Session 3aAA

Architectural Acoustics and Noise: Primary and Secondary School Special Function Spaces I

Robert C. Coffeen, Chair
Univ. of Kansas, School of Architecture and Urban Design, Marvin Hall, Lawrence, KS 66045

Chair’s Introduction—8:00

Invited Papers

8:05

3aAA1. The rest of the school: An investigation into good acoustical design of non-classroom spaces in primary and secondary schools. Benjamin Markham and Jennifer Hinckley (Acentech Inc., 33 Moulton St., Cambridge, MA 02138)

The focus on acoustics in recent standards and guidelines such as the ANSI Standard S12.60, the Leadership in Energy and Environmental Design (LEED) for Schools system, and the Collaborative for High Performance Schools (CHPS) has rightly been on classrooms. Each of them does address other school environments also, although sometimes without particular specificity. After reviewing relevant American standards and guidelines for gymnasiums, libraries, lobbies, corridors, auditoriums, and other spaces that do not fit neatly into the classroom designation, this paper addresses good practices in the acoustical design of such critical environments through the use of a series of case studies. Specifically, the authors suggest room acoustics design parameters and appropriate background noise and sound isolation criteria for non-classroom spaces in primary and secondary schools.

8:25

3aAA2. Case study in cafeteria noise solutions. Kenneth Good and Sean Browne (Armstrong World Industries, 2500 Columbiana Ave., Lancaster, PA 17601)

A recent case study was conducted addressing excessive occupant noise within a cafeteria in a Rhode Island school. A design requirement was to retain the open structure ceiling. This paper will discuss the before and after acoustic performance and the effect it had on the occupants.

8:45

3aAA3. The acoustical balancing act for music rehearsal spaces in schools. David L. Adams (D. L. Adams Assoc., Inc., 1701 Boulder St., Denver, CO 80211, dadams@dlaa.com)

Conflicting acoustical criteria are encountered in the design of music rehearsal spaces, particularly when a single space must serve both orchestra and band rehearsals, not to mention the situation when the space must also serve for chorus rehearsals. For example, the criterion commonly found in the literature for room volume or more explicitly ceiling height conflicts with the requirement for an appropriate reverberation time or design goal loudness level, which are also dependent on the group (band, orchestra, and chorus) that will use the space and for that matter on the individual music teacher. The degree of involvement of music teachers in the design of these spaces and their potential impact on the end result is discussed. Case histories are presented to illustrate the good, the bad, and the ugly with respect to music rehearsal spaces in schools.

9:05

3aAA4. Acoustical considerations for vocal music rehearsal rooms: A choir director’s perspective. Jeremy Manternach (Div. of Music Education and Music Therapy, School of Music, Univ. of Kansas, 1530 Naismith Dr., Lawrence, KS 66045, jmanter@ku.edu)

What do vocal/choral instructors value in a rehearsal space? Along with logistical considerations, these instructors also desire particular room acoustic conditions. How might their input affect the decisions of architects and acousticians? This question is investigated through examination of several school rehearsal spaces. The study will include acoustical measurements from rehearsal spaces, opinions of vocal music instructors, and feedback from vocalists who rehearse in each space.

9:25

3aAA5. Acoustical compromises: Working with a design team for maximum isolation at minimal costs in high school music facilities. Pamela J. Harght (4006 Speedway, Austin, TX 78751, pam@baiaustin.com)

Texas Public High Schools place a large emphasis on their music and drama programs, often providing facilities for their students that are far superior than many 4-year college and universities. These high school facilities include several rehearsal rooms for band, choir, orchestra, percussion, ensembles, mariachi and individual practice. In addition, they also often have a drama classroom, a black box theater, and a Performing Arts Center. When working with the design team on these projects, there are often many obstacles to work
around in order to provide the best scenario possible for the end user. These “obstacles” often include (but are not limited to) school district codes, budgets, the “convenience” factor, city codes, and space planning. This paper will showcase many high school music facilities through the south and southeast and the compromises and obstacles that were overcome with the end result goal as satisfying the music department.

9:45

3aAA6. Helping clients realize superior room acoustics while achieving other facility goals. Rob Farion (Farion Architectural Acoust., 3749 43A Ave., Red Deer, AB T4N 3G2, Canada, rob@farion.ca)

Addressing the needs of a space often involves considering many competing objectives. When room acoustic performance is one of the priorities, there is potential to address other priorities in conjunction with acoustic considerations. Integrating interior design goals with acoustic requirements often allows more resources to ultimately be available for acoustic considerations. This presentation outlines two educational facility projects where clients were assisted in meeting their goals by incorporating interior design objectives and room furnishings with acoustic performance outcomes.

10:05—10:20 Break

10:20

3aAA7. The evolution of the gymatorium and cafetorium in primary schools. Felicia Doggett (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, felicia@metropolitanacoustics.com)

Many primary schools do not have formal auditoriums, but the school still needs a space for performances and assemblies. The existence of “gymatoriums” and “cafetoriums” in these schools is essential to the student body and staff. Far from being just a gymnasium or cafeteria with a stage at one end, these rooms are fully functioning performance spaces with appropriate acoustics and audio systems. What many in the school may deem most important, however, is that they still function very well for their everyday uses of gymnasiums or cafeterias. This presentation explores several gymatoriums and cafetoriums and their acoustical and audio design elements.

10:40

3aAA8. Acoustical design of special purpose rooms in schools: Music rooms. Gary W. Siebein (Univ. of Florida School of Architecture, P.O. Box 115702, Gainesville, FL 32611), Hyun Paek, Chris, P. Jones, and Reece Skelton (Siebein Assoc., Inc., Gainesville, FL 32607)

Design strategies for music education rooms in elementary, middle, and high schools based on an impulse response-based theory are presented through a series of case studies of existing, renovated, and designed rooms. The strategies include providing sound reflections to allow the instructor to hear individual groups of students as well as to allow the students to hear each other, controlling room volume and absorption to allow early sound reflections for clear hearing, and simultaneously provide diffuse, low-level running reverberance to enhance musical qualities, providing for clear verbal communication between teacher and students and limiting background and intruding noise levels. The impulse response-based measures are reduced to a series of architectural systems that can be implemented using alternate construction systems to meet budget requirements. Standard design concept sketches communicate the intent of the acoustical design to Architects, School Board members, Design Builders and music faculty.
in beam patterns across species, indicating that directionality is indeed a frequency. Echolocation calls were recorded from five species of echolocating biosonar sound beams. Directionality increases with the size of the sound hypothesizes: Small bats emit high frequencies to obtain highly directional from their prey. Thus, we propose an alternative or supplementary small bats use higher frequencies than what is needed for effective reflection echoes efficiently at short wavelengths, i.e., high frequencies. However, very prey size since small bats feed on small prey and small objects only reflect larger bats. This correlation has been explained as an acoustic constraint by and prey. Small insectivorous bats typically emit higher frequencies than Southern Denmark, Campusvej 55, Odense C, Denmark

3aABa2. Similarities in the echolocation beam pattern of vespertilionid bats. Lasse Jakobsen and Annemarie Surlykke (Inst. of Biology, Univ. of Southern Denmark, Campusvej 55, Odense C, Denmark)

Echolocating bats can navigate and forage by sound, emitting short high-frequency sound pulses and listening to echoes reflected from obstacles and prey. Small insectivorous bats typically emit higher frequencies than larger bats. This correlation has been explained as an acoustic constraint by prey size since small bats feed on small prey and small objects only reflect echoes efficiently at short wavelengths, i.e., high frequencies. However, very small bats use higher frequencies than what is needed for effective reflection from their prey. Thus, we propose an alternative or supplementary hypothesis: Small bats emit high frequencies to obtain highly directional biosonar sound beams. Directionality increases with the size of the sound emitter relative to the wavelength. Hence small bats can counteract the decrease in directionality caused by their small size by increasing their emitted frequency. Echolocation calls were recorded from five species of echolocating bats of different sizes flying in the laboratory using a multimicrophone array and their beam pattern was calculated. The results show high similarity in beam patterns across species, indicating that directionality is indeed a constraint on echolocation frequency. [ Funded by the Oticon Foundation. ]
9:00 3aABa5. A digital model for the deformation of bat ears. Sreenath Balakrishnan (Dept. of Mech. Eng., Virginia Tech. & Inst. for Adv. Learning and Res., 150 Slayton Ave., Danville, VA 24540, sbreane@vt.edu), Li Gao, Weikai He (Shandong Univ., 250100 Jinan, China), and Rolf Müller (Virginia Tech & Inst. for Adv. Learning and Res., Danville, VA 24540)

In bats, the directivity patterns of the biosonar system are shaped by the surface geometry of the pinnae. Since many bat species are capable of large ear deformations, these beampatterns can be time-variant. To investigate this time-variance using numerical methods, a digital model that is capable of representing the pinna geometry during the entire deformation cycle has been developed. Due to large deformations and occlusions, some of the surfaces relevant to sound diffraction may not be visible and hence the geometry of the entire pinna has to be computed from limited data. This has been achieved by combining a complete digital model of the pinna in one position with time-variant sparse sets of three dimensional landmark data. The landmark positions were estimated using stereo vision methods. A finite element model based on elasticity was constructed from CT scans of the pinna post mortem. This elastic model was deformed to provide a good fit to the positions of the landmarks and retain values of smoothness and surface energy comparable to life. This model was able to handle ratios of data to degrees of freedom around 1:5000 and still effect life-like deformations with an acceptable goodness of fit.

9:15 3aABa6. Reanalyzing auditory sensitivity: The functional audiogram as modeled by the bat detecting moth ear. Matthew E. Jackson, Navdeep Asi, and James H. Fullard (Dep. of Biology, Univ. of Toronto at Mississauga, 3359 Mississauga Rd., N. Mississauga, ON L5L 1C6, Canada, mathereric.jackson@utoronto.ca)

Auditory sensitivity has often been measured by identifying neural threshold in real time (online) which can introduce bias in the audiograms that are produced. This was tested by recording auditory nerve activity of the Notodontid moth Nadaa gibbosa elicited by bat-like ultrasound and analyzing the response offline. The offline audiogram was compared to a published online audiogram showing that the bias introduced can result in a difference in both the best frequency of this moth ear and the audiogram shape. The offline neural threshold audiogram was then compared to audiograms produced using behavioral threshold definitions based on (1) spike period and (2) the latency to first spike, showing that audiograms produced using these theoretical behavioral definitions to have a similar shape. Finally, predictions on the distance at which Notodontid moths may evade bats using negative phototaxis or the acoustic startle response are made, using the number of spikes elicited as a proxy for distance.

9:30 3aABa7. Echolocation in a fresh water lake. Elizabeth vonMuggenthaler (Fauna Commun. Res. Inst., Hillsborough, NC, 27278 l@animalvoice.com), Joseph Gregory (NC), and Scott H. Mardis (Fauna Commun. Res. Inst., Vermont, Canada)

Lake Champlain is host to myriad interesting geological peculiarities, including the oldest Middle Ordovician reef bed in the world containing coral. Autogenic succession suggests animals adapt to modifications in climate. Eleven thousand year old Beluga whale skeletons have been found in Vermont and Beluga whales currently survive in the St. Lawrence Seaway. In 2003, 2005 and 2009 sites on Lake Champlain were explored using computers with National Instruments Polynesia real-time sound analysis, NI PCl-MCIA 6062-E cards, DAT recorders, GPS, amplifiers, three vector sensors, two hydrophones, and a Nagra IV-S5. ECHO Aquarium in Vermont facilitated recording of known lake inhabitants and non-biological signals studied formed the basis for the control. Neither recordings nor literature indicate that the known native creatures echolocate. Combining wavelet applications, aiding in reduction in ambient noise in this opposing environment along with conventional analysis, the experiments have been able to conduct far reaching, low-noise sound measurements and were capable of detecting signals the nature of which suggests the presence of some interesting and unexpected phenomena within the ranges and inherent structure of Beluga whale, killer whale, and dolphin echolocation. To protect Lake Champlain, further investigations into this acoustic anomaly is encouraged. [Work supported by the Radius Foundation.]

9:45 3aABa8. Identifying manatee location using dual-frequency sonar. DIDSON. Christopher Nizezcki (Dept. of Mech. Eng., UMass Lowell, 1 Univ. Ave., Lowell, MA 01854)

Over the last several years there has been much interest in detecting the West Indian manatee (Trichechus manatus latirostris) in turbid waters. Even in clear waters, manatees can be difficult to detect at appreciable distances by looking from above at the surface of the water. The standard DIDSON sonar is a multibeam sonar that provides dynamic images and operates at two frequencies (1.1 and 1.8 MHz). It is capable of providing high-resolution images of objects underwater that are typically scanned from the side or from above. Within this work, the DIDSON sonar is used to detect manatees at Homosassa Springs, FL. The manatees are ensonified from the side and appear to be viewed from the top. The manatees are readily apparent at a range up to 23 m for the measurements acquired in this preliminary sample. The presentation will present some of the images of manatees using a handheld diver DIDSON unit and describes some of the challenges and implications in using the device to detect manatees.

10:00 3aABa9. Using active acoustics to assess habitat restoration in a freshwater lake. Laura E. Madden (Dept. of Wildlife and Fisheries Sci. and Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, lem230@psu.edu) and Jennifer L. Miksis-Olds (The Penn State Univ., State College, PA 16804-0030)

Artificial structures increase bottom complexity in man-made reservoirs and aim to increase fishery production by providing improved refuge, forage, and reproduction habitat. Electrofishing is typically employed to assess the effectiveness of these structures but is limited to shallow water sites. Active acoustic technology is a non-invasive sampling method that is more flexible than electrofishing, as it is operational in both deep and shallow water sites. Active acoustics also provides a continuous data series at one location to obtain high-temporal resolution information. An acoustic water column profiler was deployed for 1 week at each of two sites: (1) a control site with no introduced structure and (2) a treatment site with artificial refuge habitat. Difference in fish abundance, vertical distribution, and diurnal behavior between the two sites was assessed from the volume backscatter time series. Variation in fish activity between the two sites was compared. This work demonstrates the utility of active acoustics in assessing the effectiveness of freshwater habitat alteration beyond the scope of conventional techniques. A more comprehensive evaluation of habitat restoration is crucial in guiding the development of future conservation efforts.
Animal Bioacoustics: Animal Hearing and Vocalization

Susan E. Parks, Cochair
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Peter Marvit, Cochair
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Contributed Papers

10:30
3aABb1. Aerial audiograms of Steller and California sea lions measured using auditory steady-state response methods. Jason Mulson (U.S. Navy Marine Mammal Program, SSC Pacific, Code 71510, 53560 Hull St., San Diego, CA 92152, jmulsow@hotmail.com), Colleen Reichmuth (Univ. of California, Santa Cruz, Santa Cruz, CA 95060), Frances Gulland (The Marine Mammal Ctr., Sausalito, CA 94965), David A. S. Rosen (Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada), and James J. Finneran (U.S. Navy Marine Mammal Program, SSC Pacific, San Diego, CA 92152)

Detection of aerial vocal signals by conspecifics is important in the reproductive behavior of the otariid pinnipeds. However, aerial hearing sensitivity measurements have only been obtained for a few otariid individuals that were trained to participate in behavioral experiments. In order to expand upon this small data set, auditory steady-state response (ASSR) methods were used to examine the aerial hearing sensitivity of Steller and California sea lions. Although ASSR thresholds were elevated relative to behavioral thresholds reported for otariids, the ASSR audiograms of the majority of individuals were similar to each other and to behavioral audiograms in terms of relative sensitivity. A marked reduction in sensitivity with increasing frequency regularly occurred between 16 and 32 kHz, indicating a consistent high-frequency cutoff. The reliability of the ASSR audiograms for both species suggests that behavioral aerial audiograms that exist for a few Steller and California sea lion individuals can be appropriately extrapolated to larger populations. The similarity of the ASSR audiograms among the Steller and California sea lions supports the notion that the otariid pinnipeds form a functional hearing group, with similar aerial hearing in terms of sensitivity and frequency range of hearing. [Work supported by ONR and NOAA Ocean Acoustics Program.]

10:45
3aABb2. Marine mammal auditory evoked potential measurements using swept amplitude stimuli. James J. Finneran (US Navy Marine Mammal Program, SSC Pacific, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

Auditory evoked potentials are routinely used to characterize hearing in marine mammals for whom behavioral testing is not practical. For frequency-specific audiometry, the most popular evoked potential method has been the measurement of the auditory steady state response (ASSR). These tests normally entail measuring the ASSR to a sequence of sinusoidally amplitude modulated tones so that the ASSR input-output function can be defined and the auditory threshold estimated. In this study, an alternative method was employed, where thresholds were estimated in response to a single amplitude modulated stimulus whose sound pressure level varied linearly with time. The tone sound pressure was therefore swept over a range of levels believed to bracket the threshold. The input-output function was obtained by analyzing the resulting grand average evoked potential using a short-time Fourier transform. The swept amplitude technique was performed with bottlenose dolphins and California sea lions, with the resulting thresholds similar to those obtained with constant amplitude tones. The tradeoffs between the swept amplitude technique and the traditional, constant amplitude approach will be discussed. [Work supported by ONR.]

