AES 133rd Convention Program

October 26 – 29, 2011

Moscone Convention Center, San Francisco, CA, USA

Session P1 **Friday, October 26**
9:00 am – 11:00 am **Room 121**

AMPLIFIERS AND EQUIPMENT

Chair: **Jayant Datta**

9:00 am

P1-1 A Low-Voltage Low-Power Output Stage for Class-G Headphone Amplifiers—*Alexandre Huffenus*, EASii IC, Grenoble, France

This paper proposes a new headphone amplifier circuit architecture, which output stage can be powered with very low supply rails from ±1.8 V to ±0.2 V. When used inside a Class-G amplifier, with the switched mode power supply powering the output stage, the power consumption can be significantly reduced. For a typical listening level of 2x100µW, the increase in power consumption compared to idle is only 0.7mW, instead of 2.5mW to 3mW for existing amplifiers. In battery-powered devices like smartphones or portable music players, this can increase the battery life of more than 15% during audio playback. Theory of operation, electrical performance and a comparison with the actual state of the art will be detailed.
Convention Paper 8684

9:30 am

P1-2 Switching/Linear Hybrid Audio Power Amplifiers for Domestic Applications, Part 2: The Class-B+D Amplifier—*Harry Dymond, Phil Mellor*, University of Bristol, Bristol, UK

The analysis and design of a series switching/linear hybrid audio power amplifier rated at 100 W into 8 Ω are presented. A high-fidelity linear stage controls the output, while the floating midpoint of the power supply for this linear stage is driven by a switching stage. This keeps the voltage across the linear stage output transistors low, enhancing efficiency. Analysis shows that the frequency responses of the linear and switching stages must be tightly matched to avoid saturation of the linear stage output transistors. The switching stage employs separate DC and AC feedback loops in order to minimize the adverse effects of the floating-supply reservoir capacitors, through which the switching stage output current must flow.
Convention Paper 8685

10:00 am

P1-3 Investigating the Benefit of Silicon Carbide for a Class D Power Stage—*Verena Grifone Fuchs*,[1,2] *Carsten Wegner*,[1,2] *Sebastian Neuser*,[1] *Dietmar Ehrhardt*[1]
[1]University of Siegen, Siegen, Germany
[2]CAMCO GmbH, Wenden, Germany

This paper analyzes in which way silicon carbide transistors improve switching errors and loss associated with the power stage. A silicon carbide power stage and a conventional power stage with super-junction devices are compared in terms of switching behavior. Experimental results of switching transitions, delay times, and harmonic distortion as well as a theoretical evaluation are presented. Emending the imperfection of the power stage, silicon carbide transistors bring out high potential for Class D audio amplification.
Convention Paper 8686

10:30 am

P1-4 Efficiency Optimization of Class G Amplifiers: Impact of the Input Signals—*Patrice Russo*,[1] *Gael Pillonnet*,[1] *Nacer Abouchi*,[1] *Sophie Taupin*,[2] *Frederic Goutti*[2]
[1]Lyon Institute of Nanotechnology, Lyon, France
[2]STMicroelectronics, Inc., Grenoble, France

Class G amplifiers are an effective solution to increase the audio efficiency for headphone applications, but realistic operating conditions have to be taken into account to predict and optimize power efficiency. In fact, power supply tracking, which is a key factor for high efficiency, is poorly optimized with the classical design method because the stimulus used is very different from a real audio signal. Here, a methodology has been proposed to find class G nominal conditions. By using relevant stimuli and nominal output power, the simulation and test of the class G amplifier are closer to the real conditions. Moreover, a novel simulator is used to quickly evaluate the efficiency with these long duration stimuli, i.e., ten seconds instead of a few milliseconds. This allows longer transient simulation for an accurate efficiency and audio quality evaluation by averaging the class G behavior. Based on this simulator, this paper indicates the limitations of the well-established test setup. Real efficiencies vary up to ±50% from the classical methods. Finally, the study underlines the need to use real audio signals to optimize the

supply voltage tracking of class G amplifiers in order to achieve a maximal efficiency in nominal operation.
Convention Paper 8687
This paper is presented by Eric Sturtzer.

Session P2 **Friday, October 26**
9:00 am – 10:30 am **Room 122**

NETWORKED AUDIO

Chair: **Ellen Juhlin**, Meyer Sound Labs

9:00 am

P2-1 Audio Latency Masking in Music Telepresence Using Artificial Reverberation **—***Ren Gang, Samarth Shivaswamy, Stephen Roessner, Akshay Rao, Dave Headlam, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Network latency poses significant challenges in music telepresence systems designed to enable multiple musicians at different locations to perform together in real-time. Since each musician hears a delayed version of the performance from the other musicians it is difficult to maintain synchronization, and there is a natural tendency for the musicians to slow their tempo while awaiting response from their fellow performers. We asked if the introduction of artificial reverberation can enable musicians to better tolerate latency by conducting experiments with performers where the degree of latency was controllable and for which artificial reverberation could be added or not. Both objective and subjective evaluation of ensemble performances were conducted to evaluate the perceptual responses at different experimental settings.
Convention Paper 8688

9:30 am

P2-2 Service Discovery Using Open Sound Control—*Andrew Eales*,[1,2] *Richard Foss*[2]
[1]Wellington Institute of Technology, Wellington, New Zealand
[2]Rhodes University, Grahamstown, Eastern Cape, South Africa

The Open Sound Control (OSC) control protocol does not have service discovery capabilities. The approach to adding service discovery to OSC proposed in this paper uses the OSC address space to represent services within the context of a logical device model. This model allows services to be represented in a context-sensitive manner by relating parameters representing services to the logical organization of a device. Implementation of service discovery is done using standard OSC messages and requires that the OSC address space be designed to support these messages. This paper illustrates how these enhancements to OSC allow a device to advertise its services. Controller applications can then explore the device's address space to discover services and retrieve the services required by the application.
Convention Paper 8689

10:00 am

P2-3 Flexilink: A Unified Low Latency Network Architecture for Multichannel Live Audio— *Yonghao Wang*,[1] *John Grant*,[2] *Jeremy Foss*[1]
[1]Birmingham City University, Birmingham, UK
[2]Nine Tiles Networks Ltd., Cambridge, UK

The networking of live audio for professional applications typically uses Layer 2-based solutions such as AES50 and MADI utilizing fixed time slots similar to Time Division Multiplexing (TDM). However, these solutions are not effective for best effort traffic where data traffic utilizes available bandwidth and is consequently subject to variations in QoS. There are audio networking methods such as AES47, which is based on asynchronous transfer mode (ATM), but ATM equipment is rarely available. Audio can also be sent over Internet Protocol (IP), but the size of the packet headers and the difficulty of keeping latency within acceptable limits make it unsuitable for many applications. In this paper we propose a new unified low latency network architecture that supports both time deterministic and best effort traffic toward full bandwidth utilization with high performance routing/switching. For live audio, this network architecture allows low latency as well as the flexibility to support multiplexing multiple channels with different sampling rates and word lengths.
Convention Paper 8690

Tutorial 1 **Friday, October 26**
9:00 am – 10:30 am **Room 133**

NOISE ON THE BRAIN—HEARING DAMAGE ON THE OTHER SIDE

Presenter: **Poppy Crum**, Dolby, San Francisco, CA, USA

Did you know that drinking a glass of orange juice every day may actually protect your hearing? Most discussions of hearing damage focus on what happens to the cochlea and inner ear. While this understanding is crucial to predicting and avoiding trauma that can lead to hearing loss, both acoustic and chemical stimuli can also have significant effects on higher brain areas. In some cases, thresholds and audiograms can look completely normal but listeners may have great difficulty hearing a conversation in a noisy environment. This session will explore the latest research regarding the effects of acoustic and chemical trauma, and how this damage manifests throughout the auditory pathway as changes in hearing sensitivity, cognition, and the experience of tinnitus. We will also consider recent research in chemically preserving hearing and combating these conditions with supplements as common as Vitamin C!

Broadcast/Media Streaming Session 1
Friday, October 26 **9:00 am – 10:30 am**
Room 131

WORKING WITH HTML5

Chair: **Valerie Tyler**, Daly City, CA, USA

Presenters: *Jan Linden*, Google, Mountain View, CA, USA

Greg Ogonowski, Orban, Diamond Bar, CA, USA
Charles Van Winkle, Adobe Systems Inc., Minneapolis, MN, USA

HTML5 is a language for structuring and presenting content for the World Wide Web, a core technology of the Internet. It is the fifth revision of the HTML standard. HTML5 has many features built into the code. One feature is the media player and how it handles media being downloaded or streamed. This session will look into the technical considerations for media to be played back as well as the user interfaces.

