compression, shear strain, or disorder in the system. We have found that a helpful theoretical framework is to consider a network representation in which the nodes (particles) are connected by weighted edges obtained from contact forces (visualized using photoelastic materials). Such network representations provide a mathematical framework which allows for the comparison of features and dynamics at different spatial scales. Collectively, these measurements highlight the importance of this force chain network in controlling the bulk properties of the material.

2:00

3pPAa3. Modelling the transmission of ultrasound pulses through grain-to-grain contacts. Bart Van Damme (Acoustics/Noise Control, Empa, Ueberlandstrasse 129, Dübendorf 8600, Switzerland, bart.vandamme@empa.ch)

Wave propagation through packed spherical particles is characterized by two distinct mechanisms, depending on the frequency of the wave content. Coherent wave pulses occur when the Hertzian contact model can be used, i.e., for frequencies low enough so that the granules behave as rigid bodies. Above a certain frequency, a chaotic time signal is the result of diffusive energy transmission through the grain contacts. This so-called coda wave is important for applications tracking the microstructure of materials, e.g., in nondestructive testing. This work looks for parallels between the two approaches by investigating the transmission of short ultrasound pulses in the unit cell of a granular material: two spheres in contact. Laser measurements on two identical large steel spheres show that the energy transmission happens in discrete steps, due to guided surface waves. The measured and modeled energy distribution evolution are similar to the one predicted by the diffusive theory. However, we show that the Hertz contact law can be applied locally in the region of the contact to quantify the pulse transmission, despite the fact that the entire sphere no longer behaves as a rigid body. This approach allows for the design of a configurable non-linear waveguide.

2:20

3pPAa4. Nonlinear elastic waves in architected soft solids. Bolei Deng (SEAS, Harvard Univ., Cambridge, MA), Jordan R. Raney (Dept. of Mech. Eng. and Appl. Mech., Univ. of Pennsylvania, Philadelphia, PA), Katia Bertoldi (SEAS, Harvard Univ., Cambridge, MA), and Vincent Tournat (LAUM, CNRS, Université du Maine, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr)

In this presentation, we analyze different regimes of elastic wave propagation in a family of architected soft solids, the rotating square structures, known to exhibit negative Poisson ratio. We show that it is possible, via a discrete model of finite size masses coupled by soft and highly deformable elastic ligaments, to describe the nonlinear wave propagation of displacement and rotation modes in a two-dimensional configuration. In turn, the geometrical characteristics and local elastic parameters of the architected structures can be put in correspondence with the dispersive and nonlinear wave properties, thus allowing for the dispersion and nonlinearity management. By exploring several designs and the influence of geometry, we show that the parameters of the governing nonlinear wave equations can be controlled, and even the type of governing equation and their dominant nonlinearity can be modified. In particular, we report that for several studied configurations, vector elastic solitons are predicted and experimentally observed. These results could be useful for the design of nonlinear elastic metamaterials, aiming at controlling high amplitude vibrations and elastic waves, or achieving amplitude dependent operations on waves.

2:40

3pPAa5. Transformable origami-inspired acoustic waveguides. Katia Bertoldi (Harvard Univ., 29 Oxford St., Cambridge, MA 02138, bertoldi@seas.harvard.edu), Vincent Tournat (Harvard Univ., Le Mans, France), Sahab Babaee, and Johannes Overvelde (Harvard Univ., Cambridge, MA)

Using foldable origami-like structures, we design reconfigurable and switchable acoustic waveguides composed of interconnected periodic arrays of hollow tubes to manipulate and guide sound. We demonstrate both numerically and experimentally that upon application of external deformation, the structure is folded and transformed to one-, two-, and three-dimensional waveguide in which sound waves travel through the tubes. The proposed design expands the ability of existing acoustic waveguides by enabling tunability induced by reversible deformation and folding shape transformation over a wide range of frequencies, opening avenues for the design of novel tunable waveguides and adaptive sound filters.

Contributed Papers

3:00

3pPAa6. Numerical investigation on the effect of structure parameter of acoustic field rotator on its operating bandwidth. Xiuhai Zhang, Zhiguo Qu, Di Tian, and Zhi Liu (Key Lab. of Thermo-Fluid Sci. and Eng. of Ministry of Education, School of Energy and Power Eng., Xi'an Jiaotong Univ., No.28, Xianning West Rd., Xi'an 710049, China, zhangxiuhai000@stu.xjtu.edu.cn)

The theory about transformation acoustics is introduced to indicate that designing an acoustic field rotator, in reality, is feasible. Then, numerical research is conducted to investigate the effect of geometrical parameters on the high cut-off frequency, including the size of building block, the aspect ratio of building block, the size of acoustic field rotator, the external diameter of acoustic field rotator, and the internal diameter of acoustic field rotator. The results suggest that the sizes of building blocks of acoustic field rotators are inversely proportional to their high cut-off frequencies in the studied frequencies. The effect of the size of acoustic field rotator mainly lies in the size of building block instead of the number of the building block. There is a linear relationship between the aspect ratio of the building block and high cut-off frequency in the studied frequencies. In addition, the layer of building block has an effect on high cut-off frequency as well. The current numerical study is helpful to give a better guidance for designing the acoustic field rotator.

3:20

3pPAa7. Comparison of results for an analogous acoustical and optical scattering problem: The double sphere. Cleon E. Dean (Phys., Georgia Southern Univ., PO Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu) and James P. Braselton (Mathematical Sci., Georgia Southern Univ., Statesboro, GA)

A comparison of results for an analogous scattering problem is made for the interactive scattering of a double sphere in both acoustics and electromagnetic scattering. For initial testing purposes, the problem is simplified to that of two identical spheres with the same size and physical properties. The acoustical scatterer is modeled to match optical characteristics of the already studied optical scattering problem [G. W. Kattawar and C. E. Dean, Opt. Lett. **8**, 48-50 (1983).]. Particular study is made of situations that lead to large side scattering responses due to resonance phenomena. The model for the acoustical scattering problem was discussed in a previous Acoustical Society of America meeting [C. E. Dean and J. P. Braselton, J. Acoust. Soc. Am. **139**, 1121 (2016)].

Session 3pPAb

Physical Acoustics: General Topics in Physical Acoustics II

Brent O. Reichman, Cochair
Brigham Young University, 453 E 1980 N, #B, Provo, UT 84604

Sarah M. Young, Cochair
Physics, Brigham Young University-Idaho, ESC N203, Provo, UT 84601

Contributed Papers

1:20

3pPAb1. Time-resolved dynamics of micrometer-sized bubbles undergoing shape oscillations. Matthieu Guédra (Univ. Lyon, Université Lyon 1, INSERM, LabTAU, F-69003, LYON, France, 151 cours Albert Thomas, Lyon 69424, France, matthieu.guedra@inserm.fr), Sarah Cleve (Univ. Lyon, Ecole Centrale de Lyon, INSA de Lyon, CNRS, LMFA UMR 5509, F-69134 Ecully CEDEX, VILLEURBANNE CEDEX, France), Cyril Mauger (Univ. Lyon, Ecole Centrale de Lyon, INSA de Lyon, CNRS, LMFA UMR 5509, F-69134 Ecully CEDEX, Lyon, France), Claude Inserra (Univ. Lyon, Université Lyon 1, INSERM, LabTAU, F-69003, LYON, France, Lyon, France), and Philippe Blanc-Benon (Univ. Lyon, Ecole Centrale de Lyon, INSA de Lyon, CNRS, LMFA UMR 5509, F-69134 Ecully CEDEX, Ecully, France)

Bubbles in a cavitation cloud can exhibit shape oscillations when insonified at large acoustic pressure amplitudes, which are generally connected to collective effects of the bubbles population (subharmonic emission, erratic motions of bubbles, coalescence, or fragmentation). The onset conditions and the oscillating properties of such shape modes are analyzed experimentally through highly time-resolved dynamics of micrometric bubbles. Single bubbles of radius ranging from 30 to 80 micrometers are nucleated with a laser pulse, trapped in a 30kHz ultrasonic field and imaged using two triggered CCD cameras at an acquisition rate of 180 kfps. A large parametric analysis over the bubbles radius and driving amplitude allows to recover the stable / unstable areas of the parametrically excited shape modes. Experimental evidence of nonzonal harmonics are reported, together with nonlinear modal interactions for sufficiently high driving amplitudes and large shape deformations, which are highlighted through (1) the subsequent excitation of nonresonant shape modes, (2) the trigger of the translational motion of the bubble, and (3) an alteration of the spherical response of the bubble. [Work supported by the French National Research Agency, LabEx CeLyA (ANR-10-LABX-0060) and granted by the ANR-MOST project CARIBBBOU (ANR-15-CE19-0003).]

1:40

3pPAb2. Triggering of surface modes by bubble coalescence at high pressure amplitudes. Sarah Cleve (Univ. Lyon, Ecole Centrale de Lyon, INSA de Lyon, CNRS, LMFA UMR 5509, F-69134 Ecully CEDEX, Lyon, FRANCE, 20 Ave. A. Einstein, VILLEURBANNE CEDEX 69621, France, sarah.cleve@ec-lyon.fr), Matthieu Guédra (Univ. Lyon, Université Lyon 1, INSERM, LabTAU, F-69003, LYON, France, Lyon, FRANCE, Lyon, France), Cyril Mauger (Univ. Lyon, Ecole Centrale de Lyon, INSA de Lyon, CNRS, LMFA UMR 5509, F-69134 Ecully CEDEX, Lyon, FRANCE, Lyon, France), Claude Inserra (Univ. Lyon, Université Lyon 1, INSERM, LabTAU, F-69003, LYON, France, Lyon, FRANCE, Lyon, France), and Philippe Blanc-Benon (Univ. Lyon, Ecole Centrale de Lyon, INSA de Lyon, CNRS, LMFA UMR 5509, F-69134 Ecully CEDEX, Lyon, FRANCE, Ecully, France)

In order to study surface instabilities of small bubbles, a single spherical bubble is usually trapped in an ultrasound field and submitted to increasing acoustic pressure until reaching the necessary threshold. In the current work, coalescence between two bubbles is used as a trigger for non-spherical oscillations. Experiments are conducted in water with air bubbles of radii ranking from of 10 to 80 μm, at a driving frequency of 30 kHz and captured at 67 kHz. While most literature deals with bubble coalescence at relatively low pressure amplitudes implying spherical bubbles, coalescence at high pressure amplitudes (in the present case up to 30 kPa) leads to surface instabilities during and after the coalescence. During the impact and the immediately following oscillations, transitory surface deformations appear. After this transitory period, the bubbles stabilize either being in a purely spherical oscillation mode or exhibiting a stable surface mode. We analyze under which conditions either of the two cases applies. Lastly, apart from conventional coalescence of two bubbles becoming one single bubble, observations of long-lasting bouncing of a pair of bubbles are reported. [Work supported by the LabEx Centre Lyonnais d'Acoustique of the Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ANR-11-IDEX-0007).]

2:00

3pPAb3. The effects of finite amplitude drop shape oscillations on the inference of material properties. Vahideh Ansari Hosseinzadeh and Ray Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, vansari@bu.edu)

Acoustic levitation of drops provides a non-contact means of isolation, stable positioning and static and dynamic manipulation. Despite a long history of the use of drop shape oscillations to infer both surface and bulk material properties, and an equally long history of observing behaviors that result in incorrect inference of material properties, researchers continue to employ finite amplitude oscillations for their experimental observations. In this study we use the inference of surface tension and viscosity of known samples of glycerin-water mixtures to illustrate the fact that finite-amplitude oscillations are the dominant mechanism for many reports of ill-behaved drop dynamics, including modal peak-splitting, vorticity, and strong field-

coupling. Since this "bad" behavior leads to incorrect inferences of surface tension and viscosity, we show that small amplitude oscillations yields recovery of correct inferences for these properties for known samples. Finally, we show results from experiments with bovine and human blood. [Work supported by NSF grant # 1438569.]

2:20

3pPAb4. To predict a thermoacoustic engine's limit cycle from its impedance measurement. Valentin Zorgnotti, Guillaume Penelet, Gaëlle Poignand (LAUM, 17 rue du Dr Leroy, Le Mans 72000, France, valentin.zorgnotti@univ-lemans.fr), and Steven L. Garrett (Grad. Prog. in Acoust., Penn. State, State College, PA)

Thermoacoustic engines are self-oscillating systems converting thermal energy into acoustic waves. Recent studies on such engines highlight much nonlinear effects responsible for the engine's saturation, leading to a limit cycle, which can be stable or not. Those effects however, are not sufficiently known, even with today knowledges, to accurately predict a limit cycle oscillations amplitude. This work suggests a new approach, based on acoustic impedance measurement at large forcing amplitudes, to predict the limit amplitude in steady state for a given engine. This method allows one to predict an engine's saturation amplitude without studying in detail its intern geometry. In the case of a quarter wave length engine, its input impedance can easily be obtained from an impedance sensor for example. Increasing the speaker's forcing leads to a nonlinear impedance depending on the acoustic field amplitude. Once measured, this function contains information such as the limit cycle amplitude, its stability and the engine's efficiency depending on parameters as the applied heating and stack's position. In the case of a standing wave prime mover, this study shows first results obtained from this method. Later on, this work should lead to steady state predictions for any, unknown, given engine.

2:40

3pPAb5. Acoustic frequency splitting in thermoacoustically driven coupled oscillators. Bonnie Andersen, Jacob H. Wright, Cory J. Heward, Emily R. Jensen, and Justin S. Bridge (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84057, bonniem@uvu.edu)

Frequency splitting, or level repulsion, occurs near the point where two resonant modes of coupled oscillators intersect as one parameter is varied such that the resonance of one passes through the resonance of the other. A thermacoustic stack, which provides internal self-sustained oscillations, placed inside the neck of a closed bottle-shaped resonator can set up standing waves of the coupled neck-cavity system. The neck behaves as a quarter-wave resonator because it is closed at the top of the bottle and open at the bottom where it is attached to the cavity. The cavity being closed at the bottom and mostly closed near the neck behaves as a half-wave resonator. A one-dimensional wave equation with appropriately applied boundary conditions is used to generate solutions of the coupled neck-cavity system. These solutions reveal mode splitting near the intersections of the uncoupled neck and cavity modes. Thermoacoustic engines with bottle-shaped resonators were tested while varying one of three geometric parameters: the neck length, the cavity length, and the cavity radius. Graphs of the coupled solutions readily illustrate mode splitting of the coupled oscillator system and in agreement with experimental results.

3:00

3pPAb6. Numerical simulation of key linear alternator performance indicators under thermoacoustic-power-conversion conditions. Ahmed Y. Abdelwahed (Mech. Dept., The American Univ. in Cairo, School of Sci. & Eng., School of Sci. and Eng., American University in Cairo, 74 South 90th St., Fifth Settlement, New Cairo, Cairo 11835, Egypt, ahmed_yassin@aucegypt.edu), A. H. Ibrahim Essawey (The American Univ. in Cairo, School of Sci. & Eng., 11835 New Cairo, Cairo, Egypt.On leave from Mech. Power Dept., Faculty of Eng., Cairo Univ., Egypt, New Cairo, Egypt), and Ehab Abdel-Rahman (Professor of Phys., Dept. of Phys., The American Univ. in Cairo, New Cairo, Cairo, Egypt)

Thermoacoustic power converters consist of thermoacoustic heat engine and linear alternator. The linear alternator converts the acoustic power generated by the thermoacoustic engine to electric output. Efficient and stable operation of a thermoacoustic power converter requires acoustic matching between the engine and the alternator. It also requires matching between the linear alternator and the connected load. An experimental setup was built to measure and analyze the linear alternator performance under different thermoacoustic power converter operating conditions. The effects of the different design and operation factors on the key linear alternator performance parameters such as mechanical stroke, the generated electric power, the acoustic-to-electric conversion efficiency, the mechanical motion loss, the electric loss, and the fluid-seal loss were investigated experimentally and numerically. The experimental results were simulated using DeltaEC and reasonable agreement was obtained.

3:20

3pPAb7. Effect of RC load on the performance of a three-stage looped thermoacoustic engine. Tao Jin (Inst. of Refrigeration and Cryogenics, Zhejiang Univ.;Key Lab. of Refrigeration and Cryogenic Technol. of Zhejiang Province, Rd. 38 West Lake District, Zhejiang University, Yuquan Campus, Hangzhou, Zhejiang 310027, China, jintao@zju.edu.cn), Yi Wang, Rui Yang, Jingqi Tan, and Ye Feng (Inst. of Refrigeration and Cryogenics, Zhejiang Univ., Hangzhou, Zhejiang, China)

Thermoacoustic heat engine is a type of machine converting thermal energy into acoustic energy with the attracting characteristics of high reliability and environmental friendliness. This work proposed a three-stage looped thermoacoustic engine, where a compliance tube is utilized as the phase adjuster, to realize the high acoustic impedance and the near traveling-wave acoustic field in the regenerator, which is significant for effective thermoacoustic conversion. In order to investigate its performance with low-grade thermal energy, the output acoustic power of the engine is measured with the variable load method. It can be found that as the resistance decreases, the efficiency of the engine may rise, but the required heating temperature also rises. Thus, there exists a trade-off between the efficiency and the heating temperature, and the relative Carnot efficiency is adopted as a main index to evaluate the performance of the system. The maximum relative Carnot efficiency of 12.6% (the corresponding efficiency was 3.2%) was achieved in our experiments, when the heating temperatures of the three stages were 120 °C.