11:00

Zebra finch song consists of complex acoustic elements repeated over several hundred milliseconds with extreme vocal-motor precision. Much less is known about how song is perceived though previous work has shown that zebra finches easily discriminate between normal and time-reversed versions of song syllables. Here, we used operant conditioning, psychophysical methods, and various synthetic song models to assess the bird’s sensitivity to local versus global temporal changes in song and to determine which cues are most salient in these complex syllables. Birds could discriminate syllable reversals in songs made up of the song envelope filled with noise, suggesting that syllable envelope cues can provide the basis for discrimination. However, birds could also discriminate syllable reversals in songs made up entirely of Schroeder harmonic complexes, stimuli that provide only phase (reversals of fine structure) cues without syllable envelope cues. Birds performed better on longer syllables, suggesting a window of temporal integration for fine structure discrimination. In contrast, humans could perceive few of the syllable reversals within these song modifications. This fine-grained perception in birds suggests that the machinery used for song perception is as precise as the machinery used for song production. [Work supported by NIH.]

11:15
3aABb4. Distance perception by non-territorial zebra finches (Taeniopygia guttata) and budgerigars (Melopsittacus undulatus). Kelly E. Radziwon and Micheal L. Dent (Dept. of Psych., Univ. of Buffalo, SUNY, 206 Park Hall, NY 14260, radziwon@buffalo.edu)

Birds use long-range acoustic signals to defend territories, attract mates, and locate conspecifics. However, long-range acoustic signals progressively degrade during their transmission from the signaler to the receiver. This degradation makes perceiving these signals more difficult, but receivers can use such information to estimate the signaler’s distance. The perception of auditory distance cues (overall amplitude, frequency-dependent attenuation, and reverberation) has typically been studied in the field with territorial birds. The present study examined auditory distance perception in a non-territorial songbird, the zebra finch, and in a non-songbird, the budgerigar in a controlled laboratory setting. Three zebra finches and three budgerigars were trained to identify 1-m (undegraded) and 75-m (degraded) recordings of three budgerigar contact calls, one male zebra finch song, and one female zebra finch call. Test stimuli were then introduced on 20% of the total number of trials. These stimuli were created manually by manipulating the natural recorded 1-m calls. By editing these calls, we could manipulate each distance cue separately to determine which cue was the most salient for the birds. Our results suggest that amplitude was the most important cue for these birds, similar to humans and other animals.
3aABb5. Direct measurements of subjective loudness in a bottlenose dolphin. Carolyn E. Schlundt (ITT Corp., 2376 Rosecrans St., San Diego, CA 92106, carolyn.melka@itt.com) and James J. Finneran (US Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA 92152)

Equal-loudness contours were measured in a bottlenose dolphin (Tursiops truncatus) trained to perform a loudness comparison task. The subject was presented two sequential tones and whistled if the first tone was louder, and produced a burst pulse or “buzzed” if the second tone was louder. Approximately 70% of trials were “known” comparisons [e.g., tones of same frequency but different sound pressure levels (SPLs)] and allowed performance to be tracked within sessions. The remaining comparisons were probe trials, consisting of a 10-kHz standard tone with a fixed SPL (either 90, 105, or 115 dB re 1 μPa) and a comparison tone, whose frequency was fixed but whose SPL varied. Presentation order of probe trials was balanced. Eleven comparison frequencies ranged from 3.5 to 113.1 kHz. Logistic regression was used to derive curves relating the probability of the comparison tone being perceived louder. The 50% point represented the SPL at which the comparison and standard tones were equally loud. The data represent the first direct measurement of equal-loudness curves in any animal and show the relationship between the frequency and subjective loudness. Loudness contours may be more appropriate for assessing behavioral effects of sound, assuming behavioral reactions are more strongly related to loudness than SPL.

3aABb6. Vocalization in a neonatal Amur tiger cub. Edward J. Walsh, Adam B. Smith (Developmental Auditory Physio. Lab., Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, smithabl@boystown.org), Douglas L. Armstrong (Omaha’s Henry Doorly Zoo, Omaha, NE 68107), and JoAnn McGee (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

Although the acoustical features of a small number of calls representing the vocal repertoire of adult tigers have been studied and reported, calls produced by developing tigers representing any subspecies have not been described. In that light, the acoustical features of calls produced by an Amur cub during the first two postnatal weeks of life will be considered. The vocal repertoire of the cub observed during recording sessions was limited, consisting primarily of neonatal cries composed of three distinct types: an intense, harmonically rich phonation, an aperiodic cry (dysphonation), and a compound cry exhibiting both dysphonation and harmonically rich phonation segments. Overall acoustic power of the harmonically rich phonation was uniformly distributed across a frequency band ranging from approximately 0.6–2.0 kHz and that was centered on roughly 1.2 kHz on the second postnatal day. While the same basic pattern was observed on postnatal day 13, inter-harmonic differences were smaller, and between days 2 and 13 the mean fundamental frequency dropped from ~260 to ~200 Hz; typically, f0 varied during each utterance at each age studied. These findings will be considered in relation to the spectrographic character of calls produced by adult Amur tigers. [Funding was provided by NSF Grant 0823417.]
This work describes a novel numerical model developed to optimize the design of a broadband, spatially averaging free fiber optic hydrophone probe (FOHP) for characterization of ultrasound fields in the frequency range 1–100 MHz. The design includes minimization of the active sensor dimensions, so its cross-section is comparable with the half acoustic wavelength at the highest frequency of interest and improvement of voltage-to-pressure sensitivity of the probe measured in V/ Pa. Initial simulations based on the assumption that bulk refractive index of sputtered material could be used indicated that a thin film gold coating (1–35 nm) could indeed enhance the voltage sensitivity of the FOHP by 16–30 dB; however, a follow up analysis revealed that determination of the coating thickness influence on the FOHP performance would require an introduction of the complex (as opposed to bulk) index of refraction of the sputtered film. The input parameters to the model, their selection criteria, and implementation of the coupled acousto-optic interaction of gold layer will be discussed. The measured prototypes produced unprecedented voltage sensitivity between 234 and 245 dB re 1 V per pPa or 2 and 560 mV/MPa, respectively. [The authors wish to acknowledge support of the NIH R01EB007117 grant.]

**Contributed Papers**

### 8:40

3aBB3. Acoustic streaming for direct delivery of therapeutics into tissue. Raghu Raghavan (1101 East 33rd St., Ste. B305, Baltimore, MD 21218)

Ultrasound has been used in medical drug delivery but largely for altering membrane permeability. We are investigating acoustic streaming for delivery of therapeutics, such as drug molecules or other particulates, into brain tissue, with other applications for the future. Acoustic streaming potentially can enhance fluid and particle flux in interstitial pathways in tissue. The conventional equations that describe streaming in fluids are not directly applicable to porous media such as the brain. In this study, a modified Biot equation is employed to develop a theoretical framework to account for the attenuation mechanisms dominant in acoustic streaming in porous media. The analytical solutions obtained for isotropic, homogeneous media are presented and compared with experiments conducted on gel phantoms, and with results reported for streaming in live monkey brains. The dependence of streaming-induced therapeutic transport on acoustic parameters and tissue properties are investigated. The theoretical framework was used to determine optimal operating parameters of ultrasound devices for streaming-based drug transport in porous tissue with particular emphasis on two objectives: (1) reduction in backflow from catheters inserted into brain tissue for direct delivery and (2) delivering drugs into the margins of a resection cavity following surgery for brain tumor.

### 8:55

3aBB4. Phase synchronization and collective instability in oscillating bubble clouds. D. Sinden, E. Stride, and N. Safiari (Dept. Mech. Eng., Univ. College London, Torrington Pl., London WC1E 7JE, United Kingdom, d.sinden@ucl.ac.uk)

The effect of bubble interactions within a bubble cloud under ultrasonic forcing is investigated. Under certain conditions, phase synchronization is found, revealing parameter regimes where the dynamics of the bubble system are primarily determined by the interactions of the bubbles, rather than the applied ultrasonic acoustic field. Due to the Bjerknes interaction, the stability thresholds for critical pressure and size leading to unpredictable oscillations for bubble clouds are lower than the stability thresholds for single bubbles. Interestingly, numerical simulations confirm that in the regimes where bubbles exhibit chaotic oscillations, the dynamics are determined by a complicated geometric object called a low-dimensional strange attractor which is qualitatively the same as that for the chaotic oscillations of a single bubble. Thus an averaged equation may also be low-dimensional. The effects of synchronization on Landau damping are discussed. The effect of delays between the interactions is also investigated. The implications for applications in high-intensity focused ultrasound, adaptive imaging, and targeted drug delivery will be discussed.

### 9:10

3aBB5. Design and implementation of a cylindrical array for passive mapping of cavitation fields produced by clinical high intensity focused ultrasound transducers. Stuart Faragher, Miklós Gyöngy, Jamie Collin (IBME, Univ. of Oxford, ORCB, Oxford OX3 7DQ, United Kingdom), Mark Hodnett (Quality of Life Div. Natl. Physical Lab., Teddington TW11 0LW, United Kingdom), and Constantin-C. Cousseios (Univ. of Oxford, Oxford OX3 7DQ, United Kingdom)

Key to the success of high-intensity focused ultrasound (HIFU) as a clinical tool is the development of standardized quality assessment procedures to assess the safety and efficacy of HIFU transducers. The present work details the development of a cylindrical sensor array to be positioned around the HIFU focus during pre-treatment quality assessment of clinical transducers, which is designed to localize cavitation activity in three dimensions by passive mapping of the broadband emissions arising from inertial cavitation. The propagation of sound from a collection of broadband sources was first modeled to determine the optimum size, number and distribution of array elements for accurate mapping, and characterization of the cavitation dynamics produced during HIFU exposure. The optimal array configuration was then manufactured from PVDF using a novel printed circuit board technique, and theoretical predictions of the spatial resolution that it could achieve were validated experimentally. Because inertial cavitation is a pressure driven phenomenon, the ability to map cavitation activity using this array could provide a novel tool for rapid mapping of pressure fields produced by clinical HIFU transducers, in addition to providing invaluable information about the evolution of cavitation dynamics during HIFU exposure.

### 9:25


Compared with the two other mainstream medical imaging methods, CT and MRI, the worst weakness of ultrasound imaging is poor resolution—the ability to resolve tiny variations in tissue structure or texture. Due to sound wave diffraction and the way imaging ultrasound signal is retrieved, a point target in object domain does not generate point image in image domain, but an image spot with sophisticated spot pattern and considerable spot size as functions of point target location relative to the transducer—the most direct cause of poor resolution. The mainstream techniques aimed at reducing the image spot, either by shortening the impulse signal or by sharpening the beam focusing, are unavoidably accompanied by the deteriorated sensitivity and depth of penetration. It is a common consent in medical ultrasound community that ultrasound imaging is very close to its theoretical resolution limit. This article presents a different approach we named E-mode imaging that uses a diffraction-theory-based spot pattern recognition technique to account for the effects of image spot pattern in image processing. With the same set of ultrasound data that B imaging is based on, E-mode imaging achieved five to ten times better resolution and diagnostic power in computer simulations.
A generalized formulation for spatial harmonic expansion suitable for calculation of scattering by arbitrary distributions of homogeneous cylinders in two dimensions and homogeneous spheres in three dimensions is presented. The method is applicable for general incident fields expressed as angular spectra of plane waves. The technique uses harmonic expansions, physical acoustic boundary conditions, and mode matching to represent scattering by each individual object as a diagonal operator. Interaction between objects is treated with a T-matrix approach in which scattering from one object is applied to another using translation operators that shift the origins of harmonic expansions between scatterers. The generalized form permits efficient calculations in both two and three dimensions. These calculations are used to assess the validity of two-dimensional approximations to three-dimensional scattering for volumes containing multiple interacting spheres that are illuminated by elevation-focused beams. Differences in the two-dimensional and three-dimensional results are attributed to out-of-plane scattering and partial-volume effects caused by significant variations in the scatterer cross section over the elevation width of the focused beam.

9:55—10:30 Break

10:45
3aBB9. Characterization of clutter in ultrasonic imaging using a nonlinear full-wave simulation method and in vivo human tissue. Jeremy J. Dahl (Dept. of Biomedical Eng., Duke Univ., Durham, NC 27708, jeremy.dahl@duke.edu) and Gianmarco F. Pinto (Laboratoire Ondes et Acoustique, ESPCI, Paris, France)

Clutter is a well-known image degradation mechanism in ultrasonic imaging that is patient dependent. Despite widespread agreement on clutter sources, there are few studies on sources and their contribution to overall image clutter. We have performed simulations of clutter using a nonlinear, full-wave method to demonstrate the characteristics and sources of ultrasonic clutter. Representations of human abdominal layers, created from histological stains of human abdominal layers, were placed at the surface of the transducer. Point targets and anechoic cysts were simulated to demonstrate the characteristics of clutter resulting from these near-field layers. We have also performed matching in vivo experimental studies in human bladders and livers. The simulations and experiments show consistent results. Clutter resulting from near-field tissue layers is spatially coherent with respect to axial motion and incoherent in the sampled pressure field. The contribution of off-axis clutter resulting from the tails of the point-spread-function is shown to be relatively small (<5%) compared to the contribution of clutter associated with near-field layers (approximately 68%). Thus, the dominating source of clutter is associated with multiple reflections in near-field tissue layers. Clutter due to off-axis scattering dominates only when an interfering target is close to the imaging target. [This work is supported by the NIH Grant R21-EB008481 from the NIBIB. Technical support was provided by the Ultrasound Division at Siemens Medical Solutions USA, Inc.]

11:10
3aBB10. Modeling nonlinear acoustic waves in media with inhomogeneities in the coefficient of nonlinearity. L. Demi (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, l.demi@tudelft.nl), M. D. Verweij (Delft Univ. of Technol., 2628 CD Delft, The Netherlands), and K. W. A. van Dongen (Delft Univ. of Technol., 2628 CJ Delft, The Netherlands)

The refraction and scattering of nonlinear acoustic waves play an important role in the realistic application of medical ultrasound. One cause of these effects is the tissue dependence of the nonlinear medium behavior. A method that is able to model these effects is essential for the design of transducers for novel ultrasound modalities. Starting from the Westervelt equation, nonlinear pressure wave fields can be modeled via a contrast source formulation, as has been done with the INCS method. An extension of this method will be presented that can handle inhomogeneities in the coefficient of nonlinearity. The contrast source formulation results in an integral equation, which is solved iteratively using a Neumann scheme. The convergence of this scheme has been investigated for relevant media (e.g., blood, brain, and liver). Further, as an example, the method has been applied to compute the one-dimensional nonlinear acoustic wave field in an inhomogeneous medium insonified by a 1 MHz Gaussian pulse propagating up to 100 mm. The results show that the method is able to predict the propagation and the scattering effects of nonlinear acoustic waves in media with inhomogeneities in the coefficient of nonlinearity. This motivates a similar extension of the three-dimensional INCS method.

11:15
3aBB11. Three-dimensional calculation of ultrasound propagation through a breast model. Jason C. Tillett (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Medical Ctr. Box 648, Rochester, NY 14642, tiltlett@ece.rochester.edu), Gheorghe Sala-Stara, Leon A. Metlay, and Robert C. Waag (Univ. of Rochester, Rochester, NY 14627)

A specimen of human breast comprised mostly of fat and connective tissue was imaged with a high-resolution magnetic resonance scanner. The volume of data was segmented to construct an acoustic model by assigning values of sound speed, density, and attenuation to voxels corresponding to fat and connective tissue. Propagation of a waveform from a point source through the model was calculated using a three-dimensional finite-difference time-domain k-space method that included perfectly matched absorbing boundary conditions to eliminate reflections at the edge of the computational domain. Because of the large 557×451×289 grid and computational demands of the three-dimensional FFT, the calculation was performed on a cluster of computers. The wave field was sampled in three orthogonal planes, one of which was offset from the origin to represent a receiving aperture. The aperture waveforms were corrected for geometry associated with the estimated location of the point source and used to calculate arrival-time and energy-level fluctuations. The root-mean-square arrival-time and energy-level fluctuations of the simulated propagation were consistent with measurements for comparable specimens of breast. The results demonstrate the feasibility of using the described combination of methods to provide insight about the way ultrasound beams are aberrated by tissue.
Session 3aEA

Engineering Acoustics and Psychological and Physiological Acoustics: Acoustic Impedance of the Ear

Daniel M. Warren, Cochair
Knowles Electronics, Inc., 1151 Maplewood Dr., Itasca, IL 60143

Susan E. Voss, Cochair
Smith College, Picker Engineering Program, Northampton, MA 01063-0100

Chair’s Introduction—8:00

Invited Papers

8:05

3aEA1. Middle-ear input impedance and middle-ear sound transfer. John J. Rosowski, Hideko H. Nakajima, Jeffrey T. Cheng, Mohamad A. Hamadeh, and Michael E. Ravicz (Eaton-Peabody Lab., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston MA 02114)

Measurements of the acoustic input impedance of the human eardrum, or tympanic membrane (TM), have been used as indicators of middle-ear sound transfer for over 70 years. Modern measurements include (i) middle-ear input impedance in single- and multi-frequency tympanometry, (ii) impedance-based estimates of ear-canal reflectance, a description of sound power absorption at the TM, and (iii) laser-Doppler measurements of the sound-induced velocity of specific points on the TM surface. The advantages and disadvantages of these techniques are discussed, and their use in identifying and estimating the effects of different middle-ear pathologies is illustrated. A new technique, opto-electronic holography, which allows measurements of the volume displacement of the visible surface of the TM, is also discussed. Common themes are (i) the impedance of the middle ear varies widely within populations of normal hearing subjects and (ii) input-impedance based estimates of sound transfer through the middle ear are hindered by the presence of flexibility within the ossicular chain. It is concluded that while middle-ear input impedance may adequately describe the acoustic load on hearing-aid receivers, this impedance, by itself, is not always an accurate indicator of pathological alterations in sound transfer through the human middle ear.