Product Design Session 1 — Friday, October 26
9:00 am – 11:00 am — Room 123

AUDIO DSP REQUIREMENTS FOR TOMORROW'S MOBILE & PORTABLE DEVICES

Presenters: **Bob Adams**, ADI
Juha Backman, Nokia Corporation, Espoo, Finland
Howard Brown, IDT
Peter Eastty, Oxford Digital Limited, Oxford, UK
Alan Kramer, SRS Labs
Cyril Martin, RIM

As the convergence of communications, entertainment, and computing races ahead, largely centered on portable and mobile devices where form factors are shrinking and style wins out over practicality of design in some instances, the challenges in delivering the audio DSP to provide good sound and differentiated effects are discussed by a panel of experts representing semiconductor manufacturers, mobile/portable device manufacturers, and DSP IP providers.

Friday October 26 — 9:00 am — Room 124

Technical Committee Meeting on Acoustics and Sound Reinforcement

Session P3 — Friday, October 26
10:00 am – 11:30 am — Foyer

POSTERS: AUDIO EFFECTS AND PHYSICAL MODELING

10:00 am

P3-1 Luciverb: Iterated Convolution for the Impatient—*Jonathan S. Abel, Michael J. Wilson*, Stanford University, Stanford, CA, USA

An analysis of iteratively applied room acoustics used by Alvin Lucier to create his piece "I'm Sitting in a Room" is presented, and a real-time system allowing interactive control over the number of rooms in the processing chain is described. Lucier anticipated that repeated application of a room response would bring out room resonances and smear the input sound over time. What was unexpected was the character of the smearing, turning a transient input into a sequence of crescendos at the room modes, ordered from high-frequency to low-frequency. Here, a room impulse response convolve with itself *L* times is shown have energy at the room modes, each with a roughly Gaussian envelope, peaking at the observed *L/2* times the frequency-dependent decay time.
Convention Paper 8691

10:00 am

P3-2 A Tilt Filter in a Servo Loop—*John Lazzaro, John Wawrzynek*, University of California, Berkeley, Berkeley, CA, USA

Tone controls based on the tilt filter first appeared in 1982, in the Quad 34 Hi-Fi preamp. More recently, tilt filters have found a home in specialist audio processors such as the Elysia *mpressor*. This paper describes a novel dynamic filter design based on a tilt filter. A control system sets the tilt slope of the filter, in order to servo the spectral median of the filter output to a user-specified target. Users also specify a tracking time. Potential applications include single-instrument processing (in the spirit of envelope filters) and mastering (for subtle control of tonal balance). Although we have prototyped the design as an AudioUnit plug-in, the architecture is also a good match for analog circuit implementation.
Convention Paper 8692

10:00 am

P3-3 Multitrack Mixing Using a Model of Loudness and Partial Loudness—*Dominic Ward*,[1] *Joshua D. Reiss*,[2] *Cham Athwal*[1]
[1]Birmingham City University, Birmingham, UK
[2]Queen Mary University of London, London, UK

A method for generating a mix of multitrack recordings using an auditory model has been developed. The proposed method is based on the concept that a balanced mix is one in which the loudness of all instruments are equal. A sophisticated psychoacoustic loudness model is used to measure the loudness of each track both in quiet and when mixed with any combination of the remaining tracks. Such measures are used to control the track gains in a time-varying manner. Finally we demonstrate how model predictions of partial loudness can be used to counteract energetic masking for any track, allowing the user to achieve better channel intelligibility in complex music mixtures.
Convention Paper 8693

10:00 am

P3-4 Predicting the Fluctuation Strength of the Output of a Spatial Chorus Effects Processor—*William L. Martens, Robert W. Taylor, Luis Miranda*, University of Sydney, Sydney, NSW, Australia

The experimental study reported in this paper was motivated by an exploration of a set of related audio effects comprising what has been called "spatial chorus." In contrast to a single-output, delay-modulation-based effects processor that produces a limited range of results, complex spatial imagery is produced when parallel processing channels are subjected to incoherent delay modulation. In order to develop a more adequate user interface for control of such "spatial chorus" effects processing, a systematic investigation of the relationship between algo- ➥

rithmic parameters and perceptual attributes was undertaken. The starting point for this investigation was to perceptually scale the amount of modulation present in a set of characteristic stimuli in terms of the auditory attribute that Fastl and Zwicker called “fluctuation strength.”
Convention Paper 8694

10:00 am

P3-5 Computer-Aided Estimation of the Athenian Agora Aulos Scales Based on Physical Modeling—*Areti Andreopoulou, Agnieszka Roginska*, New York University, New York, NY, USA

This paper presents an approach to scale estimation for the ancient Greek Aulos with the use of physical modeling. The system is based on manipulation of a parameter set that is known to affect the sound of woodwind instruments, such as the reed type, the active length of the pipe, its inner and outer diameters, and the placement and size of the toneholes. The method is applied on a single Aulos pipe reconstructed from the Athenian Agora fragments. A discussion follows on the resulting scales and the system’s advantages, and limitations.
Convention Paper 8695

10:00 am

P3-6 A Computational Acoustic Model of the Coupled Interior Architecture of Ancient Chavín—*Regina E. Collecchia, Miriam A. Kolar, Jonathan S. Abel*, Stanford University, Stanford, CA, USA

We present a physical, modular computational acoustic model of the well-preserved interior architecture at the 3,000-year-old Andean ceremonial center Chavín de Huántar. Our previous model prototype [Kolar et. al. 2010] translated the acoustically coupled topology of Chavín gallery forms to a model based on digital waveguides (bi-directional by definition), representing passageways, connected through reverberant scattering junctions, representing the larger room-like areas. Our new approach treats all architectural units as “reverberant” digital waveguides, with scattering junctions at the discrete planes defining the unit boundaries. In this extensible and efficient lumped-element model, we combine architectural dimensional and material data with sparsely measured impulse responses to simulate multiple and circulating arrival paths between sound sources and listeners.
Convention Paper 8696

10:00 am

P3-7 Simulating an Asymmetrically Saturated Nonlinearity Using an LNLNL Cascade—
Keun Sup Lee,[1] *Jonathan S. Abel*[2]
[1]DTS, Inc., Los Gatos, CA, USA
[2]Stanford University, Stanford, CA, USA

The modeling of a weakly nonlinear system having an asymmetric saturating nonlinearity is considered, and a computationally efficient model is proposed. The nonlinear model is the cascade of linear filters and memoryless nonlinearities, an LNLNL system. The two nonlinearities are upward and downward saturators, limiting, respectively, the amplitude of their input for either positive or negative excursions. In this way, distortion noted in each half an input sinusoid can be separately controlled. This simple model is applied toy simulating the signal chain of the Echoplex EP-4 tape delay, where informal listening tests showed excellent agreement between recorded and simulated program material.
Convention Paper 8697

10:00 am

P3-8 Coefficient Interpolation for the Max Mathews Phasor Filter—*Dana Massie*, Audience, Inc., Mountain View, CA, USA

Max Mathews described what he named the “phasor filter,” which is a flexible building block for computer music, with many desirable properties. It can be used as an oscillator or a filter, or a hybrid of both. There exist analysis methods to derive synthesis parameters for filter banks based on the phasor filter, for percussive sounds. The phasor filter can be viewed as a complex multiply, or as a rotation and scaling of a 2-element vector, or as a real valued MIMO (multiple-input, multiple-output) 2nd order filter with excellent numeric properties (low noise gain). In addition, it has been proven that the phasor filter is unconditionally stable under time varying parameter modifications, which is not true of many common filter topologies. A disadvantage of the phasor filter is the cost of calculating the coefficients, which requires a sine and cosine in the general case. If pre-calculated coefficients are interpolated using linear interpolation, then the poles follow a trajectory that causes the filter to lose resonance. A method is described to interpolate coefficients using a complex multiplication that preserves the filter resonance.
Convention Paper 8698

10:00 am

P3-9 The Dynamic Redistribution of Spectral Energies for Upmixing and Re-Animation of Recorded Audio —*Christopher J. Keyes*, Hong Kong Baptist University, Kowloon, Hong Kong

