Session 2pAAa

Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Adapting, Enhancing, and Fictionalizing Room Acoustics II

K. Anthony Hoover, Cochair
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Invited Papers

1:00

2pAAa1. Amplified music and measurement results regarding inflatable membrane absorber technology. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

Previous studies and experience have shown that what distinguishes the best from the less well liked venues for pop and rock music is a shorter reverberation time in the 63–250 Hz octave bands. Since a longer reverberation time in these bands is needed in order to obtain warmth at classical music concerts, variable acoustics must address these frequencies in order to provide the best results in multi-purpose halls. This paper will expand on research on recommendable acoustics for amplified music. Certified measurements from reverberation chambers and installed systems on a patented, inflatable, on/off absorption technology are presented. Since the technology can be used in the entire ceiling area, the T30 of a hall can be lowered by almost 50% in the important octave bands for pop and rock music. Absorption coefficients are almost constant across the frequency bands from 63 to 1000 Hz. The technology, which is the only passive solution to enable variability also at the important lower frequencies, is meant to be used in any hall where both classical as well as amplified music is being played, such as in music schools and performing arts centers.

1:20

2pAAa2. The role of high-frequency cues for spatial hearing in rooms. Hari Bharadwaj, Salwa Masud, and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol. (CompNet), Boston Univ., 19 Euston St., 1B, Brookline, MA 02446, hari@nmr.mgh.harvard.edu)

The ability to attend to a sound source of interest while ignoring competing sounds is vital to navigating everyday acoustic scenes. Commonly, this ability depends on the ability to focus on a sound source using acoustic spatial cues, particularly interaural time differences (ITDs) and interaural level differences (ILDs). Based on past studies of localization in anechoic settings, low-frequency ITDs have been thought to dominate perception of source location. However, reverberant environments differentially degrade ITDs and ILDs, which may affect their relative influence on localization. Moreover, a recent study suggests that ILDs play a bigger role on spatial perception in reverberant settings than in anechoic settings. Here, in a series of localization and spatial attention experiments using high-pass, low-pass and broadband sounds, we tested the hypothesis that high-frequency ILD and envelope ITD cues are important for spatial judgments in reverberant rooms. We also measured the brainstem frequency following responses (FFRs) of individual subjects in response to click trains and spoken syllables. Results suggest that compared to in anechoic space, in reverberant settings, high-frequency cues are more reliable and influential on perception and that the strength of FFR phase locking to stimulus envelope predicts how well individual listeners can direct spatial attention.

1:40

2pAAa3. Internet rooms from internet audio. Chris Chafe and John Granzow (CCRMA/Music, Stanford Univ., CCRMA/Music, Stanford, CA 94305, cc@ccrma.stanford.edu)

Music rehearsal and concert performance at a distance over long-haul optical fiber is a reality because of expanding network capacity to support low-latency, uncompressed audio streaming. Multichannel sound exchanged across the globe in real time creates “rooms” for synchronous performance. Nearby connections work well and musicians feel like they are playing together in the same room. Larger, continental-size, distances remain a challenge because of transmission delay and seemingly subtle but perceptually important cues which are in conflict with qualities expected of natural rooms. Establishing plausible, room-like reverberation between the endpoints helps mitigate these difficulties and expand the distance across which remotely located musicians perform together comfortably. The paper presents a working implementation for distributed reverberation and qualitative evaluations of reverberated versus non-reverberated conditions over the same long-haul connection.

2:00

2pAAa4. Designing a stackable diffusor/absorber tailored to a violin and cello practice room. Myoung woo Nam, Kyogu Lee (Trans-Disciplinary Studies, Seoul National Univ., D406, Iuidong 864-1, Yeongtonggu, Suwon, Gyeonggi 443-270, South Korea, mnam@snu.ac.kr), and Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts, Lowell, MA)

As distinct from a much larger concert hall, the typical practice room directs added acoustic emphasis to small room challenges such as room resonances and unwanted reflections. Although rich reverberation is not easily achieved in a small space, the proposed diffusor/absorber seeks to make practice more acoustically comfortable and rewarding. The treatment is designed to attenuate the spectral portion

of the violin and cello sound that engages the room resonances. When musicians play violin and cello, omni-directional low frequencies are primarily produced in lower elevation of the room, while more directional higher frequencies of interest to the performer are directed more to upper area. This diffusor/absorber is designed to provide more absorption for bottom area and more diffusion for upper area. In this configuration, the diffusor/absorber gives comfortable acoustical conditions for musicians to practice. Based on sound propagation characteristics [J. Meyer, "The sound of the orchestra," J. Audio Eng. Soc. **41**(4) (1993)] and formant information of violin and cello [M. Nam and K. Lee, "Analyzing string instrument formant," in *Proceeding of Acoustical Society of Korea Conference* (2011)], this design proposes a moveable acoustic panel-box suitable for a typical musician's home practice room and a small sized recording studio.

Contributed Papers

2:20

2pAAa5. Improving the indoor sound quality by using cymatic shapes. Alaa S. Algargoosh (College of Design, Univ. of Dammam, 3140 Alabbas bin Ali, Dammam 32433-4614, Saudi Arabia, alaa_algargoosh@yahoo.com), Hany Hossam Eldien (College of Architecture and Planning, Univ. of Dammam, Dammam, Saudi Arabia), and Hala El Wakeel (College of Design, Univ. of Dammam, Dammam, Saudi Arabia)

Acoustic diffusers are important components in enhancing the quality of room acoustics. This paper investigates a new type of 2D diffusers obtained by the Cymatics phenomena. Cymatics is the study of sound and vibration made visible, typically on the surface of a plate, diaphragm, or membrane. Four shapes of the diffusers were designed by the Cymatic shapes and modeled by using a quadratic residue sequence. The polar response of the diffusers was measured using DIRAC software. Polar response results were generally consistent with expectations. This type of diffusers can generate a uniform polar response over the frequency range we are interested in (400–4000 Hz). It is found that this type of acoustic diffusers can be used to maintain the acoustic energy in a room and at the same time can treat unwanted echoes and reflections by scattering sound waves in many directions.

2:40–3:00 Break

3:00

2pAAa6. Sound concentration caused by curved surfaces. Martijn Vercammen (Peutz, Lindenlaan 41, Molenhoek 6584 AC, Netherlands, m.vercammen@mook.peutz.nl)

In room acoustics, the focusing effect of reflections from concave surfaces is a well-known problem. The occurrence of concave surfaces has tended to increase in modern architecture, due to new techniques in design, materials, and manufacturing. Focusing can cause high sound pressure levels, sound coloration, or an echo. Although the problem is well known, the amount of amplification that occurs in the focusing point and the sound field around the focusing point are not. The pressure in the focusing point can only be calculated using wave-based methods. An engineering method that is based on the Kirchhoff Integral is presented to approximate the reflected sound field in and around the focusing point for a few basic geometries. It will be shown that both the amplification and the area of the focusing is strongly related to wavelength. The focusing caused by surfaces that are curved in two directions (sphere, ellipsoid) is much stronger than that caused by surfaces that are curved in only one direction (cylinders). This method enables designers to evaluate and thereby improve or redesign the geometry. The method is illustrated with a few examples.

3:20

2pAAa7. Can we use the standard deviation of the reverberation time to describe diffusion in a reverberation chamber? Margriet Lautenbach and Martijn Vercammen (Peutz, Paletsingel 2, Zoetermeer 2718 NT, Netherlands, m.lautenbach@zoetermeer.peutz.nl)

It is generally assumed that the limited diffusion properties of reverberation rooms, especially with a strongly sound absorbing sample, is the main reason for the bad reproducibility values for the sound absorption between laboratories. Reverberation rooms should be made much more diffuse to reduce the inter laboratory differences. Although there are practical ways to achieve this, it is most important that there will be a requirement in the ISO 354 standard on the diffusing quality of the sound field. One possibility is to use the standard deviation of the reverberation time for different source-microphone combinations in the reverberation room. Measurements are performed to investigate the influence of different settings of a reverberation room on the standard deviation of the reverberation time, compared to the theoretical standard deviation. This is done with the interrupted impulse method and the integrated impulse method. The results will be presented in this paper. The usefulness of this qualification method for the ISO standard will be discussed.

3:40

2pAAa8. A new third generation time variant electro-acoustic enhancement system. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

This paper describes the hardware and software implementation in a new generation of time variant electronic acoustic enhancement systems designed for use in medium and large sized venues. Examples of currently installed systems and applications will be discussed, as well as capabilities for sound file storage and playback, multi-channel film surround sound, 2D effects panning, and 3D live tracking.

4:00

2pAAa9. Optimization of electroacoustic resonators for semi-active room equalization in the low-frequency range. Etienne Rivet (Laboratoire d'Électromagnétisme et d'Acoustique, École Polytechnique Fédérale de Lausanne, EPFL - STI - IEL - LEMA, Station 11, Lausanne, Vaud 1015, Switzerland, etienne.rivet@epfl.ch), Romain Boulandet, and Hervé Lissek (Laboratoire d'Électromagnétisme et d'Acoustique, École Polytechnique Fédérale de Lausanne, Lausanne, Switzerland)

At low frequencies in listening rooms, standing waves cause large frequency-response variations within the whole space. These unwanted phenomena have a significant impact on the sound quality of an audio system rendering. Unfortunately, state-of-the-art soundproofing solutions cannot efficiently handle such low-frequency sound energy. To alleviate this problem, electroacoustic resonators can be used to damp room modes. This concept is based on the connection of direct-radiator loudspeakers to synthetic electrical loads allowing the passive dissipation of a certain part of the incoming acoustic energy of the sound field. Through judicious control of acoustic impedance, and depending of the placement of the electroacoustic resonators in the room, a significant damping of the dominant natural resonances can be achieved in order to meet the specifications of sound reproduction. This paper presents the design of prototypes of electroacoustic resonators and investigates their optimization and spatial arrangement in the perspective of semi-active room equalization.

4:20

2pAAa10. Physical and subjective factors of spatial envelopment impression of surround sound reproduction. Toru Kamekawa and Atsushi Marui (Musical Creativiry and the Environ., Tokyo Univ. of the Arts, 1-25-1, Senju, Adachi-ku, Tokyo 120-0034, Japan, kamekawa@ms.geidai.ac.jp)

How we feel the envelopment of reproduced sound? For the evaluation of spatial impression, "envelopment" is one of the key factors among several attributes in audio reproduction. In the authors' past research, three attributes were elicited from participants regarding surround sound reproduction using triadic elicitation procedure. The three attributes are "brightness," "temporal separability," and "spatial homogeneity of envelopment." In this paper, authors focused on "spatial homogeneity of envelopment" and considered related physical parameters such as ESC (ear signal correlation; defined as the correlation between left and right ears' signal). From comparison between the result of subjective test and the data measured with a dummy-head facing toward different

angles, it was found that the correlation between the ESC of different head angles and the amount of differences of ESC from focused sound to diffused sound contribute the sense of "spatial homogeneity of envelopment."

4:40

2pAAa11. Influence of low-order room reflections on sound zone system performance. Marek Olik (CVSSP, Dept. of Electron. Eng., Ctr. for Vision Speech and Signal Process. (CVSSP), Univ. of Surrey, Guildford GU2 7XH, United Kingdom, m.olik@surrey.ac.uk), Philip Jackson, Philip Coleman (CVSSP, Dept. of Electron. Eng., Univ. of Surrey, Guildford, United Kingdom), Martin Olsen, Martin Møller, and Søren Bech (Bang & Olufsen, Struer, Denmark)

Studies on sound field control methods able to create independent listening zones in a single acoustic space have recently been undertaken due to the potential of such methods for various practical applications, such as individual audio streams in home entertainment. Existing solutions to the problem have shown to be effective in creating high and low sound energy regions under anechoic conditions. Although some case studies in a reflective environment can also be found, the capabilities of sound zoning methods in rooms have not been fully explored. In this paper, the influence of low-order (early) reflections on the performance of key sound zone techniques is examined. Analytic considerations for small-scale systems reveal strong dependence of performance on parameters such as source positioning with respect to zone locations and room surfaces, as well as the parameters of the receiver configuration. These dependencies are further investigated through numerical simulation to determine system configurations which maximize the performance in terms of acoustic contrast and array control effort. The design rules for source and receiver positioning are suggested, for improved performance under a given set of constraints such as a number of available sources, zone locations, and the direction of the dominant reflection.

TUESDAY AFTERNOON, 4 JUNE 2013 — 513DEF, 12:55 P.M. TO 5:20 P.M.

Session 2pAAb

Architectural Acoustics and Noise: Dah-You Maa—His Contributions and Life in Acoustics

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Jing Tian, Cochair
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Chair's Introduction—12:55

Invited Papers

1:00

2pAAb1. Dah-You Maa, friend and scholar. Leo L. Beranek (Westwood, MA) and Ning Xiang (Grad. Prog. in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY, Xiangn@rpi.edu)

Maa, Dah-You's life as a scholar and a close friend of many both in China and North America is the theme. A number of his scholarly achievements from his UCLA and Harvard time, to the post-Cultural Revolution period, up to recent years, is reviewed. His two years as a research fellow at UCLA and Harvard which ended with his receiving his Ph.D. at Harvard University in 1940 is detailed. Then began his life in Kunming during World War II where he was a professor in the E.E. Department of the National S.-W. Associated University. He returned to Beijing and became the first Dean of Engineering in the National Peking University (now Beijing University). During the Cultural Revolution, 1966–1976, he was under house arrest. His sponsorship of the first All-China Acoustics Conference after the Cultural Revolution in 1979 heralded his position as China's leading acoustician. Excerpts from his many letters through the years complete the presentation.

1:20

2pAAb2. Prof. Dah-You Maa's Contribution to Acoustics. Jing Tian (Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxilu, Beijing 100190, China, tian@mail.ioa.ac.cn)

In his life-long career pursuing, Professor Dah-You Maa contributed greatly in many areas of acoustics, not only in acoustical research and development, but also in the promotion of acoustical education, application, and legislation. Professor Maa presented a simplified method for calculation of normal modes for room acoustics, invented micro-perforated panel absorbers and micro-perforation jet mufflers, gave a formula of jet noise power via air pressure, and acoustically designed the first and biggest Congress Hall, built the first set of acoustical laboratories, and established the acoustical standard system in China. He also supervised tens of postgraduate students working in environmental acoustics, building acoustics, speech signal processing, nonlinear acoustics, and active noise control. In this paper, the main contribution of Professor Maa is introduced. Several of his typical research works, such as micro-perforated panel absorbers and micro-perforation jet mufflers, are explained in detail, from which we can well appreciate the physical insights and theoretical skills of Professor Maa.

1:40

2pAAb3. Dah-You Maa: Most senior academic brother. David T. Blackstock (Appl. Res. Labs. & Mech. Eng. Dept., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

F. V. Hunt's ONR-supported acoustics lab at Harvard turned out 30 PhD graduates after WW II. As one of those students in the late 1950s, I gradually became aware that Hunt had had a prewar group of graduate students as well. Leo Beranek was Hunt's first Ph.D., Dah-You Maa his second (both in 1940). Maa's two 1939 JASA articles on room acoustics, one coauthored with Beranek and Hunt, were benchmark papers of the day. I became fascinated with Maa's story, partly because he seemed so completely unreachable. He had returned to China during the war; afterward the Cold War intervened. I finally met my much older academic brother in 1980, at the 10th ICA in Sydney, Australia. Thus began a warm and rewarding relationship. In 1987, he began representing China on the International Commission on Acoustics, and for 7 years, we saw each other annually at Commission meetings. I learned of his work on nonlinear standing waves, a problem in which I too shared a keen interest. The high point was at the 14th ICA in Beijing. I was finally able to see his laboratory and meet his doctoral student Ke Liu. A memorable dinner followed that evening.

2:00

2pAAb4. Dah-You Maa and the many facets of modal density. Richard H. Lyon (MIT, 60 Prentiss Lane, Belmont, MA 02478-2021, rhlyon@lyoncorp.com)

Maa's formula for the modal density of a room is known to everyone in acoustics. The wonderful part of it is that it so logical and direct that you don't have to memorize or look it up - you can simply re-derive it on the spot. When it was published by Maa in 1939, there was a competing formulation by Dick Bolt that had the same volume dependent term, but was quite different in the surface area and edge length terms. But by the time of the landmark paper by Morse and Bolt in 1944 ["Sound waves in rooms", Rev. Mod. Phys. **16**(1) (1944)], Maa's formula had won and was the relationship cited by Morse and Bolt. The uses of modal density in the response statistics of acoustical spaces and structures for the purposes of estimating impedances, energy flow, and phase statistics have grown as recognition of this fundamental property of resonant systems has grown.

2:20

2pAAb5. Maa Dah-You and the design of reverberant rooms for determination of sound power. George C. Maling (60 High Head Rd., Harpswell, ME 04079, maling@alum.mit.edu)

Reverberation rooms are useful tools for the determination of sound power emitted by machines and other sound sources. One critical factor in the design of such rooms is the mode spacing at low frequencies. A seminal contribution to our understanding of mode spacing was made by Maa in 1939, and his paper was published in the Journal of the Acoustical Society of America. Richard Bolt also made a contribution at about the same time. In this paper, we begin with Maa's work and trace the development of mode spacing statistics through the work of Richard H. Bolt and Ludwig Sepmeyer. A computer study by the author of mode spacing statistics in 200 cubic meter rooms is described. This work led to recommendations for room dimensions to be included in an international standard on determination of sound power in reverberation rooms.

2:40

2pAAb6. Remembering one of Maa Dah-You's last essential contributions to acoustics. Jiqing Wang (Inst. of Acoust., Tongji Univ., 1239 Siping Rd., Shanghai 200092, China, wongtsu@126.com)

The book *The fundamentals of Modern Acoustic Theory* written by Maa Dah-You was published (in Chinese) in March 2004 at his age of 88. This 421 pages volume with 15 chapters covers almost all main acoustic fields from basics to many extended areas, such as linear to non-linear acoustics, common acoustical measures to active control, and the aerodynamic sound and thermo-acoustics. References listed in each chapter include many Maa's pioneering work of the related acoustic fields. This book systematically reflects his deeper understanding of acoustics theory, and further account new developments of acoustic technology. Moreover, the history of acoustics, particularly of the ancient China in the beginning sections of the first chapter, and the overview of the future developments in the afterword are also the exclusive items in this book. A well prepared exercise at the end of each chapter is most helpful for better understanding and thinking of the related topics. Therefore, it is widely used as a text or main reference book in many universities and research community in China.

3:00–3:20 Break

3:20

2pAAb7. Three months in Beijing—Micro-perforated absorbers, eigenvalues, and other topics. Christian Nocke and Catja Hilge (Akustikbuero Oldenburg, Katharinenstr. 10, Oldenburg 26121, Germany, nocke@akustikbuero-oldenburg.de)

In this contribution, a personal and scientific report on a three month exchange stay at the Institute of Acoustics in Beijing will be reported. Micro-perforation and low frequencies in small rooms have been the background of an investigation under supervision of Professor Daa-You Maa in 1999. Maa's ideas from the 1930s about the distribution on eigenmodes in rooms in combination with micro-perforated absorbers as reference absorbers had been the basis of this work. Abacus and computer calculations as the basis of old/new ideas formed the background of a stay as a student. Even after more than 12 years, micro-perforation is part of the daily work as consultant often dealing with new product ideas and developments. The presentation will give a very brief overview on this work that started in Beijing. Furthermore, in remembrance of a once in a life time experience personal experiences with Professor Daa-You Maa, his wife, and his colleagues will be reported.

3:40

2pAAb8. Surface waves over rigid-porous and rough surfaces. Keith Attenborough, Imran Bashir, and Shahram Taherzadeh (DDEM, The Open Univ., Walton Hall, Milton Keynes MK7 6AA, United Kingdom, k.attenborough@open.ac.uk)

Surface waves are created by near grazing sound propagation from a point source over either a rigid-porous layer or a (slightly) rough surface. In theory, they have similar origins. Frequency- and time-domain measurements have been made on surfaces composed from parallel periodically spaced rectangular strips (width 0.0126 m, height 0.0253 m) on an acoustically hard surface. The edge-to-edge spacing between the strips has been varied between 0.003 and 0.06 m. Frequency domain predictions show that when the spacing is substantially smaller than the strip height these surfaces may be regarded as locally reacting rigid-framed hard-backed porous layers with an effective depth slightly larger than the strip height. When the spacing is comparable to the strip height or greater the surfaces behave as periodically rough surfaces. Both frequency- and time-domain results show that surface waves of comparable magnitudes are created over the range of strip spacings studied but the main frequency content of these acoustically-induced surface waves is lowered as the mean spacing is increased. These data suggest that the surface waves have a similar physical origin and that they can be created also over a micro-perforated surface.

4:00

2pAAb9. Investigation of sound scattering by a surface using the acoustical wave propagator method. Jie Pan, Hongmei Sun, and James Leader (Mech. and Chem. Eng., The Univ. of Western Australia, 35 Stirling Highway, Crawly, WA 6009, Australia, jie.pan@uwa.edu.au)

The time domain acoustical wave propagator (AWP) method is used as a tool to study the sound scattering properties of a surface. As the method allows the modeling of the excitation, absorption, and scattering of sound in a vicinity of the surface within a very short period of time, the effect of other surfaces on these process can be ignored, and the acoustical property of a specific surface can be investigated in detail without solving the sound wave equation in the whole space. Sound scattering by a finite impedance strip mounted on a rigid baffle in two different conditions is solved by the method. The result is used to illustrate the efficiency of the method.

4:20

2pAAb10. A composite sound absorber with micro-perforated panel and shunted loudspeaker. Jiancheng Tao, Qijing Jiao, Xiaojun Qiu, and Ning Han (Inst. of Acoust., Nanjing Univ., Rm. 307, Acoust. West Bldg., No.22 Hankou Rd., Nanjing, Jiangsu, China, Nanjing, Jiangsu 210093, China, jctao@nju.edu.cn)

Micro-perforated panel (MPP) backed by a rigid cavity is a widely used clean sound absorber; however, its application at low frequency is limited because a deep cavity is required to achieve good sound absorption in the low frequency range. In the present paper, a composite absorber composed by an MPP and a shunted loudspeaker is proposed. The loudspeaker is installed at the back wall of the air cavity and the acoustic impedance can be optimized by adjusting the parameters of the loudspeaker and the electronic components in the shunt to improve the sound absorption performance. The prediction model of such a finite-sized composite absorber is established based on the mode analysis solution and the equivalent circuit of the loudspeaker. Both numerical simulations and experiments show that the thickness of the proposed composite absorber can be much smaller than that of traditional MPP constructions.

4:40

2pAAb11. Active noise control in rooms. Jiri Tichy (Penn State Univ., 5552 N Citation Rd., Toledo, OH 43615, tichy@engr.psu.edu)

Maa published his ideas and achievements in the early days of the new branch of noise control based on using secondary sources to reduce the sound level of the primary sources by negative interference. He concentrated on active control of the reverberant sound in a room by a loudspeaker placed in one of the room corners and a microphone in its close vicinity. He examined the role of the complete solution of the wave equation consisting of a direct wave radiated from the sources and the reverberant field created by room modes. His work resulted in deriving a general formula for possible noise reduction that is independent of the shape, size, and the content of the room. He has shown noise reduction of 8 dB for a noise band centered at 100 Hz.

5:00

2pAAb12. A hybrid modal analysis for enclosed sound fields and its applications. Buye Xu (Signal Process. Res., Starkey Hearing Technol., 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com), Scott D. Sommerfeldt, and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In 1939, Dr. Dah-You Maa presented a paper entitled “The distribution of eigentones in a rectangular chamber at lower frequency ranges” [J. Acoust. Soc. Am. **10**, 258 (1939)]. Since then, his interest in room acoustics has not diminished. More than six decades later, Maa proposed an idea of adding a monopole solution to the normal modal expansion to improve the accuracy in simulating the near-field sound field in rooms [D. Y. Maa, Acta Acust. (Beijing) **27**, 385–388 (2002)]. Following this idea, a hybrid model that combines the free field Green’s function and a modal expansion has been proposed by the authors based on a rigorous mathematical derivation [Xu *et al.*, J. Acoust. Soc. Am. **128**, 2857–2867 (2010)]. The hybrid modal expansion can be further extended for complex sound sources by introducing the multipole expansion to the solution. In this talk, the theoretical derivation of the hybrid modal expansion will be reviewed, followed by examples demonstrating the use of the hybrid modal expansion in real-world applications.

Session 2pAB

Animal Bioacoustics, Psychological and Physiological Acoustics, Signal Processing in Acoustics, and Noise: Listening in the Natural Environment

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Peter M. Narins, Cochair
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Chair's Introduction—12:55

Invited Papers

1:00

2pAB1. Three directions in research on auditory scene analysis. Albert S. Bregman (Psychology, McGill Univ., 1205 Doctor Penfield Ave., Montreal, QC H3A 1B1, Canada, al.bregman@mcgill.ca)

Research on auditory scene analysis (ASA) began with some simple laboratory phenomena such as streaming and illusory continuity. Subsequently, research has gone in three directions, downwards toward underlying mechanisms (by neurophysiologists), upwards toward system organization (by computer scientists), and sideways toward other species (by neurobiologists). Each direction has its problems. The downward approach sometimes takes a phenomenon-oriented view of ASA, leading to simple explanations of a single ASA demonstration, such as streaming, with no obvious connection to any larger system. Research done by the upward approach usually takes the form of a computer program to achieve ASA in a working system, often ignoring known facts about human ASA, in favor of mathematically understood principles. The sideways approach often finds that non-human animals can respond to an important sound despite the presence of other interfering sounds. However, there is no reason to believe that a frog, a fish, and a human accomplish this by means of the same mechanisms. So finding out how some animal does this, while interesting in its own right, may shed little light on how humans do it. I will describe some properties of the human ASA system that should be borne in mind when manufacturing explanations.

1:20

2pAB2. Mechanisms of perceiving communication sounds in scenes. Sarah M. Woolley (Psychology, Columbia Univ., 406 Schermerhorn Hall, 1190 Amsterdam Ave., New York, NY 10027, sw2277@columbia.edu)

Vocal communicators must perceive the vocal signals of social partners in complex auditory scenes that include distracting background sounds. The auditory system must therefore parse auditory scenes into multiple information streams and/or accurately encode individual vocalizations despite the presence of competing sounds. Explaining mechanisms whereby neural representations of vocalizations are extracted from neural representations of scenes is an important part of understanding how auditory processing leads to perception of communication signals in complex scenes. We study how songbirds recognize individual vocalizations (songs) in scenes of conspecific choruses. We combine behavioral studies with neurophysiological studies of song and scene coding in midbrain and cortex. We find dramatic transformations in the neural coding of songs and scenes between different regions of auditory cortex. Neural representations of individual songs are dense and non-selective in the midbrain and primary cortex, but are sparse and highly selective in higher cortex. Sparse coding neurons produce background-invariant responses to individual songs in scenes, providing a potential neural mechanism for the perception of individual communication vocalizations in complex auditory scenes. Acoustic manipulations of song and pharmacological manipulations of neural coding suggest that sparse and background-invariant representations of songs in higher cortex are due to context-dependent inhibition.

1:40

2pAB3. The search for a neural basis of communication: Learning, memory, perception. and performance of vocal signals. Jonathan Prather (Dept. of Zoology and Physiol., Univ. of Wyoming, 1000 E Univ Ave. - Dept. 3166, Laramie, WY 82071, jprathe2@uwyo.edu)

Brain mechanisms for communication must establish a correspondence between sensory perception and motor performance of individual signals. A class of neurons in the swamp sparrow forebrain is well suited for that task. Recordings from awake and freely behaving birds reveal that those cells express categorical auditory responses to changes in note duration, a learned feature of their songs, and the neural response boundary accurately predicts the categorical perceptual boundary measured in field studies. Extremely precise auditory activity of those cells represents not only songs in the adult repertoire but also songs of others and tutor songs, including those imitated only very few times or perhaps not at all during development. Furthermore, recordings during singing reveal that these cells also express a temporally precise auditory-vocal correspondence, and limits on auditory responses to extremely challenging tutor songs may contribute to the emergence of a novel form of song syntax. Therefore, these forebrain neurons provide a mechanism through which sensory perception may influence motor performance to enable imitation. These cells constitute the projection from a premotor cortical-like area into the avian striatum (HVCX neurons), and data from humans implicate analogous or homologous areas in perception and performance of the sounds used in speech.

Contributed Paper

2:00

2pAB4. Call perception in mice. Erikson Neilans, David Holfoth, Kelly E. Radziwon (SUNY, Univ. at Buffalo, 207 Park Hall, Buffalo, NY 14260, eneilans@buffalo.edu), Christine V. Portfors (School of Biological Sci., Washington State Univ., Vancouver, WA), and Micheal L. Dent (SUNY, Univ. at Buffalo, Buffalo, NY)

Acoustic communication in laboratory mice is a relatively recent subject of experimental study, often yielding disparate findings. For example, researchers often manually place mouse ultrasonic vocalizations (USVs) into categories based on spectrotemporal characteristics, but the numbers and types of categories differ widely between laboratories. Here, we attempt to determine what cues CBA/CaJ mice use to discriminate between vocalizations by testing them in an operant conditioning paradigm. The mice were trained to discriminate a repeating background containing one USV from several target USVs. The targets were different call types used by Holmstrom *et al*. (2010) and manipulations of the background calls, such as removing the frequency modulation, shifting the entire call up or down in frequency, shortening or lengthening the call, or reversing the entire call. Results show that large frequency shifts were easy for the mice to discriminate, while reversing the calls and removing the frequency modulation were much more difficult. For most calls, similarity in spectrotemporal characteristics yielded poor discrimination performance. These results are the first to show that mice can discriminate between some vocalizations but not others, and that they may place different meaning to different call types, though not necessarily the call types designated by humans.

2:20–2:40 Break

Invited Papers

2:40

2pAB5. Auditory processing for contrast enhancement of salient communication vocalizations. Alex G. Dunlap, Frank G. Lin (Biomed. Eng., Georgia Inst. of Technol. and Emory Univ., Atlanta, GA), and Robert C. Liu (Biology, Emory Univ., 1510 Clifton Rd. NE, Rm. 2006, Atlanta, GA 30322, robert.liu@emory.edu)

In a natural acoustic environment, coherent representations of auditory objects and sources are streamed from the myriad sounds that enter our ears. Features of those sounds that are familiar and behaviorally salient to us are detected and discriminated into invariant precepts that inform us about our external world. Research into how this occurs is increasingly converging on the idea that there is a transformation from the auditory periphery wherein an initial acoustically faithful representation by neurons becomes progressively altered to enhance the population neural representation of perceptually relevant aspects of the sound. How this occurs may vary for sounds whose meanings are acquired in different ways, perhaps depending on what actions and decisions must be executed upon recognition. We have investigated this process in a natural social context in which mouse mothers “learn” about the meaning of pup ultrasound vocalizations through their maternal care. Here we discuss our recent studies in awake mice using electrophysiological, behavioral, immunohistochemical, and computational methods. Our results suggest that experience with natural vocalizations may alter core auditory cortical neural responses so that the contrast in activity across the neural population enhances the detection and discrimination of salient calls.

3:00

2pAB6. Influences of perceptual continuity on everyday listening. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu), Golbarg Mehraei (Speech and Hearing Biosci. and Technol., Harvard/MIT, Cambridge, MA), Scott Bressler, and Salwa Masud (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

In the natural environment, listeners face the challenge of parsing the sound mixture reaching their ears into individual sources, and maintaining attention on a source of interest through time long enough to extract meaning. A number of studies have shown that continuity of certain acoustic features (including pitch, location, timbre, etc.) allows the brain to group sound from one acoustic sound source together through time to form an auditory object or stream. This presentation reviews results demonstrating that auditory feature continuity has important consequences on how listeners maintain attention on a stream through time. For instance, continuity of a sound feature that a listener knows is irrelevant to the task at hand nonetheless impacts the ability to maintain auditory attention based on some other sound feature. Moreover, the influence of auditory feature continuity decreases as the time between events in a given sound stream increases. Taken together, these behavioral results support the idea that auditory attention operates on auditory objects, rather than on individual sound features, and that feature continuity has an obligatory influence on the formation of auditory streams, and therefore on how selective auditory attention allows us to communicate in everyday settings.

3:20

2pAB7. Bugs and bats: Neural analysis of behaviorally relevant sounds in crickets. Gerald Pollack (Dept. of Biol., McGill Univ., 1205 Dr. Penfield Ave., Montreal, QC H3A1B1, Canada, gerald.pollack@mcgill.ca)

Hearing in crickets is specialized to serve particular behavioral functions, namely intraspecific communication and predator avoidance. Male crickets produce species-specific acoustic signals (songs) that attract distant females, promote copulation, and contribute to agonistic interactions with rivals. Crickets also hear the echolocation calls of aerially hunting bats, which evoke avoidance responses. These clear behavioral functions of hearing, combined with the relative simplicity of the cricket’s nervous system, make it possible to address questions about how behaviorally relevant sensory signals are analyzed at the level of single, uniquely identifiable nerve cells. Cricket songs and bat calls differ both in rhythm and in spectrum, and neurons throughout the auditory processing chain are specialized for processing these two sorts of signal. I will focus on specializations that are evident at early stages of auditory processing, i.e., primary sensory neurons and the first-order interneurons with which they interact.

Contributed Paper

3:40

2pAB8. New observations and modeling of an unusual spatiotemporal pattern of fish chorusing off the southern California coast. Gerald L. D'Spain, Heidi H. Batchelor, and Tyler A. Helble (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

The purpose of this paper is to present new results on an unusual spatiotemporal pattern of fish chorusing off the southern California coast. Characteristics of this fish chorus have been reported previously; it occurs at night in the late spring and summer months in shallow, sandy bottom regions just outside the surf zone. The background sound levels increase by up to 30 dB and cycle in level with a period of 30–35 s all night long. In this paper, recent results from measurements made by a set of high spatial resolution sensor systems spanning a 50-km stretch of coastline out to 20 km offshore over a 2-month time period are presented. These data allow the spatial dependence and long-term temporal variability of the chorus to be examined at high spatial resolution. Refinements to a numerical model that predicts this chorusing behavior are required to account for some aspects of these new observations. [Work supported by the Office of Naval Research, Code 322-MMB.]

Invited Papers

4:00

2pAB9. The influence of anthropogenic noise on the evolution of communication systems. Peter M. Narins (Integrative Biol. & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095, pnarins@ucla.edu)

Many species of animals, including man, face the formidable task of communicating in noisy environments. In this talk, I shall discuss the effects of anthropogenic (man-made) noise on the calling behavior of anuran amphibians. Moreover, the role of spectral, temporal, and spatial separation in minimizing masking by background noise will be examined. For example, presenting high-level, periodic (or aperiodic) tones at the males co-note frequency to males of the Puerto Rican treefrog, *Eleutherodactylus coqui* results in a clear shift in their calling pattern in an attempt to minimize acoustic overlap with the interfering playback stimulus. Amphibians also have a remarkable ability to shift their call timing in response to small intensity shifts in the background noise. Males of *E. coqui* are capable of reliably detecting a change in interfering tone intensity as small as 2–4 dB. Finally, I shall present behavioral evidence that anthropogenic noise may act as a strong selective force in sculpting the acoustic communication systems of several species of Old World frogs. Some techniques for visualizing sound interference will be discussed. [Supported by grants from the NIDCD (Grant No. DC-00222), and the UCLA Academic Senate (3501).]

4:20

2pAB10. Active listening in a complex environment. Melville Wohlgemuth and Cynthia F. Moss (Psych. and ISR, Univ. of Maryland, Biol.-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

Spatially guided behaviors in echolocating bats depend upon the dynamic interplay between auditory information processing and adaptive motor control. The bat produces ultrasonic signals and uses information contained in the returning echoes to determine the direction and distance of objects in space. With this acoustic information, the echolocating bat builds a 3-D auditory representation of the world, which it uses to guide a suite of coordinated motor behaviors, including head and pinna movements, as well as the timing, duration, frequency characteristics, and directionality of sonar signals. Adaptive echolocation behaviors shape the acoustic information available to the bat's sonar imaging system and provide a window to its perception of complex scenes. In a complex environment, an echolocating bat encounters multiple reflecting surfaces that return a cascade of echoes from each sonar transmission. The work presented here will focus on adaptive echolocation behaviors of the big brown bat as it tracks a selected prey item in the presence of multiple objects, both obstacles and other prey. Data suggest that bats can successfully segregate streams of echoes from closely spaced objects through finely tuned adaptive sonar signal control.

Contributed Papers

4:40

2pAB11. Your attention, please! Determining saliency of competing audio stimuli in natural scenarios. Francesco Tordini (Dept. of Elec. and Comput. Eng., McGill, 3480 University St., McConnell Eng. Bldg., Montreal, QC H3A 2A7, Canada, tord@cim.mcgill.ca), Albert S. Bregman (Dept. of Psych., McGill, Montreal, QC, Canada), Anupriya Ankolekar, Thomas Sandholm (Hewlett-Packard Lab., Palo Alto, CA), and Jeremy R. Cooperstock (Dept. of Elec. and Comput. Eng., McGill, Montreal, QC, Canada)

Perceptual saliency is a precursor to bottom-up attention modeling. While visual saliency models are approaching maturity, auditory models remain in their infancy. This is mainly due to the lack of robust methods to gather basic data, and oversimplifications such as an assumption of monaural signals. Here we present the rationale and initial results of a newly designed experimental paradigm, testing for auditory saliency of natural sounds in a binaural listening scenario. Our main goal is to explore the idea that the saliency of a sound depends on its relation to background sounds by using more than one sound at a time, presented against different backgrounds. An analysis of the relevant, emerging acoustical correlates together with other descriptors is performed. A review of current auditory saliency models and the deficiencies of conventional testing approaches are provided. These motivate the development of our experimental test bed and more formalized stimulus selection criteria to support more versatile and ecologically relevant saliency models. Applications for auditory scene analysis and sound synthesis are briefly discussed. Some initial conclusions are drawn about the definition of an expanded feature set to be used for auditory saliency modeling and prediction in the context of natural, everyday sounds.