Session 3pPAc

Physical Acoustics: Topics in Physical Acoustics (Poster Session)

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

All posters will be on display from 1:20 p.m. to 3:40 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:20 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 3:40 p.m.

Contributed Papers

3pPAc1. Metal scaffolds for acoustic trapping of polystyrene particles in water suspensions. Iciar Gonzalez and Zuriñe Bonilla del Rio (Consejo Superior de Investigaciones Científicas CSIC, Serrano 144, Madrid 28006, Spain, iciar.gonzalez@csic.es)

The concept of ultrasonic 3D caging is based in wells designed to host three orthogonal half-wave acoustic resonances defined by its three dimensions generating a single-pressure-node in the center where particles collect to aggregate. A replacement of the planar shapes in these cavities by curved walls introduce changes in the pressure patterns, generating different effects on the particles exposed to the acoustic field. Here we present an experimental study of the acoustic behavior of aqueous suspensions of micron-sized polystyrene particles (Cv~1%) exposed to ultrasounds at a frequency f~1MHz in a metal scaffold made up of stainless steel wires crossed-linked orthogonally to form a mesh with a light of 1mm (something larger than $\lambda/2$). Once applied the acoustic field, the particles are rapidly attracted by the rods of the mesh distances much larger than their diameter ($\Phi_p = 6\mu m$), where are trapped and remain adhered to the scaffold during the acoustic actuation and even later, providing stable aggregates. Hydrodynamic mechanisms associated to viscous disturbances and mutual radiation pressure induced by the metal rods could be responsible of this massive trapping effect on the polymeric particles according to previous theoretical and experimental studies carried by the authors in aerosols.

3pPAc2. Improving infrasonic location estimates for underground nuclear explosions Improving infrasonic location estimates for underground nuclear explosions. Fransiska K. Dannemann, Philip Blom (Los Alamos National Lab., P.O. Box 1663, MS D446, Los Alamos, NM 87545, fransiska@lanl.gov), Junghyun Park (Southern Methodist Univ., Dallas, TX), Omar Marcillo (Los Alamos National Lab., Los Alamos, NM), Brian W. Stump (Southern Methodist Univ., Dallas, TX), and Il-Young Che (Korean Inst. of GeoSci. and Mineral Resources, Daejeon, South Korea)

Infrasound data from underground nuclear explosions conducted by North Korea in 2006, 2009, 2013 and 2016 were recorded on six seismo-acoustic arrays co-operated by Southern Methodist University (SMU) and the Korean Institute of Geosciences and Mineral Resources (KIGAM). No infrasound signals were observed during the 2006 test, while signals from the others have been used to determine event locations and yield estimations. Prior location studies have demonstrated that wind corrections for back azimuth deviation improve location estimates. Additionally, recent improvements to the Bayesian Infrasonic Source Localization (BISL) methodology have shown to reduce 90% confidence contours for location by 40% through the utilization of propagation-based likelihood priors for celerity and backazimuth deviation from seven years of archival atmospheric specifications. Relocations of the 2009, 2013 and 2016 nuclear explosions will be presented to demonstrate the application of BISL to underground nuclear explosions.

3pPAc3. Seismo-Acoustic numerical simulation of North Korea nuclear tests. Gil Averbuch (Dept. of GeoSci. and Eng., Delft Univ. of Technol., Graswinckelstraat 64, Delft 2613 PX, Netherlands, g.averbuch@tudelft.nl), Jelle D. Assink, Pieter S. Smets, and Läslo G. Evers (R&D Dept. of Seismology and Acoust., KNMI, De Bilt, Netherlands)

A seismo-acoustic event is an event in which seismic energy transfers to acoustic energy in the oceans and/or atmosphere and vice versa. Although measurements confirms the coupling between the mediums, a numerical investigation of these events may provide us with a deeper understanding of the phenomena. In this presentation, a finite element, elasto-acoustic Fast Field Program (FFP) is presented and is applied to the recent 2013 and January 6th, 2016 North Korea underground nuclear tests. The aim of this study is to model the elastic and acoustic wave fields of these events and use this information to estimate the source depths. The modeled depths will be compared to those from seismo-acoustical observations.

3pPAc4. Photoacoustic effect of ethene: Sound generation due to plant hormone gasses. David W. Ide and Han Jung Park (Chemistry, Univ. of Tennessee at Chattanooga, 615 McCallie Ave., Chattanooga, TN 37403, hanjung-park@utc.edu)

Ethene (C_2H_4), which is produced in plants as they mature, was used to study its photoacoustic properties using photoacoustic spectroscopy. Detection of trace amounts, with N_2 gas, of C_2H_4 gas was also applied. The gas was tested in various conditions- temperature, concentration of the gas, gas cell length, and power of the laser to determine their effect on the photoacoustic signal, the ideal conditions to detect trace gas amounts, and concentration of C_2H_4 produced by an avocado and banana. A detection limit of 10 ppm was determined for pure C_2H_4. A detection of 5% and 13% (by volume) concentration of C_2H_4 produced for a ripening avocado and banana, respectively, in closed space.

3pPAc5. Cavitation detection in a nozzle flow. Huy K. Do, Purity Dele-Oni, Tony Tang, Daniel Poe, James Bird, Sheryl Grace, Emily Ryan, and Ray Holt (Mech. Eng., Boston Univ., 44 Parker Hill Ave., Apt. 2, Boston, MA 02120, xhuydo@bu.edu)

The presence of cavitation inside fuel injector nozzles has been linked not only to damage associated with cavity collapse near the walls, but also more intriguingly to improved spray atomization. Previous studies have shown that cavitation is associated with increased spray angle. Our goal is to investigate the underlying mechanics. In this talk we describe our initial efforts to employ both acoustic techniques (passive cavitation detection, or PCD) and optical techniques (optical cavitation detection, or OCD) to characterize nozzle cavitation. Experiments are conducted with acrylic nozzles of various geometry. Unfocused single element transducers are used for

PCD, while digital imaging is used for OCD. Cavitation onset thresholds and development are studied as functions of flow rate, nozzle geometry, and upstream fluid preparation. Cavitation characterization results will be compared with an in-house computational code being developed to model cavitation in fuel injectors.

3pPAc6. A transfer function approach for measuring the characteristic impedance and propagation constant of porous materials. Zhehao Huang and Xiaolin Wang (Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxi Rd., Beijing 100190, China, wangxl@mail.ioa.ac.cn)

A four-microphone one-load transfer function (4M1L) method using an impedance tube is proposed for estimating the characteristic impedance and propagation constant of porous materials, referring to the standard two- and four-microphone methods (ISO 10534-2, ASTM E2611-09). The material in this single measurement method does not have to be geometrically symmetrical and homogeneous. Even the material is geometrically asymmetrical and inhomogeneous, like a multi-layered system, an equivalent-impedance model can be assumed. Moreover, this method can also be used in the presence of a mean flow. First, in this presentation, the measuring theories of conventional two-microphone transfer function (2M2L), three- and four-microphone transfer matrix (3M2L, 4M2L) methods are discussed and compared with this 4M1L method,. Second, a direct measurement for various multi-layered materials mounted with a hard termination is used to verify this measurement method and material characterization by the equivalent-impedance model. Last, measurement and calculation results are presented and compared with the conventional 4M2L method. The 4M1L method can be of help in acoustic properties measurement of sound absorbing materials.

3pPAc7. Fiber maps from acoustic anisotropy in rodent cardiac tissue. Michelle L. Milne (Phys., St. Mary's College of Maryland, 47645 College Dr., St. Mary's City, MD 20686-3001, mlmilne@smcm.edu) and Charles S. Chung (Physiol., Wayne State Univ., Detroit, MI)

Previous studies have demonstrated that 3D myocardial fiber maps of excised large mammal hearts can be generated from ultrasonic images by utilizing the acoustic anisotropy of cardiac tissue. The goal of this paper is to demonstrate that detection of acoustic anisotropy and the creation of myocardial fiber maps using ultrasound is also feasible in rodents. Acoustic anisotropy of rat myocardium was confirmed using 2-mm diameter cores taken from the left-ventricular free wall using a 21 MHz probe (VisualSonics Vevo210) in B-mode. The relationship between fiber orientation and ultrasonic backscatter was obtained. These data were confirmed in segments of left ventricular free wall that were scanned and subsequently histologically sectioned serially from the epi- to endo-cardium. Subsequently, a series of long-axis images were taken from intact rat hearts to generate 3D rodent fiber maps. Preliminary data from mouse hearts using a 40 MHz probe produces similar results. We conclude that it is feasible to obtain a cardiac fiber map from ex vivo rodent hearts using echocardiography. Further development of this method may allow for *in-vivo* fiber direction analysis in live rodents. [Work supported by the American Heart Association (14SDG20100063 to CSC) and a grant from The Patuxent Partnership (MLM).]

3pPAc8. Rayleigh surface wave in a porothermoelastic solid half-space. Baljeet Singh (Dept. of Mathematics, Post Graduate Government College, Sector 11, Chandigarh 160011, India, bsinghgc11@gmail.com)

In the present paper, the Rayleigh wave at a stress free thermally insulated surface of a generalized porothermoelastic solid half-space is considered. The governing equations of generalized porothermoelasticity are solved for general surface wave solution. The particular solutions satisfying the required radiation conditions are obtained. These solutions are applied to boundary conditions at stress free thermally insulated surface. In order to satisfy the relevant boundary conditions, a secular equation for wave speed of Rayleigh wave is obtained. Numerical simulations are done using an experimental data given by Yew and Jogi (1976) [J. Acoust. Soc. Am. 60, 2-8]. The wave speed is computed to observe the effects of frequency, porosity, coefficients of thermal expansion, and thermoelastic coupling coefficients.

3pPAc9. Bianisotropic acoustic metasurfaces for independent control of reflection and transmission wave. Choonlae Cho and Namkyoo Park (Elec. and Comput. Eng., Seoul National Univ., Bldg. 301-913, 1 Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, lemon03@snu.ac.kr)

We propose bianisotropic acoustic metasurfaces which manipulate reflection and transmission wave-front independently. We design one-dimensional bianisotropic meta-atom controlling density (ρ), inverse of bulk modulus (B^{-1}) and bianisotropy (ξ) near zero-index point in water, based on separation of characteristic oscillations. Through derivation of the density, bulk modulus and bianisotropy as a function of S-parameters, we show full-access to the reflection and transmission amplitude and phase in the tailored bianisotropic media. Utilizing bianisotropic meta-atom, we numerically demonstrate independent control of reflection and transmission wave. Furthermore, design process of the metasurfaces of independent manipulation of forward- and backward- reflection waves is presented.

3pPAc10. Generation of ultrasonic finite-amplitude waves through a multiple scattering medium by time reversal in a waveguide. Gonzalo A. Garay, Nicolás Benech, Carlos Negreira, and Yamil Abraham (Instituto de Física, Facultad de Ciencias, UdelaR, Iguá 4225, Montevideo, Montevideo 11400, Uruguay, ggaray@fisica.edu.uy)

Time-reversal process has been studied and applied in several acoustical systems. Two particular systems have gain our attention: an acoustical waveguide and a multiple scattering medium. Applied in a waveguide, time-reversal process can be employed to generate finite-amplitude waves by using low-power electronics. We used seven low-power ultrasonic transducers attached to the waveguide. After a 1-bit time-reversal, the waveform clearly shows time-trace distortion typical of shock waves. We analyzed the spectrum of the focal spot. As expected in a non-linear regime, it showed higher order harmonics. We find that for low input amplitude levels, the second harmonic have a lower amplitude in the focal point than in the surrounding region. As the input amplitude increase, the second harmonic's amplitude in the focal point reaches the maximum level. In a second stage, we interpose a multiple scattering medium between the guide and the receiver. Its width was larger than the mean free path. The nonlinear wave is still present but with lower amplitude. However, we observe a narrower focal spot with reduced side-lobes. Thus, the multiple scattering medium improves the quality of acoustic focalization while it still allows the formation of shock waves.

3pPAc11. Finite element models of crystallized white dwarf cores: A gateway to undergraduate physical acoustics and computational modeling of complex systems. Kenneth A. Pestka II, Robert C. Highley, and Laura K. Deale (Chemistry and Phys., Longwood Univ., 201 High St., Farmville, VA 23909, pestkaka@longwood.edu)

In this work we present details of several finite element (FE) models of white dwarf stars with cores composed of layered crystalline carbon, oxygen and neon. The FE models, produced by undergraduate physics majors, are constructed using a commercially available software package Femap with NX Nastran. These models can be used to understand the effect of stellar composition and crystallization ratios on white dwarf behavior including vibrational modes, surface velocity and variation in luminosity. While the nature of these ultra-dense stellar remnants can appear quite exotic, the physical acoustic principles required to build and analyze the FE models are directly related to those commonly utilized by research acousticians. This gateway project was designed for undergraduate physics majors in order to illustrate this connection and to encourage interest in applied physical acoustics. The inherent interdisciplinary nature of the project also provides an opportunity for undergraduate physics majors to explore fields that are often considered disparate while improving their computational modeling skills.

3pPAc12. Inhibition of Rayleigh-Bénard convection through acceleration modulation for thermoacoustic devices. Anand Swaminathan (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, azs5363@psu.edu), Steven L. Garrett (151 Sycamore Dr., State College, PA), and Robert W. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA)

The ability to dynamically stabilize Rayleigh-Bénard convection using acceleration modulation is of interest to groups who design and study thermoacoustic machines, as the introduction of unwanted convection can have deleterious effects on the desired operation and efficiency of the device. These performance issues caused by suspected convective instability have been seen both in traveling wave thermoacoustic refrigerators and in cryogenic pulse tube chillers. This presentation reports the results of an experiment intended to determine the vibratory, fluidic, and geometric conditions under which a small, rectangular container of statically unstable fluid may be stabilized by vertical vibration, evaluating the computational methods of R. M. Carbo [J. Acoust. Soc. Am. **135** 654 (2014)]. Measurements are obtained using a long-displacement kinematic shaker of a unique design with the convecting gas characterized using both thermal transport measurements and flow visualization employing tracer particles illuminated by a diode laser light sheet phase-locked to the shaker. [Work supported by the Julian Schwinger Foundation for Physics Research, the Pennsylvania Space Grant Consortium Graduate Research Fellowship, and the Paul S. Veneklasen Research Foundation.]

3pPAc13. Computation of nonlinear acoustic waves using smoothed particle hydrodynamics. Yong Ou Zhang (School of Transportation, Wuhan Univ. of Technol., Wuhan, China), Sheng Wang (School of Automotive Eng., Wuhan Univ. of Technol., Wuhan, China), Zhixiong Gong (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Webster Physical Sci. 754, Pullman, WA 99164-2814, Pullman, Washington 99164-2814, zhixiong.gong@wsu.edu), Tao Zhang, Tianyun Li (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei Province, China), and Qing Zhi Hou (School of Comput. Sci. and Technol., Tianjin Univ., Tianjin, China)

A Lagrangian approach for solving nonlinear acoustic wave problems is presented with direct computation from smoothed particle hydrodynamics. The traditional smoothed particle hydrodynamics method has been applied to solve linear acoustic wave propagations. However, nonlinear acoustic problems are common in medical ultrasonography, sonic boom research, and acoustic levitation. Smoothed particle hydrodynamics is a Lagrangian meshfree particle method that shows advantages in modeling nonlinear phenomena, such as the shock tube problem, and other nonlinear problems with material separation or deformable boundaries. The method is used to solve the governing equations of fluid dynamics for simulating nonlinear acoustics. The present work also tests the method in solving the nonlinear simple wave equation based on Burgers' equation. Effects of initial particle spacing, kernel length, and time step are then discussed based on the wave propagation simulation. Different kernel functions are also evaluated. The results of numerical experiments are compared with the exact solution to confirm the accuracy, convergence, and efficiency of the Lagrangian smoothed particle hydrodynamics method.

3pPAc14. State changes in lipid interfaces observed during cavitation. Shamit Shrivastava and Robin Cleveland (Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, shamit.shrivastava@eng.ox.ac.uk)

Here we investigate the cavitation phenomenon at a lipid interface of multilaminar vesicles (MLVs) subjected to acoustic shock waves. The lipid membranes contain a fluorescent dye, Laurdan, which produces a fluorescence emission sensitive to the thermodynamic state of the interface. Fluorescence emissions were measured at 438nm and 470nm using two photomultiplier tubes (with 8 MHz bandwidth) from which the temporal evolution of the interface's thermodynamic state was determined with submicrosecond resolution. Acoustic emissions were recorded simultaneously in order to detect the presence of cavitation. Different lipids were used to prepare the MLVs in order to observe cavitation phenomenon as a function of the state of the interface. It was deduced that the interface behaves as an adiabatic system decoupled from the bulk, where the entropy increase due to vaporization during cavitation is compensated by the entropy decrease resulting from condensation and dehydration of the lipids. These results show that cavitation physics critically depends on the thermodynamics of the interface. While applied here on a simple system of pure lipid MLVs, the thermodynamic approach is applicable to native biological membranes and cavitation phenomenon in general. [Work supported by UK EPSRC EP/L024012/1.]