This paper details a novel approach to upmixing any n channels of audio to any arbitrary n+ channels of audio using frequency-domain processing to dynamically redistribute spectral energies across however many channels of audio are available. Although primarily an upmixing technique, the process may also help the recorded audio regain the sense of “liveliness” that one encounters in concerts of acoustic music, partially mimicking the effects of sound spectra being redistributed throughout a hall due to the dynamically changing radiation patterns of the instruments and the movements of the instruments themselves, during performance and recording. Preliminary listening tests reveal listeners prefer this technique 3 to 1 over a more standard upmixing technique.
Convention Paper 8699

10:00 am

P3-10 Matching Artificial Reverb Settings to Unknown Room Recordings: A Recommendation System for Reverb Plugins—*Nils Peters*,[1,2] *Jaeyoung Choi*,[1] *Howard Lei*[1]
[1]International Computer Science Institute, Berkeley, CA, USA
[2]University of California Berkeley, Berkeley, CA, USA

For creating artificial room impressions, numerous reverb plugins exist and are often controllable by many parameters. To efficiently create a desired room impression, the sound engineer must be familiar with all the available reverb setting possibilities. Although plugins are usually equipped with many factory presets for exploring available reverb options, it is a time-consuming learning process to find the ideal reverb settings to create the desired room impression, especially if various reverberation plugins are available. For creating a desired room impression based on a reference audio sample, we present a method to automatically determine the best matching reverb preset across different reverb plugins. Our method uses a supervised machine-learning approach and can dramatically reduce the time spent on the reverb selection process.
Convention Paper 8700

Workshop 1 **Friday, October 26**
10:00 am – 12:00 noon **Room 130**

FORENSIC AUTHENTICATION OF DIGITAL AUDIO

Presenters: **Catalin Grigoras**, University of Colorado at Denver, Denver, CO, USA
Jeff M. Smith, National Center for Media Forensics, Denver, CO, USA; University of Colorado Denver, Denver, CO, USA

Following the successful presentation of numerous papers on the subject of digital audio authentication at the recent 46th International Conference of the AES on Audio Forensics, this workshop will summarize and present the latest developments in the field. Digital audio authentication methods including data analysis, database management and validation, inter-laboratory testing, and methods for ENF analysis will be discussed. Dr. Catalin Grigoras and Jeff M. Smith of the National Center for Media Forensics will share research and findings related to these topics and explore implications that the latest developments mean for forensic audio analysis and digital media authentication.

Friday October 26 **10:00 am** **Room 124**
Technical Committee Meeting on Archiving, Restoration, and Digital Libraries

Session P4 **Friday, October 26**
10:30 am – 11:30 am **Room 122**

AUDIO IN EDUCATION

Chair: **Jason Corey**, University of Michigan, Ann Arbor, MI, USA

10:30 am

P4-1 Teaching Audio Processing Software Development in the Web Browser—*Matthias Birkenstock, Jörn Loviscach*, Fachhochschule Bielefeld (University of Applied Sciences), Bielefeld, Germany

Web-based learning progresses from lectures and videos to hands-on software development problems that are to be solved interactively in the browser. This work looks into the technical underpinnings required and available to support teaching the mathematics of audio processing in this fashion. All intensive computations are to happen on the client to minimize the amount of data transfer and the computational load on the server. This setup requires editing source code and executing it in the browser, dealing with the audio computation in the browser, playing back and visualizing computed waveforms. Validating the user's solution cannot be done by a sample-by-sample comparison with a "correct" result, but requires a tolerant comparison based on psychoacoustic features.
Convention Paper 8701

11:00 am

P4-2 *[Paper withdrawn.]*

11:30 am

P4-3 Distance Learning Strategies for Sound Recording Technology—*Duncan Williams*, University of Plymouth, Plymouth, UK

This paper addresses the design of a full credit remote access module as part of an undergraduate degree course in Sound Recording Technology at a public university in Texas. While audio engineering has been historically regarded by industry, and to a certain extent the corresponding educational sector, as a vocational skill, and as such, one that must be learned in practice, the client university required that all content be delivered, facilitated, and assessed entirely electronically—a challenge that necessitated a number of particular pedagogical approaches. This work focuses on the advantages and disadvantages of such a system for technical and vocational content delivery in practice.
Convention Paper 8703

Live Sound Seminar 1 **Friday, October 26**
10:30 am – 12:00 noon **Room 120**

POWER FOR LIVE EVENTS

Chair: **Kenneth Fause**, Auerbach Pollock Friedlander, San Francisco, CA, USA

Panelists: *Steve Bush*, Meyer Sound Labs, Inc.
Marc Kellom, Crown Audio, Inc.
Randall Venerable, Generators Unlimited, Inc.

A discussion of power for live events where the panelists will focus on practical considerations. Power for modern amplifiers—how much is really needed? Power factor of loads—real power, reactive power, why this matters. Non-linear loads and harmonics. High-quality audio sys- ➥

tems co-existing with theatrical lighting, rigging, video, catering, and cooling—Electromagnetic Compatibility in the real world. Safety and regulatory issues.

Broadcast/Media Streaming Session 2
Friday, October 26 10:45 am – 12:15 pm
Room 131

FACILITY DESIGN

Chair: **John Storyk**, Walters-Storyk Design Group, Highland, NY, USA

Presenters: *Kevin Carroll*, Sonic Construction LLC
Cindy McSherry-Martinez, Studio Trilogy
Paul Stewart, Genelec

A panel of leading studio contractors and installation experts will provide a real-world users survey of specific products and acoustic materials commonly (and rarely) incorporated in professional critical listening environments. Optimal options for doors, glass, HVAC, variable acoustic panels, furniture, equipment racks, and other integral components of today's high-end (and budget conscious) TV and radio broadcast facilities will be discussed in detail. This is not an advertorial event. Contractor recommendations are based on personal field experience with these products. Their success is contingent on their ability to provide clients with cost-effective solutions to a myriad of technical and aesthetic issues.

Product Design Session 2 Friday, October 26
10:45 am – 12:30 pm Room 132

AVB NETWORKING FOR PRODUCT DESIGNERS

Chair: **Robert Silfvast**, Avid

Panelists: *John Bergen*, Marvell
Jeff Koftinoff, Meyer Sound
Morten Lave, TC Applied Technologies
Lee Minich, LabX Technologies
Matt Mora, Apple Inc.
Dave Olsen, Harman International
Michael Johas Teener, Broadcom

This session will cover the essential technical aspects of Audio Video Bridging technology and how it can be deployed in products to support standards-based networked connectivity. AVB is an open IEEE standard and therefore promises low cost and wide interoperability among products that leverage the technology. Speakers from several different companies will share insights on their experiences deploying AVB in real products. The panelists will also compare and contrast the open-standards approach of AVB with proprietary audio-over-Ethernet technologies.

Game Audio Session 1 Friday, October 26
11:00 am – 12:30 pm Room 123

A WHOLE WORLD IN YOUR HANDS: NEW TECHNIQUES IN GENERATIVE AUDIO BRING ENTIRE GAME WORLDS INTO THE REALMS OF MOBILE PLATFORMS

Presenter: **Stephan Schütze**, Sound Librarian

"We can't have good audio; there is not enough memory on our target platform." This is a comment heard far too often especially considering it's incorrect. Current technology already allows for complex and effective audio environments to be made with limited platform resources when developed correctly, but we are just around the corner from an audio revolution.

The next generation of tools being developed for audio creation and implementation will allow large and complex audio environments to be created using minimal amounts of resources. While the new software apps being developed are obviously an important part of this coming revolution it is the techniques, designs, and overall attitudes to audio production that will be the critical factors in successfully creating the next era of sound environments.

This presentation will break down and discuss this new methodology independent of the technology and demonstrate some simple concepts that can be used to develop a new approach to sound design. All the material presented in this talk will benefit development on current and next gen consoles as much as development for mobile devices.

Friday October 26 11:00 am Room 124

Technical Committee Meeting on Hearing and Hearing Loss Prevention

Historical Program
H1 - POPULAR MISCONCEPTIONS ABOUT MAGNETIC RECORDING HISTORY AND THEORY —THINGS YOU MAY HAVE MISSED OVER THE PAST 85 YEARS
Friday, October 26, 11:15 am – 12:15 pm
Room 121

Presenters: **Jay McKnight**, Magnetic Reference Lab, Cupertino, CA, USA
Jeffrey McKnight, Creativity, Inc., San Carlos, CA, USA

• Who really discovered AC Bias? The four groups that re-discovered it around 1940, including one person you've probably never heard of.
• How does AC bias actually work?
• Is a recording properly described by "surface induction"?
• The story of the "effective" gap length of a reproducing head, and a correction to Westmijze's Gap Loss Theory.
• Does Wallace's "Thickness Loss" properly describe the wavelength response?
• Does disconnecting the erasing head make a quieter recording?