8:25

3aEA2. Effects of middle-ear disorders on ear-canal reflectance measures in human cadaver ears. Susan E. Voss (Picker Eng. Program, Smith College, Ford Hall, 100 Green St., Northampton, MA 01063, svoss@smith.edu), Gabrielle R. Merchant, and Nicholas J. Horton (Smith College, Northampton, MA 01063)

The development of acoustic reflectance measurements may lead to noninvasive tests that provide information currently unavailable from standard audiometric testing. Few systematic measurements exist on the effects of various middle-ear pathologies on the ear-canal reflectance, where reflectance is calculated from impedance measurements. This work uses a human cadaver preparation to examine how a variety of middle-ear pathologies affect power reflectance: static middle-ear pressures, middle-ear fluid, fixed stapes, disarticulated incudo-stapedial joint, and tympanic-membrane perforations. The change from normal in power reflectance, induced by the simulation of pathology, generally depends on frequency and the extent of the pathology (except for the disarticulation). The largest changes from normal occur as increases in power reflectance near 1000 Hz with both large static pressures and large amounts of middle ear fluid. Disarticulation of the stapes results in reductions in power reflectance at the lower frequencies, while fixation of the stapes results in low-frequency increases in power reflectance. Tympanic membrane perforations generally produce reductions in power reflectance for frequencies below 1000–2000 Hz that are systematically related to the size of the perforation. These preliminary measurements may help assess the utility of power reflectance as a diagnostic tool for middle-ear disorders.

8:45

3aEA3. Modeling diagnosis of the human middle ear. Pierre Parent (Mimosa Acoust., Urbana IL) and Jont Allen (Dept. of ECE, Univ. of IL, Urbana IL, jontalle@illinois.edu)

Tympanic membrane (TM) and ossicular (OSC) models have been studied for many years. Most are implemented in the frequency-domain via Kirchhoff’s equations. However, the frequency domain is less intuitive than the time-domain models and does not apply given non-linear phenomena. This research presents a human time-domain TM+OSC implementation, extending our cat model [J. Acoust. Soc. Am. 122, 918–931 (2007)] as a cascade of slightly mismatches transmission lines. The canal plane-wave volume velocity is uniformly distributed across the canal cross-section, and updated at ~1 MHz (to accurately simulate sound propagation). The time-domain model includes both forward and backward waves, enabling for the first time, the computation of one-way delays, thus explaining why multi-modal propagation observed on the TM are not critical. Results are validated with data from normal and pathological ears. Additionally, comparisons between the acoustic impedance in living and dead (temporal bones) ears shows significant temporal-bone artifacts. Because of ear-dependent impedance mismatch, canal standing waves are observed. It is frequently assumed that the umbo pressure is the best reference pressure. In fact the “forward-pressure” \( P_f \) is a better estimate of the ideal reference, the cochlear pressure [J. Acoust. Soc. Am. 125, 1605–1611 (2009)].
9:05


An eardrum impedance simulator has been incorporated into life-size replicas of the human ear canal. The simulator branches off from the ear canal at the center of the nominal eardrum position. It has been designed so that sound pressure distributions in an ear canal will have the same standing wave ratios as those measured previously in human subjects over the 4–10-kHz range. The key component of the eardrum simulator is a micro-perforated plate, whose holes provide damping through viscous and thermal boundary layer losses. The optimum design specifies 109 holes of 0.06 mm diameter in 0.003-in. shim stock. Measurements of sound pressure distribution in replica ear canals are underway to evaluate the acoustical performance of the eardrum simulator. The sound field is generated by an occluding hearing aid test fixture.

3aEA5. Forward-pressure level for in-the-ear calibration. Stephen T. Neely and Michael P. Gorga (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, neely@boystown.org)

When acoustic impedance of the ear canal is known, sound pressure measured by a microphone in the ear canal can be decomposed into forward and reflected components. This decomposition provides a means to overcome problems associated with standing waves that may cause in-the-ear sound level calibration to be unreliable at high frequencies. Compared to total pressure measured at the sound source, we have demonstrated that forward pressure provides a more reliable estimate of sound quantity entering the ear, especially at frequencies above 4 kHz. Specifically, we have demonstrated increased reliability in measurements of distortion-product otoacoustic emissions (DPOAEs) when we use forward-pressure level (FPL) calibrations. Also, we have shown that behavioral threshold measurements for stimuli calibrated in FPL are more reliable than conventional calibration methods in the 4–10 kHz range. FPL calibration may provide advantages at high frequencies (1) for assessing hearing status with DPOAEs and (2) for fitting hearing aids using real-ear measurements. Objective methods of assessing hearing loss and fitting hearing aids at high frequencies are needed because the 6–12 kHz range is known to be important to young children for learning speech. [Work supported by Grants DC2251, DC8312, and DC4662 from the NIH/NIDCD.]

3aEA6. In-situ calibrated sound signals and hearing sensitivity. Robert Withnell (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, rwithnel@indiana.edu), Pat Jeng (Mimosa Acoust., Champaign, IL 61820), William Shofner (Indiana Univ., Bloomington, IN 47405), and Jon Allen (Univ. of Illinois, Urbana, IL 61801)

The acoustic input impedance of the ear provides data on the acoustico-mechanical function of the ear and can be used diagnostically to assess outer and middle ear function. Sound signals delivered to the ear (canal) can be quantified in-situ if the complex reflectance of the ear has been determined. Hearing sensitivity can be expressed in terms of the forward-going sound pressure wave, or after correcting for acoustic delay, the fraction of the forward-going sound pressure wave transmitted to the middle ear. The latter is an estimate of the signal the cochlea receives. Data will be presented examining the virtues of expressing hearing sensitivity in terms of an in-situ calibrated sound signal.

10:05—10:25 Break

10:25


The effects of a vent or open fitting are determined by the vent and ear acoustics. The dominant effects are a high-pass filter applied to the hearing-aid frequency response and a low-frequency direct signal path through the vent into the ear canal. Because these effects occur predominantly at low frequencies, simple lumped-parameter acoustics can be used to model them. The vent is modeled as an acoustic mass, and the ear canal and middle ear impedances as equivalent acoustic volumes. The results of using these simple approximations will be compared to calculations using acoustic tube models for the vent and ear canal and the Zwislocki middle-ear impedance model.

3aEA8. A comparison of two methods for estimating sound pressure level at the tympanic membrane at high frequencies for hearing aids. Tao Zhang, Karrie Recker (Starkey Labs., Inc., 6600 Washington Ave. South, Eden Prairie, MN 55344-3405, tao\_zhang@starkey.com), Janice LoPresti (Knowles Electrons LLC, Itasca, IL 60143), Matt Kleffner (Starkey Labs., Inc., Eden Prairie, MN 55344-3405), and William Ryan (Knowles Electrons LLC, Itasca, IL 60143)

For hearing aid fittings, it is important to know the delivered sound pressure level (SPL) at the tympanic membrane (TM) over a wide frequency range. Because it is infeasible to measure the SPL at the TM clinically, an indirect estimation based on measurements away from the TM is often used in practice. Furthermore, it is highly desirable that such a method does not require a clinician to use...
extra accessories to perform extra calibration in a clinic. Two such methods were proposed in the past. One used a set of empirically derived correction factors to estimate the SPL at the TM (Recker et al. 2009), the other used a lumped-element model to estimate the SPL at the TM [LoPresti et al. (2009)]. In this study, both methods were evaluated using actual hearing aid prototypes. An ITC hearing aid was built for each of ten participants. For each participant, the SPL in the ear canal was measured using a hearing aid microphone. The SPL at the TM was estimated using each method separately. The results were compared with the measured SPL at the TM. The relative strength and weakness of each method were discussed with the implications for clinical applications.

**Contribution Papers**

11:05  
3aEA9. Assess eardrum characteristics using standing wave technique.  
Wei Li Lin and Karrie Recker (Starkey Labs., Inc., 6600 Washington Ave. S., Eden Prairie, MN 55344, weili_lin@starkey.com)

It has been shown that eardrum reflection coefficients can be derived accurately up to 8 kHz based on standing wave patterns in the human ear canal, but not as reliably in higher frequencies [Lawton et al., J. Acoust. Soc. Am. 79, 1003 (1986)]. The primary purpose of this study is to investigate various factors affecting the accuracy of this type of measurements and extract important characteristics of ear canal and eardrum. In addition to reflection coefficients, eardrum impedance and the effect of loading on the eardrum are also examined. A model ear canal is constructed using a tube with variable loading conditions at one end. Measurements are taken along the center axis and contour of the tube when a stimulus is presented with frequency components up to 16 kHz, under both occluded and open situations. Results obtained from standing wave properties in the model ear canal are then compared to data acquired in real ear and in an ear simulator.

11:20  
3aEA10. Capturing low-frequency cochlear impedance in time domain models: The role of viscosity.  
Michael J. Rapson and Jonathan C. Tapson (Dept. of Elec. Eng., Univ. of Cape Town, Private Bag, Rondebosch, 7701, South Africa, michael.rapson@uct.ac.za)

Viscous effects in the cochlear fluid have been shown to be negligible for frequencies greater than 100 Hz [M. Viergever, Ph.D. thesis]. However, they become important at lower frequencies; hence viscous damping is sometimes included in helicotrema models [S. T. Neely and D. O. Kim, J. Acoust. Soc. Am. 79, 1472–1480 (1986)]. Modeling viscosity in time-domain computational cochlear models for wide-band input signals is problematic. Since the boundary layer thickness is inversely proportional to frequency, Navier–Stokes formulations require fine spatial resolutions, and fluid potential formulations are inherently inviscid models. Lumping all viscous effects into the helicotrema is not physiologically accurate as “the helicotrema does not appear to be a dominant constriction” [Lynch et al., J. Acoust. Soc. Am. 72, 108–130 (1982)]. An alternative is proposed whereby a fluid potential formulation is augmented with frequency dependent viscosity corrections that are based on prior analysis of the cochlear geometry and become negligible at higher frequencies. For simplified fluid boundary conditions, the predictions of this fluid model are compared to Navier–Stokes, potential, and lumped viscosity models for frequencies in the range 10 Hz–10 kHz. The simulations are also compared to physiological data for the input impedance of the cochlea.

**Invited Papers**

8:30  
3aMU1. What is known about banjo science and what is not yet known. Thomas D. Rossing (Stanford Univ., Stanford, CA 94305, rossing@ccrma.stanford.edu)

Banjos generally come in three different types: five-string banjo with a resonator, five-string banjo without a resonator, and four-string banjo with a resonator. Only recently have banjos been the object of much scientific study, and most of these studies have been directed at the American five-string banjo. These investigations have considered the modes of vibration of the principal parts, how they couple together, and how they radiate sound. In general, peaks in the driving point mobility correlate well with modes of vibration in the banjo head and with the radiated sound. Systematic modifications of some of banjo parts have been made by various investigators, and the effects of these modifications are considered.
3aMU2. A world beyond the banjo. Paul A. Wheeler (Dept. of ECE, Utah State Univ., 4120 Old Main Hill, Logan, UT 84322-4120, paul.wheeler@usu.edu)

The American banjo, a long-necked plucked lute developed by slaves from Africa and an important instrument in Bluegrass music, derives its characteristic timbre from the membrane belly of the instrument. Even though the banjo is the best known membrane-style lute in music of the western hemisphere, there are many musical traditions across the eastern hemisphere which have used their own version of the banjo long before the development of the American banjo. This paper will introduce some of these instruments and the musical cultures using them. These include the ngoni of western Africa (possibly the source of the American banjo), the cumbus and tar from the Middle East, the rawap of central Asia, the sarod of India, and the shamisen and sanxian of Japan and China. As diverse as these cultures may be, the characteristic timbre of the banjo is an important element in their music.

3aMU3. Time-resolved studies of banjo head motion. Thomas R. Moore and Laurie A. Stephey (Dept. of Phys., Rollins College, Winter Park, FL 32789)

Almost all of the radiated sound from a banjo can be attributed to the motion of the membranic head. This motion depends critically on the physical properties of the membrane as well as which string is plucked and the direction of the initial displacement. We present time-resolved studies of the motion of the membrane of an American five-string banjo after plucking and discuss the interaction between the string and the vibrating membrane.

3aMU4. The banjo: Developing a plausible model from first principles. Joe Dickey (3960 Birdsville Rd., Davidsonville, MD 21035, dickey@jhu.edu)

The banjo is one of the few musical instruments which is amenable to analytical modeling. It is fundamentally two connected canonical wave bearing systems, and as such, it can be solved. Furthermore, the requisite computer code runs quickly, making it practical to examine the instrument’s response as functions of a variety of parameters. The parameters are the things banjo players tweek in their attempt to get the sound they want; things like head tension, bridge mass, and string gauge. This tutorial will start with first principles in both the physics and the mathematics and end with movies of the calculated head response and radiated sound as functions of time and frequency. Numerical models can never be exact. Real systems, and particularly musical instruments, are just too complex for this, but models can provide useful guidance by illuminating what might happen as parameters are changed.

3aMU5. The banjo, string theory for the rest of us. Joe Dickey (3960 Birdsville Rd., Davidsonville, MD 21035, dickey@jhu.edu)

An analytical model of a five string banjo is used to calculate the dynamic response of a plucked banjo and the subsequent radiation of sound. This is the easy part: the difficult part is to identify and quantify the subtle differences between bad and good sound, whether that sound is measured or calculated. As a step toward this, the author has defined figures of merit (FOMs) to quantify radiated sound characteristics related to loudness, brightness, and decay, and to examine these FOMs as functions of the tension and propagation loss factor of the membranic head, the mass of the bridge, and the location of the pluck on the string. The calculated response does exhibit dual decay constants for radiated sound and a diminution of the sixth overtone. Qualitative agreement between calculated and measured mode patterns will be presented.

3aMU6. Membrane modes and air resonances of the banjo using physical modeling and microphone array measurements. Florian Pfeifle and Rolf Bader (Musikwissenschaftliches Institut, Universitaet Hamburg, Neue Rabenstrasse 13, 20354 Hamburg, Germany, florian.pfeifle@haw-hamburg.de)

A fullscale physical model of a banjo is implemented using a finite-difference formulation. The model consists of strings, bridge, resonance membrane, and the air enclosed in the geometry. The resulting time series of plucked strings in the model radiated from the membrane is integrated with respect to a room position in front of the banjo membrane and analyzed. Furthermore, the modes of the membrane are displayed as two-dimensional mode shapes for the plucked string as well as for the free vibrating membrane. For verification and refinements of the model, measurements on a banjo were performed using a 11×11 microphone array. The recorded time series were back-propagated to the surface of the membrane showing its vibrational patterns. These measured modes are compared to the modeled mode shapes in the low- to mid-frequency range. Furthermore, the air modes of the open sound holes are investigated with respect to their phase relations to the membrane. Radiation strength of these frequencies are strongly influenced by the opposite phase relations of these patterns. Also open-back and closed-back banjos are compared. The radiation behavior of these lower modes in the far field is then used to be implemented into the physical model of the banjo.