3pPAc15. Lagrangian meshfree particle method for modeling acoustic wave propagation in moving fluid. Yong Ou Zhang (School of Transportation, Wuhan Univ. of Technol., Wuhan, Hubei, China), Qing Zhi Hou (School of Comput. Sci. and Technol., Tianjin Univ., Tianjin, China), Zhixiong Gong (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Webster Physical Sci. 754, Pullman, WA 99164-2814, Pullman, Washington 99164-2814, zhixiong.gong@wsu.edu), Tao Zhang, Tianyun Li (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei Province, China), Jian Guo Wei (School of Software Eng., Tianjin Univ., Tianjin, China), and Jian Wu Dang (School of Comput. Sci. and Technol., Tianjin Univ., Tianjin, China)

Introducing the Lagrangian approach to acoustic simulation is supposed to reduce the difficulty in solving problems with deformable boundaries, complex topologies, or multiphase media. Specific examples are sound generation in the vocal track and bubble acoustics. As a Lagrangian meshfree particle method, the traditional smoothed particle hydrodynamics (SPH) method has been applied in acoustic computation but in a quiescent medium. This study presents two Lagrangian approaches for modeling sound propagation in moving fluid. In the first approach, which can be regarded as a direct numerical simulation method, both standard SPH and the corrective smoothed particle method (CSPM) are utilized to solve the fluid dynamic equations and obtain pressure change directly. In the second approach, both SPH and CSPM are used to solve the Lagrangian acoustic perturbation equations; the particle motion and the acoustic perturbation are separated and controlled by two sets of governing equations. Subsequently, sound propagation in flows with different Mach numbers is simulated with several boundary conditions including the perfected matched layers. Computational results show clear Doppler effects. The two Lagrangian approaches demonstrate convergence with exact solutions, and the different boundary conditions are validated to be effective.

3pPAc16. Diffraction effects of an acoustic beam propagating through a flowing medium. Kjell E. Frøysa (Elec. Eng., Western Norway Univ. of Appl. Sci., Postbox 7030, Bergen 5020, Norway, kef@hvl.no)

Ultrasonic transit time difference flow meters are today industrially accepted for custody transfer measurements of oil and for natural gas. Such meters are currently planned to be used also subsea, where the calibration possibilities are few. In such subsea applications the velocity of sound measured by these meters will be a powerful input for estimation of density and calorific value of the flowing oil or gas. The ultrasonic transit time measurements in such meters are carried out in a flowing oil or gas, over a range typically between 6 and 40 inches. For precise transit time measurements over such ranges, diffraction corrections may be of high importance. Diffraction effects for an acoustic beam in a flowing medium are therefore studied numerically. The flow direction and the propagation direction of the acoustic beam will be different. The investigation is based on the solution for the acoustic field from a point source in a homogeneous flowing medium. Acoustic beams are modeled using two-dimensional arrays of point sources. The results will be compared to the no-flow case in order to identify effects of flow on the diffraction correction for application of precise transit time measurements in a flowing medium.

3pPAc17. A novel acoustic cell processing platform for cell concentration and washing. Jason P. Dionne (Flodesign Sonics, 499 Bushy Hill Rd., Simsbury, CT 06070, j.dionne@fdsonics.com), Brian Dutra, Kedar C. Chitale, Goutam Ghoshal, Chris Leidel (Flodesign Sonics, Wilbraham, MA), and Bart Lipkens (Flodesign Sonics, Springfield, MA)

FloDesign Sonics has developed a technology to enable a single use (gamma irradiated) continuous cell concentration and wash application for manufacturing of cell-based therapies. The device has been designed to be able to process several liters of a suspended cell culture, e.g., T-cells, at concentrations of 1 to 10 M cells/ml. The cell suspension flows through the device and the acoustic radiation force field is used to trap and hold the cells in the acoustic field. After concentrating the cells, one or multiple washing steps are accomplished by flowing the washing fluid through the device, using the acoustic field to trap the cells while displacing the original cell culture fluid. The holdup volume of the device is about 30 ml. Results are shown for prototypes with a 1x0.75 inch flow chamber driven by 2 MHz PZT-8 transducers operating at flow rates of 1-2L/h, measured cell recoveries of 90% have been achieved with concentration factors of 20 to 50 for Jurkat T-cell suspensions, depending on cell concentration and initial volume of the cell suspension. Scaling strategies used previously for cell clarification will be used to scale up the current cell concentration device to accommodate larger volumes.

3pPAc18. Effect of ultrasound pressure and bubble-bubble interaction on the nonlinear attenuation and sound speed in a bubbly medium. Amin Jafari Sojahrood (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca), Qian Li (Biomedical Eng., Boston Univ., Boston, MA), Hossein Haghi, Raffi Karshafian (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada), Tyrone M. Porter (Biomedical Eng., Boston Univ., Boston, MA), and Michael C. Kolios (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada)

The presence of bubbles changes the attenuation and sound speed of a medium. These changes in medium properties depend on the nonlinear behavior of bubbles which are not well understood. Previous studies employed linear models for the calculation of the attenuation and sound speed of bubbly mediums. These predictions are not valid in the regime of nonlinear oscillations. In addition, bubble-bubble interactions are often neglected. In this work, we have numerically simulated the attenuation and sound speed of a bubbly medium by solving a recently developed nonlinear model and considering the bubble-bubble interactions. A cluster of 52 interacting bubbles was simulated, with sizes derived from experimental measurements. Broadband attenuation measurements of monodisperse solutions were performed with peak pressures ranging within 10-100 kPa. The bubble solutions had mean diameters of 4-6 micron and peak concentrations of 1000 to 15000 bubbles/ml. At lower concentrations (with minimal microbubble interactions) model predictions are in good agreement with experimental measurements. At higher concentrations, new secondary peaks in plots of attenuation and sound speed as a function of frequency appear. By simulating bubble-bubble interactions, the numerical results could predict the frequency shift of the peaks of attenuation and sound speed, and the generation of secondary peaks.

3pPAc19. Smoothed particle acoustics with variable smoothing length and its application to sound propagation with complex boundary. Futang Wang, Qing Zhi Hou, Zhe Wang (School of Comput. Sci. and Technol., Tianjin Univ., No.135 Yaguan Rd., Haihe Education Park, Tianjin 300350, China, futangwang@tju.edu.cn), Yong Ou Zhang (School of Transportation, Wuhan Univ. of Technol., Wuhan, China), and Jianwu Dang (School of Comput. Sci. and Technol., Tianjin Univ., Tianjin, China)

Lagrangian smoothed particle hydrodynamics (SPH) method has shown its high potential for solving acoustic wave propagations in complex domain with multi-mediums. Typical applications are sound wave propagation in speech production and multi-phase flow. For these problems, SPH with adaptive particle distribution might be more efficient, with analog to mesh-based methods with adaptive grids. If the fluid flow or moving boundary is taken into account, initially evenly distributed particles will become irregular anyway (dense somewhere and sparse somewhere else). For irregular particle distribution, conventional SPH with constant smoothing length suffers from low accuracy, phase error and instability problems. The main aim of this work is to apply variable smoothing length into SPH and apply it to 2D sound wave propagation in a domain with complex boundary. In addition, the effects of several strategies for variable smoothing length on phase error in smoothed particle acoustics are fully investigated by numerical examples and theoretical analysis. Numerical results indicate that the phase error is reduced by the use of variable smoothing length.

Session 3pPP

Psychological and Physiological Acoustics: A Celebration of Nat Durlach and His Contributions to Sensory Communications

Barbara Shinn-Cunningham, Cochair
Boston University, 677 Beacon St., Boston, MA 02215

H. Steven Colburn, Cochair
University of Rhode Island, Boston University, 44 Cummington Mall, Boston, MA 02215

Chair's Introduction—1:15

Invited Papers

1:20

3pPP1. The contributions of Nathaniel (Nat) Durlach to binaural hearing research. H. Steven Colburn (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, colburn@bu.edu)

The important and extensive work of Nat Durlach in the area of binaural hearing will be reviewed. Nat started working on this topic in the context of work on bat sonar processing at Lincoln Laboratories, with an early publication on his hearing modeling in 1960. Nat's thinking about the sonar problem and signal processing in noise led to his long-term interest and work in human hearing, and he joined the Sensory Communications group at MIT in 1963. In addition to his well-known work on the Equalization-Cancellation (EC) model, his important contributions to other binaural hearing models and experiments will be discussed, including work in detection, discrimination, and estimation. Nat's binaural work also allows a consideration of his deep thinking about problems, his approach to modeling in general, and his distinctive style of interacting with other scientists, both young and old. The ongoing impact of Nat's binaural hearing work, his personal example for approaching research, and his deep influence on the personal lives of his students and colleagues continues strongly into the future.

1:40

3pPP2. Nat Durlach and the context-coding and trace modes of intensity perception. Louis D. Braida (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA 02139, ldbraida@mit.edu)

Nat came to M.I.T. from Lincoln Laboratory and became involved in the teaching of the subject 6.37, Sensory Communication. Nat soon realized that there were certain problems with teaching the subject. Not only were there many different ways to quantify the magnitude of stimuli (detection, discrimination, identification, category and ratio scaling, to name a few), but also the time variable affected comparisons in ways that were not independent of the range of intensities. This bothered Nat as a mathematician. However within two years of effort, Nat assembled a unifying picture that permitted some of the roadblocks to be overcome. All one-interval experiments were assumed to use the context-coding mode exclusively and were described in terms of two experimental parameters. Two-interval experiments were more complicated, using the trace mode that interacted with context-coding mode, and requiring an additional parameter. Nat assumed that an optimal combination of modes was used. Subsequent work extended these models to the cases where Weber's Law did not hold and to the comparison of loudness of different types of stimuli. As the result of Nat's efforts, 14 papers appeared in the Journal.

2:00

3pPP3. All I really need to know I learned from Lou and Nat: Nat Durlach. Michael Picheny (Watson Multimodal, IBM TJ Watson Res. Ctr., POB 218, Yorktown Heights, NY 10598, picheny@us.ibm.com)

Nat Durlach was one of my two primary mentors in graduate school at MIT. In the context of my PhD thesis, Nat taught me the invaluable lesson of how to think. In this talk I will describe how I applied to Speech Recognition what I learned from him about the value of reviewing prior literature, how to write a strong research proposal, and the need to question basic assumptions. These are eternal gifts Nat bequeathed to me, and I will highlight them in a series of applications using examples over my career at IBM. Specifically, I will draw from work in making advances in core speech recognition, the creation of the interdisciplinary multi-site NSF MALACH project on providing access to large spoken archives of speech, and building one of the early Speech Recognition systems for Mandarin. In addition, I will also describe applications of our work in Clear speech to a set of speech recognition related problems, including the issue of "Sheep and Goats"—why speech recognition works well on some speakers but not others—and also work done some years ago on using Speech Recognition to improve perception and understandability of speech by non-native speakers of English.

2:20

3pPP4. Contributions of Nat Durlach to the field of haptics: Research on tactual communication of speech and manual sensing of mechanical properties. Charlotte M. Reed (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Rm. 36-751, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, cmreed@mit.edu) and Hong Z. Tan (School of Elec. and Comput. Eng., Purdue University, West Lafayette, IN)

As a sensory scientist, Nat's earliest contributions were through his theoretical and experimental work in the area of binaural hearing. Early in his career, however, he demonstrated an interest in comparative sensory processing as evidenced by a study to determine if the masking-level difference observed in audition would be found for stimulation on the skin (it was). Nat's interest in the sense of touch continued through his research concerned with the use of the tactual sense as a substitute for hearing in the communication of speech and language for individuals with profound hearing impairment. This research encompassed studies with experienced deaf-blind users of natural methods of tactual communication (to establish an "existence proof" for the information-bearing capacity of the tactual sense) as well as research on methods for encoding and displaying acoustic signals for presentation through tactile aids. In addition, Nat also spearheaded efforts concerned with the development of haptic displays for use in virtual environment and teleoperator systems. His research in this area was concerned with advancements in knowledge regarding manual sensing and manipulation through a set of basic psychophysical studies. In this talk, we will summarize some of Nat's important contributions in both of these areas.

2:40

3pPP5. The contributions of Nathaniel I. Durlach to the study of informational masking. Gerald Kidd (Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, gkidd@bu.edu)

Beginning in the year 2000, Nat Durlach collaborated with our research group on the study of auditory masking and, in particular, the study of informational masking. During this collaboration of more than one and a half decades, Nat's energy, enthusiasm, creativity, and insight provided the stimulus for many research projects and ultimately a variety of publications. These many contributions to our understanding of auditory masking are embodied in journal articles, letters to the editor, grant proposals and papers at scientific conferences. This presentation will review the diverse body of empirical work that Nat drew on to formulate his views on informational masking and the implications of his work on this topic for contemporary auditory theory.

3:00–3:20 Panel Discussion

TUESDAY AFTERNOON, 27 JUNE 2017 ROOM 201, 1:20 P.M. TO 3:40 P.M.

Session 3pSAa

Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics: Acoustic Metamaterials IV

Christina J. Naify, Chair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Contributed Papers

1:20

3pSAa1. Space-time modulation of electric boundary conditions in a one-dimensional piezoelectric phononic crystal. Charles Croënne (ISEN, IEMN UMR CNRS 8520, Lille, France), Olivier Bou Matar (Ecole Centrale de Lille, IEMN UMR CNRS 8520, Villeneuve d'Ascq, France), Jérôme 0. Vasseur (Université de Lille, IEMN UMR CNRS 8520, Lille, France), Anne-Christine Hladky-Hennion (ISEN, IEMN UMR CNRS 8520, Lille, France), Pierre A. Deymier (Univ. of Arizona, Tucson, AZ), and Bertrand Dubus (ISEN, IEMN UMR CNRS 8520, 41 boulevard Vauban, Lille cedex 59046, France, bertrand.dubus@isen.fr)

A phononic crystal constituted by a one-dimensional piezoelectric material with a periodic distribution of electrodes submitted to space and time-dependent electrical boundary conditions is considered in this work. Interaction of an incident elastic pulse propagating with such phononic crystal is studied using a specific Finite Difference Time Domain model. Simulations are conducted in the case of periodic grounded electrodes "moving" at constant subsonic or supersonic speed. Various nonlinear phenomena resulting from this interaction are observed: Brillouin-like acoustic scattering, non-reciprocity of transmission at fundamental frequency, parametric amplification of the incident wave. The dispersion curve of the wave propagating in the phononic crystal with space-time modulated electrical boundary conditions is also deduced from simulation results.

1:40

3pSAa2. Exploring phononic crystal tunability using dielectric elastomers. Michael A. Jandron (Naval Undersea Warfare Ctr., Naval Undersea Warfare Ctr., Code 8232, Bldg. 1302, Newport, RI 02841, michael.jandron@navy.mil) and David Henann (School of Eng., Brown Univ., Providence, RI)

Tunable phononic crystals give rise to interesting opportunities such as variable-frequency vibration filters. By using soft dielectric elastomers, which undergo large deformations when acted upon by an external electric field, the frequency ranges of these band gaps may be adjusted, or new band gaps may be created through electrical stimuli. In this talk, we will discuss our finite-element-based numerical simulation capability for designing electrically-tunable, soft phononic crystals. The key ingredients of our finite-

element tools are (i) the incorporation of electro-mechanical coupling, (ii) large-deformation capability, and (iii) an accounting for inertial effects. We present a demonstration of our simulation capability to the design of phononic crystals consisting of both square and hexagonal arrays of circular-cross-section threads embedded in a dielectric elastomeric matrix. Finally, we will consider electro-mechanical instabilities as alternative route for enhanced tunability. [This work was currently funded through the Naval Undersea Warfare Center Research program.]

2:00

3pSAa3. Long-range elastic metamaterials. Antonio Carcaterra (Mech. and Aerosp. Eng., La Sapienza, Univ. of Rome, Tarquinia, VT, Italy), Francesco Coppo, Federica Mezzani, and Sara Pensalfini (Mech. and Aerosp. Eng., La Sapienza, Univ. of Rome, Via Eudossiana 18, Rome 00184, Italy, sara.pensalfini@uniroma1.it)

The problem of wave propagation control in one-dimensional systems, including electrical charges and dipole magnetic moments is investigated. The waveguide is characterized by long-range and nonlinear interaction forces of Coulomb and Lorentz nature. Wave propagation properties are derived by a method based on an equivalent partial differential equation that replaces the discrete equation of motion of the chain. The paper shows how the waves propagating in these special systems have characteristics, such as phase and group velocity, that are function of the electrical and magnetic property distribution along the chain. Of interest are also possible wave-stopping phenomena. The paper presents an outline of some basic principles developed by some of the authors in recent theoretical papers and shows also numerical experiments illustrating wave propagation in metamaterials characterized by long-range elastic-electromagnetic interactions.