5:00

2pAB12. A neuroethological analysis of the information in propagated communication calls. Frederic E. Theunissen, Solveig Mouterde (Psychology, UC Berkeley, UC Berkeley, 3210 Tolman Hall, Berkeley, CA 94720, theunissen@berkeley.edu), and Nicolas Mathevon (Univ. of Lyon/Saint-Etienne, Saint-Etienne, France)

The detection and recognition of communication signals in natural soundscapes is a difficult task that animals and birds in particular excel at. We have used a neuroethological approach to quantify the recognition

performance for propagated communication signals in the zebra finch, specifically regarding the information about individual identity. The propagated signals were analyzed using a regularized discriminant function analyses on a complete spectrographic representation of the signals. We found (1) a reduction in the informative frequency range a long distances yielding a frequency band sweet-spot, (2) that call duration and pitch are important parameters at short distances, and (3) that frequency modulation gains are important parameters at longer distances. Operant conditioning experiments showed that female songbirds were able to discriminate male calls at up to 128 m but not at 256 m. Finally, neurophysiological recordings showed a similar pattern in that high neural discrimination for calls was observed at 16 m and that this information degraded as a function of distance. We are currently analyzing the tuning properties of neurons that showed the most invariant responses to propagated sounds and hypothesized that these will be tuned to the parameters that we found were the most informative in the discriminant function analysis.

TUESDAY AFTERNOON, 4 JUNE 2013 519A, 1:00 P.M. TO 5:20 P.M.

Session 2pBAa

Biomedical Acoustics, Physical Acoustics, and Acoustical Oceanography: Bubbles Bubbles Everywhere II

Ronald A. Roy, Cochair
Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215

Thomas J. Matula, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., WA 98105-6698

Contributed Papers

1:00

2pBAa1. Spatially and temporally resolved single bubble sonoluminescence and its entrainment in Rayleigh-Taylor jets. Jonathan R. Sukovich, Phillip A. Anderson (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jonsukovich@gmail.com), Ashwinkumar Sampathkumar (Frederic L. Lizzi Center for Biomed. Engineering, Riverside Research, New York, NY), and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

Previous investigations of the temporal and spatial evolution of single bubble sonoluminescence (SBSL) have shown events to last on the order of tens to hundreds of picoseconds with spatial extents of less than 1 μm. Here we present observations of the temporal and spatial evolution of laser-nucleated SBSL events in a high-pressure spherical resonator. Using high-speed imaging, we observe large, long-lived SBSL events reaching diameters of up to 50 μm and lasting on the order of 30 ns. Observations of events entrained in Rayleigh-Taylor jets resulting from instabilities in the final stages of the bubbles collapses will also be presented. We observe the light emitting region entrained in these jets to reach velocities well in excess of Mach 1 and to travel up to 100 μm before being extinguished. The size and duration of events, and the velocity of those entrained in Rayleigh-Taylor jets, will be compared to the maximum radius and collapse velocity of the bubbles responsible for generating them to develop a better understanding of the dynamics leading to, and the mechanisms responsible for light emissions during highly energetic collapse events. [Work supported by Impulse Devices, Inc.]

1:20

2pBAa2. High pressure phase transitions in the fluid region surrounding the collapse point of large single bubbles in water. Jonathan R. Sukovich, Phillip Anderson (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jonsukovich@gmail.com), Ashwinkumar Sampathkumar (Frederic L. Lizzi Ctr. for Biomed. Eng., Riverside Res., New York, NY), and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

Observations from imaging experiments will be presented, which have shown persistent, long-lived spherical objects to form in the fluid region surrounding large, single bubbles in highly over-pressured water. Objects have been observed to form in a region of fluid where pressures are first predicted to exceed 0.8 GPa, and to extend radially inward to where fluid pressures are predicted to reach 6 GPa. These pressures bound those requisite for transitions in water to the crystalline phases of Ice-VI and Ice-VII, at 1.1 GPa and 2.1 GPa, respectively. The objects have been observed to behave in a fashion more consistent with a highly viscous fluid. They support and recover from large shape deformations, as well as support fluid flows within them. While water does have phases which are known to exhibit properties of highly viscous fluids, they have only been observed to form at or near cryogenic temperatures, typically via hyperquenching or quasi-static pressurization at low temperatures. Here, we present evidence for a high pressure liquid-liquid phase transition in water surrounding collapsing bubbles at room temperature. [Work supported by Impulse Devices, Inc.]

1:40

2pBAa3. Experimental characterization of light emission during shock-driven cavity collapse. Phillip Anderson, Nicholas Hawker, Matthew Betney (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, phillip.anderson@eng.ox.ac.uk), Brett Tully (Oxyntix Ltd., Oxford, United Kingdom), Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Dept. of Mech. Eng., Boston Univ., Boston, MA)

The authors describe experimental work examining the collapse of a cavity by a strong shockwave. A millimeter size cavity is cast in Phytagel, which is then impacted by a metallic projectile accelerated by a compressed gas gun, reaching velocities up to 500 m/s. The impact generates a strong shockwave that propagates into the gel at greater than sonic velocity. Schlieren images are presented that illustrate both this process and the subsequent cavity collapse at a sub-microsecond timescale. As the shockwave reaches the cavity, it is shown to cause a rapid asymmetric collapse process characterized by the formation of a high-speed transverse jet. The pressure of the shockwave is found to be 100+ MPa as measured via a custom-built fiber-optic probe hydrophone. Previous work examining shock-driven cavity collapse observed luminescence, postulated to be due to the high-speed impact of the transverse jet on the far bubble wall; this experimental observation is replicated. Further, the light emission is characterized as a function of impact velocity and thus of shockwave pressure. This reveals that shock-driven cavity collapse shares many of the unique features that make the more widely studied SBSL-type collapse interesting.

2:00

2pBAa4. Simulation of warm dense matter in intense bubble collapse. Brett Tully (Oxyntix Ltd., Dept. of Eng. Sci., Parks Rd., Oxford OX1 3PJ, United Kingdom, brett.tully@oxyntix.com), Nicholas Hawker, Matthew Betney, and Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Previous work by the authors includes computational study of shock-bubble interaction. This work demonstrated strong compression and heating within the bubble, with gas reaching densities of order of magnitude 1 g/cc and temperatures of 10 eV. These conditions correspond to the warm dense matter regime. This paper addresses limitations of previous work through utilization of various equations of state (EOS) appropriate for the modeling of dense plasma. This is achieved through the design and implementation of a generic interface based on tabulated EOS data. Any EOS may be utilized through the framework, requiring only knowledge of pressure and energy as functions of density and temperature. The solutions to various issues such as table interpolation, tabulated change of variables, arbitrary calculation of entropy, and calculation of thermodynamic derivatives are presented. In addition, the trade-offs between CPU time, memory requirement and computational accuracy are discussed. Validation work is presented and a comparison of different EOS is also explored. The EOS used include but are not limited to the EOS for air utilized by Moss *et al*. (1994) to study SBSL, SESAME tabulated EOS and QEOS-type formulations. Finally, conditions attained during shock-bubble interaction are re-examined.

2:20

2pBAa5. Percussoluminescence. Nicholas Hawker and Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, nicholas.hawker@eng.ox.ac.uk)

The phenomenon of light emission from a bubble driven to collapse via ultrasound has a long history of study. Light emission is widely believed to occur due to the formation of plasma within the bubble during the final moment of collapse and one important focus of the literature has been on understanding the exact thermodynamic conditions created. There are, however, other phenomena that demonstrate a similar light emission, including but not limited to other types of bubble collapse. This paper examines light emission in seeming disparate phenomena and discovers that a common thread exists. The terminology and fundamental theory appropriate for sonoluminescence is found to be decreasingly relevant and a new term, percussoluminescence (PCL), is established. This is substantiated through the presentation of a 1D theory, detailed numerical simulation, and illustrative experimental results. The implications of this new perspective are explored.

2:40–3:00 Break

3:00

2pBAa6. Investigating the acoustic response of gold nanoparticle coated microbubbles. Mehrdad Azmin (UCL Mech. Eng., Univ. College London, London, United Kingdom, m.azmin@ucl.ac.uk), Paul Rademeyer, Graciela Mohamedi (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Mohan Edirisinghe (Mech. Eng., Univ. College London, London, United Kingdom), Luis Liz-Marzan (Biofunctional Nanomaterials, CIC biomaGUNE, San Sebastian, Spain), and Eleanor Stride (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Recent work has shown that incorporating solid nanoparticles into the coatings of contrast agent microbubbles can be used to control their stability and to provide other functional characteristics for example in multimodality imaging. The aim of this study was to investigate the influence of nanoparticle characteristics and concentration on the response of the microbubbles to ultrasound excitation. Theoretical models were first derived to simulate the effects of adsorbing different types and concentrations of nanoparticle on to the surface of a bubble in both a monolayer and a layer of finite thickness. The results indicate that the particles modify the symmetry of the microbubble oscillations and enhance their nonlinear character. Experimentally, microbubbles coated with a surfactant and varying concentrations of gold nanoparticles of different sizes and surface properties were produced using either sonication or microfluidics. The attenuation and backscattering coefficients from the bubble suspensions and the scattered response from individual bubbles were measured for a range of frequencies (1–7.5 MHz) and pressures (50–500 kPa). The nanoparticles were found to enhance the nonlinear character of the bubble response in agreement with the theoretical results. Both the degree of enhancement and stability of the microbubbles was dependent upon the nanoparticle surface chemistry.

3:20

2pBAa7. Investigating the sensitivity of microbubble acoustic response for biosensing applications. Caroline J. Harfield (Inst. of Biomed. Eng., Dept. of Eng. Sci., Oxford Univ., Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, caroline.harfield@stcatz.ox.ac.uk), Gianluca Memoli (Acoust. Group, National Physical Lab., London, United Kingdom), Philip Jones (Dept. of Phys. and Astronomy, Univ. College London, London, United Kingdom), Nick Ovenden (Dept. of Mathematics, Univ. College London, London, United Kingdom), and Eleanor Stride (Inst. of Biomed. Eng., Dept. of Eng. Sci., Oxford Univ., Oxford, United Kingdom)

Microbubbles are currently used as contrast agents for diagnostic imaging on account of their high scattering efficiency and non-linear response to ultrasound. The exact nature of this response depends not only upon the bubble size and imposed sound field but also the bubble environment: physical properties of the surrounding liquid, bubble surface coating, ambient temperature, pressure, and proximity to other bubbles or surfaces. This dependence can potentially be exploited for the microscale interrogation of a liquid to detect, e.g., changes in viscosity or the presence of particular chemical species. To facilitate this, the sensitivity of the microbubble acoustic response to changes in its environment must be analyzed. The aim of this study was to provide a theoretical framework for this. A modified Rayleigh-Plesset equation was derived to describe the radial bubble motion, including the effects of gas diffusion and adsorption/desorption of a surfactant coating, and coupled to an equation describing microbubble translation. The presence of a rigid boundary was also included in the simulations. A sensitivity analysis was performed for the effect of each of the physical variables upon the bubble response, which indicated high sensitivity to species altering the dynamic surface tension and proximity to a boundary.

3:40

2pBAa8. Modeling of microbubbles pushed through clots via acoustic radiation force. Ascanio Guarini and E. C. Everbach (Engineering, Swarthmore College, 500 College Ave., Swarthmore, PA 19081, ceverba1@swarthmore.edu)

Previous studies have shown that thrombi, which may completely block the blood flow in a vessel, can be dissolved by ultrasound acting on echo-contrast agent microbubbles. The presumed mechanism is acoustic cavitation, the radial oscillations of the bubbles, which can exert locally large forces on the fibrin ropes that make up the clot matrix. However, the movement of the bubbles through the clot in the absence of flow suggests that acoustic radiation force also plays an important role. Because detailed mechanistic modeling of this process is not available, we present here a heuristic study in which microbubble transit times in gels of various porosities were measured and described by a simplified percolation theory. Results suggest considerations for optimizing the penetration of active microbubbles in sonothrombolysis.

4:00

2pBAa9. Use of dolphin-like pulses to enhance target discrimination and reduce clutter. Tim Leighton, Gim-Hwa Chua, Paul R. White (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk), Kenneth Tong, Hugh Griffiths (Dept. of Electron. and Elec. Eng., Univ. College London, London, United Kingdom), and David Daniels (Cobham Tech. Services, Leatherhead, United Kingdom)

Building on the earlier success of using twin inverted pulses to supress bubble scatter and so reduce clutter when detecting targets in bubbly water (proven in ship wakes in field trials), the same nonlinear processing scheme is generalized to make use of dolphin-like pulses. Their performance in reducing clutter and enhancing target discrimination is demonstrated in the laboratory, and the opportunities for using the same scheme to improve the detection of hidden electronic devices or semiconductor devices by radar are discussed.

4:20

2pBAa10. Acoustic and optical observations of methane gas seeps in the Gulf of Mexico. Thomas C. Weber, Yuri Rzhanov, Kevin Jerram, Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, weber@ccom.unh.edu), and Dave Lovalvo (Eastern Oceanics/NOAA Office of Ocean Exploration, West Redding, New Hampshire)

In 2011 and 2012, measurements of acoustic backscatter from natural methane seeps were made in the northern Gulf of Mexico in water depths between 1000 and 2000 m. The measurements were made using a calibrated 18 kHz echo sounder with an 11 degree beamwidth in order to estimate the depth-dependent target strength (TS). The TS data indicate a wide variation in the rate of gas seepage from the seafloor. Several of these seeps were revisited with a remotely operated vehicle in order to optically assess the bubble size distribution and to estimate the rate at which gas bubbles were exiting the seafloor. The optical data show bubble sizes between 1 and 10 mm radius, and similar rates of gas seepage ranging from a few bubbles per second to several tens of bubbles per second. Together, these data help to suggest the requirements for acoustically estimating gas flux from the seafloor over large regions.

4:40

2pBAa11. Effect of shell thickness on sound propagation through encapsulated bubbles: A resonator approach. Craig N. Dolder and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

The acoustics of elastic-shelled bubbles are of interest in applications ranging from biomedical imaging, to fisheries acoustics, to underwater noise abatement. Multiple models exist to predict the velocity and attenuation of sound propagating through water containing encapsulated bubbles, but existing measurements have yet been able to confidently verify model accuracy for void fractions above about 10^{-3}. The effect of shell thickness was studied in this work using tethered, rubber-encapsulated bubbles bearing shells of varying thickness, deployed within a one-dimensional resonator apparatus. In previous work, resonator modes below the individual bubble resonance frequency (IBRF) were exploited to extract inferences of sound speed, but this is a regime where there is little difference between competing model prediction. In the present work, an increased understanding of the modal field inside the resonator has extended this technique to just below IBRF and to well above IBRF, both regimes where model behavior diversifies, thus providing a new opportunity for model verification. Measurement-model comparisons will be shown for encapsulated bubbles with radii ranging from 13 mm to 30 mm, void fractions ranging from 8.4×10^{-4} to 1.1×10^{-2}, and shell thicknesses ranging from 0.085 to 0.16 mm. [Work supported by the Office of Naval Research.]

5:00

2pBAa12. Attenuation of sound in water through collections of very large bubbles with elastic shells. Kevin M. Lee (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu) and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

The ultimate goal of this work is to accurately predict the attenuation through a collection of large (on the order of 10 cm radius) tethered encapsulated bubbles used in an underwater noise abatement system. Measurements of underwater sound attenuation were performed during a set of lake experiments, where a low-frequency compact electromechanical sound source was surrounded by different arrays of encapsulated bubbles with various individual bubbles sizes and void fractions. The measurements are compared with an existing predictive model [J. Acoust. Soc. Am. **97**, 1510–1521 (1995)] of the dispersion relation for linear propagation in liquid containing encapsulated bubbles. Although the model was originally intended to describe ultrasound contrast agents, it is evaluated here for large bubbles, and hence low frequencies, as a design tool for the underwater noise abatement system, and there is fairly good quantitative agreement between the data and the model. Refinements to the model to incorporate multiple scattering effects, which may be important at high void fractions, via an effective medium approach [J. Acoust. Soc. Am. **111**, 168–173 (2002)] and comparison with the data will also be discussed. [Work supported by Shell Global Solutions.]

TUESDAY AFTERNOON, 4 JUNE 2013 518C, 1:20 P.M. TO 3:20 P.M.

Session 2pBAb

Biomedical Acoustics: Imaging and Characterization in Elastic Media

Guy Cloutier, Chair

Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada

Contributed Papers

1:20

2pBAb1. Dynamic quantitative ultrasound imaging of mimicked breast lesions during shear wave propagation to emphasize differences in tissue statistical backscatter properties. Marzieh Alavi (Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, Univ. of Montreal Hospital Res. Ctr., Montreal, QC, Canada, marziehalavi@gmail.com), Francois Destrempes, Emmanuel Montagnon, and Guy Cloutier (Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, Montreal, QC, Canada)

The main motivation in this study was to increase the accuracy of the breast tissue characterization by combining quantitative ultrasound (QUS) with ultrasound (US) dynamic elastography. To demonstrate that, an agar-gelatin breast mimicking phantom with two inclusions containing the same density of agar (US scatterers) but different proportions of gelatin corresponding to different mechanical properties was made. Transient plane shear waves (SW) at 200 Hz were transmitted through the phantom while the displacement of scatterers was imaged at 5 MHz with an ultrafast imaging technique. With segmented inclusions, the reciprocal (beta parameter) of the effective density of scatterers of a general distribution model of the echo envelope and its normalized range (normalized by the mean of beta during SW propagation) were estimated for each inclusion. The results showed that the relative difference of beta between the surrounding medium and both inclusions A and B were 55.6% (A) and 0.9% (B), respectively, whereas differences (in %) of the beta normalized range were 46.2% (A) and 52.6% (B), respectively. The static value beta failed to distinguish inclusion B from the surrounding; however, the dynamic range of beta succeeded in that task for the two inclusions. Thus, dynamic QUS might add information to QUS in a static framework.

1:40

2pBAb2. Shear wave elastography for characterizing muscle tissue in myofascial pain syndrome. Diego Turo, Paul Otto (Dept. of Bioengineering, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, diego-turo@gmail.com), Tadesse Gebreab (National Inst. of Health, Bethesda, MD), Katherine Armstrong, Lynn H. Gerber (Dept. of Rehabilitation Sci., George Mason Univ., Fairfax, VA), and Siddhartha Sikdar (Dept. of Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Myofascial pain syndrome (MPS) affects 85% of chronic pain sufferers in a specialty pain center. Neck and low-back are commonly affected by MPS. Myofascial trigger points (MTrPs) are characteristic findings of MPS and are palpable tender nodules in the muscles of symptomatic subjects. Mechanical characterization of MTrPs and surrounding tissue can offer important insight about the pathophysiology of the MPS, which is currently poorly understood. In this study, we propose an inexpensive technique, based on ultrasound shear wave elastography, to objectively measure mechanical properties of MTrPs and surrounding tissue in the upper trapezius. In an ongoing clinical study, we recruited 34 subjects: 12 healthy controls, 10 with not spontaneously painful MTrPs (latent) and 12 with symptomatic chronic neck pain (>3 months) and active (spontaneously painful) MTrPs. Shear wave elastography was performed on the upper trapezius of all subjects using the Ultrasonix RP system and an external vibrator. Voigt' s model was used to estimate shear modulus G and viscosity μ of the interrogated tissue. Preliminary analysis demonstrates that symptomatic muscle tissue in subjects with neck pain is stiffer ($G = 8.40 \pm 8.31$ kPa, mean $\pm$ standard deviation) compared to muscle in control subjects ($G = 2.86 \pm 2.48$ kPa) ($p < 0.05$), and that active MTrPs are more viscous ($\mu = 17.10 \pm 9.46$ Pa*s) than surrounding tissue ($\mu = 10.59 \pm 5.96$ Pa*s). Latent MTrPs ($\mu = 24.31 \pm 12.72$ Pa*s) and surrounding tissue ($\mu = 20.09 \pm 7.48$ Pa*s) are more viscous than normal tissue ($\mu = 10.96 \pm 4.17$ Pa*s).

2:00

2pBAb3. Electromagnetic-acoustic imaging of stiffness and dielectric properties in gels. Ning Zhang, Robin Cleveland, and David J. Edwards (Eng. Sci., Univ. of Oxford, Inst. Biomed. Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, robin.cleveland@eng.ox.ac.uk)

Here we present a novel multi-modal imaging method, electromagnetic-acoustics (EMA) that combines acoustic radiation force and electromagnetic scattering. Experiments were carried out in gels using a focused megahertz ultrasound source, amplitude modulated at 160 Hz, to create an oscillating radiation force resulting in vibration of the gel in the focal region. At the same time an EM signal was transmitted into the gel and the backscattered EM signal recorded. The tissue that was in motion resulted in a Doppler shift of the EM signal which manifested itself as frequency modulation of the EM wave. The modulated component was detected by means of a demodulator and lock-in amplifier and the amplitude of the modulated signal we call the EMA signal. By steering the ultrasound beam through the sample, an EMA image can be created which has spatial resolution of the ultrasound but is sensitive to shear and dielectric properties. We show that the EMA signal is sensitive to changes in elasticity and conductivity in a gel. EMA may have utility in biomedical imaging by detecting diseases, which have contrast in dielectric properties without the cost and complexity of an magnetic resonance imaging system.

2:20

2pBAb4. Speckle tracking in multiple-scattered pressure field. Ayse Kalkan-Savoy (Biomed. Eng. and Technol., UMass-Lowell, 1 University Ave., Lowell, MA 01854, ayse.k.savoy@gmail.com) and Charles Thompson (Elec. and Comput. Eng., UMass-Lowell, Lowell, MA)

Speckle tracking using ultrasound B-scan image sequences to quantify myocardial strain has the potential to become a standard method in echocardiography. A pressure field for a modulated Gaussian pulse traveling in inhomogeneous media is constructed. This model is based on computation of pressure field using Kirchoff integral formulation in frequency domain. Each pressure term of Neumann Series is converted into time domain and used in computation of Pade approximants. Pressure field in time and space is constructed utilizing Pade approximants and includes multiple-scattering effects. The scatterers of interest are large enough in size and contrast to contribute to strong and multiple-scattering. The signals at transducer point for each time frame are analyzed and speckles are tracked utilizing an optical flow algorithm. A velocity field which can be utilized to quantify strain is computed and visualized. The efficiency of this method will be discussed.

2:40

2pBAb5. Validation of three-dimensional strain tracking by volumetric ultrasound image correlation in a pubovisceral muscle model. Anna S. Nagle, Ashok R. Nageswaren, Balakrishna Haridas, and T. Douglas Mast (SEEBME, Univ. of Cincinnati, 3940 CVC, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, nagleas@mail.uc.edu)

Little is understood about the biomechanical changes leading to pelvic floor disorders such as stress urinary incontinence. In order to measure regional biomechanical properties of the pelvic floor muscles *in vivo*, a 3D strain tracking technique employing correlation of volumetric ultrasound images has been implemented. In this technique, local 3D displacements are determined as a function of applied stress and then converted to strain maps. To validate this approach, an *in vitro* model of the pubovisceral muscle, with a hemispherical indenter emulating the downward stress caused by intra-abdominal pressure, was constructed. Volumetric B-scan images were recorded as a function of indenter displacement while muscle strain was measured independently by a sonomicrometry system (Sonometrics). Local strains were computed by ultrasound image correlation and compared with sonomicrometry-measured strains to assess strain tracking accuracy. Image correlation by maximizing an exponential likelihood function was found more reliable than the Pearson correlation coefficient. Strain accuracy was dependent on sizes of the subvolumes used for image correlation, relative to characteristic speckle length scales of the images. Decorrelation of echo signals was mapped as a function of indenter displacement and local tissue orientation. Strain measurement accuracy was weakly related to local echo decorrelation.

3:00

2pBAb6. Measurement of surface acoustic wave in soft material using swept-source optical coherence tomography. Yukako Kato, Yuji Wada, Yosuke Mizuno, and Nakamura Kentaro (Tokyo Inst. of Technol., 4259-R2-26 Nagatsudacho Midoriku, Yokohama 226-8503, Japan, ykato@sonic.pi.titech.ac.jp)

In endoscopic elastography, it is needed to observe small area with high spatial resolution. Optical coherence tomography (OCT) is one of the candidate imaging methods, which has the depth resolution of several 10 μm. In this study, we try to find the propagation velocity of surface acoustic wave (SAW) using a swept-source OCT (SS-OCT). The depth scanning rate in the SS-OCT is rather fast, which is determined by the wavelength sweep of the light source as fast as 20–100 kHz. However, on the other hand, the lateral scanning is limited up to 100 times per second, since it is performed using a mechanical moving mirror. We develop a theory to estimate SAW velocity of tissues ranging from 1 to 20 m/s from data taken by the slow lateral scanning of less than 1 m/s using the OCT. The present method is tested for agar samples with different concentrations and also for several tissue samples. Vibrations are excited on the sample surface using a small stick connected to a loudspeaker. The measurements are carried out at many frequencies from 500 to 1000 Hz. The dependence of the SAW velocity on the concentration successfully agreed the previous results.

Session 2pEAa

Engineering Acoustics: Computational Methods in Transducer Design, Modeling, Simulation, and Optimization I

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60134

Chair's Introduction—12:55

Invited Paper

1:00

2pEAa1. Virtual prototyping of condenser microphones using the finite element method for detailed electric, mechanic, and acoustic characterization. Mads Jakob Herring Jensen (COMSOL A/S, Diplomvej 373, Lyngby 2800, Denmark, mads@comsol.dk) and Erling Sandermann Olsen (Brüel & Kjaer Sound and Vib. Measurement A/S, Naerum, Denmark)

Recent development and advances within numerical techniques and computers now enable the modeling, design, and optimization of many transducers using virtual prototypes. Here, we present such a virtual prototype of a Brüel & Kjær Type 4134 condenser microphone. The virtual prototype is implemented as a model using the finite element method with COMSOL Multiphysics and includes description of the electric, mechanic, and acoustic properties of the transducer. The acoustic description includes thermal and viscous losses explicitly solving the linearized continuity, Navier-Stokes, and energy equations. The mechanics of the diaphragm are modeled assuming a pre-stressed membrane, electrostatic attraction forces, and acoustic loads. The model includes electric description of the active and passive capacitances of the microphone cartridge as well as an external circuit model representing the preamplifier. Different modes of the system are studied, including the important first rocking mode of the membrane. The model has no free fitting parameters and results in the prediction of the frequency dependent sensitivity, capacitance, and mechanical impedance. The model results show good agreement with measured data.

Contributed Papers

1:20

2pEAa2. New planar nano-gauge detection microphone: Analytical and numerical acoustic modeling. Cécile Guianvarc'h, Thierry Verdot (LVA-Insa de Lyon, Villeurbanne, France), Jaroslaw Czarny (CEA-Leti, Grenoble, France), Emmanuel Redon, Kerem Ege, Jean-Louis Guyader (LVA-Insa de Lyon, 25 Bis, avenue Jean Capelle, Villeurbanne 69621, France, emmanuel.redon@insa-lyon.fr), Arnaud Walther, and Philippe Robert (CEA-Leti, Grenoble, France)

The miniaturization of microphones is of great interest for several fields, such as medical applications (audio implants), or consumer electronics (cell phones). Almost all existing miniature microphones rely on electrostatic transduction and offer good performances (sensitivity, frequency bandwidth). However, their sensitivity, proportional to the membranes area, would be dramatically reduced in case of extreme miniaturization. A new concept of microphones developed by CEA-LETI, which uses membranes moving in the plane of the substrate and inducing strain on piezoresistive Si nano-gauges (M&NEMS technology), seems promising for its miniaturization potential without significant decrease of sensitivity. The design and optimization of such planar piezo-resistive microphone require a deep understanding of its acoustic and vibroacoustic behavior. Regarding the small dimensions of the slits (1–100μm) and the sharp discontinuities in the microphones structure, viscous and thermal effects in the boundary layers and turbulent perturbations are of great importance, and must then be taken into account with high accuracy in device modeling. The aim of the present work is to provide accurate analytical and numerical (FEM) models able to gather all these effects in a consistent manner, and to suggest an experimental method to check their validity.

1:40

2pEAa3. Noise minimization in micromachined piezoelectric microphones. Robert Littrell (Baker-Calling, Inc., 1810 14th St., Ste. 210, Santa Monica, CA 90404, rlittrell@bakercalling.com) and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Piezoelectric MEMS microphones have been built for more than 30 years and they offer some advantages over other technologies such as improved linearity, simple construction, no need for a bias voltage or charge, and the ability to withstand high temperatures. Despite these advantages, a relatively high noise floor has always limited their utilization. Traditionally, the noise of these sensors has been minimized by viewing the preamplifier or amplifier as a black box with fixed gain and noise. This leads the designer to minimize noise by maximizing microphone sensitivity. By viewing the microphone-amplifier system together, we develop a different method of optimization, leading to lower noise. Further, by including the back cavity compliance of the package in the optimization, we can determine absolute limits on the minimum achievable noise floor with very few assumptions. To date, we have built piezoelectric MEMS microphones utilizing aluminum nitride with a 32 dBA noise floor. We can compute a minimum achievable noise floor of 24 dBA for the same sensing structure with a 2 mm^3 back cavity volume.

2:00

2pEAa4. Directional robustness of an *in-situ*, dual dipole omni microphone array for hearing instruments. Thomas Burns (Starkey Hearing Technol., 6600 Washington Ave. S, Eden Prairie, MN 55344, tburns@starkey.com)

The three critical factors for providing stable directional performance for typical microphone arrays used in hearing aids include the relative sensitivity and phase between the microphones in addition to the placement of the hearing instrument behind the user's ear. A directional system is robust if these factors can operate over a wide range of levels without degrading the directional performance. In this study, dual dipole microphones were arranged symmetrically around an omnidirectional microphone such that all inlets were collinear. Compared to an endfire array, whether it be a delay-and-sum or Blumlein configuration, this dual-dipole-omni array is remarkably more robust, yielding very little degradation in the Directivity Index for the aforementioned critical factors varying as much as +/− 3 dB, +/− 30 ms, and the directional axis of the hearing instrument varying +/− 20 degrees on the ear.

2:20

2pEAa5. The efficiency of receivers used in hearing aid devices. Michael Salameh (Research, Starkey Hearing Technol., 6600 Washington Ave. S., Eden Prairie, MN 55344, michael_salameh@starkey.com)

Current drain is one of the most important considerations in the hearing aid design. It is determined mainly by the receiver impedance. Recently, many additional features and technologies such as wireless are used in the hearing aid devices. These capabilities increase the current drain further. Efficiency evaluation becomes an important step in the receiver selection for hearing aid devices. Balanced armature receivers are widely used in hearing aids. In comparison with the moving-coil loudspeakers, these receivers are designed for better efficiency in closed acoustical loads found in different hearing aid styles. However, the receiver efficiency for hearing aid applications is not well defined, evaluated, or reported. In this paper, the receiver efficiency including the effect of the receiver size, impedance, and acoustical load is discussed. The evaluation of the receiver efficiency is revisited and a new approach is suggested. The simulation and the measurement results are presented.

2:40

2pEAa6. Cantilever mode piston transducer array. John Butler and Alexander L. Butler (Image Acoust., Inc., 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

The underwater sound cantilever mode transducer uses a piston, tail mass, dual piezoelectric cantilever benders, and mechanical arms for translating lateral bending cantilever motion to piston motion in a direction normal to the cantilever motion. This compact design provides a wide band response and exceptionally low resonance, especially under array loaded conditions. We present the operation of the cantilever mode transducer along with its unique response under array loading conditions where the resonance frequency decreases as the array gets larger. These results are demonstrated through a number of finite element array models and measured results on a 2x2 array. The single crystal PIN-PMN-PT results show a significant source level improvement over ceramic PZT-8 active material when driven up to 5 V/mil.

3:00

2pEAa7. Applications of network synthesis and zero-pole analysis in transducer modeling. Daniel Warren (R&D, Knowles Electron., 1151 Maplewood Dr., Itasca, IL 60143, daniel.warren@knowles.com)

When a transducer is an immutable component of a larger system being simulated, it is sufficient that the transducer model correctly reproduce behavior only at the available ports of the transducer. The behavior of two-port electroacoustic transducer should be completely characterized by three transfer functions related to the electrical and acoustic termination impedance and the transfer impedance. To the extent that the transducer could be represented as a analog circuit of passive linear elements, the same circuit could be exactly represented by rational polynomials of the Laplace s, embodied as a set of poles and zeros of the transfer functions. This invites the reverse process of identifying poles and zeros by nonlinear curve fit of rational polynomials to measured transfer data, perhaps even synthesizing a circuit directly from the identified poles and zeros. Measured transducer transfer data have been fit demonstrating both the utility and the pitfalls of this method. Curve fit transfer functions can be a compact and faithful representation of complex data over frequency, but have no predictive value outside the given data. Judicious selection of the number of poles and zeros, initial values, proper constraints, and some physical insight are necessary for stable curve fits. Further investigation into the relationship among the transfer functions of a physical system may lead to a meaningful model derived from curve fits.

3:20

2pEAa8. Improved estimation of direction of arrival of sound sources for hearing aids using gyroscopic information. Alan W. Boyd (Dept. of Electron. and Elec. Eng., Univ. of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, alan.boyd@strath.ac.uk), William M. Whitmer, W. Owen Brimijoin, and Michael A. Akeroyd (Inst. of Hearing Res., Med. Res. Council, Glasgow, United Kingdom)

Determining the direction of arrival (DOA) of a sound source is important in spatial audio signal processing, as it can lead to substantial improvement in noise reduction performance. Techniques such as generalized cross correlation with phase transform (GCC-PHAT) and adaptive eigenvalue decomposition (AED) perform optimally when the measurement microphones are fixed in place. However, hearing-aid microphones move with the listener's head movements, which can result in momentarily inaccurate directional estimates and noise artifacts in the output signal. Techniques such as GCC-PHAT experience degraded short-term performance in the presence of multiple signals and noise. The system presented measures instantaneous head movement velocity using a micro-electromechanical systems (MEMS) gyroscope attached to binaurally communicating hearing aids. Estimates of DOA for physically stationary sources are shifted based on the gyroscope's head-movement information. Using GCC-PHAT with gyroscopic input can produce robust *in situ* DOA estimates for several sources in reverberant environments. In addition, the gyroscope allows an adaptive beamformer to be steered to a target direction, compensating for head movements on a very short timescale during DOA estimation. Results show improved localization performance over a standard GCC-PHAT system during head movements.

3:40

2pEAa9. Miniaturized electrostatic receiver with small-sized backing electrode. Alexey Podkovskiy, Petr Honzík, Stéphane Durand, Nicolas Joly, and Michel Bruneau (LUNAM Université, LAUM (Laboratoire d'acoustique de l'université du Maine), UMR CNRS 6613, Avenue Olivier Messiaen, Le Mans F-72085, France, petr.honzik@gmail.com)

A miniaturized electrostatic receiver design, having a central cylindrical backing electrode of small radius surrounded by a flat annular cavity behind the circular membrane, can lead to both a higher sensitivity and a larger frequency bandwidth compared to the ones achieved with other designs, while bringing a geometrical simplicity which is advantageous from the point of view of microfabrication. An appropriate computational method, relying on a specific 2-D axisymmetrical simulation using an adaptive mesh and accounting for both viscous and thermal boundary layer effects, provides results against which analytical results can be tested. An analytical approach, which leads to solutions based on the eigenmode expansion of the membrane displacement, the acoustic pressure field depending on the radial coordinate in the central fluid gap but being assumed quasi-uniform in the annular cavity, is much faster in terms of running time and appears to be sufficiently accurate to achieve final optimization of this kind of devices.

4:00

2pEAa10. How a hearing aid transducer works. Noori Kim and Jont B. Allen (UIUC, 1085 Baytowner df 11, Champaign, IL 61822, nkim13@illinois.edu)

The oldest magnetic earphone, the balanced armature receiver (BAR), is the most widely used receivers in modern hearing-aid instruments, where the efficiency of the power (battery life) and the size of the device, as well as the larger frequency bandwidth, are critical parameters. Since these miniature loudspeakers remain one of the most expensive components of the

hearing-aids, a detailed studying of them is a cornerstone of understanding the hearing-aid system, and we believe that the appropriate and rigorous analysis of this transducer is critical. The motivation of this study started from the modeling of a widely used commercial hearing-aid receiver ED series, manufactured by Knowles Electronics, Inc. Our proposed model includes a semi-inductor and a gyrator along with the two-port network glue which enables us an intuitive design of the electromagnetic transducer. Based on the BAR model, we will investigate and discuss the roles of each physical component in the BAR such as a coil, magnets, an armature, a diaphragm, and the rear volume of the receiver. Ultimately, this work will deliver a fundamental and innate answer for the question, "How does a hearing-aid transducer work?"

4:20

2pEAa11. Numerical study of the cross-talk effects in acoustical transducer arrays and correction. Abdelmajid Bybi, Jamal Assaad (OAE Dept., CNRS UMR 8520, IEMN, Univ. of Valenciennes and Hainaut Cambrésis, Valenciennes, France), Anne-Christine Hladky-Hennion (ISEN Dept., UMR CNRS 8520, IEMN, Lille, France), Farouk Benmeddour, Sébastien Grondel, and Frederic Rivart (OAE Dept., CNRS UMR 8520, IEMN, Univ. of Valenciennes and Hainaut Cambrésis, Campus Mont Houy, Valenciennes 59313, France, farouk.benmeddour@univ-valenciennes.fr)

Cross-talk in acoustical transducer arrays is an undesirable phenomenon which decreases seriously the performances of these sensors. Indeed, when one element of the array is driven, it generates parasitic displacement fields at the radiating surfaces of the neighboring elements, which changes the directivity of the antenna. To well understand this phenomenon a transducer array similar to those used in medical imaging and NDT applications is modeled by finite element method. The research work, investigated systematically the effects of the cross-talk. First, it inspected the acoustical and mechanical cross-talk throughout the propagating medium and the filling material. Second, it studied the influence of the matching layer and the backing on the acoustical performances of the transducer. It is shown that the filling material and the matching layer are the major factor contributing to this phenomenon. In order to cancel the cross-talk a correction method previously developed by the author has been used. This solution consisted in applying adapted electrical voltages on each neighboring element of the active one in the purpose to reduce the displacement field on their active surface. This method was tested numerically and the obtained results clearly demonstrated its ability to reduce the cross-talk.