A short concert will follow the session in which techniques of Bluegrass 5-string banjos are demonstrated.
Session 3aNcA

NOISE-CON: Plenary

Courtney B. Burroughs, Chair
Noise Control Engineering, Inc., 1241 Smithfield St., State College, PA 16801

Chair’s Introduction—8:00

Invited Paper

8:05

3aNcA1. Effects of building mechanical system noise on worker performance and perception. Lily M. Wang (Architectural Engr. Prog., Peter Kiewit Inst., Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, lwang4@unl.edu)

This paper presents results from a number of studies that investigated the effects of noise from building mechanical systems on human task performance and perception. Three phases of research were conducted, each of which utilized a different set of noise signals produced by building mechanical systems: (1) broad-band noise at different levels and spectral qualities, (2) tonal noise conditions, and (3) noise conditions with time-varying fluctuations. In each phase, six different noise signals (many based on in-situ measurements) were reproduced in an office-like setting. Thirty participants completed tasks (e.g., typing, grammatical reasoning, and math) plus subjective questionnaires, while exposed to each noise condition for up to 1 h. In general, no statistically significant differences were found in task performance across the various noise signals tested. However, higher annoyance/distraction responses from the test subjects were often significantly correlated with reduced typing performance. These higher annoyance/distraction responses were also closely correlated to higher subjective ratings of loudness, followed by roar, rumble, and tones or fluctuations. Of particular interest is that a greater perception of low-frequency rumble was significantly linked to reduced performance on both the routine and cognitively demanding tasks. [Work supported by the American Society of Heating, Refrigeration and Air-Conditioning Engineers.]

8:45—8:50 Questions

8:50—9:00 Announcements

Session 3aNcb

NOISE-CON, Noise, and Structural Acoustics and Vibration: Aircraft Interior Noise

Vincent Cotoni, Chair
ESI, 12555 High Bluff Dr., Ste. 250, San Diego, CA 92130

Contributed Papers

9:15

3aNcb1. Sound transmission through double wall aircraft structures/mounting effect. Neple Pascale (Airbus Operations SAS, 316 Rte. de Bayonne, 31060 Toulouse Cedex 09, France, pascale.neple@airbus.com), Atalla Noureddine, and Bolduc Maxime (GAUS, Univ. of Sherbrooke.)

In the context of noise control treatment identification for turbulent boundary layer excitation, the main objective of this work is to assess the impact of mechanical links on the sound transmission loss through a double wall aircraft structure. An assembly of a stiffened panel made of composite fiber reinforced plastic (with a critical frequency around 6000 Hz) and a trim panel with a honeycomb core (with a critical frequency around 3000 Hz) has been considered in the whole study. The transmission loss of this double panel filled with glass wool blankets has been measured under diffuse sound field and point force excitation in the 100–10000 Hz frequency range for a fully decoupled configuration and a totally coupled one (rigid mounting). Extensive simulation using transfer matrix methodology has been conducted on the double panel for different glass wool filling rates, density, sound barriers, and add-on damping patches, for fully decoupled and rigid configuration. The main result is that the rigid mounting decreases significantly (up to 10 dB at 1000 Hz) the double wall transmission loss compared to a fully decoupled mounting and the acoustic efficiency of the passive treatments as well, whatever the excitation field.
In this paper, the use of polyimide foam as a lining in double panel applications is considered. Polyimide foam has a number of attractive functional attributes, not the least of which is its high fire resistance, thus making its use desirable in some sound transmission applications. The configuration studied here consisted of two 0.04×94 thick, flat aluminum panels separated by 5 in., with a 3 in. thick layer of foam centered in that space. Random incidence transmission loss measurements were conducted on this buildup, and conventional poro-elastic models were used to predict the performance of the lining material. The Biot parameters of the foam were determined by a combination of direct measurement (for density, flow resistivity and Young’s modulus) and inverse characterization procedures (for porosity, tortuosity, viscous and thermal characteristic length, Poisson’s ratio, and loss factor). The inverse characterization procedure involved matching normal incidence standing wave measurements of absorption coefficient and transmission loss of the isolated foam with finite element predictions. When the foam parameters determined in this way were used to predict the performance of the complete double panel system, reasonable agreement between the measured transmission loss and predictions made using commercial statistical energy analysis codes was obtained.

### 3aNCb2. Validation of a polyimide foam model for use in transmission loss applications


The vibroacoustics behavior of aircraft-type stiffened panels is classically analyzed using deterministic methods such as the finite element and boundary element method at low frequencies or energy based methods at higher frequencies. In the present work, a general semi-analytical method based on modal expansion technique is developed to predict the vibration and acoustics radiation of both metallic and composite flat stiffened panels. Both unidirectional and bidirectional stiffened panels with eccentric stiffeners and various shapes are analyzed using the same matrix formulation. The presented model is also able to predict the response of both regular and irregular stiffened panels. The contributions of the force and moment modal coupling at each beam location are accounted for together with the effect of interaction between ribs in the case of orthogonal stiffened plate. The response to various types of excitations (point force, diffuse acoustic field, and turbulent boundary layer) are presented in terms of their joint acceptance. The model is numerically validated by comparison with the FEM/BEM and hybrid FEM/SEA methods for various configurations and excitations. Excellent agreement is found.

### 3aNCb3. Analysis of sound transmission through periodic structures typical for aircraft fuselages

Vincent Cotoni (ESI Group, 12555 High Bluff Dr., San Diego, CA 92130, vincent.cotoni@esi-group.com)

A novel numerical method is applied to calculate radiation properties and sound transmission loss of periodic structures typical for aircraft fuselages. The concept rests on that an accurate statistical energy analysis representation of the structure is determined from a small finite element (FE) model consisting of one or a few cells of the periodic structure. The FE model can be kept small and computationally efficient and can for this reason be used for parametric studies of the effect of design changes. Four different structures are analysed: a flat and a curved rib-stiffened panels typical of an aircraft fuselage, and an aluminium honeycomb inner floor panel. Calculated transmission loss results are found to compare well to measured data. Also mechanical power inputs due to point force excitations as well as radiation efficiencies are calculated and compared to measured data.

### 3aNCb4. Vibroacoustic characterization of a commercial airliner bulkhead

Albert Allen and Mark Moeller (Spirit AeroSystems, P.O. Box 780008, MC K95-79, Wichita, KS 67278, albert.allen@spiritaero.com)

The commercial airliner bulkhead design candidate of concern is an internal divider wall consisting of an aluminum forward panel and liner aft panel sandwiching an array of closely spaced interstitial aluminum I-beams. The main purpose of the bulkhead, which separates the cargo bay from the occupied area, is to protect the crew from being compromised by cargo during a hard landing. Occupied area noise requirements also affect the bulkhead design as it is a primary path of noise transmission from the cargo bay. Vibroacoustic performance of the bulkhead analyzed with statistical energy analysis is supported by coupon testing of a representative bulkhead test section. Fibrous acoustic treatments were also evaluated experimentally for noise reduction potential. The experimental methods, results, and component level modeling complement each other by demonstrating a vibroacoustically appropriate bulkhead design and good bulkhead modeling practices.

### 3aNCb5. A semi-analytical method to predict the vibroacoustics response of composite and isotropic stiffened panels

Abderrazak Mejdi and Noureddine Atalla (Departement de Gnie mécanique, Univ. of Sherbrooke, 2500 Boul. de l’Universit Sherbrooke, PQ J1K 2R1, Canada, abderrazak.mejdi@usherbrooke.ca)

The vibroacoustics behavior of aircraft-type stiffened panels is classically analyzed using deterministic methods such as the finite element and boundary element method at low frequencies or energy based methods at higher frequencies. In the present work, a general semi-analytical method based on modal expansion technique is developed to predict the vibration and acoustics radiation of both metallic and composite flat stiffened panels. Both unidirectional and bidirectional stiffened panels with eccentric stiffeners and various shapes are analyzed using the same matrix formulation. The presented model is also able to predict the response of both regular and irregular stiffened panels. The contributions of the force and moment modal coupling at each beam location are accounted for together with the effect of interaction between ribs in the case of orthogonal stiffened plate. The response to various types of excitations (point force, diffuse acoustic field, and turbulent boundary layer) are presented in terms of their joint acceptance. The model is numerically validated by comparison with the FEM/BEM and hybrid FEM/SEA methods for various configurations and excitations. Excellent agreement is found.

### 3aNCb6. Quiet honeycomb panel

Dan Palumbo and Jake Klos (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, d.l.palumbo@nasa.gov)

Sandwich honeycomb composite panels are lightweight and strong and, therefore, provide a reasonable alternative to the aluminum ring frame/stringer architecture currently used for most aircraft airframes. The drawback to honeycomb panels is that they radiate noise into the aircraft cabin very efficiently provoking the need for additional sound treatment which adds weight and reduces the material’s cost advantage. A series of honeycomb panels incorporating different strategies aimed at reducing the honeycomb panels’ radiation efficiency while at the same time maintaining its strength. The majority of the designs were centered around the concept of creating areas of reduced stiffness in the panel by adding voids and recesses to the core. The effort culminated with a reinforced/recessed panel which had 6 dB higher transmission loss than the baseline solid core panel while maintaining comparable strength. Attempts were made to damp the panels’ vibration energy by the addition of lightweight particles to the honeycomb cells. These designs were very difficult to build given the particles’ tendency to pollute the bond interface between the honeycomb and the face sheet. Well constructed panels exhibited very little benefit from the treatment that could not be attributed to the added mass alone.

### 3aNCb7. Prediction of the cabin noise of business jet using statistical energy analysis

Washington de Lima and Apoorv Ravindran (Cessna Aircraft Co., One Cessna Blvd., M.S. C3-353, Wichita, KS 67215-1424, washdelima@hotmail.com)

The application of deterministic method (FEM and BEM) to predict vibroacoustic response of complex system like an aircraft fuselage is limited to low frequency due to large size of computational model for high frequency and the high frequencies modes are very sensitive to variation in the system properties which introduces substantial uncertainty. Energy based method such as energy statistical analysis (SEA) is an alternative that avoids these difficulties. This paper presents the vibroacoustic response of business jet using SEA. The SEA model is validated using in-flight tests with the aircraft in various acoustic package configurations.
A freighter version of a commercial passenger airplane platform was developed. The new architecture includes an occupied space immediately aft of the flight deck and forward of the cargo space. A bulkhead separating the occupied space from the cargo space is designed to protect people from being compromised by cargo during a hard landing. Occupied space noise requirements affect the bulkhead design as it is a primary path of noise transmission from the cargo bay. Statistical energy analysis (SEA) noise models were developed to evaluate the noise performance of the bulkhead. Models were developed for ground test, flight test, and component configurations, respectively. The component model is described in a companion paper. A fourth, requirements model, was developed to demonstrate bulkhead noise compliance. The model development and correlation efforts are detailed in this document. SEA model results are shown to correlate reasonably well with the test data.

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10:45
3aNCc6. Evaluation of stiffeners for reducing noise from horizontal vibrating screens. David Yantek (CDC-NIOSH-PRL, 626 Cochran Mill Rd., Pittsburgh, PA 15236, dyantek@cdc.gov) and Shawn Catlin (Mallett Technol., Inc.)

Sound levels around vibrating screens in coal preparation plants often exceed 90 dB(A). The National Institute for Occupational Safety and Health (NIOSH) is developing noise controls to reduce noise generated by horizontal vibrating screens. NIOSH determined the vibration mechanism housings and the screen body to be the dominant noise sources on a horizontal vibrating screen. Researchers used beamforming, experimental modal analysis, and operating deflection shape analysis to examine noise radiated by the screen body. Based on these results, finite element analysis and a NIOSH-written program were used to estimate the sound power level reduction resulting from adding rib stiffeners to key locations on the screen sides. Rib stiffeners made from two sizes of steel channel and two different cross-sections, C and T, were evaluated. In addition, the effects of orienting the stiffeners horizontally and vertically were examined. Finally, the stiffeners were evaluated with the ends welded to the existing ribs on the screen sides and with the ends free. The results indicate that for a broadband input, the smaller T cross-section oriented vertically with the ends welded to the existing ribs was the best option. This configuration reduced the predicted A-weighted sound power level by 7 dB.

11:00
3aNCc7. An investigation of the noise dynamics in a southern Illinois underground coal mine. Marek Szary, Yoginder Chugh, William Bell (Carbondale College of Eng., Southern Illinois Univ., 1230 Lincoln Dr., Carbondale, IL 62901-6603, szary@engr.siu.edu), and Joseph Hirschi (Illinois Clean Coal Inst.)

Noise in an underground coal mine has dominant components generated mainly from three sources: (1) continuous mining machine (CMM), (2) roof bolters, and (3) cars/vehicles which are transporting personnel and/or coal. Each of these three noise sources also has a number of well defined sub-sources with their own noise characteristics. The CMM noise is comprised mainly of noise generated by coal cutting drum, wet scrubber for dust control, and coal transport conveyor (called also the CMM’s tail). Roof bolter’s noise is generated during the drilling of the roof bolt holes in the bolting process. Personnel and coal transportation vehicles generate noise from the power driven system. The personnel most exposed to these noises are operators of these machines and associated support personnel. Three selected techniques with appropriate instrumentation were used to monitor exposure of the personnel to the noise and noise energy over a period of time. The most common technique is based on the use of personal noise dosimeters. The sound level meters (both pressure and power) were also used to collect noise data in form of instantaneous readings and also to check calibration of other sound measuring instruments. Most useful information was obtained from continuous recordings of the noise over time. This paper discusses the variability or dynamics of the generated noise in both frequency and time domains.
of 61.5 and 89.0 dBA, respectively, while the operator and helper working on a surface coal auger were exposed to higher average sound pressure levels of 95.4 and 94.0 dBA, respectively.

11:30

This paper describes results from vibration study conducted for a construction of proposed building structures on a covered landfill site. Construction site is surrounded by various land uses, including mobile homes, single dwelling units, and infrastructures. Deep dynamic compaction (DDC) method is utilized to provide soil compactions necessary for the construction of the buildings and parking lot foundations. Concern with the potential structural damage and the human perceptions of the ground vibration exist because of close proximity of the existing structures (i.e., mobile homes, single homes, roadways, and bridges) to the compaction area. DDC induced ground-vibrations level were modeled using actual vibration measurements "pilot tests" and the site’s soil dynamic characteristics. Vibration mitigation plan were developed to minimize the impacts.

9:15
3aNCd2. Vibration impact of 150 MW cogeneration power plant on a semiconductor fabrication facility. Ahmad Bayat and Jon Byron Davis (490 Post St., Ste. 1427, San Francisco, CA 94102, abayat@vaconsult.com)

A 150 MW cogeneration power plant construction was planned next to an extremely vibration-sensitive semiconductor fabrication facility in Taiwan. Both facilities were under construction simultaneously. We were asked to evaluate the impact of the power plant vibration sources on the semiconductor facility and determine that they can coexist at only about 30 m separation distance. Some of the power plant sources included large rotating equipment such as 7000-HP heat extraction blowers, turbines, and pumps. Dynamic forcing functions were extracted from a similar plant. In situ transfer function mobilities were obtained using seismic load drop equipment. Predictions were made which were in-line with the as-built results with the exception of two blowers. Their foundation design had to be retrofitted with deep pile foundations to transfer their vibration to deeper stiffer soil.

9:45
3aNCd3. Techniques for measuring the propagation speed of vibration waves in the ground. Anthony Nash (Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, anthony.nash@cmsalter.com)

Characterizing the transmission of ground vibration is important for assessing impacts upon buildings located adjacent to a rail right-of-way. This type ground-borne vibration transmission is typically dominated by surface or "Rayleigh" waves. When predicting the impacts, it is helpful to understand the attenuation of vibration from one location to another along the surface of the ground. In many cases, measuring the time-averaged rms vibration at spaced locations away from a line source is sufficient if one wishes only to quantify the reduction in ground borne energy density. For other purposes, however, it is essential to know the speed (and length) of the propagating wave front. This paper discusses successful (and not so successful) techniques for measuring the propagation speed of Rayleigh waves in the ground using both steady state and transient "point" sources. In terms of meaningful data, the steady state vibration method was found to be inferior to the transient vibration method. The steady state technique generated contradictory information on phase and amplitude at the spaced measurement locations. In contrast, the transient technique always indicated that (1) the Rayleigh wave propagated outward from the source and (2) the vibration amplitudes decreased between the near and far measurement locations.

10:00

Rail transit operations often produce groundborne vibration with energy that extends into the audible frequency range. It is often necessary to predict potential ground/structure-borne vibration and noise transmission in the design of new buildings in close proximity to existing rail lines or for new rail facilities/alignments. In a detailed analysis, key factors include source vibration characteristics (spectral force input to soil and/or track structure), ground vibration propagation characteristics (spreading and material damping losses), building foundation type (soil/structure interaction and coupling losses), and structural design of upper floors (floor and wall resonances and attenuation through the building). This paper focuses on methods for obtain-
ing vibration propagation characteristics in rock which is an efficient transmitter of vibration at audible frequencies. Three methods for obtaining one-third octave band response functions are compared: empirical methods, finite element analysis, and seismic modeling.

10:15—10:30 Break

**3aNCd5. Observed frequency and depth effects on attenuation of ambient ground vibration.** Hal Amick and Michael Gendreau (Colin Gordon & Assoc., 150 North Hill Dr., Ste. 15, Brisbane, CA 94005, hal.amick@colingordon.com)

This article presents observed relationships between depth and frequency with regard to ambient ground vibration. Typically, site vibration studies are conducted on undeveloped locations (grass fields, parking lots, etc.) to assess the ambient vibration conditions for a potential building site. However, the vibration-sensitive equipment of concern is frequently placed below grade, such that the final ambient vibration conditions it experiences might be quite different from that shown in the field survey. An earlier study [NoiseCon (2007)] presented data illustrating how several factors—including building foundation stiffness, ground stiffness, and building geometry—all have an effect on the propagation from “ambient” vibration sources. This presentation extends that work, specifically examining the relationship between depth, frequency, and extent of attenuation associated with below-grade spaces.