2:20

3pSAa4. Bi-anisotropy in acoustic scattering problems. Li Quan and Andrea Alu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu)

Electric and magnetic dipole moments dominate the scattering properties of small nanoparticles in optics. Ordinarily, electric dipole moments are excited by the local electric field, and magnetic dipole moments by the magnetic field. Bi-anisotropy, or magneto-electric coupling, largely enriches the electromagnetic response of materials. Similarly, in acoustics the dominant scattering contributions from objects smaller than the wavelength, monopole and dipole moments, are commonly excited by pressure and velocity, respectively. The recent interest in Willis acoustic materials, for which these responses are coupled, leads us to analyze the opportunities offered by the analogues of bi-anisotropic nanoparticles in acoustics and their relevance in practical applications to tailor sound. We also present an extraction procedure to determine the acoustic polarizability tensor of an arbitrary object and relate its bi-anisotropic response to its geometry and material properties.

2:40

3pSAa5. Pressure enhancement in water-based passive acoustic metamaterials. Bogdan Ioan Popa (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu)

It has recently been shown that anisotropic passive acoustic metamaterials can enhance significantly the pressure of sound waves (Chen *et al.*, Nature Commun. 2014). The effect was attributed to strong wave compression inside carefully designed anisotropic metamaterials and was shown to lead to directional acoustic sensors whose sensing threshold is greatly improved. The improved sensing strategy was demonstrated in air but cannot be trivially ported to a water environment where it could find many applications in sonar systems, ultrasound imagers, or underwater communication systems. In this presentation, we will demonstrate new strategies different from Chen *et al.* that lead to pressure amplification in passive metamaterials, and, more importantly, we show that these strategies are suitable to water-based applications. We will further show that the wave compression phenomenon proposed by Chen *et al.* is not a necessary requirement, and both isotropic and anisotropic metamaterial structures can be used to obtain strong pressure enhancements. We will present specific metamaterial structures for underwater operation designed using the new approach, and their significant pressure amplification ability and directivity will be quantified.

3:00

3pSAa6. On the unique features of mechanical metamaterials with nonlinear local oscillators. Priscilla B. Silva, Varvara G. Kouznetsova, and Marc G. Geers (Dept. of Mech. Eng., Eindhoven Univ. of Technol., Mech. of Mater. Group, P.O. Box 513, Eindhoven 5600 MB, Netherlands, p.brandao.silva@tue.nl)

Over the last two decades, metamaterials have attracted a high number of researches motivated by the possibility of designing structures capable of manipulating wave propagation. The basic mechanism underlying the behavior of mechanical metamaterials is their negative effective dynamic parameters (mass and/or stiffness). Within the framework of mechanical metamaterials, most of the developments up to now have considered linear material behavior only. There is a natural need to understand the effect of nonlinear material behavior on the wave propagation through such engineered composites. In this paper, the dynamic behavior of a discrete lattice system composed of a series of nonlinear local resonators is investigated. By making use of the harmonic balance method, approximating dispersion relations are derived. Unlike previous works, super/sub-harmonic generation has been considered and revealed new phenomenon: the possibility of generating multiple transmission dips. The analysis also showed the tunability and multistability features of the system. The semi-analytical predictions were verified with direct numerical simulations.

3:20

3pSAa7. Negative refraction and superresolution by a steel-methanol phononic crystal. Ukesh Koju and Joel Mobley (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., P.O. Box 1848, University, MS 38677, ukoju@go.olemiss.edu)

Negative refraction and the associated lensing effect of a two-dimensional (2D) phononic crystal (PC) in the MHz regime were studied both experimentally and numerically. The PC consists of a hexagonal array of steel cylinders (r = 0.4 mm, a = 0.5 mm) in a methanol matrix for use in an aqueous medium. FEM simulations of the pressure field show negative refraction of plane waves through a prism shaped crystal and superresolution lensing through a rectangular crystal. These phenomena were observed with hydrophone scans of the transmitted pressure fields through the steel-methanol PC in a water tank.

Session 3pSAb

Structural Acoustics and Vibration: Energy Methods in Acoustics and Vibration II

Donald B. Bliss, Cochair
Mechanical Engineering, Duke University, 148B Hudson Hall, Durham, NC 27705

Linda P. Franzoni, Cochair
Dept. of Mech. Eng. and Materials Sci., Duke Univ., Box 90271, Durham, NC 27708-0271

Otto von Estorff, Cochair
Institute of Modelling and Computation, Hamburg University of Technology, Hamburg 21073, Germany

Contributed Papers

1:20

3pSAb1. Blocking oblique wave sound transmission through a multi-element barrier using alternate resonance tuning. Meredith C. Fleming (Phys., Duke Univ., Durham, NC), Donald B. Bliss, and Mauricio Villa (Mech. Eng., Duke Univ., Mech. Eng., 148B Hudson Hall, Durham, NC 27708, donald.bliss@duke.edu)

Alternate Resonance Tuning (ART) utilizes a structural barrier subdivided into dynamic panel subsystems with different resonant behaviors. The underlying idea involves adjacent panel subsystems tuned differently to take advantage of the outofphase vibratory behavior, causing the 180° phase shift at different frequencies. In the intermediate range between the two resonance frequencies, the panels vibrate out of phase, leading to strong cancellation of the transmitted sound field. The method has been demonstrated analytically and experimentally for waves striking a barrier at normal incidence. The current work considers the effectiveness of ART to block the transmission of incident oblique waves, and considers both discrete frequencies and angles, and more realistic broadband random incidence fields. The research goal is to show that flexibility and controlled resonant behavior in subsystems can substantially block sound transmission, even for low structural damping. The subsystems alter the vibrating surface wavenumber spectra to reduce coupling between the structure and the acoustic field. Not only is the transmission of incident oblique waves reduced, but the transmitted and reflected waves radiate at a variety of angles due to the modification of the surface wavenumber spectrum. Applications include the development of lightweight flexible sound blocking barriers for vehicles and architectural spaces.

1:40

3pSAb2. Reflection and transmission of acoustic energy by an elastic plate with structural discontinuities forced by multiple angle oblique broadband sound waves. Mauricio Villa, Donald B. Bliss, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90300 Hudson Hall, Durham, NC 27708, mauricio.villa@duke.edu)

An analysis is presented for acoustic reflection and transmission from an infinite fluid-loaded plate with spatially periodic discontinuities. The plate, with similar or dissimilar acoustic fluids on both sides, is excited by an oblique wave incident acoustic field. The fully-coupled structural/acoustic problem is treated by the method of Analytical-Numerical Matching (ANM). The ANM framework separates the problem into global numerical and local analytical solutions, and handles rapid spatial variation around the structural discontinuities in closed form, improving the numerical accuracy and convergence rate. The ANM approach includes a novel way to handle difficulties associated with coincidence frequencies. The periodic spatial discontinuities, modeled by various boundary conditions, create deviations from specular directivity with multiple reflection and transmission angles, the effect being most pronounced at structural resonances. The periodic discontinuities redirect part of the structural energy into reverberant resonant substructures having wavenumbers different from the oblique wave forcing, reradiating with different directivity angles. Discrete frequency and broadband diffuse results are presented. These results are also compared to a baffled finite barrier with a fluid loading correction introduced to the structural wavenumber. The goal is to develop efficient methods for structural-acoustic reflection and transmission of broadband acoustic energy between coupled acoustic domains.

2:00

3pSAb3. The energy method for solving electroacoustic problems using combining finite element analysis and analytical methods. David A. Brown (ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net), Xiang Yan, and Boris Aronov (BTech Acoust. LLC, Fall River, MA)

The energy method for solving electroacoustic transducer problems requires calculation of the electrical, mechanical and electromechanical energies of a piezoelectric body from the vibration mode shapes, which are often difficult to obtain analytical particularly with structures having complex boundary conditions. An alternative approach is to determine the vibration mode shapes either experimentally or by finite element analysis and to proceed with computing the energies analytically. The frequency response can then be determined from an equivalent electromechanical circuit approach. An example of this hybrid approach applied to a flextensional transducer will be presented with comparison to experimental results. [Work supported by ONR Code 321.]

2:20

3pSAb4. Nonlinear unsteady energy analysis of structural systems. Antonio Culla, Gianluca Pepe (Dept. of Mech. and Aerosp. Eng., Univ. of Rome La Sapienza, via Eudossiana 18, Rome 00184, Italy, antonio.culla@uniroma1.it), and Antonio Carcaterra (Dept. of Mech. and Aerosp. Eng., Univ. of Rome La Sapienza, Tarquinia, VT, Italy)

The problem of vibration of large systems undergoing shocks and unsteady loads is one of the field of great interest in vibro-acoustic engineering. Statistical Energy Analysis-SEA is one of the most acknowledged methods in this field. However, SEA has many limitations, and is based on several questionable hypotheses. In the present paper, on the basis of a new theory of vibration thermodynamics, the authors consider a set of systems characterized by (i) unsteady loads, such as shocks, (ii) nonlinear coupling between the different subcomponents. The analysis is carried on considering

3p TUE. PM

different prototype systems, starting from a very simple pair of nonlinear resonators, a 2-dof system, up to consider a system of plates coupled through nonlinear joints. It is shown how the energy flow relationship between subsystems pair, comes out to be a power series of the energy storage difference. These results are systematically considered in the light of the thermodynamic theory of vibrating systems, showing how a general energy approach to complex systems is feasible.

2:40

3pSAb5. Diffuse elastic waves in a nearly axisymmetric body: Distribution, transport, and dynamical localization. Richard Weaver and John Yoritomo (Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL, r-weaver@uiuc.edu)

We report measurements on the distribution and evolution of diffuse ultrasonic waves in elastic bodies with weakly broken axisymmetry. Aluminum cylinders with dimensions large compared to wavelength were excited by transient point sources at the center of the upper circular face. The resulting power spectral density was then examined as a function of time and frequency and position. It was found that this energy density showed a marked concentration at the center at early times, a concentration that subsequently slowly diminished towards a state of uniformity across the face, over times long compared to ultrasonic transit time across the sample. The evolution is attributed to scattering by symmetry breaking heterogeneities. Relaxation did not proceed all the way to uniformity and equipartition, behavior shown to be consistent with Enhanced Backscatter and with Dynamical Anderson Localization among subspaces of different angular momentum.

3:00

3pSAb6. Anderson localization amongst weakly coupled substructures. John P. Coleman, John Y. Yoritomo, and Richard Weaver (Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL 61801, jpcolem2@illinois.edu)

It is shown that if vibrational energy is put into one substructure and allowed to diffuse into other weakly coupled substructures (as in the case of statistical energy analysis or reverberant sound diffusing from one room into others through small windows), at infinite time the energy density in the starting room will still be above its equipartition value. We offer a theory to predict the amount of this Anderson localization in such systems of weakly coupled substructures. We compare these predictions with numerical results obtained using random matrix substructures.

3:20

3pSAb7. A higher order shear deformation model of a periodically sectioned plate. Andrew J. Hull (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk develops a higher order shear deformation model of a periodically sectioned plate. A parabolic deformation expression is used with periodic analysis methods to calculate the displacement field as a function of plate spatial location. The problem is formulated by writing the transverse displacement field and the in-plane rotations as a series solution of unknown wave propagation coefficients multiplied by an exponential indexed wavenumber term in the direction of varying structural properties multiplied by an exponential constant term in the direction of constant structural properties. These expansions, along with various structural properties written using Fourier summations, are inserted into the governing differential equations that were derived using Hamilton's principle. The equations are now algebraic expressions that can be orthogonalized and written in a global matrix format whose solution is the wave propagation coefficients, thus yielding the transverse and in-plane displacements of the system. This new model is validated with finite element theory and Kirchhoff plate theory for a thin plate simulation and verified with comparison to experimental results for a 0.0191 m thick sectional plate.

TUESDAY AFTERNOON, 27 JUNE 2017 — BALLROOM A, 1:20 P.M. TO 3:40 P.M.

Session 3pSC

Speech Communication: Aging and Development (Poster Session)

Christina Kuo, Chair
Communication Sciences and Disorders, James Madison University, MSC4304, 801 Carrier Drive, Harrisonburg, VA 22807

All posters will be on display from 1:20 p.m. to 3:40 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:20 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 3:40 p.m.

Contributed Papers

3pSC1. Linguistic masking release in older adults. Sarah Alageel (Dept. of Otolaryngol. and Commun. Sci., King Faisal Specialist Hospital and Res. Ctr., Zahrawi St., Al Maather, Riyadh 12713, Saudi Arabia, Salageel97@kfsrc.edu.sa), Stanley Sheft, and Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., Chicago, IL)

Past studies of speech-on-speech masking in children and young adults (YA) indicate that intelligibility of target speech can improve when target and masker speech are in different languages. We investigated whether such linguistic release from masking is obtained in older adults (OA) with age-typical hearing abilities. All participants were asked to recognize English sentences in the presence of two-talker maskers spoken in either English or Spanish presented at four different Signal-to-Noise ratios (SNR) to each group. Differences in energetic masking between the masker languages were minimized. Overall sentence recognition accuracy was greater for YA participants. However, both groups equally benefited from a linguistic mismatch in the target-masker language with a significant masking release for the Spanish language masker. The magnitude of masking release increased

as SNR decreased, ranging across SNR conditions from 1 to 27 percentage points. Age, hearing-in-noise ability and hearing sensitivity of OA listeners did not correlate with masking release. Results confirm previous findings of masking release associated with a linguistic mismatch between target and masker speech, and indicate that in speech-on-speech masking older listeners can improve speech intelligibility by utilizing nonenergetic linguistic differences between the target and masker speech.

3pSC2. The impact of context and competition on speech comprehension in younger and older adults revealed using eye-tracking and pupillometry. Nicole Ayasse and Arthur Wingfield (Volen National Ctr. for Complex Systems, Brandeis Univ., 415 South St., MS 062 Brown Psych. Office, Waltham, MA 02453, nayasse@brandeis.edu)

Although younger and older adults can use context effectively to understand spoken language, at times the context may fit multiple semantic competitors, and choosing the correct one can be crucial for comprehension. Given the ambiguities present in the real world, and the inhibitory control deficit common in aging, it is critical to understand how adults of all ages comprehend sentences. An experiment is reported to explore the interplay of context and competition in sentence comprehension using a variation on a visual world eye-tracking paradigm. Spoken sentences were presented with either high or low expectancy (context) for a sentence-final (target) word and with either high or low response entropy (uncertainty or competition); these target words were then paired with either a contextual competitor or an unrelated lure. Results support the expectation that lower context and greater competition slow comprehension and increase cognitive effort. Results will be discussed in terms of aging and individual differences. [Work supported by NIH Grants RO1 AG 019714 and T32 GM 084907.]

3pSC3. Rhythmic characteristics in aging: A study on Zurich German. Elisa Pellegrino (Univ. of Zurich, Plattenstrasse 54, Zuerich 8032, Switzerland, pellegrino.elisa.1981@gmail.com), Lei He, Natalie Giroud, Martin Meyer, and Volker Dellwo (Univ. of Zurich, Zurich, Switzerland)

Age-related changes in speech production influence both speech segmental and suprasegmental characteristics. Previous research focussed on changes in voice quality, vowel formant patterns, f0 and speech rate due to aging but only little attention has been paid on speech rhythm (durational and dynamic). In this study we analyzed the segmental durational variability as well as the syllabic intensity variability between 60 Zurich German speakers ranging in age between 20 and 81 years. Speakers read 90 sentences in Zurich German. Between-speaker durational variability across age was quantified through a variety of different rhythmic variables (%V, ΔCLn, ΔVLn, rPVI-C, nPVI-V, %VO, varcoVO, nPVI-VO, ΔPeak, varcoPeak, and nPVI-Peak). Intensity variability was computed by taking the standard deviations, variation coefficients and PVIs of average syllable intensity and syllable peak intensity values across sentences. Results based on durational measurements show that with aging there is an increase in %V and r-PVI-C. Aged voices also present lower variability in the consonantal and especially in the vocalic intervals (nPVI-V and ΔVLn). We argue that changes in the physical characteristics as well as in the neural control mechanisms of the articulators might play a significant role in age-related rhythmic changes.

