# Session 1aAA

## Architectural Acoustics and Education in Acoustics: Computer Modeling in Buildings and the Environment as an Education Tool

Norman H. Philipp, Cochair
*Geiler and Associates, LLC, 1840 E. 153rd Cir., Olathe, KS 66062*

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

**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**1aAA1. Using computer building modeling and auralization as teaching tools.** Robert C. Coffeen (School of Architecture, Design & Planning, Universtiy of Kansas, 1465 Jayhawk Blvd, Lawrence, KS 66045, coffeen@ku.edu)

Acoustic building modeling in computer programs is very useful in the understanding of room acoustics for venues of various types by architecture and architectural engineering students. Models provide calculation of reverberation time using the Sabine and similar equations as interior materials are changed. Ray tracing can be used to understand the effect of disturbing sound reflections from interior surface shapes and locations. Being able to create impulse responses in a model allows the estimation of reverberation time using Schroeder integration. And, transferring impulse responses to a measurement and analysis program allows determination of early decay time as well as T10, T20, T30 and other sound decay cutoff times. In addition, more advanced students can determine Sound Transmission Class STI, Strength G, Inter-aural Cross Correlation Coefficient IACC, and other acoustic parameters. But, one of the most useful items that can be produced by model impulse responses is auralization. This allows students to hear a simulation of room sound as reverberation time and other acoustic parameters are changed. Examples of using one of the several modeling and analysis programs will be presented.

**9:25**

**1aAA2. Simple interactive virtual auralizations as educational tools.** Christopher L. Barnobi (Stewart Acoustical Consultants, 7330 Chapel Hill Rd, Suite 101, Raleigh, NC 27607, chris@sacnc.com)

This presentation provides an overview and demonstration of some 'classic' acoustic phenomena using a computer simulation. The computer program provides a visual rendering of an environment with a source and receiver. By allowing the user to vary some of the parameters of the environment, a user can see and hear the differences in spaces by changing the surroundings. The goal is to highlight the well known environmental factors that impact sound such as volume in a room. A variety of parameters and environments will be explored.

**9:45**

**1aAA3. Education technology in architectural acoustics: A hands-on program for teaching.** Norman H. Philipp (School of Architecture, Design & Planning, University of Kansas, Lawrence, KS 66045, philipp.norman@gmail.com)

Through the implementation of educational technology a web-based educational tool is being developed to aide in the teaching of architectural acoustics to architecture students at the undergraduate and graduate level. As the first step, its scope has been limited to reverberation time in architectural acoustics. The overall objective is to provide a dynamic educational tool for both educators and students to improve their understanding and retention of the principles of architectural acoustics.

**10:05–10:25 Break**

**10:25**

**1aAA4. Finite difference simulation methods as an educational tool.** Jonathan Botts, Ning Xiang, and Todd Brooks (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th St., Greene Building, Troy, NY 12180, botts.jonathan@gmail.com)

Finite difference methods can be a valuable and unexpected tool in acoustics education. As a wave acoustic simulation method, it provides instructive time-domain visualizations particularly useful for illustrating broadband wave effects like diffraction and interference. Furthermore, the knowledge required to implement these simulations can be taught in just a few hours of instruction with or without calculus in contrast to several other wave acoustic methods. The exposition provides opportunities for discussion of basic numerical methods as well as the physics of wave and diffusion processes. We present example projects from students of mixed science, engineering, and music backgrounds. After only four hours of instruction, they were able to independently simulate various room geometries with impedance boundary conditions along with a variety of other acoustical systems. Beyond strictly educational value, this is a flexible and free tool that the modern acoustician can use for research, physics-based simulation, or creation of broadband virtual sound fields.

**10:45**

**1aAA5. Accuracy of acoustic simulations and the effects of material databases.** Ronald Sauro (NWAA Labs, Inc, 90 Tower blvd, Elma, WA 98541, audio_ron@msn.com)

A discussion of the effects of material databases on the accuracy of predictions emanating from acoustic simulation programs. This brings to light the lack of inherent accuracy in these programs because of the lack of accuracy in the measurement of absorption, scattering and diffusion parameters. Some of the predictions can be shown to be off as much as 300 to 500 per cent. We look at possible corrections in these measurements and how they can improve these predictions.

## *Contributed Papers*

**11:05**

**1aAA6. Parallelized finite difference time domain room acoustic simulation.** Cameron Fackler, Jonathan Botts, and Ning Xiang (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, 110 8th St, Greene Building, Troy, NY 12180, facklc@rpi.edu)

A parallelized room acoustics simulator based on the finite difference time domain (FDTD) method is developed, utilizing a Blue Gene/L supercomputer. Wave-based methods such as FDTD are desirable for use in room acoustics simulations since they account for effects such as diffraction and interference. However, such methods require large amounts of computational power and memory, especially when simulating large volumes or high frequencies. To utilize the power of modern computing systems and move toward large-scale simulations of realistic concert halls, a parallel FDTD implementation is written in the C++ programming language with the Message Passing Interface (MPI) library. The volume to be simulated is partitioned into blocks, which efficiently update shared interfaces between nearest neighbors using the Blue Gene architecture's point-to-point communication network. Several compact explicit FDTD schemes are compared using simulations of various spatial volumes, executed on varying numbers of processors. The use of a Blue Gene/L supercomputer demonstrates substantial speedup over an equivalent serial implementation.

**11:20**

**1aAA7. Acoustic simulation of the church of San Francesco della Vigna.** Braxton B. Boren (Music, New York University, New York, NY 10012, bbb259@nyu.edu) and Malcolm Longair (Cavendish Laboratory, University of Cambridge, Cambridge, Cambridgeshire, United Kingdom)

San Francesco della Vigna is the oldest church in Venice for which there is evidence that acoustic considerations were taken into account in the architectural design. Francesco Zorzi, a humanist scholar, recommended that the church have a flat wooden coffered ceiling to improve the intelligibility of the sermons preached there. But instead of Zorzi's recommended flat ceiling, the church was built with a plaster vault ceiling. Using measured acoustic data from the CAMERA project, a virtual model of the church was constructed in Odeon whose simulated parameters matched the measured values at different source-receiver combinations. After obtaining a good match to the measured values, this virtual model was then altered to reconstruct the flat ceiling recommended by Zorzi. This ceiling was then placed at the two different heights at which it might have been built in Zorzi's time. The simulations show that the more absorptive ceiling might have slightly reduced the long reverberation time in the church. However, the ceiling would have been too high to make any significant change in the D50, which still remains extremely low. Thus this simulation indicates that Zorzi's ceiling would not have made the impact on speech intelligibility he had expected.

# Session 1aAO

## Acoustical Oceanography and Underwater Acoustics: Memorial Session in Honor of Clarence S. Clay I

Dezhang Chu, Cochair
*NOAA Fisheries, NWFSC, Seattle, WA 98112*

John K. Horne, Cochair
*School of Aquatic and Fishery Sciences, University of Washington, Seattle, WA 98195*

J. Michael Jech, Cochair
*Northeast Fisheries Science Center, Woods Hole, MA 02543*

Timothy K. Stanton, Cochair
*Woods Hole Oceanographic Institution, Woods Hole, MA 02543-1053*

**Chair's Introduction—8:55**

### *Invited Papers*

**9:00**

**1aAO1. C.S.Clay: A distinguished acoustician.** Ivan Tolstoy (Knocktower, Knockvennie, Castle Douglas DG7 3PA, United Kingdom, tstanton@whoi.edu)

Clay's well-known work in ocean acoustics earned him an international reputation. His contributions ranged from studies of sound propagation in shallow water to the application of filter theory in noisy environments, diffraction from rough surfaces, sound scatter by fish and, generally speaking, the execution and design of numerous experiments. Less well known, perhaps on account of his modesty, was his role at Columbia University's Hudson labs in designing a fully digitized microphone array for the study of low frequency atmospheric waves (in the 1 to 600 sec. period band), which led to the detection of gravity and acoustic-gravity waves from several nuclear tests. After leaving Columbia, Clay accepted a professorship at the University of Wisconsin where, among other things, he taught geophysics and continued research on sound scatter. His presence and participation at Acoustical Society meetings will be sorely missed.