**3aNCd6. Variation in measured force density levels of light rail vehicles.** Shankar Rajaram and Hugh Saurenman (ATS Consulting, 801 S. Grand Ave, Ste. 575, Los Angeles, CA 90017, srajaram@atsconsulting.com)

The procedure recommended by the Federal Transit Administration for detailed predictions of ground-borne vibration is based on measurements at an existing rail system and measurements at the target site. A force density level (FDL) is derived from train vibration and line source transfer mobility (LSTM) measurements at the existing system. This FDL is then combined with the LSTM measured at the target site to develop predictions. This empirical procedure usually provides more reliable predictions than computer modeling procedures. FDL is assumed to characterize the trains and track support system and LSTM is assumed to characterize the effects of local geology. Implicit assumptions when applying this procedure are that FDL is independent of the local geology and that the vehicle and track support system will be the same at the target site as they are at the vehicle test site. This paper considers the differences in FDLs measured at light rail systems in Los Angeles, Minneapolis, Phoenix, and Portland. Issues discussed are how to adjust for train speed, why the speed adjustments are system specific, causes for FDL variations when the vehicles and track support systems are nominally the same, and procedures for obtaining consistent FDL measurements.

10:45

**3aNCd7. Analysis and design of new floating slab track for special trackwork using finite element analysis.** James Phillips (Wilson, Ihrig & Assoc., Inc., 5776 Broadway, Oakland, CA 94618, jphillips@wiai.com)

This paper discusses the development of a new vibration isolation system for tunnel crossovers in a future rail line extension. The previous design included ballasted track on top of continuous concrete slabs on rubber vibration isolators. The goal of the new design was to replace the ballast and tie track with rails fastened directly to discrete slabs similar to the vibration isolation system utilized on running portions of track within existing tunnels. Finite element analysis (FEA) was used to evaluate the dynamic interaction between the vehicle, rails, resilient rail fasteners and the vibration isolation system. The results of the analyses guided the design in determining the size and thickness of the inertia slabs, the stiffness and design of the rubber vibration isolation pads, and the rail fastener selection.
Session 3aNSa

Noise, Physical Acoustics, and INCE: Military Noise Environments—Continuous and Impulsive I

Richard L. McKinley, Chair

Air Force Research Lab., 2610 Seventh St., Wright Patterson Air Force Base, OH 45433-7901

Chair’s Introduction—8:05

Invited Papers

8:10

3aNSa1. Hearing loss mitigation through applied acoustics. Kurt Yankaskas (Office of Naval Res. (Code 342), 875 North Randolph St., Arlington, VA 22203-1995, kurt.d.yankaskas@navy.mil)

Hearing loss in the Navy and Marine Corps has been accelerating for decades, reducing combat effectiveness and creating hazardous conditions. This paper will discuss applied acoustics in current and future programs and its related challenges. Source noise reduction limits the amount of energy transmitted to the structure or into the environment. Acoustic modeling, insulation/isolation, and personal hearing protection are vital tools in this process, but often adding weight and complexity. Source noise solutions such as isotropic superfinishing treatment of gears increases transmitted power and reduces fuel consumption and lubricant temperatures while increasing gear life and maintenance intervals. Great strides are also currently being made in the arena of personal hearing protection to reduce hearing loss rates in environments where reducing source noise is a viable option, such as the aircraft carrier flight deck. Technological advances in insulation and isolation and possible operational changes will also be discussed, adding to the tools required to approach this issue from an integrated systems approach. In conclusion, a systems integration approach to incorporating noise reduction into the design of vessels, vehicles, and platforms from the start of the design process will reduce life cycle costs, benefiting the Navy and Marine Corps.

8:30

3aNSa2. Navy aircraft carrier near-field jet noise and flight deck crew hearing protection. James A. Janousek (Naval Air Systems Command, 48110 Shaw Rd., Bldg. 2187, Patuxent River, MD 20670, james.janousek@navy.mil), Valerie S. Bjorn (Naval Air Systems Command, Arnold AFB, TN 37389-6000), Richard L. McKinley (Air Force Res. Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433-7901), and James K. Wilt (Naval Air Systems Command, Patuxent River, MD 20670)

Military jet operations on board US Navy aircraft carriers produce excessive acoustical noise fields in which personnel work. For those working in these noise environments, hearing impairment generally starts and worsens as they continue to perform countless high-stakes jet aircraft launch and recovery missions. Several noise measuring surveys were completed to characterize near-field jet noise on US Navy flight decks. These data set the foundation for numerous efforts to improve personal hearing protection and reduce jet engine noise.

8:50

3aNSa3. Near-field noise of high-performance military aircraft and required hearing protection. Richard L. McKinley (Air Force Res. Lab., 2610 Seventh St., Wright-Patterson Air Force Base, OH 45433-7901, richard.mckinley@wpafb.af.mil)

The near-field acoustic environment of a high-performance military jet aircraft is very complex and intense. The overall sound pressure levels at high-engine power settings can exceed 170 dB very near the aircraft and can be in the 130–150 dB range at personnel locations. This paper describes the measured near-field noise environments from several Air Force and Navy high performance aircraft such as the F-14, F-15, F-16, F-18, and F-35. This paper will also describe a technique of calculating the hearing protection requirements in octave bands using a baseline exposure criterion and noise dose.

9:10

3aNSa4. Characterization of near- and far-field nonlinearity in high-performance military jet aircraft noise. Tracianne B. Neilson, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, tbn@byu.edu), J. Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC 28801)

This study addresses the characterization of nonlinearity in military jet aircraft noise data in both the near and far fields. Previous work [Gee et al., J. Acoust. Soc. Am. 123, 4082–4093 (2008)] showed using a generalized Burgers equation-based model that nonlinear propagation effects were significant in far-field, ground run-up data from the F-22 Raptor for different engine powers and propagation angles. In this investigation, those findings are further vetted using data collected from the F-35 Joint Strike Fighter over greater propagation ranges and angles. The nonlinear nature of the near fields of the F-22 and F-35 jet noise is characterized through the calculation of maps of the waveform and waveform time derivative skewness as well as the crest factor. These results provide insights regarding short-range shock formation and the apparent origin of the nonlinearity-producing region of the plume. [Work supported by the Air Force Research Laboratory.]
On the battlefield a soldier is continuously exposed to various types of noise. Much of this exposure is noise generated by his own weaponry and weapons of close by troops. The exposure levels are between 160-dB peak for small arms and 190-dB peak at the soldier’s ear for some antitank weapons, with A-durations from 0.3 ms (small caliber) to 4 ms for large caliber weapons (e.g., Howitzers). As these parameters feed into the damage risk criteria which are in use by the armed forces of different countries, it is important that the recorded pressure time histories are as precise as possible. The paper will present the acoustic levels to which a soldier may be exposed by different types of weapon and under different circumstances. It also will present measurement techniques to use for reliable results.

Military noise environments, and, in particular, the noise environments faced by dismounted soldiers on the battlefield, are characterized by wide variations in ambient level. Situations can quickly and unexpectedly change from quiet conditions where the sound of a snapping twig might alert the listener to hostile enemy activity to extreme noise conditions where firefights, explosions, or loud machinery create levels of noise that can cause hearing loss in a matter of minutes or seconds. This poses a unique challenge for the designers of military hearing protection, who must produce systems that provide enough comfort and acoustic transparency to convince users to consistently wear them in quiet conditions to ensure that they will be in place if an unexpected blast or other potentially damaging noise exposure occurs. In this talk, we discuss some of the issues that can influence the willingness of listeners to wear hearing protection for extended periods in quiet environments, including acoustic factors that have been shown to influence situational awareness while wearing hearing protection. Additionally, lessons on the factors impacting the comfort of hearing devices drawn from studies of how consistently hearing aids are worn by hearing impaired listeners will be discussed.

**Contributed Papers**

10:45

*3aNSa5. Improved system for detection, localization, and classification of military impulse noise.* Jeffrey Allanach, Justin Borodinsky (Appl. Physical Sci. Corp., 475 Bridge St., Ste. 100, Groton, CT 06340, jallanach@aphysci.com), Jeffrey Vipperman, and Matthew Rhudy (Univ. of Pittsburgh, Pittsburgh, PA 15261)

Impulse noise monitoring systems are needed for quantifying the magnitude and time of impulsive noise events to ensure compliance and to provide an archival record of noise emanating from military installations. Currently, there are several commercial systems; however, they report an overwhelming number of false events generated by windborne noise and distant non-military acoustic events. This can bias noise statistics to the point where a meaningful assessment of the acoustic sound levels from a site is not possible. APS, in collaboration with the University of Pittsburgh, have developed an improved noise monitoring system for mitigating windborne and other sources of non-military noise. This system includes a collection of remote sensors capable of detection, localization, and classification. Each sensor uses an acoustic array and real-time signal processing codes to estimate the noise source location. Mitigation of windborne events is accomplished using cross-channel correlation analysis, beamforming, and classification. A prototype of this system is installed at the Marine Corps Base Camp Lejeune where it has been automatically reporting and archiving detected impulse noise. In this paper, we will present the specialized array processing algorithms, system design, and its performance as determined by months of detection statistics accumulated in the field.

10:25

*3aNSa6. Measurement and characterization of army impulsive noise sources.* Joel T. Kalb (U.S. Army Res. Lab., Aberdeen Proving Ground, MD 21005-5425, joel.kalb@us.army.mil)

Peak pressure levels in army noise environments range from 140- to 195-dB near soldiers firing energetic weapons and are all dangerous to unprotected cars. Blast reverberation in confined spaces and steady noise in armored vehicles is also hazardous, so complete assessment requires the widest possible measurement range. To accurately measure pressures in the free-field or under hearing protectors, microphone considerations include type, location, orientation, mounting, shape, and shielding from flash or mechanical shock. Blast-waves rise instantaneously and return to free-field ambient within 0.3–15 ms, while in enclosures this can increase to 100 ms, requiring wide bandwidth signal processing. In addition to hazard, acoustic detection and annoyance at greater distances from weapons involve propagation effects in the atmosphere and at the ground which alter the wave-shape.

11:00

*3aNSa9. Aircraft jet source noise measurements of a Lockheed Martin F-22 fighter jet using a prototype near-field acoustical holography measurement system.* Michael M. James (Blue Ridge Res. and Consulting, 13 1/2 W. Walnut St., Asheville, NC 28801, michael.james@blueridgeres.com), Kent L. Gee, Alan T. Wall (Brigham Young Univ., Provo, UT 84602), J. Micah Downing, Kevin A. Bradley (Blue Ridge Res. and Consulting, Asheville, NC 28801), and Sally Anne McInerny (Univ. of Alabama at Birmingham, Birmingham, AL 35205)

Military jet aircraft generate high levels of noise which require innovative measurement and analysis methods to characterize the jet noise. A near-field acoustic holography (NAH) system is being developed to provide model refinement and benchmarking, evaluate performance of noise control devices, and predict ground maintenance personnel and community noise exposure. A prototype NAH system was used to perform jet source noise measurements of an F-22 at Holloman AFB, NM. This NAH system utilizes a patch and scan measurement approach using a 90 channel microphone array and over 50 reference microphones. The microphone array was used to make pressure measurements on a planar surface extending from the jet nozzle to 75 ft along the plume and over 6 ft in height. These measurements were made parallel either to the shear layer or to the nozzle center line of the jet plume at three different offset distances. Measurements were made every 6 in. on the surface which results in over 1800 measurements per offset distance. The ground based fixed reference microphones were located parallel to the nozzle center line. The measurement approach and pressure mea-
measurements for four power conditions detailing the source size, frequency content, and ground effect will be detailed. [Work supported by AFRL SBIR.]

11:15
3aNSa10. Application of near-field acoustical holography to high-performance jet aircraft noise. Alan T. Wall (371 E 720 S, Orem, UT 84058), Kent L. Gee (Brigham Young Univ., Provo, UT 84602), Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC 28801), and Michael D. Gardner (Graduate Program in Acoust., Penn State Univ., State College, PA 16802)

Near-field acoustical holography (NAH) measurements were made to visualize the sound field in the jet exhaust region of an F-22 Raptor. This is one of the largest-scale applications of NAH since its development in the 1980s. A scan-based holographic measurement was made using a 90-microphone array with 15-cm regular grid spacing for four engine power settings. The array was scanned through 93 measurement positions along three different measurement planes in a region near 7 m from the jet centerline and 23 m downstream. In addition, 50 fixed reference microphones were placed along the ground 11.6 m from the jet centerline, spanning 30.8 m. The reference microphones have been used to perform virtual coherence on the measurement planes. Statistically optimized NAH has been used to backpropagate the sound field to the source region for low frequencies and to identify jet noise characteristics. Ground reflection interference and other non-ideal measurement conditions must be dealt with. Details relating to jet coherence-lengths and their relation to reference microphone requirements will be discussed. Preliminary results of this ongoing work will be presented. [Work supported by Air Force SBIR.]

11:30
3aNSa11. Experiments on the directivity of sound emission from firearms. Morten Huseby and Haakon Fykse (Norwegian Defence Establishment, Pb. 25, 2027 Kjeller, Norway)

Personnel firing weapons will be exposed to noise created by the expanding gunpowder gas escaping from the muzzle. The noise from a rifle is directive, up to 20 dB louder in the shooting direction than behind the weapon. Usually personnel firing a weapon is taking advantage of the lower noise level behind the weapon. However, today firearms will be used by soldiers in complex scenarios where the head of one soldier might be directly to the side or even slightly in front of the muzzle of another soldier’s weapon. Most countries training regulations have not been updated to include all effects of the directivity of firearms in complex scenarios. Unexpected results were obtained from previous measurements on weapons with flash suppressors attached to the end of the barrel. The noise may be considerably larger in front of the weapon with flash suppressor fitted. The angular resolution of previous experimental data was not high enough to provide an explanation. This motivated us to design an experimental setup where the pressure wave from the weapon is measured, at 80 cm from the muzzle, in 180 directions relative to the firing direction. The measurements are done with and without a flash suppressor mounted.

11:45

Reduction in impulsive noise hearing hazard by earplugs and earmuffs is calculated with an electroacoustic lumped-parameter model of HPD attenuation using real ear attenuation at threshold (REAT) data. Energy flows into the occluded volume along three paths, each considered as a piston: (1) the rigid protector mass moving against the skin, (2) leakage at the support, and (3) transmission through the protector material (a second piston within the rigid piston). Circuit elements are adjusted so loss matches REAT data assuming path 1 is important at low, 2 at middle, and 3 at high REAT frequencies. Applying the model to 384 REAT data sets for ANSI S12.6 method B naive users gives statistical frequency distributions of occluded volume and leakage elements. For a given free-field impulsive noise, the model pressure predictions under the protector are compared to measurements on human and acoustical manikin ears to check validity of assumptions. The hearing hazards of the measured waveforms and the predicted waveforms are calculated with our previously developed auditory hazard analysis algorithm for humans ear model. The result is a cumulative frequency distribution of hazard based on user fit data useful in finding the best protector for a given impulsive noise.
Session 3aNSb

Noise, Architectural Acoustics, INCE, and ASA Committee on Standards: Standards in Psychoacoustics

Klaus Genuit, Chair
HEAD acoustics GmbH, Ebertstrasse 30a, Herzogenrath, 52134, Germany

Invited Papers

8:00

3aNSb1. Need for standardization of psychoacoustics. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

In the last decades, psychoacoustic parameters gained more and more importance in the context of sound perception and assessment in various applications. In particular, loudness was introduced as a better parameter than A-weighted level, because it shows a much better correspondence with the subjective impression of “volume” (loudness). This parameter was the object of standardization in the last years and few loudness standards were established (DIN 45631/A1, ISO 532, ANSI S3.4). The calculation of sharpness is also standardized (DIN 45692) and the standardization of roughness is recently under consideration by the DIN. These efforts have led to a global acceptance and a widespread use of psychoacoustic parameters in the automotive and IT-field. However, since the human signal processing is complex and the development of valid calculation models difficult, psychoacoustic phenomena must further be studied. Moreover, the validity, accuracy, and applicability of psychoacoustic parameters to specific noises must be determined. Especially, in the context of the description and classification of environmental noise, the potential of psychoacoustics has not been fully used so far. This paper will show the current status of psychoacoustics and will discuss the needs for the future.