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Friday, October 26, 12:30 pm – 2:00 pm
Room 134

Opening Remarks:
• Executive Director Bob Moses
• President Jan Pedersen
• Convention Co-chairs Jim McTigue/Valerie Tyler

Program:
• AES Awards Presentation
• Introduction of Keynote Speaker
• Keynote Address by Steve Lillywhite

Awards Presentation
Please join us as the AES presents special awards to those who have made outstanding contributions to the Society in such areas of research, scholarship, and pub-

lications, as well as other accomplishments that have contributed to the enhancement of our industry. The awardees are:

BOARD OF GOVERNORS AWARD
- **Kyungwhoon Cheun**
- **Ricardo Escallón**
- **Jay Fouts**
- **Cesar Lamstein**
- **Gunther Melendez**
- **John Oh**
- **Ville Pulkki**
- **Jeff. M. Smith**
- **Michael Santucci**
- **Marcela Zorro**

FELLOWSHIP AWARD
- **Juha Backman**
- **David Bialik**
- **Poppy Crum**
- **Carlos Indio Gauvron**
- **Garry Margolis**
- **Jorge Moreno**
- **Woon-Seng Gan**
- **John Storyk**

DISTINGUISHED SERVICES MEDAL
- **John Vanderkooy**

HONORARY MEMBERSHIP AWARD
- **Ronald E. Uhlig**

Keynote Speaker

"Listen With Your Ears, And Not Your Eyes"

Multi-Platinum record producer (and Commander of The British Empire/CBE recipient) **Steve Lillywhite**, has collaborated with stars ranging from The Rolling Stones, U2, Peter Gabriel, Morrissey, Counting Crows, and The Pogues to The Killers, Dave Matthews Band, and Thirty Seconds to Mars. Over the past thirty years, he has made an indelible impact on contemporary music, and he continues to hone the razor edge with innovative new projects. His AES Keynote address will focus on the importance of "studio culture," and on inspiring and managing the creative process. He will also stress the importance of embracing new technology while avoiding the trap of becoming enslaved by it. Steve Lillywhite's studio philosophy emphasizes the axiom "Listen with your ears and not your eyes."

Session P5 **Friday, October 26**
2:00 pm - 6:00 pm **Room 121**

MEASUREMENT AND MODELS

Chair: **Louis Fielder**, Dolby, San Francisco, CA, USA

2:00 pm

P5-1 Measurement of Harmonic Distortion Audibility Using a Simplified Psychoacoustic Model—*Steve Temme, Pascal Brunet, Parastoo Qarabaqi,* Listen, Inc., Boston, MA, USA

A perceptual method is proposed for measuring harmonic distortion audibility. This method is similar to the CLEAR (Cepstral Loudness Enhanced Algorithm for Rub & buzz) algorithm previously proposed by the authors as a means of detecting audible Rub & Buzz, which is an extreme type of distortion [1,2]. Both methods are based on the Perceptual Evaluation of Audio Quality (PEAQ) standard [3]. In the present work, in order to estimate the audibility of regular harmonic distortion, additional psychoacoustic variables are added to the CLEAR algorithm. These variables are then combined using an artificial neural network approach to derive a metric that is indicative of the overall audible harmonic distortion. Experimental results on headphones are presented to justify the accuracy of the model.
Convention Paper 8704

2:30 pm

P5-2 Overview and Comparison of and Guide to Audio Measurement Methods—*Gregor Schmidle, Danilo Zanatta*, NTi Audio AG, Schaan, Liechtenstein

Modern audio analyzers offer a large number of measurement functions using various measurement methods. This paper categorizes measurement methods from several perspectives. The underlying signal processing concepts, as well as strengths and weaknesses of the most popular methods are listed and assessed for various aspects. The reader is offered guidance for choosing the optimal measurement method based on the specific requirements and application.
Convention Paper 8705

3:00 pm

P5-3 Spherical Sound Source for Acoustic Measurements—*Plamen Valtchev,*[1] *Dimitar Dimitrov*,[2] *Rumen Artarski*[3]
[1]Spherovox, Sofia, Bulgaria
[2]BMS Production
[3]Thrax, Sofia, Bulgaria

A spherical sound source for acoustic measurements is proposed, consisting of a pair of coaxial loudspeakers and a pair of compression drivers radiating into a common radially expanding horn in full 360-degree horizontal plane. This horn's vertical radiation pattern is defined by the enclosures of the LF arrangement. The LF membranes radiate spherically the 50 to 500 Hz band, whereas their HF components complete the horizontal horn reference ellipsoid-like diagram in both vertical directions to a spherical one. The assembly has axial symmetry, thus perfect horizontal polar pattern. The vertical pattern is well within ISO 3382 specifications, even without any "gliding." Comparative measurements against a purposely built typical dodecahedron revealed superior directivity, sound power capability, and distortion performance.
Convention Paper 8706

3:30 pm

P5-4 Low Frequency Noise Reduction by Synchronous Averaging under Asynchronous Measurement System in Real Sound Field—*Takuma Suzuki, Hiroshi Koide, Akihiko Shoji, Kouichi Tsuchiya, Tomohiko Endo, Shokichiro Hino*, Etani Electronics Co. Ltd., Ohta-ku, Tokyo, Japan

An important feature in synchronous averaging is the synchronization of sampling clock between the transmitting and receiving devices (e.g., D/A

and A/D converters). However, in the case where the devices are placed apart, synchronization becomes difficult to gain. For such circumstances, an effective method is proposed that enables synchronization for an asynchronous measurement environment. Normally, a swept-sine is employed as a measuring signal but because its power spectrum is flat, the signal-to-noise ratio (SNR) is decreased in a real environment with high levels of low frequency noise. To solve this, the devised method adopts the means of "enhancing the signal source power in low frequencies" and "placing random fluctuations in the repetitive period of signal source." Subsequently, its practicability was verified.
Convention Paper 8707

4:00 pm

P5-5 Measurement and Analysis of the Spectral Directivity of an Electric Guitar Amplifier: Vertical Plane—*Agnieszka Roginska,*[1] *Justin Mathew,*[1] *Andrew Madden,*[1] *Jim Anderson*,[1] *Alex U. Case*[2]
[1]New York University, New York, NY, USA
[2]University of Massachusetts—Lowell, Lowell, MA, USA

Previous work presented the radiation pattern measurement of an electric guitar amplifier densely sampled spatially on a 3-D grid. Results were presented of the directionally dependent spectral features on-axis with the driver, as a function of left/right position, and distance. This paper examines the directionally dependent features of the amplifier measured at the center of the amplifier, in relationship to the height and distance placement of the microphone. Differences between acoustically measured and estimated frequency responses are used to study the change in the acoustic field. This work results in a better understanding of the spectral directivity of the electric guitar amplifier in all three planes.
Convention Paper 8708

4:30 pm

P5-6 The Radiation Characteristics of a Horizontally Asymmetrical Waveguide that Utilizes a Continuous Arc Diffraction Slot—*Soichiro Hayashi, Akira Mochimaru, Paul F. Fidlin*, Bose Corporation, Framingham, MA, USA

One of the unique requirements for sound reinforcement speaker systems is the need for flexible coverage control—sometimes this requires an asymmetrical pattern. Vertical control can be achieved by arraying sound sources, but in the horizontal plane, a horizontally asymmetrical waveguide may be the best solution. In this paper the radiation characteristics of horizontally asymmetrical waveguides with continuous arc diffraction slots are discussed. Waveguides with several different angular variations are developed and their radiation characteristics are measured. Symmetrical and asymmetrical waveguides are compared, and the controllable frequency range and limitations are discussed.
Convention Paper 8709

5:00 pm

P5-7 Analysis on Multiple Scattering between the Rigid-Spherical Microphone Array and Nearby Surface in Sound Field Recording—*Guangzheng Yu, Bo-sun Xie, Yu Liu*, South China University of Technology, Guangzhou, China

The sound field recording with a rigid spherical microphone array (RSMA) is a newly developed technique. In room sound field recording, when an RSMA is close to a reflective surface, such as the wall or floor, the multiple scattering between the RSMA and the surface occurs and accordingly causes the error in the recorded signals. Based on the mirror-image principle of acoustics, an equivalent two-sphere model is suggested, and the multipole expansion method is applied to analyze the multiple scattering between the RSMA and reflective surface. Using an RSMA with 50 microphones the relationships among the error in RSMA output signals caused by multiple scattering and frequency, direction of incident plane wave, and distance of RSMA relative to reflective surface are analyzed.
Convention Paper 8710