**9:20**

**1aAO2. C.S. Clay—A scientist of outstanding vision, brilliance, and versatility.** Christopher Feuillade (Facultad de Física, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago, Región Metropolitana, Chile, chris.feuillade@gmail.com)

During a scientific career of more that 60 years, Clarence Clay made many important research contributions spanning a wide range of topics in ocean acoustics, acoustical oceanography, geophysical exploration, signal processing, SONAR system applications and techniques, and more. His achievements are detailed in over 135 peer-reviewed articles, abstracts, technical reports, and at least five patents. In addition he was the author, or co-author, of four widely read textbooks. His research work is recognized internationally for the far-reaching significance of many of the advances he made. In order to attempt a proper appreciation of the range and versatility of Clay's accomplishments, within the constraints of a 30 minute presentation, this talk will consist of a series of brief overviews of a representative selection of topics on which he worked. These include: the theory and measurement of rough surface scattering, particularly at the ocean boundaries; matched-filter signal transmission and detection, time reversal, and matched-field processing; acoustic methods and models for scattering from individual fish, and for fish detection and abundance estimation; the theory and experimental investigation of time domain scattering from wedges, and its incorporation into wedge assemblage models; and the theory and applications of seismic exploration and profiling.

**9:50**

**1aAO3. A review of Clarence Clay's research contributions in the area of fisheries acoustics.** John Ehrenberg (Hydroacoustic Technology Inc, 715 NE Northlake Way, Seattle, WA 98105, jehrenberg@htisonar.com)

This presentation provides an overview of the significant contribution that Clay and his students made to the understanding of the statistical nature of the acoustic signals scattered from fish and the methods for removing the effect of the beam pattern from the received echo statistics to measure the underlying fish backscattering statistics. By making measurements of the acoustic scattering from live fish, he showed that the probability density function of the envelope amplitude of the echo signal scattered from the fish could be modeled by a Ricean PDF. He further showed that as the ratio of the length of the fish to the acoustic wavelength became large, the PDF became Rayleigh distributed. Clay and his students were interested in using the measured echo statistics to obtain information about the size distribution of the fish producing the scattering. They developed a method for deconvolving the effect of the acoustic beam pattern from the received echo statistics to provide an estimate of the fish scattering PDF. The effectiveness of the technique was demonstrated for a fish population in a Wisconsin lake.

**10:10–10:25 Break**

## *Contributed Papers*

**10:25**

**1aAO4. Estimating numerical density of scatterers in monotype aggregations using the statistics of broadband echoes: Applications to fish echoes.** Wu-Jung Lee, Timothy K. Stanton, and Andone C. Lavery (Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543, wjlee@whoi.edu)

The statistics of echoes from active sonar systems can yield important information on aggregations of scatterers. This study explores the use of echo statistics for estimating the numerical density of scatterers in monotype aggregations. Here, "monotype" refers to scatterers with the same scattering amplitude distribution in the considered frequency range. The signals are broadband, and the geometry involves direct paths between the sonar and the scatterers without interference from boundaries. Model probability density functions (pdf's) of envelope amplitudes of matched-filter outputs are numerically generated by varying the number of Rayleigh scatterers randomly-distributed in a half-space shell while accounting for the frequency-dependent system response, scatterer response, and beampattern effects. The shape of the echo pdf as observed by the sonar receiver is highly non-Rayleigh when there are few scatterers in the beam, and gradually approaches the Rayleigh distribution when the number of scatterers increases. This model is applied to broadband fish echoes (30-70 kHz) collected in the ocean through a best-fit procedure. The inferred numerical density of fish is comparable to the density estimated using corresponding measurements of volume backscattering strength and modeled target strengths.

**10:40**

**1aAO5. Ice cream and the application of backscatter models.** John K. Horne (School of Aquatic and Fishery Sciences, University of Washington, Box 355020, Seattle, WA 98195, jhorne@u.washington.edu)

I was invited to visit Clay and colleagues at the Center for Limnology at the University of Wisconsin, Madison in October 1991. As an acoustics neophyte, I had lots of questions that Clay patiently took the time to answer while we ate ice cream at the Memorial Union. That discussion led to the development of the Kirchhoff Ray-mode (KRM) model and increased the use of acoustic scattering models to investigate how fish reflect sound. Acoustic scattering models enable investigation of factors or conditions that cannot be replicated or isolated in field or experimental measures. The iterative combination of models with measures improves accuracy of model predictions and the understanding of how the physics of sound interacts with biology to produce acoustic data. Both the structure and application of backscatter models have evolved in their complexity and realism. Examples will be used to illustrate advances in and insight gained through modeling, with special consideration of Clay's contributions. The talk will conclude with speculation on what Clay would see as the next step.

**10:55**

**1aAO6. Low frequency acoustical scattering properties of large schools of swim bladder fish.** María P. Raveau (Facultad de Ingeniería, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Región Metropolitana, Chile) and Christopher Feuillade (Facultad de Física, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago, Chile, chris.feuillade@gmail.com)

The collective back scattering behavior of fish schools has previously been described by a school scattering model [J. Acoust. Soc. Am., **99**(1), 196-208 (1996)], which incorporates both multiple scattering effects between neighboring fish, and coherent interactions of their individual scattered fields. In the present work, the school scattering model has been extended, and used to investigate the back- and forward-scattering properties of the acoustic field, and transmission through, large schools of swim bladder fish, at frequencies close to the swim bladder resonance frequency. Results show that their frequency and spatially-varying scattering behavior depends strongly upon the number of fish in the school ensemble, the species specific swim bladder size, the average spacing between fish, and the size and shape of the school. Results will also be presented of a comparison between the school model and fish absorption data obtained during the experiment Modal Lion, performed in the Gulf of Lion in September 1995, and reported by Diachok [J. Acoust. Soc. Am., **105**(4), 2107-2128 (1999)]. [Work supported by ONR.]

**11:10**

**1aAO7. Acoustic characterization of thecosome pteropods and recent field measurements in the context of ocean acidification.** Andone C. Lavery (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02543, alavery@whoi.edu), Gareth L. Lawson, Peter H. Wiebe (Biology, Woods Hole Oceanographic Institution, Woods Hole, MA), Timothy K. Stanton, Jonathan R. Fincke (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA), and Nancy J. Copley (Biology, Woods Hole Oceanographic Institution, Woods Hole, MA)

One of Clay's passions was modeling the scattering physics of marine organisms, a passion that has transcended into new generations of scientists. The focus of this presentation is thecosome pteropods, a widely and patchily distributed group of shelled zooplankton that are important members of pelagic ecosystems as they constitute important prey for a variety of other zooplankton and top predators. Acoustic techniques are well suited to sampling pteropods on relevant spatial and temporal scales as they secrete aragonite shells that make them highly efficient scatterers of sound. However, pteropod shells are complex and very susceptible to an increasingly corrosive seawater environment due to ocean acidification. Understanding the scattering physics is key to using acoustics as a quantitative remote sensing tool. Here we report on recent field measurements that combine the use of broadband (30-600 kHz) and narrowband (43, 120, 200, and 420 kHz) acoustic scattering techniques, as well as supporting in situ measurements (nets, optics, CTD and ocean chemistry) to investigate the distribution, abundance and size of pteropods in both the northwest Atlantic and the northeast Pacific in relation to the oceanic chemistry. Existing scattering models are tested, and improvements and modifications to the acoustic instrumentation and models are suggested.

**11:25**

**1aAO8. Echo statistics: Pursuing one of Clay's visions.** Timothy K. Stanton (Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, MS #11, Woods Hole, MA 02543-1053, tstanton@whoi.edu) and Dezhang Chu (National Marine Fisheries Service - Northeast, National Oceanic and Atmospheric Administration, Seattle, WA)

One of the first papers that Clay gave to one of the authors (TKS) when he entered Clay's office in 1980 was one of Clay's papers involving echo statistics and, specifically, accounting for beampattern effects in using single beam echoes from resolved fish to estimate their target strength and abundance. That, and a multitude of conversations with Clay helped propel TKS, and later DC, into a career where echo statistics was an integral aspect of their work. We will review our work on echo statistics associated with a variety of scatterers–the seafloor, sea ice, fish, zooplankton, and machined objects. A key aspect of this work has been connecting the physics of the scattering process and sonar parameters with parameters of the statistical functions (such as shape parameter).