4:40

2pEAa12. Development and performance evaluation of virtual auditory display system to synthesize sound from multiple sound sources using graphics processing unit. Kanji Watanabe (Faculty of Systems Sci. and Technol., Akita Prefectural Univ., 84-4 Ebinokuchi, Tsuchiya, Yuri-Honjo, Akita 015-0055, Japan, kwatanabe@akita-pu.ac.jp), Yusuke Oikawa (Grad. School of Systems Sci. and Technol., Akita Prefectural Univ., Yuri-Honjo, Japan), Sojun Sato, Shouichi Takane, and Koji Abe (Faculty of Systems Sci. and Technol., Akita Prefectural Univ., Yuri-Honjo, Japan)

Head-related transfer function (HRTF) is characterized as sound transmission from sound source to listener's eardrum. When a listener hears a sound that is filtered with the HRTFs, the listener can localize a virtual target (sound image) as if the sound had come from the position corresponding to that at which the HRTFs were measured. A moving sound image can be generated to switch HRTFs of successive direction in real-time. While many virtual auditory displays (VADs) based on synthesis of HRTFs have been proposed, most of them can synthesize only a few sound images due to lack of computation power. In this article, the VAD system implemented based on graphics processing unit (GPU) was introduced. In our system, the convolution of HRTFs is parallelized on GPU to realize a high-speed processing. In addition, the multiple HRTFs each of which is corresponding to sound sources at different position are processed in parallel to control multiple sound image simultaneously. In this article, the performance of our system was evaluated not only objectively but also subjectively. The results showed that our current system can present at least 40 sound images simultaneously in real-time.

TUESDAY AFTERNOON, 4 JUNE 2013 | 512BF, 1:00 P.M. TO 5:00 P.M.

Session 2pEAb

Engineering Acoustics: Controlling Sound Quality

Stephen Butler, Chair
NUWC, Newport, RI 02841

Contributed Papers

1:00

2pEAb1. Subjective and objective evaluation of sound quality of radio programs transmitted via Digital Audio Broadcast (DAB+) System. Andrzej B. Dobrucki and Maurycy J. Kin (Chair of Acoust., Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, Wroclaw 50-370, Poland, andrzej.dobrucki@pwr.wroc.pl)

The work presents results of research on the sound quality of different radio programs transmitted via Digital Audio Broadcasting (DAB+). This assessment has been provided with a use of psychoacoustic model as well as standard listening tests, using an Absolute Category Rating (ACR) method of scaling, and Comparison Category Rating (CCR) method. Results have shown that sound quality gets worse when bit-stream is of the lowest values (48 kbit/s or 24 kbit/s). Application of the Spectral Band Replication processor significantly improves the perceived quality, which is satisfying for bit streams higher than 64 kbit/s, particularly for jazz and popular music. The assessment with ACR method (recommended for broadcast by International Telecommunication Union) showed better notes than CCR one. It means that recommended method is less critical. Also results obtained with psychoacoustic model are more similar to obtained with CCR method. The attributes of spatial impression change in different ways. The greatest distortion has been observed for the perspective and spaciousness of sound image, while the sound color as well as localization stability and accuracy of phantom sources remained almost the same.

1:20

2pEAb2. Objective evaluation of sound quality for attacks on robust audio watermarking. Akira Nishimura (Media and Cultural Studies, Tokyo Univ. of Information Sci., 4-1, Onaridai, Wakaba-ku, Chiba 2658501, Japan, akira@rsch.tuis.ac.jp), Masashi Unoki (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi, Japan), Kazuhiro Kondo (Grad. School of Sci. and Eng., Yamagata Univ., Yamagata, Japan), and Akio Ogihara (Faculty of Eng., Kinki Univ., Higashihiroshima, Japan)

Various attacks on robust audio watermarking have been proposed. Reversible signal processing attacks, such as sampling frequency conversion, degrade sound quality of the distributed watermarked audio (stego audio)

and disturb extraction of hidden data so that copyright detection systems using automated crawling are invalidated. Reversible signal processing of the attack can recover sound quality of the degraded audio data. In order to prove validity and security of audio watermarking system, analysis of the presumed attacks or reversible signal processing on stego audio, is required. However, these attacks on audio signal also degrade sound quality of commercial music where such pieces of music are considered to be not suitable for appreciation. Therefore, degradation of sound quality induced by various attacks should be taken into account to decide if the intensity of the attacks are realistic. In this study, objective audio quality measurement (PEAQ) was applied to the audio signals including typical perceptual coding, MP3, tandem MP3, MPEG4AAC, and reversible signal processing of sampling frequency conversion, noise addition, frequency shift, bandpass filtering, and echo addition. The results indicate requirements for robustness and criteria of the attacks on high quality and robust audio watermarking technology.

1:40

2pEAb3. Measurement of acoustic transmission properties of a handset with a piezoelectric vibrator using a head and torso simulator. Toshiharu Horiuchi and Tsuneo Kato (User Interface Lab., KDDI R&D Lab., Inc., 2-1-15 Ohara, Fujimino, Saitama 356-8502, Japan, to-horiuchi@kddilabs.jp)

This paper presents equalizing an acoustic transmission property from a handset with a piezoelectric vibrator at a pinna to an eardrum with that of a normal headphone by hearing the measured sound through a head and torso simulator (HATS). Recently, a piezoelectric vibrator that vibrates a pinna to produce sounds was adopted as a receiver of smartphones to improve perceived quality within noisy environments. The HATS, used for handset testings in accordance with ITU-T recommendations, has a silicon-rubber pinna simulator to reproduce realistic acoustic properties with its human-like shape and stiffness. However, the handset with the built-in piezoelectric vibrator is beyond the scope and was not tested on the HATS before. This paper clarifies the difference of frequency responses between the pinna simulator and the real pinna based on a subjective assessment that adjusts the loudness of pure tones through the pinna simulator to be equalized auditorily to those through the real pinna. We used B&K HATS Type 4128-D. The results indicated a flat response for the pinna simulator, while a low-pass-like response for the real pinna with a cutoff at 1.5 kHz. Thereby, the actual sound can be simulated from the sound measured by the HATS with the responses.

2:00

2pEAb4. The effect of firefighting protective equipment on head related transfer functions. Joelle I. Suits, Theodore F. Argo (Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keaton St., Austin, TX 78712, jsuits@utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX), Ofodike A. Ezekoye (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Craig A. Champlin (Austin, Texas)

Personal Alert Safety System (PASS) devices are used in the fire service to locate trapped or injured personnel. When a firefighter becomes incapacitated, the device emits an audible alarm to help rescue teams locate the downed firefighter. These devices have been successful, but there are still cases in which PASS is not effective, and the present project seeks to provide science-based guidance for improvements to PASS. One part of this complex problem is the effect of the protective equipment (helmet, eye protection, hood, coat) that is worn by firefighters on hearing. Since this has not previously been studied, it has not been accounted for in the current design of the PASS signal. To address this deficiency, head related transfer function (HRTF) measurements have been taken with a KEMAR acoustic mannequin wearing various combinations of the aforementioned equipment. Results indicate a reduced received level at the ear when the full complement of gear is worn, as might be expected, potentially causing a reduced detection range. In addition, the helmet and eye protection devices cause significant disruption of the normal HRTF patterns, which could potentially interfere with localization. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

2:20

2pEAb5. An escape guiding system utilizing the precedence effect for evacuation signal. Takahiro Fujikawa and Shigeaki Aoki (Electron. Information and Commun. Eng., Kanazawa Inst. of Technol., 7-1 Ohgigaoka, Nonoichi, Ishikawa 921 - 8501, Japan, aoki_s@neptune.kanazawa-it.ac.jp)

This research aims at building an escape guiding system using audio signals. The system utilizes the precedence effect for the listener to perceive easily the direction of an emergency exit. In one of the ordinary escape guiding systems, two or more loudspeakers are set on the ceiling of a passageway. Since the precedence effect is generated by delaying suitably the audio guidance signal radiated from the loudspeakers, the proper direction of the emergency exit is recognized. However, the effective listening area is limited. That is, the ordinary escape guiding systems is not effective in areas under loudspeakers. In this paper, a new method that the loudspeakers are set beside in a passageway is proposed. The generation and disappearance of the precedence effect of an audio signal in the new configuration are investigated. In the listening tests, the time delay and the difference of intensity between loudspeakers are parameters. The installation angle of the loudspeaker is another parameter. Male voice and female voices are used as emergency guidance. The guidance effect of the configuration in setting beside the loudspeaker is confirmed and the test results are discussed.

2:40

2pEAb6. Reduction of sound leakage in handheld devices using open loop control. GunWoo Lee, Aran Cha, SeoungHun Kim, YoungTae Kim (Samsung Electron. Co., Ltd., 416, Martan 3-dong, Yeongtong-gu, Suwon-si 443-742, South Korea, gw325.lee@samsung.com), and JungWoo Choi (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

Recently, handheld devices with sound functionality are popular. A problem encountered by using such devices is sound leakage due to inappropriate volume setting, which should be reduced not to disturb people around the user. In previous study, we presented an active control (ANC) technique to control such sound leakage in handheld devices. The least squares (LS) optimum control under various positions of error sensors was investigated. Based on the results, desirable and feasible microphone-loudspeaker setups were suggested. In this paper, we present a novel open loop leakage reduction scheme, and it is compared with adaptive noise control method. To achieve this, control available frequency bands and spatial ranges are studied. And the influence of leakage reduction performance by the receiver and control speaker's radiation pattern are analyzed. Also the effects of the physical environment including the user's hand for the leakage control performance are studied. Finally, the proposed method is implemented on the mobile phone mock-up, and the performance of actual measured leakage reduction is investigated.

3:00

2pEAb7. Active control applied to simplified wind musical instrument. Thibaut Meurisse, Adrien Mamou-Mani, René Caussé (Instrumental Acoust., IRCAM, 1 place Stravinsky, Paris, France, thibaut.meurisse@ircam.fr), and David Sharp (Acoust. Res. Group, The Open Univ., Milton Keynes, United Kingdom)

Musicians have always been interested in the evolution of their instruments. This evolution might be done either to adapt an instrument's quality to musicians' and composers' needs, or to enable it to produce new sounds. In this study, we want to control the sound quality and playability of wind instruments, using active control. The active control makes it possible to modify the input impedance (frequency, gain, and damping) of these instruments and to modify the instrument's quality. Simulations and first experiments on a simplified reed instrument are presented. We simulate a control of the modes (frequency, damping) of a cylinder using two different approaches: classic feedback and modal active control. Then, we apply these control methods on a simplified reed instrument with embedded microphone and speaker. Finally, the effects on sound and playability of the instrument is studied.

3:20

2pEAb8. Predicting speech transmissibility using ray tracing and statistical energy analysis. Sascha Merz, Vincent Cotoni, and Phil Shorter (ESI Group, 12555 High Bluff Drv, Ste. 250, San Diego, CA 92130, sascha.merz@esi-group.com)

Statistical Energy Analysis (SEA) is widely used for predicting interior noise across many different industries. SEA typically describes steady-state reverberant responses, for example the response at interior passenger locations in transportation applications due to steady-state exterior sources. However, many applications exist where the direct field is also important. This includes prediction of Speech Transmissibility for audio systems in automotive applications or public address systems in train and aircraft applications. For predicting Speech Transmissiblity, the transient response at a given receiving location due to transient excitation applied at a particular source location must be known. Geometrical methods such as ray tracing are often used for describing the first few reflections of the direct field; however, they are not well suited for describing the reverberant field and typically include approximate statistical assumptions about late time reflections. These assumptions often do not consider the detailed distribution of sound package in interior spaces. An alternative approach is to use ray tracing for predicting the direct field and low order reflections and to use SEA to predict late time reflections and background noise. This approach is computationally efficient and can use information contained in existing SEA models. The method is discussed and validation examples are presented.

3:40

2pEAb9. Comparison of precedence effect behavior in anechoic chamber with that in ordinary room. Koji Abe, Shouichi Takane, Sojun Sato, and Kanji Watanabe (Electron. and Information Systems, Akita Prefectural Univ., 84-4 Ebinokuchi Tsutiya, Yuri-Honjyo 015-0055, Japan, koji@akita-pu.ac.jp)

The precedence effect is well known as one of auditory illusions occurred by using multiple sound sources with similar sound output. When a sound is followed by similar sound separated with relatively short time delay, a single fused sound image is localized at the source position corresponding to the first-arriving sound. This feature is applicable to public address systems, which make audience perceive the sound image different from the actual sound source positions prepared for the system, with some sound reinforcement achieved. In spite of many studies in this phenomenon, the behavior of the precedence effect has been investigated for limited sound source arrangements in laboratory environments like anechoic chamber. On the other hand, this behavior in the ordinary room is not obvious, and it is effective to clarify the difference of the behavior of the precedence effect in anechoic chamber from that in the ordinary room for the application of the precedence effect to the public address system. In this study, the similar sound sources were installed both in the lecture room and in the anechoic chamber, and the behavior of the precedence effect was compared each other with the given time and level difference among sound sources.

4:00

2pEAb10. Vertical sound image control using level differences between parametric speakers. Kumi Maeda, Takanori Nishino, and Hiroshi Naruse (Grad. School of Eng., Mie Univ., 1577 Kurimamachiya-cho, Tsu 5148507, Japan, nishino@pa.info.mie-u.ac.jp)

Horizontal sound image can be controlled by using level difference between two loudspeakers; however, vertical sound localization is difficult. In this report, we propose a method of controlling a sound image with sound level differences between two parametric speakers that have a super-directivity. Our proposed system uses sounds that are reflected on a wall. In our experiments, two parametric speakers were arranged 56.6 and 226.0 cm high, respectively. Vertical sound localization was evaluated by subjective tests. Subjects were seven males and one female. Test signals were a white noise whose duration was 0.5 s. Sound level differences between parametric speakers were $-\infty$, -9, -6, -3, 0, 3, 6, 9, and ∞ dB. Test signals were presented three times to each subject in a random order. Both parametric speakers were arranged at angles of $0°$, $\pm 5°$, and $\pm 10°$ from the horizontal plane, respectively. Answers were examined by the Wilcoxon rank-sum test. From the results, good performances were obtained when parametric speakers were arranged at angles of $\pm 10°$. [Work supported by a Grant-in-Aid for Scientific Research (24500203).]

4:20

2pEAb11. Characteristics of whistle noise from mufflers with perforated pipes. Tatsuya Yamada, Takehiko Seo, Masato Mikami (Grad. School of Sci. and Eng., Yamaguchi Univ.,2-16-1 Tokiwadai, Ube, Yamaguchi, Japan, 201, 3-44 Wakamatsucho, Ube, Yamaguchi, Japan, Ube, Yamaguchi 755-8611, Japan, s045ve@yamaguchi-u.ac.jp), and Takashi Esaki (Sango Co., Miyoshi, Aichi, Japan)

As a countermeasure to reduce exhaust gas pulsation noise in a wide frequency range, the exhaust system employs expansion, resonance and sound absorbing structures with perforated pipes. However, whistle noise is generated near holes under some conditions. In straight-through-type mufflers, sound absorbing materials covering punching holes can suppress whistle noise generation. On the other hand, in the expansion-cavity-type muffler, which has inlet and outlet insertion pipes, effective countermeasures have not been clarified yet. The purpose of this study is to improve understanding of whistle noise generation mechanism in mufflers with perforated pipes. First, we measured sound pressure of whistle noise radiating from a straight-through-type sub-muffler with a perforated pipe with steady flow. Results show that the frequency of predominant whistle noise became higher stepwisely with increasing the flow velocity and was higher with smaller hole diameter. Strouhal number based on the hole diameter, the frequency of predominant whistle noise and flow velocity existed within a certain range while the hole diameter and flow velocity were varied. Next, we measured the sound pressure of whistle noise radiating from an expansion-cavity-type main muffler with a perforated pipe. The whistle noise generation is discussed in comparison with that for the sub-muffler.

4:40

2pEAb12. Back scattering attenuators (silencers). Giora Rosenhouse (89 Hagalil St., Haifa 32684, Israel, fwamtech@bezeqint.net)

Back scattering silencers are a kind of "sonic crystals." The frequency range of "sonic crystals" modeling includes the whole audio range and ultrasound, up to phonon waves. Here we concentrate on applications in the audio domain. However, the present investigation is applicable at the higher frequencies as well¸ because of the high scalability of the system. The paper defines and analyses 2-D and 3-D back scattering silencers, made of arrays of rigid or soft obstacles (cylinders, spheres or prolate/oblate spheroids for example), in order to attenuate plane waves by multiple scattering along a wave guide. This effect is strong especially at certain band-gaps along the frequency domain. Each obstacle reflects secondary waves that are partially reflected. Thus, along each row of the array, the sound waves lose a certain amount of the energy that adds to the total amount of attenuation. Specifically, the paper analyses back scattering silencers built of meshes of either cylinders or prolate spheroids, where each obstacle is located symmetrically within a fluid cell and each cell is identical to the others.

Session 2pED

Education in Acoustics: Teaching Methods in Acoustics

Preston S. Wilson, Chair
Appl. Res. Lab., Univ. of Texas at Austin, 1 University Station, TX 78712-0292

Contributed Papers

1:20

2pED1. A design of an impedance tube for teaching acoustic material properties and laboratory techniques. Chelsea E. Good, Aldo A. Glean, Joseph F. Vignola, John A. Judge, Teresa J. Ryan, Nicole Bull (Mech. Eng., Catholic Univ. of America, 620 Michgan Ave. NE, Washington, DC 20064, 26good@cardinalmail.cua.edu), and Diego Turo (Bioengineering, George Mason Univ., Fairfax, VA)

The design and implementation of an adaptable acoustic impedance tube for instructional use will be presented. This system, which was designed for use in a graduate-level laboratory environment, is in accordance with ASTM standard E1050-98. The system is configurable for both horizontal and vertical measurements. Vertical use enables acoustic property measurement of granular materials, allowing characterization of a wide variety of media. The academic content is delivered by engaging students in measurements using the impedance tube. Specific acoustics content objectives include modeling of sound propagation in porous materials and the role of acoustic parameters such as porosity, tortuosity, and flow resistivity in quantifying acoustic behavior of materials. Instructional goals also include mastery of computer based data acquisition and processing. This aspect of modern laboratory practice is a critical part of our graduate acoustics curriculum. Students participated actively in the development of the software used to collect data and calibrate the impedance tube. The system design required the students to solve practical problems such as cutting and positioning of the sample as well as more demanding tasks such as design of a multilayer porous medium with desired acoustic properties.

1:40

2pED2. Popular papers with short case stories on acoustics and vibration for practical engineers and students. Roman Vinokur (Engineering, ResMed Motor Technol., 9540 De Soto Ave., Chatsworth, CA 91311, romanv@resmed.com)

One of the reasons for using foam wedges or cones in hemi-anechoic rooms is a gradual change of the acoustical impedance in order to reduce the reflection of incident sound waves from the sound absorbing walls. By analogy, popular papers on science (in particular, in acoustics and vibration) facilitate a smooth introduction to new theories because of their small cognitive "impedance" to understanding the written information. Such papers are relevant mostly for extramural reading but they help engineers and students to promptly perceive important effects and applications via interesting case stories and simplified physics and mathematics. Generally speaking, this approach is not new: in particular, it was successfully applied by Perelman in his book "Physics for Entertainment." But in author's opinion, for better effectiveness such texts should be limited in size and include 3–4 related short case stories from actual engineering or consulting practice, history, news, or literature. To illustrate this method, several one-page papers published in the "Sound and Vibration" magazine will be briefly discussed: "Vibroacoustic Measurements without Transducers," "A Common Myth about Mechanical Resonance," "Only the Best Will Do," and "Haunted Buildings and Other Acoustical Challenges."

2:00

2pED3. Mechanical bent-type models of the human vocal tract consisting of blocks. Takayuki Arai (Dept. of Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

In our previous work, we developed several physical models of the human vocal tract and reported that they are intuitive and helpful for students studying acoustics and speech science. Models with a bent vocal tract can achieve relatively realistic tongue movements. These bent-type models had either a flexible tongue or a sliding tongue. In the former case, the tongue was made of a flexible gel-type material so that we could form arbitrary tongue shapes. However, this flexibility meant that training is needed to achieve target sounds. In the latter case, the tongue was made of an acrylic resin, and only a limited number of vowel sounds can be achieved because so few sliding parts are available to change the tongue shape. Therefore, in this study, we redesigned the mechanical bent-type models so that they now consist of blocks. By placing the blocks at the proper positions, the block-type model can produce five intelligible Japanese vowels. We also designed a single bent-type model with sliding blocks that can produce several vowel sounds. [This work was partially supported by a Grant-in-Aid for Scientific Research (24501063) from the Japan Society for the Promotion of Science.]

2:20

2pED4. Teaching acoustics at architect students using digital tools. Delphine Bard (Eng. Acoust., Lund Univ., John Ericssonväg 1, Lund 221oo, Sweden, delphine.bard@construction.lth.se), Tina-Henriette Kristiansen (Architecture, Lund Univ., Lund, Sweden), and Eva Frühwald Hansson (Solid Mechanics, Lund Univ., Lund, Sweden)

At the School of Architecture, Lund University Sweden, courses are taught in different ways. A large part of the education during year one and two is held as "studios," doing creative (individual) project work. Usually acoustics courses rather correspond to the traditional engineering education style, using lectures, exercises, small project works, and final written examination. The problem is that many students do not know how to use the gathered information in their creative works. The aim of the study covered by this paper was to improve upon the existing teaching/learning of the fundamental acoustics principles scheme by introducing new methodologies. In order to achieve our goal, we gave the students two different assignments: In the first assignment, the students had to produce short educational movies to explain and teach acoustics principles to their peers. In the second assignment, they should implement the new knowledge gathered in the first assignment into their individual creative projects.

2:40

2pED5. Black box measurements – Using a family of electrical circuits as a tool for self-guided learning in acoustical engineering. Bernardo H. Murta, Sergio Aguirre, Jessica Lins, Stephan Paul, Eric Brandao (Departamento de Estruturas e Construção Civil, Universidade Federal de Santa Maria, Rua Dezenove de Novembro, 289, 302, Santa Maria, Rio Grande do Sul 97060160, Brazil, be.murta@gmail.com), and Pascal Dietrich (Inst. of Tech. Acoust., RWTH Aachen Univ., Santa Maria, Brazil)

A partnership between Brazil's first undergraduate program in Acoustical Engineering and the Institute of Technical Acoustics of RWTH Aachen University yielded in a didactic project that uses the engineering software MATLAB with the ITA-Toolbox to teach acoustic measurements. Simple electrical circuits are used to mimic typical behavior of acoustical systems. This low-cost solution has proven to be didactically very effective since it helps students to identify themselves with the measurement tasks. Two hardware solutions were developed-a simple oscillator circuit integrated into connectors of audio cables and a desktop box containing seven different transfer characteristics ranging from ideal linear and time-invariant to nonlinear and time-varying behavior. Undergraduate students of Acoustical Engineering used both devices in classroom experiments for self-guided learning by comparing their results to published results. Students were able to learn the fundamental concepts of acoustical measurements and to handle measurement tasks. Besides the practical experiences and the learning effect, the students were also encouraged to step into the open source routines of the software, understand the signal processing steps, adapt routines, and even write their own ones, e.g., a GUI that provides effective control of the measurement via touch-screens.

3:00

2pED6. A for play! Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

The task was straightforward; design and build a playhouse to be raffled for a charitable organization. A team consisting of students, volunteers, and faculty banded together to not only meet the requirements but to exceed the typical preconceived ideas of a totally enclosed miniature home. Based upon the needs of juvenile clients, the design team focused more on "play" than on "house" when working out conceptual ideas. The playful design was based upon the enclosure being partially open to allow air flow, sunlight, and the ability for the inhabitants to have an aural connection to the outside. The idea of having partially open space on the lower level, a mere 5'-0" x 5'-0" footprint, flanked by stepped bands of cedar and cypress yielded a particular acoustical presence. The space is not only visually unique, but the selection of materials, how they were cut and assembled, and the scale in relation to a seated child enhance the fun factor by creating an enveloping and somewhat amplified acoustic. This project provided pedagogical opportunities within an atypical learning environment. The final inhabitable playhouse exceeded our visual and acoustical expectations of a small space and prove acoustics "plays" an intrinsic role despite occupant age.

TUESDAY AFTERNOON, 4 JUNE 2013 — 512DH, 1:00 P.M. TO 5:40 P.M.

Session 2pMU

Musical Acoustics: Musical Preference, Perception, and Processing

Jean-François Petiot, Cochair
IRCCyN, Ecole Centrale de Nantes, 1 rue de la noe, BP92101, NANTES 44321, France

Richard L. King, Cochair
Music Res., McGIll Univ., 555 Sherbrooke West, Montreal, QC H3A 1E3, Canada

Contributed Papers

1:00

2pMU1. Modeling of the subjective quality of saxophone reeds. Jean-François Petiot, Pierric Kersaudy (IRCCyN, Ecole Centrale de Nantes, 1 rue de la noe, BP92101, NANTES 44321, France, jean-francois.petiot@irccyn.ec-nantes.fr), Gary Scavone, Stephen McAdams (CIRMMT, Schulich School of Music, McGill Univ., Montreal, QC, Canada), and Bruno Gazengel (LAUM, Université du Maine, Le Mans, France)

The subjective quality of cane reeds used on saxophones or clarinets may be very different from one reed to another even though the reeds have the same shape and strength. The aim of this work is to understand the differences in the subjective quality of reeds and to explain them with objective measurements. A subjective study, involving a panel of 10 musicians, was first conducted on a set of 20 reeds of the same strength. Second, signal recordings during saxophone playing (*in vivo* measurements) were made of the pressures in the player's mouth, in the mouthpiece and at the bell of the instrument. These measurements enable us to deduce specific parameters, such as the threshold pressure or the spectral centroid of the notes. After an analysis of the subjective and objective data (assessment of the agreement between the assessors and the main consensual differences between the reeds), correlations between the subjective and objective data were performed. To propose a model of the subjective quality, a machine learning approach was proposed using partial least-squares (PLS) regression and PLS discriminant analysis. Results show interesting performance of the model in cross validation and open the potential for an objectification of the perceived quality.

1:20

2pMU2. Perceptual evaluation of violins: A comparison of intra-individual agreement in playing vs. listening tasks for the case of richness. Charalampos Saitis, Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke Str. West, Montreal, QC H3A 1E3, Canada, charalampos.saitis@mail.mcgill.ca), Claudia Fritz (Lutheries-Acoustique-Musique, Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Paris, France), and Bruno L. Giordano (Inst. of Neurosci. and Psychol., Univ. of Glasgow, Glasgow, United Kingdom)

In previous studies by the authors, it was shown that there is a significant lack of agreement between violinists when evaluating different instruments in terms of perceived richness in free-playing tasks. A new experiment was designed to further investigate the perceptual evaluation of richness using both a constrained-playing task, which was recorded, and a subsequent listening task (using the previously recorded sounds). The goal was to compare the evaluation of richness from playing vs. listening tasks in order to better understand whether they are based more on auditory feedback or tactile and

proprioceptive cues in the wider context of correlating audio features extracted from the recordings with richness judgments. Skilled violinists were asked to rank five different instruments by playing only certain notes on the G-string. Subsequently, the players were asked to listen to their recordings and rank the violins. Results appeared to show a higher inter-individual agreement relative to previous studies. Furthermore, the rankings in the playing task were generally different from those in the listening task, indicating that the evaluation of richness is based on different criteria in the two cases. Results from matching the tristimulus and spectral centroid of recorded tones to richness judgments will be presented at the conference.

1:40

2pMU3. Music of the body: An investigation of skull resonance and its influence on musical preferences. Jitwipar Suwangbutra, Rachelle Tobias, and Michael S. Gordon (Dept. of Psych., William Paterson Univ., 300 Pompton Rd., William Paterson U., Wayne, NJ 07470, jitwipar@gmail.com)

Musical preferences can be attributed to environmental and biological factors. This research analyzes the specific influence of body resonance, and in particular, how the resonant properties of the skull might contribute to auditory perception of music and musical preferences. To examine this issue resonances were sampled from a set of participants and analyzed using FFTs. The fundamental frequencies of each participant's head was correlated against their preference amongst a set of novel melodies presented in each of the 12 major keys. Using this method the spectral properties of the melody could be directly related to the resonant properties of a listener's skull to evaluate their influence. While results were subtle, participants were found to be influenced in their judgments of loudness and musical preference for the melodies. Conclusions from this research support speculation on an embodied model of cognition for musical interactions.

2:00

2pMU4. Preferences for melodic contours transcend pitch. Jackson Graves, Christophe Micheyl, and Andrew J. Oxenham (Psychology, Univ. of Minnesota, 1849 Washington Ave. S, Apt 435, Minneapolis, MN 55454, grave276@umn.edu)

The question of what makes a good melody has interested composers, music theorists, and psychologists alike. Many of the observed principles of good "melodic continuation" involve the melodic contour—the pattern of rising and falling pitch within a sequence. Recent work has shown that contour perception can extend beyond pitch to other auditory dimensions, such as timbre and loudness. Here we show that the generalization of contour perception to non-traditional dimensions also extends to melodic preferences and expectations. We find that subjective continuation ratings for sequences that vary in brightness or loudness generally conform to the same contour-based expectations as pitch sequences. The results support the hypothesis that contour perception is a general auditory phenomenon, and suggest that the well-known preference for narrow ranges and small intervals in melodies is not unique to the dimension of pitch. [Work supported by NIH Grant R01DC05126 and by the Undergraduate Research Opportunities Program of the University of Minnesota.]

2:20

2pMU5. Development of a new series of tests to assess the effectiveness of hearing aids for the perception of music. Martin Kirchberger (Health Sci. & Technol., ETH Zürich, Universitätsstrasse 2, Zürich 8092, Switzerland, martin.kirchberger@phonak.com), Frank A. Russo (Psychology, Ryerson Univ., Toronto, ON, Canada), Peter Derleth, and Markus Hofbauer (Sci. & Technol., Phonak AG, Stäfa, Switzerland)

A new series of tests has been designed to assess the effectiveness of hearing aids for the perception of music. Within each subtest, discrimination thresholds for low-level acoustic dimensions are determined adaptively using a 2AFC method within the context of a musical judgment regarding melody, harmony, timbre or meter. The presented test stimuli are synthesized and either unprocessed or processed by different hearing aid signal processing algorithms before being played back via loudspeaker. The battery will be used to evaluate different hearing aid algorithms with regard to their benefit for functional hearing in music. A group of six normal hearing control participants (6.7 dB HL) and five hearing impaired participants (34 dB HL) each performed the melody subtest and the harmony subtest twice. The hearing impaired subjects had higher discrimination thresholds than the control group. A comparison of the results from both administrations suggests that these two subtests have good test-retest reliability.

2:40

2pMU6. The examination of the performance motion and emotional valence by a pianist. Yuki Mito, Hiroshi Kawakami (College of Art, Nihon Univ., 2-42-1 Asahigaoka Nerima-Ku, Tokyo 176-8525, Japan, mitotic@hotmail.com), Masanobu Miura (Faculty of Sci. and Technol., Ryukoku Univ., Shiga, Japan), and Yukitaka Shinoda (College of Sci. and Technol., Nihon Univ., Tokyo, Japan)

Until now, we examined the performance motion by the snare drum. Particularly, we analyzed an association between emotion and motion using motion capture. We analyzed total of six emotions [tenderness, happiness, sadness, fear, anger, and non-emotion from Juslin (2001)]. The analysis method averaged the performance motion of six emotions. We named the method as "Motion Averaging Method (MAM)." We calculated the difference of the each emotion by the MAM and revealed the characteristic of motion. From the result, we achieved results in snare drum motion. Therefore, in this study, we considered the association between emotion and motion in the keyboard. The subject was a professional pianist. A musical piece was an etude of the music dictation. Motion capture system is the MAC 3D System of Motion Analysis Corp., which is an optical motion capture system. The measurement marker bonded 33 points to the upper body. As a result, we understood the difference in emotional valence by the center of gravity of the head, an arm, the body, and the hand. Particularly, the non-emotion was move less than other five emotions. Then, we were able to express an association between five emotions of Juslin and motion as a figure.

3:00

2pMU7. Human ability of counting the number of instruments in polyphonic music. Fabian-Robert Stöter, Michael Schoeffler, Bernd Edler, and Jürgen Herre (Int. Audio Lab. Erlangen, Am Wolfsmantel 33, Erlangen 91058, Germany, fabian-robert.stoeter@audiolabs-erlangen.de)

There are indications that humans are only able to correctly count up to three voices in polyphonic music pieces of homogeneous timbre, where each voice is played by the same instrument. A more general case, where voices are played by instruments of inhomogeneous timbre, has not been fully addressed so far. In order to approach this question we conducted a listening experiment with 62 participants to find out whether both scenarios — instrumentation by inhomogeneous or homogeneous timbre — share the same outcome. This paper describes the design of the experiment including an analysis of the results, which show that both scenarios are related. Furthermore, a detailed analysis of the error rates in correctly counting the number of instruments reveals that there are significant differences between non-musician and musician listeners, in particular regarding the upper auditory limit of the number of correctly counted instruments. Based on these results, models for the perception of instruments in auditory streams can be developed.

3:20–3:40 Break

3:40

2pMU8. Loudspeakers and headphones: The effects of playback systems on listening test subjects. Richard L. King, Brett Leonard (Music Res., Schulich School of Music, McGill Univ., 555 Sherbrooke West, Montreal, QC H3A 1E3, Canada, richard.king@mcgill.ca), and Grzegorz Sikora (Automotive Audio, Bang & Olufsen, Pullach, Germany)

Many modern listening test designers disagree on the best playback system to be used when conducting tests. The preference often tends toward headphone-based monitoring in order to minimize the possibility of undesirable acoustical interactions with less than ideal testing environments. On the other hand, most recording and mixing engineers prefer to monitor on loudspeakers, citing a greater ability to make critical decisions on level balances and effects. While anecdotal evidence suggests that differences exist between systems, there is little quantified, perceptually based data to guide both listening test designers and engineers in what differences to expect

when alternating between monitoring systems. Controlled tests are conducted with highly trained subjects manipulating the level of solo musical elements against a backing track, using both headphones and loudspeakers. This task serves to make the results equally applicable to critical mixing tasks and rigorous listening tests. The results from both playback systems are compared, showing a defined difference in the mean levels set on the two different monitoring systems. Likewise, the variance seen across subjects is larger when monitoring on headphones than on loudspeakers, lending credence to the hypothesis that tests conducted on one playback system may not be equally applicable to the other.

4:00

2pMU9. Modeling listener distraction resulting from audio-on-audio interference. Jon Francombe, Russell Mason, Martin Dewhirst (Dept. of Music and Sound Recording, Inst. of Sound Recording, Univ. of Surrey, Guildford, Surrey GU2 7XH, United Kingdom, j.francombe@surrey.ac.uk), and Søren Bech (Bang & Olufsen, Struer, Denmark)

As devices that produce audio become more commonplace and increasingly portable, situations in which two competing audio programs are present occur more regularly. In order to support the design of systems intended to mitigate the effects of interfering audio (including sound field control, noise cancelation or source separation systems), it is desirable to model the perceived distraction in such situations. Distraction ratings were collected for a range of audio-on-audio interference situations including various target and interferer programs at three interferer levels, with and without road noise. Time-frequency target-to-interferer ratio (TIR) maps of the stimuli were created using a simple auditory model. A number of feature sets were extracted from the TIR maps, including combinations of mean, standard deviation, minimum and maximum TIR taken across the duration of the program item. In order to predict distraction ratings from the features, linear regression models were produced. The models were evaluated for goodness-of-fit (RMSE) and generalizability (using a K-fold cross-validation procedure). The best model performed well, with almost all predictions falling within the 95% confidence intervals of the perceptual data. A validation data set was used to test the model, suggesting areas for future improvement.

4:20

2pMU10. Effects of audio latency in a disc jockey interface. Laurent S. Simon, Arthur Vimond (INRIA, INRIA Rennes, Bretagne Atlantique, Campus Universitaire de Beaulieu, Rennes Cedex 35042, France, laurent.s.simon@inria.fr), and Emmanuel Vincent (INRIA Nancy, Rennes, France)

This study presents an evaluation of the disturbance caused by audio latency in a DJ-ing task. An experiment was conducted, during which subjects were asked to synchronize one piece of music to a reference piece of music using a common DJ-ing interface. Synchronization was performed by adjusting the speed of one of the music pieces to that of a reference piece of music and time-aligning both pieces. Latency was introduced between the interface and the audio output, varying between 0 and 550 ms. The average synchronization time was estimated as a function of subjects, beat-per-minute difference between the pieces of music, and latency. Results showed that for trained DJs, synchronization time increased significantly above 130 ms of audio latency, whereas for naive subjects, latency had no influence on the synchronization time.

4:40

2pMU11. Evaluating the absolute volume of digital sound source measurement and standard measuring unit. Doo-Heon Kyon and Myung-Jin Bae (Electron. Eng. Dept., Soongsil Univ., Hyeongnam Eng. Hall #1212, Sangdo-dong, Seoul 156-030, Democratic People's Republic of Korea, kdhforce@gmail.com)

Listeners do not know the actual volume of sound before playing a sound source, so they have to adjust the volume through trials and errors. Moreover, they have to change the volume repetitively because each sound source has different volume. If we can identify the absolute volume of a sound source at the perspective of listener, the volume of all sound sources can be effectively standardized. This study evaluated a method to measure the absolute volume of a digital sound source and suggested the dB(N) as a measuring unit. The pink noise was used as a reference sound source, to be used for measuring the absolute volume. The pink noise was set as 60 dB(N), which is equal to sound output of 60 dB(A). The volume was adjusted until the pink noise and the target sound matches into a recognizable volume by reducing the pink noise or target sound source under the given environment. Subsequently, the difference is reflected to 60 dB(N) to determine the absolute volume. The accuracy of measured results was confirmed through a music listening test and suggested how to develop the volume system using the absolute volume.

5:00

2pMU12. Refining the stereo technique for augmented ambience gradient: Improvements in stereo image, spatial envelopment, and mixing flexibility. David J. Tagg and Kevin Fallis (Sound Recording, McGill Univ., 74 Cameron Cres, Toronto, Ontario M4G 2A3, Canada, kevin.fallis@mail.mcgill.ca)

While working on location, recording engineers are often challenged by insufficient monitoring. Poor (temporary control room) acoustics and/or mandatory headphone monitoring can make judgments regarding microphone choice and placement difficult. This compromised monitoring often leads to timbral, phase, and stereo image problems. We are often forced to choose between the improved spatial imaging of near-coincident techniques and the attractive acoustic envelopment from spaced omni-directional mics. This research reviews a new technique: Stereo Technique for Augmented Ambience Gradient (STAAG), which aims to improve stereo imaging, ambient envelopment, and flexibility in the mix. Building on a preliminary study, this research realizes ideal microphone angle/spacing combinations to promote spatial accuracy, investigates the quality of the ambient envelopment compared to omnidirectional-based techniques, and the ability of STAAG to allow an engineer to manipulate the direct to reverberant energy ratio during post-production without corrupting the stability of the stereo image.