8:20

3aNSb2. Comparative study of a proposed time varying loudness standard for common sounds. Colin Novak, Helen Ule (Dept. of Mech., Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B 3P4, Canada, novak1@uwindsor.ca), and Tomasz Letowski (Army Res. Lab. Human Res. and Eng. Directorate)

Several calculation procedures exist and are standardized for the determination of loudness of steady sounds. While loudness models for unsteady sounds do presently exist, they differ in their outcomes and have not been yet sufficiently validated using round robin or other appropriate evaluation methodology. Recently, the Deutsches Institut fur Normung (DIN) announced that it will soon issue a new standard describing a method for calculation of loudness of time-varying sounds. However, the DIN 45631-A1 takes a less common approach in that it does not fully describe a methodology to calculate the unsteady loudness. It instead offers a more general procedure along with open-ended checks and balances for verifying the progress through the various calculation steps. The caveat of this is that while conditions specified by this standard may be satisfied, varying results may be achieved due to the procedures’ open architecture. Conversely, the checks and balances may be enough such that identical results are achieved for the same sounds. This study investigates several commercial applications using this standard and compares their results for several common inputs. The results of the study will be discussed in terms of the extent to which compliance with DIN 45631-A1 results in identical loudness scores.

8:40

3aNSb3. An overview of issues concerning loudness measurement versus time. Wade R. Bray (HEAD Acoust., Inc., 6964 Kensington Rd., Brighton, MI 48116, wbray@headacoustics.com)

An overview of present methodologies for loudness measurement, especially considering time-varying sounds, is given. An ideal loudness-measuring method would mimic both the spectral and temporal acuities of human hearing including the relationship of those acuities at all audible frequencies. Due in part to practical considerations in their development history, none of the present set of loudness-measurement tools offers fully accurate representation of subjective time-frequency impressions. The most general limitations are insufficient spectral resolution and frequency-dependent time resolution. A recently standardized method well matches the temporal behavior of subjective loudness perception but only above a certain frequency due to lower-frequency time-resolution limits imposed by prior standards necessitating compliance. Examples will be shown of trade-offs and varying balances among several critical factors: time and frequency resolution at all audible frequencies, representation of temporal masking and growth/release effects, and appropriate spectral representations over varying combinations of broadband and narrowband signals.

9:00

3aNSb4. Loudness models applied to technical sounds. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, 52134 Herzogenrath, Germany, roland.sottek@head-acoustics.de)

Loudness evaluation has become a central focus for assuring better consideration of subjective sound intensity phenomena than frequency-weighted levels such as dB(A). Different standards are available for evaluating loudness of stationary sounds (ANSI, ISO, and DIN). In principle, standards are very helpful in daily life, even if the method does not consider the most recent research results. But a variety of standards generating different results, at least for the most important technical or industrial sounds, may lead to confusion. Considering that most of these sounds are time variant, a single model of time-varying loudness is preferable. At present the recently
published extension of the German standard for time-varying loudness (DIN 45631/A1) is the sole standardized method, applicable also for stationary sounds. There are other models allowing for loudness evaluation of time-variant sounds. Some years ago, a Hearing Model was borne out of a widespread interest to have a single standardized psychoacoustic model describing and explaining many phenomena at once. The Hearing Model is based on the physiology of human hearing and has been validated by testing against previously conducted psychoacoustic research results. The different methods will be compared and validated with subjective tests using technical sounds.

Contributed Papers

9:20
3aNSb5. Safe lifetime occupational exposure-1 LONE (lifetime occupational noise exposure), Robert D. Bruce, Arno S. Bommer, Kimberly A. Lefkowitz, and Noel W. Hart (CSTI Acoust., 16155 Park Row, Ste. 150, Houston, TX 77084)

For many years, it has been recognized by those working in the field of industrial noise that understanding how much noise is needed to cause hearing loss over a lifetime is difficult to communicate to most people, even those who have an understanding of logarithms. The concept of expressing noise exposure in industrial situations without decibels is the focus of this paper. Eldred (“Sound Exposure without Decibels” Internoise-86) discussed this approach for community noise. ANSI Standard S3.44-1996 defines sound exposure with units of Pascals squared seconds, or PASQUES, as noted by Eldred. This paper proposes that a safe value for lifetime occupational exposure to noise be expressed in terms of PASQUES. The authors will discuss the pros and cons of such an approach and offer $1.5 \times 10^6$ million PASQUES as the upper limit for a safe lifetime exposure to occupational noise.

9:35
3aNSb6. Earmuff noise attenuation by microphone in real ear technique, Samir N. Y. Gerges (Mech. Eng., Federal Universit of Santa Catarina, Florianopolis, Sc, Brazil, samir@emc.ufsc.br), Danilo A. Agurto, Jorge Arenas (Universidad Austral de Chile, Chile), Jose Bento Coelho, and Miguel M. Neves (IST, Lisbon, Portugal)

When using a hearing protector at the workplace, it is necessary to quantify its noise attenuation by laboratory measurements. This paper aims at developing an objective methodology for measuring hearing protector noise attenuation of earmuff type using “microphone in real ear” (MIRE) method. The methodology uses the “insertion loss” (IL) as parameter, which is the difference between the sound pressure level with and without the hearing protector, using MIRE, and then calculating the hearing protector IL. The results for four different hearing protectors are compared with the subjective method “real-ear attenuation at threshold” (REAT). Factors such as physiological noise and bone conduction are quantified using the relation between MIRE and REAT method. Results are compared with values obtained from the literature.
3aNSc2. Laboratory measurements of sound transmission at low frequencies. Chris Fuller (Dept. of Mech. Eng., VaTech, 133 Durham Hall, Blacksburg, VA 24061), Kathleen Kondylas (NEVA Assoc., Newburyport, MA 01950), and Natalia Levit (DuPont, Richmond, VA 23234, natalia.v.levit@usa.dupont.com)

The ability of acoustic materials to attenuate sound is usually determined in accordance with ASTM E-90 or ISO 140-3, both based on the diffuse field theory. However, the reality is that the size of typical reverberant chambers does not provide a sufficiently diffuse field at low frequencies. As a result, there is significant variation in sound transmission data of tested materials below 150 Hz. Several attempts were undertaken to improve the accuracy of measuring TL such as increasing field diffuseness by using reflecting surfaces, installing near-field array of loudspeakers, and using sound intensity probes on the receiving sides. None of these proved totally satisfactory. This work presents studies on using various approaches to more accurately determine acoustic mitigation of low-frequency transmitted noise. A metal box consisting of a very rigid frame supporting elastic panels on all sides was acoustically excited by a loudspeaker suspended inside of the box. The panels and box interior were designed to have very low-fundamental frequencies. The box was suspended by its frame in an anechoic room. An intensity probe was used to measure the transmitted sound from the box panels. The acoustic performance of various acoustic materials was compared for frequencies below 500 Hz.

10:30

3aNSc3. Acoustic impedance of Earth’s ground surface at infrasonic frequencies. Allan J. Zuckerwar (Analytical Services and Mater., Hampton, VA 23666, ajzuckerwar@yahoo.com)

The acoustic impedance of Earth’s ground surface is known to have a significant impact upon measurements of outdoor infrasound, e.g., sonic boom rise times. Current acoustic impedance models assume a rigid frame, or a no-slip condition, and thus lead to unbounded values of the real and imaginary parts of the acoustic impedance as the frequency approaches zero. The model of Tarnow [J. Acoust. Soc. Am. 105, 234–240 (1999)], on the other hand, considers relative motion between a porous frame (soil) and the air in the pores. It is shown here that this model yields finite values of the ground impedance in the limit to zero frequency. Further, extrapolation of selected experimental data to zero frequency by the method of the Richardson extrapolation also leads to finite values of the acoustic ground impedance. Finally, it is shown that the reflection of long-wavelength infrasound from bedrock may reveal resonance effects that will impact the ground impedance.

10:45

3aNSc4. Experience of the measurement and assessment of residential low-frequency noise complaints. David C. Waddington, Andrew T. Moorhouse, and Mags D. Adams (Acoust. Res. Ctr., Univ. of Salford, Salford M5 4WT, United Kingdom, d.c.waddington@salford.ac.uk)

The practical application of a procedure for the assessment of low-frequency noise (LFN) complaints is described. The development of the assessment method was published recently [J. Acoust. Soc. Am. 126, 1131–1141 (2009)]. The procedure includes guidance notes and a pro-forma report with step-by-step instructions for field measurements, complemented with an interview-based questionnaire together with an event log completed by the complainant. It does not provide a prescriptive indicator of nuisance but rather gives a systematic procedure to help environmental health practitioners to form their own opinion. Examples of field measurements and application of the procedure are presented. The procedure and examples are likely to be of interest to environmental health practitioners involved in the assessment of LFN complaints. [Work funded by the Department for Environment, Food and Rural Affairs (Defra), UK.]

11:00

3aNSc5. Research into the human response to vibration from railways in residential environments. David C. Waddington, Andrew T. Moorhouse, James Woodcock, Nathan Whittle, Sharron Henning, Eulalia Peris, Gennaro Sica, Andy Steele, Phil A. Brown, and Mags D. Adams (Univ. of Salford, Salford M5 4WT, United Kingdom, d.c.waddington@salford.ac.uk)

This paper describes progress in research being carried out at the University of Salford to develop a method by which human annoyance to vibration in residential environments can be assessed. The objective of this study is to yield a robust relationship between vibration exposure and human response, therefore providing a reliable basis for the development of standards and guidance for the assessment of vibration in residential buildings. The vibration sources to be considered are those affecting residents that are outside their control, such as construction, road, and rail activities. Noise is also a consideration. The protocol involves the measurement of vibration outside and inside individual residences and a social study questionnaire based on face-to-face interviews with householders. Work so far has concentrated on the response of people in their own homes to railway noise and vibration. Approximately 1000 case studies have been obtained, and examples of early field measurements and results are presented. This work is likely to be of interest to acoustical consultants and environmental health officers involved in the assessment of vibration complaints and to planners and practitioners involved in the design of buildings. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK.]

11:15

3aNSc6. Measurements of exploding balloon demonstrations. Julia A. Vernon, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, julia.vernon@yahoo.com), and Jeffery H. Macedone (Dept. of Chemistry and Biochemistry, Brigham Young Univ., Provo, UT 84602)

Exploding balloons are popular demonstrations in introductory chemistry and physical science classes and as part of outreach programs. However, as impulsive noise sources, these demonstrations constitute a possible hearing damage risk to both the demonstrator and the audience. Measurements of various hydrogen, hydrogen-oxygen, and acetylene-oxygen balloons were made in both a large lecture hall and an anechoic chamber. Condenser microphones (6.35 and 3.2-mm diameters) were placed at various angles, and distances from the balloon and time waveform data were collected at a sampling frequency of 192 kHz. For all balloon sizes tried, hydrogen-only balloons were found to produce peak sound pressure levels less than 140 dB at distances greater than or equal to 2 m. On the other hand, large (but reasonably sized) hydrogen-oxygen and acetylene balloons can result in peak levels exceeding 160 dB at a distance of 2 m, which constitutes a significant hearing risk for unprotected listeners at typical distances.

11:30

3aNSc7. Ultrasonic noise from manmade electrical devices. Trevor Jenny (tdwheats@hotmail.com), Brian E. Anderson (Dept. of Phys. and Astron., Acoust. Res. Group, Brigham Young Univ., Provo, UT 84602), and James A. TenCate (Los Alamos Natl. Lab., Los Alamos, NM 87545)

Measurements of ultrasonic noise emitted from manmade electrical devices have been made in an ultrasonic anechoic chamber. Until now, little has been published about the levels and frequency content of ultrasonic noise emitted from electronic devices such as laptops, cell phones, and other electronic devices. The most common source of ultrasonic noise in these devices is dc to dc power supplies. Measurements are made over the range of 20–200 kHz emitted by electronic devices (e.g., laptops). The results of these measurements and the challenges associated with making the measurements will be discussed (including anechoic chamber qualification in the ultrasonic frequency range). [This work has been funded by the Los Alamos National Laboratory.]
Session 3aPA

Physical Acoustics, Structural Acoustics and Vibration, and Noise: Sonic Boom I

Natalia V. Sizov, Cochair
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Victor W. Sparrow, Cochair
Pennsylvania State Univ., Graduate Program in Acoustics, 201 Applied Sciences Bldg., University Park, PA 16802

Invited Papers

8:00

3aPA1. Sonic boom research in NASA’s supersonic fundamental aeronautics project. Peter G. Coen (NASA Langley Res. Ctr., MS 264, Hampton, VA 23681)

The Supersonics Project of NASA’s Fundamental Aeronautics Program is a broad based effort designed to address the technical challenges associated with flight in the supersonic speed regime. One of the key environmental challenges that must be overcome for successful commercial supersonic flight is sonic boom. The NASA Supersonics Project has chosen this challenge as a focus for its research activity over the next several years. In cooperation with the FAA and the ICAO Committee on Aviation Environmental Protection (CAEP) NASA has initiated the development of a roadmap for research into the community response to sonic booms with noise levels significantly less than those of the Concorde and current supersonic aircraft. The purpose of the roadmap is to define the key research that should be accomplished in order to provide a firm technical basis for future noise-based standards for sonic boom acceptability. This presentation will discuss the elements of this roadmap and outline some of the recent research contributions made by the NASA Supersonics Project.

8:20

3aPA2. Random focusing of nonlinear N-waves in fully developed turbulence: Laboratory scale experiment and theoretical analysis. Philippe Blanc-Benon (CNRS, LMFA UMR 5509 Ecole Centrale de Lyon, 69134 Ecully Cedex, France, philippe.blanc-benon@ec-lyon.fr), Mikhail V. Averianov (Moscow State Univ., Moscow 119991, Russia), S. Ollivier (CNRS, LMFA UMR, 69134 Ecully Cedex, France), and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow 119991, Russia)

The high-amplitude shock wave generated by a supersonic aircraft propagates through the atmosphere toward the ground and generates an acoustic field with non-uniform pressure distributions strongly influenced by atmospheric turbulence. Recent numerical simulations based on generalized KZK-type equation including the effects of moving inhomogeneous media will be discussed. Formation of multiple focusing and defocusing zones is predicted. Nonlinear effects are significant not only in the random focusing zones but also in shadow zones of lower-pressure levels due to scattering of high frequencies from the areas of focusing. A statistical analysis is performed, and the results are compared to experimental data obtained in the controlled laboratory scale experiments conducted in the ECL anechoic wind tunnel. A high-power spark source is used to generate N-waves. Correlation length scales and spectra of the turbulent velocity field are measured. Statistical distributions and mean values for peak positive pressure and shock arrival time are obtained and found to be in a good agreement with modeling. In focusing areas, waveforms with amplitudes more than four times higher than those measured without turbulence are observed. Pressure amplitude probability density distributions are shown to possess autosimilarity properties when changing the intensity of turbulence. [Work supported by RFBR, French Government.]

8:40

3aPA3. A comparison of wide-angle and narrow-angle progressive wave equations for modeling sonic boom propagation through turbulence. Andrew Piacsek (Dept. of Phys., Central Washington Univ., 400 E. Univ. Way, Ellensburg, WA 98926-7422, piasek@cwu.edu), B. Edward McDonald (Naval Res. Lab, Washington, DC 20375), and Victor Sparrow (Penn State Univ., University Park, PA 16802)

The high-angle formulation of the nonlinear progressive wave equation model (NPE2) [McDonald, Wave Motion 31, 165–171 (2000)] has been adapted for atmospheric propagation and is used to perform numerical simulations of sonic boom propagation through atmospheric turbulence. Turbulence is incorporated into the model as perturbations of the local ambient sound speed, which have a random spatial distribution determined by a von Karman energy spectrum. Wave forms, peak overpressures, and wavefront profiles computed at fixed ranges from the onset of turbulence are compared to corresponding solutions produced by the original (narrow-angle) version of the NPE in order to quantitatively assess the importance of wide-angle accuracy for sonic boom propagation.
3aPA4. The influence of non-standard atmospheric conditions on the potential number of acoustic signature occurrences in the continental United States. Joe Salamone (Gulfstream Aervosp., 500 Gulfstream Rd M/S R-07, Savannah, GA 31402)

The potential number of supersonic flight operations and their resulting acoustic signature occurrences are important components for assessing the overall impact of supersonic flight over populated areas. Gulfstream Aerospace has recently published estimates for the number of acoustic signatures that would occur per day, on average, over the continental United States based on actual flight movements from large cabin business jets. The previous analysis of the flight movement data assumed propagation from a supersonic vehicle flying at a constant altitude and Mach number through a standard atmosphere and also assumed great circle routing between the origin and destination airports. The present work will examine the influence that realistic flight profiles and non-standard atmospheric conditions have on the location and extent of the acoustic signature occurrences. The non-standard atmospheric conditions will consist of seasonally averaged, upper-air profiles from nearly 100 locations across the continental United States. The realistic flight profiles will be based on estimates for the climb capability of the supersonic vehicle and possible air traffic routing trajectories.