**11:40–12:00 Panel Discussion**

# Session 1aEA

## Engineering Acoustics, Signal Processing in Acoustics, and Animal Bioacoustics: Broadband, Complex Pulses for Echolocation

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

**Chair's Introduction—8:25**

### *Invited Papers*

**8:30**

**1aEA1. Broadband synthetic aperture chirp reflection profiling.** Steven Schock (Ocean and Mechanical Engineering, Florida Atlantic University, 777 Glades Road, Boca Raton, FL 33431, sschock@fau.edu), Jason Sara (Edgetech, Boca Raton, FL), and Kenneth M. Walsh (K M Engineering Ltd., Middletown, RI)

A newly developed towed chirp subbottom profiler transmits FM pulses with a bandwidth of three octaves to generate high resolution reflection profiles of the seabed. The broad bandwidth of the pulses, generated with two arrays of piston sources, produces the temporal resolution needed for resolving fine sediment layering. A 40 channel horizontal hydrophone array, embedded in vehicle wings, provides the acoustic aperture for enhancing the across track resolution of subsurface features and reducing sediment scattering noise. After application of a matched filter, synthetic aperture processing of hydrophone data generates a large aperture along the track of the sonar vehicle thereby improving along track image resolution and obtaining a further reduction in scattering noise. The reductions in backscattering from sediments yield imagery with improved subsurface penetration. The reflection profiles are stacked envelopes of coherently summed data formed by time domain focusing on planar surfaces oriented over a range of discrete slopes. This work was funded by the National Science Foundation.

**8:55**

**1aEA2. Odontocete biosonar signals: Functional anatomy or signal design.** Whitlow W. Au (HI Institute of Marine Biology, University of Hawaii, 46-007 Lilipuna Road, Kaneohe, HI 96744, wau@hawaii.edu)

There are between 67 and 76 species of odontocetes (toothed whales) and presumably all have biosonar capabilites. There are three fundamental biosonar signal types that can be categorized by types of marine mammals that produce these signals. Whales and dolphins that can emit whistle signals (except for sperm whales) project short broadband clicks containing about 5 to 7 cycles with decaying exponential envelope and Q (center frequency over bandwidth) between 2 and 3. Porpoises do not whistle and produce polycyclic narrow band high frequency biosonar signals with approximately 20 or more cycles with a modified sinusoidal amplitude envelope and Q around 14. Biosonar waveforms of beaked whales (also non-whistling animals) typically have 10-15 cycles with a linear FM component and Q around 4. This presentation will discuss the characteristics of the three different biosonar signal types and suggest some motivation factors involved with the use of the different signals. The type of prey and their habitat will also be included in the discussion. It will be shown that in some cases, the signal type is motivated by anatomical constraints of the odontocete and in other cases, the backscatter characteristics of the prey may be the most important factor.

**9:20**

**1aEA3. Implications of the variety of bat echolocation sounds for understanding biosonar processing.** James A. Simmons (Neuroscience, Brown University, 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu), Matthias Hoffmann-Kuhnt, Tzi Ming Leong (National University of Singapore, Singapore), Shizuko Hiryu, Hiroshi Riquimaroux (Doshisha University, Kyotanabe, Kyoto, Japan), Jeffrey M. Knowles (Neuroscience, Brown University, Providence, RI), and Cynthia F. Moss (University of Maryland, College Park, MD)

The variety of echolocation sounds used by different species of bats have implications for target ranging. Signals recorded at individual sites reveal species stacked in different frequency bands, perhaps to avoid cross-interference. Search-stage signals include short single-harmonic or multi-harmonic tone-bursts, or very shallow FM bursts. These narrowband sounds have abrupt onsets to evoke phasic on-responses that register echo delay, but with limited acuity. Wider FM sweeps used for searching by other bats evoke on-responses at many more frequencies for better delay acuity. These sound types may signify foraging in the open, within broad spaces bounded relatively remotely by trees or the ground. Intervals between broadcasts are consistent with biosonar operating ranges set by the boundaries of the scene in relation to atmospheric attenuation. Most species make transitions to wider signal bandwidth during interception by increasing FM sweep-width or adding harmonics. Additionally, wideband, multi-harmonic FM sounds are used by species that frequently fly vegetation; they use harmonic processing to suppress surrounding clutter and perceive the path to the front. These observations suggest basic echo-delay processing to determine target range, with increasing bandwidth first to improve delay acuity and then to determine target shape. [Work supported by ONR and NSF.]

9:45

**1aEA4. Dolphins use "packets" of broadband clicks during long range echolocation tasks.** James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

When echolocating, dolphins typically emit a single short duration, high-frequency, broadband "click," then wait for the echo to return before emitting another click. However, previous studies have shown that dolphins and belugas performing long-range echolocation tasks may instead emit a burst, or "packet," of several clicks, then wait for the packet of echoes to return before emitting another packet of clicks. The exact reasons for the use of packets, rather than individual clicks, is unknown. In this study, the use of packets by dolphins was examined by having trained bottlenose dolphins perform long-range echolocation tasks. The tasks featured the use of "phantom" echoes produced by capturing the dolphin's outgoing echolocation clicks, convolving the clicks with the impulse response of a physical target to create an echo waveform, then broadcasting the delayed, scaled echo waveform back to the dolphin. Dolphins were trained to report the presence of phantom echoes or a change in phantom echoes. At ranges below 75 m, the dolphins rarely used packets of clicks. For ranges greater than 75 m, the likelihood of packet use was related to both target range and echo strength. [Work supported by the SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

**10:10–10:25 Break**

10:25

**1aEA5. Environmentally neutral complex broadband biomimetic waveforms for active sonar.** Peter F. Dobbins (Advanced Systems, Ultra Electronics Sonar Systems, Waverley House, Hampshire Road, Weymouth, Dorset DT4 9XD, United Kingdom, peter.dobbins@ultra-sonar.com)

There is an expanding requirement to reduce the impact of man-made sound, including active sonar transmissions, on marine mammals in the defence, offshore and other sectors. This is driven partly by increased public interest in these animals, but mainly by legislation such as the US Marine Mammal Protection Act, and similar regulatory and licensing requirements throughout the world. Typically, such requirements are met using monitoring by Marine Mammal Observers or Passive Acoustic Monitoring. Having detected animals within a specified range, some form of mitigating action such as shutting down the sound source is then necessary. However, in general it is difficult to ensure the absence of marine mammals before transmitting, so it is desirable to look for forms of sonar waveform that are potentially less harmful to marine life. One way this might be achieved is to use signals derived from natural sounds such as the vocalisations of the animals themselves. It might be expected that such sounds would appear more familiar, thus reducing possible abnormal behavioural impacts. This paper reviews the use of such waveforms and presents a preliminary estimate of their performance in practical sonar systems, along with an assessment of the potential impacts on marine life.

10:50

**1aEA6. Measuring the covertness of broadband sonar waveforms.** Joonho D. Park and John F. Doherty (Electrical Engineering, Pennsylvania State University, State College, PA 16802, jdp971@psu.edu)

In underwater environment, the platform uses active sonar to estimate the range to the target, covertly. The target employs detectors designed to find anomalies in its ambient environment, including man-made waveforms such as ones transmitted by the platform. However, the structure of the waveform is unknown. In this scenario, we try to measure the covertness of the waveforms transmitted by the platform in various scenarios using a quantity related to relative entropy, and the performance of range estimation using the covert waveform. At the target, we measure the maximum probability of detection of the sonar waveform, for a specified false alarm rate. At the platform, we measure the probability of estimating the range to the target correctly. The performances of both processes are dependent on the amount of information one has about the ambient environment, the structure of the transmitted waveform, and the actual range between the platform and the target. We show how the performances are dependent on the accuracy of the knowledge about these elements.