3pSC4. Speed of lexical access relates to quality of neural response to sound, not cognitive abilities, in younger and older adults. Alexis R. Johns (Memory and Cognition Lab (Volen Complex), Brandeis Univ. MS 013, Waltham, MA 02453, ajohns@brandeis.edu), Emily B. Myers, Erika Skoe (Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs, CT), and James S. Magnuson (Psychol. Sci., Univ. of Connecticut, Storrs, CT)

Previous work has demonstrated a relationship between age-related declines in inhibitory control and difficulties identifying words of low lexical frequency and high neighborhood density (Sommers & Danielson, 1999). We hypothesized that declines in consistency of the auditory brainstem response (ABR; neural response to a repeated sound; Anderson *et al.*, 2012) might also impede lexical access in older adults. We measured audiometric thresholds, ABR consistency, vocabulary, inhibitory control, and working memory in two groups (younger: 18-23yo, n = 41; older: 54-76yo, n = 41). We used mean target fixation proportion from 200 to 750 ms after word onset in a visual world task as a proxy for lexical access speed as listeners identified spoken words (varying on low/high: lexical frequency, neighborhood density, and cohort density; Magnuson *et al.*, 2007). ABR consistency significantly predicted speed of word identification across variations in neighborhood and cohort densities, but contra previous findings, cognitive measures did not improve model fits. Interactions involving age, vocabulary, and lexical frequency suggest age-related linguistic expertise influences lexical access of uncommon words. We conclude that older adults exhibit increases in phonological competition due to declines in auditory encoding, suggesting that a consistent neural response to sounds leads to more efficient speech processing and lexical access.

3pSC5. Velar-vowel coarticulation across the lifespan and in people who stutter: Findings and model. Stefan A. Frisch and Nathan D. Maxfield (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD1017, Tampa, FL 33620, sfrisch@usf.edu)

The study of anticipatory coarticulation provides insight into the speech production planning process. In the present study, the task involved repeating velar-vowel consonant combinations in a carrier sentence (e.g., for /ke/, "Say a cape again"). Data for velar-vowel coarticulation were analyzed using the Articulate Assistant Advanced software to create tongue traces that were quantified following the procedures for average of the nearest neighbor point-to-point distance between curves (Zharkova & Hewlett 2009, Journal of Phonetics). There were 126 participants total in child (8-12), young adult (18-39), and older adult (55-75) ages and groups of typical speakers (n = 21, 23, 29) and people who stutter (n = 15, 23, 11). Data analysis found a decrease in coarticulatory influence of the vowel on the velar across the lifespan, but no differences in coarticulation for people who stutter. Analysis of variability found greater variability for children and people who stutter. A two allophone model of coarticulation provided the best fit to the data, replicating Frisch & Wodzinski (2016, Journal of Phonetics).

3pSC6. Influences on lip-reading ability: Aging, sensory, and cognitive functions. Katherine M. Dawson and D. H. Whalen (Speech-Language-Hearing, City Univ. New York Graduate Ctr., 365 5th Ave., New York, NY 10016, kdawson2@gradcenter.cuny.edu)

To explore the previously reported effect of aging on lip-reading ability, older (60-75 years) and younger (18-35) adults gave oral responses to sentences from a modified version of the build-a-sentence (BAS) test [Tye-Murray *et al.*, Int. J. Acoust., **47**(S2), S31-S37, (2008)]. These sentences have predictable syntactic form but contain some words selected from a randomized list of nouns, e.g., 'The duck watched the cop'. Participants identified these nouns from videos of a single female talker, and responses were transcribed by the experimenter. The 30 participants (15 native English speakers in each age group) were also tested for general cognitive function, vision, hearing, working memory (WM), attention, inhibition, and speech motor ability. They were also asked to gauge their own lip-reading abilities. Preliminary results indicate that the cognitive variables have effects independent of age. Sentence length and syntactic complexity contributed differently in the two age groups, perhaps due to WM capacity.

3pSC7. Three-dimensional analysis of liquid sounds produced by first graders. Olivia Foley, Amy W. Piper, and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

The liquids /r/ and /l/ in American English are complex sounds whose productions are highly variable among adult speakers, but there is currently little knowledge about how children articulate these sounds, which are typically acquired late in development. In this study, the Goldman-Fristoe Test of Articulation (GFTA-3) was administered to typically developing first graders (6 and 7 years old) while a 4D ultrasound system imaged the tongue with synchronous audio and webcam video recordings. The GFTA-3 contains words in which /r/ and /l/ occur in a variety of syllable positions and phonetic contexts. Among 14 first grade participants in this study, the so-called "bunched /r/" is overwhelmingly preferred over the so-called "retroflexed /r/." In contrast, three-dimensional tongue shapes in the production of syllable initial /l/ are substantially more variable with two basic configurations ("coronal" and "dorsal"), while syllable final /l/ productions are more consistently "dorsal." Examples of 3D tongue shapes will be presented, along with results from a Principal Components Analysis.

3pSC8. Naturalistic coding for prelinguistic speech at 9 and 12 months of age from two Mandarin-learning children differing in auditory function. Hsin-yu Li and Li-mei Chen (National Cheng Kung Univ., 1 University Rd., Tainan, Taiwan, claion11@gmail.com)

This study investigated the feasibility of Fagan's (2005) naturalistic coding system for prelinguistic speech of Mandarin-learning children. This system includes 10 categories: single vowel, single consonant, consonant combination, vowel combination, syllable containing glottal sound and vowel, syllable containing supra-glottal consonant and vowel (SGCV), reduplicated vowel, reduplicated consonant, reduplicated CV, and reduplicated "SGCV" (RSGCV). The first 50 clear utterances produced by one normal-hearing (NH) child and one hearing-impaired (HI) child in the audio recordings at 9 and 12 months old were transcribed into 10 categories for comparison. Major findings are (1) compared with NH child, HI child demonstrated no obvious changes in vocalization from 9 to 12 months old; (2) at 12 months old, canonical babbling ratio of HI child was 0, while those of NH child were 0.84 (utterance as unit) and 0.48 (syllable as unit); (3) compared with NH child, HI child did not manipulate supra-glottal sounds and had limited consonant inventory; (4) HI child produced no RSGCV (e.g., /baba/) while NH child was at reduplicated babbling stage at 12 months old. Naturalistic coding system revealed difference in prelinguistic speech between a NH child and a HI child. More participants should be included to verify the findings.

3pSC9. Irregular pitch periods as a feature cue in the developing speech of English-learning children. Helen Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu), Stefanie Shattuck-Hufnagel (RLE, MIT, Cambridge, MA), and John Pereira (ECE Dept., Union College, Schenectady, NY)

Changes in phonation patterns have long been studied as correlates of various linguistic elements, such as the occurrence of irregular pitch periods (IPPs) at significant locations in prosodic structure (in phrase-initial, phrase-final, and pitch accented contexts) and word-final voiceless stops, especially /t/. But less is known about the development of this phonation pattern in children [cf. Song *et al.*, JASA, **131**, 3036-50, 2012], particularly in toddlers between the ages of 2;6 and 3;6. The study of its course of acquisition may shed light on the mechanisms involved, since child vocal folds are very different physiologically from those of adults, and change strikingly during development. Monosyllabic target words from the Imbrie Corpus of speech from 10 toddlers 2 1/2 to 3 1/2 years old, ending in /t, d/ were examined for evidence of IPPs. Preliminary results based on three adult/child pairs suggest that both adults and children produce IPPs preceding coda /t/ about 50% of the time. But children produce fewer IPPs before coda /d/ than adults do (38% v 8%), suggesting (like earlier reports) that children are not simply imitating the cues produced by the adults around them. Data from additional adult/child pairs will be presented.

3pSC10. Development of acoustic speech discrimination abilities in school-aged children. Pamela Trudeau-Fisette (Phonet. Lab., Dept. of Linguist, Université du Québec à Montréal, Montréal, QC, Canada), Melinda Maysounave, Camille Vidou, and Lucie Menard (Phonet. Lab., Dept. of Linguist, Université du Québec à Montréal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

The acquisition of speech perception skills is challenging for children. Although some studies have shown that categorical perception boundaries become steeper during childhood and are sometimes shifted in children compared to adults, very few experiments on discrimination abilities in children have been conducted. To investigate this, we conducted a perceptual discrimination task in school-aged children. Sixty-seven 6- to 12-year-old native Quebec French-speaking children were asked to complete a speech perception task that used an AXB scheme. The children were asked to discriminate synthesized 5-formant vowels belonging to three continuums: /i-e/ and /e-ø/ (in which stimuli were equally stepped along F1) and /e-ø/ (in which stimuli were equally stepped along F2). There was a significant effect of age on peak discrimination scores, with older children having higher peak scores than younger children. There was no effect of age on the location of the categorical boundary on the three tested continuums. These findings support the hypothesis that speech development and learning experience play significant roles in the establishment of strong phonological targets.

3pSC11. A bear called Baddington? Variability and contrast enhancement in accented infant-directed speech. Jessamyn L. Schertz (Dept. of Lang. Studies, Univ. of Toronto Mississauga, Dept. of Lang. Studies, Ste. 301, Erindale Hall, 3359 Mississauga Rd., Mississauga, ON L5L1C6, Canada, jessamyn.schertz@utoronto.ca), Helen Buckler, Chris Klammer, and Elizabeth Johnson (Dept. of Psych., Univ. of Toronto Mississauga, Mississauga, ON, Canada)

This work examines the realization of the English stop voicing contrast in read speech directed to infants (IDS) and adults (ADS), as well as in words in isolation, as produced by three groups of speakers: native speakers of Canadian English (where /b/ and /p/ differ in aspiration), native speakers of languages in which /b/ and /p/ differ in phonetic voicing (e.g., Spanish), and native speakers of languages which have a 4-way stop contrast /b, p, b^h, p^h/, where both aspiration and voicing are contrastive (e.g., Hindi). In words in isolation, speakers from both "accented" groups tended to produce English voiceless stops as unaspirated, and voiced stops as phonetically voiced. However, there was variability in accented speakers' voiceless stops, as well as in native speakers' use of phonetic voicing in voiced stops, and this variability appeared to be augmented in the read speech conditions (ADS and IDS). We test the hypotheses (1) that IDS results in phonologically-informed contrast enhancement, with accent-specific modifications expected for the three groups of speakers, and (2) that speakers aim for more precision in phonetic targets when talking to infants, resulting in less within-speaker variability in realization of the contrast in IDS as compared to ADS.

3pSC12. Do mothers enhance the tonal contrasts in their monosyllabic Cantonese tones directed to their infants? Puisan Wong and Hoi Yee Ng (Speech and Hearing Sci., The Univ. of Hong Kong, Rm. 757 Meng Wah Complex, Pokfulam NA, Hong Kong, pswResearch@gmail.com)

Introduction Some studies reported that adults enhanced acoustic differences of phonemes when speaking to young children. We examined whether mothers improved the contrasts among the lexical tones when speaking to infants. Methods Nineteen native Cantonese-speaking mothers produced the six Cantonese tones in 45 monosyllabic words to an adult and their 7- to 12-month-old infants. The 1709 words were low-pass filtered to preserve the pitch contours but eliminate lexical information. Five judges categorized the tones of the filtered words. Acoustic analysis was performed. Results Infant-directed tones had higher fundamental frequency (F0) and longer duration, but were identified with lower, though not significantly different, accuracy. Larger mean F0 differences were found in four pair of tones in infant-directed speech. However, these increased acoustic contrasts did not lead to higher perceptual accuracy of these tone pairs. Despite substantial perceptual confusion between T2 (HR) and T5 (LR) has been reported in previous tone studies, the difference in the slopes of the two tones was not enhanced in infant-directed speech. Conclusion Mothers acoustically modified their tones when speaking to infants. However, little evidence supported that mothers enhanced the phonetic contrasts of the tones in infant-directed speech. [Work supported by Research Grants Council of Hong Kong.]

3pSC13. Differentiating infant cry from non-cry vocalizations based on perception of negativity and acoustic features. Hyunjoo Yoo, Gavin Bidelman, Eugene Buder, Miriam van Mersbergen, and David K. Oller (School of Commun. Sci. and Disord., The Univ. of Memphis, 4055 N Park Loop, Memphis, TN 38117, hyoo2@memphis.edu)

This study seeks to determine how human listeners discriminate cry vs. non-cry sounds by investigating acoustic factors that may contribute to the perception of negativity in infant vocalizations (e.g., cry, whine, and vowel-

like sounds). The assumption is that identification of cry is self-evident; therefore, there has been no attempt to systematically differentiate cry from non-cry vocalizations. Twelve exemplars each of cry, whine, and vowel-like sound segments (36 total) were selected from archival audio recordings of infant vocalizations. Categories were selected from expert-judged audio signals of vocal development. Adult listeners identified each utterance as either cry, whine, or vowel-like sound as quickly and accurately as possible. They also judged the extent of negativity of each utterance. Acoustic features of each utterance were analyzed in association with the categories and degrees of negativity. Results suggest a continuum of negativity from cries (most negative) to vowel-like sounds (least negative), and that acoustic variables are gradated across the negativity continuum. However, preliminary results suggest that peak F0, peak RMS, and spectral slope best differentiate the categories.

3pSC14. Amplitude-modulation detection and speech recognition in normal and hearing-impaired listeners. Sarah Verhulst (Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium, s.verhulst@ugent.be) and Anna Warzybok (Oldenburg Univ., Oldenburg, Germany)

Even though temporal speech envelopes may form a salient cue when listening to speech in noisy backgrounds, the relationship between speech intelligibility and the sensitivity to detecting temporal envelopes (i.e., amplitude modulation detection) is not well understood. This study measured speech reception thresholds in quiet, stationary and speech-modulated noise in three listener groups: young (yNH) and older normal-hearing (oNH) listeners and hearing-impaired (HI) listeners. In addition to broadband speech and noise signals, we adopted low and high-pass filtered versions of the stimuli to study the contribution of different coding mechanisms in basal and apical cochlear regions. For the same listeners, amplitude-modulation (AM) detection thresholds were measured in quiet and in the presence of broadband masking noise for 70 dB AM tones of 0.5, 2 and 4 kHz with either 100 or 5 Hz modulation frequency. Even though group trends were clearer than individual differences, AM detection thresholds showed a relationship to speech recognition in the low-pass filtered speech-modulated noise condition, and for yNH and oNH listeners in the high-pass filtered condition. Overall, this study sheds light on the importance of temporal envelope coding sensitivity for speech recognition and its relationship to near- and supra-threshold hearing deficits.

TUESDAY AFTERNOON, 27 JUNE 2017 ROOM 302, 1:20 P.M. TO 3:20 P.M.

Session 3pSP

Signal Processing in Acoustics, Engineering Acoustics, and Architectural Acoustics: Signal Processing for Directional Sensors IV

Kainam T. Wong, Chair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, DE 605, Hung Hom KLN, Hong Kong

Invited Papers

1:20

3pSP1. Modelling and estimation of the spatial impulse response in reverberant conditions. Ivan J. Tashev, Hannes Gamper, and Lyle Corbin (Microsoft Res. Labs, Microsoft Corp., One Microsoft Way, Redmond, WA 98052, ivantash@microsoft.com)

Modern audio signal processing and speech enhancement relies more and more on machine learning approaches, which require vast amount of data for training. One of the ways to create a dataset for training is by convolving measured impulse response between the sound source and the device with clean speech and adding noise. This approach is limited to the pair of used sound source and microphone, as it incorporates not only the reverberation of the room, but also the radiation pattern of the sound source (typically mouth simulator or head and torso simulator) and the directivity patterns of the microphones in the device under test. In this paper we propose using a spherical loudspeaker array as a transmitter and a spherical microphone array as a receiver to create a sound source and receiver independent impulse response. During the dataset synthesis this spatial impulse response is modified to model the impulse response between transmitter and receiver with given directivity patterns.

1:40

3pSP2. Minimum Energy Method (MEM) microphone array back-propagation for measuring musical wind instruments sound hole radiation. Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de), Jost L. Fischer (Inst. of Systematic Musicology, Univ. of Hamburg, Hambug, Germany), and Markus Abel (Inst. of Phys., Univ. of Potsdam, Potsdam, Germany)

Using a 128 microphone array, the sound source distribution of musical wind instruments from their blowing and finger holes are measured. The Japanese \emph{shakuhachi} flute, the Chinese \emph{dizi} transverse flute, the Balinese \emph{suling} bamboo flute and flue organ pipes are investigated. The sound radiation is measured by a rectangular microphone array in the near field, and back-

propagated onto the flute radiation plane using the Minimum Energy Method (MEM). Here, the radiation is assumed to be composed of as many sound sources as there are microphones, where the source positions can be chosen arbitrarily in space. Using a regularization parameter α the virtual sound sources, which are monopoles for $\alpha = 0$ are narrowed for higher α. Calculating the reconstruction energy on the radiation plane while using different α, this energy becomes minimal for the correct α. The ill-posedness of the solved Fredholm integral at the presence of measurement noise is met by easing α slightly, therefore stabilizing the solution in a robust way. This is especially necessary for problems like musical wind instruments, where the sound sources are the air itself and often a precise sound source position cannot be known in advance. The flutes show complex radiation behavior and sometimes even a coupling of the sound field between finger holes outside the flute. Coupling between organ pipes can also lead to synchronization between the pipes.