The focusing effects of aircraft maneuvers and accelerations on shaped, or minimized, sonic boom signals have been investigated. The spectrally accurate boom prediction methodology of Pilon [AIAA J. 45, 2149–2156 (2007)] has been enhanced to include the effects of focusing through an implementation of the nonlinear Tricomi equation solver of Auger and Coulouvrat [AIAA J. 40, 1726–1734 (2002)]. The newly enhanced prediction tool was used to predict boom focus from aircraft accelerations, turns, and dives. Idealized near-field shaped boom signals were used as inputs to the code. Both fully resolved, continuous boom signals and booms containing zero-thickness shock waves were used as inputs to the focusing algorithm, in order to highlight the differences in perceived loudness on the ground. The new tool and results of the study will be used to help design accelerations and maneuvers for future supersonic cruise aircraft.

3aPA6. Three-dimensional finite-difference time-domain simulation of low-amplitude sonic boom diffraction around a rectangular test structure and building spiking effect. Sang I. Cho and Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, stc142@psu.edu)

A finite-difference time-domain (FDTD) approach is used for a three-dimensional numerical modeling of pressure loading on the exterior surface of a building due to an incident low-amplitude sonic boom. For these “low-booms,” the linearized Euler equations suffice to describe the boom-structure interaction. A second-order centered-difference scheme and fourth-order Runge–Kutta time integration scheme are applied. The FDTD simulation models a field test performed by Virginia Tech in 2008, where a rectangular test structure representing a residential building was instrumented to measure its vibroacoustic response to a boom generated by detonation cord. By matching the geometries of the simulated structure to the actual test structure and using the waveform of a measured boom as the input to the simulation, a direct comparison of numerical and experimental data is made possible. The simulation result agrees well with the experiment and also shows expected pressure doubling and diffraction effects. An interesting observation is made in both numerical and experimental data and is called the “building spiking” effect, which can be attributed to frequency-selective diffraction by the building. [Work supported by NASA. The authors appreciate Virginia Tech and NASA making the 2008 detonation cord test data available for this work.]

10:00—10:20 Break

10:20


Civil supersonic flight over land will only be allowed when sound levels are low enough to be acceptable to the public, and thus the sound field inside buildings needs to be well understood. To understand the field inside buildings, the sound field outside the buildings as well as the diffraction and loading on the building must be determined. A model to propagate sonic booms in urban landscapes was developed using a combined ray tracing and radiosity method. This model includes both diffuse and specular reflections for a single building configuration. The impact of diffuse reflections on the propagating and diffracting sound field will be examined, and multiple building configurations are also considered. The model results will be compared with data from NASA’s SonicBOBS (sonic booms on big structures) experiment conducted in September of 2009. [Work supported by NASA.]

10:40

3aPA8. Vibro-acoustic response of a simplified residential structure exposed to sonic booms. Part I: Numerical model. Marcel Remillieux, Joseph Corcoran, Ryan Haac, Ricardo Burdisso (Virginia Tech, 142 Durham Hall, Blacksburg, VA, 24061, mremilli@vt.edu), and Georg Reichard (Myers-Lawson School of Construction, Virginia Tech)

This paper presents a numerical model to predict the vibro-acoustic responses of simplified residential structures exposed to sonic booms. The model is validated experimentally in a companion paper. The dynamics of the fluid-structure system, including their interaction, is computed in the time domain using a modal-decomposition approach. In the dynamic equations of the system, the structural displacement is expressed in terms of summations over the in vacuo modes of vibration. The pressures inside the interior volumes are expressed as summations over the acoustic modes of rooms with perfectly reflecting surfaces. The structural modes are computed nu
Numerically using the finite element method. A shell element was specifically derived to model the structural components of typical residential buildings, e.g., plaster-wood walls, windows, and doors. The acoustic modes are computed for rectangular geometries using analytical expressions. Using modal decomposition, the dynamics of the fluid-structure system may be formulated by a finite set of ordinary differential equations (modal equations). These equations are then integrated with a Newmark algorithm to solve for the vibro-acoustic response of the system in the time domain. The system response may also be predicted in the frequency domain, by taking the Fourier transform of the time-domain response.

11:00
3aPA9. Vibro-acoustic response of a simplified residential structure exposed to sonic booms. Part II: Experimental validation. Marcel Remillieux, Joseph Corcoran, Ryan Haac, Ricardo Burdisso (Virginia Tech, 142 Durham Hall, Blacksburg, VA 24061, mremilli@vt.edu), and Georg Reichard (Myers-Lawson School of Construction, Virginia Tech, Blacksburg, VA 24061)

In a companion paper, a model was derived to predict the vibro-acoustic responses of simplified residential structures exposed to sonic booms. In the present paper, the experimental validation of the numerical model is presented. First, the experimental setup is described, including the structure, instrumentation, and external pressure loading. The structure was a single room made of plaster-wood walls and includes two double-panel windows and a door. The structure was extensively instrumented with accelerometers and microphones to record its vibro-acoustic response. Sonic booms with realistic amplitudes and durations were generated by an explosive technique. Subsequently, numerical predictions on the vibration of the structure and pressures inside the room are compared to experimental data, showing a fairly good agreement overall.

11:20
3aPA10. Distributed field measurements of low amplitude sonic booms. Joseph Gavin (Gulfstream Aerosp., 32 Lummus Dr, Savannah, GA 31408, joseph.gavin@gulfstream.com), Jack Arnold, Bryan Nadeau (Natl. Instruments, Austin, TX), Jacob Klos (NASA Langley Res. Ctr., Hampton, VA. 23681), and Edward A. Haering, Jr. (NASA Dryden Flight Research Ctr., Edwards, CA)

Quiet supersonic flight over land appears increasingly viable, based on significant advancements in aircraft shape optimization. One of the next critical milestones will involve community scale surveys, measuring the public response to events that sound as benign as distant thunder. The present paper describes a measurement infrastructure to reduce the field costs associated with those tests. Specifically, a scalable grid of NI CompactRIO data acquisition systems has been developed and deployed in a risk reduction test. Highlights of the system include communications via a wireless rf network and synchronization using low cost GPS receivers. This paper highlights some of the development challenges, solutions, and recent results from component and system level risk reduction testing at the NASA Dryden Flight Research Center.

11:40
3aPA11. Detailed finite element analysis of sonic boom transmission into residential building structures. Beom Soo Kim, W. Victor Sparrow (Graduate Program in Acoust., 201 Appl. Sci. Bldg., University Park, PA 16802, buk104@psu.edu), V. Natalia Sizov, and T. Ben Manning (Wyle Labs., Arlington, VA 22202)

Finite element modeling is a powerful and cost-effective tool for solving problems in structural dynamics and acoustics, particularly at low frequencies. In the present work the transient responses of the sonic boom transmission from outdoors to indoors were studied. Detailed finite element models of individual rooms from two residential structures were built. The simulated geometries were matched to the physical geometries of two houses exposed to sonic booms that were heavily instrumented by NASA in the summers of 2006 and 2007. The residential building walls were composed of studs and sheathing, and these were modeled using MSC Nastran/PATRAN software. To match the normal mode frequencies of the walls, it was necessary to not only match material coefficients but also the actual connections between the studs and the sheathing. The fluid structure coupling between the walls and the air was utilized to analyze the acoustic response of the sonic boom in the room interior. In order to verify the correctness of the model, the vibration responses of the structure and the acoustic cavity were compared with measured data. [Work supported by NASA.]
Session 3aPP

Psychological and Physiological Acoustics: Sound Source Localization

Joseph W. Hall, III, Chair
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Contributed Papers

9:00 3aPP1. Sound source localization of a dipole by the plainfin midshipman fish (Porichthys notatus), David G. Zedddies (Marine Acoust. Inc., 41 Fairfax Dr., Arlington, VA 22203), Richard R. Fay (Loyola Univ.-Chicago, Chicago, IL 60626), Peter W. Alderks, Andrew Acob, and Joseph A. Sisneros (Univ. of Washington, Seattle, WA 98195)

Localization of a dipole sound source was studied in female plainfin midshipman fish (Porichthys notatus). Experiments were conducted and videotaped in a 3.65-m-diameter tank using a dipole underwater speaker system placed near the center of the tank. The sound was a 90-Hz tone, approximately the fundamental frequency of the male’s advertisement call. Pressure and particle motion components of the sound field were mapped with 9-cm resolution. Pressure was measured using an eight-hydrophone array, and particle motion vectors calculated from the pressure gradients. Mapping confirmed that the projector was operating as a dipole. Gravid fish were released 70 cm from the sound source at two different positions relative to the dipole axis: one near the dipole axis and one near the pressure null axis. Twenty-five positive responses were recorded from each release site. The phonotactic response pathways along the dipole axis consisted of slightly curved tracks to the sound source, whereas pathways from the null axis consisted of greatly curved tracks to the source that followed the particle motion vectors. Results confirm that fish can locate a dipole sound source and are sensitive to the direction of acoustic particle motion. [Work supported by NSF.]

9:15 3aPP2. Generating correlated noise, William M. Hartmann and Yun Jin Cho (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824)

Research on binaural hearing frequently presents partially correlated noise to a listener’s two ears. Experimentally, there are two classic methods to generate noise with a controlled amount of cross-correlation: the two-generator method and the three-generator method [J. C. R. Licklider and E. Dzendolek, Science, 107, 121–124 (1948)]. Analytic formulas were developed and numerical experiments were done to study the properties of the two methods, particularly to determine how well the actual correlations approach targeted correlations. It was found that both methods give very similar results, with variance that is greatest when the targeted cross-correlation is zero and which decreases monotonically to zero as the targeted value increases to 1.0. The variance decreases inversely as the first power of the number of degrees of freedom in the noise, whatever the targeted correlation. Paradoxically, a far less popular method, the symmetrical generator method, is far more successful. The variance is zero when the targeted cross-correlation is either zero or 1.0, and the variance decreases inversely as the square of the number of degrees of freedom. The symmetrical generator method is a reasonable alternative to the summing of digitally orthogonalized noises. [Work supported by the NIDCD DC-00181.]

9:30 3aPP3. A binaural localization model that resolves front-back confusions through head movements, Jonas Braasch, John T. Strong, and Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

It is well known that head movements are instrumental in resolving front/back confusions in human sound localization. A mechanism for a binaural model is proposed here to extent current cross-correlation models to compensate for head movements. The algorithm tracks sound sources in the head-related coordinate system (HRCS) as well as in the room-related coordinate system (RRCS). It is also aware of the current head position within the room. The sounds are positioned in space using an HRTF catalog at 1 deg azimuthal resolution. The position of the sound source is determined through the inter-aural cross-correlation (IACC) functions across several auditory bands, which are mapped to functions of azimuth and supersonor. The maxima of the cross-correlation functions determine the position of the sound source, but unfortunately, usually two peaks occur—one at or near the correct location and the second one at the front/back reversed position. When the model is programmed to virtually turn its head, the degree-based cross-correlation functions are shifted with current head angle to match the RRCS. During this procedure, the IACC peak for the correct hemisphere will prevail if integrated over time for the duration of the head movement, whereas the front/back reversed peak will average out.

9:45 3aPP4. Modeling the precedence effect for multiple echoes, Matthew J. Goupell and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI 53705)

A typical and controlled way to study the precedence effect has been to use two pairs of binaural click stimuli with different lead-lag delays. For lead-lag delays between 1 and 5 ms, interaural differences in the lead click pair are more easily discriminated than those in the lag click pair. This work attempts to move from the classic, but unrealistic, two-source paradigm to a more realistic situation, much like real rooms in which multiple echoes occur. Using a three-source paradigm, listeners were presented three click pairs (lead, lag1, and lag2) via headphones. Discrimination of directional changes in each of the three click pairs was measured with a 2AFC task for delays between 0 and 130 ms. Results show that due to interactions amongst the three sources “echo thresholds” were affected in a non-monotonic way. Temporal order effects were observed for long delays, which led to poorer lead discrimination, and better lag2 discrimination. Summing localization was observed for short delays between lead and lag1, and lag1 and lag2. A simple weighting model [Shinn-Cunningham et al., J. Acoust. Soc. Am. 93, 2923–2932 (1993)] was adapted to include this multiple source data and the complex relationships between the three sources. [Work supported by NIH-NIDCD R01 DC03083—R.Y. L.]

10:00 3aPP5. Investigation into and modelling of head movement for objective evaluation of the spatial impression of audio, Changeun Kim, Russell Mason, and Tim Brookes (Inst. of Sound Recording, Univ. of Surrey, Guildford GU2 7XH, United Kingdom)

Research was undertaken to determine the nature of head movements made when judging spatial impression and to incorporate these into a system for measuring, in a perceptually relevant manner, the acoustic parameters which contribute to spatial impression: interaural time and level differences and interaural cross-correlation coefficient. First, a subjective test was conducted that showed that (i) the amount of head movement was larger when evaluating source width and envelopment than when judging localization
and timbre and (ii) the pattern of head movement resulted in ear positions that formed a sloped area. These findings led to the design of a binaural signal capture technique using a sphere with multiple microphones, mounted on a simulated torso. Evaluation of this technique revealed that it would be appropriate for the prediction of perceived spatial attributes including both source direction and aspects of spatial impression. Reliable derivation of these attributes across the range of ear positions determined from the earlier subjective test was shown to be possible with a limited number of microphones through an appropriate interpolation and calculation technique. A prototype capture system was suggested as a result, using a sphere with torso, with 21 omnidirectional microphones on each side. [Work supported by the Engineering and Physical Sciences Research Council (EPSRC), UK, Grant No. EP/D049253.]

10:15—10:30 Break

10:30 3aPP6. Tuning of head-related transfer functions using principal component analysis. Kimberley J. Fink and Laura R. Ray (Thayer School of Eng., Dartmouth College, 8000 Cummings Hall, Hanover, NH 03755)

Modeling head related impulse responses (HRIRs) to determine head-related transfer functions (HRTFs) can be used to create a virtual auditory display through head phones. However, non-individualized HRTFs can create errors in sound localization and perception of the source as coming from inside vs outside of the head. We present a method, based on model-order reduction and principal component analysis, to “tune” generalized HRTFs derived from the CIPIC database of HRIRs in order to individualize an HRTF. The impulse response for each individual in the CIPIC database is modeled at given azimuth locations on the horizontal plane. Model order reduction is used to eliminate poles and zeros of the identified HRTF at high frequencies, while preserving the frequency response below ~8000 Hz. Principal component analysis is then performed on the reduced-order models in order to reduce the dimensionality of the dataset and identify a subset of weighting variables that are adjusted during HRTF customization. The sensitivity of weights on the frequency response is also studied in order to correlate weights with HRTF features. A prototype auditory display has been developed that allows a user to tune HRTFs in real-time, while listening to a 44.1-kHz source presented from a chosen azimuth.

10:45 3aPP7. Effects of individualized headphone equalization on front/back discrimination of virtual sound sources. Abhishek Guru, William L. Martens, and Doheon Lee (Faculty of Architecture, Design and Planning, Univ. of Sydney, New South Wales 2006, Australia, agur4608@uni.sydney.edu.au)

Individualized head related transfer functions (HRTFs) were used to process brief noise bursts for a two-interval forced choice front/back shift between virtual sound source locations presented via two models of head phones, the frequency responses of which could be made nearly flat for each of 21 listeners using their own individualized headphone equalization filters. In order to remove tone coloration differences between virtual sources processed using individualized HRTFs measured in front or in back of each listener, the spectral centroid of the “back” source was adjusted to more nearly match that of the source processed using the “front” HRTF. This manipulation resulted in chance levels of discrimination performance for 12 out of 21 listeners. For the remaining nine listeners showing good discrimination, the virtual sources presented using individualized headphone equalization supported significantly better front/back discrimination rates than did virtual sources presented without correction to headphone responses.

11:00 3aPP8. Reverberation disrupts spatial selective auditory attention. Dorea R. Ruggles and Barbara G. Shinn-Cunningham (Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215)

When there are multiple, competing speech streams, listeners can selectively attend to a desired target using spatial cues like interaural time differences (ITDs). However, reverberation smears fine temporal information needed to encode ITDs. Reverberation may therefore interfere with the ability to selectively attend to a target from a particular direction. Subjects were asked to report a stream of digits simulated to be directly in front of them in the presence of two masker streams symmetrically positioned at 15 deg left and right. Because the maskers were statistically identical to the target (except in direction), listeners had to focus spatial attention on the target while ignoring the masker streams. Three levels of reverberation were simulated using a rectangular room model to measure the influence of reverberation on performance. Initial results showed that spatial selective attention degraded as reverberation increased. However, overall performance varied dramatically from listener to listener in ways that were not predicted by age or memory span. The current study explores whether individual differences are predicted by performance on two discrimination tasks that rely on signal fine temporal structure: frequency modulation (a monaural task) and ITD (a binaural task). [NIDCD, ONR, and NSF provided funding to support this work.]