11:15

**1aEA7. Considerations in designing piezocomposite transducers and arrays for broadband sonar systems.** Barry Doust, Tim Mudarri, Connie Ursch, Joe Aghia, and Brian Pazol (Electroacoustics, Materials Systems Inc., 543 Great Rd., Littleton, MA 01460, bdoust@matsysinc.com)

The term broadband is commonly used to refer to the capability of a sonar system. Recent advances in signal processing and system electronics have re-defined capabilities for these systems and introduced new complex broadband pulse requirements for the sonar transducer. Advances in 1-3 piezocomposite technology have addressed this need through optimization of materials and innovative electroacoustic designs to provide high performance broadband solutions. Taking full advantage of this transducer technology requires careful consideration of the total system performance including dynamic range of receiver electronics, available transmit voltage/current/power, directivity and array configuration. This generalized study considers the design options available for optimizing transducers and arrays for broadband operation. Several Langevin or sandwich style Piezocomposite transducer configurations will be presented for both monostatic (two-way) and bistatic (separate receive and transmit) transducer systems and the trades in terms of total system performance. The study includes comparison of design optimization methodologies based on peak frequency response, impedance phase center or maximum transmit power factor, and combined two way system response. Emphasis will be on conventional piezoceramic materials with some discussion of second generation single crystal materials and their potential in future systems.

### *Contributed Paper*



11:40

**1aEA8. A generalized sinusoidal frequency modulated waveform for active sonar.** David A. Hague and John R. Buck (Electrical and Computer Engineering, University of Massachusetts Dartmouth, North Dartmouth, MA 02747, david.a.hague@gmail.com)

Pulse-Compression or Frequency Modulated (FM) active sonar waveforms provide a significant improvement in range resolution and reverberation suppression over Continuous Wave (CW) waveforms. The Sinusoidal FM (SFM) waveform modulates its instantaneous frequency (IF) by a sinusoid to achieve high Doppler sensitivity while maintaining desirable reverberation suppression. This allows the SFM waveform to resolve target velocities much better than the Doppler tolerant Hyperbolic FM waveform. However, the SFM suffers from poor range resolution as the Auto-Correlation Function (ACF) contains many ambiguous peaks generated by the periodicity of the SFM's IF. The periodic sidelobes in the ACF for the SFM signal are similar to those exhibited by periodic CW waveforms, which motivated the development of FM waveforms to improve range resolution. This suggests that modifying the SFM waveform to use an aperiodic modulating function should improve range resolution while preserving Doppler sensitivity. This talk presents an active sonar waveform where the IF function is itself an FM chirp waveform, and for which the SFM is a special case. This generalized sinusoidal FM waveform resolves target range and velocity with reverberation suppression comparable to other well-established FM waveforms. [Work supported by ONR and the SMART Program.]

MONDAY MORNING, 22 OCTOBER 2012 — ANDY KIRK A/B, 10:00 A.M. TO 12:00 NOON

## Session 1aMU

## Musical Acoustics: General Topics in Musical Acoustics

Thomas R. Moore, Chair
*Department of Physics, Rollins College, Winter Park, FL 32789*

### *Contributed Papers*

10:00

**1aMU1. Choir hearing responses: Rehearsal versus performance configurations.** Glenn E. Sweitzer (Sweitzer LLP, 4504 N Hereford Dr, Muncie, IN 47304, glenn.sweitzer@gmail.com)

Choir member responses to hearing (sung parts) in a rehearsal room are compared with those for its associated performance stage. Anonymous scaled responses from each choir member are gathered simultaneously using a personal response system. The protocol is repeated in a rehearsal room for the choir voices configured by 1) part versus mixed; and 2) on risers versus flat floor. On the performance stage, the protocol is repeated for same. Prior to each set of responses, the choir sings a prayer familiar to the choir members. The responses vary widely between rehearsal and performance venues, and by configuration in each. These findings suggest that choir member response may be largely ignored in the design and operation of choir rehearsal and performance facilities. Potentials for improving choir member hearing in existing rehearsal and performance venues is discussed.

10:15

**1aMU2. The origins of longitudinal waves in piano strings.** Brandon August, Nikki Etchenique, and Thomas R. Moore (Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

The importance of longitudinal waves in piano strings has been previously identified by several investigators. Recent experimental work has provided insight into the origin of these waves and their relationship to the transverse string motion. These measurements indicate that there are multiple regimes in which longitudinal waves are created through different processes.

10:30

**1aMU3. Automatic transcription of monophonic piano music.** Fatemeh Pishdadian and Jill K. Nelson (Electrical and Computer Engineering, George Mason University, 4400 University Drive, ECE Department, Volgenau School of Engineering, Fairfax, VA 22030-4444, fpishdad@masonlive.gmu.edu)

Automatic music transcription refers to the process of transforming an acoustic musical signal into a written symbolic representation, e.g. a score. This process consists of extracting the parameters of note events, for instance pitches, onset times, and durations, from a raw acoustic signal. We have developed a novel algorithm for transcription of monophonic piano music, which addresses the challenges of pitch and note sequence detection in two stages: (1) The K-Nearest Neighbor (KNN) classification method is employed to identify K pitch candidates, based on spectral information, for each note event. (2) The most likely note sequence is determined by running a best-first tree search over the note candidates based on both spectral information and note transition probability distributions. The proposed two-step approach provides the performance gain achieved by incorporating note transition probabilities while maintaining significantly lower computational complexity than existing support vector machine and hidden Markov model methods. The algorithm was evaluated on a database comprised of excerpts from Bach's inventions. By performing a low complexity tree search based on note transition information, we achieve approximately 10% improvement over using only spectral information, correctly classifying roughly 85% of the notes in the database.

10:45

**1aMU4. Multiple-timbre fundamental frequency tracking using an instrument spectrum library.** Mert Bay and James W. Beauchamp (Electrical & Computer Engineering, University of Illinois at Urbana-Champaign, Champaign, IL 61820, mertbay@illinois.edu)

Recently many researchers have attempted automatic pitch estimation of polyphonic music (e.g., Li et al., IEEE Trans ASLP, 2009). Most of these attempts have concerned themselves with the estimation of individual pitches (F0s) while not associating the estimated pitches with the particular instruments that produce them. Estimating pitches for each instrument will lead to full music transcription. Individual instrument F0 tracks can be used in music information retrieval systems to better organize and search music. We propose a method to estimate the F0 tracks for a set of harmonic instruments in a sound mixture, using probabilistic latent component analysis (PLCA) and collections of basis spectra indexed by F0 and instrument learned in advance. The PLCA model is extended hierarchically to explain the observed input mixture spectra as a sum of basis spectra from note(s) of various instruments. The polyphonic pitch tracking problem is posed as inferring the most likely combination of the active note(s) from different instruments. Continuity and sparsity constraints are enforced to better model how the music is produced. The method was trained on a common instrument spectrum library an evaluated using an established polyphonic audio dataset.

11:00

**1aMU5. Absolute pitch is associated with a large auditory digit span: A clue to its genesis.** Diana Deutsch and Kevin Dooley (Department of Psychology, University of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0109, ddeutsch@ucsd.edu)

Absolute pitch (AP) is very rare in North America and Europe, and its genesis is unclear. Its prevalence is far higher among tone language speakers, and among those with early onset of musical training. However, most nontone language speakers with early and extensive musical training do not possess AP. To test the hypothesis that an unusually large auditory memory is involved in the genesis of AP, at least in nontone language speakers, we recruited 7 AP possessors, and 20 AP nonpossessors. All subjects were primary speakers of English, had begun musical training at $\leq$ age 6, and were UCSD students or recent graduates. The two groups were matched for age and years of musical experience. All subjects were administered an auditory digit span test, followed by a visual digit span test, with digits presented 1/sec. While the average auditory digit span was 8.1 digits for the AP nonpossessors, it was 10.0 digits for the AP possessors. This difference between the two groups was highly significant ($p = 0.0015$, 1-tailed). The AP possessors also marginally outperformed the nonpossessors on the visual digit span test; however this difference was nonsignificant. These new findings provide a clue to a genetic component of AP.

11:15

**1aMU6. Perception of musical and lexical tones by musicians and nonmusicians.** Chao-Yang Lee and Allison Lekich (Communication Sciences and Disorders, Ohio University, Grover W225, Athens, OH 45701, leec1@ohio.edu)

This study explores the relationship between musical and linguistic pitch perception. We asked whether the ability to identify musical tones is associated with the ability to identify lexical tones. English-speaking musicians and nonmusicians were asked to identify Taiwanese level tones produced by multiple speakers. Because pitch range varies across speakers and the tones were produced in isolation, participants had to estimate relative pitch height without cues typically available for speaker normalization. The musician participants were also asked to identify synthesized musical tones without a reference pitch. The results showed that both musicians and nonmusicians were able to identify Taiwanese tones above chance, but only for tones in the extremes of the speakers' overall pitch range. Preliminary data from the musicians show that musical tone identification accuracy was low and not associated with accuracy in the Taiwanese tone task. Implications of these findings for the music-speech relationship are discussed.