5:30 pm

P5-8 Calibration of Soundfield Microphones Using the Diffuse-Field Response—*Aaron Heller*,[1] *Eric M. Benjamin*[2]
[1]SRI International, Menlo Park, CA, USA
[2]Surround Research, Pacifica, CA, USA

The soundfield microphone utilizes an array of microphones to derive various components of the sound field to be recorded or measured. Given that at high frequencies the response varies with the angle of incidence, it may be argued that any angle of incidence is as important as another, and thus it is important to achieve a calibration that achieves an optimum perceived response characteristic. Gerzon noted that "Above a limiting frequency $F \approx c/(\pi\ r)$ [. . .] it is found best to equalise the nominal omni and figure-of-eight outputs for an approximately flat response to homogeneous random sound fields." In practice, however, soundfield microphones have been calibrated to realize a flat axial response. The present work explores the theoretical ramifications of using a diffuse-field equalization target as opposed to a free-field equalization target and provides two practical examples of diffuse-field equalization of tetrahedral microphone arrays.
Convention Paper 8711

Session P6 — Friday, October 26
2:00 pm – 6:00 pm — Room 122

SPATIAL AUDIO OVER LOUDSPEAKERS

Chair: **Rhonda Wilson**, Dolby Labs, San Francisco, CA, USA

2:00 pm

P6-1 Higher Order Loudspeakers for Improved Surround Sound Reproduction in Rooms—*Mark A. Poletti*,[1] *Terence Betlehem*,[1] *Thushara Abhayapala*[2]
[1]Industrial Research Limited, Lower Hutt, New Zealand;

[2]Australian National University, Canberra, ACT, Australia

Holographic surround sound systems aim to accurately reproduce a recorded field in a small region of space around one or more listeners. This is possible at low frequencies with well-matched loudspeakers and acoustically treated rooms. At high frequencies the region of accurate reproduction shrinks and source localization is compromised. Furthermore, in typical rooms reflections compromise quality. High quality reproduction therefore requires large numbers of loudspeakers and the use of techniques to reduce unwanted reverberation. This paper considers the use of higher-order loudspeakers that have multiple modes of radiation to offer an extended frequency range and zone of accurate reproduction. In addition, if a higher-order microphone is used for calibration, room effects can be effectively removed.
Convention Paper 8712

2:30 pm

P6-2 A Model for Rendering Stereo Signals in the ITD-Range of Hearing—*Siegfried Linkwitz*, Linkwitz Lab, Corte Madera, CA, USA

Live sounds at a concert have spatial relationships to each other and to their environment. The specific microphone technique used for recording the sounds, the placement and directional properties of the playback loudspeakers, and the room's response determine the signals at the listener's ears and thus the rendering of the concert recording. For the frequency range, in which Inter-aural Time Differences dominate directional hearing, a free-field transmission line model will be used to predict the placement of phantom sources between two loudspeakers. Level panning and time panning of monaural sources are investigated. Effectiveness and limitations of different microphone pairs are shown. Recording techniques can be improved by recognizing fundamental requirements for spatial rendering. Observations from a novel 4-loudspeaker setup are presented. It provides enhanced spatial rendering of 2-channel sound.
Convention Paper 8713

3:00 pm

P6-3 A Method for Reproducing Frontal Sound Field of 22.2 Multichannel Sound Utilizing a Loudspeaker Array Frame—*Hiroyuki Okubo, Takehiro Sugimoto, Satoshi Oishi, Akio Ando*, NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

NHK has been developing Super Hi-Vision (SHV), an ultrahigh-definition TV system that has a 7,680 x 4,320 pixel video image and a 22.2 multichannel three-dimensional sound system. A loudspeaker array frame (LAF) integrated into a flat panel display can synthesize wavefront of frontal sound source and localize sound images on the display and back of the viewer by using technology to simulate sound propagation characteristics. This makes it possible to listen to 22.2 multichannel sound without installing 24 discrete loudspeakers surrounding the listener in the room. In this paper we describe the prototype of the LAF and its performance focusing on frontal sound reproduction.
Convention Paper 8714

3:30 pm

P6-4 Low-Frequency Temporal Accuracy of Small-Room Sound Reproduction—*Adam J. Hill*,[1] Malcolm *O. J. Hawksford*[2]
[1]University of Derby, Derby, Derbyshire, UK
[2]University of Essex, Colchester, Essex, UK

Small-room sound reproduction is strongly affected by room-modes in the low-frequency band. While the spectral impact of room-modes is well understood, there is less information on how modes degrade the spatiotemporal response of a sound reproduction system. This topic is investigated using a bespoke finite-difference time-domain (FDTD) simulation toolbox to virtually test common subwoofer configurations using tone bursts to judge waveform fidelity over a wide listening area. Temporal accuracy is compared to the steady-state frequency response to determine any link between the two domains. The simulated results are compared to practical measurements for validation.
Convention Paper 8715

4:00 pm

P6-5 Experiments of Sound Field Reproduction Inside Aircraft Cabin Mock-Up—*Philippe-Aubert Gauthier*,[1,2] *Cédric Camier*,[1,2] *Felix A. Lebel*,[1] *Y. Pasco*,[1,2] *Alain Berry*[1,2]
[1]Université de Sherbrooke, Sherbrooke, Quebec, Canada
[2]McGill University, Montreal, Quebec, Canada

Sound environment reproduction of various flight conditions in aircraft mock-ups is a valuable tool for the study, prediction, demonstration, and jury testing of interior aircraft sound quality and comfort. To provide a faithful reproduced sound environment, time, frequency, and spatial characteristics should be preserved. Physical sound field reproduction approaches for spatial sound reproduction are mandatory to immerse the listener body in the proper sound field so that localization cues are recreated. Vehicle mock-ups pose specific problems for sound field reproduction. Confined spaces, needs for invisible sound sources, and a singular acoustical environment make the use of open-loop sound field reproduction technologies not ideal. In this paper preliminary experiments in an aircraft mock-up with classical multichannel least-square methods are reported. The paper presents objective evaluations of reproduced sound fields. Promising results along with practical compromises are reported.
Convention Paper 8716

4:30 pm

P6-6 Wave Field Synthesis with Primary Source Correction: Theory, Simulation Results, and Comparison to Earlier Approaches—*Florian Völk, Hugo Fastl*, Technical University of Munich, Munich, Germany

Wave field synthesis (WFS) with primary source correction (PSC) extends earlier theoretical ➥

derivations by the correct synthesis of primary point sources at a reference point. In this paper the theory of WFS with PSC is revised with respect to other derivations, extended for the application to focus points and validated by numerical simulation. A comparison to earlier approaches to WFS concludes the paper.
Convention Paper 8717

5:00 pm

P6-7 Limitations of Point-Source Sub-Woofer Array Models for Live Sound—*Ambrose Thompson, Josebaitor Luzarraga Iturrioz, Phil Anthony*, Martin Audio, High Wycombe, UK

This paper examines the validity of applying simple models to the kind of highly configurable, low frequency arrays typically used for live sound. Measurements were performed on a single full-sized touring sub-woofer array element at different positions within a number of different array configurations. It was discovered that radiation was rarely omnidirectional and in some cases more than 20 dB from being so. Additionally, the in-situ polar response significantly differed from that obtained with the cabinet in isolation, the degree of difference (2–10 dB) was strongly dependent on array type and element position. For compact arrays we demonstrate, via the application of the "acoustic center" concept, that even when elemental radiation approaches omnidirectional behavior some array configurations are particularly susceptible to errors arising from commonly applied assumptions.
Convention Paper 8718

5:30 pm

P6-8 Improved Methods for Generating Focused Sources Using Circular Arrays—*Mark A. Poletti*, Industrial Research Limited, Lower Hutt, New Zealand

Circular loudspeaker arrays allow the reproduction of 2-D sound fields due to sources outside the loudspeaker radius. Sources inside the array can be approximated by focusing sound from a subset of the loudspeakers to a point. The resulting sound field produces the required divergence of wave fronts in a half-space beyond the focus point. This paper presents two new methods for generating focused sources that produce lower errors than previous approaches. The first derives an optimum window for limiting the range of active loudspeakers by matching the field to that of a monopole inside the source radius. The second applies pressure matching to a monopole source over a region where the wavefronts are diverging.
Convention Paper 8719

Tutorial 2 **Friday, October 26**
2:00 pm – 4:00 pm **Room 130**

FORENSIC DIGITAL DATA ANALYSIS

Presenter: **Doug Lacey**, DL Technology LLC, Fredericksburg, VA, USA

The analysis of digital audio is often required in forensics when the provenance of recorded material is in doubt or when carved material is recovered during investigations. Methods often involve spectral and statistical analyses to determine the nature of an audio recording but this tutorial will explore additional analyses of the data that composes a digital audio file expanding on a recent workshop given at the 46th International Conference of the AES on Audio Forensics. Following a short introduction and demonstration of the classical "hex editor" which allows the viewing of the raw contents of a file, interpretation of a file's structure, analysis of embedded metadata that otherwise may not be apparent, and real case examples will be shared.