11:15 3aPP9. The effects of intervening interference on working memory for sound location. Aurora Grossmann, Dennis Ries, and Traci Hamilton (School of Hearing, Speech, & Lang. Sci., Ohio Univ., Grover Ctr., Athens, OH 45701, ag303003@ohio.edu)

Inter-aural phase differences were used to study working memory for perceived auditory position quantified as a change in the difference limen (DL) in equivalent auditory angle across 300-, 5000-, and 15 000-ms retention intervals. Data were obtained for the medium and long intervals both in the presence and absence of intervening tones. Intervening stimuli within the medium and long inter-comparison intervals produced a significant increase in the DLs compared those obtained in the corresponding quiet conditions. The DL for the 300-ms interval was roughly equivalent to that obtained for the medium interval without intervening tones while that obtained for 15 000-ms interval was significantly greater than that obtained for either of the shorter intervals. The results suggest that the temporal decay of information within AWM of a listener regarding the location of a sound within their environment is so gradual that it can be maintained in trace memory for tens of seconds in the absence of intervening acoustic signals. Conversely, the presence of interpolated sounds within the retention interval may facilitate the use of context memory, resulting in a less detailed, but relevant representation of the location that is resistant to further degradation.
Session 3aSA

Structural Acoustics and Vibration and Physical Acoustics: Damping Mechanisms

J. Gregory McDaniel, Chair
Boston Univ., Dept. of Aerospace and Mechanical Engineering, 110 Cummington St., Boston, MA 02215

Chair’s Introduction—9:10

Invited Papers

9:15

3aSA1. Damping in auxetic materials and structures. Fabrizio Scarpa (Dept. of Aerosp. Eng., Univ. of Bristol, 83 Woodland Rd., Bristol BS8 1US, United Kingdom, fscarpa@bris.ac.uk)

Negative Poisson’s ratio (NPR) (otherwise called auxetics) materials and structures feature enhanced energy absorption under cyclic loading both at low- and high-frequency excitations. We show in this paper experimental and simulation results related to open cell PU-PE foams, honeycombs, and composites, all exhibiting a negative Poisson’s ratio value. The open cell auxetic foams provide a significant damping capacity at low frequency and high preload as well as under broadband excitation compared to the conventional foams from which they are derived. We show also that the acoustic absorption properties of the auxetic foams are significantly enhanced compared to conventional PU-PE under 500 Hz, and their loss factor increased using their specific shape memory property. We describe also the vibration damping properties of NPR honeycombs with different unit cell topologies and the damping capacity of NPR carbon fiber composites under flexural cyclic loading.

9:35

3aSA2. Intrinsic damping, relaxation processes, and internal friction in vibrating systems. Allan D. Pierce (Dept. of Mech. Engr., Boston Univ., Boston, MA 02215, adp@bu.edu)

Vibrations lose energy because of (1) coupling of the system to other systems or to the environment or of (2) intrinsic damping within the system itself. The present paper attempts to survey the current state of understanding for the principal mechanisms of the latter type with explicit physics (rather than purely phenomenological considerations), as simply as practical. Some mechanisms are evidenced in the attenuation of waves in unbounded homogeneous media, others are caused by the presence of boundaries. Some explicitly require quantum-mechanical concepts for their explanation, others can be explained in terms of classical mechanics and thermodynamics. Specific mechanisms that are to be discussed include those associated with thermal conductivity, grain and domain boundary effects, interstitial atom diffusion, relaxation by diffusion, inter-grain diffusion, micro-eddy current losses, hysteresis losses, thermoelastic internal friction, thermal damping associated with transverse vibrations of thin bodies, phonon-phonon interactions, phonon-electron interactions, internal fluids, and dislocations. The concept of relaxation often provides a convenient unified description. Rather than use the local averaged particle displacement as a sole description of the local vibrations, one considers additional local variables, loosely referred to as hidden variables. Each such variable, when disturbed, relaxes to its equilibrium value with a characteristic relaxation time.

9:55


This work considers passive damping treatments that dynamically interact with plates over areas that are much smaller than a flexural wavelength. Each treatment is modeled as a frequency-dependent admittance presented to the plate at a point. The frequency-dependence is required to satisfy the causality and minimum phase conditions, which are implicitly enforced via a specialized Fourier series in frequency. The Fourier coefficients, as well as the positions of the attachments, are regarded as optimization parameters to be varied so as to globally decrease plate vibrations over a frequency band of interest. Various cost functions are considered, such as the sum of modal loss factors over the band as well as acoustic radiation efficiencies. Examples involving a simply supported plate excited by a point force illustrate the approach. [Work supported by ONR under Grant No. N000140810531.]

10:15—10:30 Break
The idea in transformation acoustics is to map one region into another so that the new region has the same acoustic appearance to observers as the original. It is analogous to transformation optics and is closely related to devices for cloaking, but it is not restricted to cloaking. This talk will explore the implications of transformation acoustics for radiation damping and scattering. Mechanisms for realizing transformed regions, which we call acoustic metafluids, will be described. The objective in these materials is to bend the waves so that energy can be steered around a given region, and thereby reduce the scattering strength. But it is not as obvious how these materials influence the radiation from an active source inside. In this talk we will discuss the implications of transformation acoustics on radiation using fundamental concepts, including acoustical reciprocity, mechanical impedance, and energy flux.

Contributed Papers

10:50
3aSA5. Radiation reduction using layers of three fluids. Adam J. Nagy and Andrew N. Norris (Dept. of Mech. and Aerosp. Engng., Rutgers Univ., Piscataway, NJ 08854, adnagy866@gmail.com)

Significant reduction in target strength and radiation signature can be achieved by surrounding an object with multiple concentric layers comprised of three acoustic fluids. The idea is to make a finely layered shell with the thickness of each layer defined by a unique transformation rule. The shell has the effect of steering incident acoustic energy around the structure and, conversely, reducing the radiation strength. The overall effectiveness and the precise form of the layering depend upon the densities and compressibilities of the three fluids. The best results are obtained if one fluid has density equal to the background fluid, while the other two densities are much greater and much less than the background values. Optimal choices for the compressibilities are also found. Simulations in two dimensions and three dimensions illustrate the effectiveness of the three-fluid shell.

11:05
3aSA6. Transmission loss of membrane-type locally resonant acoustic materials. Christina Naify (Dept. of Mater. Sci., Univ. of Southern California, 3651 Watt Way, VHE 602, Los Angeles, CA 90089), Chia-Ming Chang, Bill Carter, Geoff McKnight (HRL Labs., Malibu, CA 90265-4797), and Steve Nutt (Univ. of Southern California, Los Angeles, CA 90089)

Membrane-type acoustic metamaterials were fabricated, characterized, and analyzed. Thin plates which obey the acoustic mass law have low transmission loss at low frequencies. Acoustic metamaterials with negative dynamic mass density have been shown to demonstrate a significant (ten times) increase in transmission loss (TL) over mass law predictions for a narrow band (100 Hz) at low frequencies (100–1000 Hz). The peak TL frequency can be tuned to specific values by varying the membrane and mass properties. In this work, transmission loss magnitude as a function of frequency was measured for variations in the mass magnitude, mass placement on the membrane, membrane size, and membrane thickness using an impedance tube setup. The dynamic properties of membranes constructed from different materials and thicknesses were measured and compared to the results of coupled field acoustic-structural FEA modeling to understand the role of tension and element quality factor. In addition, the behavior and limitations of stacking multiple membranes in series to increase the reflection bandwidth were explored.

11:20
3aSA7. Damping mechanism concepts in ocean wave energy conversion: A simplified model of the Pelamis converter. Amadou G. Thiam and Allan D. Pierce (Boston Univ., Boston, MA 02215, thiam@bu.edu)

The conversion of mechanical energy to another energy form is often built into a structural acoustics model by adding some damping mechanism to the model. Also, in ocean wave energy conversion, one damping mechanism, which is occasionally of great importance in the determination of the limiting upper efficiency of a device, is the re-radiation of wave energy back into the ocean. Present paper focuses on a simplified model of the Pelamis wave energy converter, with the model consisting of a modified Euler-Bernoulli beam oriented head-on to incoming ocean waves and with energy conversion accounted for by a damping term, where the additional bending moment is proportional to the time derivative of the beam curvature. The wave energy converted is ideally equal to the energy dissipated by this term. The beam is taken as having circular cross-section with a mass density that is one-half that of water. An incident wave causes a transverse force along the bottom of the beam so that it heaves up and down in a snake-like fashion, and this heaving causes ocean waves to be re-radiated back into the ocean, causing an additional apparent damping. Consideration is limited to a linear analysis and to a deep-water ocean.

11:35
3aSA8. Rapid step response with limited ringing and light damping. Joseph Vignola and John Judge (Dept. of Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE Washington, DC 20064, vignola@cua.edu)

The time domain step response of an underdamped simple harmonic oscil- lator can be characterized as having two parts. The first is a transition from the initial position where the structure is accelerated toward a target position by a step force that introduces energy to the system. The second is a ringdown process, involving decaying oscillations about the target position as energy is removed from the structure. This study reports how an array of substantially smaller subordinate oscillators attached to the primary structure can dramatically reduce the time constant of the ringdown (the settling time) without slowing the first phase (rise time) in the way added classical damping does. In addition to having a more rapid initial response, a lightly damped oscillator with such an array of attachments requires less energy to change states than a more heavily damped structure. The approach is based on the concept of apparent damping, where vibration energy is transferred into and trapped by the set of attachments. Numerical results show that the distribution of the individual masses, isolated natural frequencies, and quality factors of the attachments, referred to as a subordinate oscillator array, can be tailored to achieve the rapid step response.

11:50

Some of the most challenging vibration control problems involve the selection and placement of frequency-dependent damping devices, such as multilayered viscoelastic materials. In complex structures involving the simultaneous use of several passive damping devices, it is difficult to determine the effectiveness of each device directly from measurements. This question motivates the present analysis, which seeks to quantify the power being dissipated by the structure at selected locations for a given force distribution. The analysis begins with an expression for power flow to the structure from a specified force distribution. This expression involves the real part of the admittance matrix looking into the structure at a set of measurement points. By expanding this matrix as a sum over damping elements that mechanically connect measurement points, one can spatially map the power flow into the structure. While thermodynamics requires that this power flow to the structure be equal to the power dissipated by the structure, it does not require equality of the spatial distributions. In order to understand the spatial distribution of power dissipation, time windowing of the impulse responses is introduced. Examples are developed to explore the effects of the time window. [Work supported by ONR under Grant No. N000140810531.]
Session 3aSC

Speech Communication: Exploring the Relationship Between Cognitive Processes and Speech Perception

Amee Shah, Chair
Cleveland State Univ., Speech and Hearing, 2121 Euclid Ave., Cincinnati, OH 44115

Chair’s Introduction—8:00

Invited Papers

8:05

3aSC1. Perceptual expectations, attention, and speech recognition. Howard C. Nusbaum (Dept. of Psych., The Univ. of Chicago, 5848 S. Univ. Chicago, IL 60637, hcnusbaum@uchicago.edu)

Research has demonstrated substantial plasticity in the adult speech perceiver. When listeners receive feedback about the classification of a speech signal, this feedback can shape attention affecting phoneme perception. This demonstrates that it is possible to shape perceptual attention prospectively changing subsequent speech perception. These changes can also reduce the cognitive load of speech perception. The effect of expectations on speech perception has been reported for a wide range of studies from sinewave speech perception to phonetic categorization and effects of talker variability. Neural changes occurring during learning and the neural effects of expectations both appear to involve regions of motor cortex implicated in speech production. These neural changes suggest a model of speech perception similar to analysis-by-synthesis in which sensory processes are tuned by activity within the speech motor system. Although such sensory-motor interactions have been implicated in speech perception due to coarticulation and the encoding of speech into sound, the role of expectations in speech perception may extend beyond segmental perception. We consider evidence that the interpretation of prosodic signals also depends on perceptual expectations, suggesting that the role of expectations may be to constrain recognition given the many-to-many relationship that holds broadly in speech.

8:35

3aSC2. What do comparisons between younger and older adult listeners tell us about speech processing? Margaret K. Pichora-Fuller (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd. N, Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca)

The perception of speech, the recognition of words, and the understanding of spoken language involve the dynamic and interactive processing of cues provided by the incoming signal and information stored in memory. Even when accuracy is high, the relative contributions of bottom-up and top-down processes may explain variations in the speed and effort required when listening to speech. Listening is fast when the quality of the incoming signal is optimal, but it is slowed as signal quality is reduced. Likewise, listening can be speeded when expectations constrain the likely alternatives or when priming implicitly facilitates the recognition of the signal, whereas it can be slowed if the context is incongruent with the signal or if context is used to resolve ambiguities or repair misperceptions in a compensatory fashion. Within-subjects comparisons on off-line and on-line measures in different listening conditions, including simulations of auditory aging and hearing loss, are used to investigate how listening effort varies and how listening is speeded or slowed depending on signal-driven and knowledge-driven factors. Comparisons between younger and older participants are used to evaluate how long-standing reductions in auditory temporal processing and compensatory changes in brain organization may alter how signal-driven and knowledge-driven processes interact.

9:05

3aSC3. The talker-specific nature of spoken language processing. Lynne C. Nygaard (Dept. of Psych., Emory Univ., 36 Eagle Row, Atlanta, GA 30322, lnygaard@emory.edu)

During spoken communication, listeners must contend with a physical speech signal that carries information not only about the linguistic content of a talker’s utterance but also about a host of talker-specific attributes such as talker identity and emotion. Traditionally, the recovery of linguistic structure has been thought to be independent of the recovery of these indexical or surface characteristics of speech. Recent research, however, suggests that listeners remember the perceptual quality of spoken words, flexibly adapt to informative indexical variation in speech, and use this variation to guide their understanding of spoken words. This talk will review evidence for indexical specificity effects across a range of perceptual and memory tasks and will examine the linguistic, attentional, and cognitive constraints that may mediate the role of surface form in spoken language processing. Implications of these effects for accounts of linguistic representation and processing will be discussed. [Work supported by NIDCD.]
3aSC4. Interpreting speech in context. Mark Pitt (Dept. of Psych., Ohio State Univ., 1835 Neil, Ave., Columbus, OH, 43210, pitt.2 @osu.edu)

One of the challenges in understanding how humans recognize spoken language is to reconcile the robustness of verbal communication with the many forms of ambiguity and variability in the speech signal. How much of the discrepancy is due to our incomplete understanding of the acoustics of speech (i.e., what is the critical information) versus the recruitment of mental processes (e.g., memory) to aid recognition? My approach to these explanations has been to study the processing of words that have been degraded in controlled ways (e.g., digital manipulation) and by using conversation-style speech that contains naturally produced ambiguities. Recognition is measured in isolation and in sentences using a variety of tasks whose purpose is to measure the source, magnitude, and time course of ambiguity resolution. I will present an overview of this work in the context of a few experimental settings, highlighting methodological issues along the way. Together, the findings suggest that word recognition is a highly fluid process that depends on the rapid integration of multiple sources of information.

10:05—10:20 Break

3aSC5. Investigating whether ease of lexical processing affects listeners’ ratings of the strength of talkers’ foreign accent. Amee Shah (Dept. of Health Sci., Cleveland State Univ., 2121 Euclid Ave., MC 429, Cleveland, OH 44115) and Conor McLennan (Dept. of Psych., Cleveland State Univ., 2121 Euclid Ave., CB 175, Cleveland, OH 44115)

Many research studies in spoken word recognition have examined the role that different types of surface variability have on listeners’ perception of spoken words (e.g., influence of same versus different talker manipulations). In our ongoing research program, we are investigating the other direction of this relationship. Our initial two experiments investigated whether linguistic complexity affects listeners’ overt subjective impressions of the strength of talkers’ foreign accents. Ease of lexical processing was manipulated in two ways, namely, sentence context and long-term repetition priming. In the sentence context experiment, participants made accent ratings on the last word of each sentence presented to them; this word was either related or unrelated to the sentence context. In the long-term repetition priming experiment, participants made their accent ratings to isolated spoken words, which they either had (primed) or had not (unprimed) heard earlier in the experiment. Stronger accent ratings in the unrelated sentence context and unprimed conditions were predicted. Response time to provide their accent rating responses to sentence context and priming manipulations was also studied. Our results should lead to a greater understanding of the role that lexical processing plays in listeners’ perception of foreign accents and perhaps other types of surface variability.

10:50


Context plays a critical role in human speech processing. Rapid, efficient integration of context and input is accomplished by interactive processing: bottom-up input information and top-down context information work together to constrain the percept. One domain where this is most clear is the effects of word context on perception of speech sounds. Behavioral experiments and computational model simulations show that interactive processing confers both benefits (robust perception in noisy environments and tuning in response to changes in input patterns) and costs (errors and delays in perception when the input is inconsistent with the context). Context effects are also prevalent is the constraining effect of global communication context on activation of different meanings of ambiguous words (i.e., homophones): when only highly imageable meanings are consistent with the context (a word-to-picture matching task), concrete noun meanings (e.g., the tree-related meaning of bark) become more activate than less imageable meanings (e.g., the dog-related meaning of bark). These findings are consistent with an interactive graded constraint satisfaction view of speech perception in which bottom-up input and top-down context simultaneously constrain the final percept.