11:30

**1aMU7. Measurement and analysis of timing asynchronies in ensemble performance.** Gang Ren, Stephen Roessner, Samarth Hosakere Shivaswamy (Dept. of Electrical and Computer Engineering, Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu), Dave Headlam, and Mark Bocko (Dept. of Electrical and Computer Engineering; Dept. of Music Theory, Eastman Sch. of Music, Univ. of Rochester, Rochester, NY)

Timing asynchrony is an important timing descriptor of ensemble music performance. In this paper the timing asynchrony is measured as offsets between "concurring" music onsets. Specifically we measure the offset between music note onsets that are prescribed to be synchronized according to the music score. First, we conduct measurements on the multi-track audio with separate instrument tracks. These multi-track materials are recorded by using acoustically isolated recording booth or by conducting multiple recording rounds. We then perform statistical analysis both on individual asynchrony points and on asynchrony points at key coordination locations, such as the start and the end of a music phrase. We emphasize in the analysis that the musical timing asynchronies should not be treated only as performance discrepancies because part of these micro-deviation patterns provide artistic "lively" elements. We also generalize the proposed framework to mixed-down polyphonic recordings. For polyphonic recordings where clear onsets can be identified we perform similar measurement and analysis algorithms as for separated multi-track recordings. For mixed-down polyphonic recordings where a clear separation of instrument tracks is not possible, we use onset dispersions, which measure the offset range of sonic partials onsets in an coordination points, as an alternative timing asynchrony descriptor.

11:45

**1aMU8. The implementation of pyschoacoustical signal parameters in the wavelet domain.** Matt Borland and Stephen Birkett (SYDE, University of Waterloo, 3-98 John St. W., Waterloo, ON N2L1C1, Canada, mjborlan@uwaterloo.ca)

Introducing Wavelet techniques into the psychoacoustical analysis of sound signals provides a powerful alternative to standard Fourier methods. In this paper the reformulation of existing psychoacoustical signal parameters using wavelet methods will be explored. A major motivation for this work is that traditional psychoacoustical signal analysis relies heavily on the Fourier transform to provide a frequency content representation of a time signal, but this frequency domain representation is not always accurate; especially for sounds with impactive components. These impactive events can have a more significant contribution to the calculation of psychoacoustical signal parameters by reformulating existing psychoacoustic parameters in the Wavelet domain that are dependent on the Fourier transform. To provide concrete examples of the results simulated "plucked string" sounds are analyzed with analogous Fourier and Wavelet domain signal parameters to demonstrate the difference in performance achieved using Wavelet methods for sounds which have impactive components.

## Session 1aNS

## Noise, Architectural Acoustics and Physical Acoustics: Sound Absorption of Micro-Perforated Structures

Li Cheng, Cochair
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Ning Xiang, Cochair
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**Chair's Introduction—8:55**

### *Invited Papers*

**9:00**

**1aNS1. On the use of micro-perforates for machinery and vehicle noise control.** Mats Åbom and Sabry Allam (The Marcus Wallenberg Laboratory, KTH-The Royal Inst of Technology, Stockholm 10044, Sweden, matsabom@kth.se)

A micro-perforated plate (MPP) is a perforated plate with holes typically in the sub-millimeter range and perforation ratio around 1%. The values are typical for applications in air at standard temperature and pressure (STP). The underlying acoustic principle is simple: To create a surface with a built in damping that effectively dissipates sound waves. To achieve this, the specific acoustic impedance of a MPP is normally tuned to be of the order of the characteristic wave impedance in the medium (400 Pa*s/m in air at STP). The traditional application for MPP's has been building acoustics, normally in the form of a so called panel absorber to create an absorption peak at a selected frequency. However, MPP's made of metal are also well suited for machinery and vehicle noise control. For instance MPP's have the potential to be used instead of porous materials in dissipative mufflers, which not only can save weight but also offer a non-fibrous alternative. Furthermore, since MPP's have a large steady flow resistance they can be used as acoustically absorbing guide vanes at duct bends or as a fan housing. One important issue for these applications is the effect of flow on the MPP impedance. This issue plus a number of applications related to vehicle noise, have been studied at KTH during the last decade and this paper aims at summarizing the main results.

**9:20**

**1aNS2. The effect of flexibility on the acoustical performance of microperforated materials.** J. S. Bolton (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, 140 S. Martin Jischke Drive, West Lafayette, IN 47907-2031, bolton@purdue.edu)

In conventional models of microperforated materials, the solid layer in which the holes are formed is usually considered to be rigid. However, microperforated materials are often thin, say less than 0.5 mm in thickness, and are sometimes made of lightweight polymeric materials. Experimental measurements suggest that when the mass per unit area of a microperforated material is less than approximately 0.5 kg per square meter, the motion of the solid layer becomes important. The solid layer is driven into motion both by the incident pressure acting on its surface and by viscous forces generated within the perforations. The ability of a microperforate to dissipate acoustical energy depends on there being relative motion between the air in the perforations and the solid layer: motion of the solid may either help or hurt this effect, particularly when the solid layer is supported on a grid-like structure, since the individual segments of the microperforate then exhibit modal behavior. In this presentation, models of this behavior will be described, and examples will be given in which essentially membrane-like behavior is modified by the presence of the perforations, and conversely, in which essentially rigid microperforated layer behavior is modified by vibration of the solid layer.

**9:40**

**1aNS3. Micro-perforated elements in complex vibro-acoustic environment: Modelling and applications.** Xiang Yu (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong SAR, China), Laurent Maxit, Jean-Louis Guyader (Laboratoire Vibrations Acoustique, Institut National des Sciences Appliquées (INSA) de Lyon, Lyon, France), and Li Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, FG611, Hung Hom, Kowloon, Hong Kong SAR 999077, China, mmlcheng@inet.polyu.edu.hk)

Micro-perforated structures/panels (MPP) are widely used in various architectural, industrial and environmental applications for providing efficient sound absorptions. More recently, they found their use in compact mechanical systems, in which the property of the MPP is shown to be strongly influenced by the surrounding vibro-acoustic environment, which is drastically different from the one dimensional Kundt tube configuration, usually used in existing works. Unfortunately, very little has been done in this regard, due to the fact that modeling such a vibro-coustic system with MPPs as integrative elements is a very challenging task, not even to mention the optimization. Recently, a vibro-acoustic formulation based on the Patch Transfer Function (PTF) approach was proposed to model micro-perforated structures in a complex vibro-acoustic environment. As a sub-structuring approach, PTF allows assembling different vibro-acoustic subsystems, including micro-perforations and the flexibility of a MPP, through coupled surfaces. The proposed

formulation provides explicit representation of the coupling among subsystems, with enhanced capability of handling system complexities and facilitating system optimization. In this talk, the overall approach will be reviewed and applied to a number of typical examples. The versatility and efficiency of the method as well as the underlying physics are demonstrated.

**10:00–10:30 Break**

**10:30**

**1aNS4. A straightforward method toward efficient and precise impedance measurement for microperforation panels under flow conditions.** Xiaodong Jing (School of Jet Propulsion, Beijing University of Aeronautics and Astronautics, No. 37, Xueyuan Road, Beijing 100191, China, jingxd@buaa.edu.cn)

This paper addresses the problem how to measure acoustic impedance of microperforation panels (MPPs) under flow conditions. Since 1960s, many different methods have been proposed to tackle this problem, mainly motivated by the aim of reducing noise emission from aeroengines, ventilators and other fluid machines. It has been found that the presence of flow favorably enhances the damping of MPPs due to the mechanism of sound-vortex interaction. This, however, leads to flow-dependent acoustic impedance whose determination is rather difficult. Despite considerable efforts over the past decades, there is still stringent need for developing efficient and precise impedance measurement method under flow conditions in order to fully explore the potentials of MPPs. Towards this goal, a straightforward method has been put forward to measure the acoustic impedance of an MPP lined in a flow duct (JASA, 124(1), 227-234). The basic principle is that the dominate axial wavenumber is extracted from the measured wall sound pressure by means of Prony method, thereby the unknown acoustic impedance is algebraically solved from the dispersion equation. In this paper, the straightforward method is further extended to incorporate the effects of flow boundary layer and higher-order acoustic modes that are essentially important for practical applications.