Workshop 3 **Friday, October 26**
2:00 pm – 4:00 pm **Room 133**

WHAT EVERY SOUND ENGINEER SHOULD KNOW ABOUT THE VOICE

Chair: **Eddy B. Brixen**, EBB-consult, Smørum, Denmark

Panelists: *Henrik Kjelin*, Complete Vocal Institute, Denmark
Cathrine Sadolin, Complete Vocal Institute, Denmark

The purpose of this workshop is to teach sound engineers how to listen to the voice before they even think of microphone picking and knob-turning. The presentation and demonstrations are based on the "Complete Vocal Technique" (CVT) where the fundamental is the classification of all human voice sounds into one of four vocal modes named Neutral, Curbing, Overdrive, and Edge. The classification is used by professional singers within all musical styles and has in a period of 20 years proved easy to grasp in both real life situations and also in auditive and visual tests (sound examples and laryngeal images/Laryngograph waveforms). These vocal modes are found in the speaking voice as well. Cathrine Sadolin, the developer of CVT, will involve the audience in this workshop, while explaining and demonstrating how to work with the modes in practice to achieve any sound and solve many different voice problems like unintentional vocal breaks, too much or too little volume, hoarseness, and much more. The physical aspects of the voice will be explained and laryngograph waveforms and analyses will be demonstrated by Henrik Kjelin. Eddy Brixen will demonstrate measurements for the detection of the vocal modes and explain essential parameters in the recording chain, especially the microphone, to ensure reliable and natural recordings.

Game Audio Session 2 **Friday, October 26**
2:00 pm – 3:30 pm **Room 123**

**THE FUTURE IS NOW:
MIND CONTROLLED INTERACTIVE MUSIC**

Presenters: **Adam Gazzaley**, Neuroscience Imaging Center, UCSF
Jim Hedges, Zynga
Kyle Machulis, Nonpolynomial Labs
Nicolas Tomasino, IGN Entertainment
Richard Warp, Leapfrog

If one thing is clear from the games industry over the last 20 years, it is that consumers are seeking an ever-more immersive environment for their gaming experience, and in many ways biofeedback is the "final frontier," where a player's emotions, reactions, and mood can directly influence the gameplay. Whether the feedback comes from autonomic processes (stress or arousal, as in Galvanic Skin Response) or cognitive function (EEG signals from the brain), there is no doubt that these "active input" technologies, which differ from traditional HCI inputs (such as hardware controllers) in their singular correspondence to the individual player, greatly enhance the contextual responsiveness and "reality" of a game. These technologies are already robust enough to be integrated via audiovisual mappings into the immersive world of gaming. Things are about to get a lot more real.

Live Sound Seminar 2 Friday, October 26
2:00 pm – 3:30 pm Room 120

LIVE SOUND ENGINEERING:
THE JUXTAPOSITION OF ART AND SCIENCE

Presenter: **Chuck Knowledge**, Chucknology, Dublin, CA, USA

In this presentation we examine the different disciplines required for modern live event production. The technical foundations of audio engineering are no longer enough to deliver the experiences demanded by today's concert goers. This session will discuss practical engineering challenges with consideration for the subjective nature of art and the desire of performing artists to push the technological envelope. Focal points will include:

- Transplanting studio production to the live arena.
- Computer-based solutions and infrastructure requirements.
- The symbiosis with lighting and video.
- New technologies for interactivity and audience engagement.
- New career paths made possible by innovation in these fields.

Attendees can expect insight to the delivery of recent high-profile live events, the relevant enabling technologies, and how to develop their own skill set to remain at the cutting edge.

Student/Career Event
SC-1 OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Friday, October 26, 2:00 pm – 3:30 pm
Room 132

Chair: **Ezequiel Morfi**

Vice Chair: **Colin Pfund**

The first Student Delegate Assembly (SDA) meeting is the official opening of the Convention's student program and a great opportunity to meet with fellow students from all corners of the world. This opening meeting of the Student Delegate Assembly will introduce new events and election proceedings, announce candidates for the coming year's election for the North & Latin American Regions Vice Chair, announce the finalists in the four Student Recording Competition categories and the new Student Design Competition, and announce all upcoming student/education related events of the convention. Students and student sections will be given the opportunity to introduce themselves and their activities, in order to stimulate international contacts. The SDA leaders will then lead a dialogue to discuss important issues significant to all college audio students.

All students and educators are invited to participate in this meeting. Election results and Recording Competition and Design Competition Awards will be given at the Student Delegate Assembly Meeting–Part 2 on Monday, October 29.

Friday October 26 2:00 pm Room 124

Technical Committee Meeting on Perception and Subjective Evaluation of Audio Signals

Friday October 26 2:00 pm Room 125

AES Standards Committee Meeting SC-02-02 on Digital Input/Output Interfacing

Broadcast/Media Streaming Session 3
Friday, October 26 2:15 pm – 3:45 pm
Room 131

BROADCAST AUDIO NETWORK TECHNIQUES

Chair: **Dan Braverman**, Radio Systems

Panelists: *Tag Borland*, Logitek Electronic Systems Inc., Houston, TX, USA
Andreas Hildebrand, ALC NetworX GmbH
Kelly Parker, Wheatstone
Greg Shay, Telos Alliance/Axia, Cleveland, OH, USA

Broadcasting, especially in the studio arena has suffered mightily from lack of standards (or as the old joke goes; "from liking standards so much we created too many!"). Without any analogous MIDI-like control or serial protocol, integrating today's studio remains a science project. But audio over IP presents—and more aptly demands—a standard protocol if our new industry hardware and peripherals are literally going to communicate.

This session will overview the current implemented broadcast VOIP standards with an emphasis on interoperability challenging the participating manufacturers to reveal their plans, issues and hurtles in adapting and implementing a standard.

Session P7 Friday, October 26
3:00 pm – 4:30 pm Foyer

POSTERS: AMPLIFIERS, TRANSDUCERS, AND EQUIPMENT

3:00 pm

P7-1 Evaluation of t_{rr} Distorting Effects Reduction in DCI-NPC Multilevel Power Amplifiers by Using SiC Diodes and MOSFET Technologies
—*Vicent Sala, Tomas Resano, Jose Luis Romeral, Jose Manuel Moreno*, UPC-Universitat Politecnica de Catalunya, Terrassa, Catalunya, Spain

In the last decade, the Power Amplifier applications have used multilevel diode-clamped-inverter or neutral-point-clamped (DCI-NPC) topologies to present very low distortion at high power. In these applications a lot of research has been done in order to reduce the sources of distortion in the DCI-NPC topologies. One of the most important sources of distortion, and less studied, is the reverse recovery time (t_{rr}) of the clamp diodes and MOSFET parasitic diodes. Today, with the emergence of Silicon Carbide (SiC) technologies, these sources of ➥

distortion are minimized. This paper presents a comparative study and evaluation of the distortion generated by different combinations of diodes and MOSFETs with Si and SiC technologies in a DCI-NPC multilevel Power Amplifier in order to reduce the distortions generated by the non-idealities of the semiconductor devices.
Convention Paper 8720

3:00 pm

P7-2 New Strategy to Minimize Dead-Time Distortion in DCI-NPC Power Amplifiers Using COE-Error Injection—*Tomas Resano, Vicent Sala, Jose Luis Romeral, Jose Manuel Moreno*, UPC-Universitat Politecnica de Catalunya, Terrassa, Catalunya, Spain