Contributed Paper

2:00

3pSP3. Capturing sound of target source located beyond noise source by line microphone array. Akio Ando, Kodai Yamauchi, and Kazuyoshi Onishi (Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 3190 Gofuku, Toyama 930-8555, Japan, andio@eng.u-toyama.ac.jp)

We had developed a method that captured the target sound when the noise source was located in front of the target source. It used a line microphone array whose direction was toward the noise source and created a spatial null sensitivity point at the noise source position. Our previous method, however, could reduce the noise only in a narrow area around the null point. To extend the noise reduction area, we developed a new method that creates some adjacent null points with more than two microphones. The experimental result showed that, if the target source was located 8 m distant from the microphone, the new method with three microphones could create a -20 dB noise reduction area whose center was 6 m distant from the microphone with a radius of 0.5 m, and captured the target sound without changing its timbre.

Invited Paper

2:20

3pSP4. Model-error-compensated reiterative adaptive beamforming on a physically shaded cylindrical array. Jonathan Botts (ARiA, 209 N. Commerce St, Ste 300, Culpeper, VA 22701, jonathan.botts@ariacoustics.com), Jason E. Summers, and Charles F. Gaumond (ARiA, Washington, DC)

The authors formulate and compare beamspace and element-space formulations of Reiterative Superresolution (RISR) [S. D. Blunt *et al*., IEEE Trans. Aero. & Elec. Sys. **47**, 332-346 (2011)] on a physically shaded cylindrical array. The beamspace adaptive approach minimizes computational load while use of a model-based structured covariance estimates enables adaptive beamforming with Doppler sensitive waveforms having little sample support. While structured covariance estimates reduce sample-support requirements and facilitate stable inversion, they also introduce degradation due to model-mismatch errors. In space-time adaptive processing (STAP), covariance matrix tapers (CMT) have been used to counter the effects of off-axis arrivals and other forms of model mismatch [J. R. Guerci, Space-Time Adaptive Processing for Radar (Artech, 2014)]. In adaptive pulse compression, CMT have been used to increase range sidelobe suppression in the presence of Doppler [T. Cuprak, M.S. Thesis (2013)]. In this work, we have formulated CMT to address model mismatch by accounting for interbeam arrivals. We also consider the degree to which CMT can be used to address model-mismatch in RISR resulting from the physical shading effects on the directionality of elements in a large cylindrical array. [Portions of this material are based upon work supported by the Naval Sea Systems Command.]

Contributed Paper

2:40

3pSP5. Comparison of time domain noise source localization techniques: Application to impulsive noise of nail guns. Thomas Padois (Mech., ETS, 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, Thomas.Padois@etsmtl.ca), Marc-André Gaudreau (Mech., Cegep Drummondville, Montréal, QC, Canada), Olivier Doutres (Mech., ETS, Montréal, QC, Canada), Franck C. Sgard (IRSST, Montreal, QC, Canada), Alain Berry (Mech., Université de Sherbrooke, Sherbrooke, QC, Canada), Pierre Marcotte (IRSST, Montréal, QC, Canada), and Frédéric Laville (Mech., ETS, Montréal, QC, Canada)

Microphone array techniques are an efficient tool to detect acoustic source positions. The standard technique is the delay and sum beamforming. In the time domain, the generalized cross correlation of the microphone signals is used to compute the Spatial Likelihood Functions (SLF) of all microphone pairs and the noise source map is provided by the arithmetic mean of these functions. To improve the former noise source map, which means narrowing the main lobe and removing side and spurious lobes, several techniques have been developed in the past. In this work, the performances of three of these techniques (in terms of source position detection, amplitude estimation and computation time) are compared in the case of both synthetic and real data: (1) energetic and geometric criteria are applied in order to remove the SLF with useless information, (2) the arithmetic mean is replaced by the generalized mean and (3) linear inverse problem is solved with sparsity constraint. In the case of real data, the source to be located and quantified is an impulsive noise radiated by nail guns which is recorded by a spiral arm microphone array.

Invited Paper

3:00

3pSP6. Directional information extracted from time reversal scanning to image stress corrosion crack orientation. Brian E. Anderson (Phys. and Astronomy, Brigham Young Univ., MS D446, Provo, UT 84602, bea@byu.edu), Timothy J. Ulrich, Pierre-yves Le Bas (Detonator Technol., Los Alamos National Lab., Los Alamos, NM), Marcel Remillieux (Geophys. Group, Los Alamos National Lab., Los Alamos, NM), and Brent O. Reichman (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The time reversed elastic nonlinearity diagnostic (TREND) is a nondestructive inspection technique based on the principle of time reversal and used to scan the surface of a sample to identify and characterize cracks and other defects. TREND utilizes a series of individual time reversal experiments conducted on a grid of points in a region of interest to map out the spatial extent of surficial expressions of cracks and subsurface features that are less than a wavelength below the surface. The focal signatures from each of these experiments can be used to determine not only the location but also the orientation of the crack. We will discuss how this technique can be applied to a stainless steel sample with stress corrosion cracking (SCC) using stationary piezoelectric transducers broadcasting ultrasonic waves and a scanning laser vibrometer setup used to detect the three dimensional vibration of the sample surface. The orientation of the crack is critical to estimate how soon a crack might penetrate through the wall thickness of a structure. [This work was funded by the U.S. Dept. of Energy, Fuel Cycle R&D, Used Fuel Disposition (Storage) campaign and through a Nuclear Energy University Program Integrated Research Project (IRP-15-9318).]

TUESDAY AFTERNOON, 27 JUNE 2017 ROOM 309, 1:20 P.M. TO 2:20 P.M.

Session 3pUWa

Underwater Acoustics, Acoustical Oceanography, Engineering Acoustics, and Signal Processing in Acoustics: A Century of Sonar II

Michael A. Ainslie, Cochair
Underwater Tech. Dept., TNO, P.O. Box 96864, The Hague 2509JG, Netherlands

Kevin D. Heaney, Cochair
OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039

Invited Papers

1:20

3pUWa1. The impact of naval requirements on sonar development. David L. Bradley (Univ. of New Hampshire, 6934 Traveler's Rest Circle, Easton, MD 21601, dlb25@psu.edu)

A "walk" through history from WWI to today provides an overview of sonar system evolution. The time and spatial constraints of Naval sonar systems are not necessarily consistent with those required for scientific studies of the ocean and its boundaries. The pressure from these competing purposes has resulted in major advances, but also compromises in sonar design and implementation. The circumstances of the Cold War led to complications in the relations between ocean scientists and undersea warfare specialists, in turn leading to competition for limited resources and consequent weak cooperation between the communities. The result was a slowing of the development pace of acoustic systems. More recently, re-convergence has occurred and improved, multi-purpose sonar systems are the norm.

1:40

3pUWa2. A century of sonar performance prediction. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

Developments in sonar technology went hand in hand with increased scientific understanding of the factors affecting sonar performance The first known reference is from Lichte (Physikalische Zeitschrift 1919) where cylindrical spreading for propagation loss was accounted for in deep water (convergence zone, CZ) propagation. Users of sonar in the 1920s encountered an "afternoon effect", a temporary dip in performance that occurred during warm summer evenings, was explained in the 1930s by Iselin and Batchelder in terms of vertical temperature gradients in the sea. A significant amount of work occurred in both World War I(Klein 1962, Wood 1965) and World War II to understand, and therefore be able to predict, the capabilities of sonar systems to detect submerged targets.

Understanding of sonar performance increased rapidly during WW2 and continued to advance during the Cold War. Advances in the understanding of sonar performance modeling before, during and after the Second World War are described.

Contributed Paper

2:00

3pUWa3. Liquid filled encapsulation for thermoacoustic sonar projectors. Nathanael K. Mayo and John B. Blottman (Div. Newport, Naval Undersea Warfare Ctr., Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841-1708, nathanael.mayo@navy.mil)

Thermoacoustic projectors produce sound by rapidly heating and cooling a material with low heat capacity. These "thermophones" were originally demonstrated in 1917 using thin platinum filaments [Arnold, H., I.B Crandall, (1917) Phys. Rev. 10(1):22-38], but were very limited in their efficiencies and bandwidth until the much more recent discovery of new nanomaterials. The first underwater thermophones were made by Aliev *et al.* in 2010 [Aliev, A. E *et al*, (2010) *Nano letters 10* (7), 2374-80] which utilized a carbon nanotube (CNT) sheet submerged in deionized water. These devices worked well for demonstrational purposes, but the CNT sheet would become damaged when retracted from the water, which made characterization difficult. More recently, new methods for enhancing the robustness of freestanding sheets have been developed which allow few layer CNT sheets to be repeatedly dipped and withdrawn from a water bath without damage. Such methods have enabled revisiting the study of submerged thermoacoustic projectors. Our studies of CNT thermophones in various liquid baths give evidence to the mechanism for acoustic wave generation in these systems. The potential to improve the impedance matching between the thermoacoustic source and surrounding fluid media suggest enhanced designs for compact sonar transducers.

TUESDAY AFTERNOON, 27 JUNE 2017 ROOM 306, 1:20 P.M. TO 3:20 P.M.

Session 3pUWb

Underwater Acoustics: Sound Propagation and Scattering in Three-Dimensional Environments IV

Ying-Tsong Lin, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Frédéric Sturm, Cochair
Acoustics, LMFA, Centre Acoustique, Ecole Centrale de Lyon, 36, avenue Guy de Collongue, Ecully 69134, France

Invited Paper

1:20

3pUWb1. The internal wave prediction component of the Integrated Ocean Dynamics and Acoustics Project. Timothy F. Duda, Ying-Tsong Lin, James Lynch, Arthur Newhall, Weifeng G. Zhang, Karl R. Helfrich (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), and Pierre F. Lermusiaux (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

The accuracy of ocean sound propagation modeling depends on properly representing the environment. The water column portion of this has phenomena and features that are time- and space-dependent, covering a huge range of scales. Nonlinear internal gravity waves (nonlinear internal waves, NIW) in shallow areas are important features that are moving, evolving and anisotropic, and whose effects can be handled properly only with 3D sound modeling. Thus, 3D NIW field predictions would be needed for comprehensive modeling. One challenging part of our project to make acoustic condition forecasts from available data involves making this 3D NIW field prediction using data-assimilating regional models that do not faithfully handle NIW. Our methods for extracting internal tide signals from the models, analyzing their propagation into regions where they transform in reality to NIW (but not in the models), and predicting NIW conditions in these regions are explained here. Outstanding challenges such as how to parameterize internal-tide coupled mode propagation on slopes and how to model highly nonlinear crossing NIW groups will be presented. The question of what constitutes an effective NIW prediction for acoustic purposes will be addressed.

Contributed Papers

1:40

3pUWb2. Four dimensional sound speed environments in ocean acoustic simulations. EeShan C. Bhatt and Henrik Schmidt (Mech. Eng., MIT, Rm. 5-223, 77 Massachusetts Ave., Cambridge, MA 02139, eesh@mit.edu)

Current ocean acoustic simulation environments often rely on a single depth-varying sound speed profile. This work introduces a robust and configurable Octave/C++ implementation of a four-dimensional sound speed environment (4D SSP) for use with Bellhop in the MIT/LAMSS software package for acoustic simulations. This tool shows considerable advances in that the 4D SSP environment can be created from MIT GCM model output, CTD data, or manually varied from existing historical profiles. By considering the fluctuations in sound speed in longitude, latitude, depth, and time, output pressure data better approximates experimental data. Two experiments in the Arctic and the Santa Barbara Channel (SBC) are simulated in 1-D and 4-D to compare to experimental data taken. A framework for acoustic data assimilation using this new simulated data environment in tandem with experimental data is shown to derive the true sound speed field. [Work supported by ONR under the Information in Ambient Noise MURI.]

2:00

3pUWb3. Three-dimensional numerical simulation of sound waves propagating near the west coast of Brittany. Frédéric Sturm (Ctr. Acoustique, LMFA, UMR CNRS 5509, Université de Lyon, Laboratoire de Mécanique des Fluides et d'Acoustique, Université de Lyon, Ctr. Acoustique, Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully Cedex 69134, France, frederic.sturm@ec-lyon.fr)

Numerical results of sound wave propagation in a realistic three-dimensional (3-D) oceanic environment are reported. The region of interest is the west coast of Brittany, near the harbor of Brest, France. The numerical simulations were performed running a fully 3-D parabolic equation based code considering omnidirectional point sources emitting at low frequencies (e.g. 50—100 Hz) and located in the water column near the entrance of the strait linking the roadstead of Brest to the Atlantic Ocean (also known as 'Goulet de Brest'). Numerical simulations clearly show that, depending on position of the acoustic sources, sound waves can be strongly affected by out-of-plane propagation effects resulting from complicated multiple reflections off the sloping bottom and channeling effects due to the three-dimensionally varying bathymetry in this particular region, and hence can predict interesting modal arrivals at specific receivers (with typical source-receiver distances of 30 km), not predicted by two-dimensional models. Note that the 3-D effects predicted here for a realistic marine environment are very similar to the ones described in detail for (now classical) benchmark problems (e.g. 3-D wedge and 3-D canyon test cases), though the environmental parameters are different. Several source depths and positions are investigated.

2:20

3pUWb4. Three-dimensional sound propagation and scattering in an ocean with surface and internal waves over range-dependent seafloor. Ying-Tsong Lin and James Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation in an oceanic waveguide can be influenced by environmental fluctuations on the boundaries at the sea surface and the sea floor and also in the ocean interior. These fluctuations can in fact cause three-dimensional acoustic propagation and scattering effects, especially when the horizontal/azimuthal gradients of the fluctuations are significant. Many studies have only been considering individual environmental factor, but the current work presented in this talk is investigating the joint effects by surface and internal waves over range-dependent seafloor consisting of sand waves, ripples, or scours. This scenario represents better the reality in some dynamic areas on the edge of continental shelf (shelf-break), continental slopes, submarine canyons, and also riverine and estuarine environments. Two research methods are taken here: one is theoretical analysis utilizing acoustic mode theory, and the other is numerical modeling with three-dimensional parabolic-equation models. The frequency dependency of the joint effects will be analyzed, as well as the dependencies on the source and receiver positions, acoustic mode numbers, and/or ray angles. Numerical examples of underwater sound propagation and scattering with realistic environmental conditions will be presented with statistical analysis on the temporal and spatial variability. [Work supported by ONR.]

2:40

3pUWb5. Parameter dependence of acoustic quantities in a nonlinear internal wave duct. Brendan J. DeCourcy, Matthew Milone (Mathematical Sci., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, decoub@rpi.edu), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Parameter dependence of acoustic quantities in a nonlinear internal wave duct BJD, Matt Milone, YTL Ocean features with 3-D spatial variability in shallow water can significantly affect acoustic propagation. One example is a curved front modeled with a discontinuous sound speed change over a sloping shelf [Lin and Lynch, JASA-EL (2012)], which has an extension to a continuous sound-speed change. An approach using normal modes and perturbation approximations yields convenient formulas that show how acoustic quantities depend on environmental parameters [DeCourcy *et al*., ASA, Salt Lake City (2016)]. Another common 3-D example is nonlinear internal waves, with wave fronts that pairwise can produce acoustic ducting, radiating, and scattering effects often observed in field data. The previous approach is applied to this feature, using a well model with two sound-speed jumps for such a duct [Lin *et al*., (2013)]. Approximate formulas for acoustic wavenumbers and phase speeds are determined in order to estimate sensitivity to changes in environmental parameters. All mode types will be considered (whispering gallery, fully bouncing, and leaky), highlighting differences from those in the single-front example. [Work supported by ONR Grants N00014-14-1-0372 and N00014-11-1-0701.]

3:00–3:20 Panel Discussion

Session 3eED

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

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

This workshop for Boston area Girl Scouts (age 12–17) consists of a hands-on tutorial, interactive demonstrations, and a panel discussion about careers in acoustics. The primary goals of this workshop are to expose the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success.

Please e-mail Keeta Jones (kjones@acousticalsociety.org) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. to 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. to 7:30 p.m.).

We will provide many demonstrations, but feel free to contact us if you would like to bring your own. Following is a description of one of the demonstrations.

Contributed Papers

3eED1. Demonstration of nonlinear tuning curve vibration of granular medium supported by a clamped circular elastic plate using a soil plate oscillator. Emily V. Santos and Murray S. Korman (Physics Dept., U. S. Naval Academy, Annapolis, MD 21402, santosemily08@gmail.com)

A demonstration will be conducted in order to show how a soil plate oscillator (SPO) filled with granular material will create a nonlinear system due to shifting peaks in a tuning curve with incremental increases in the swept drive amplitude. An SPO has two flanges clamping an elastic plate which supports a circular column of granular material. The plate (with a magnet and accelerometer fastened to the underside) is driven below by an amplified swept sinusoidal current applied to an AC coil. Past results have used masonry sand, granular edible uncooked materials, and glass beads in order to study the nonlinear tuning curve response near a resonance with (1) a fixed soil column while changing drive amplitude and (2) mass loading of a soil column versus the resonant frequency response of the system at a fixed drive amplitude. When the experiments are performed at a fixed drive but with a changing mass layer the resonant frequency increases then decreases with increased added mass due to an increase in flexural rigidity of the granular media disk layer which dominates the effect of adding mass. SPO tuning curves resemble the nonlinear mesoscopic elastic behavior of resonant effects of geomaterials such as sandstone.