**10:50**

**1aNS5. Hybrid silencer by using micro-perforated plate with side-branch cavities.** Xiaonan Wang, Yatsze Choy, and Li Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong 852, mmyschoy@polyu.edu.hk)

A plate silencer consists of an expansion chamber with two side-branch cavities covered by light but extremely stiff plates. It works effectively with wide stopband from low-to-medium frequencies only if the plate is extremely stiff, to ensure a strong reflection of acoustic wave to the upstream in the duct. However, a plate with slightly weak bending stiffness will result in non-uniform transmission loss (TL) spectra with narrowed stopband. In this study, a hybrid silencer is proposed by introducing micro-perforations into the plate to elicit the sound absorption in order to compensate for the deficiency in the passband caused by the insufficient sound reflection in certain frequency range due to plate with weaker stiffness. A theoretical model, capable of dealing with the strong coupling between the vibrating micro-perforated plate and sound fields inside the cavity and the duct, is developed. Through a proper balancing between the sound absorption and reflection, the proposed hybrid plate silencer with moderately stiff plates is shown to outperform the typical plate silencer with very stiff plate. Whilst releasing the harsh requirement on the bending stiffness of the plate, the proposed hybrid silencer provides a more flattened and uniform TL and a widened stopband by about 30%.

## Session 1aPA

## Physical Acoustics, Architectural Acoustics, and Noise: Recent Developments in Computational Acoustics for Complex Indoor and Outdoor Spaces

D. Keith Wilson, Cochair
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Dinesh Manocha, Cochair
*Computer Science, University of North Carolina at Chapel Hill, Chapel Hill, NC 27599-3175*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**1aPA1. Modeling reflections from rough surfaces in complex spaces.** Samuel Siltanen, Alex Southern, and Lauri Savioja (Department of Media Technology, Aalto University, PO Box 15400, Aalto FI-00076, Finland, Lauri.Savioja@aalto.fi)

Acoustic reflections from rough surfaces occur both in indoor and outdoor environments. Detailed modeling of such reflections is possible with wave-based modeling algorithms. However, their time and resource consumption limits the applicability of such algorithms in practical modeling tasks where the modeled space is large or even unbounded. On the other hand, geometrical acoustics modeling techniques are more efficient in most cases, but they are not able to capture the wave interaction with the rough surface. The presented solution combines a geometric modeling algorithm with a theoretical model of the rough surface. The geometric algorithm is an efficient beam tracer. The theoretical model assumes long wavelength compared to the dimensions of the surface details. The model shows that the effect of the rough surface can be approximated with a exponential decay tail after an impulse in the time domain impulse response. A further approximation is to present multiple such reflections with a decay resembling a gamma distribution. Several complex example cases with a large number of reflections are shown. In addition, comparison of the theoretical model to finite-difference time-domain algorithm modeling results is given. The results support the applicability of the presented approach to practical modeling tasks.

**8:25**

**1aPA2. Computational modeling of broadband acoustic fields in enclosures with specular reflection boundaries using a first-principles energy method.** Donald B. Bliss, Krista A. Michalis, and Linda P. Franzoni (Mechanical Engineering, Duke University, Durham, NC 27708, dbb@duke.edu)

Steady-state sound fields in enclosures with specular reflection boundaries are modeled with a first-principle energy-intensity boundary element method using uncorrelated broadband directional sources. The specular reflection field is represented by a limited set of spherical harmonics, orthogonal on the half-space. For each boundary element, the amplitudes of these harmonics are determined from the incident field from all other elements and sources, and are subject to an energy conservation integral constraint using a Lagrange multiplier method. The computational problem is solved using an iterative relaxation method starting from the 3-D diffuse reflection solution. At each iteration, directivity harmonics are estimated by post-processing and the influence matrix is refined accordingly. For internal sources, simple first reflection images improve accuracy with virtually no penalty on computation time. Convergence occurs in relatively few relaxation steps. Extrapolating to an infinite number of boundary elements and iterations gives very accurate results. Results are compared to exact benchmark solutions obtained from a frequency-by-frequency modal analysis, and to a broadband image method. The method of absorption scaling is verified for 3-D cases, and showing that the spatial variation in rooms is largely determined by source position and the relative distribution of absorption, but not the overall absorption level.

**8:45**

**1aPA3. An edge-source integral equation for the calculation of scattering.** U. Peter Svensson (Acoustics Research Centre, Department of Electronics and Telecommunications, Norwegian University of Science and Technology, NTNU IET, Trondheim - NO-7491, Norway, svensson@iet.ntnu.no) and Andreas Asheim (Department of Computer Science, Katholieke Universiteit, Heverlee, Belgium)

A new integral equation for the scattering from rigid, or pressure-release, convex polyhedra and disks is presented. It is based on edge diffraction as a complement to the geometrical acoustics components, and uses directional edge sources as unknowns in the frequency-domain integral equation. Comparisons with reference results show that the new method gives correct results down to 0 Hz in spite of being a geometrically based method [Asheim & Svensson, Report TW610, KU Leuven, Dept. of Computer Science, 2012]. The two-dimensional unknowns are solved for all edges, straight or curved, of the scattering object, but this reduces to a one-dimensional unknown for certain geometries such as axisymmetric scattering from a circular thin disk, or plane-wave incidence onto a polygonal cylinder. The general integral equation can be solved with the regular Nyström technique, iteratively or by direct inversion. The scattered field is computed in a post-processing stage, and is added to the geometrical acoustics and first-order diffraction components, which are computed separately. The formulations for non-convex external, as well as internal, geometries are laid out, and time-domain versions of the integral equation are linked to previously published work on edge diffraction impulse responses.

**9:05**

**1aPA4. A fast wave-based hybrid method for interactive acoustic simulation in large and complex environments.** Tian Xiao (EE Boost Inc., 618 Powers Ferry RD, Cary, NC 27519, xiao@eeboost.com)

Modern acoustic applications require fast and accurate simulation in large and complex environments. Current methods are rather limited to very simple environments or not able to perform fast interactive simulations. Our objective is to develop an innovative and practical method to perform real-time or near real-time accurate interactive simulation in large and complex environments, such as urban environments up to one kilometer. A new method, the embedded hybrid method using immersed boundary within k-space PSTD has been proposed and developed to achieve the objective. Its performance, accuracy, parallel hardware acceleration, and capabilities to handle all kinds of environmental complexities have been rigorously validated and verified by a number of examples. It shows that with the utilization of modern many-core GPUs, the method can perform real-time or near real-time accurate interactive simulation in large environments of hundreds to thousands of meters with all kinds of complexities including (1) inhomogeneous and absorptive medium, (2) curved and arbitrarily-shaped objects, (3) and moving sources, receivers, and objects, and time-varying medium.

**9:25**

**1aPA5. Adaptive rectangular decomposition: A spectral, domain-decomposition approach for fast wave solution on complex scenes.** Nikunj Raghuvanshi (Microsoft Research, 1 Microsoft Way, Redmond, WA 98052, nikunjr@gmail.com), Ravish Mehra, Dinesh Manocha, and Ming C. Lin (Computer Science, University of North Carolina at Chapel Hill, Chapel Hill, Washington)

Computational wave propagation is increasingly becoming a practical tool for acoustic prediction in indoor and outdoor spaces, with applications ranging from noise control to architectural acoustics. We discuss Adaptive Rectangular Decomposition (ARD), that decomposes a complex 3D domain into a set of disjoint rectangular partitions. Assuming spatially-invariant speed of sound, spectral basis functions are derived from the analytic solution and used to time-step the field with high spatio-temporal accuracy within each partition. This allows close-to-Nyquist numerical grids, with as low as 3 points per wavelength, resulting in large performance gains of ten to hundred times compared to the Finite-Difference Time-Domain method. The coarser simulation grid also allows much larger computational domains. ARD employs finite-difference interface operators to transfer waves between adjoining rectangular partitions. We show that efficient, spatially-compact interface operators can be designed to ensure low numerical errors. Numerical solutions obtained with ARD are compared to analytical solutions on simple geometries and good agreement is observed.