The DCI-NPC topology has become one of the best options to optimize energy efficiency in the world of high power and high quality amplifiers. This can use an analog PWM modulator that is sensitive to generate distortion or error, mainly for two reasons: Carriers Amplitude Error (CAE) and Carriers Offset Error (COE). Other main error and distortion sources in the system is the Dead-Time (td). This is necessary to guarantee the proper operation of the power amplifier stage so that errors and distortions originated by it are unavoidable. This work proposes a negative COE generation to minimize the distortion effects of td. Simulation and experimental results validates this strategy.
Convention Paper 8721

3:00 pm

P7-3 Further Testing and Newer Methods in Evaluating Amplifiers for Induced Phase and Frequency Modulation via Tones, Amplitude Modulated Signals, and Pulsed Waveforms—*Ronald Quan*, Ron Quan Designs, Cupertino, CA, USA

This paper will present further investigations from AES Convention Paper 8194 that studied induced FM distortions in audio amplifiers. Amplitude modulated (AM) signals are used for investigating frequency shifts of the AM carrier signal with different modulation frequencies. A square-wave and sine-wave TIM test signal is used to evaluate FM distortions at the fundamental frequency and harmonics of the square-wave. Newer amplifiers are tested for FM distortion with a large level low frequency signal inducing FM distortion on a small level high frequency signal. In particular, amplifiers with low and higher open loop bandwidths are tested for differential phase and FM distortion as the frequency of the large level signal is increased from 1 kHz to 2 kHz.
Convention Paper 8722

3:00 pm

P7-4 Coupling Lumped and Boundary Element Methods Using Superposition—*Joerg Panzer*, R&D Team, Salgen, Germany

Both, the Lumped and the Boundary Element Method are powerful tools for simulating electroacoustic systems. Each one can have its preferred domain of application within one system to be modeled. For example the Lumped Element Method is practical for electronics, simple mechanics, and internal acoustics. The Boundary Element Method on the other hand enfolds its strength on acoustic-field calculations, such as diffraction, reflection, and radiation impedance problems. Coupling both methods allows to investigate the total system. This paper describes a method for fully coupling of the rigid body mode of the Lumped to the Boundary Element Method with the help of radiation self- and mutual radiation impedance components using the superposition principle. By this, the coupling approach features the convenient property of a high degree of independence of both domains. For example, one can modify parameters and even, to some extent, change the structure of the lumped-element network without the necessity to resolve the boundary element system. This paper gives the mathematical derivation and a demonstration-example, which compares calculation results versus measurement. In this example electronics and mechanics of the three involved loudspeakers are modeled with the help of the lumped element method. Waveguide, enclosure and radiation is modeled with the boundary element method.
Convention Paper 8723

3:00 pm

P7-5 Study of the Interaction between Radiating Systems in a Coaxial Loudspeaker—
Alejandro Espi,[1] *William A. Cárdenas,*[2] *Jose Martinez*,[1] *Jaime Ramis*,[2] *Jesus Carbajo*[2]
[1]Acustica Beyma S.L., Moncada (Valencia), Spain
[2]University of Alicante, Alicante, Spain

In this work the procedure followed to study the interaction between the mid and high frequency radiating systems of a coaxial loudspeaker is explained. For this purpose a numerical Finite Element model was implemented. In order to fit the model, an experimental prototype was built and a set of experimental measurements, electrical impedance, and pressure frequency response in an anechoic plane wave tube among these, were carried out. So as to take into account the displacement dependent nonlinearities, a different input voltage parametric analysis was performed and internal acoustic impedance was computed numerically in the frequency domain for specific phase plug geometries. Through inversely transforming to a time differential equation scheme, a lumped element equivalent circuit to evaluate the mutual acoustic load effect present in this type of acoustic coupled systems was obtained. Additionally, the crossover frequency range was analyzed using the Near Field Acoustic Holography technique.
Convention Paper 8724

3:00 pm

P7-6 Flexible Acoustic Transducer from Dielectric-Compound Elastomer Film—*Takehiro Sugimoto*,[1,2] *Kazuho Ono*,[1] *Akio Ando*,[1] *Hiroyuki Okubo*,[1] *Kentaro Nakamura*[2]
[1]NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

[2]Tokyo Institute of Technology, Midori-ku, Yokohama, Japan

To increase the sound pressure level of a flexible acoustic transducer from a dielectric elastomer film, this paper proposes compounding various kinds of dielectrics into a polyurethane elastomer, which is the base material of the transducer. The studied dielectric elastomer film utilizes a change in side length derived from the electrostriction for sound generation. The proposed method was conceived from the fact that the amount of dimensional change depends on the relative dielectric constant of the elastomer. Acoustical measurements demonstrated that the proposed method was effective because the sound pressure level increased by 6 dB at the maximum.
Convention Paper 8725

3:00 pm

P7-7 A Digitally Driven Speaker System Using Direct Digital Spread Spectrum Technology to Reduce EMI Noise—*Masayuki Yashiro, Mitsuhiro Iwaide, Akira Yasuda, Michitaka Yoshino, Kazuyki Yokota, Yugo Moriyasu, Kenji Sakuda, Fumiaki Nakashima*, Hosei University, Koganei, Tokyo, Japan

In this paper a novel digital direct-driven speaker for reducing electromagnetic interference incorporating a spread spectrum clock generator is proposed. The driving signal of a loudspeaker, which has a large spectrum at specific frequency, interferes with nearby equipment because the driving signal emits electromagnetic waves. The proposed method changes two clock frequencies according to the clock selection signal generated by a pseudo-noise circuit. The noise performance deterioration caused by the clock frequency switching can be reduced by the proposed modified delta-sigma modulator, which changes coefficients of the DSM according to the width of the clock period. The proposed method can reduce out-of-band noise by 10 dB compared to the conventional method.
Convention Paper 8726

3:00 pm

P7-8 Automatic Speaker Delay Adjustment System Using Wireless Audio Capability of ZigBee Networks—*Jaeho Choi, Myoung woo Nam, Kyogu Lee*, Seoul National University, Seoul, Korea

IEEE 802.15.4 (ZigBee) standard is a low data rate, low power consumption, low cost, flexible network system that uses wireless networking protocol for automation and remote control applications. This paper applied these characteristics on the wireless speaker delay compensation system in a large venue (over 500-seat hall). Traditionally delay adjustment has been manually done by sound engineers, but our suggested system will be able to analyze delayed-sound of front speaker to rear speaker automatically and apply appropriate delay time to rear speakers. This paper investigates the feasibility of adjusting the wireless speaker delay over the above-mentioned ZigBee network. We present an implementation of a ZigBee audio transmision and LBS (Location-Based Service) application that allows to calculation a speaker delay time.
Convention Paper 8727

3:00 pm

P7-9 A Second-Order Soundfield Microphone with Improved Polar Pattern Shape—*Eric M. Benjamin*, Surround Research, Pacifica, CA, USA

The soundfield microphone is a compact tetrahedral array of four figure-of-eight microphones yielding four coincident virtual microphones; one omnidirectional and three orthogonal pressure gradient microphones. As described by Gerzon, above a limiting frequency approximated by $f_c = \pi c/r$, the virtual microphones become progressively contaminated by higher-order spherical harmonics. To improve the high-frequency performance, either the array size must be substantially reduced or a new array geometry must be found. In the present work an array having nominally octahedral geometry is described. It samples the spherical harmonics in a natural way and yields horizontal virtual microphones up to second order having excellent horizontal polar patterns up to 20 kHz.
Convention Paper 8728

3:00 pm

P7-10 Period Deviation Tolerance Templates: A Novel Approach to Evaluation and Specification of Self-Synchronizing Audio Converters—*Francis Legray*,[1] *Thierry Heeb*,[2,3] *Sebastien Genevey*,[1] *Hugo Kuo*[1]
[1]Dolphin Integration, Meylan, France
[2]Digimath, Sainte-Croix, Switzerland
[3]SUPSI, ICIMSI, Manno, Switzerland

Self-synchronizing converters represent an elegant and cost effective solution for audio functionality integration into SoC (System-on-Chip) as they integrate both conversion and clock synchronization functionalities. Audio performance of such converters is, however, very dependent on the jitter rejection capabilities of the synchronization system. A methodology based on two period deviation tolerance templates is described for evaluating such synchronization solutions, prior to any silicon measurements. It is also a unique way for specifying expected performance of a synchronization system in the presence of jitter on the audio interface. The proposed methodology is applied to a self-synchronizing audio converter and its advantages are illustrated by both simulation and measurement results.
Convention Paper 8729