5:20

2pMU13. Real-time concatenative synthesis for networked musical interactions. Chrisoula Alexandraki (Dept. of Music Technol. and Acoust., Technol. Educational Inst. of Crete, E. Daskalaki Perivolia, Rethymnon, Crete 74100, Greece, chrisoula@staff.teicrete.gr) and Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Hamburg, Germany)

The recent proliferation of Networked Music Performances has led to the investigation of low-latency, low-bitrate musical encoding schemes, including audio codecs and control protocols that specifically address the requirements of live musical interactions across the Internet. This work presents an alternative perspective inspired by the "synthesis by analysis" approach, tightly constrained in terms of processing latencies and rendering quality. The entire process is fully automated and involves an offline processing phase (that takes place prior to performance) and an online real-time analysis-synthesis phase. The offline phase involves processing a solo recording of each musician's part so as to acquire (a) audio segments corresponding to each note in the performance, and (b) a trained Hidden Markov Model to be later used for online analysis. During live performance, the online analysis process encodes the position of the performance on a music score and re-synthesizes the audio waveform by concatenating the audio segments of the offline phase. Although the synthesized waveform originates from an offline solo recording, it is synchronized to the live performance at note level, so as to allow for rendering a wide range of musical tempi as well as their expressive variations. The paper presents the complete methodology and reports on implementation details and preliminary evaluation results.

Session 2pNSa

Noise: Community Response to Low-Amplitude Sonic Booms

Alexandra Loubeau, Cochair
NASA Langley Res. Ctr., MS 463, Hampton, VA 23681

Juliet A. Page, Cochair
Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202

Chair's Introduction—12:55

Invited Papers

1:00

2pNSa1. Overview of the waveform and sonicboom perception and response program. Juliet A. Page (Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202, juliet.page@wyle.com)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted in California in November 2011 was designed to test and demonstrate the applicability and effectiveness of techniques to gather data relating human subjective response to multiple low-amplitude sonic booms. It was a practice session for future wider scale testing of communities, eventually using a purpose built low-boom demonstrator aircraft. The WSPR program addressed the following: design and development of an experimental design to expose people to low-amplitude sonic booms; development and implementation of methods for collecting acoustical measures of the sonic booms in the neighborhoods where people live; design and administration of social surveys to measure people's reactions to sonic booms; and assessment of the effectiveness of various elements of the experimental design and execution to inform future, wider-scale testing. The low boom community response pilot experiment acquired sufficient data to assess and evaluate the effectiveness of the various physical and psychological data gathering techniques and analysis methods. Results include a comparison of survey modes, techniques for correlating subjective and objective data, assessment of single event and cumulative daily sonic boom subjective and percent highly annoyed results, and methods for analysis of empirical boom data.

1:20

2pNSa2. Low amplitude sonic boom noise exposure and social survey design. Kathleen K. Hodgdon (Appl. Res. Lab., Penn State, ARL North Atherton St., P.O. Box 30, State College, PA 16804-0030, kkh2@psu.edu) and Juliet Page (Wyle, Arlington, VA)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted a field study relating subjective response to noise from multiple low-amplitude sonic booms. The team was led by Wyle and included researchers from Penn State, Tetra Tech, and Gulfstream Aerospace Corp. The test exposed residents in the Edwards Air Force Base (EAFB) Housing area to two weeks of low-amplitude sonic booms while recording their responses via surveys. The noise exposure design balanced DNL across test days, the number of low, medium, and high booms, and the separation of booms between AM and PM flight sequences. Survey instruments consisted of a Baseline survey, a Single Event survey, and a Daily Summary survey. The WSPR low boom survey included a question on strength of annoyance, followed by questions on the strength of perception of five additional variables that contribute to the annoyance response. Three modes of administration were utilized for both the single event and daily summary surveys: paper/pen, web-based, and Mobile (Apple) device. The survey followed recommendations published by The International Commission on the Biological Effects of Noise (ICBEN). The data from the low boom field test provide a measure of the acceptance of low booms in an acclimated community.

1:40

2pNSa3. A flight research overview of the Waveforms and Sonicboom Perception and Response Project, the National Aeronautics and Space Administration's pilot program for sonic boom community response research. Larry J. Cliatt, Edward A. Haering, Michael D. Holtz, Thomas P. Jones, Erin R. Waggoner (NASA Dryden Flight Res. Ctr., P.O. Box 273, M.S.2228, Edwards, CA 93523, larry.j.cliatt@nasa.gov), Scott L. Wiley (Tybrin Corp., Edwards, CA), Ashley K. Parham (NASA Dryden Flight Res. Ctr., Edwards, CA), and Franzeska F. Houtas (Tybrin Corp., Edwards, CA)

To support the National Aeronautics and Space Administration's (NASA) ongoing effort to bring supersonic commercial travel to the aerospace industry NASA Dryden, in cooperation with other government and industry organizations, conducted a flight research experiment to identify the methods, tools, and best practices for a large-scale sonic boom community human response test. The name of the project was Waveforms and Sonicboom Perception and Response (WSPR). Such tests go toward building a dataset that governing agencies like the Federal Aviation Administration and International Civil Aviation Organization will use to establish regulations for acceptable sound levels of overland sonic booms. This paper focuses on NASA's role in the project on essential elements of community response testing including recruitment, survey methods, instrumentation systems, flight planning, and operations. Objectives of the testing included exposing a residential community with sonic boom doses designed to simulate those produced by the next generation of commercial supersonic aircraft. The sonic booms were recorded with an instrumentation array that spanned the community. Human

response data was collected using multiple survey methods, and was correlated to acoustic metrics from the sonic booms. The project resulted in lessons-learned and the findings of appropriate methods necessary to implement a successful large-scale test.

2:00

2pNSa4. Objective data collection and analysis for the waveform and sonic boom perception and response program. Brian Cook, Joe Salamone (Acoust. and Vib., Gulfstream Aerosp., 3 Innovation Dr., M/S R-4P, Savannah, GA 31408, brian.cook@gulfstream.com), Chris Hobbs, and Juliet Page (Wyle, Arlington, VA)

The Waveform and Sonic boom Perception and Response (WSPR) program experiment was conducted in November 2011. Low-amplitude sonic booms were created by planned NASA F-18 supersonic flights executing a unique dive maneuver. The WSPR program was designed to simultaneously collect objective sonic boom acoustic data and subjective response data from residents in the Edwards Air Force Base residential community. Sonic Boom field kits were developed for the WSPR program consisting of a digital data acquisition system with networked nodes, deployable for extended periods of time. The Sonic Boom Unattended Data Acquisition System (SBUDAS) purposely developed for sonic boom community noise testing was deployed and details of the measurement system and all aspects of the objective data collection process are described. Data analysis during testing provided vital information to the flight planners for experimental execution. This paper also explains the post-experimental analysis of the objective data achieved by creation of a measurement data archive, predictions of sonic boom exposure at subject household locations, an automated algorithm to locate sonic booms within the recorded data and computation of a variety of indoor and outdoor metrics.

2:20

2pNSa5. A comparison of survey implementation methods. Peg Krecker, Carrie Koenig (TetraTech, Madison, WI), Clifton Wilmer, and Juliet A. Page (Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202, juliet.page@wyle.com)

As part of a pilot program to measure subjective response to low level sonic booms, 52 residents at Edwards Air Force Base were recruited to answer questions about their reactions to low-amplitude sonic booms. Over a two-week period, participants completed brief surveys (12 items) each time they heard a sonic boom and a short summary form at the end of each day. The study used three modes of survey administration—paper, Web, and Apple mobile device—to support analysis of the effectiveness of different approaches. Previous research on subjective response to sonic booms or other impulsive noise with similar measurement objectives has used in-person surveys, telephone surveys, or computer-assisted self-administered methods similar to a Web survey, but study designs prevented direct comparisons of the methods on the same sample of participants. We examine data quality across the three modes and the paper will present results on completion rates by survey mode, variation in completion rates over time, and differences in the timeliness of response for web and Apple participants. Qualitative interviews with a subset of participants yield further insights into each approach.

2:40

2pNSa6. Statistical analysis of community response to low amplitude sonic boom noise. Kathleen K. Hodgdon (Appl. Res. Lab., Penn State Univ., ARL North Atherton St., P.O. Box 30, State College, PA 16804-0030, kkh2@psu.edu), Juliet Page (Wyle, Arlington, VA), Trent Gaugler, Daisy Phillips, Durland Shumway, and James Rosenberger (Dept. of Stat., Penn State Univ., University Park, PA)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted a field study of subjective response to noise from multiple low-amplitude sonic booms. The team was led by Wyle and included researchers from Penn State, Tetra Tech and Gulfstream Aerospace Corp. The test exposed residents in the Edwards Air Force Base (EAFB) Housing area to two weeks of low-amplitude sonic booms while recording their responses via surveys. There were 52 participants divided across three response modes. The response instruments included Baseline Surveys, Single Event Surveys submitted each time a participant heard a boom, and Daily Surveys submitted at the end of each day. The analysis included assessments of single events and cumulative daily ratings of annoyance and categorical variables including loudness, interference, startle, vibration, and rattle. The WSPR daily annoyance data was analyzed by computing percent highly annoyed (%HA) and relating it to the cumulative noise exposure and by relating the subjective annoyance rating directly to the daily noise exposure. The WSPR design was established to cover the full range of noise exposures and annoyance factors so that sufficient data would be gathered to facilitate analyses of %HA and noise metrics. The statistical analyses examining these relationships will be presented.

3:00

2pNSa7. Relationships among near-real time and end-of-day judgments of the annoyance of sonic booms. Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St., Woodland Hills, CA 91367, sf@fidellassociates.com), Richard Horonjeff (Consultant in Acoust. and Noise Control, Boxborough, MA), and Linda Fidell (California State Univ., Emerita, Morro Bay, CA)

A recent social survey of the annoyance of low amplitude sonic booms included both prompt and delayed-response questions about the annoyance of sonic booms heard by respondents in the home over the course of two weeks. Interviews were conducted via smartphone with 49 voluntary test participants. Most of the prompt annoyance judgments were made within about a minute of notice of a sonic boom. The delayed response judgments were solicited in the evening, at a time of the respondent's choosing. Prior analyses showed that dosage-response relationships between the prevalence of high annoyance and sonic boom amplitude were well predicted by CTL analysis. The current analyses investigated how individual, within-day, prompt annoyance judgments were related to end-of-day judgments of the annoyance of sonic booms. Preliminary analyses suggest that end-of-day annoyance judgments are not simply a linear sum or averaging of the annoyances of individual sonic booms.

3:20

2pNSa8. Community response to low-amplitude sonic booms. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Sonic boom research conducted at NASA is oriented toward understanding the potential impact of sonic boom noise on communities from new low-boom supersonic aircraft designs. This research contributes to knowledge in key areas needed to support development of a new noise-based standard for supersonic aircraft certification. Partnerships with several industry, government, and academic institutions have enabled the execution of a pilot low boom community response test to develop and assess experimental methodologies, including sonic boom data acquisition, subjective data collection, and data analysis. Areas of additional research are identified and a prioritization of issues is performed to guide design of a potential follow-on pilot test. Lessons learned from these community response tests will facilitate future community testing with actual low-boom aircraft in communities not familiar with sonic booms.

Contributed Papers

3:40

2pNSa9. The impact of including diffraction when predicting the effect of listener environment on the perceived loudness of outdoor sonic booms. Amanda B. Lind and Victor Sparrow (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, State College, PA 16801, Amanda.Blair.Lind@gmail.com)

The human impact of sonic booms varies with listening environment. Given the incident sonic boom waveform, the specular field around a realistic geometry has been predicted via a c++ implementation of image source method (ISM) tailored to outdoor applications. This work explores the necessity of including the diffracted field when predicting time series and PLdB, both in and out of shadow zones. The impulsive nature of the excitation, and the sensitivity of the PLdB to temporal details, constrains appropriate diffraction modeling techniques to those capable of time domain accuracy. Uniform Theory of Diffraction (UTD) and Biot Tolstoy Medwin (BTM) approaches are considered. The benefits and challenges of each approach are explored, particularly with regards to scalability and bandwidth. The importance of accurately predicting diffraction in this application is evaluated through comparison with booms recorded around a building corresponding to the simulated geometry. [Work supported by the FAA/NASA/Transport Canada PARTNER Center Excellence and the Applied Research Laboratory. Experimental data courtesy of NASA.]

4:00

2pNSa10. Effect of room characteristics on perception of low-amplitude sonic booms heard indoors. Clothilde Giacomoni and Patricia Davies (Purdue Univ., 2911 Horizon Dr., Apt. 4, West Lafayette, IN 47906, cgiacomo@purdue.edu)

Supersonic flight over inhabited territories of the United States has been banned by the Federal Aviation Association. While research has been conducted to determine the effects of sonic booms on the general population when heard outdoors, little work has been done on people's perception of sonic booms as heard indoors. A sound's waveform will change in its transmission from outdoors to indoors due to several factors, one of which is the indoor acoustic environment. This can be changed using different room sizes, shapes or which materials are covering each surface (wall, ceiling, or floor). A subjective test, designed to determine which of these room characteristics has an effect on people's ratings of annoyance, has been completed. It was found that smaller rooms and square rooms are rated as more annoying than larger rooms or rooms with a corridor-like or rectangular shape, and that rooms with lower reverberation times were rated as less annoying than rooms with higher reverberation times.

4:20–5:20 Panel Discussion

TUESDAY AFTERNOON, 4 JUNE 2013 — 511CF, 12:55 P.M. TO 5:00 P.M.

Session 2pNSb

Noise and Architectural Acoustics: Soundscape and its Application

Brigitte Schulte-Fortkamp, Cochair
Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany

Bennett M. Brooks, Cochair
Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066

Chair's Introduction—12:55

Invited Papers

1:00

2pNSb1. Soundscape workshop report: Perception Lexicon. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com) and Brigitte Schulte-Fortkamp (Tech. Univ. Berlin, Berlin, Germany)

A workshop was held at the 164th Meeting of the ASA in Kansas City on methodology standardization for the advancement of the developing field of soundscape measurement, analysis, and design. The workshop focused on the terminology lexicon used for soundscape subject interviews. Interviews of local experts, residents, and other users and inhabitants of the sonic environment can yield insights into both personal reactions and universal observations. The specific terminology used in this process may significantly affect the outcomes. As the success of a new research or development project can depend on the lessons learned from previous projects, the standardization of interview techniques becomes increasingly important. Workshop participants were invited to develop a standardized lexicon of descriptors for field use in interview questionnaires. After a review of soundscape objectives and procedures, the participants reviewed key issues and assessed available lexicon term types. The group then proposed, developed, and prioritized terms which could

describe a soundscape as it is contextually perceived by an interview subject. Two key conditions were considered for each proposed lexicon term: would the term be commonly understood, and would the term describe a reaction to the soundscape of sufficiently notable intensity. The results of the workshop are presented.

1:20

2pNSb2. Characterizing the soundscape of tranquil urban spaces. Bert De Coensel, Michiel Boes, Damiano Oldoni, and Dick Botteldooren (Dept. of Information Technol., Ghent Univ., St.-Pietersnieuwstraat 41, Ghent B-9000, Belgium, bert.decoensel@intec.ugent.be)

Tranquil spaces provide restorative environments for urban residents and visitors and are therefore essential for health and quality of life. Tranquil spaces may be characterized through a combination of acoustical criteria, such as relatively low (percentile) sound levels and the relative absence of non-fitting sounds, and non-acoustical criteria, such as the presence of natural elements within the visual scene. Public urban parks and courtyards as well as private urban backyards are typically considered to be the most tranquil spots within a city. Current state-of-the-art in distributed measurement technology allows for long-term sound monitoring at these places. In this paper, the soundscape at a number of urban parks and backyards in the cities of Ghent and Antwerp is investigated through a detailed analysis of sound measurements performed over an extended period of time. An analysis of percentile sound levels, noise events and indicators for temporal and spectral structure is presented, and novel computational methods are applied to estimate the relative occurrence of sounds arising from various sources.

1:40

2pNSb3. What will be the influence of e-mobility on soundscape? Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath-Kohlscheid 52134, Germany, klaus.genuit@head-acoustics.de)

The increasing electrification of the powertrain after 125 years of continuous development of the internal combustion engine will not only lead to a sound pressure level reduction of vehicle exterior noises but to a complete change of sound quality. With this expected development road traffic noise affected persons hope for quiet cities and a better quality of life. The creation and successful preservation of quiet zones in cities and to avoid harmful effects of noise exposure are special focuses in European noise policy. However, different surveys have shown the increased risk of accidents for pedestrians and cyclists with respect to collisions with quiet vehicles, which caused a lively discussion about acoustical warning systems for the prevention of crashes. But it is obvious, that major conflicts between quietness and safety arise. Consequently, to address this issue, on the one hand, sustainable concepts must be developed for the successful avoidance of accidents and on the other hand the general traffic noise must be minimized. The sound of electric vehicles will influence in a significant way our soundscape at places and cities in the future. This is a special challenge for psychoacoustics to provide helpful contribution besides the A-weighted sound pressure level.

2:00

2pNSb4. Plan for research to better understand visitor response to the soundscape in national parks. James H. Boyle (Univ. HS, Univ. of Illinois, 3805 Deerfield Dr., Champaign, IL 61822, jhhboyle@gmail.com) and Paul D. Schomer (Schomer and Assoc., Inc., Champaign, IL)

Since 91% of national park visitors come to a park, at least in part, for the natural sounds, analysis of the perceptions visitors have of sound in national parks is extremely significant. In the summer of 2011, several pilot surveys were tested at Rocky Mountain National Park in Colorado for their abilities to provide an accurate reflection of visitors' perceptions of the soundscape. The 2011 surveys were modified in various ways to create the current 2013 survey, which uses "pleasantness" of park sound as the main visitor response metric. The testing plan for the 2013 survey is non-labor intensive for respondents and researchers; rather than closely monitoring activities of respondents, and having them closely monitor the sounds they hear, this survey requires respondents to complete a short questionnaire every 30 to 60 min and an end-of-hike survey. Much like how day-night sound level (DNL) has traditionally been predicted for a cluster of houses around airports or highways, global sound measurements conducted along the trail will be correlated with visitor survey results in order to develop the means to predict how an average park visitor will respond to the soundscape.

2:20

2pNSb5. When do we judge sounds? Relevant everyday situations for the estimation of ecological validity of indoor soundscape experiments. Jochen Steffens (FH Duesseldorf, Josef-Gockeln-Strasse 9, Duesseldorf 40476, Germany, jochen.steffens@fh-duesseldorf.de)

This paper introduces a model of indoor soundscape evaluation, which is based on the results of listening experiments on household appliances. The model reveals, among other things, permanently changing action and attention processes and learning effects which occur in everyday settings. However, what are the relevant situations which we want to reconstruct in our experiments? When do people perceive and evaluate sounds in their everyday life like they do under test conditions? This theoretical knowledge is essential for the estimation of ecological validity of soundscape experiments in general. Hence, this contribution deals with approaches to determine meaningful real-life situations. The interviews held in the course of the studies show that many subjects construct extremely critical context situations while they are assessing sounds. These contexts are often not representative for these peoples' reality. Many participants, in fact, state that they no longer consciously perceive and evaluate the sounds of the appliances in their personal life. Decreasing attention over time can lead to a great influence of the first impression on subsequent perceptions and cognitions (primacy effect). Furthermore, the effect of memory processes on retrospective sound evaluations will be discussed.

2:40

2pNSb6. The practical SoundLab for architects: Sound parameters as a design tool. Juergen Bauer (Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Co Waterford, Ireland, jbauer@wit.ie)

The experience gained from exploring acoustics with architectural students in Waterford Institute of Technology (Ireland) suggests that a studio situation can be exploited and used as a case study for young designers to become familiar with basic sound phenomena. From phase 1 of previous research, the student "audiovisual design workshop" concluded that class rooms that perform poorly

acoustically can help to strengthen the students' understanding of the sonic environment, in a way that the students were exposed day after day to high reverberation, low speech intelligibility, and the "Lombard" effect in their own studio. However, becoming aware of poor acoustics necessitates a creative follow-up: how to design and create good acoustics, or how to develop the environment as a "soundscape"? Consequently, phase 2 of our audiovisual design workshop introduced the practical sound lab: our students were required to design and to build acoustic panels for their own studio environment. These prototypes would be analyzed as part of an Acoustic Laboratory, with specialist advise from an Acoustic Consultant. The paper summarizes the findings from the audiovisual design laboratory. It also reflects on how to compile international case studies of buildings with good acoustics that can be used as references for designing architects.

3:00–3:20 Break

Contributed Papers

3:20

2pNSb7. Evaluation of noise climate in a campus environment using geospatial technology. Rajasekar Elangovan (Ctr. for Excellence and Futuristic Developments, Larsen & Toubro Construction, L&T ECC, Manapakkam, Chennai, TamilNadu 600089, India, erajas@gmail.com), Pari YadavaMudaliar (GeoSpatial Technol., EDRC, Larsen & Toubro Construction, Chennai, India), Venkateswaran Rajan, Khaleel Elur Rahaman Anser Basha (Ctr. for Excellence and Futuristic Developments, Larsen & Toubro Construction, Chennai, India), and Ravi Muttavarapu (GeoSpatial Technol., EDRC, Larsen & Toubro Construction, Chennai, India)

This study focuses on experimental evaluation of outdoor noise climate in an office campus, which is spread across 27 acres and investigation of its impact on the adjoining office buildings. For this purpose, a base map of the campus was prepared from the recent high spatial resolution satellite image using ESRI Arc/GIS and the same was used for planning appropriate locations for capturing noise levels and its spectral characteristics. Mapping grade Trimble GPS was used to stake out the measurement locations and noise levels were recorded using 01 dB solo and Norsonic type 118 sound level meters. Captured noise levels were plotted in GIS environment and appropriate spatial interpolation was carried out in order to give a continuous graphical representation of sound levels. A wide variation of noise levels was observed across the campus with LAeq ranging from 50 dB(A) to 80 dB(A). Low frequency noise was found to be predominant compared to mid and high frequency noise. Major noise sources and the propagation pattern were determined through the mapping. The data thus obtained is used to investigate the noise impact on the office buildings. The measurements were also accompanied by a subjective evaluation of the outdoor noise annoyance.

3:40

2pNSb8. On the noise generated by park visitors along hiking trails. Lucas Newman-Johnson (Univ. HS, Univ. of Illinois, 808 w. White St., Champaign, IL 61821, newmanj1@uni.illinois.edu) and Paul D. Schomer (Schomer and Assoc., Champaign, IL)

In assessing the sound environment of a park, visitor noise along trails may be important because of its affect upon: (1) the un-altered park environment, (2) wildlife, and (3) other park visitors. Just as noise in residential communities has been correlated with population density we set out to see if the noise level along trails would correlate with visitor density. In the summer of 2011, measurements were made using two sites, one on each of two trails. These measurements included one second Leq measurements at an array of four trailside microphones, and recording of the number of park visitors entering the measurement zone during each minute. Examination of the data revealed little correlation between a 5 min measurement of the Leq and density of trail visitors. The problem was that the smallest time increment in which we could accurately portray the number of park visitors was about 5 min, using the one minute totals, and in a 5 min period visitor noise would rarely equal or exceed the measurement ambient. Thus, no relation between visitor density and trailside noise could be developed. Additional analysis was done with data to better understand the limiting factors to the measurement.

4:00

2pNSb9. Investigating soundscape affordances through activity appropriateness. Frederik L. Nielbo (Ctr. for Semiotics, Aarhus Univ., Jens Chr. Skous Vej 2, Bldg. 1485, Rm. 525, Aarhus DK-2200, Denmark, norfln@hum.au.dk), Daniel Steele (Ctr. for Interdisciplinary Res. in Music Media and Technol., McGill Univ., Montréal, QC, Canada), and Catherine Guastavino (School of Information Studies, McGill Univ., Montréal, QC, Canada)

Central to the concept of soundscape is the understanding of the acoustic environment in context. Previous research indicates that people understand soundscapes through their potential for activities. One way to look at activities is through the concept of affordances—defined as the actionable properties of an object. In this study, the object is a location and time in the city. Fifteen participants listened to stereo recordings of eight outdoor sites in Paris and Montreal. In each trial, they evaluated on a continuous scale how appropriate the soundscapes were for a given activity. Four activities were considered and presented in random order: studying for an exam, meeting up with a friend, riding a bike and relaxing. Participants justified their ratings in free-format comments. A 8(Soundscapes) x 4(Activities) factorial ANOVAs revealed significant effects of Soundscape and Activities and Soundscape*Activities on appropriateness ratings. Certain soundscapes were found to accommodate specific activities only while others were found to potentially accommodate all activities or none (prominent mechanical/traffic noise). We also analyzed comments to further understand how participants envision utilizing the soundscape/environment and attribute meanings to the various sounds present.

4:20–5:00 Panel Discussion

Session 2pPA

Physical Acoustics: Material Characterization

Noureddine Atalla, Cochair
GAUS Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

Nico Declercq, Cochair
Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France

Contributed Papers

1:00

2pPA1. Probing acoustics of liquid foams by optical diffusive wave spectroscopy. Benjamin Dollet, Marion Erpelding, Reine-Marie Guillermic, Caitlin Sample, Juliette Pierre, Arnaud Saint-Jalmes, and Jérôme Crassous (Institut de Physique de Rennes, CNRS/Université Rennes 1, Campus Beaulieu, Bâtiment 11A, Rennes 35042, France, benjamin.dollet@univ-rennes1.fr)

Sound propagation through liquid foams, which are dispersions of gas bubbles in a continuous liquid phase, is not well known yet. To characterize foam acoustics at the local scale, we have studied the effect of an external acoustic wave on bubble displacements inside an aqueous foam. We quantify these displacements by using a technique based on optical diffusive wave spectroscopy, that we specially developed to resolve tiny deformations in materials. Bubble displacements induce a modulation on the photon correlation curve. Measurements for various sound frequencies and amplitudes are interpreted using a light diffusion model. It allows us to unravel a nontrivial acoustic displacement profile inside the foam; in particular, we find that the acoustic wave creates a localized shear in the vicinity of the solid walls holding the foam, as a consequence of inertial contributions. This study of how bubbles "dance" inside a foam as a response to sound turns out to provide new insights on foam acoustics and sound transmission into a foam, foam deformation at high frequencies, and analysis of light scattering data in samples undergoing nonhomogeneous deformations.

1:20

2pPA2. Reduction of ultrasonic multiple scattering applied to flaw detection with array probes in polycrystalline materials. Sharfine Shahjahan (Site des Renardieres, EDF R&D, Moret-sur-Loing, France), Alexandre Aubry (Institut Langevin - ESPCI ParisTech & CNRS, Universite Paris Diderot, 1 rue Jussieu, Paris 75005, France), Fabienne Rupin, Bertrand Chassignole (Site des Renardieres, EDF R&D, Moret-sur-Loing, France), and Arnaud Derode (Institut Langevin - ESPCI ParisTech & CNRS, Universite Paris Diderot, Paris, France, arnaud.derode@espci.fr)

Flaw detection using ultrasonic evaluation of coarse-grain steels is perturbed by a high structural noise due to scattering. This leads to a decrease of the detection capabilities, particularly at high frequencies and large depths for which multiple scattering dominates. Recent academic studies have shown that the contribution of multiple scattering could be dramatically reduced. These results were obtained on a model random medium made of parallel steel rods immersed in water. The ability to detect a target could be significantly increased using a specific filtering method, based on the full matrix capture (F.M.C.) combined with a smart post-treatment based on random matrix theory, in supplement with the DORT method (i.e., decomposition of the time-reversal operator). Here, the same technique to separate simple and multiple scattering contributions is now applied to a real material. Experimental results were obtained on a nickel-based alloy (Inconel600®) with a thermically induced coarse grain structure and manufactured flaws (side drilled holes) at different depths. The experimental set-up used a multi-element ultrasonic array. Results are presented and compared to other detection techniques, at various depths and frequencies. Despite a dominant multiple scattering noise, a significant improvement of the detection performances is observed.

1:40

2pPA3. Negative and density-near-zero acoustic metamaterials based on quasi-two-dimensional phononic crystals. Victor Manuel Garcia-Chocano, Rogelio Graciá-Salgado, Daniel Torrent, and José Sánchez-Dehesa (Dept. of Electron. Eng., Polytechnic Univ. of Valencia, C/Camino de Vera S/N, Departamento de Ingeniería Electrónica, Valencia 46022, Spain, vicgarch@upvnet.upv.es)

A phononic crystal consisting of an array of cylindrical boreholes in a two dimensional waveguide has been fabricated and characterized. Reflectance and transmittance have been measured in a slab with seven layers of scatterers. The acoustic bands as well as the effective parameters have been extracted from experimental data, showing that the proposed structure behaves as an acoustic metamaterial with negative bulk modulus. In addition it is shown that the inclusion of anisotropic effects through angular-dependent structures inside the boreholes leads to metamaterials with double negative parameters. This feature allows the observation of interesting phenomena such as an acoustic tunneling, which has been predicted through full wave simulations in a density-near-zero metamaterial.

2:00

2pPA4. Impulse response of a medium in a three layered media. Ambika Bhatta (Elec. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu), Charles Thompson, Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), and Vineet Mehta (MIT, Lincoln Lab., Lexington, MA)

This paper investigates the point source response of a layered medium separating two semi-infinite impedance medium. A detailed numerical method to evaluate Laplace transform of the reflection coefficient for each boundary is presented. A description of the branch integral dominantly contributing to the reflection coefficient in Laplace domain for single and multiple reflections is given. The analysis utilizes the image theory to evaluate the order of the reflection and complex pressure amplitude at each image source location. The impulse response is also evaluated numerically from the Green's function obtained from the reflection coefficient and complex pressure due to image sources. Fresnel's plane wave reflection coefficient validates the solution when the height of layer containing the source and the observation position is relatively large. The obtained expression for reflection coefficient for large wave number and observation position is verified numerically.

2:20

2pPA5. Experimental demonstration of a three-dimensional acoustic cloak based on a cancelation effect. Jose Sanchez-Dehesa, Victor M. García-Chocano, Alfonso Climente, Francisco Cervera (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Camino de vera s.n., Edificio 7F, Valencia, Valencia ES-46022, Spain, jsdehesa@upvnet.upv.es), Vicente Gomez-Lozano (Centro de Tecnologías Físicas, Universitat Politecnica de Valencia, Valencia, Spain), Lorenzo Sanchis (UMDO (Unidad Asociada al CSIC-IMM), Universidad de Valencia, Valencia, Spain), Rafael LLopis-Pontiveros, and Martínez-Pastor Juan (UMDO (Unidad Asociada al CSIC-IMM), Universidad de Valencia, Valencia, Spain)

A three-dimensional acoustic cloak has been designed, fabricated, and experimentally characterized. The cloak is made of 60 concentric tori acoustically rigid that surround a sphere of radius 4 cm, which is considered as the cloaked object. The major radii and positions of the tori along the symmetry axis are determined by a cancelation condition; i.e., the scattering cross section by the sphere and the tori must be zero. An optimization algorithm that combines a genetic algorithm and simulated annealing is employed to satisfy such a condition. The operational frequency of the one-directional cloak is 8.67 kHz with a bandwidth of about 120 Hz.

2:40

2pPA6. Ice thickness determination with audio sound. Anurupa Shaw (Georgia Tech Lorraine, 133 Rue Du Fort de Queuleu, D107, Residence Lafayette, Metz 57070, France, shaw.anurupa@gmail.com) and Nico F. Declercq (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., Lab. for Ultrasonic Destructive Evaluation "LUNE", Georgia Tech-CNRS UMI2958, Georgia Tech Lorraine, Metz-Technopole, France)

The objective of this study is to determine the thickness of Ice by analyzing the sound spectrum generated by dispersion of lamb waves propagating in ice. In winters when the lakes and rivers freeze, it is important to know the thickness of the ice layer before one intends to walk on it. When we throw a stone on the ice layer, we can hear a fluting noise. This is recorded for different thicknesses of ice and the sound spectrum is compared with the results simulated using a parameterized model. This model is created using a combination of plane waves for different incident angles and frequencies to generate dispersion curves for different thicknesses of ice. The frequencies of the reflected sound are then compared with the frequencies of musical notes in order to assign different musical notes to different thicknesses of ice.

3:00

2pPA7. Shaving foam: A complex system for acoustic wave propagation. Juliette Pierre (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Université Rennes 1 - CNRS UMR 6251 263 av. Général Leclerc, Rennes 35042 Cedex, France, juliet.pierre@gmail.com), Valentin Leroy (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Paris, France), Arnaud Saint-Jalmes, Benjamin Dollet, Imen Ben Salem, Jérôme Crassous, Reine-Marie Guillermic (Institut de Physique de Rennes, UMR CNRS 6251 - University Rennes 1, Rennes, France), Wiebke Drenckhan (Laboratoire de Physique du Solide, UMR CNRS 8502 - Univ. Paris-Sud, Orsay, France), and Florence Elias (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Paris, France)

While liquid foams have applications in an increasing number of industrial areas (food, cosmetic, or petroleum industry), it remains difficult to non-invasively probe their structure and/or composition. Since the propagation of acoustic waves is very sensitive to parameters such that the liquid fraction, the bubble size distribution, or even the nature of the liquid phase, acoustic spectroscopy could be a very powerful tool to determine the structure and/or composition of liquid foams. In this context, we present an investigation of the acoustic properties of a useful and common foam, often considered as a model system: shaving foam. Phase velocity and attenuation of acoustic waves in a commercial shaving foam (Gillette) were measured over a broad frequency range (0.5 to 600 kHz), using four different experimental setups: an impedance tube (0.5–6 kHz), an acousto-optic setup based on diffusive wave spectroscopy (1–10 kHz), and two transmission setups with narrow-band (40 kHz) and broad-band (60–600 kHz) transducers. We present the results and discuss the advantages and shortcomings of each setup in terms of a potential spectroscopy technique.

3:20

2pPA8. Lamb modes and acoustic microscopy for the characterization of bonded structures. Alaoui Ismaili Naima, De Mello Da Silva Camilla (Institut Electronique du Sud (IES), UMR CNRS 5214, University of Montpellier 2, Place Eugène Bataillon, Montpellier 34095, France, naima.alaoui-ismaili@insa-lyon.fr), Ech-Cherif El-Kettani Mounsif (Laboratoire Ondes et Milieux Complexes (LOMC), UMR CNRS 6294, Univ. of Le Havre, Le Havre, France), Despaux Gilles (Institut Electronique du Sud (IES), UMR CNRS 5214, Univ. of Montpellier 2, Montpellier, France), Rousseau Martine (Institut Jean Le Rond d'Alembert UMR CNRS 7190, Univ. of Pierre et Marie Curie, Paris, France), and Izbicki Jean-Louis (Laboratoire Ondes et Milieux Complexes (LOMC), UMR CNRS 6294, Univ. of Le Havre, Le Havre, France)

This paper is a contribution to the evaluation of the bonding by means of guided waves, when the quality of the glue is degraded by addition of grease. Experimental dispersion curves are compared to the theoretical ones obtained either from a perfect welded tri-layer model or from a second model using delaminated conditions. It is shown that the second model is more convenient when the quality of the glue is highly degraded. On another hand, the parameters of Jones are extracted from experimental data by inverse problem. It is shown that the values of these parameters strongly diminish and the variation is of logarithmic type when the quality of the glue is progressively degraded. Acoustic microscopy is also performed, and it provides images of the inner structure giving a qualitative explanation of the values of the parameters of Jones.

3:40

2pPA9. Experimental measurements of the coherent field resulting from the interaction of an ultrasonic shock wave with a multiple scattering medium. Nicolas Viard, Bruno Gianmarinaro, Arnaud Derode, and Christophe Barrière (Institut Langevin, 1, rue Jussieu, Paris 75005, France, nicolas.viard@espci.fr)

Whereas multiple scattering and shock wave formation are known to be antagonistic phenomena, this work concentrates on the interaction of an ultrasonic shock wave with a random multiple scattering medium. The shock wave is generated by long distance propagation of a short pulse (four periods at a 3.5 MHz central frequency) in water before it encounters the scattering medium (a slab-shaped random set of parallel metallic rods). Transmitted waves are recorded over hundreds of positions along the lateral dimension of the slab to estimate the ensemble-averaged transmitted field $\langle \Phi(t) \rangle$, also known as the coherent wave. Experiments are repeated for different thicknesses L of the slab and different emission amplitudes. The elastic mean free path l_e (i.e., the typical distance for the decreasing of the coherent intensity $|\langle \Phi(t) \rangle|^2$ due to scattering) is determined as well as the harmonic rate of the averaged transmitted wave. Experimental results are discussed and compared to the linear case.

4:00

2pPA10. Frequency-resolved measurements of the diffusion constant for ultrasonic waves in resonant multiple scattering media. Nicolas Viard and Arnaud Derode (Institut Langevin, 1, rue Jussieu, Paris 75005, France, nicolas.viard@espci.fr)

Experimental measurements of the diffusion constant for ultrasonic waves (around 3 MHz) propagating in water through a random set of scatterers (parallel metallic rods arranged as a slab) are presented. The slab thickness is around 10 times the transport mean free path. Transmitted waves are recorded over hundreds of emitting/receiving positions in order to estimate the ensemble-averaged transmitted intensity $\langle I(x,t) \rangle$. Focused beamforming is performed on both faces of the sample in order to mimic a set of point-like sources and receivers. In theory, under the diffusion approximation, the ratio of the off-axis intensity $\langle I(x,t) \rangle$ to the on-axis intensity $\langle I(0,t) \rangle$ shows a simple gaussian dependence on the lateral dimension x, independently from absorption or boundary conditions. This yields a simple way to estimate the diffusion constant D and therefore characterize the scattering medium. Based on that method, broadband as well as frequency-resolved measurements of the diffusion constant are presented in controllable model media, such as these forests of steel rods. Experimental results and difficulties for measuring a reliable value for D on a real sample are discussed.