**9:45**

**1aPA6. Real-time sound propagation and noise modeling in outdoor environments using Equivalent Source Formulation.** Ravish Mehra, Dinesh Manocha, Lakulish Antani (Computer Science, University of North Carolina at Chapel Hill, Columbia Street, Chapel Hill, NC 27599-3175, dmanocha@gmail.com), and Nikunj Raghuvanshi (Microsoft Research, Microsoft, Redmond, WA)

We address the problem of wave-based sound propagation in outdoor and urban environments. The goal is to accurately simulate acoustic effects, including interference, diffraction, scattering, and higher-order wave effects in large outdoor scenes. We give an overview of a precomputed wave-based solver that is based on equivalent source method and is mainly applicable to large, open scenes [Mehra et al. 2012]. As part of a preprocessing step, it computes a per-object transfer function that models the scattering behavior of each object, and handles pair-wise acoustic coupling between objects using inter-object transfer functions. The runtime component involves fast summation over all outgoing equivalent sources for all objects at the listener location. We highlight its runtime performance and memory efficiency and use it for noise modeling and prediction in outdoor scenes spanning a few hundreds of meters. The sound field is computed in three dimensions, modeling frequency-dependent propagation above ceilings, around buildings and corners, and high-order interactions.

**10:05–10:25 Break**

## *Contributed Papers*

**10:25**

**1aPA7. Coupling of parabolic equation method with the scattering of sound.** Santosh Parakkal, D. Keith Wilson, and Sergey N. Vecherin (US Army, ERDC-CRREL-NH, 72 Lyme Road, Hanover, NH 03755, Santosh.Parakkal@usace.army.mil)

The problem of sound scattering by an infinitely long penetrable and impenetrable cylinder suspended over a realistic impedance ground is investigated. The analytical approach using the image source method in the scattering of sound involves expressing the total sound field at any receiver point (over a locally reacting ground) as the sum total of the direct field, the ground reflection from the source, scattered field by the actual cylinder and finally by its image. The exact solutions can then be expressed as an infinite series, containing Bessel and Hankel functions of increasing order. The coefficients of the scattered field are determined by matching the desired boundary condition (for a rigid circular cylinder, for example, the normal component of velocity is zero on the boundary). Although the preceding approach is commonly used in theoretical treatments of sound scattering, to the best of our knowledge this is the first time it has been attempted numerically with coupling to the Parabolic Equation (PE) method. Presented is a Two-dimensional case of sound scattering of PE generated acoustic field by an impenetrable (soft or rigid) and penetrable cylindrical obstacle over a finite impedance ground in a non-refractive atmosphere.

**10:40**

**1aPA8. Time-domain simulation of long-range sound propagation in an atmosphere with temperature gradient.** Z. C. Zheng and Guoyi Ke (Aerospace Engineering, University of Kansas, 1530 W 15th Street, Lawrence, KS 66045, zzheng@ku.edu)

A numerical model for linearized Euler equations using finite difference in time-domain (FDTD) simulation is developed to simulate sound propagation with temperature gradient in the atmosphere. The speed of sound in the air varies with the temperature at different altitude above the ground due to the effect of temperature gradient. For sound propagation at long ranges, an algorithm of moving-frame method is implemented with parallel computation. The numerical results are compared with analytical solutions for sound propagation with downward and upward refraction caused by the speed of sound linearly increasing (downward refraction) or decreasing (upward refraction) with altitude. The 2D normal mode analytical solutions are used to compare the downward refraction results, and the residue series analytical solutions are used to compare the upward refraction results. The comparison show that the numerical simulation results agree very well with the analytical solutions for both downward refraction and upward refraction cases. Several examples of long- and short-range simulation results are then presented.

**10:55**

**1aPA9. Wine glass resonance experiment and demonstration.** Benjamin C. Thines (University of Central Arkansas, Conway, AR 72034, thinesbc@gmail.com)

Breaking a wine glass with sound is a visually striking achievement and a great way to get potential students interested in Physics. The goal of this project is to not only break the wine glass but to build an apparatus that is portable and easily setup for lecture room demonstrations as well as outreach to area schools. The apparatus should also provide enough visibility for a room full of observers to easily see the process. In order to be able to observe the small deflections of the object a variable frequency strobe will be employed. A strobe has the benefit of being able to see in real time what is going on at a much higher frequency than the human eye would normally perceive. In a larger setting a camera could be used to relay the relatively small image of the wine glass to a projector for better visibility. From a more technical stand point, the project will provide an opportunity to experiment with resonance on a variety of different shapes and compositions of items. In order to prepare for the final demonstration, many different wine glasses will be tested in the test chamber.

**11:10**

**1aPA10. Aperiodicity and ground effects on the sonic crystal noise barriers.** Shahram Taherzadeh, Imran Bashir, Keith Attenborough, and Alvin Y. Chong (MCT, The Open University, Walton Hall, Milton Keynes, Bucks MK7 6AA, United Kingdom, s.taherzadeh@open.ac.uk)

Sonic crystal structures consisting of periodically-arranged solid vertical cylinders can act as sound barriers at certain frequencies. Their performance depends on the filling fraction which is determined by the spacing and cylinder radius. To be effective the filling fraction must be high. This means that periodic arrays with relatively low filling fractions such as in trees belts are not effective as traffic noise barriers. The effects of partially perturbing the positions of sonic crystal elements have been investigated by modelling and laboratory measurements and have been shown to improve the insertion loss of the periodic structure. It is argued that partial perturbation of regular tree planting near highways will improve their noise attenuation. Furthermore, much previous research assumes the sonic crystal structure to be in the free field, i.e. no account has been taken of the presence of the ground surface. With a conventional, wall type, barrier the ground effect is reduced by presence of the barrier. Laboratory measurements have been made of periodic and aperiodic arrays of cylinders placed with their axes normal to acoustically hard and soft surfaces. It is found that the ground effects and sonic crystal band gap effects are additive.



**11:25**

**1aPA11. Partial field decomposition of jet noise using optimally located virtual reference microphones.** Alan T. Wall, Kent L. Gee, Tracianne B. Neilsen (Dept. of Physics and Astronomy, Brigham Young University, N283 ESC, Provo, UT 84602, alantwall@gmail.com), and Michael M. James (Blue Ridge Research and Consulting, Asheville, NC)

The application of partial field decomposition (PFD) techniques based on a singular value decomposition to jet noise fields is useful for estimating the number of incoherent (equivalent) noise sources within a jet and for implementing near-field acoustical holography, but it does not generally provide physically meaningful partial fields (i.e. partial fields related to individual sources). Among several PFD methods that were designed to generate physically meaningful partial fields, the method developed by Kim et al. [JASA 115(4), 2004] finds the optimal locations of references in a sound field and places virtual references at those locations. In past investigations this method has been successfully applied to locate discrete numerical and physical sources and to generate partial fields related to each source. In this study, Kim's method is applied to a full-scale jet installed on a military aircraft in an attempt to obtain physically meaningful partial fields. The partial fields obtained using these optimally located virtual references are compared to the partial fields obtained from other PFD methods.

**11:40**

**1aPA12. Observations with grazing and vertical incidence methods of ground impedance estimation.** Michael J. White (US Army ERDC/CERL, PO Box 9005, Champaign, IL 61826, michael.j.white@usace.army.mil), George W. Swenson, and Jeffrey D. Borth (Department of Electrical and Computing Engineering, University of Illinois at Urbana-Champaign, Urbana, IL)

At locations near a ground surface of interest, the ground impedance may be evaluated using a loudspeaker suitably disposed to broadcast toward both the ground and two vertically-separated microphones. By evaluating the complex gain ratio between the pair for a single tone, the usual approximation to the Green function for the monopole field above a locally-reacting ground can be inverted to find the surface impedance. This inversion is somewhat sensitive to noise, but it can also be found to vary according to the placement of microphones and speaker, apart from speaker directivity. Recently the method of Soh et al. [Soh et al. JASA 128:5 EL286 2010] was proposed for measuring ground impedance some distance from the source. Because the method relies more directly on the boundary condition at the ground, it may offer some benefit for use at shorter distances as well. We discuss the comparisons between measurements interpreted by both techniques, consider noise entering the estimation process and placement effects.