Plenary Session and Awards Ceremony

Michael R. Stinson, Cochair
President, Acoustical Society of America

Jorge Patricio, Cochair
President, European Acoustics Association

Presentation of Certificates to New ASA Fellows

Douglas A. Abraham – For contributions to our understanding of the effect of non-Rayleigh reverberation on active sonar

Joshua G. Bernstein – For contributions to our understanding of normal and impaired pitch and speech perception

David Braslau – For contributions to noise mitigation and quieter communities

Tim Colonius – For contributions to numerical modeling of cavitation, medical acoustics, and aeroacoustics

Elisa E. Konofagou – For contributions to diagnostic and therapeutic applications of ultrasound

Ying-Tsong Lin – For contributions to three-dimensional computational and shallow water acoustics

Tyrone M. Porter – For contributions to therapeutic ultrasound

James A. TenCate – For contributions to nonlinear acoustics of earth materials

Blake S. Wilson – For the development and enhancement of cochlear implants

Introduction of ASA Award Recipients and Presentation of ASA Awards

Student Mentor Award to Daniel A. Russell

William and Christine Hartmann Prize in Auditory Neuroscience to Cynthia Moss

Medwin Prize in Acoustical Oceanography to Jennifer Miksis-Olds

R. Bruce Lindsay Award to Bradley E. Treeby

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Blake S. Wilson

Gold Medal to William M. Hartmann

Vice President’s Gavel to Ronald A. Roy

President’s Tuning Fork to Michael R. Stinson

Introduction of EAA Award Recipients and Presentation of EAA Awards

The EAA AWARD for lifetime achievements in acoustics to Hugo Fastl

The EAA AWARD for contributions to the promotion of Acoustics in Europe to Antonio Pérez Lopéz.

ACOUSTICAL SOCIETY OF AMERICA

R. BRUCE LINDSAY AWARD

Bradley E. Treeby

2017

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is under 35 years of age on 1 January of the year of the Award and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942
Leo L. Beranek	1944
Vincent Salmon	1946
Isadore Rudnick	1948
J. C. R. Licklider	1950
Osman K. Mawardi	1952
Uno Ingard	1954
Ernest Yeager	1956
Ira J. Hirsh	1956
Bruce P. Bogert	1958
Ira Dyer	1960
Alan Powell	1962
Tony F. W. Embleton	1964
David M. Green	1966
Emmanuel P. Papadakis	1968
Logan E. Hargrove	1970
Robert D. Finch	1972
Lawrence R. Rabiner	1974
Robert E. Apfel	1976
Henry E. Bass	1978
Peter H. Rogers	1980
Ralph N. Baer	1982
Peter N. Mikhalevsky	1984
William E. Cooper	1986
Ilene J. Busch-Vishniac	1987
Gilles A. Daigle	1988
Mark F. Hamilton	1989
Thomas J. Hofler	1990
Yves H. Berthelot	1991
Joseph M. Cuschieri	1991
Anthony A. Atchley	1992
Michael D. Collins	1993
Robert P. Carlyon	1994
Beverly A. Wright	1995
Victor W. Sparrow	1996
D. Keith Wilson	1997
Robert L. Clark	1998
Paul E. Barbone	1999
Robin O. Cleveland	2000
Andrew J. Oxenham	2001
James J. Finneran	2002
Thomas J. Royston	2002
Dani Byrd	2003
Michael R. Bailey	2004
Lily M. Wang	2005
Purnima Ratilal	2006
Dorian S. Houser	2007
Tyrone M. Porter	2008
Kelly J. Benoit-Bird	2009
Kent L. Gee	2010
Karim G. Sabra	2011
Constantin-C. Coussios	2012
Eleanor P. J. Stride	2013
Matthew J. Goupell	2014
Matthew W. Urban	2015
Megan S. Ballard	2016

CITATION FOR BRADLEY E. TREEBY

. . . for contributions to the modeling of biomedical ultrasound fields

BOSTON, MASSACHUSETTS • 27 JUNE 2017

Bradley Treeby grew up outside the small town of Albany in South Western Australia. He received a Bachelor of Engineering degree with 1st Class Honors from the Department of Mechanical Engineering at the University of Western Australia in 2003. Continuing as a graduate student in the same department, Brad began his first in-depth studies in acoustics while working under the supervision of Professors Roshun Paurobally and Jie Pan in the Department's Centre for Acoustics, Dynamics and Vibration. Brad's dissertation work examined the effect of hair on human sound localization cues, bringing challenges that necessitated the development of novel scattering models valid over non-rigid surfaces. For his research accomplishments he received the Robert and Maude Gledden Postgraduate Research Scholarship (2004), the F. S. Shaw Memorial Postgraduate Scholarship for excellence in applied mechanics research (2005), and earned the title of Doctor of Engineering in 2007.

Upon completing his degree, Brad moved to the UK to become a Research Fellow at University College London (UCL). Working with Dr. Ben Cox in the in the Department of Medical Physics and Bioengineering, Brad began investigating methods for fast tissue-realistic modeling of wave propagation relevant to photoacoustics.

From this time onward Brad focused his career on the development of fast and accurate models for describing ultrasound waves traveling through the human body. His work at UCL on pseudospectral time domain models for acoustic wave propagation grew into what is today the well-known open source "k-Wave" Matlab toolbox for modelling biomedical ultrasound fields.

"Brad developed k-Wave all the way from theoretical principles, through coding and validation, to its applications to real world problems. In doing so, he has demonstrated considerable expertise and attention to detail in an impressive range of areas of acoustics," notes Cox.

Key steps leading to k-Wave's widespread use include Brad's theoretical work on using fractional Laplacian terms to model tissue-realistic acoustic absorption and dispersion, his extension of the model from the linear to the nonlinear regime, his developments to make the model computationally efficient, his formation of the model for large-scale simulations on high performance computing architectures, his validation of the models against experimental measurements, and his active daily support to users through an online user forum. Since this freely-distributed software was released in 2009 there have been seven subsequent releases, each overseen by Brad. k-Wave now has some 9,000 registered users in at least 70 countries, and a paper describing the first release of the toolbox has been cited over 375 times since its publication in 2010 (Treeby, B.E. and B. T. Cox "k-Wave: MATLAB toolbox for the simulation and reconstruction of photoacoustic wave fields," J Biomed Opt 2010; 15: 021314).

Between 2010 and 2013 Brad spent time at Australian National University (ANU) serving as a research fellow. While continuing his research throughout this period, Brad's dedication as an educator and mentor became clear during his time: He received both a Top Supervisor Award and a Dean's Commendation for Teaching Excellence in his relatively short tenure at ANU.

In 2013 Brad returned to London to establish a new Biomedical Ultrasound Group– or "BUG"– at UCL and has nurtured its growth since. Currently supported by an Early Career Fellowship from the Engineering and Physical Sciences Research Council, he has maintained notable levels of grant funding, allowing growth of the lab's experimental and computational capabilities. Brad is very active in the application of full wave models to clinical problems, notably for accurate targeting and dosimetry in therapeutic ultrasound. To this end, he has established collaborations with a wide circle of clinicians, scientists, manufacturers, and other interested parties from around the world. His ongoing research, "sits at the interface between physical acoustics, biomedical ultrasound, numerical methods, and high performance computing," which has earned much respect and attention among the ultrasound modeling community and has attracted both graduate students and postdocs to his lab.

Brad's affinity for acoustics is clearly not limited to the scientific. He is an accomplished guitarist and vocalist, whose most prominent works stem from the band of his namesake, Brad Treeby & the Simplists. Founded in Perth in 2008, the group moved to London along with Brad and remained active through 2013 with their "unmistakable syndicate of beats and melody forged from the combination of acoustic guitar, vocals, bass, and human beat-box."

While he is clearly accomplished in numerical methods, Brad is equally interested and active in experimental work. As his mentor and colleague Cox has observed a hallmark of Brad's character as an investigator with his "insistence that research needs a symbiotic approach that includes not only modelling and experimentation, but also an eye on the end goal, on the application. Because of his k-Wave software, some might mistakenly categorize Brad as just an expert in numerical methods of acoustics. That would be an injustice."

We are delighted to congratulate Bradley E. Treeby on behalf of his colleagues, friends, and supporters throughout the Acoustical Society of America on being selected for the 2017 R. Bruce Lindsay Award.

NATHAN J. MCDANNOLD

ACOUSTICAL SOCIETY OF AMERICA
HELMHOLTZ-RAYLEIGH INTERDISCIPLINARY
SILVER MEDAL
in

Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics

Blake S. Wilson

2017

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

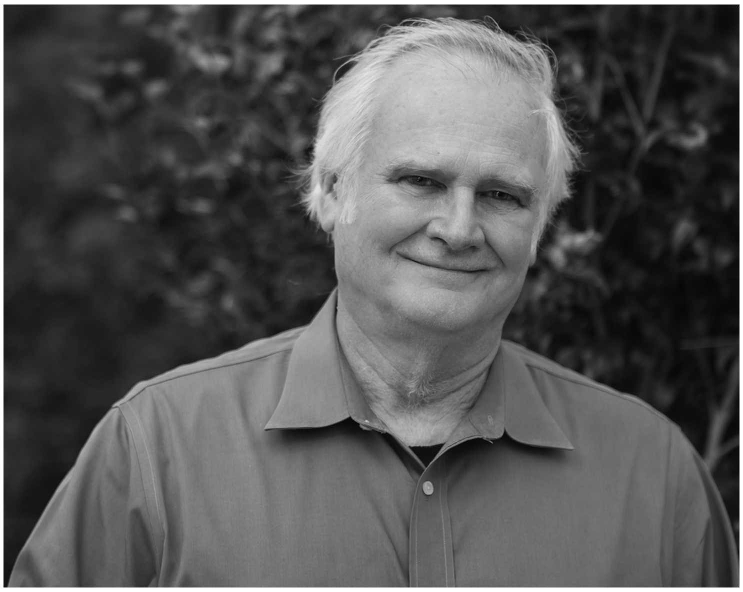
PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997
David E. Weston	1998
Jens P. Blauert	1999
Lawrence A. Crum	2000
William M. Hartmann	2001
Arthur B. Baggeroer	2002
David Lubman	2004
Gilles A. Daigle	2005
Mathias Fink	2006
Edwin L. Carstensen	2007
James V. Candy	2008
Ronald A. Roy	2010
James E. Barger	2011
Timothy J. Leighton	2013
Mark F. Hamilton	2014
Henry Cox	2015
Armen Sarvazyan	2016

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993



ENCOMIUM FOR BLAKE S. WILSON

. . . for contributions to the development and adoption of cochlear implants

BOSTON, MASSACHUSETTS • 27 JUNE 2017

Blake S. Wilson directed the Neuroscience Program, then the Center for Auditory Prosthesis Research, at Research Triangle Institute in North Carolina, over a 20+ year period beginning in 1983. During this time, Blake and his research teams developed a suite of highly effective signal processing strategies for cochlear implants–devices that restore hearing and speech understanding to infants born deaf and to adults who have lost most, or all, of their hearing. Today, the signal-processing strategies developed by Blake and his teams, or direct descendants of those strategies, are the heart of the cochlear implants used worldwide by over 400,000 individuals ranging in age from a few months to over 100. Cochlear implants are the first and most successful neural prosthesis for a sensory system and have been described as one of the most significant medical developments in the second half of the twentieth century. Blake's work has been central to this remarkable achievement.

Blake received a B.S. in Electrical Engineering from Duke University in 1974 and probably set a record for the number of humanities courses, predominately English, taken by an EE major. Having educated both sides of his brain, he immediately went to work down the road at the Research Triangle Institute (now RTI International) where he stayed for 33 years working his way up from Research Engineer to Senior Fellow. He became Chief Strategy Officer for MED-EL GmbH, a manufacturer of cochlear implants, in 2007, and in 2008 he founded the Duke Hearing Center, with Debara L. Tucci, M.D. The next year he became director of the MED-EL Laboratory for Basic Research.

Blake's first paper was on "pinna reflections as a cue for localization." (J. Acoustical Soc. Am. 56, 957-962, 1974). Other early publications reported results on bat biosonar and the effects of microwave action on the auditory system. His introduction to the problems resulting from deafness came from a project in which the outputs of speech analyses were sent to LED displays mounted on the frame of eyeglasses worn by the deaf in an effort to disambiguate visual information about speech.

This experience lead to a successful bid on a contract from the Neural Prosthesis Program at the National Institutes of Health (NIH) in 1983 to design signal-processing strategies for cochlear implants. At this point in the development of cochlear implants, despite the presence of as many as 22 electrodes in the cochlea, speech-understanding scores were very poor. What was missing was a highly effective signal-processing strategy. In 1989, Blake and his team invented and began to test such a strategy--the continuous interleaved-sampling strategy or CIS, building on previous work at the University of San Francisco by Michael Merzenich, at Stanford University by Robert White and Blair Simmons, and at the Massachusetts Eye and Ear Infirmary by Donald Eddington and William Rabinowitz. In this strategy, speech is filtered into a number of bands, the energy in each band is estimated and pulses, proportional to the energy in the band, are output to electrodes in the scala tympani. Two aspects of this strategy were critical to its success. First, the pulses were sequenced over time in an interleaved fashion across electrodes so that vector summation of electric fields, that would arise from simultaneous pulse outputs, was minimized.

Second, the rate of stimulation was much higher than had been used before and, as a consequence, both spatial information about place of cochlear stimulation and temporal information were represented. The results of the first clinical test of this strategy were published in 1991 in *Nature* (Nature 352: 236-238, 1991). Scores on tests of sentence understanding in quiet improved significantly with the majority of the patients achieving scores of greater than 90% correct. This paper heralded a new era in the field of sensory prosthetics and is the most cited paper in the field of cochlear implants. Later work by Blake and his research teams produced multiple variants of this strategy, all of which are used in cochlear implants today.

Blake has been the recipient of many awards as is fitting for a person making discoveries that have restored functional hearing for deaf infants and deafened adults. These include the Lasker-DeBakey Clinical Medical Research Award (2013) "for the development of the modern cochlear implant" and the Fritz J. and Dolores H. Russ Prize from the National Academy of Engineering (2015) "for engineering cochlear implants that allow the deaf to hear." As befits his multidisciplinary work, he was awarded two honorary degrees in medicine in 2015, one from Uppsala University, Sweden, and one from the University of Salamanca, Spain.

Two aspects of Blake's career are critical to an appreciation of his achievements. First, his discoveries with respect to signal processing for cochlear implants came when he was armed with only a baccalaureate degree in EE. He did not have a Ph.D. program to teach him the research enterprise, or a postdoctoral fellowship in an important laboratory to sharpen his skills. He made his discoveries by building a multi-disciplinary research team and then spending decades of long hours in the laboratory. Having made the odd discovery or two, he then acquired a D.Sc. degree from the University of Warwick in the U.K., a Doctor of Engineering degree from the University of Technology, Sydney, and a Ph.D. in EE from his Alma Mater, Duke University.

Second, early on in his tenure at RTI, Blake, working in conjunction with the RTI administration, made the well-considered decision to place all of his work in the public domain and, in doing so, to relinquish rights to his intellectual property. This decision was made in order to speed the adoption of his work by manufacturers of cochlear implants. A conservative estimate of the value of his intellectual property rights is 10's of millions of dollars.

Given his long time residence in the Raleigh Durham area, it is no surprise that Blake is an avid fan of Duke basketball and Coach K. Indeed, it was Duke basketball that gave Blake his largest audience. Several years ago, during a nationally televised game, the TV camera was panning over the audience and settled on Blake, who looked very comfortable in the 'standing room only' section. He is a tennis fan, as well as enthusiastic player, and can regularly be seen poring over scientific texts while sporting the colorful shoes of his on-court heroes.

There is no question that Blake's pioneering research on cochlear implants has linked auditory physiology with auditory perception, and speech perception and spoken-language processing in adults and children. His life's work has made a major contribution to improving the quality of life for many profoundly deaf individuals. For these reasons, we are pleased to congratulate Blake Wilson for being awarded the ASA Helmholtz-Raleigh Interdisciplinary Silver Medal in Speech Communication, Psychological and Physiological Acoustics and Signal Processing in Acoustics.