4:20

2pPA11. Wavetrain-long waves interaction in a non-homogeneous, non-stationary medium. Alexander Voronovich (Physical Sci. Div., NOAA/ESRL, 325 Broadway, Boulder, CO 80305, alexander.voronovich@noaa.gov)

A weakly nonlinear wavetrain of the acoustic waves apart from higher harmonics generates also a low-frequency, large-scale motion which in turn affects both the wavetrain itself and other wavetrains, thus leading to an effective wavetrains interaction. A closed set of equations describing interaction of a wavetrain with inhomogeneous medium including a recoil effect on the background motion is derived using Hamiltonian form of the equations of motion (viscosity is neglected). Corresponding energy and momentum conservation equations are derived which account for the energy and pseudo-momentum exchange between the wavetrain and a background, large-scale motion (the latter includes the nonlinearly generated component due to the wavetrains). A large-scale motion due to a single wavetrain propagating in an initially homogeneous medium is calculated. Far from the wavetrain nonlinearly generated large-scale current is confined by an expanding sphere with the velocity aligned with the wavetrain propagation direction; the velocity appears to be constant in the planes perpendicular to this direction. It is demonstrated that interaction between acoustic wavetrains lead to their effective repulsion which in a complex way depends on mutual location and velocities of the wavetrains. Generalization of the approach to the case of integral gravity waves will be also briefly mentioned.

4:40

2pPA12. Gaussian closure technique for Bouc's hysteretic model under white noise excitation. Holger Waubke (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, holger.waubke@oeaw.ac.at)

Bouc developed a hysteretic model for materials like rubber under dynamic excitation. The response of a hysteretic system under white noise excitation is normally estimated by means of the statistical linearization or a related method. Disadvantages of this method are the assumption of a Gaussian nature of the random distributions, the high computational efforts caused by the iterations needed, and the instability of the iteration in certain parameter regions. Using the assumption of Gaussian random distributions, the Gaussian closure technique can be applied. Analytic solutions of the integrals occurring in this approximation were found and are presented. These solution allow for an explicit time step procedure for the random moments in the transient case. For the stationary case, a fast and stable iteration about a set of non linear equations is needed. Both procedures allow to calculate the moments in a fast manner and allow to solve problems with more than one degree of freedom with limited computational efforts.

5:00

2pPA13. Numerical model of a causal and fractional all-frequency wave equation for lossy media. Margaret Wismer (Bloomsburg, 400 E Second St., Bloomsburg, PA 17815, mwismer@bloomu.edu)

A numerical algorithm, to simulate a lossy acoustic wave equation with fractional time derivative terms, is presented. The inclusion of fractional derivatives yields a causal acoustic wave equation which can model and predict power law attenuation for which the level of absorption is proportional to frequency raised to a non-integer power. The fractional Zener wave equation is derived from the fractional Zener stress-strain constitutive relation and contains two absorption terms. One term which includes the Laplacian, similar to the traditional wave equation in viscous fluids, has a coefficient, which is proportional to a relaxation time. The other term is a fractional time derivative, higher than second order, and is proportional to the creep or retardation time. Both relaxation and retardation parameters will affect the frequency behavior of the attenuation. The inclusion of two terms enables the modeling of power-law attenuation in all frequency ranges. It results in more stable numerical simulations as both the overall mass and stiffness matrices of the finite element algorithm are changed according to the level of absorption. Results show the animation of diffraction patterns of focused and planar acoustic waves by inclusions with different values for the two different parameters, retardation time and relaxation time.

TUESDAY AFTERNOON, 4 JUNE 2013 514ABC, 12:55 P.M. TO 4:20 P.M.

Session 2pPPa

Psychological and Physiological Acoustics: Celebrating a "Long" Career: Explorations of Auditory Physiology and Psychoacoustics

Jungmee Lee, Cochair
Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208

Elizabeth A. Strickland, Cochair
Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—12:55

Invited Papers

1:00

2pPPa1. Glenis Long's contribution to animal psychoacoustics. Richard Fay (MBL, 179 Woods Hole Rd., Falmouth, MA 02540, rfay@luc.edu)

Glenis and I were psychology graduate students together in 1970–1971. She was interested in auditory psychophysics, but soon developed an interest in comparative psychophysics which she pursued in a post-doc and a faculty position in Germany, where she obtained a behavioral audiogram for the horseshoe bat and later studied masking in the same species. This audiogram was one of the very first psychophysical investigations of hearing in any bat and the very first for the horseshoe bat. It confirmed a rather complex frequency response function with sensitivity peaks at about 20 kHz and 60 kHz, and a very sharp peak at about 80 kHz. The masked thresholds indicated rather normal critical masking ratios (CR) except in a narrow region at about 80 kHz where the CRs are up to 15 dB lower than expected, suggesting a critical band of about 300 Hz at 80 kHz that is presumably used in Doppler-shift compensation. In 1981–1983, Glenis went on to investigate frequency and rate modulation discrimination in the chinchilla, for the first time, and to study tone-on-tone masking in the chinchilla. Since these early works, Glenis has maintained her comparative interests with studies on birds, kangaroo rats, frogs, and fleas on cats.

1:20

2pPPa2. Understanding subtle changes in auditory function with otoacoustic emissions. Linda J. Hood (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, MCE South, 8310, Nashville, TN 37232, linda.j.hood@vanderbilt.edu), Shanda Brashears (A. I. du-Pont Hospital for Children, Wilmington, DE), Glenis Long (The Grad. Ctr., City Univ. of New York, New York, NY), and Carrick Talmadge (Univ. of Mississippi, Oxford, MS)

Otoacoustic emissions (OAEs), a sensitive measure of cochlear processing, may be altered by subtle changes in auditory function that are not measurable by usual clinical methods. We studied auditory function in carriers of genetic mutations related to recessive hereditary hearing loss where we hypothesized that carrying a single mutation copy may compromise auditory function and be reflected when sensitive assays are used. Parents and siblings who were confirmed carriers of recessive mutations associated with GJB2 (connexin 26) or Usher syndrome, as well as obligate carriers of mutations related to recessive hearing loss of unknown genetic origin, were compared to age and gender matched control subjects. All participants had normal pure tone and middle ear responses. Metrics included transient OAEs, distortion product OAEs, and OAE fine structure. DPOAE fine structure was specifically explored based on the ability to isolate components that could be differentially affected by genetic mutations. The results support the hypothesis that carriers of gene mutations related to hearing loss display subtle auditory abnormalities that can be observed in OAEs. These findings will be related to other studies of subtle changes in OAEs in disorders affecting auditory function. [Work supported by NIH NIDCD R01-DC03679 and VU Development Funds.]

1:40

2pPPa3. Brain activity and perception of gaze-modulated tinnitus. Pim Van Dijk, Margriet J. Van Gendt, Kris Boyen, Emile De Kleine (Otorhinolaryngology, Univ. Med. Ctr. Groningen, P.O. Box 30001, Groningen 9700 RB, Netherlands, p.van.dijk@umcg.nl), and Dave R. Langers (NIHR Nottingham Hearing Biomed. Res. Unit, Univ. of Nottingham, Nottingham, United Kingdom)

We studied the correspondence between brain activity and tinnitus in subjects with gaze-modulated tinnitus. These subjects are able to modulate their tinnitus by peripheral gaze of the eyes. This is a rare form of tinnitus that primarily occurs in subjects that underwent acoustic schwannoma surgery. The voluntary control of the tinnitus allows for a controlled experiment to study the perceptual characteristics of tinnitus and the corresponding brain activity as assessed by functional MRI. Eighteen subjects with gaze-modulated tinnitus participated in the study. The effect of gaze on tinnitus was diverse. Most commonly, the largest effect on tinnitus was observed for horizontal gaze toward the surgery side. When the loudness of tinnitus changed, it was usually an increase. In addition, changes of the pitch and apparent bandwidth of the tinnitus were reported. Peripheral gaze corresponded to increase of activity in the cochlear nucleus and inferior colliculus, a decrease of activity in the medial geniculate body, and a reduction of deactivation in the auditory cortex. The inhibition of the medial geniculate body in the thalamus contrasts with the excitation that is typically observed in response to external sound stimuli. It suggests that abnormal functioning of the thalamus plays a role in tinnitus.

2:00

2pPPa4. Improving the usability of the distortion product otoacoustic emisssions-sweep method: An alternative artifact rejection and noise-floor estimation. Manfred Mauermann (Med. Phys., Univ. of Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26111, Germany, manfred.mauermann@uni-oldenburg.de)

The DPOAE-sweep method combined with a specific least-squares-fit (LSF) analysis provides a fast method to measure and analyze distortion product otoacoustic emissions (DPOAE) with a high frequency resolution. In studies using this technique the noise reduction, artifact rejection and noise estimation are typically realized in a “classical” way, i.e., as temporal averaging with preceding elimination of time epochs exceeding an artifact threshold level and a noise estimation based upon the analysis of the difference of two buffers of epoch averages. However, the choice of an artifact threshold is arbitrary to some extent and different choices can lead to differences in the estimation of DPOAE levels. The two-buffer technique is ambiguous as well since a different grouping of the epochs into the buffers leads to rather different noise estimates. Therefore, the present study proposes an alternative approach, which provides unique noise estimators for a given set of data, a robust artifact rejection without the need to select an arbitrary rejection threshold, as well as estimators including confidence intervals for the DPOAE levels and phases. The “classical” and suggested estimators for DPOAE levels, DPOAE phases, and noise levels are compared based on Monte Carlo simulations and real measured data sets.

2:20–2:40 Break

2:40

2pPPa5. The relevance of otoacoustic emission fine structure. Sumitrajit Dhar (Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, s-dhar@northwestern.edu)

The quasi-periodic fluctuations in otoacoustic emission (OAE) amplitude and phase, known as fine structure, provide critical insight into the mechanisms of OAE generation and propagation. Whether fine structure is relevant from the perspective of hearing health remains an open question. Starting with work done under the tutelage of Professor Long, our group has conducted a decade of work investigating the clinical relevance of fine structure in general, and distortion product (DP) OAE components in particular. These studies cover a significant portion of the human lifespan (0 to ~80 yr) and the collective results provide insights into the avenues of application that hold promise. Drawing from the results of several studies conducted in our laboratory and those of collaborators and colleagues we will make the case for focused clinical application of the DPOAE components that constitute fine structure. These constituent elements appear to be ideal for monitoring of cochlear status and expose physiological changes prior to their evidence in other measures of

auditory function. However, harnessing the information encapsulated in fine structure directly is not without challenge and the examination of the constituent elements of fine structure may prove more profitable for clinical applications.

3:00

2pPPa6. Noise-induced hearing loss and strategies for its prevention in the New Zealand population: The Kiwi connection. Peter R. Thorne, Gavin Coad, Ravi Reddy, and David Welch (Audiology, Univ. of Auckland, School of Population Health, Private Bag 92019, Auckland 1142, New Zealand, pr.thorne@auckland.ac.nz)

Celebrating Glenis Long's outstanding contribution to auditory physiology and psychoacoustics, this presentation covers the New Zealand connection and particularly her involvement in our research program into the monitoring and prevention of noise-induced hearing loss (NIHL) in the country. A large multidisciplinary project is being undertaken to investigate the nature of occupational hearing loss in New Zealand and establish a national approach to prevent NIHL. This includes estimates of NIHL prevalence and the design and evaluation of education and prevention programs to reduce the impact of noise. Using a modeling approach, we have estimated that NIHL contributes to 17–25% of cases of hearing impairment in New Zealand and is therefore a significant modifiable risk factor. A key component of our project is monitoring of noise injury and we have also studied distortion product otoacoustic emissions (DPOAE) as a measure of early injury. To assess DPOAEs as a measure of injury, we recorded them using swept pure tones and extracted DPOAE components using a least-squares fit approach in noise and non-noise exposed individuals. OAE findings were compared with measures of auditory function. We found that the generator component correlated more strongly with auditory threshold and thus may be a better physiological index of noise injury. Overall, these findings have informed a national strategy involving government and community agencies to mitigate the effects of noise.

3:20

2pPPa7. Demonstration of distributed distortion-product otoacoustic emission components using onset-latency techniques. Brenda L. Lonsbury-Martin (Otolaryngology, Loma Linda Univ. Med. Ctr., Res. Service (151), 11201 Benton St., Loma Linda, CA 92357, blonsbury-martin@llu.edu), Glen K. Martin, and Bart B. Stagner (Res. Service, VA Loma Linda Healthcare System, Loma Linda, CA)

An oversimplified notion is that DPOAEs originate from a restricted region on the basilar membrane (BM). In actuality, DPOAEs are a distributed process involving the interaction of many wavelets, most likely generated over a broad region at, and basal to the overlap place of the primary tones. In the present study, DPOAEs were measured in rabbits as time waveforms by using phase rotation to cancel all components in the final average, except the 2f1-f2 DPOAE. At times, f2 was turned off for 6 ms producing a gap so that the DPOAE was no longer generated. These procedures allowed the DPOAE onset as well as the decay during the gap to be observed in the time domain. Results showed that complexities emerged near the onset of the DPOAE time waveform as the f2/f1 ratio decreased, and at the beginning of the gap when f2 was turned off. Such complexities were unaffected by interference tones (ITs) near the DPOAE. However, these complexities were removed by ITs presented above f2, which can be explained by the interactions of distributed DPOAE components with different phase relationships.

3:40

2pPPa8. Distortion-product otoacoustic emission generator and reflection components in newborns, infants, and adults. Beth Prieve (Commun. Sci. and Disord., Syracuse Univ., 805 S. Crouse Ave., Syracuse, NY 13244, baprieve@syr.edu), Glenis Long (Speech-Lang.-Hearing Program, City Univ. of New York Grad. Ctr., New York, NY), and Carrick Talmadge (National Ctr. for Physical Acoust., University, MS)

Glenis Long and colleagues (Talmadge *et al.*, J. Acoust. Soc. Am. **105**, 275) were among the first to model and describe the characteristics of distortion-product otoacoustic emissions (DPOAEs) using two sub-components from different, cochlear sources. It is now accepted that there is a generator component that results from inter-modulation distortion created by nonlinearity in the outer hair cell near the f2 place. The reflection component predominantly arises from coherent reflection near the characteristic place corresponding to the frequency of the distortion product. Because the two components are generated through different mechanisms, it has been hypothesized that they may be differentially affected by human development. The goal of this presentation is to discuss ongoing research of DPOAE components in newborns, infants and adults. Analysis of the components indicates that the relationship between growth rates for generator and reflection components are significantly different between infants and adults. Furthermore, the phase functions for both components are different among groups. Possible sources for these differences will be discussed. [Funded by the March of Dimes Birth Defects Foundation].

4:00–4:20 Panel Discussion

Session 2pPPb

Psychological and Physiological Acoustics: Speech, Attention, and Impairment (Poster Session)

Jayaganesh Swaminathan, Chair
Boston Univ., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215

Contributed Papers

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

2pPPb1. Factors affecting frequency discrimination in school-aged children and adults. Crystal N. Taylor (Allied Health Sci., UNC Chapel Hill, Dept. of Allied Health Sci., CB 7190, Chapel Hill, NC 27599, Crystal_Taylor@med.unc.edu), Emily Buss (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, Chapel Hill, NC), and Lori J. Leibold (Allied Health Sci., UNC Chapel Hill, Chapel Hill, NC)

Auditory frequency discrimination is a basic ability that may limit the maturation of speech and language skills in some listeners. Despite its importance, the factors affecting frequency discrimination in school-aged children are poorly understood. The goal of the present study was to evaluate effects related to memory for pitch, musical training, and the utilization of temporal fine-structure cues. Listeners were normal-hearing children, 5.1–13.6 years old, and adults. One subgroup of children had musical training (>150 h) and the other did not. The standard stimulus was either a 500- or a 5000-Hz pure tone, and the target stimulus was either a tone of higher frequency or a frequency-modulated tone (2- or 20-Hz rate) centered on the standard frequency. As commonly observed, mean frequency discrimination thresholds tended to be elevated in younger listeners. This developmental effect was smaller for FM detection than for pure-tone frequency discrimination, consistent with an effect of memory for pitch. The child/adult difference tended to be smaller for musically trained than untrained children. Children were not particularly poor at 2-Hz FM detection for the 500-Hz standard, a condition thought to rely on temporal fine-structure cues. [Work supported by NIDCD R03DC008389.]

2pPPb2. Cognitive and auditory influences on speech recognition by younger, middle-aged, and older listeners. Karen S. Helfer and Angela Costanzi (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu)

In this study, we measured hearing thresholds and cognitive abilities (working memory, processing speed, executive function, and inhibition) in our participants to determine how these factors relate to speech understanding in the presence of competing speech. Participants were younger, middle-aged, and older adults, with the older adults having hearing thresholds that ranged from normal to a moderate hearing loss. The target stimuli for this study were recordings of TVM sentences [Helfer and Freyman (2009)] presented with various types of maskers (one or two other TVM sentences spoken by same-sex maskers, samples of running speech spoken by one or two same-sex talkers, and single-channel signal-envelope-modulated noise) at several signal-to-noise ratios. This poster will discuss speech recognition performance in the framework of multimasker penalties and will detail connections among hearing thresholds, cognitive abilities, and speech understanding. [Work supported by NIH DC012057.]

2pPPb3. Effects of reverberation and spatial diffuseness on the speech intelligibility of public address sounds in subway platform for young and aged people. Yong Hee Kim and Yoshiharu Soeta (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, yh.kim@aist.go.jp)

This paper investigated how young and aged people response differently on the public address sounds in subway platform according to various reverberation and diffuseness. Both word intelligibility (WI) and listening difficulty (LD) tests were adopted as experimental method. Twelve simulated sound fields at the same position were prepared according to the combination of four different reverberation times (RT) and three different interaural cross-correlation coefficients (IACC). Auralized sounds with anechoic test words of actual station names were presented to young and aged subjects at the fixed sound pressure level. As results, LD results showed significant correlation with RT for both young and aged subjects, whereas significant correlation of WI results and RT was found only in aged subjects. Aged subjects showed worse speech intelligibility performances on public address sounds than young subjects due to their worse hearing level. RT was found as the most important factor to determine speech intelligibility for both young and aged subject, whereas aged subject showed better speech intelligibility performances with lower IACC. From the regression analysis, LD rating was estimated from the measured RT and IACC. Additionally, the effects of word familiarity, individual noise sensitivity and hearing level on the speech intelligibility were discussed.

2pPPb4. Auditory influence on tactile perception changes with age. Simon P. Landry (École d'orthophonie et d'audiologie, Université de Montréal, 7077 avenue du Parc, Montreal, QC H3N 1X7, Canada, simon.landry.4@umontreal.ca), Jean-Paul Guillemot (Département de kinanthropologie, Université du Québec à Montréal, Montreal, QC, Canada), and François Champoux (École d'orthophonie et d'audiologie, Université de Montréal, Montreal, QC, Canada)

Characteristics of auditory interaction with vision have been extensively studied. However, auditory system integration with other sensory modalities, such as the tactile system, lacks such thorough investigations. The objective of this study was to examine the effects of age on audiotactile integration in humans. Thirty-one participants between the ages of 20 and 65 were divided into three groups according to their age. Audiotactile integration was assessed using the "auditory flash illusion" in which 1, 2, 3, or 4 tactile stimuli were accompanied with 0, 1, 2, 3, or 4 auditory stimuli. Participants were asked to ignore auditory stimulations and report the number of tactile stimulations perceived. All participants were tested with task relevant auditory and tactile stimuli as a control measures and were shown to have similar abilities. However, groups differed during the experimental conditions. The youngest group reported a greater number of tactile stimuli than actually presented during the illusory experimental conditions. Participants in the middle and older age groups did not report this illusory tactile perception. These results suggest that age reduces predisposition to audiotactile integration. These results are consistent with developmental studies for multisensory integration in other sensory modalities.

2pPPb5. The influence of auditory training on measures of temporal resolution in younger and older adults. Meital Avivi-Reich (Psychology, Univ. of Toronto Mississauga, 1909-35 Charles st.W, Toronto, ON m4y 1r6, Canada, me_avv@yahoo.com), Stephen R. Arnott (Rotman Res. Inst., Baycrest, Toronto, ON, Canada), Tamara Tavares, and Bruce A. Schneider (Psychology, Univ. of Toronto Mississauga, Toronto, ON, Canada)

Deterioration in the ability to perceived rapid changes in auditory input is thought to contribute to the difficulties older adults experience when communicating in noise. Studies have demonstrated that the performance of

young adults on auditory tasks improves with training. However, few studies have tested the degree to which practice improves auditory performance in older adults. A previous study examined the extent to which younger and older adults benefited from training when the task was to detect a gap in a narrow-band noise centered at 1 kHz. Significant improvements occurred in both age groups, indicating that auditory learning can still occur later in life. The present study examines if training improves performance when more than one auditory filter is activated. Twenty-four younger and older participants were trained for 10 days to detect a gap between a 2-kHz and a 1-kHz noise. Performance was assessed one day and one month after the last training session along with the extent to which the benefits generalized to other frequencies and the untrained ear. In addition, event-related potentials (ERPs) were obtained pre- and post-training to assess cortical changes in the response to temporal gaps. Initial results show improvement throughout training in both age groups.

2pPPb6. Intelligibility of voiced and whispered speech in noise in listeners with and without musical training. Dorea Ruggles, Ariane Riddell (Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, druggles@umn.edu), Richard Freyman (Univ. of Massachusetts-Amherst, Amherst, MA), and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN)

Musicians have been shown to exhibit more robust neural coding of periodicity than non-musicians; they have also been reported to exhibit an advantage in understanding speech in noise. This study tested the hypothesis that the musicians' speech intelligibility advantage arises from more efficient coding of voiced (periodic) speech. This was tested by comparing intelligibility of normal speech in noise with that of whispered (unvoiced) speech in musicians and non-musicians. Listeners with less than 2 years of formal musical training were categorized as non-musicians; listeners who began musical training before age 10 and who currently play more than 10 h/wk were included as musicians. Listeners heard grammatically correct nonsense sentences that were either (1) voiced, (2) whispered, or (3) whispered with subband amplitude distributions matched to the voiced speech. Masking noise was either continuous or gated with a 16 Hz, 50% duty cycle. In contrast to the earlier study, preliminary data suggest no advantage for musicians over non-musicians in understanding voiced or whispered speech in either continuous- or gated-noise conditions. The results suggest that more investigation is needed to fully understand the nature of auditory and speech processing advantages imparted by musical training. [Wok supported by NIH Grant No. R01DC05216.]

2pPPb7. Level-dependent effects on speech intelligibility. Patricia Pérez-González and Enrique A. Lopez-Poveda (Neurosci. Inst. of Castilla y Leon, Univ. of Salamanca, Pintor Fernando Gallego 1, Salamanca, Salamanca 37007, Spain, patripg@usal.es)

The effect of noise on speech intelligibility is typically measured using fixed-level speech (or noise) and varying the speech-to-noise ratio (SNR). An assumption of this procedure is that intelligibility mostly depends on the SNR and barely depends on speech level. The effective SNR, however, (i.e., the SNR in the internal stimulus representation), possibly depends on peripheral compression. Indeed, compression could facilitate or hinder intelligibility for negative and positive SNRs, respectively. Insofar as compression varies with level, speech intelligibility might also vary with speech level. Here, we tested these hypotheses by measuring percent correct digit triplet identification as a function of speech level for fixed SNRs. Measurements were carried out for normal-hearing subjects and for hearing-impaired subjects with linear cochlear responses, as assessed using the temporal masking curve method. Results for both groups suggest that the detrimental effect of the noise on intelligibility is larger for speech levels near threshold, particularly for negative SNRs, a result that cannot easily be explained by compression. Alternative explanations for the result are discussed.

2pPPb8. Talker effects in speech band importance functions. Eric W. Healy, Sarah E. Yoho, Carla L. Youngdahl, and Frederic Apoux (Speech and Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu)

The literature is somewhat mixed with regard to the influence of (a) the particular speech material (e.g., sentences or words) versus (b) the particular talker used to create the recordings, on band-importance function (BIF) shape. One possibility is that previous techniques for creating BIFs are not sensitive enough to reveal these influences. In the current investigation, the role of talkers was examined using the compound technique for creating BIFs. This technique was developed to account for the multitude of synergistic and redundant interactions that take place among various speech frequencies. The resulting functions display a complex microstructure, in which the importance of adjacent bands can differ substantially. It was found that the microstructure could be traced to acoustic aspects of the particular talkers employed. Further, BIFs for IEEE sentences based on ten-talker recordings displayed less microstructure and were therefore smoother than BIFs based on one such talker. These results together suggest that the compound technique is sensitive enough to reveal acoustic aspects of the particular talker employed. It is further suggested that multiple talkers, rather than smoothing of the functions, be used if the goal is to describe speech more generally.

2pPPb9. Psychometric effects of adding realism to a speech-in-noise test. Virginia Best, Gitte Keidser, Jörg M. Buchholz, and Katrina Freeston (National Acoust. Lab. and the HEARing Cooperative Res. Ctr., 126 Greville St., Chatswood, NSW 2067, Australia, virginia.best@nal.gov.au)

The speech reception threshold (SRT) is routinely measured in the laboratory to assess speech understanding in noise, but is often reported to be a poor predictor of performance in real world listening situations. The overall goal of this work is to determine whether introducing realistic aspects to speech tests can better capture individual differences and ultimately produce more relevant performance measures. We examined the psychometric effects of (a) transplanting a standard sentence-in-noise test into a simulated reverberant cafeteria environment, and (b) moving from sentence recall to a new ongoing speech comprehension task. Participants included normal hearers and hearing-impaired listeners (who were tested with and without their hearing aids). SRTs in the cafeteria environment were significantly correlated with standard SRTs, but were poorer overall and more sensitive to hearing loss. The comprehension task, despite having very different demands to sentence recall, produced similar SRTs under these conditions. The benefit of hearing aids was weakly correlated across the two listening environments and the two listening tasks. These manipulations promise to be useful for the creation of realistic laboratory tests that are engaging and challenging, yet controlled enough to be useful for psychophysical experiments.

2pPPb10. The Glasgow Monitoring of Uninterrupted Speech Task: A naturalistic measure of speech intelligibility in noise. Alexandra MacPherson and Michael A. Akeroyd (MRC Inst. of Hearing Res. (Scottish section), Queen Elizabeth Bldg., 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, alex@ihr.gla.ac.uk)

When listening to speech in noisy environments parts of the speech signal are often missed due to masking, degradation, or inattention. Sometimes the message can be reconstructed, but reallocating resources to recover the missed information can affect the efficiency and speed at which the message is understood. A slowing of processing will be particularly detrimental if there are few opportunities for understanding to catch up, e.g., when listening to the radio. It is likely then that the monitoring of uninterrupted flows of continuous speech will be especially sensitive to hearing or listening impairments. We developed a new task to test this. The Glasgow Monitoring of Uninterrupted Speech Task (GMUST) requires participants to listen to a 10-min segment of continuous speech while monitoring a scrolling transcript of the speech on the screen in front of them for any word substitutions. The proportion of word substitutions listeners are able to correctly identify is then used as a measure of speech intelligibility. When compared to speech reception thresholds (SRTs) given by a standard speech-in-noise test pilot results indicate higher SRTs for the GMUST. This suggests that the GMUST could be a more sensitive measure of deficits in speech-in-noise understanding than standard speech-in-noise tests.

2pPPb11. Magnitude of speech-reception-threshold manipulators for a spatial speech-in-speech test that takes signal-to-noise ratio confounds and ecological validity into account. Filip M. Rønne, Søren Laugesen, Niels S. Jensen, Renskje K. Hietkamp, and Julie H. Pedersen (Eriksholm Res. Ctr., Rørtangvej 20, Snekkersten 3070, Denmark, fmr@eriksholm.com)

Measuring speech-reception threshold (SRT) using adaptive procedures is popular, as testing yield results with desirable statistical properties. However, SRT measures have drawbacks related to the unbounded nature of the signal-to-noise

ratio (SNR) at which the SRT is achieved. Often the SRT will be a double-digit negative number, which compromises the ecological validity of the result. If testing involves hearing aids, it means that these devices and the signal-processing algorithms in them may be operating in conditions for which they were not intended. Further, the commonly observed large spread in SRT (both between- and within-group) has the possibility to cause SNR confounds that may lead to faulty conclusions. One way to address these issues is to provide the experimenter with SRT manipulators, to control the SNR at which testing takes place for the individual listener. The present work aims at developing a spatial speech-in-speech test with a selection of SRT manipulators for the experimenter to choose from. The manipulators investigated in this study are as follows: the spatial separation between target and maskers, the number of spatially separated maskers, changing the masker gender, and scoring in words versus sentences. The magnitudes of the SRT manipulators were investigated using 20 hearing-aid users as listeners.

2pPPb12. Binaural speech recognition in continuous and intermittent noises in people with hearing loss. Chantal Laroche, Jean-Grégoire Roveda, Julie Levionnois, Christian Giguère, and Véronique Vaillancourt (Audiology/SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H 8M5, Canada, claroche@uottawa.ca)

Multiple tests have been developed to quantify speech recognition in noise, but the characteristics of the masking noise vary significantly across tests and can considerably impact performance and clinical interpretation. Using the HINT, speech perception in 24 young adults with normal hearing was assessed using both the standard masker (a continuous speech-spectrum noise) and an intermittent version of the masker at an ON-OFF rate of 16 Hz. Intermittency helps in extracting speech from the "quiet" segments of the noise. Speech recognition thresholds for frontal speech were better in the intermittent than the continuous noise by an amount of 13 and 10 dB for noise maskers located in front or side, respectively, The average difference in thresholds between the noise front and side conditions, called the binaural advantage, was 4.7 and 8.4 dB for the intermittent and continuous noises, respectively. Data collected with people presenting different hearing loss profiles also show binaural and intermittency advantages, but to a lesser degree. Considering that people encounter a wide a range of fluctuating noises in daily life, these results motivate adding an intermittent noise condition to the HINT protocol to better reflect the challenges faced by individuals with hearing loss.

2pPPb13. Spatial release from masking for noise-vocoded speech. Jayaganesh Swaminathan, Christine R. Mason, Timothy M. Streeter (Boston University, 635 Commonwealth Avenue, Room 320, Boston, MA 02215, jswamy@bu.edu), Virginia Best (National Acoust. Lab., Chatswood, NSW, Australia), and Gerald Kidd, Jr (Boston Univ., Boston, MA)

Spatially separating a speech target from interfering masker(s) generally improves target intelligibility; an effect known as spatial release from masking (SRM). This study assessed the contribution of envelope cues to SRM. Target speech was presented from the front (0° azimuth) and speech maskers were either colocated or symmetrically separated from the target in azimuth (±15°, ±30°, ±45° and ±90°) using KEMAR head-related transfer functions. The target and maskers were presented either as natural speech or as noise-vocoded speech. For the vocoded speech, intelligibility was conveyed only by the envelopes from M frequency bands. Experiment 1 examined the effects of varying the number of frequency bands for the vocoder, and the degree of target-masker spatial separation, on SRM. Experiment 2 examined the effects of low-pass filtering the envelopes of the vocoded speech bands on SRM. Preliminary results for experiment 1 indicated that SRM improved as the number of spectral channels providing independent envelope cues increased for all spatial separations. Preliminary results for experiment 2 showed no difference in SRM between low and high envelope-frequency cutoffs. Potential implications for studying hearing-impaired and cochlear-implant subjects will be discussed. [Work supported by NIH-NIDCD and AFOSR.]

2pPPb14. Can envelope recovery account for speech recognition based on temporal fine structure? Frederic Apoux, Carla L. Youngdahl, Sarah E. Yoho, and Eric W. Healy (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

Over the past decade, several studies have demonstrated that normal-hearing listeners can achieve high levels of speech recognition when presented with only the temporal fine structure (TFS) of speech stimuli. Initial suggestions to explain these findings were that they were the result of the auditory system's ability to recover envelope information from the TFS (envelope recovery; ER). A number of studies have since showed decreasing ER with increasing numbers of analysis filters (the filters used to decompose the signal) while intelligibility from speech-TFS remains almost unaffected. Accordingly, it is now assumed that speech information is present in the TFS. A recent psychophysical study, however, showed that envelope information remains in the TFS after decomposition, suggesting a possible role of ER in speech-TFS understanding. The present study investigated this potential role. In contrast to previous work, a clear influence of analysis filter bandwidth on speech-TFS understanding was established. In addition, it was shown that near perfect speech recognition from recovered envelopes can be achieved with as many as 15 analysis filters. Finally, the relationship between analysis and auditory filter bandwidths was explored in ER. Taken together, the present findings suggest that a role of ER in speech-TFS understanding cannot be excluded.

2pPPb15. Complementary correlation may offer a new approach to better understand temporal fine structure coding. Adrian KC Lee (Speech & Hearing Sci. and Inst. for Learning & Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98195, akclee@uw.edu), Les E. Atlas, and Xing Li (Elec. Eng., Univ. of Washington, Seattle, WA)

Historically, sound has been separated into a slow-varying envelope and a component that rapidly varies, called the temporal fine structure (TFS). Perceptual studies suggest that users of cochlear implants can benefit from better TFS coding strategies, as they lead to improved speech understanding in noise. Yet given the standard signal processing methodology used in the auditory neuroscience community, namely the Hilbert transform, findings in such prior studies have been limited by their use of the heavily distorted estimates of the TFS information (through the Hilbert phase of the related analytic signal). Complementary correlation is a new mathematical tool with potential to advance our understanding of temporal coding in neuroscience. This new mathematical formulation can be defined simply by dropping the usual complex conjugation operation from the canonical definitions of correlation, coherence, variance, magnitude-squared estimators of power, and other similar common second-order statistical quantities and their estimators. We will show that a complementary correlation approach, which provides a more complete characterization of TFS information, will provide measurable perceptual benefits by reducing distortion in the original speech signal.

2pPPb16. The roles of temporal envelope and temporal fine structure in speech synthesis for cochlear implants for tonal language speakers. Nantaporn Saimai, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Khlongluang, Pathumthani 12120, Thailand, 5310030076@student.tu.ac.th), Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand), and Chai Wutiwiwatchai (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

Since most cochlear implants (CIs) have been developed for non-tonal languages, their level of hearing improvement is significantly decreased when used by speakers of tonal language, e.g., Thai. Temporal envelope (TE) and temporal fine structure (TFS) are important acoustic cues for languages with lexical tones. Specifically, TE is shown to carry manner and voicing cues for consonants, while TFS correlates with vowel formant transitions. Therefore, TFS and TE are expected to enhance intelligibility of lexical tones for CI patients. We proposed the use of six-channel bandpass filters to extract spectral information. Then, TE is extracted by half-wave rectification and smoothed by lowpass filler at 500 Hz cutoff frequency. TFS is extracted by the Hilbert transform to construct carrier signals. TE from each channel is modulated with its corresponding carrier signal and then combined to generate synthesized speech. Synthesized speech tokens from this study and two others [Fu *et al*. (1998) and Chen and Zhang (2008)] are evaluated by 16 Thais with normal hearing. The results showed that the intelligibility scores from the proposed algorithm are significantly higher than the other two for initials (by 32.2%) and final consonants (by 16.7%) and significantly higher for tones (by 48.8%) than Fu *et al*.

2pPPb17. The role of peripheral spectro-temporal coding in congenital amusia. Marion Cousineau (Département de Psychologie, Université de Montréal, Pavillon 1420 boul. Mont Royal, Entrance #1430 Ste. 0-120, Outremont, QC H2V 4P3, Canada, marioncousineau@gmail.com), Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), and Isabelle Peretz (Département de Psychologie, Université de Montréal, Montreal, QC, Canada)

Congenital amusia, a neurogenetic disorder, affects primarily pitch and melody perception. Here we test the hypothesis that amusics suffer from impaired access to spectro-temporal fine-structure cues associated with low-order resolved harmonics. The hypothesis is motivated by the fact that tones containing only unresolved harmonics result in poorer pitch sensitivity in normal-hearing listeners. F0DLs were measured in amusics and matched controls for harmonic complexes containing either resolved or unresolved harmonics. Sensitivity to temporal-fine-structure was assessed via interaural-time-difference (ITD) thresholds, intensity resolution was probed via interaural-level-difference (ILD) thresholds and intensity difference limens, and spectral resolution was estimated using the notched-noise method. As expected, F0DLs were elevated in amusics for resolved harmonics; however, no difference between amusics and controls was found for F0DLs using unresolved harmonics. The deficit appears unlikely to be due to temporal-fine-structure coding, as ITD thresholds were unimpaired in the amusic group. In addition, no differences were found between the two groups in ILD thresholds, intensity difference limens, or auditory-filter bandwidths. Overall the results suggest a pitch-specific deficit in fine spectro-temporal information processing in amusia that cannot be ascribed to defective temporal-fine-structure or spectral encoding in the auditory periphery. [Work supported by Fyssen Foundation, Erasmus Mundus, CIHR, and NIH grant R01DC05216.]

2pPPb18. Comodulation masking release and monaural envelope correlation perception in listeners with cochlear hearing loss. Heather Porter, John H. Grose, Joseph W. Hall, and Emily Buss (Otolaryngol. - Head & Neck Surgery, Univ. of North Carolina at Chapel Hill, 170 Manning Dr., Chapel Hill, NC 27599-7070, heather_porter@med.unc.edu)

This study investigated the comparative dependence of comodulation masking release (CMR) and monaural envelope correlation perception (MECP) on the degree of envelope similarity in pre-senescent adult listeners with normal hearing (NH) or mild-to-moderate cochlear hearing loss (CL). A 1600-Hz pure-tone signal was used to measure CMR as a function of degree of envelope correlation in 100-Hz-wide noise bands centered at 727, 1093, 1600, 2300, and 3268 Hz. The same noise band configuration was used to measure MECP thresholds for both comodulated and independent standards. Envelope correlation was adjusted by mixing comodulated and independent maskers at variable intensity ratios. The five-band complex was 85 dB SPL. Signal thresholds improved monotonically (i.e., CMR increased) with increasing degrees of envelope correlation for all listeners. Results for CL listeners were most similar to data from previous NH listeners at a 72 dB SPL masker level. For MECP, performance patterns for the two conditions were uniform across NH listeners, whereas those for CL listeners exhibited greater individual differences. Finally, CMR and MECP performance appeared to be related in listeners with CL. The pattern of results will be discussed in terms of the effects of CL on sensitivity to envelope similarity. [Work supported by NIDCD R01DC001507.]