# Session 1aSA

## Structural Acoustics and Vibration: Damping Applications and Modeling

Benjamin M. Shafer, Chair
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### *Invited Papers*

**9:00**

**1aSA1. Damping: Some often overlooked facts.** Eric E. Ungar (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138-1118, eungar@acentech.com)

Damping - dissipation of mechanical energy - has significant effects on only some types of vibrations. Damping can result from many mechanisms, many of which cannot readily be modeled, but prediction of details of motions requires correct representations of the dominant mechanisms. The assumption of viscous damping permits one to analyze vibrations via relatively easily solved linear differential equations, but can lead to results that do not represent reality. Several measures of damping are based on simple models involving frequency-independent viscous damping, but realistic damping behavior often may be better represented by a frequency-dependent loss factor. The chemical properties of plastics and elastomers generally are not known well enough to permit assessment of the behavior of such materials without dynamic measurements. Structural configurations with relatively high damping may be obtained by combining high-damping polymeric materials with efficient structural materials only if the configurations are such that for a given deformation the high-damping material stores a considerable fraction of the total mechanical energy. This is manifest in the behavior of free-layer and constrained-layer damping treatments and in their design equations, which also indicate that a damping material that is very good for one of these types of treatments may not be good for the other.

**9:25**

**1aSA2. Acoustical performance of damped gypsum board in double wood stud wall assemblies.** John LoVerde and David W. Dong (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com)

A common assembly for demising walls in multifamily residential projects is double wood studs with multiple layers of gypsum board on one or both sides. While improving the acoustical performance, installing multiple layers of gypsum board adds significant cost to the project; complicates scheduling, materials storage, and delivery; and may require additional inspections. On some types of projects, removing these complications has been desired. In these cases, the use of damped gypsum board has been pursued to reduce the number of layers of gypsum board on the demising wall while attempting to achieve equivalent acoustical performance. Acoustical testing was performed to determine if damped gypsum board was a suitable alternative to the standard gypsum board assemblies in typical demising constructions. Laboratory and field tests were performed on double wood stud walls with a variety of gypsum board configurations, and the results are presented.

**9:50**

**1aSA3. Effects of boundary conditions on the transmission of impulsive sound at low frequencies through building components into enclosed spaces.** Marcel C. Remillieux (Mechanical Engineering, Virginia Tech, 149 Durham Hall, Blacksburg, VA 24061, mremilli@vt.edu)

The transmission of impulsive sound at low frequencies, through building components, into enclosed spaces is solved numerically. This investigation is directed at what is essentially a transient problem. The shapes of the vibro-acoustic waveforms, in particular peak values, are of principal concern since they relate directly to the auditory response of occupants and possibility to structural damage. The case of an N-wave impinging upon a window panel backed by a rigid rectangular enclosure is considered in this study. At low frequencies, the vibro-acoustic response of a typical building component is dominated by a few modes. Besides, the response is very sensitive to boundary conditions. Therefore, the vibro-acoustic response of the system can be altered by tuning the boundary conditions of the structural components. Three types of boundary conditions are examined: simply-supported, fixed, and visco-elastic. It is demonstrated that the peak amplitudes of the vibro-acoustic waveforms can be reduced significantly by the appropriate choice of boundary conditions.

**10:15–10:30 Break**

**10:30**

**1aSA4. Predicting structural and acoustic-radiation loss factors using experimental modal analysis.** Micah R. Shepherd, Stephen A. Hambric, and John B. Fahnline (Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

Standard damping measurements cannot typically distinguish between structural losses and losses due to acoustic radiation. Since the trend in aerospace engineering is to use lighter and stiffer materials, aerospace panels often have their critical frequency at low to mid frequencies making the acoustic losses of comparable or greater value than the structural loss factors. A procedure for measuring

acoustic radiation loss factors is presented based on experimental modal analysis. Experimental modal analysis was performed on a composite panel with carbon-fiber facesheets and an aluminum honeycomb core. The loss factors using the standard decay and modal methods are compared to an energy-based method based on the power injection method. The acoustic radiation loss factors were then estimated using boundary element computations of the radiated noise with the modally reconstructed velocity as input. The acoustic radiation damping was then removed from the total damping to predict the losses due to structural damping only.



## *Contributed Papers*

10:55

**1aSA5. Vibration damping mechanisms for the reduction of sting in baseball bats.** Daniel A. Russell (Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802, drussell@engr.psu.edu)

When the impact between a baseball and a bat occurs outside the “sweet-spot” region, the resulting vibration in the handle often produces a stinging sensation in the hands. Pain from a poorly hit ball is primarily felt in the fleshy web between thumb and forefinger in the top (distal) hand, and also to a lesser degree in the heel of the bottom (proximal) hand, and the sensation of sting is more prevalent in aluminum bats than in wood. Several bat manufacturers have attempted to minimize handle vibration in aluminum bats through various methods including handle grips, foam injected into the hollow handle, two-piece construction joining composite handles to aluminum barrels, and the insertion of vibration absorbers in the taper region and in the knob of the handle. This paper will assess and compare the performance of several such vibration damping applications implemented in baseball bats. Experimentally measured damping rates corresponding to the bending mode shapes responsible for sting will be compared for a variety of bat designs. The effectiveness of available commercially implemented damping mechanisms will be compared using frequency response functions and time signals.

11:10

**1aSA6. An approach to increase apparent damping with reduced subordinate oscillator array mass.** Aldo A. Glean, Joseph F. Vignola, John A. Judge, Teresa J. Ryan (Mechanical Engineering, Center for Acoustics, Vibrations and Structures, Catholic University of America, 620 Michigan Avenue, N.E, Washington, DC 20064, 10glean@cardinalmail.cua.edu), and Patrick F. O’Malley (Mechanical Engineering, Benedictine College, Atchison, KS)

The case of a lightly damped oscillator (primary mass) adorned with a set of substantially less massive oscillators is known as a subordinate oscillator array (SOA). An SOA can function as an energy sink on the primary, extracting vibration energy from it, thus adding apparent damping to the system. Low apparent Q is achieved by increasing non-dimensional bandwidth, which is the ratio of the bandwidth to the fundamental frequency of the primary. The mass of the subordinate set required to achieve the most rapid energy transfer from the primary is proportional to non-dimensional bandwidth squared. We have shown the limit of apparent damping achievable in these systems is the inverse of non-dimensional bandwidth. In practice, the utility of this result is limited because a great deal of mass (~30% of primary) is required to increase the apparent damping in the system such that Q_apparent→1. This work will focus on an alternative design strategy that produces comparable increase in apparent damping with less added mass. We describe numerical optimizations in which the non-dimensional bandwidth of the isolated natural frequencies of the SOA elements and the distribution of those isolated natural frequencies are used to minimize the total mass of the SOA.

11:25

**1aSA7. Viscous boundary layer correction on a pressure-field acoustic model.** Yizeng Li (Department of Mechanical Engineering, University of Michigan - Ann Arbor, 2350 Hayward Avenue, Ann Arbor, MI 48109, liyizeng52@hotmail.com), Lei Cheng (BOSE Corporation, Framingham, MA), and Karl Grosh (Department of Mechanical Engineering, University of Michigan - Ann Arbor, Ann Arbor, MI)

Fluid viscosity plays an important role in many acoustics and structural acoustics problems. For example, using an inviscid approximation to the flow of fluid-loaded micro-electro-mechanical systems and micro-scale biological structures results in large errors in the predicted response. Using a linearized Navier–Stokes solution, however, increases the number of unknowns by at least a factor of three compared to an inviscid approximation where pressure is the only degree of freedom. In this work, an approximate boundary condition is developed to include fluid viscosity for coupled fluid-structure systems. The viscous effect is included as a correction term to the inviscid boundary condition, written in terms of second order in-plane derivatives of pressure. This is the key step enabling the development of a variational formulation that is directly amenable for approximation in a finite element method (FEM) code as only a minor modification to existing structural acoustic code. Hence, this approach retains the great computational advantage over the conventional viscous FEM formulation. We show results demonstrating the accuracy of the approximate boundary condition as compared to the full three dimensional Navier-Stokes solution.