MICHAEL DORMAN
FAN GANG ZENG
JOHN HANSEN

ACOUSTICAL SOCIETY OF AMERICA

GOLD MEDAL

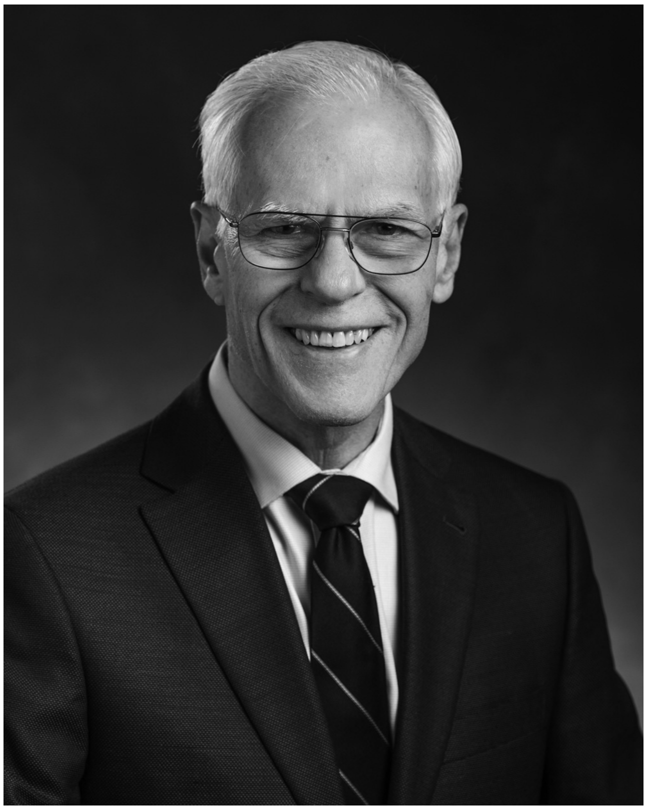
William M. Hartmann

2017

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954
Floyd A. Firestone	1955
Harvey Fletcher	1957
Edward C. Wente	1959
Georg von Békésy	1961
R. Bruce Lindsay	1963
Hallowell Davis	1965
Vern O. Knudsen	1967
Frederick V. Hunt	1969
Warren P. Mason	1971
Philip M. Morse	1973
Leo L. Beranek	1975
Raymond W. B. Stephens	1977
Richard H. Bolt	1979
Harry F. Olson	1981
Isadore Rudnick	1982
Martin Greenspan	1983
Robert T. Beyer	1984
Laurence Batchelder	1985
James L. Flanagan	1986
Cyril M. Harris	1987
Arthur H. Benade	1988
Richard K. Cook	1988
Lothar W. Cremer	1989
Eugen J. Skudrzyk	1990
Manfred R. Schroeder	1991
Ira J. Hirsh	1992
David T. Blackstock	1993
David M. Green	1994
Kenneth N. Stevens	1995
Ira Dyer	1996
K. Uno Ingard	1997
Floyd Dunn	1998
Henning E. von Gierke	1999
Murray Strasberg	2000
Herman Medwin	2001
Robert E. Apfel	2002
Tony F. W. Embleton	2002
Richard H. Lyon	2003
Chester M. McKinney	2004
Allan D. Pierce	2005
James E. West	2006
Katherine S. Harris	2007
Patricia K. Kuhl	2008
Thomas D. Rossing	2009
Jiri Tichy	2010
Eric E. Ungar	2011
William A. Kuperman	2012
Lawrence A. Crum	2013
Brian C. J. Moore	2014
Gerhard M. Sessler	2015
Whitlow W. L. Au	2016



ENCOMIUM FOR WILLIAM M. HARTMANN

. . . for contributions to research and education in psychological acoustics and service to the society

BOSTON, MASSACHUSETTS • 27 JUNE 2017

William (Bill) Hartmann studied electrical engineering at Iowa State University starting in 1957. His (incomplete) conversion to the study of physics began in 1960 when the United States launched the Echo satellite, a large metalized balloon designed to reflect both microwave signals and sunlight so that it could be seen by the entire world. Bill was given the task of using orbital parameters from the US Naval Observatory to predict the appearance of the Echo over Ames, Iowa, which he achieved by programming the Cyclone, a five-ton vacuum-tube machine with 1024 words of 40-bit memory stored on Williams tubes. At the time, both artificial earth satellites and computing were hot topics. Perhaps this is why he was awarded a Rhodes scholarship to study at Oxford University in the UK, from 1961 to 1965.

At Oxford, Bill studied condensed matter theory under the direction of Roger Elliott. He completed a two-part thesis – neutron scattering from liquid and solid hydrogen, and infra-red absorption from defective rare-gas crystals. Ever drawn to sources of cold neutrons, Bill then took a post-doctoral position at Argonne National Laboratory (1965-1968). There, he determined how to introduce short-range order into the lattice dynamics of alloys such as copper-gold. Easily the most important result of Bill's stay at Argonne was meeting Christine Rein, a widely travelled school teacher and summer-time tennis instructor. She and Bill married in 1967 and they have been playing tennis ever since. Chris is a good friend to many members of the ASA. They have two children, Mitra, a professor of engineering at Northwestern, and Daniel, a biomechanical engineer with Eli Lilly.

Bill joined the faculty of the Department of Physics at Michigan State University in 1968. In the early 1970s, his life took a sudden turn beginning with "Switched on Bach," which got Bill hooked on electronic music. Op-amp chips made it easy to build analog music synthesis electronics, and Bill began to explore making music electronically and to teach a course in musical acoustics. In making electronic music, Bill discovered effects that did not seem to make sense. His perceptions did not square with what he knew his electronics were generating. Bill began to do perceptual experiments, mostly on pitch, to try to understand what he was hearing. By accident, he met David Wessel, then assistant professor of Psychology at Michigan State, who told Bill that there were people who made a living by studying such problems in auditory perception. Bill demanded to know where such people were and whether any of them ever wrote about what they were doing. David explained that such people could be found at meetings of the Acoustical Society of America (ASA), and that they published in the *Journal of the Acoustical Society of America* (JASA). David advised Bill to learn the tricks of the trade with David Green at Harvard University. So Bill and Chris packed up the family and spent the academic year (1976-1977) with David Green.

Bill joined the ASA in 1976 and attended the fall meeting in San Diego. He went right to work and organized a special session on electronic music for the next meeting. Bill has missed only one ASA meeting since then. His occasional performances of rap at jam sessions are not to be missed.

Bill's contributions to our field can be divided into three broad categories: research, education, and service to the ASA. We consider them in that order.

Bill's research has spanned a broad set of interrelated topics in acoustics and psychoacoustics and he has made seminal contributions in multiple areas. His work is characterized by a rigorous specification of the problem, careful measurements of complex phenomena, a rigorous mathematical description and analysis of the results, and discussion of implications for more general problems. His very considerable technical and mathematical skills enable him to perform analyses and construct models in a way that would be difficult or impossible for many others working in the same fields.

Bill has made major contributions in several areas: (1) the ability of humans to localize sounds in space, especially when room reverberation and other sources are present;

(2) pitch phenomena, including pitches created from differences in the sounds at the two ears; (3) the perceptual analysis of mixtures of sounds from different sources, often called "scene analysis"; and (4) modulation detection, including AM, FM, and mixed modulations. His more recent work includes modeling that incorporates current knowledge and ideas about processing in the auditory periphery and neural coding up to the midbrain.

As an example of the nature of Bill's contributions in the areas listed above, we consider his substantial contributions to knowledge in the field of pitch perception. Some of Bill's early work was concerned with the perception of frequency modulation. He proposed an influential model to explain the detection of frequency modulation. He also explored the influence of the envelope amplitude of a sound on its pitch and he measured how the pitch of a single mistuned harmonic in a complex tone was influenced by the amount of mistuning. Somewhat later, he conducted an important series of experiments on the effects of mistuning a harmonic in a complex tone on pitch perception and auditory scene analysis. Bill also showed that the "octave enlargement effect" occurs for Huggins pitch (a pitch created by binaural interaction), demonstrating that the effect was unlikely to have a peripheral origin, as previously assumed. All of these papers had a considerable influence on theories of pitch perception.

Bill's interest in auditory scene analysis stemmed from his work on pitch perception, and he was one of the first to give a comprehensive overview of the role of pitch in auditory scene analysis. He published a highly influential paper on the factors that influence the perceptual organization of rapid sequences of sounds [Hartmann, W. M., and Johnson, D. (1991). "Stream segregation and peripheral channeling," Music Percept. 9, 155-184]. He also studied the role of spatial and temporal factors that influence the ability to understand speech in the presence of other sounds.

Bill has contributed greatly to education in acoustics via his two books, *Sound, Signals, and Sensation* (now in its fifth printing) and *Principles of Musical Acoustics* - as well as many chapters in other books. In addition, Bill has educated many undergraduate and postgraduate students over the years, several of whom have gone on to become distinguished researchers in their own right. Bill has organized six special sessions at various ASA meetings. He was the ASA Technical Program Chair for the Acoustics'08 Paris meeting held jointly with the European Acoustics Association, the largest ASA meeting ever held, with over 5,000 registrants.

Bill has made many additional contributions to the ASA. He was elected a Member of the Executive Council (1992-1995), and later as Vice President (1998-1999), President-Elect (2000-2001) and President (2001-2002) of the ASA. He has served on many ASA committees, including the Medals and Awards Committee, Committee on Education in Acoustics, Investments, Panel on Public Policy, and the Technical Committee on Psychological and Physiological Acoustics. He was chair of the Technical Committee on Musical Acoustics, the Books+ Committee, and the Rules and Governance Committee. He was an active member of the Re-Creation Committee (1993-1994) and Vision 2010 (2004 and 2005), which set the direction for the future of the Society.

In 2011 the William and Christine Hartmann Prize in Auditory Neuroscience was established by the ASA to recognize and honor research that links auditory physiology with auditory perception or behavior in humans and other animals. The prize was underwritten by a substantial donation from Bill and Chris.

Bill was awarded ASA's Science Writing Award for Professionals in Acoustics in December 2000. In 2001, he received the ASA Helmholtz-Rayleigh Interdisciplinary Silver Medal "for research and education in psychological and physiological acoustics, architectural acoustics, musical acoustics, and signal processing", showing the diverse areas of acoustics in which he has made substantial contributions.

In summary, Bill has made tremendous contributions to knowledge and education in acoustics and he has shown outstanding loyalty and devotion to the ASA. We congratulate him most warmly on the award of the Gold Medal of the Acoustical Society of America.

BRIAN C. J. MOORE
H. STEVEN COLBURN

ENCOMIUM FOR HUGO FASTL

BOSTON, MASSACHUSETTS • 27 JUNE 2017

EAA AWARD for Lifetime Achievements in Acoustics

Hugo Fastl is an expert in the field of psychoacoustics and related fields. He not only has established the foundations of models for various psychoacoustic quantities and sound quality metrics, but he also serves as a very prominent reference and central node in a large network of international research and standardization.

Hugo Fastl first studied music (string bass) at the Music Conservatory Munich (graduated 1969), followed by electrical engineering at Technical University of Munich (graduated 1970), and received his Ph.D. from the Technical University of Munich in 1974 with supervision from Eberhard Zwicker. He then continued his academic career by adding a second doctorate (habilitation) in 1981; both dissertations made outstanding contributions to hearing research, particularly on masking thresholds as a measure for temporal and spectral resolution of the hearing organ. In 1987, he served as visiting professor at the University of Osaka in Japan, and since 1988 he has served as professor and head of the "Technical Acoustics" group at the Technical University of Munich.

His work covers a wide range of topics, including psychoacoustic quantities (masking, loudness, sharpness, roughness, and fluctuation strength), audio communication and speech intelligibility, noise abatement on vehicles and road surfaces, sound quality and sound design, musical acoustics (pitch perception, pitch strength), and audiology (hearing devices, cochlea implants, and the eponymous "Fastl noise" for speech audiometry). Even with his enormous efforts in fundamental research, Hugo Fastl has always found ways to transfer his results into practical and industrial applications. He has also been key in promoting psycho-acoustical quantities within acoustic engineering, successfully bridging the fields of diverse disciplines. In particular, his book "Psychoacoustics – Facts and Models", written initially with Eberhard Zwicker and subsequently revised and published under his own name multiple times, has gained very high international recognition lasting until today – the 2013 edition alone scores about 4500 citations on Google Scholar. No education in Psychoacoustics fails to integrate this remarkable work.

For his research, Hugo Fastl followed a strong multidisciplinary approach in collaboration with Japanese researchers and in supervising many joint projects in the field of Psychology. The intercultural difference in multimodal perception was one of the most significant aspects in these projects.

Among various memberships in national and international councils and boards, Hugo Fastl has been active in the German Acoustical Society (as chair of its TC Hearing Acoustics 1991-1997, as treasurer 1996-2004, as conference chair 2005, and finally as president 2004-2007). From 2004 to 2010, he also served as the treasurer of the ICA. He has received various prestigious national and international awards: among them, the 1990 Fellow of the Acoustical Society of America, the 1998 Research Award of the Japan Society for the Promotion of Science, the 2003 Rayleigh Medal of the British Institute of Acoustics, the 2010 Helmholtz Medal of the German Acoustical Society, and in 2014 he was appointed as Distinguished International Member of INCE-USA.

At their core, Hugo Fastl's interests are based in psychological and physiological acoustics. Ultimately, however, a rich balance of academic and applied work characterizes his career. This has enabled his participation in a large number of projects and the support of numerous young researchers in his field. His graduate and postgraduate students carry on his ideas and inspiration, and they do so very successfully in industry and academia alike.

The EAA now honors Hugo with its award for lifetime achievements. This sounds final and complete, especially since in the present case the honorable colleague has already crossed the age limit of university positions in Germany.

Not so with Hugo Fastl, who does not even think about resting on his achievements. Anyone who knows him is aware that retirement is by no means a consideration. So we will be happy to meet him again at national and international events, to discuss acoustics, and share a Bavarian beer at DAGA 2017 in Munich. In recognition of his outstanding contributions, we congratulate Hugo Fastl for being awarded the EAA Award for Lifetime Achievements in Acoustics.

MICHAEL VORLAENDER
BRIGITTE SCHULTE-FORTKAMP

ENCOMIUM FOR ANTONIO PÉREZ LÓPEZ

BOSTON, MASSACHUSETTS • 27 JUNE 2017

EAA AWARD for contributions to the promotion of Acoustics in Europe

Antonio Pérez López is perhaps the person who will be considered in the history of European Acoustics as the most representative of a European Acoustician.

He graduated from the University of Madrid with a degree in Physics and later he got the Diploma in Audiophonology from the same University. He continued his studies in England obtaining a Master D.I.C. by the Imperial College, University of London. He started his career in Acoustics at the Spanish National Research Council in Madrid and served as the Chief of the Noise Lab. Later, he worked as the General Director of the Spanish Branch of the German Group "Rheinhold & Mahla".

From the very beginning, Antonio showed a unique ability to combine science with education and production. He was able to speak with students at the University presenting, in a professional way, the scientific aspects of sound and noise, to share new ideas with scientists working exclusively in the research sector, and at the same time to decide on the production of new materials related to noise and vibration control. He also demonstrated his administrative skills working in the industry and taking care of its running issues including the working relations of the employees of all ranks.

His involvement in all possible aspects of acoustics, in connection with his personal character and his belief in the necessity for collaboration between people and the dissemination of knowledge, dictated the way he dealt with the scientific unions devoted to science and in particular to acoustics. First, he was involved in Spanish acoustics becoming founding member of the Spanish Acoustical Society (Sociedad Española de Acústica (SEA)) and serving it as General Secretary and later as President, a position he holds until now. Under the leadership of Antonio the SEA has been now widely recognized as a model society in Europe. Checking the activities of the SEA which are rich of innovations and consistent activities, people will soon recognize that Antonio is behind the scenes.

Feeling European, he also dedicated his activities to the European Acoustics Association (EAA). He was the person to resolve the legal status of EAA by registering the association under the Spanish Law and undertaking all the necessary preparatory actions. He, also, offered the headquarters of the Spanish Acoustical Society to host the office of EAA. Thereafter, Antonio was appointed as the Director of the EAA office in Madrid, but being determined to promote the idea of a unified Europe he offered himself as the vital link between all the EAA boards and the National Societies. He was the person to look after all the European National Societies, especially the small or new ones, being in close contact with their boards and trying to show them the road map to success. Being member of the EAA executive council he was the person to ensure the continuation of the function of the EAA and its relation with other international Federations, Commissions, and/or Associations. He was always "pulling the strings" in the background, suggesting and implementing innovations in administrative, educative, and scientific aspects, mitigating conflicts, thinking forward, and giving the EAA a friendly and cooperative spirit.

Antonio has many beliefs: He believes in Science, in Solidarity, in Europe, in People, in Cooperation. He treats everybody as if he or she is a member of his family, trying to educate, comfort, introduce motivations, discuss future plans, and resolve current issues. He is always a good friend, “father” or “brother” to his collaborators. He is a real volunteer in life and science. He likes and believes in youth. He is able to communicate effectively with the young people. Among the many activities he has devoted to the youth, one may mention the production of educational material for young students aiming at their introduction to sound and noise, his support to the Young Acousticians Network of the EAA always suggesting the Presidents and the board to invest on the young scientists, and his continuous efforts to organize seminars and special courses for young acousticians.

Antonio is well known worldwide for his activities, having served the science of acoustics from different posts. He is a Member of the board of the International Commission for Acoustics since 2007 and Treasurer since 2010. He has been President of the FIA (Ibero-American Federation of Acoustics), currently being member of the board. He has organized many International Conferences and Symposia which remain in history of acoustics for their success. He has established good relations with many non-european countries for instance Morocco, Tunisia, Nigeria, and Algeria bringing to them the air of European acoustics and asking them to actively participate in European acoustics events.

Antonio will keep going, showing the way to all his friends and fellow acousticians.

MICHAEL TAROUDAKIS