2pPPb19. Attentional switching when listeners respond to semantic meaning expressed by multiple talkers. Ervin R. Hafter (Dept. of Psych., Univ. of California, Berkeley, CA, hafter@berkeley.edu), Jing Xia, Sridhar Kalluri (Starkey Hearing Res. Ctr., Berkeley, CA), Rosa Poggesi, Claes Hansen, and Kelly Whiteford (Psychology, Univ. of California, Berkeley, Berkeley, CA)

The “cocktail party problem” asks how we know what one person says when others are speaking at the same time. With interest in the difficulty faced by older and hearing-impaired listeners in multi-talker environments, the present experiment looks at the speed of shifting attention between talkers. In our simulated cocktail party, a subject sits among multiple talkers who are each telling a different story. A sequence of questions are drawn from the various stories and presented visually for subjects to answer with manual responses. Pay is based on correct answers, and attention is assessed by comparing response accuracy on questions from the talker identified visually as the primary and from the other talkers. Attention is manipulated by varying the primary talker at random, from question to question. The speed of shifting attention is measured by varying the time from when the new primary is identified to the moment when the relevant information appears in that story. In a related study, bilingual subjects must shift attention to talkers speaking either English or Spanish. This allows determination of the additional time needed to switch languages within a multi-talker environment.

2pPPb20. Dividing attention between two segregated tone streams. Laurent Demany and Catherine Semal (INCIA, CNRS and Universite de Bordeaux, BP 63, 146 rue Leo Saignat, Bordeaux F-33076, France, laurent.demany@u-bordeaux2.fr)

Listeners were presented with pure-tone sequences which had a high speed (25–50 tones/s) and consisted of two interleaved melodies, M1 and M2, spanning separate frequency ranges (624–786 Hz and 1483–1869 Hz). M1 and M2 were renewed from sequence to sequence and could be either (1) perfectly correlated, or (2) perfectly anticorrelated, or (3) independent of each other. The main task, performed in a 2I-2AFC paradigm, was to discriminate sequences of type 1 (perfect correlation) from sequences of type 2 or 3. This appeared to be relatively easy when, in the type-1 sequences, each note of M2 immediately preceded or followed the corresponding note of M1. In that case, however, listeners were unable to tell whether M2 preceded or followed M1, which shows that the two melodies were perceptually segregated. In another series of experimental sessions, listeners knew that M2 would always follow M1 after a fixed delay, corresponding to one tone in some sessions and three tones in other sessions. The main discrimination task was performed better for the one-tone delay, but performance was still well above chance for the three-tone delay. Overall, the data suggest that two segregated tone streams can be attended to simultaneously.

2pPPb21. Selective auditory or visual attention reduces physiological noise in the ear canals of human subjects. Kyle P. Walsh, Edward G. Pasanen, and Dennis McFadden (Psychology, Univ. of Minnesota, 75 East River Rd., Minneapolis, Texas 55455, kpwalsh@umn.edu)

A nonlinear version of the stimulus-frequency OAE (SFOAE), called the nSFOAE, was used to measure cochlear responses from human subjects while they simultaneously performed behavioral tasks requiring selective auditory attention (dichotic or diotic listening), selective visual attention, or relatively little attention. The auditory- and visual-attention tasks both were digit-recall tasks, where the nSFOAE-stimuli were interleaved with seven spoken (or displayed) digits. Unlike many previous studies, the required motor behavior always was the same across all tasks, including the inattention tasks. A 30-ms recording in the quiet followed every nSFOAE-eliciting stimulus to provide an estimate of the magnitude of each subject’s physiological noise in each experimental condition. For every subject, physiological-noise magnitudes were higher (noisier) in the inattention tasks, and lower (quieter) in the selective auditory- and visual-attention tasks. The differences in noise levels were about 3–6 dB, on average, and the effect sizes for those differences all were greater than 2.5. Our interpretation is that the efferent innervation of the cochlea is activated maximally during selective attention (be it auditory or visual), potentially to the benefit of the observer. [Work supported by NIDCD grant DC00153.]

2pPPb22. Build-up of auditory stream segregation induced by tone sequences of constant or alternating frequency and the resetting effects of single deviants. Nicholas R. Haywood and Brian Roberts (Psychology, School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

Three experiments investigated the dynamics of auditory stream segregation. Experiment 1 used a 2.0-s constant-frequency inducer (10 repetitions of a low-frequency pure tone) to promote segregation in a subsequent, 1.2-s test sequence of alternating low- and high-frequency tones. Replacing the final inducer tone with silence reduced reported test-sequence segregation substantially. This reduction did not occur when either the 4th or 7th inducer was replaced with silence. This suggests that a change at the induction/test-sequence boundary actively resets build-up, rather than less segregation occurring simply because fewer inducer tones were presented. Furthermore, experiment 2 found that a constant-frequency inducer produced its

maximum segregation-promoting effect after only three tone cycles—this contrasts with the more gradual build-up typically observed for alternating sequences. Experiment 3 required listeners to judge continuously the grouping of 20-s test sequences. Constant-frequency inducers were considerably more effective at promoting segregation than alternating ones; this difference persisted for ~10 s. In addition, resetting arising from a single deviant (longer tone) was associated only with constant-frequency inducers. Overall, the results suggest that constant-frequency inducers promote segregation by capturing one subset of test-sequence tones into an on-going, pre-established stream and that a deviant tone may reduce segregation by disrupting this capture.

2pPPb23. Annoyance perception for hearing impaired listeners: A revisit. Susie Valentine, Martin McKinney, and Tao Zhang (Starkey Hearing Technol., 6600 Washington Ave. S., Eden Prairie, MN 55344, susie_valentine@starkey.com)

For hearing impaired (HI) listeners, it is well known that certain sounds are much more annoying than others even though they may have similar spectral shape and level. For example, HI listeners often report paper rustling noise as highly annoying. A common approach to deal with this complaint is to reduce high frequency gain. While this approach may mitigate the complaint, it can create audibility issues for speech. A more effective approach is to determine the underlying cause of annoyance and then design an algorithm to selectively reduce it. While existing literature on annoyance perception for HI listeners is scant, a previous attempt was made to investigate this perception using real-world recordings [Vishnubhotla *et al.* (2012)]. The study showed a large variability of annoyance ratings across listeners that may have been due to subjective associations with the sound sources. In this study, we use abstract psychoacoustic stimuli designed carefully to avoid possible confounding subjective associations. A magnitude estimation method was used to measure the annoyance of each stimulus in hearing impaired listeners. All stimuli were presented over headphones in a sound treated room. Results will be presented along with implications for hearing aid applications.

2pPPb24. Acoustic correlates of tinnitus-like sounds. Jennifer Lentz and Yuan He (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu)

Although many people describe their tinnitus using complex terms (such a tea-kettle, crickets, and roaring), past studies of tinnitus have focused using pure tones and noises as stimuli. Therefore, this study was developed to begin to address the usefulness of using complex, dynamic sounds in the assessment of tinnitus. In a previous study, a free-classification task was used to ascertain the perceptual dimensions of tinnitus-like sounds in normally hearing listeners. Sounds were representative of those commonly used to describe tinnitus (e.g., ringing, tonal, noisy, pulsing, and clicking sounds). Listeners placed icons associated with each sound on a grid and placed similar sounds in clusters. Multi-dimensional scaling conducted on the classification data revealed three different perceptual dimensions. This study evaluated the acoustics of the stimuli to determine the nature of the perceptual dimensions. These analyses estimated a variety of temporal and spectral stimulus properties (e.g., autocorrelation statistics, spectral statistics, envelope characteristics, etc.). The acoustic characteristics were then correlated with the ordering along the three perceptual dimensions. Results suggest a noisy versus tonal dimension, an envelope-based dimension stimulus (choppy versus smooth), and a dimension related to dynamic stimulus characteristics.

2pPPb25. Intervention for restricted dynamic range and reduced sound tolerance: Clinical trial using a Tinnitus Retraining Therapy protocol for hyperacusis. Craig Formby (Commun. Disord., Univ. of Alabama, 700 University Boulevard East, Tuscaloosa, AL 35487, cformby@as.ua.edu), Monica Hawley, LaGuinn P. Sherlock, Susan Gold (Otorhinolaryngology, Univ. of Maryland School of Medicine, Baltimore, MD), Jason Parton, Rebecca Brooks, and JoAnne Payne (Commun. Disord., Univ. of Alabama, Tuscaloosa, AL)

Hyperacusis is the intolerance to sound levels that normally are judged acceptable to others. The presence of hyperacusis (diagnosed or undiagnosed) can be an important reason that some persons reject their hearing aids. Tinnitus Retraining Therapy (TRT), a treatment approach for debilitating tinnitus and hyperacusis, routinely gives rise to increased loudness discomfort levels (LDLs) and improved sound tolerance. TRT involves both counseling and the daily exposure to soft sound from bilateral noise generator devices (NGs). We implemented a randomized, double-blind, placebo-controlled clinical trial to assess the efficacy of TRT as an intervention for reduced sound tolerance in hearing-aid eligible persons with hyperacusis and/or restricted dynamic ranges. Subjects were assigned to one of four treatment groups (2x2): Devices: NGs or placebo NGs and Counseling: Yes or No. They were evaluated at least monthly on a variety of audiometric tests, including LDLs, the Contour Test for Loudness for tones and speech, and word recognition measured at each session's comfortable and loud levels. Eighty percentage of the participants who received full treatment benefited significantly; whereas the other treatment groups demonstrated $\leq$ 45% treatment efficacy. Treatment dynamics and examples of improved word recognition post-treatment will be described. [Work supported by NIH R01 DC04678.]

2pPPb26. Relationship between distortion and working memory for digital noise-reduction processing in hearing aids. Kathryn Arehart (Speech, Lang., Hearing Sci., Univ. of Colorado, UCB 409, Boulder, CO 80309, kathryn.arehart@colorado.edu), Pamela Souza (Dept. of Commun. Sci. and Disord. and Knowles Hearing Ctr., Northwestern Univ., Evanston, IL), Thomas Lunner (Eriksholm Res. Ctr., Oticon A/S, Linköping, Sweden), Michael Syskind Pedersen (Oticon A/S, Smørum, Denmark), and James M. Kates (Speech, Lang., Hearing Sci., Univ. of Colorado, Boulder, CO)

Several recent studies have shown a relationship between working memory and the ability of older adults to benefit from specific advanced signal processing algorithms in hearing aids. In this study, we quantify tradeoffs between benefit due to noise reduction and the perceptual costs associated with distortion caused by the noise reduction algorithm. We also investigate the relationship between these tradeoffs and working memory abilities. Speech intelligibility, speech quality, and perceived listening effort were measured in a cohort of elderly adults with hearing loss. Test materials were low-context sentences presented in fluctuating noise conditions at several signal-to-noise ratios. Speech stimuli were processed with a binary mask noise-reduction strategy. The amount of distortion produced by the noise reduction algorithm was parametrically varied by manipulating two binary mask parameters, error rate, and attenuation rate. Working memory was assessed with a reading span test. Results will be discussed in terms of the extent to which intelligibility, quality, and effort ratings are explained by the amount of distortion and/or noise and by working memory ability. [Funded by NIH, Oticon, and GN ReSound.]

2pPPb27. Evaluation of a binaurally synchronized dynamic-range compression algorithms for hearing aids. Stephan Ernst, Giso Grimm, and Birger Kollmeier (Med. Phys., Univ. of Oldenburg, Carl von Ossietzky Universität Oldenburg, Oldenburg 26111, Germany, stephan.ernst2@uni-oldenburg.de)

Binaural cues such as interaural level differences (ILD) are used, among other cues, to organize auditory perception and to segregate sound sources in complex acoustical environments. Dynamic-range compression working independently at each ear in a bilateral hearing aid, however, can alter these ILDs, potentially affecting sound source segregation. Binaural synchronization of compression algorithms might thus be necessary to preserve potentially beneficial spatial cues. This study presents a binaurally linked model-based fast-acting dynamic compression algorithm designed to approximate the normal-hearing basilar membrane input-output function in hearing-impaired listeners. Aim of the evaluation was to assess the effect of binaural synchronization on speech intelligibility and listening effort in spatial masking conditions in comparison to bilateral fitting. Spatially symmetric and asymmetric masking conditions were used. A conventional multiband dynamic compression algorithm both implemented in a bilaterally independent and in a binaurally linked version, was tested as a reference. Hearing impaired listeners were aided individually with the algorithms for both experiments. Results indicate a small preference toward the model-based algorithm in challenging masking conditions. However, no benefit of binaural-synchronization could be found even for the fast-acting compressor, suggesting a dominant role of the better ear in all experimental conditions. [Work funded by BMBF 01EZ0741 and DFG FOR1732.]

2pPPb28. Assessing the contribution of spectral cues to recognition of frequency-lowered consonants. Kelly Fitz (Signal Process. Res., Starkey Hearing Technol., 6600 Washington Ave. South, Eden Prairie, MN 55344, kelly_fitz@starkey.com), Christophe Micheyl (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), Susie Valentine (Audiology Res., Starkey Hearing Technol., Eden Prairie, MN), and Tao Zhang (Signal Process. Res., Starkey Hearing Technol., Eden Prairie, MN)

Commercially available strategies for restoring audibility of critical high frequency cues to patients with severe high frequency hearing loss translate information from high-frequency regions with unaidable hearing to lower-frequency regions with aidable hearing. Methods for synthesizing lowered spectral features, rather than generating them from the signal, have been proposed, though no commercially available hearing aid uses such a method. We assessed consonant discrimination under three configurations of a spectral feature synthesis method intended for use in frequency lowering. Lowered consonants were rendered using one or two narrowband noise components presented in a low frequency region with aidable hearing. Different configurations conveyed different spectral cues intended to distinguish among lowered consonants. In a short pilot study, preliminary analysis found no significant difference in consonant matching accuracy between the different configurations, suggesting that listeners were not making use of additional spectral cues when they were available. However, there were some observable trends in the data, as well as open questions about the possible impact of training and acclimatization on listener performance. We will present the findings of a more extensive study to determine whether listeners with training can make use of enhanced spectral cues to distinguish among frequency-lowered consonants.

2pPPb29. Spatial release from masking in simulations of cochlear implants for single-sided deafness. Joshua G. Bernstein (Audiology and Speech Ctr., Walter Reed National Military Med. Ctr., 8901 Wisconsin Ave., Bethesda, MD 20889, joshua.g.bernstein.civ@health.mil), Nandini Iyer (Battlespace Acoust. Branch, Air Force Res. Lab., Wright Patterson Air Force Base, OH), and Douglas S. Brungart (Audiology and Speech Ctr., Walter Reed National Military Med. Ctr., Bethesda, MD)

Previous studies have shown that single-sided-deaf (SSD) patients implanted with cochlear implants (CIs) can receive spatial release-from-masking (SRM), likely due to head-shadowing effects. This study investigated the possibility that SSD-CI patients might also obtain SRM benefits from binaural-integration cues when the target speech is masked by a spatially separated fluctuating masker. Experiment 1 measured psychometric functions for word-recognition performance in the presence of stationary noise, modulated noise, and one or two same- or opposite-gender interfering talkers. The first ear received an unprocessed mixture containing the target and masker(s). The second ear received no signal (SSD), an unprocessed mixture containing just the maskers (NH-Binaural), or a mixture containing just the maskers that was processed with an eight-channel noise vocoder (SSD+CI). The results show that SRM occurs in the NH-Binaural condition for all masker types, but that it only occurs in the SSD+CI condition with same-gender interfering talkers. Experiment 2 revealed that SRM occurred in the SSD+CI condition with as few as two vocoder channels, and that maximum performance occurred with six or more channels. These results suggest that CIs for SSD have the potential to produce SRM in situations where monaural cues are insufficient for concurrent speech-stream segregation.

2pPPb30. Speech intelligibility of hearing impaired participants in long-term training of bone-conducted ultrasonic hearing aid. Toshie Matsui, Ryota Shimokura, Tadashi Nishimura, Hiroshi Hosoi (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Med. Univ., Shijo-cho 840, Kashihara City 634-8522, Japan, tomatsui@naramed-u.ac.jp), and Seiji Nakagawa (Living Informatics Res. Group, National Inst. of Adv. Industrial Sci. and Technol., Ikeda City, Hyogo, Japan)

Bone-conducted ultrasonic hearing aid (BCUHA) system is the unique device to provide the auditory sensation to profoundly hearing-impaired persons without any surgical operations. To clarify effects of long-term hearing training with this device, two deaf participants engaged the BCUHA training for 9 months. They were trained to use BCUHA through repetition of sentences read aloud and free conversation, and then they took part in word recognition tests and monosyllabic identification tests. Both participants showed that they could recognize words above chance using auditory sensation only provided by BCUHA device if alternatives or context were presented to them during the trials. Besides, it was observed that monosyllabic intelligibility score with both of auditory and visual cue had much increased with the day of training than the score with auditory cue only and that with visual cue only. The result suggests that the long-term training with BCUHA achieves efficient integration of auditory and visual cue of speech such as cochlear implant users showed in previous studies.

2pPPb31. Development of a novel hearing-aid for the profoundly deaf using bone-conducted ultrasonic perception: Assessments of the modulation type with regard to articulation, intelligibility, and sound quality. Seiji Nakagawa, Chika Fujiyuki, Yuko Okubo, Takayuki Kagomiya, and Takuya Hotehama (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, s-nakagawa@aist.go.jp)

Bone-conducted ultrasound (BCU) is perceived even by the profoundly sensorineural deaf. We have developed a novel hearing-aid using BCU perception (BCU hearing aid: BCUHA) for the profoundly deaf. In the BCUHA, ultrasonic sinusoids of about 30 kHz are amplitude-modulated by speech and presented to the mastoid. Generally, two sounds are perceived: one is a high-pitched tone due to the ultrasonic carrier, with a pitch corresponding to a 8–16 kHz air-conducted (AC) sinusoid, and the other is the envelope of the modulated signal. As a method of amplitude modulation (AM), double-sideband with transmitted carrier (DSB-TC) modulation had been used; however, the DSB-TC is accompanied by a strong high-pitched tone. In this study, two new AM methods, double-sideband with suppressed carrier (DSB-SC) and transposed modulation (TM), that can be expected to reduce the high-pitched tone were newly employed in the BCUHA, and their resulting articulations, intelligibilities, and sound qualities were evaluated. The results showed that DSB-TC and TM had higher articulation and intelligibility scores than DSB-SC. Further, in terms of sound quality, the TM speech was closer than other types of BCU speech to AC speech. These results provide useful information for further development of the BCUHA.

2pPPb32. Communication aid utilizing bone-conducted sound via teeth by means of mouthpiece form actuator. Mikio Muramatsu (Dept. of Intermedia Art and Sci., Waseda Univ., Tokyo, Japan), Junko Kurosawa (Information Technol. Res. Organization, Waseda Univ., Tokyo, Japan), Yasuhiro Oikawa, and Yoshio Yamasaki (Dept. of Intermedia Art and Sci., Waseda Univ., 3-4-1, Okubo, Shinjuku-ku, Tokyo, Japan, yoikawa@waseda.jp)

Since bone-conducted sound is conveyed to cochlea directly, without passing through eardrum, it is audible even for hard-of-hearing people whose inner ears are still normal. In this study, we utilize bone-conducted sound via teeth so as to support sound communication. We implement a bone-conducted actuator on a tooth, while actuators of prevalent hearing aids are attached to mastoid, forehead, or jaw in general. Teeth convey sound excitation more easily, because they are bare bones, not covered with skin. Our hearing aid is made in the form of mouthpiece, and it can be readily put on and taken off from a tooth. Plus, we carry out experiments regarding sound localization and thresholds of bone-conducted sound via teeth, using this actuator. The results offer hearing characteristic of bone-conducted sound via teeth and show that examinees can perceive right and left using bone-conducted sound via teeth. In addition, we also attempt to record vibrations of teeth through a microphone, which is embedded on the mouthpiece form actuator. The aim of this study is to realize a hybrid actuator that enables both hearing and recording simultaneously and to suggest a new communication aid system not only for hard-of-hearing people but also for the robust.

Session 2pSA

Structural Acoustics and Vibration: Memorial Session in Honor of Miguel Junger

David Feit, Cochair
ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502

Joel M. Garrelick, Cochair
505 Tremont St., Apt 209, Boston, MA 02116

Chair's Introduction—1:35

Invited Papers

1:40

2pSA1. Recollections of my father: Miguel C. Junger. Sebastian Junger (P.O. Box 906, Truro, MA 02666, sjunger5@aol.com)

Thoughts and reflections on my father as such an inspirational and unforgettable person.

2:00

2pSA2. Miguel Junger: Legacy contributions to the field of structural acoustics. David Feit (ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, feit.d@att.net)

The field of structural acoustics, i.e., the theory of acoustic radiation and scattering from elastic structures, developed primarily in the last half of the 20th century in response to Navy needs. Miguel Junger and colleagues at Cambridge Acoustical Associates under the sponsorship of the Office of Naval Research Sound and Structures made seminal contributions to the field. This presentation reviews some of this research, which ultimately was included in the book, "Sound, Structures and Their Interaction."

2:20

2pSA3. Three curious results of radiation loading calculations. Joel M. Garrelick (505 Tremont St., Unit 209, Boston, MA 02116, joelgarrelick@aol.com)

One of the many joys in reading Miguel C. Junger's contributions to the journal and elsewhere was the physical interpretations that accompanied theoretical results, which were at times unanticipated but always enlightening. In three instances however, he found the task of offering such an explanation particularly challenging and was beguiled by the fact. This is documented in his paper "Three apparently paradoxical results of sound radiation theory" [J. Acoust. Soc. Am. **106**(3, Pt. 1), 1589–1590 (1999)]. The specific results are as follows: (1) At frequencies above coincidence, the power radiated by a point driven infinite thin plate exposed to a low impedance fluid is equal to the power radiated by that plate when submerged in a high impedance fluid. (2) At low frequencies, the entrained mass that is associated with a translating circular piston fully submerged in an acoustic medium is equal to that acting on the piston when it is baffled and exposed to the medium on one side only. (3) The low frequency admittance of a fluid loaded infinite thin plate driven by a point force exhibits a spring-like reactance and yields a phase angle magnitude that is equal to that of the plate when driven by a line force. These three topics are revisited in this paper with an attempt to distinguish between paradox and coincidence.

2:40

2pSA4. Acoustic waves in violently collapsing bubbles. Thomas L. Geers (Mech. Eng., Univ. of Colorado, Campus Box 427, Boulder, CO 80302, geers@spot.colorado.edu)

Among Miguel Junger's many contributions to acoustical science and engineering were his papers and presentations on bubble acoustics. Among his many contributions to the well being of his colleagues at Cambridge Acoustical Associates was the mentoring of this presenter during the latter years of graduate study at MIT. Hence, this presentation in this session. In an evaluation of five reduced models for spherically symmetric bubble collapse and rebound [J. Appl. Phys. **112**, 054910 (2012)], it was found that some recent models, which incorporate wave propagation in both the external fluid and internal gas, did not perform as well as the long-established model by Keller and Kolodner, which incorporates wave propagation in the fluid but not in the gas [J. Appl. Phys. **27**, 1152–1161 (1956)]. Performance was assessed through comparisons against response histories produced by finite-difference solution of the Euler equations under adiabatic conditions. Further investigation revealed that neither acoustic-wave nor shock-wave propagation in the gas was apparent, but that a standing wave in the gas was. This prompted an enhancement of the Keller and Kolodner model that accounts for the standing wave. The formulation and evaluation of the enhanced model is the subject of this presentation.

3:00–3:20 Break

3:20

2pSA5. Poisson coupling in the in vacuo dynamics of an infinite cylindrical shell. Rudolph Martinez (Acoustics, Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, rmartinez@aphysci.com)

This paper applies a perturbation analysis to the "n = 0" in vacuo dynamics of an infinite cylindrical shell as presented in their exact form by Junger and Feit in *Sound, Structures, and Their Interaction*. The small parameter in the asymptotic theory is the square of the Poisson ratio ν. Two sets of results emerge: (1) Relating to the system's dispersion relation and therefore irrespective of the type of loading, the approximate roots of the governing cubic polynomial display explicitly the mutual contamination and influence of axial compression and latent skin flexure, the latter becoming actual past the ring frequency ωra/cp = 1. (2) In the parametric range of small values of ωa/cp, and for a radial ring force, our asymptotic expansions establish that the shell's normal-to-surface response is one of local reaction to zeroth order in ν2. It is not until the analysis is carried out to O(ν2) that wave propagation from axial compression and nearfield skin flexure begin to assert themselves away from the driven station.

3:40

2pSA6. Sound radiation by parallel coated plates separated by a fluid layer: Now and Then. Ann Stokes (Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, astokes@aphysci.com)

In 1987, I had the opportunity to work with Miguel Junger on a paper entitled "Sound Radiation by parallel coated plates separated by a fluid layer." A critical piece of the analysis involved analytically inverting a 5x5 matrix, which resulted in an extremely complex algebraic expression for the far-field of a point-excited 5-layer configuration. In order to understand the physical mechanisms, and validate the results, Dr. Junger developed asymptotic forms of the solution for familiar configurations and made use of spring-mass models to understand resonance enhancement effects. This talk introduces our current semi-analytical waveguide analysis of sound radiation by parallel, multi-layer plates. An advantage of the waveguide approach is that finite-length structures can be modeled, and axial discontinuities, including ribs and wavebearing bulkheads, can also be included very efficiently. Physical mechanisms are interpreted in terms of the propagating and evanescent waves of the structures. A focus of this talk is the extension of the perfectly matched layer (PML) model of the exterior fluid to waveguide models.

4:00

2pSA7. Cooperating with Miguel on improvements of the acoustical product - SOUNDBLOX. Klaus Kleinschmidt (Retired, 100 Newbury Court, Concord, MA 01742, ksquare@comcast.net)

Some 15 years after joining Miguel's consulting firm Cambridge Acoustical Associates around 1960 he asked me to help him with improving the sound absorbing quality of a patented slotted concrete block. The original concept of the block, sold under the trademark, SOUNDBLOX, was create a Helmholtz resonator by providing a slot in one face of a standard concrete block whose natural frequency would match the fundamental frequency of a common noise source, transformer hum. A broader absorption spectrum was desired to expand the rather limited applications of the original design to compete with certain acoustical materials widely used in schools, gymnasiums, auditoriums, and swimming pools. This presentation will describe the nature of our cooperation and the successes and failures of various concepts. The fact that the product, first introduced in the late 1950's, is still on the market speaks well of our collaboration.

Contributed Papers

4:20

2pSA8. Sound radiation from finite surfaces. Jonas Brunskog (Acoust. Technol., DTU Elec. Eng., Elektrovej, Bldg. 352, Kgs. Lyngby DK-2800, Denmark, jbr@elektro.dtu.dk)

A method to account for the effect of finite size in acoustic power radiation problem of planar surfaces using spatial windowing is developed. Cremer and Heckl presents a very useful formula for the power radiating from a structure using the spatially Fourier transformed velocity, which combined with spatially windowing of a plane waves can be used to take into account the finite size. In the present paper, this is developed by means of a radiation impedance for finite surfaces, which is used instead of the radiation impedance for infinite surfaces. In this way, the spatial windowing is included in the radiation formula directly, and no pre-windowing is needed. Examples are given for the radiation efficiency, and the results are compared with results found in the literature.

4:40

2pSA9. Modal contributions to sound radiated from a fluid loaded cylinder. Herwig Peters, Nicole Kessissoglou (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, UNSW, Sydney NSW 2052 Australia, z3268667@student.unsw.edu.au), and Steffen Marburg (LRT4 - Inst. of Mech., Universitat der Bundeswehr Munchen, Neubiberg, Germany)

A modal decomposition technique to compute the individual modal contributions to the sound radiated from a cylindrical shell submerged in water is presented. The wet structural modes are calculated by means of a polynomial approximation and symmetric linearization of the underlying nonlinear eigenvalue problem. A Krylov subspace technique is used to reduce the model size of the structural domain, while the fluid domain remains unchanged. Results for the radiated sound power and sound pressure directivity are presented for groups of circumferential modes with common mode number. Under axial and transverse excitation, the cylinder breathing and bending modes are respectively the major modes contributing to the radiated sound at low frequencies. The contribution of the rigid body modes to the radiated sound is also observed.

5:00

2pSA10. Acoustic radiation mode shapes for control of plates and shells. William R. Johnson (Mech. Eng., Brigham Young Univ., 1085 N. 1750 W., Provo, UT 84604, will.johnson@byu.edu), Pegah Aslani (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Daniel R. Hendricks (Mech. Eng., Brigham Young Univ., Provo, UT)

During the advent of active structural acoustic control, attempts were made to target and control structural vibration mode shapes to reduce radiated sound power. In the late 1980s and early nineties work on acoustic radiation mode shapes developed an alternative way to target structural acoustic radiation. By attempting to control the radiation mode shapes, contributing structural modes could be more easily targeted. Radiation mode shapes have been examined previously for rectangular plates. The method has been extended to demonstrate radiation mode shapes of circular plates and cylindrical shells. Certain spatial derivatives of plate vibration have been found to be highly correlated with the most efficiently radiating radiation mode shapes at low frequencies. A weighted sum of these spatial derivatives is proposed as a new, generalized control metric.

Session 2pSCa

Speech Communication: Variability in Speech Intelligibility: Behavioral and Neural Perspectives

Rajka Smiljanic, Cochair
Linguistics, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198

Bharath Chandrasekaran, Cochair
Communi. Sci. and Disorders, Univ. of Texas at Austin, Austin, TX 78712

Sven Mattys, Cochair
Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom

Chair's Introduction—12:55

Invited Papers

1:00

2pSCa1. Processing speech of varying intelligibility. Rajka Smiljanic (Linguistics, Univ. of Texas-Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu) and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas-Austin, Austin, TX)

In this talk, we examine how variation in intelligibility impacts speech processing with insights from behavioral and neuroimaging studies. We discuss a set of experiments that explore the extent to which listener-oriented speaking style changes, sentence context, and visual information contribute to enhanced word recognition in challenging listening conditions. We further examine whether these same enhancements impact speech processing beyond word recognition, namely recognition memory for sentences. The results show that both signal-related and contextual enhancements lead to improved speech recognition in noise and, crucially, to a substantially better sentence recall. We then discuss studies examining neural mechanisms involved in processing speech of varying intelligibility using fMRI. Previous fMRI studies have examined speech intelligibility by using artificially degraded speech stimuli. Few studies have examined natural variation in intelligibility. Here we present neuroimaging data from two studies that examine natural variations in speech intelligibility (native vs. non-native speech; audio versus audiovisual speech). Overall, combined insights from behavioral and neuroimaging studies provide important additions to our understanding of how different sources of variability in the speech signal affect speech processing and memory representations.

1:20

2pSCa2. The impact of variation in phoneme category structure on consonant intelligibility. Valerie Hazan, Rachel R. Romeo, and Michele Pettinato (Speech, Hearing and Phonet. Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

Newman *et al*. [J. Acoustic. Soc. Am. **109**, 1181–1196 (2001)] suggested that phoneme identification accuracy and speed for a given talker was affected by the degree of variability in their production of phoneme categories. This study investigates how intra-talker variability in the production of two phoneme contrasts varies with age and gender, and how this variability affects speed of perceptual processing. Multiple iterations of tokens differing in initial consonants (/s/-/ʃ/, /p/-/b/) were collected via picture elicitation from 40 adults and 31 children aged 11 to 14; measures of within-category dispersion, between-category distance, overlap, and discriminability were obtained. While females produced more discriminable categories than males, children produced farther yet more dispersed—and thus similarly discriminable—categories than adults. Variability was contrast-specific rather than a general talker characteristic. Tokens with initial /s/-/ʃ/ from pairs of adult and child talkers varying in between-category distance or overlap were presented for identification. The presence of overlap had a greater effect on identification accuracy and speed than between-category distance, with strongest effects for adult speakers, but reaction time correlated most highly with within-category dispersion. These data suggest that talkers who are less consistent in their speech production may be perceived less clearly than more internally consistent talkers.

Contributed Paper

1:40

2pSCa3. On the tolerance of spectral blur in the perception of spoken words. Robert E. Remez, Chloe B. Cheimets, and Emily F. Thomas (Dept. of Psych. and Prog. in Neurosci. & Behavior, Barnard College, Columbia Univ., 3009 Broadway, New York, NY 10027, remez@columbia.edu)

How does a listener resolve linguistic properties conveyed by speech? Many descriptions of perception attribute a causal role to brief spectral details in narrow frequency ranges. Perceptual standards allow far more variety, revealed by the robustness of perception of many kinds of distorted speech. The present study considered the effects of spectral blur on the recognition of spoken words. Listeners heard successive presentations of noise-vocoded easy and hard words. The number of spectral channels composing the word increased with each presentation, reducing blur within a trial. Four conditions counterbalanced the number of presentations of each word in a trial, 3 or 5, and the severity of initial blur, either 1 or 5 channels. In all conditions, the final presentation had 9 bands, yielding a net blur reduction of 4 or 8 bands. These conditions were also tested with the instruction that the

words would become clearer during each trial. A control used two repetitions of each word at 9 spectral bands. Across the tests, exposure to spectral blur impaired the recognition of easy and hard words alike regardless of the listener's belief during presentation. However, intelligibility of hard words declined sharply when subjects were instructed to attend to the continuity and successive decrease in blur within a trial. The pattern of results exposes the role of attention, uncertainty, and spectral resolution in the phonetic contribution to word identification.

Invited Papers

2:00

2pSCa4. Intelligibility of interrupted speech in listeners of different age and hearing status. Valeriy Shafiro, Stanley Sheft, Robert Risley (Commun. Disord. & Sci., Rush Univ. Med. Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), and Brian Gygi (NIHR Nottingham Hearing Biomed. Res. Unit, Nottingham, United Kingdom)

In most real-world communicative environments, speech signals are fragmented and incomplete due to masking by other concurrent sounds. Experiments in the perception of gated and time-compressed speech provide a useful approach for systematically investigating factors involved in the perception of temporally fragmented speech. In the current work, the intelligibility of spoken sentences was measured following periodic signal interruption at various square-wave gating rates with time compression applied by either omitting or doubling silent intervals during gated-off times. Across interruption rates (0.5–16 Hz), speech perception of younger and older normal-hearing and older hearing-impaired adults was similar for slow interruption rates, but differed substantially for the faster rates. This rate-dependent variation was consistently found for different interruption methods and conditions. Importantly, speech perception at fast interruption rates correlated strongly with pure-tone thresholds and performance on speech-in-noise tests, while it did not correlate with results from tests of working memory or spectro-temporal pattern discrimination. These findings suggest that different perceptual processes are involved in the perception of interrupted speech at slow and at faster interruption rates, and have implications for developing diagnostic tests applicable to real-world listening environments. [Work supported by NIH.]

2:20–2:40 Break

2:40

2pSCa5. Impaired speech recognition under a cognitive load: Where is the locus? Sven Mattys (Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom, sven.mattys@york.ac.uk)

Improving the validity of speech-recognition models requires an understanding of the conditions in which speech is experienced in everyday life. Listening conditions leading to a degradation of the signal—noise, competing talkers, disordered speech—have received most of the attention in that literature. But what about adverse conditions that do not alter the integrity of the signal, such as listening to speech under a non-auditory cognitive load (CL)? Drawing upon a variety of behavioral methods, this presentation investigates the effects of a concurrent attentional or mnemonic task on the relative reliance on acoustic cues and lexical knowledge during speech-perception tasks. The results show that listeners under CL downplay the contribution of acoustic detail and increase their reliance on lexical-semantic knowledge. However, greater reliance on lexical-semantic knowledge under CL is a cascaded effect of impoverished phonetic processing, not a direct consequence of CL. Ways of integrating CL into the functional architecture of existing speech-recognition models are discussed.

Contributed Paper

3:00

2pSCa6. Development of the auditory evoked potential to amplitude rise time and rate of formant transition of speech sounds. Antoine Shahin and Allen Carpenter (Otolaryngology, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212, tonyshahin@gmail.com)

We investigated the morphology of the N1-P2 auditory evoked potential (AEP) to changes in amplitude rise time (ART) and rate of formant transition (RFT) of consonant-vowel pairs (CVs) in 4–6-year olds and adults. In the AEP session, individuals listened passively to the CVs /ba/, /wa/, and a /ba/ with a superimposed slower-rising /wa/ envelope (/ba/wa). In the behavioral session, individuals listened to the same stimuli and judged whether they heard a /ba/ or /wa/. In 6-year olds and adults, the N1-P2 amplitude reflected a change in RFT (/ba/wa and /wa/) but not in ART (/ba/ and /ba/wa). In contrast, in the 4–5-year olds, the poorly developed N1-P2 did not show specificity to changes in RFT or ART. Behaviorally, 6-year olds and adults relied more strongly on RFT cues (classified /ba/wa as /ba/) during phonetic judgments, as opposed to 4–5-year olds, which utilized both cues equally (chance level). Our findings suggest that following age 4–5, the development of the N1-P2 AEP, representing maturation of synaptic connections in the superficial layer of the auditory cortex, reflects a shift toward weighting of spectral dynamics more than amplitude dynamics during /ba/ - /wa/ phonetic categorization.

Invited Paper

3:20

2pSCa7. Cortical responses to degraded speech are modulated by linguistic predictions. Jonathan E. Peelle (Dept. of Otolaryngol., Washington Univ. in St. Louis, 660 South Euclid, Box 8115, St. Louis, MO 63110, peellej@ent.wustl.edu)

Our perceptual experience is formed by combining incoming sensory information with prior knowledge and expectation. When speech is not fully intelligible, non-acoustic information may be particularly important. Predictions about this degraded acoustic signal can be provided intrinsically (if the speech is still partially intelligible) or extrinsically (for example, by presenting a written cue). I will discuss results from studies in which the neural response to speech was measured using magnetoencephalography (MEG), with speech clarity parametrically manipulated using noise vocoding. In one study, we found that during sentence processing the phase of ongoing

cortical oscillations is matched to that of the acoustic speech envelope in the range of the syllable rate (4–8 Hz). Critically, this phase-locking was enhanced in left temporal cortex when speech is intelligible. In a separate study of single word listening, accurate predictions provided by written text enhanced subjective clarity and changed the response in early auditory processing regions of temporal cortex. Both experiments thus highlight neural responses in brain regions associated with relatively low-level speech perception. Together these findings support the ability of linguistic information to provide predictions that shape auditory processing of spoken language, particularly when acoustic clarity is compromised.

3:40–4:20 Panel Discussion

TUESDAY AFTERNOON, 4 JUNE 2013 516, 2:20 P.M. TO 5:00 P.M.

Session 2pSCb

Speech Communication: Speech Intelligibility (Poster Session)

Yingjiu Nie, Chair

Speech Lang. Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., SE, Minneapolis, MN 55455

Contributed Papers

All posters will be on display from 2:20 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:20 p.m. to 3:40 p.m. and contributors of even-numbered papers will be at their posters from 3:40 p.m. to 5:00 p.m.

2pSCb1. Effect of speech clarity on perception of interrupted meaningful and anomalous sentences. Rajka Smiljanic (Linguistics, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu), Stan Shaft (Commun. Disord. and Sci., Rush Univ. Med. Ctr., Chicago, IL), Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas-Austin, Austin, TX), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Med. Ctr., Chicago, IL)

The influence of speech clarity on the perception of interrupted speech was examined for sentences distinguished by the presence of semantic-contextual cues. Semantically meaningful and anomalous sentences produced in either conversational or "clear" speech were periodically interrupted at gating rates ranging from slow (0.5 Hz) to fast (24 Hz) and presented to 32 native English listeners. At slow rates, speech perception may be based on integration of whole syllables and words, with "glimpsing" of (sub)phonemic segments playing a role at faster rates. Our results show that semantic context and speech clarity had a significant rate-dependent impact on the intelligibility of interrupted speech. At the lowest rates, intelligibility differences between conditions were minimal. Overall, interruption was most deleterious for anomalous conversational sentences. Such effects were seen even at the highest gating rate of 24 Hz for which interruption effects are generally minimal. The magnitude of the clear-speech benefit varied with gating rate for the two types of sentences, starting at 1 Hz for meaningful and 2 Hz for anomalous sentences. Acoustic-phonetic enhancements of clear speech thus "shifted" contextual benefit to lower gating rates. The implications of these results for our understanding of different time scales of speech processing will be discussed.

2pSCb2. Speech understanding in babble noise and syllabic-range temporal processing in aging. Pierre Divenyi (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, pdivenyi@ccrma.stanford.edu)

The ability to understand speech in multitalker noise was measured in a group of 46 elderly (71.6 +/− 6.03 years) having at worst moderate presbycusic hearing loss, using the SPIN test with the target and an 8-talker babble presented either monaurally (left or right) or, using generic HRTF spatialization, with the target at 45 degrees right or left flanked by two independent 4-talker babble on each side. The subjects' temporal processing was measured as (1) the reverberation time yielding a 15% decrease in word comprehension and (2) as the minimum modulation index necessary to segregate two streams generated by modulating, at an average frequency of 4.375 Hz, two carriers consisting of harmonics 4, 5, and 6 having fundamental frequencies of 107 and 189 Hz, that produced two simultaneous three-pulse envelopes, one accelerating and one decelerating. With effects of hearing loss under control, the data exhibited correlations between intelligibility in babble and temporal processing that were significant for a speech-to-babble ratio of 4 dB but not for one of 8 dB. The results thus suggest that under unfavorable acoustic conditions the elderly achieve speech understanding by focusing on prosodic rhythm on the syllabic level. [Work supported by NIA and the VA Medical Research.]

2pSCb3. The effects of temporal envelope confusion on listeners' phoneme and word recognition. Yingjiu Nie, Adam Svec, Peggy B. Nelson, and Benjamin Munson (Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, niex0008@umn.edu)

Broadened auditory filters in listeners with hearing loss may result in listeners' increased reliance on temporal envelope cues for understanding speech. Previous data have shown that background noise may affect hearing-impaired (HI) listeners by negatively affecting the temporal envelope cues in speech. The current study investigates additional HI listeners' understanding of vocoded spondees in the presence of fluctuating and stationary background noise. Stimuli were 8- and 32-channel noise vocoded double spondees, high-pass filtered at 1426 Hz. New data confirmed the previous finding that temporal envelope confusion in HI listeners resulted in speech understanding that is poorer in fluctuating noise (at a rate of 4 Hz) than in stationary noise. Preliminary analysis suggests HI listeners experience significant envelope confusion for both 8- and 32-channel vocoded stimuli. Additional analysis of phoneme errors suggests that envelope confusion affects HI listeners' perception of both consonants and vowels. Further analysis of j-factors will indicate the relationship of phoneme to whole word understanding for vocoded speech in noise. Results confirm the importance of temporal envelope cues for phoneme and syllable recognition for listeners with hearing loss. [Work supported by NIH DC008306 to PB Nelson.]

2pSCb4. Quality of voices processed by hearing aids: Intra-talker differences. Ramesh Kumar Muralimanohar, Caleb Kronen, Kathryn Arehart, James Kates (Dept. of Speech, Lang., and Hearing Sci., UCB 409, Univ. of Colorado at Boulder, Boulder, CO 80309, muralima@colorado.edu), and Kathleen Pichora-Fuller (Dept. of Psych., Univ. of Toronto, Toronto, ON, Canada)

The effects of hearing aid signal processing depend on the voice characteristics of a talker. For example, we have found that the perceptual and acoustic consequences of hearing aid signal processing vary across talkers

and that these effects can be explained, in part, by the acoustic differences between the different voices. However, we find that different utterances spoken by the same talker are also differentially affected by the hearing aid signal processing. In this study, we quantified the acoustic and perceptual consequences of hearing aid signal processing on several utterances spoken by the same talker. The voices were presented in quiet and in the presence of babble noise (10 dB SNR). The voices were processed with varying amounts of frequency compression, wide dynamic range compression and noise suppression (spectral subtraction). We considered intra-talker variability using an objective quality metric (Hearing Aid Sound Quality Index) and perceptual ratings of quality. We analyzed sentences from several different speech corpora, including the Hearing-in-Noise Test (HINT), IEEE, and TIMIT. The results showed interactions between the effects of signal processing and the acoustic characteristics of specific utterances spoken by an individual. [Work supported in part by GN ReSound and NSERC.]

2pSCb5. Perceptual confusability of French vowels. Kathleen C. Hall (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, kathleen.hall@ubc.ca) and Elizabeth Hume (Linguistics, Univ. of Canterbury, Christchurch, New Zealand)

The confusability of sounds is argued to both reflect phonological structure [e.g., Boomershine *et al.* (2008)] and be a source of phonological variability and change [e.g., Ohala (1981), Hume (1998)]. We present the results of a perception task in which 25 Parisian French-speaking participants identified the French vowels [i e ɛ y ø œ ə a u o ɔ ɔ̃ ɛ̃ ɑ̃], or ∅, in an aC_Ca context, using standard orthography in key words. The durations of the vowel in the vowel-containing stimuli were manipulated to be 0, 15, 25, 50, or 100% of their original durations. We can therefore determine which vowels are most confusable with each other (and thus likely to be the target for either mergers or dissimilatory processes) and which are most confusable with zero (and thus likely to be the target of processes such as deletion, assimilation, and metathesis). Results show high accuracy for [a i y u], even with very short durations; some degree of confusability within the nasal vowels; high confusability rates within the mid-front rounded vowels; and a tendency for zero to be confused with one of the mid-front rounded vowels. These results align with observed phonological patterns in French.

2pSCb6. Effects of phonetic experience on neural processing of English /r/ and /l/ by Korean and Japanese listeners. Lee Jung An, Brett A. Martin, and Glenis R. Long (Ph.D. Prog. in Speech-Lang.-Hearing Sci., The Grad. Ctr., CUNY, 365 Fifth Ave., 7th Fl., New York, NY 10016, lan@gc.cuny.edu)

The effects of phonetic experience on behavioral and neurophysiological processing of English /r/ and /l/ by Koreans and Japanese were compared to speakers of American English. Although English /r/ and /l/ are not phonemic in both Korean and Japanese languages, Koreans have a pseudo phonetic [ɾ]-[ll] model available for perception of English, /r/-/l/ sounds in medial position, while Japanese do not. Speech stimuli were a continuum of synthetic stimuli ranging from perceived /iri/ to perceived /ili/. To date, five subjects in each language group have been tested. As predicted, behavioral results show that English medial /r/ and /l/ were perceived in a categorical manner by Americans, in a categorical-like manner by Koreans and in a non-categorical manner by Japanese. Neural responses tapped by the ACC did not differ significantly between language groups for P1-N1-P2, suggesting little effect of phonetic experience on the encoding of these sounds. In contrast, the T-complex (Tb latency and Ta morphology) differed significantly between groups. The T-complex morphology had double-peaks in the Japanese group. These findings suggest that the T-complex may index the effects of phonetic experience on speech perception.

2pSCb7. Perception of Thai distinctive vowel length in noise. Chutamanee Onsuwan (Linguistics, Thammasat Univ., 2 Prachan Rd., Pranakhon, Bangkok, Bangkok 10200, Thailand, consuwan@tu.ac.th), Charturong Tantibundhit, Nantaporn Saimai, Tanawan Saimai (Elec. and Comput. Eng., Thammasat Univ., Khlongluang, Pathumthani, Thailand), Patcharika Chootrakool, and Sumonmas Thatphithakkul (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

A forced choice identification perception experiment using 150 monosyllabic rhyming-word stimulus pairs (with identical consonants and tone) in four conditions of white Gaussian noise was conducted to explore vowel confusions in Thai, a language with nine monophthongs and length (short-long) contrast for all vowels (e.g., /i/-/i:/ and /o/-/o:/). Each stimulus containing speech and noise portions is equal in length. Perceptual results of 18 vowels from 36 Thai listeners at a noise level (SNR) of -24 dB, where the percent intelligibility is the most interpretable, showed that stimuli with short vowels are more accurately perceived than those with long vowels (93.46 vs. 85.64%) with /o:/ and /e:/ as the most confusable. Interestingly, asymmetrical confusions are observed with very few short vowels being misperceived as long vowels, but a larger number of long vowels misperceived as short. Consistent with previous studies of perception of English vowels in white noise [e.g., Benki (2003)], the findings confirm perceptual robustness of vowel height (correlating with F1) over vowel front/backness (correlating with F2). Lastly, an analysis for listeners' misidentified responses shows that the listeners generally favor short over long vowels.

2pSCb8. Effects of semantic predictability and dialect variation on vowel production in clear and plain lab speech. Rory Turnbull and Cynthia G. Clopper (Linguistics, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43215, turnbull@ling.osu.edu)

Speech addressed to a non-native or hearing impaired listener features longer, more peripheral vowels. In addition, more extreme dialect-specific forms are produced in semantically predictable contexts, and less extreme forms (more standard forms) in unpredictable contexts. This study investigated the interactions between predictability and speaking style on Southern American English monophthongization of the vowel /aj/. The Midland dialect of American English served as the comparison. Participants read a set of sentences with monosyllabic target words in sentence-final position. Target words varied in semantic predictability based on the preceding sentential context. Each set of sentences was produced twice by each participant—first as if talking to a friend ("plain" speech) and again as if talking to a non-native or hearing impaired listener ("clear" speech). The duration, dispersion, and trajectory length of the vowel in each target word were measured. Preliminary results suggest that, as expected, Southern /aj/ has a shorter trajectory length than Midland /aj/, and in both dialects, /aj/ has a shorter trajectory length in clear speech than plain speech. However, these processes do not interact with each other or with semantic predictability, suggesting that style and predictability effects are independent of the realization of some dialect variants.

2pSCb9. Continuous recognition memory for spoken words in noise. Susanne Brouwer (Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, smbrouwer@hotmail.com)

Previous research has shown that talker variability affects recognition memory for spoken words [Palmeri *et al.*, (1993)]. This study examines whether additive noise is similarly retained in memory for spoken words. In a continuous recognition memory task, participants listened to a list of spoken words mixed with noise consisting of a pure tone or of high-pass filtered white noise. The noise and speech were in non-overlapping frequency bands. In experiment 1, listeners indicated whether each spoken word in the list was "old" (heard before in the list) or "new." Results showed that listeners were as accurate and as fast at recognizing a word as old if it was repeated with the same or different noise. In experiment 2, listeners also indicated whether words judged as "old" were repeated with the same or with a different type of noise. Results showed that listeners benefitted from hearing words presented with the same versus different noise. These data suggest that spoken words and temporally overlapping but spectrally non-overlapping noise are retained or reconstructed together for explicit, but not for implicit recognition memory. This indicates that the extent to which noise variability is retained seems to depend on the depth of processing.

2pSCb10. When spectral smearing can increase speech intelligibility. James A. Bashford, Richard M. Warren, and Peter W. Lenz (Psychology, Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201, bashford@uwm.edu)

Sentences were reduced to an array of 16 effectively rectangular bands (RBs) having center frequencies ranging from 0.25 to 8 kHz spaced at $^1/_3$-octave intervals. Four arrays were employed, each having uniform subcritical bandwidths which ranged from 40 to 5 Hz. The 40 Hz width array had intelligibility near ceiling, and the 5 Hz array about 1%. The finding of interest was that when the subcritical speech RBs were used to modulate RBs of noise having the same center frequency as the speech, but having bandwidths increased to a critical (ERBn) bandwidth at each center frequency, these spectrally smeared arrays were considerably more intelligible in all but the 40 Hz (ceiling) condition. For example, when the 10 Hz bandwidth speech array having an intelligibility of 8% modulated the ERBn noise array, intelligibility increased to 48%. This six-fold increase occurred despite the elimination of spectral fine structure and the addition of stochastic fluctuation to speech envelope cues. (As anticipated, conventional vocoding with matching bandwidths of speech and noise reduced the 10-Hz-speech array intelligibility from 8% to 1%.) These effects of smearing confirm findings by Bashford, Warren, and Lenz (2010) that optimal temporal processing requires stimulation of a critical bandwidth. [Work supported by NIH.]

2pSCb11. Can physical metrics identify noise reduction settings that optimize intelligibility? Gaston Hilkhuysen and Mark Huckvale (Speech, Hearing and Phonet. Sci., Univ. College London, 31, Chemin Joseph Aiguier, Marseille 13402, France, ghilkhuysen@gmail.com)

Noise reduction algorithms often include adjustable parameters. Their settings can have important consequences for the intelligibility of an enhanced noisy speech signal, but it is not clear how to choose the best settings. In previous work, we have found that even experienced listeners prefer non-optimal settings, which degrade intelligibility. Measuring the effect of settings directly using an intelligibility listening task is often infeasible because of time, cost, or the large number of parameter combinations. In this paper we investigate whether physical intelligibility metrics can provide an efficient mean to optimize settings for noise reduction. A number of intelligibility metrics can be found in the recent research literature. An evaluation of these metrics has shown disappointing performance in their abilities to predict the absolute intelligibility of enhanced noisy speech. However, to find optimal settings, it may be sufficient to predict the relative change in intelligibility caused by a parameter change. Five metrics were used to predict the optimal settings of two parameters of a noise reduction system. The change in each metric with parameter settings are compared to listeners' performance on an intelligibility test with the enhanced signals. We show which metrics are best suited.

2pSCb12. Perception of English vowels and use of visual cues by learners of English and English native speakers. Yasna I. Pereira (Speech, Hearing and Phonet. Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, yasnap@hotmail.com)

English vowels may be difficult to discriminate for many learners of English (L2 learners). Research in L2 speech perception has shown that the use of visual cues improves speech perception, at least for visually salient contrasts. This study investigated the use of visual cues in the perception of English vowels by L2 Advanced learners (Spanish native speakers) and English native speakers (ENS). Thirty-seven L2 learners and 20 ENS were given a vowel test that presented real CVC words in audio (A), audiovisual (AV), and video-alone (V) mode. The A and AV conditions were presented in noise (-10 dB SNR) to ENS and in quiet to L2 learners. For ENS, identification rates were significantly higher in AV than in A condition, suggesting there were visual cues to vowel identity. For L2 learners, A scores were significantly lower than for ENS, and AV scores did not differ significantly from results in A mode. This suggests low sensitivity to visual cues to vowel identification, though L2 learners achieved better than chance scores when forced to attend to visual information in the V mode. These results support previous findings of relatively poor sensitivity to visual cues to phoneme identity in L2 learners.

2pSCb13. The development of clear speech strategies in 9–14 year olds. Michèle Pettinato and Valerie Hazan (Speech, Hearing and Phonet. Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, m.pettinato@ucl.ac.uk)

This study investigated the development of global clear speech strategies of child talkers. Two groups of 20 talkers aged 9–10 (children) and 13–14 (teens) were recorded in pairs while they carried out spot the difference picture tasks (diapix), either hearing each other normally (NB condition) or with one talker hearing the other via a three-channel noise vocoder (VOC condition). Acoustic-phonetic analyses focused on the talker having to overcome the communication barrier. Data were compared to those for twenty of the adults in Hazan and Baker [J. Acoust. Soc. Am. **130**, 2139–2152 (2011)]. The three age groups did not differ in task transaction time for NB, but children took significantly longer to complete the task in VOC than teens or adults who took equally long. Children spoke at a slower speech rate overall than teens, while teens and adults did not differ; all groups significantly reduced their speech rate in VOC relative to NB. Adults hyperarticulated vowels in VOC, but children and teens showed only minor adaptations. These results suggest that although 9–10 year olds use some strategies to clarify their speech in difficult conditions, other strategies continue to develop into late adolescence.

2pSCb14. Consonant confusability and its relation to phonological dissimilarity. Sameer ud Dowla Khan (Linguistics, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, sameeruddowlakhan@gmail.com)

Consonant similarity can be measured indirectly through a language's phoneme inventory, lexicon (e.g. cooccurrence restrictions), or phonology (e.g., processes that take similarity or dissimilarity into account). It can also be measured more directly as confusability in a perception task. Thus far, consonant similarity in Bengali has only been measured indirectly, through the inventory, lexicon, and phonology. Previous studies [Khan (2006)] claim that Bengali speakers judge the similarity of consonants in echo reduplication, where the initial consonant of the base is systematically replaced with a phonologically dissimilar consonant in the reduplicant, e.g., kashi "cough" > kashi-tashi "cough, etc." but thonga "bag" > *thonga-tonga > thonga-fonga "bags, etc."). This measurement of similarity assumes a set of features assigned language-specific weights; for example, [voice] is weighted more heavily that [spread glottis], to explain why speakers treat the pair [t, th] as more similar than the pair [t, d]. But does the measurement of similarity inherent in the echo reduplicative construction correspond directly to the relative perceptibility of different consonant contrasts? The current study examines data collected in a perception experiment, comparing the relative confusability of Bengali consonants produced in noise with the claims of phonological notions of similarity associated with echo reduplication.

2pSCb15. Ambiguity related to French liaisons: The role of bottom-up and top-down processes. Mireille Babineau and Rushen Shi (Psychology, Université du Québec à Montréal, Département de Psychologie, Université du Québec à Montréal, C.P. 8888 succursale Centre-ville, Montreal, QC H3C 3P8, Canada, babineau.mireille@courrier.uqam.ca)

French liaison is a phonological process involving the surfacing of an underlying floating consonant as the onset of a vowel-initial word when the word is preceded by a liaison-causing word. We used vowel- and consonant-initial ambiguous targets following four liaison-related contexts /z/, /n/, /t/, /r/ (e.g., ces onches - ces zonches). Targets included nouns and pseudonouns. Quebec-French-speaking adults performed two tasks (production, discrimination). One bottom-up (acoustic cues) and three top-down (noun token frequency, word onset probability, contextual liaison knowledge) factors were investigated in the production task. Participants had to produce the last word upon hearing each phrase. Their productions thus reflected their interpretation of the onset of the liaison-ambiguous words. Perception of acoustic cues was also tested in the discrimination task: participants judged if two phrases were different or same. No perception of any acoustical distinction was revealed in the tasks: participants often produced the intended target incorrectly, and differently intended phrasal pairs were judged same (e.g., ces onches - ces zonches). Effects of top-down information were found in the production task. Among the top-down factors, liaison knowledge related to liaison-causing words had a dominant impact on participants' interpretation of the ambiguous phrases, showing the importance of contextual knowledge in lexical recognition.

2pSCb16. The identification of high pass filtered vowels. Jeremy Donai and Dwayne Paschall (Speech-Lang.-Hearing Sci., Texas Tech. Univ. Health Sci. Ctr., 3601 4th St., Lubbock, TX 79430, jeremy.donai@ttuhsc.edu)

Vowels are typically described according to their spectral prominences (i.e., formants). Previous studies have shown that the first three formants provide important information for identification (Hillebrand *et al*. (1995), Miller (1989), Molis (2005), Peterson and Barney (1952)]. The present study

measured identification accuracy for six naturally produced vowels spoken by a male and female talker with these spectral prominences removed. The six hVd tokens for each talker were high-pass filtered to remove the first three formants from the vowels and then identified by 24 normal hearing listeners. Results suggest that listeners identified a majority of the tokens above chance levels. The average identification of the male vowels was 29% (range, 17%–47%), with two vowels identified with nearly 50% accuracy. Average identification for the six female vowels was 53% (range, 37%–72%), with 5 of 6 vowels being identified with over 40% accuracy.

2pSCb17. The effects of surgical masks on speech perception in noise.
Kelsi J. Wittum, Lawrence L. Feth, and Evelyn M. Hoglund (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, wittum.2@osu.edu)

Surgical masks and blood shields worn by anesthesiologists and surgeons in hospital operating rooms may negatively impact speech communication and put patients at needless risk. Young adult subjects listened to sentences from the Speech Perception in Noise Test (SPIN) recorded by a male and female talker. All eight SPIN lists were recorded under three different speaking conditions: (1) speaking normally without any obstruction, (2) wearing a typical surgical mask, and (3) wearing a surgical mask with an attached blood shield. Multi-talker babble was mixed with the SPIN sentences at several signal-to-noise ratios to simulate conversation in noisy environments. Speaker gender and recording conditions were counterbalanced across listeners to control for learning and fatigue effects. SPIN test scores for each of the three types of recordings and both talker genders were compared in order to determine the degradation that blood-shields and surgical masks may have speech communication in the operating room. [Research supported by research grants from the Division of Social and Behavioral Sciences and a scholarship from the College of Arts and Sciences at The Ohio State University.]

2pSCb18. The role of high-frequency envelope fluctuations for speech masking release.
Søren Jørgensen and Torsten Dau (Elec. Eng., Ctr. for Appl. Hearing Res., DTU, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, sjor@elektro.dtu.dk)

The speech-based envelope power spectrum model [sEPSM; Jørgensen and Dau (2011), Jørgensen *et al*. (2013)] was shown to successfully predict speech intelligibility in conditions with stationary and fluctuating interferers, reverberation, and spectral subtraction. The key element in the model was the multi-resolution estimation of the signal-to-noise ratio in the envelope domain (SNR_{env}) at the output of a modulation filterbank. The simulations suggested that mainly modulation filters centered in the range from 1 to 8 Hz contribute to speech intelligibility in the case of stationary maskers whereas modulation filters tuned to frequencies above 16 Hz might be important in the case of fluctuating maskers. In the present study, the role of high-frequency envelope fluctuations for speech masking release was further investigated in conditions of speech-on-speech masking. Simulations were compared to various measured data from normal-hearing and hearing-impaired listeners [Festen and Plomp (1990), Christiansen *et al*. (2013)]. The results support the hypothesis that high-frequency envelope fluctuations ($>$30 Hz) are essential for speech intelligibility in conditions with speech interferers. While the sEPSM reflects effects of energetic and modulation masking in speech intelligibility, the remaining unexplored effect in some conditions may be attributed to, and defined as, "information masking."

2pSCb19. Regional linguistic variations in Canadian French: Do they affect performance on speech perception in noise?
Josée Lagacé, Stéphanie Breau-Godwin, and Christian Giguère (Health Sciences Faculty, University of Ottawa, 451 Smyth Rd, Ottawa, ON K1H 8M5, Canada, josee.lagace@uottawa.ca)

Many audiologists working with a Canadian French population use word recognition tests to measure speech perception abilities in noise. The TMB test ("Test de Mots dans le Bruit") includes four lists of 35 words presented in babble noise. The test is intended to measure the pre-cognitive perceptual stage of auditory processing and does not require understanding of the phonetic differences between speech sounds at a cognitive level. Previous studies examining performance on auditory tests similar to the TMB showed differences between populations speaking the same language but with different accentuations, such as the English spoken in the United States versus the United Kingdom. Variations in performance were attributed to accentuation differences between the speaker and the listener. To the authors' knowledge, no study appears to have investigated the effect of regional linguistic variations of Canadian French on word recognition in noise. Normative data for the TMB are being collected in three regions of Canada: Moncton, Montréal, and Ottawa. Participants are all native speakers of Canadian French, but there are important linguistic variations across the three regions. Knowledge of the effect of regional linguistic variations on the TMB performance will help refine interpretation of test results in the audiology clinics.

2pSCb20. The role of across-frequency envelope processing for speech intelligibility.
Alexandre Chabot-Leclerc, Søren Jørgensen, and Torsten Dau (Ctr. for Appl. Hearing Res., Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds plads, Bldg. 352, Kongens Lyngby 2800, Denmark, alech@elektro.dtu.dk)

Speech intelligibility models consist of a preprocessing part that transforms the stimuli into some internal (auditory) representation, and a decision metric that quantifies effects of transmission channel, speech interferers, and auditory processing on the speech intelligibility. Here, two recent speech intelligibility models, the spectro-temporal modulation index [STMI; Elhilali *et al*. (2003)] and the speech-based envelope power spectrum model [sEPSM; Jørgensen and Dau (2011)] were evaluated in conditions of noisy speech subjected to reverberation, and to nonlinear distortions through either a phase jitter process or noise reduction via spectral subtraction. The contributions of the individual preprocessing stages in the models and the role of the decision metrics were analyzed in the different experimental conditions. It is demonstrated that an explicit across-frequency envelope processing stage, as assumed in the STMI, together with the metric based on the envelope power signal-to-noise ratio, as assumed in the sEPSM, are required to account for all three conditions. However, a simple weighting of the across-frequency variance of the modulation power at the output of the (purely temporal) modulation filterbank is assumed to be sufficient to describe the data, i.e., a joint two-dimensional modulation filterbank might not be required.

2pSCb21. Using landmark detection to measure effective clear speech.
Suzanne E. Boyce, Sarah Hamilton (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Mail Location 379, Cincinnati, OH 45267, Suzanne.Boyce@uc.edu), Joel MacAuslan (S.T.A.R. Corp., Bedford, MA), Jean Krause (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Rajka Smiljanic (Dept. of Linguist., Univ. of Texas at Austin, Austin, TX), and Ann Bradlow (Dept. of Linguist., Northwestern Univ., Evanston, IL)

While the relationship of speaking style to intelligibility under challenging conditions has been established, it is a common observation that some speakers seem to be more intelligible than others for most listeners. In previous work, we have reported that automatic measures based on the technique of Landmark Detection appear to track differences between Clear and Conversational speaking style. One question that remains is whether Landmark measures can be used to predict which speakers are most likely to produce highly intelligible speech. In this study, we took advantage of a set of previously acquired databases covering a total of 31 American English speakers who produced Clear and Conversational Speech to examine correlations between our Landmark-based measures and the Clear Speech productions of highly intelligible speech. Across these databases, we had data on intelligibility for 13 speakers. Results showed that speakers with high overall intelligibility in Clear Speech showed significantly different patterns on Landmark-based automatic measures, compared to speakers with more moderate performance on intelligibility measures. Applications of these results to problems in speech technology, linguistic education, and clinical practice will be discussed.

2pSCb22. Comprehending speech at artificially enhanced rates.
Lucia da Silva, Adriano V. Barbosa, and Eric Vatikiotis-Bateson (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC, Canada, helena.rozario@gmail.com)

This study is part of a larger study comparing the production, perception, and long-term comprehension of natural fast speech and artificially sped-up speech. In the 1950's, Fairbanks and colleagues made the interesting claim that artificially compressing speech to half its duration (twice the

rate) and listening to it twice is comprehended better than listening to the original natural speed utterance once. Since this finding has been disputed [T. Sticht, J. Expr. Ed. **37**, 60–62 (1969)], we focus here on providing baseline results for perception of naturally produced speech at conversational and fast rate and of speech sped up to twice its original speed. Specifically, we compare how well the different stimuli are perceived when presented once, twice in rapid succession, and twice with delays of hours and days between presentations. We evaluate perception for both audio-only and audiovisual stimuli.

2pSCb23. Influence of cochlear implantation on sentence intelligibility and duration. Olga Peskova (Commun. Sci. and Disord., Callier Adv. Hearing Res. Ctr., The Univ. of Texas at Dallas, Office JO3.104, 800 W.Cambell Rd., Richardson, TX 75080, oxp100020@utdallas.edu), Nirmal Kumar Srinivasan (Dept. of Elec. Eng., Univ. of Texas at Dallas, Richardson, TX), Sujin Shin, Madhu Sundarrajan, and Emily Tobey (Commun. Sci. and Disord., Callier Adv. Hearing Res. Ctr., The Univ. of Texas at Dallas, Richardson, TX)

Cochlear implants (CI) allow children with hearing loss (HL) to achieve speech perception and production outcomes that make their spoken speech understandable to normal hearing adult listeners. This capability is characterize by wide variability of scores. In order to understand the factors that contribute to the overall variability, we investigated the effects of duration of cochlear implantation on speech intelligibility and sentence duration over time. Participants were 105 children implanted between the ages of 2 and 4 and tested at 2 time points there they were 8 and 16 years old. Participants repeated McGarr sentences, which vary in length from 3 to 5 to 7 syllables. Recording were analyze using acoustic software to designate the beginning and end of each sentence in listeners who only heard one sentence from one child. Speech intelligibility scores were related statistically to the duration of each sentence. Durations of sentences that approximated that of normal hearing listeners were those with high intelligibility judgment. In addition, it appears that the children with the longest experience of CI use continue to improve their intelligibility. [Sponsored by NIH.]

2pSCb24. Influence of duration on the perception of consonants /x/ and /j/ in Chinese. Li Feng (Unit of Perceptual Psych., Kyushu Univ., Kyushu University, 4-9-1Shiobaru Minami-ku, Fukuoka, Japan, Fukuoka, Fukuoka 815-8540, Japan, rihou1986@yahoo.co.jp), Nakajima Yoshitaka, and Ueda Kazuo (Dept. Human Sci., Kyushu Univ., Fukuoka, Japan)

In the current study, we investigated whether Mandarin Chinese native speakers' perception of /x/ and /j/ was affected by the duration of consonant parts. Two perceptual experiments, in which the method of constant stimuli was employed, were conducted. Six normal-hearing adults (four females) took part in the experiment. Their average age was 24 yr. All subjects were native speakers of Mandarin Chinese. In the first experiment, Chinese syllables that begin with /x/ or /j/ were extracted from a speech database, and the consonant parts were manipulated in terms of duration. As the duration of /x/ was decreased or the duration of /j/ was increased, to a certain extent, the consonant which had been originally /x/ was perceived as /j/, and vice versa. Synthesized noises instead of recorded consonants were utilized in the second experiment, similar effects of the consonant duration appeared.

2pSCb25. Quality of older voices processed by hearing aids: Acoustic factors explaining inter-talker differences. Huiwen Goy, Margaret K. Pichora-Fuller (Psychology, Univ. of Toronto, 3359 Mississauga Rd. North, Mississauga, ON L5L 1C6, Canada, huiwen.goy@utoronto.ca), Pascal van Lieshout (Dept. of Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada), and Kathryn H. Arehart (Dept. of Speech, Lang., Hearing Sci., Univ. of Colorado Boulder, Boulder, CO)

Hearing aid signal processing algorithms are often evaluated with professional recordings of voices. However, hearing aid users often listen to older talkers who may have poorer voices than younger talkers. The purpose of this study was to quantify the extent to which the acoustic and perceptual consequences of hearing aid digital signal processing algorithms differ for talkers that vary in their vocal characteristics. There were six older talkers (3 males and 3 females) selected from a larger database of 79 talkers; their voices had good, moderate, or poor quality based on perceptual data from younger and older listeners. The voices were presented in quiet and in the presence of babble noise (10 dB SNR). The voices were processed with varying amounts of frequency compression, wide dynamic range compression, and noise suppression (spectral subtraction). There were interactions between signal processing and voice characteristics as measured using the Hearing Aid Speech Quality Index (HASQI) and listener ratings of perceived sound quality of the processed speech. In this paper, we examine the possible acoustic sources that may explain these interactions, including inter-talker differences in formant space and pitch variation. [Research supported by NIH, GN Resound, NSERC, and the Canada Research Chairs program.]

2pSCb26. The effect of background noise on the ability to perceive and remember unrelated words in nonnative listeners. Meital Avivi-Reich, Caterina Y. Leung, and Bruce A. Schneider (Psychology, Univ. of Toronto Mississauga, 1909-35 Charles st.W, Toronto, ON m4y 1r6, Canada, me_avv@yahoo.com)

Nonnative listeners find it more difficult to meet the challenges presented by additional background noise than do native listeners [Ezzatian *et al*., Speech Commun. **52**, 919–929 (2010)], but it is not known whether it is more difficult for them to remember what was said in noisy situations than it is for native listeners. Previous studies have acknowledged that the effect of background noise on the ability to perceive and remember unrelated words is greater in older adults than younger adults. The present study investigates auditory memory performance in nonnative younger adults, using a paired-associate paradigm in three conditions: quiet, continuous babble and babble during word presentation only. Noise levels were adjusted to equate for individual differences in the ability to identify single words in noise. The initial results suggest that nonnative listeners perform similarly to native young adults in the quiet and continuous conditions but worse in the babble during the word-presentation-only condition. These results suggest that stream segregation may be slower in nonnative listeners when the masker and the target words start at the same time.

2pSCb27. Advantage of talker differences and spatial separation for speech-on-speech listening in younger and older adults with good audiograms. Jana Besser (ENT/Audiology, VU Univ. Med. Ctr., De Boelelaan 1118, Amsterdam 1081HV, Netherlands, besser.jana@gmail.com), Kathleen Pichora-Fuller (Dept. of Psych., Univ. of Toronto, Mississauga, ON, Canada), and Joost M. Festen (ENT/Audiology, VU Univ. Med. Ctr., Amsterdam, Netherlands)

Older adults have more difficulty than younger adults understanding speech when there is competing speech, even if they have good audiograms. Age-related differences in listening may be due to declines in auditory temporal processing and/or cognition. We administered the LiSN-S [Cameron *et al*. (2011)] to measure speech reception thresholds (SRTs) in younger and older adults with good audiograms. There were four test conditions, in which the target and competing speech were presented with the same or different voices at the same or different locations. Compared to younger listeners, older listeners obtained worse SRTs in all test conditions and they realized less advantage from talker differences and spatial separation between the target and competing speech. For both groups, the results obtained in the four test conditions were strongly associated with each other. We also assessed cognitive abilities and auditory temporal processing in the older adults. LiSN-S results in this group were strongly associated with measures of cognition, measures of temporal processing (tapping the use of fine structure and gap cues), as well as pure-tone averages (PTA) for 9 and 10 kHz, but not PTAs for frequencies in the standard audiometric range.

Session 2pSP

Signal Processing in Acoustics: Acoustic Signal Processing for Various Applications

Harry A. DeFerrari, Cochair
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Philippe-Aubert Gauthier, Cochair
Université de Sherbrooke, 51, 8e Ave. Sud, Sherbrooke, QC J1G 2P6, Canada

Contributed Papers

1:00

2pSP1. Ideal signals and processing for continuous active sonar. Harry A. DeFerrari (Appl. Marine Phys., Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, hdeferrari@rsmas.miami.edu)

The ideal signal for continuous active sonar would be linear in both time and Doppler and have no time or Doppler leakage. Here two signals/signal processing methods are presented that have half the requisite condition—zero time leakage. But we show a way to use the time property to eliminate Doppler interference (clutter) from zero Doppler reverberation (bottomed) sources and from direct arrivals. Here we develop an m-sequence processed with a matched filter approach and a special class of inverted binary sequences processed with a matched-inverse filter. In both cases, there is a perfect pulse correlation property; that is, a large signal when they line up in time and hard zero when they do not. This property is used to develop a Complete Ortho-Normal (CON) data sets in both waveform and pulse response space. The CON pulse response allows elimination of the easily identified zero Doppler reverberation arrivals. Then, reprocessing for Doppler, the zero Doppler signals are cleanly removed as is all of their Doppler leakage. In effect one has the ideal clutter free signal with leakage from direct arrivals and bottom reverberation removed and a target signal displayed in a time-Doppler plane against a noise only background.

1:20

2pSP2. Toward blind reverberation time estimation for non-speech signals. João F. Santos (INRS-EMT, Institute National de la Recherche Scientifique, 800, Rue de La Gauchetière Ouest, Ste. 6900, Montreal, QC H5A-1K6, Canada, jfsantos@emt.inrs.ca), Nils Peters (Qualicomm Technol. Inc., Berkeley, CA), and Tiago H. Falk (INRS-EMT, Institute National de la Recherche Scientifique, Montreal, QC, Canada)

Reverberation time (RT) is an important parameter for room acoustics characterization, intelligibility and quality assessment of reverberant speech, and for dereverberation. Commonly, RT is estimated from the room impulse response (RIR). In practice, however, RIRs are often unavailable or continuously changing. As such, blind estimation of RT based only on the recorded reverberant signals is of great interest. To date, blind RT estimation has focused on reverberant speech signals. Here, we propose to blindly estimate RT from non-speech signals, such as solo instrument recordings and music ensembles. To estimate the RT of non-speech signals, we propose a blind estimator based on an auditory-inspired modulation spectrum signal representation, which measures the modulation frequency of temporal envelopes computed from a 23-channel gammatone filterbank. We show that the higher modulation frequency bands are more sensitive to reverberation than the modulation bands below 20 Hz. When tested on a database of non-speech sounds under 23 different reverberation conditions with early decay time (EDT) ranging from 0.26 to 7.6 s, a blind estimator based on the ratio of high-to-low modulation frequencies outperformed two state-of-the-art methods and achieved correlations with EDT as high as 0.80 for solo instruments and 0.75 for ensembles.

1:40

2pSP3. Feedback active noise control in a crew rest compartment. Delf Sachau and Oliver Pabst (Mechatronics, Helmut-Schmidt-Univ., Holstenhofweg 85, Hamburg, Hamburg D-22043, Germany, sachau@hsu-hh.de)

Active systems for noise cancelation are typically used to reduce low frequency noise (below 500 Hz). In this study, a method to reduce the delay in the control loop of feedback noise control systems is discussed. There are certain applications in which we may choose to control a specific frequency range; hence, a selection of the control-band must be performed. This selection is commonly done with analog filters introducing delay in the signal path. However, in feedback systems, the reference signal is calculated from the error signal which makes their performance more sensitive to these delays. In this contribution, the filtered-error feedback filtered-reference LMS algorithm (FE-FBFxLMS) is analyzed. Consequently, control-band selection is performed in an auxiliary loop removing the additional delay required in the signal path. This method of band selection also allows a reduction of the delay due to the anti-aliasing filters by using faster low-order filters in terms of group delay. With this delay reduction, higher noise attenuation is expected at the cost of a slower convergence rate. This method is applied to control broadband noise in a single channel feedback system in simulation and experiment.

2:00

2pSP4. A polar-coordinate-discretized wave equation finite-difference time-domain simulation for controlling the emission characteristics of sound source. Kota Nakano, Masato Nakayama, Takanobu Nishiura, Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Noji-higashi, Kusatsu, Shiga 525-8577, Japan, cm010064@ed.ritsumei.ac.jp), and Toshiyuki Kimura (Universal Commun. Res. Inst., National Inst. of Information and Commun. Technol., Soraku-gun, Japan)

In sound field simulation with conventional WE-FDTD, the simulated field and wave equation are discretized with symbol time domain and spatial domain with Cartesian coordinate, and sound pressure distribution is calculated. Spatial-discretization with Cartesian coordinate easily achieves calculation and sound synthesis. However, the discretized Cartesian coordinate cannot model sound emission characteristic exactly, because the emission characteristics is under spherical diffusion and discretize Cartesian coordinate cannot define spherical. Emission characteristic of sound source is indispensable for sound field reproduction. Then, we propose a new approach for WE-FDTD method to simulate emission characteristic of sound source object. Polar coordinate discretization is employed in the new approach instead of Cartesian coordinate. The wave equation with spatial Cartesian coordinate is projected onto polar coordinates, and discretized. The discretized polar coordinate is better way to define spherical wave surface than Cartesian one, and it is expected that the emission characteristics are simulated more exactly. The objective evaluations were conducted for the proposed approach. According to the result, the sound field simulation with the new approach can flexibly define exact emission characteristics. However, the discretized interval of polar coordinate is coarse in distance. Verifying the interval limit is important issue in future work.

2:20

2pSP5. Wind noise reduction using empirical mode decomposition. Kohei Yatabe and Yasuhiro Oikawa (Intermedia Art and Sci., Waseda Univ., 59-407 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, k.yatabe@asagi.waseda.jp)

One common problem of recorded sound outdoors is interfusion of wind noise, which has highly non-stationary characteristics. Although there are a lot of noise reduction methods which produce good results for general kinds of noises, most methods perform worse for wind noise due to its non-stationary nature. Therefore, wind noise reduction need special technique to overcome this non-stationarity. Empirical mode decomposition (EMD) is a relatively new method to decompose a signal into several nonlinear and non-stationary bases which are modeled as amplitude and frequency modulated sinusoids that represent wind noise well. Thus, EMD has a possibility to reduce wind noise from recorded sounds in entirely different way from ordinary methods. In this paper, a preliminary discussion of applying EMD to wind noise reduction is presented. Since EMD decomposes a signal into monocomponent bases, it is easier to treat them as analytic signals via Hilbert transform. Our method utilize this characteristics in order to reduce wind noise. The experiment is performed on female voice superimposed with wind noise and shows it possibility and effectiveness.

2:40

2pSP6. Efficient speech encryption using chaotic cat map for code-excited linear prediction based coders in packet networks. Fatiha Merazka (Telecommunications, Univ. of Sci. & Technol. Houari Boumediene, P.O. Box 32 El Alia, Algiers 16111, Algeria, fmerazka@usthb.dz)

The increasing importance of multimedia applications is placing a great insist on content protection and customer privacy. Communications can be intercepted, especially over wireless links. Since encryption can effectively prevent eavesdropping, its use is widely advocated. The codec G. 729 based CS-ACELP algorithm is standardized as voice codec by ITU-T for multimedia and Voice over Internet Protocol (VoIP) applications. In this paper, we introduce a speech encryption method based chaotic cat map algorithm. Cat map extended to two-dimensional NxN matrix. It takes concepts from linear algebra and uses them to change the positions of the values of the matrix. The result after applying the Cat Map will be shuffled signals that contain the same values of the original signals. We applied our encryption scheme to the standard ITU-T G.729 standard speech coder to evaluate its performance. Simulation results show that G.729 based cat map encryption is very efficient since the encrypted speech is similar to a white noise. The perceptual evaluation of speech quality (PESQ) and enhanced modified bark spectral distortion (EMBSD) tests for speech extracted from TIMIT database confirm the efficiency of our proposed scheme.

3:00–3:20 Break

3:20

2pSP7. An investigation into the relationship between sound quality and demodulation ratio for parametric loudspeakers. Daisuke Ikefuji, Masato Nakayama, Takanobu Nishiura, and Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, cm000074@ed.ritsumei.ac.jp)

Recently, parametric loudspeakers with a powerful ultrasound have been utilized for reproduction of sound to particular area. Amplitude of the ultrasound is modulated with a target audible sound. The emitted amplitude modulated wave is demodulated into the target audible sound in the air. Demodulation ratio affects sound quality of the reproduced sound with parametric loudspeakers. Moreover, demodulation ratio depends on the distance between the parametric loudspeaker and the listener. Therefore, the listener should utilize parametric loudspeakers at the suitable distance, which is required for sufficient demodulation of the amplitude modulated wave. Thus, we have proposed the criterion for measuring the demodulation ratio. However, it had not been conducted to investigate a relationship between subjective sound quality and demodulation ratio. In this paper, we investigate the relationship for estimation of the suitable distance. We carried out subjective evaluation experiment for confirming sound quality at each distance. As a result, we confirmed the consistency about fluctuation tendency of the subjective sound quality and demodulation ratio at each distance.

3:40

2pSP8. An adaptive noise power spectral density estimation of noisy speech using generalized gamma probability density function. Xin Dang (Information Sci. and Technol., Grad. School of Sci. and Technol., Grad. School of Eng., Hamamatsu, Shizuoka 432-8012, Japan, f5045013@ipc.shizuoka.ac.jp) and Takayoshi Nakai (Information Sci. and Technol., Grad. School of Sci. and Technol., Hamamatsu, Shizuoka, Japan)

An estimation of the power spectral density (PSD) of noise is a crucial part to retrieve speech in a noisy environment. A novel estimation method for non-white noise of noisy speech on the basis of a generalized Gamma distribution is proposed. Because of highly non-stationary nature of speech, its probability density function (PDF) is difficult to derive using any modeling technique, while a segmental noise is more stationary and can be fitted more accurately by a generalized Gamma PDF, which is a natural extension of the Gaussian modeling of a non-white components distribution. In the experiment, different types of non-white noises are added to the clean speech signal at different SNRs to study the estimation of noise using different types of PDF. It is found that non-white noise spectrums fit more accurately on the generalized Gamma PDF with adaptive parameters instead of a Gaussian distribution function. The reported generalized Gamma PDF model shows the best performance to estimate the noise spectral amplitudes as compared with Minimum Statistics (MS), Speech absence Probability (SAP), and MMSE based PSD estimation methods. The performance of the proposed noise estimation is good when it is integrated with the speech enhancement technique as demonstrated by both the subjective and objective measures.

4:00

2pSP9. Markov random field in speech enhancement: Application for tonal languages. Tanawan Saimai, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Khlongluang, Pathumthani 12120, Thailand, 5310030068@student.tu.ac.th), Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand), and Chai Wutiwiwatchai (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

This paper proposed speech enhancement algorithm based on Markov random field (MRF) model for Thai, a tonal language. Firstly, a noisy speech signal is transformed using the short time Fourier transform (STFT). In so doing, noise is removed and speech is preserved, especially harmonics information as f0 patterns are relevant perceptual cues for lexical tones. The voice activity detector is used to classify each STFT time frame into voiced and unvoiced. Harmonics information is retrieved from each voiced time frame, where four neighborhoods of the analyzed STFT coefficients include its adjacent time frames (left, right) and nearest harmonics (top, bottom). For the unvoiced, four adjacent coefficients (left, right, top, and bottom) are used. A two-state MRF model is used to classify STFT coefficients into speech and noise. Those with speech state are retained, while the rest is set to zero. The enhanced speech is estimated by the inverse STFT. Results from quality evaluation test on four sets of Thai rhyming words corrupted by white noise at SNR levels of 0, 5, and 10 dB showed that the proposed algorithm significantly improved SNR of noisy speeches compared with spectral subtraction (1.3 dB on average) and Wiener filtering (1.9 dB on average).

4:20

2pSP10. Dereverberation of a closed test section of a wind tunnel with a mult imicrophones cesptral method. Daniel Blacodon and Jean Bulté (DSNA/ACOU, ONERA, 29, avenue de la Division Leclerc, BP N° 72, Châtillon 92322, France, daniel.blacodon@onera.fr)

Today, the manufacturers of aircraft want to perform at the same time aeroacoustic and classical aerodynamic tests in the wind tunnels with closed test sections in order to reduce test duration and costs involved in new aircraft developments. However, this kind of wind tunnels is not optimal for

acoustics testing because the measurements are often masked by background noise (i.e., noise generated by the wind tunnel flow mechanism) and reverberations on the walls without acoustic liner within the test section. This is a serious problem because these unwanted phenomena limit the accuracy of identification, and quantification of acoustic sources. The reduction of background noise can be obtained with subspace techniques or noise subtraction based on a noise reference measurement. Concerning the reduction of the contamination of the echos, it can be achieved with an inverse filtering technique or with a single microphone cepstral method. A new approach is proposed in this study to remove the spurious effects of the echos. It is based on multi microphones cesptral technique. This new technique has been successfully applied on numerical simulations, and experimental data. The theoretical developments of this method and the obtained results will be presented in the full paper.

TUESDAY AFTERNOON, 4 JUNE 2013 — 511AD, 1:00 P.M. TO 5:40 P.M.

Session 2pUWa

Underwater Acoustics and Acoustical Oceanography: Ocean Ambient Noise

Rex K. Andrew, Chair
Appl. Phys. Lab., 1013 NE40th St., Seattle, WA 98105

Contributed Papers

1:00

2pUWa1. Seabed characterization using ambient noise and compact arrays on an autonomous underwater vehicle. Peter L. Nielsen (Res. Dept., STO-CMRE, V.S. Bartolomeo 400, La Spezia 19126, Italy, nielsen@cmre.nato.int), Martin Siderius (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR), Jim Miller (Research Dept., STO-CMRE, La Spezia, Italy), Steven Crocker (Sensors & SONAR Systems Dept., NUWC, Newport, RI), and Jennifer Giard (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Estimating the seabed geoacoustic properties at various fidelity levels has been a research topic for several decades. The majority of the applied seabed characterization techniques often require significant involvement of surface vessels, complex experimental setup, and human interaction. Technical advances in underwater autonomy and the development of energy efficient electronics provide new opportunities to optimize underwater environmental surveys in particular of the seabed. In 2012, the CMRE conducted the GLASS'12 experiment in the Mediterranean Sea with the objective to investigate the feasibility of utilizing a hybrid autonomous underwater vehicle equipped with a compact nose array for long-duration seabed characterization over large areas. The vehicle has the capability of operating in traditional propulsion and glider mode, and the nose-mounted array consists of a 5-element vertical and 4-element tetrahedral array. The sound sources used as information carrier were ambient noise, e.g., sea surface generated noise and loud distant sources of opportunity. The experimental setup together with the newly developed autonomous equipment will be presented and examples of inferred reflection loss and sub-bottom profiling from the ambient noise are compared to ground truth measurements. [Work supported by the STO-CMRE, ONR-G Grant No. N62909-12-1-7040, the ONR N-STAR/ILIR program.]

1:20

2pUWa2. Monterey Bay ambient noise profiles using underwater gliders. Tarun K. Chandrayadula, Chris W. Miller, and John E. Joseph (Oceanography, Naval Postgrad. School, 650 Sloat Ave., # 3, Monterey, CA 93940, tkchandr@nps.edu)

In 2012, during two separate week-long deployments, underwater gliders outfitted with external hydrophones profiled the upper 100 m of Monterey Bay. The environment contains various noises made by marine mammals, ships, winds, and earthquakes. Unlike hydrophone receivers moored to a fixed location, moving gliders measure noise variability across a wide terrain. However, underwater mobile systems have limitations such as instrument and flow noise, that are undesired. In order to estimate the system noise level, the hydrophones on the gliders had different gain settings on each deployment. The first deployment used a 0 dB gain during which the ambient noise recordings were dominated by the glider. The second used two hydrophones, one with a 0 dB gain and the other with 20 dB. Apart from system sounds, the higher-gain hydrophone also recorded far-away sources such as whales and ships. The noise recordings are used to estimate the spectrograms across depth and record time. The spectrograms are integrated with the glider engineering data to estimate histograms of noise power as a function of depth and glider velocity. The statistics from the two different deployments are compared to discuss the value of gliders with external hydrophones in ambient noise studies.

1:40

2pUWa3. Measuring the spatial characteristics of the ambient noise field from an autonomous underwater vehicle. Stephanie Fried and Henrik Schmidt (Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 5-204, Cambridge, MA 02139, eowyn@mit.edu)

For autonomous underwater vehicles (AUVs), the primary method of sensing the local environment is through acoustics. The local noise field contains a wealth of information the AUV uses—from target tracking to communication to general understanding of the environment. A measure of the spatial composition of the ambient noise field can provide details about the physical environment as well as information for the AUV to incorporate into its control decisions. The challenge is in accurately measuring the directionality of the noise field from a single line array and continuously updating this measure to reflect changes in the environment and additional information as the AUV moves. Here we present a method for continuously assessing the spatial characteristics of an ocean ambient noise field measured by an AUV with a towed hydrophone array.

2:00

2pUWa4. Synthetic-array beamforming for bottom-loss estimation using marine ambient noise. Lanfranco Muzi and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu)

Previous studies have shown that, using a vertical line array, the bottom loss can be estimated in an ambient-noise field from the output power of beams steered toward the sea surface and beams reflected off the seabed. With short arrays, the low angular resolution of the bottom-loss estimate is one of the main limitations of the approach. *Synthetic-array processing* is proposed as a technique that can improve the angular resolution of the bottom-loss estimate to a level comparable to that of an array with twice as many physical sensors (at equal inter-sensor spacing). The proposed technique follows naturally from a new derivation in frequency-wavenumber

domain of the bottom-loss estimation procedure. The conditions under which the approach can be successfully applied are analyzed, with particular regard to the need for the array cross spectral density matrix to be "close" to Toeplitz. A metric is proposed for assessing this particular circumstance in practice. The technique is illustrated using synthetic data from simulations and then applied to data from several experimental campaigns. Results show that a 16-element synthetic array can achieve an angular resolution comparable to that of a 32-element array.

2:20

2pUWa5. Flow noise measurements at strong tidal current area in Uldolmok Waterway. Myungkwon Ko and Jee Woong Choi (Dept. of Marine Sci. and Convergence Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 426-791, South Korea, buymk@hanyang.ac.kr)

Flow noise is a kind of hydrodynamic noise, which is created by turbulent flow in the boundary layer around the hydrophone. Although the turbulent pressure is not a true acoustic noise in that its influence decreases rapidly with distance, it acts as a self-noise source. In general, since the spectrum level of flow noise has been reported to increase rapidly with the increment of flow speed, it is possible to monitor the current velocity from the flow noise measurements. Uldolmok waterway in Korea is one of the locations where currents are very strong, with maximum speed of about 5 m/s. The measurements of flow noise were conducted in Uldolmok waterway using two different shapes of hydrophone. In this talk, the flow noise spectra for various flow speeds will be presented for the frequency range of 20–100 Hz, and the comparison of spectra between two different-shape hydrophones will be given.

2:40

2pUWa6. On the origins of ambient biological sounds in shallow water tropical ecosystems. Simon E. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 3725 Miramar St. Apt. C, La Jolla, CA 92037, sfreeman@ucsd.edu), Forest L. Rohwer, Allison Gregg, Laura Coleman (Rohwer Lab., San Diego State Univ., San Diego, CA), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Although discovered more than 60 years ago, the origins of much ambient underwater biological noise remain unclear. Snapping shrimp sounds dominate some environments but elsewhere the shallow-water biological sound field is often heterogeneous. Here we show that dominant components of underwater ambient noise recorded on coral reefs around five islands in the central Pacific may be caused by the interaction of hard-shelled benthic macro-organisms with the substrate. Recordings show a consistent, nightly 4.7 to 6.9 dB increase in estimated pressure spectral density level in the 11 to 17 kHz band with a spectral peak centered between 14 to 15 kHz. Underwater time-lapse photography reveals a marked night-time increase in benthic invertebrate activity at most locations, temporally consistent with the increase in pressure spectral density level. Intensity-filtered recordings of an example species, the hermit crab *Clibanarius diugeti*, in quiet aquarium conditions reveal that transient sounds produced by interaction between the crustacean's carapace, shell, and coral substrate are spectrally consistent with the central Pacific recordings. Passive acoustic monitoring of such ambient noise may be useful as a complementary ecological survey technique to SCUBA-based visual observations, which are typically poor in estimating the abundance and diversity of cryptobenthic organisms.

3:00

2pUWa7. Prediction of underwater noise and far field propagation due to pile driving for offshore wind farms. Stephan Lippert, Tristan Lippert, Kristof Heitmann, and Otto von Estorff (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Denickestr. 17, Hamburg 21073, Germany, s.lippert@tu-harburg.de)

Wind energy plays a key role toward a greener and more sustainable energy generation. Due to limited onshore areas and possible negative effects on human living space, offshore wind parks become increasingly popular. During construction by pile driving, however, high levels of underwater sound emission are observed. To avoid negative effects on marine mammals and other sea life, different approaches, like, e.g., bubble curtains or cofferdams, are currently investigated to cut down the sound pressure levels. In order to predict the expected underwater noise, both with and without sound damping measures, numerical simulation models are needed to avoid complex and costly offshore tests. Within this contribution, possible modeling strategies for the prediction of underwater noise due to pile driving are discussed. Different approaches are shown for the direct adjacencies of the pile and for the far field sound propagation. The effectivity of potential noise mitigation measures is investigated using a detailed finite element model of the surroundings of the pile. The far field propagation in the kilohertz range at distances of several kilometer from the pile, on the other hand, is computed by a model based on wavenumber integration. Finally, the model validation with corresponding offshore tests is addressed.

3:20

2pUWa8. Wind-dependence of low-frequency ambient noise in the deep-ocean sound channel. Stephen Nichols (Grad. Prog. in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu) and David L. Bradley (Appl. Res. Lab., The Penn State Univ., State College, PA)

In the low-frequency range (1–125 Hz), the deep-ocean ambient noise field is produced by seismic, marine life, ship traffic, and wind-dependent hydrodynamic noise mechanisms. This study focuses on the contribution of wind-related source mechanisms to the overall ambient noise field, as well as previous attempts to understand the physics of these mechanisms. The Comprehensive Nuclear-Test Ban Treaty Organization (CTBTO) hydroacoustic monitoring system has produced nearly continuous recordings of the low-frequency deep-ocean ambient noise field at sites in the Pacific, Atlantic, and Indian Oceans, each spanning several years in length. Additionally, wind speed data have been recorded at the host island of each station by the National Oceanic and Atmospheric Administration (NOAA). Correlation techniques are used with these two datasets to determine the relationship between wind speed and the sound level in different frequency bands, and to determine the prominence of wind-related noise in the combined ambient noise spectrum. Results from the three sites are compared to each other to assess the uniformity of wind-generated noise over the world's ocean basins.

3:40

2pUWa9. Model for underwater noise radiated by submerged wind turbine towers. Todd Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (ARL:UT, P.O. Box 8029, Austin, TX 78713, hamilton@mail.utexas.edu)

Sustained tonal noise radiated by towers supporting offshore wind turbines contains energy in frequency bands that may disturb marine mammals, or interfere with passive sonar and seismic sensors and underwater communication equipment. Understanding the generation and propagation of underwater noise due to the operation of wind farms is important for determining strategies for mitigating the environmental impact of these noise sources. An analytic model based on a Green's function approach was previously developed for the sound radiated in the water column by a pulsating cylindrical structure embedded in horizontally stratified layers of viscoelastic sediment [Hay *et al*., J. Acoust. Soc. Am. **130**, 2558 (2011)]. This model has since been adapted to include relaxation and viscous losses in seawater and empirical loss factors for the sedimentary layers. In order to validate the model simulations were compared with reported measurements collected near an operating wind turbine that include radial acceleration of the tower, taken to be the source condition, and sound pressure levels in the water column. For long-range propagation over range-dependent environments, the analytic model has been coupled to a parabolic equation code. Simulations are presented for several bathymetries, sediment types, and tower array configurations. [Work supported by Department of Energy DE-EE0005380.]

4:00

2pUWa10. A quasi-analytic model of the underwater sound signal from impact driving of an offshore semi-infinite pile. Marshall V. Hall (Marshall Hall, 9 Moya Crescent, Kingsgrove, NSW 2208, Australia, marshallhall@optushome.com.au)

A quasi-analytic model is derived for the underwater sound signal radiated when an offshore pile is struck on its face by a hammer. The pile is modeled as a semi-infinite cylindrical shell of an elastic solid. The impact generates a pulse of vibration that travels down the pile at the longitudinal

sound-speed. At a given distance below the pile face, the axial displacement after the peak has arrived decreases exponentially with time. There are two coupled equations of motion for the axial and radial displacements. A closed form expression is derived for the radiated sound pressure (which is proportional to the radial acceleration) in terms of the Poisson ratio and Young's Modulus of the pile material, the hammer velocity, contact area between hammer and pile, pile radius, hammer mass, the pile's longitudinal sound-speed, and the sound-speed and density of the external medium. This model is applied to a published scenario for which the radiated sound pressure had been computed using a Finite Element Model, but is found to produce a different result. Some assumptions used in the model are identified that may explain the difference.

4:20

2pUWa11. Implementing physical constraints for noise only normal mode shape estimation. Ian M. Rooney, John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts at Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, irooney@umassd.edu), and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Many underwater acoustic tasks employ a vertical line array to sample a continuous wave pressure field. In the absence of strong local sources, the noise sampled by an array consists of both a spatially correlated and a spatially uncorrelated component. The correlated component is generally due to distant sources and can be described in terms of the modes of the waveguide. If the number of propagating modes is less than the array size, the diffuse noise lies within a lower dimensional subspace than the array data. The mode shapes defining this subspace can be estimated from noise measurements. Propagation physics constrain the mode shapes defining this subspace to be real, but the basis vectors obtained from a singular value decomposition of noise snapshots are generally complex. A phase rotation for each basis vector is required to rectify this. This work proposes a weighted average of the phase angles for each element that minimizes the variance of the rotation angle estimate. Simulations compare the proposed algorithm against prior approaches such as rotating each basis vector by the phase of the largest magnitude sample, rotating by the average of the phase, or taking the magnitude and ignoring the phase. [Work supported by ONR.]

4:40

2pUWa12. Analysis of the vertical structure of deep ocean noise using measurements from the SPICEX and PhilSea experiments. Kathleen E. Wage, Mehdi Farrokhrooz (Elec. and Comput. Eng. Dept., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA)

There are open research questions about the vertical structure of low-frequency ambient noise in deep water. For example Gaul *et al.*'s [*IEEE JOE* (2007)] analysis of the Church Opal data set showed that noise decreases substantially (on the order of 20 dB) below the critical depth, whereas other researchers have reported more modest reductions [Morris, *J. Acoust. Soc. Am.* (1978)]. Two deep water experiments provided a unique opportunity to measure ambient noise using large vertical arrays. In 2004–2005, SPICEX used two arrays to sample a North Pacific environment. One array was centered on the sound channel axis, and the other array had hydrophones above and below the critical depth. In 2010–2011, the PhilSea experiment deployed a single array with 150 hydrophones spanning the full water column. Both experiments made repeated short measurements (each 2–3 min long) of the field at the arrays. This talk compares the ambient noise observed during SPICEX and PhilSea with results reported in the literature. Since these data sets contain receptions over the period of a year, we focus on the seasonal dependence of the noise field. In addition to investigating noise level as a function of depth, we consider wind dependence and vertical directionality.

5:00

2pUWa13. Modeling of underwater piling noise mitigation using an array of soft spheres in the ocean. Keunhwa Lee (Ocean Eng., Seoul National Univ., Kwanak-ro 1, Seoul 151742, South Korea, nasalkh2@snu.ac.kr), Kyungmin Baik (CFFA Div. of Physical Metrology, Korea Res. Inst. of Standards and Sci., Daejeon, South Korea), and Woojae Seong (Ocean Eng., Seoul National Univ., Seoul, South Korea)

The ocean noise generated by marine piling affects severely fish, other marine life, and fishery activities. Accordingly, a few kind of noise mitigation system are presented. Among them, the noise mitigation system using soft scatterers such as air bubbles or rubber spheres is reported to show higher noise reduction than the classical cofferdam system composed of mass-absorbing materials. In this proceeding, a numerical scheme is developed to model and design the noise mitigation system using an array of soft spheres. This scheme is originally based on self-consistent equation of Zhen Ye for multiple scattering [Z. Ye and A. Alvarez, Phys. Rev. Lett. **80**, 3503 (1998)]. We generalize the original self-consistent equation for the oceanic waveguide using the waveguide green function. This generalized self-consistent equation is useful to model the noise propagation through an array of soft spheres in the ocean and assess the ability of the noise mitigation system. The effect of the oceanic waveguide on the noise reduction is studied and the validity of effective medium approach for a bubbly layer is also analyzed numerically.

5:20

2pUWa14. Spatial filtering in ambient noise crosscorrelation. Olivier Carriere, Peter Gerstoft, and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr, La Jolla, CA 92093-0238, ocarriere@ucsd.edu)

Ambient noise crosscorrelation is an attractive approach for estimating the properties of a propagation medium (presence of scatterers, wave propagation speed, etc.) without having to deploy active acoustic source. According to the theory, Green's function between pair of receivers can be extracted from the crosscorrelation of the received signals, given that the noise source is diffuse. In practice, this ideal condition is rarely met in the ocean and directional sources, often related to human activity in the vicinity of the receivers, may bias the travel time estimates. Using array processing, matrix spatial filters can be built independently from the environment properties (as opposed to model-based techniques) to filter specific directions of arrival. Such filter might be useful to remove an interferer or to restrict the effective source distribution to a reduced cone area. Here we study the use of matrix spatial filters for removing unwanted contributions in the crosscorrelations. Based on theory and simulations, we discuss the effect of array size, dimension and geometry, processed frequency band, and environmental conditions in the filtered crosscorrelations. An example of application to real data from the Shallow Water '06 experiment further illustrates the potential of the method.

Session 2pUWb

Underwater Acoustics and Acoustical Oceanography: Arctic Acoustics and Applications

Stan E. Dosso, Cochair
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Juan I. Arvelo, Cochair
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Chair's Introduction—12:55

Invited Papers

1:00

2pUWb1. Under-ice shallow-water sound propagation and communication in the Baltic Sea. Erland Sangfelt, Sven M. Ivansson, and Ilkka Karasalo (Underwater Res., Swedish Defence Res. Agency, FOI kista, Stockholm SE-16490, Sweden, erland.sangfelt@foi.se)

Sound propagation under ice in the Baltic is of interest for military as well as civilian purposes. Important questions are, for example, how sonar systems are affected by ice in shallow waters and how marine mammal life under ice is affected by ship traffic. Changing climate and increasing ship traffic are factors of great concern also in the Arctic region. Modeling results for sound propagation under ice in the Baltic Sea are presented for low as well as high frequencies. The sound propagation is influenced by bottom as well as ice-cap interaction in these shallow waters. The low- and high-frequency modeling is performed with wavenumber and ray models, respectively. Both types of models are amended to include a solid ice layer on top of the water column. In addition, the modeling efforts are extended to study the performance of an underwater communication system developed at FOI. The modeling results are evaluated using data from sea trials in the Gulf of Bothnia under ice-covered and ice-free conditions.

1:20

2pUWb2. Propagation of seismic exploration noise in the Marginal Ice Zone. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no) and Hanne Sagen (Nansen Environ. and Remote Sensing Ctr. (NERSC), Bergen, Norway)

This paper presents data from ambient noise recordings in the Marginal Ice Zone (MIZ) in the Fram Strait (deep water) and Barents Sea (shallow water) with focus on noise due to seismic exploration activity. Data were recorded on fields of sonobuoys deployed from P-3C aircraft in June 2011 and May 2012, each covering a 100 km x 100 km area. Recordings from three bouys in each area, from open water to within the solid ice cover, are analyzed for low-frequency (10 Hz–1 kHz) noise levels, with broadband noise due to distant seismic exploration identified. Strong attenuation of this noise component with distance into the MIZ is observed in the shallow-water data, whereas noise levels decrease less with distance in the deep-water data despite under-ice conditions. Propagation loss is modeled using a raytrace model including under-ice reflection loss due to an elastic ice cover, with environment input from the TOPAZ coupled ocean-sea ice model and from satellite images. Properties of the noise field, modeling results, and model-data discrepancies will be discussed. [Data collected under the ACOBAR and WIFAR projects at NERSC.]

1:40

2pUWb3. Modelled and measured sound levels from a seismic survey in the Canadian Beaufort Sea. Marie-Noël R. Matthews and Alexander O. MacGillivray (JASCO Appl. Sci., 202 - 32 Troop Ave., Dartmouth, NS B3B 1Z1, Canada, Marie-Noel.Matthews@jasco.com)

JASCO Applied Sciences performed acoustic modeling and measurements to calculate marine mammal exclusion zones for Chevron Canada Limited's 2012 Sirluaq 3-D seismic program in the Canadian Beaufort Sea. The Sirluaq survey was located in deep water (>650 m), on and beyond the continental slope, and presented unique challenges from both modeling and measurement standpoints. The modeling was performed with JASCO's Marine Operations Noise Model (MONM), which uses a parabolic-equation-based algorithm to accurately predict $N \times 2$-D sound propagation in ocean environments. Sound levels were measured with five calibrated Autonomous Multichannel Acoustic Recorder (AMAR) systems at distances of 50–50 000 m from the airgun array, in water depths ranging from 50 to 1500 m. The sensors were laid out to capture sound levels in both the broadside (perpendicular to survey line) and endfire (along the survey line) directions. The high-resolution digital recordings of seismic sounds were analyzed to determine peak and rms sound pressure levels (SPL), and sound exposure levels (SEL) as functions of range from the airgun array. The model estimates were generally conservative; however, the model predictions at the specific depth of the receivers accurately predicted the existence of a shadow zone and the overall transmission loss trend.

2:00

2pUWb4. Arctic ambient noise measurements in support of the Northern Watch Project. Garry J. Heard, Nicos Pelavas, Sean Pecknold (Atlantic, Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, garry.heard@gmail.com), Carmen E. Lucas (Dartmouth, Nova Scotia, Canada), and Bruce Martin (Jasco Res., Dartmouth, NS, Canada)

During August 2012, acoustic recording systems were deployed in Barrow Strait as part of the Defence Research and Development Canada (DRDC) Northern Watch Technology Demonstration Project. Two Starfish Sensor Cubes each with a 1-m cube of seven hydrophones operating in the frequency range of 5–750 Hz, and two single-hydrophone, Autonomous Multichannel Acoustic Recorders (AMAR) providing a 30-kHz signal bandwidth were deployed. The Starfish were deployed for two one-week intervals. One AMAR was deployed for two weeks partially overlapping the Starfish deployment. The second AMAR was deployed for a period of one year with recovery planned for August 2013. The observed underwater noise picture is one of high variability ranging from an extremely quiet to a noisy environment. Noise sources included: A 500-m long iceberg grounded within 500 m of one of the Starfish; a large ice island (4–5 sq-km) that passed within 4 km of the sensors; a small number of motoring vessels; significant wind events that caused rapid and strong variations in the noise field; and a small number of marine mammal detections. After our departure, a large number of Beluga whales were observed visually. The remaining AMAR may detect these late summer visitors.

Contributed Papers

2:20

2pUWb5. Acoustic manifestations of frozen bubbles. Alexey Maksimov (Phys. of the Ocean, Pacific Oceanological Inst. Far Eastern Branch of the Russian Acad. of Sci., 43, Baltic St., Vladivostok 690041, Russian Federation, maksimov@poi.dvo.ru)

The hydrocarbon seeps emitting buoyant bubble plumes from seafloor vents have been actively investigated in different regions of the World Ocean. In winter, rising bubbles, which have reached the sea surface, freeze in ice. These clouds of the frozen bubbles are observed in Arctic seas and represent a common element of an ice cover of lakes. Deposits of gas hydrates were discovered in many marginal seas of the World Ocean. The nature of the relationship between the acoustic and physical properties of gas hydrates is still under debate for both saturated sediments and sediment containing free-gas bubbles. Acoustic manifestations of bubbles frozen in ice and gas hydrate are the subjects of the current study. On the basis of the general solution of the elastic wave scattering problem for the sphere, the scattering cross section for a bubble frozen in an ice has been found. The derived expressions coincide in the limiting cases with known results: a bubble in a rubber-like media, where the longitudinal wave speed is significantly faster than the transverse wave speed and an empty cavity in the elastic media. The structure of low-frequency resonances of bubbles clouds of the simplest geometry has been investigated.

2:40

2pUWb6. Source bearing and range estimation using an ice-mounted tri-axial geophone. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper presents results from Arctic field trials to estimate the bearing and range of an acoustic source in the water column using seismic particle motion measured at a tri-axial geophone on the sea ice surface. Measurements were carried out on smooth, rough, and ridged annual ice, and on a multi-year ice floe. At each site, a hammer seismic study was carried out to selectively excite various ice seismic waves and investigate their propagation properties. Impulsive acoustic sources were deployed in the water at a variety of bearings and ranges from 0.2 to 50 km. Source bearings are estimated by applying polarization filters to suppress shear waves with transverse particle motion and computing the incident power as a function of radial look angle; the inherent 180-degree ambiguity is resolved by requiring prograde particle motion in the vertical-radial plane. Results indicate good bearing estimation (<10-degree average errors) at all ranges with little dependence on ice type. Source range is estimated from the time difference between the water-borne arrival and the critically-refracted longitudinal plate (Lp) wave. Results are limited due to strong attenuation of the Lp wave, with good range estimation to <1 km for smooth annual ice and <0.5 km for other sites.

3:00

2pUWb7. Measurements and modeling of transmission loss variability in Barrow Strait. Sean Pecknold, Nicos Pelavas, and Garry Heard (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca)

During the summer of 2012, a field trial was held in Barrow Strait, south of Devon Island in the Canadian Arctic. The trial included a set of acoustic transmission loss experiments recorded on Starfish Sensor Cubes, which include a 1-m cube of seven hydrophones operating in the frequency range of 5–750 Hz. The transmission loss runs consisted of 10-min and 20-min duration transmissions of 400 and 500 Hz tones made at a discrete set of distances up to 60 km from the recorders. Supporting environmental measurements included sets of CTD (conductivity-temperature-depth) profiles and bathymetric measurements. The effects of the measured environmental properties and variability are investigated via propagation modeling, and compared to the experimental data acquired during these experiments.

3:20

2pUWb8. A parabolic equation for under ice propagation. Adam M. Metzler (Environ. Sci. Lab., Appl. Res. Lab.: The Univ. of Texas at Austin, ARL-Environ. Sci. Group, P.O. Box 8029, Austin, TX 78713, ametzler@arlut.utexas.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, Texas)

Parabolic equation methods are useful to accurately and efficiently model propagation in the ocean for range-dependent environments. Most methods treat environments where the ocean surface is a flat perfect reflector. For some problems, the ocean surface characterized as a random rough scattering water-air boundary produces significant effects on the overall propagation. A rough surface parabolic equation has been developed [A. P. Rosenberg, J. Acoust. Soc. Am. **105**, 144 (1999)] that extends the split-step Padé approach in the RAM implementation, and others have extended split-step Fourier methods. For an upper ice surface, elasticity should be incorporated into the rough scattering boundary. In this paper, a parabolic equation is presented that captures effects of seismo-acoustic interactions and scattering from a rough surface modeled as rigid as opposed to pressure release. Particular attention is given to upward-refracting sound speed profiles, typically found in Arctic environments, which encourage interactions with an ice surface. The scattering technique can be extended to environments with fluid/ice interfaces. [Work supported by ARL:IR&D.]

3:40

2pUWb9. Effective ice model for under-ice propagation using the fluid-fluid parabolic equation. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Richard L. Campbell (OASIS Inc., Seattle, Washington), and Lee Freitag (Woods Hole Oceanogr. Inst., Woods Hole, Washington)

An approach is presented to permit efficient computation of the waterborne acoustic field in the presence of sea ice. The range-dependent wide-angle parabolic equation (PE) is used to model the acoustic field with a hard, lower-density layer (ice) placed above the ocean and seafloor. The ice layer is characterized by its thickness, compressional speed, density, and attenuation. Acoustic loss due to sea ice is primarily driven by conversion to shear waves, and in this model the effect will be approximated by attenuation within the ice layer. Rough interface scattering at the air-ice and ice-water interfaces will be handled by generating range-dependent realizations from a data-derived ice thickness model. An inversion for ice parameters is conducted using the work of Jin *et al*. [J. Acoust. Soc. Am. **96**(5)] and by matching the transmission loss of Diachok [J. Acoust. Soc. Am. **59**(5)]. Model-data comparisons between this three-layer PE and measurements the taken in the Fram Strait in 2010 and in the Canada Basin in 2011 will be presented.

TUESDAY EVENING, 4 JUNE 2013 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday are as follows:

Architectural Acoustics	513abc
Engineering Acoustics	512ae
Musical Acoustics	512dh
Physical Acoustics	519b
Psychological and Physiological Acoustics	514abc
Structural Acoustics and Vibration	512cg