3:00 pm

P7-11 Loudspeaker Localization Based on Audio Watermarking—*Florian Kolbeck, Giovanni Del Galdo, Iwona Sobieraj, Tobias Bliem*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Localizing the positions of loudspeakers can be useful in a variety of applications, above all the calibration of a home theater setup. For this aim, several existing approaches employ a micro- ➥

phone array and specifically designed signals to be played back by the loudspeakers, such as sine sweeps or maximum length sequences. While these systems achieve good localization accuracy, they are unsuitable for those applications in which the end-user should not be made aware that the localization is taking place. This contribution proposes a system that fulfills these requirements by employing an inaudible watermark to carry out the localization. The watermark is specifically designed to work in reverberant environments. Results from realistic simulations confirm the practicability of the proposed system.
Convention Paper 8730

Friday October 26 3:00 pm Room 124

Technical Committee Meeting on Human Factors in Audio Systems

Special Event
AL SCHMITT & FRIENDS: AES ROUNDTABLE OF AWARD-WINNING ENGINEERS
Friday, October 26, 3:30 pm – 5:00 pm
Room 134

Moderator: **Al Schmitt**

Panelists: *Chuck Ainley* (Mark Knopfler, Dire Straits)
Ed Cherney (Bob Dylan, The Rolling Stones)
Frank Filipetti (Foreigner, KISS)
George Massenburg (Earth Wind & Fire, James Taylor)
Elliot Scheiner (Foo Fighters, Eric Clapton)

A roundtable discussion of modern recording techniques and trends will be hosted by 21-time Grammy-winning engineer and producer Al Schmitt. The panel includes some of the biggest names in audio production: Elliot Scheiner (Foo Fighters, Eric Clapton), Ed Cherney (Bob Dylan, The Rolling Stones), George Massenburg (Earth Wind & Fire, James Taylor), Chuck Ainley (Mark Knopfler, Dire Straits) and Frank Filipetti (Foreigner, KISS).

Topics covered at the roundtable will include:

- Understanding the differences of recording at home, in a project studio and in a larger traditional studio
- How to effectively record drums
- How and when to use effects
- Techniques for vocal thickening
- Obstacles that were encountered and resolved during the recording of classic albums
- Diplomacy in the studio with artists, executives and other stakeholders

The event will be sponsored by the Sennheiser Recording Sound Academy and is open to all AES attendees.

Tutorial 3 Friday, October 26
4:00 pm – 6:00 pm Room 132

SOCIAL MEDIA FOR ENGINEERS AND PRODUCERS

Presenter: **Bobby Owsinski**

Facebook, Google+, Twitter, and YouTube are important elements for developing a fan base or client list, but without the proper strategy they can prove ineffective and take so much time that there's none left for creating. This presentation shows engineers, producers, audio professionals, and musicians the best techniques and strategy to utilize social media as a promotional tool without it taking 20 hours a day.

Topics covered include:

- Your mailing list—old tech, new importance
- Social Media management strategies
- Optimizing your YouTube presence
- The secrets of viral videos
- The online video world is larger than you think
- Search Engine Optimization basics
- Using Facebook and Twitter for marketing
- The secret behind successful tweets
- Social media measurement techniques
- What's next?

Broadcast/Media Streaming Session 4
Friday, October 26 4:00 pm – 5:30 pm
Room 131

AUDIO ENCODING FOR STREAMING

Chair: **Fred Willard**, Univision

Panelists: *Casey Camba*, Dolby Labs, San Francisco, CA, USA
Robert Reams, Streaming Appliances/DSP Concepts, Mill Creek, WA, USA
Samuel Sousa, Triton Digital
Jason Thibeault, Limelight

This session will discuss various methods of encoding audio for streaming. What is most efficient for particular uses? Various bitrates and compression algorithms are used, what are the advantages, what are the disadvantages?

Game Audio Session 3 Friday, October 26
4:00 pm – 6:00 pm Room 133

HOW TO USE THE INTERACTIVE REVERBERATOR: THEORETICAL BASES AND PRACTICAL APPLICATIONS

Chair: **Steve Martz**, THX Ltd., San Rafael, CA, USA

Presenters: *Toshiki Hanyu*, Nihon University, Funabashi, Chiba, Japan
Takumi Higashi, CAPCOM Co. Ltd., Okaka-shi, Oasaka-fu, Japan
Tomoya Kishi, CAPCOM Co., Ltd., Okaka-shi, Oasaka-fu, Japan
Masataka Nakahara, ONFUTURE Ltd., Tokyo, Japan; SONA Corp., Tokyo, Japan

The interactive reverberator, applying realistic computed acoustic responses interactively for video game scenes, is a very important technology for in-game sound processing. The presenters have developed the interactive reverberator whose acoustical properties can be adjusted easily even after the calculated results are given. It has been already implemented into a Capcom's middleware, MT-framework and a trial run has been conducted successfully. How to setup initial parameters of the interactive reverberator? Do they need to be adjusted again by hearing impression? How to minimize difference in reverb timber between interactive scenes and cut scenes? The workshop introduces algorithm and basic functions of the interactive reverberator that the authors developed and also show practical operations of using it with 5.1ch run-time demos.

Live Sound Seminar 3 **Friday, October 26**
4:00 pm – 5:30 pm **Room 120**

PRACTICAL APPLICATION OF AUDIO NETWORKING FOR LIVE SOUND

Chair: **Kevin Kimmel**, Yamaha Commercial

Panelists: *Kari Eythorsson*, Optocore GmbH
Steve Seable, Yamaha Commercial
Rob Smoot, Walt Disney Company
Kieran Walsh, Audinate

This panel will focus on the use of several audio networking technologies, including A-Net, Dante, EtherSound, Cobranet, Optocore, Rocknet, and AVnu avb and their deployment in live sound applications. Network panelists will be industry professionals who have experience working with various network formats.

Networked Audio Session 1 **Friday, October 26**
4:00 pm – 6:00 pm **Room 123**

ERROR-TOLERANT AUDIO CODING

Chair: **David Trainor**, CSR, Belfast, Northern Ireland, UK

Panelists: *Bernhard Grill*, Fraunhofer IIS, Erlangen, Germany
Deepen Sinha, ATC Labs, Newark, NJ, USA
Gary Spittle, Dolby Laboratories, San Francisco, CA, USA

Two important and observable trends are the increasing delivery of real-time audio services over the Internet or cellular network and also the implementation of audio networking throughout a residence, office or studio using wireless technologies. This approach to distributing audio content is convenient, ubiquitous, and can be relatively inexpensive. However the nature of these networks is such that their capacity and reliability for real-time audio streaming can vary considerably with time and environment. Therefore error-tolerant audio coding techniques have an important role to play in maintaining audio quality for relevant applications. This workshop will discuss the capabilities of error-tolerant audio coding algorithms and recent advances in the state of the art.

Product Design Session 3 **Friday, October 26**
4:00 pm – 6:00 pm **Room 130**

DON'T MAKE YOUR PRODUCT A NOISE NIGHTMARE

Presenter: **Bill Whitlock**, Jensen Transformers, Inc., Chatsworth, CA, USA; Whitlock Consulting, Oxnard, CA, USA

Audio systems that operate on AC power will experience interference in the form of ground voltage differences, magnetic fields, and electric fields. Although, theoretically, balanced interfaces are immune to such interference, realizing high immunity in real-world, mass-produced equipment is not trivial. Designers who use ordinary balanced input stages fail to recognize the critical importance of very high common-mode impedances, which are the advantage of transformers. Because legacy test methods don't account for signal sources, most modern audio gear has poor immunity in real-world systems. The new IEC test for CMRR is explained as well as excellent alternatives to ordinary input stages. Other ubiquitous design errors, such as the "pin 1 problem" and the "power-line prima donna" syndrome are described as well as avoidance measures.

Friday October 26 **4:00 pm** **Room 124**

Technical Committee Meeting on Audio Forensics

Friday October 26 **5:00 pm** **Room 125**

AES Standards Committee Meeting SC-02-08 on Audio File Transfer and Exchange

Friday October 26 **5:30 pm** **Room 124**

Technical Committee Meeting on Network Audio Signals

Special Event
ORGAN CONCERT
Friday, October 26, 8:00 pm – 9:00 pm
St. Mark's Lutheran Church
1111 O'Farrell Street
San Francisco CA

Organist Graham Blyth's concerts are a highlight of every AES convention at which he performs. This year's concert is being held at St. Mark's Lutheran Church. The organ is a 2-manual and pedal Taylor & Boody, with a stop-list chosen to represent the North German Organs of the Baroque period. The program is going to be all Bach and will include the Dorian Toccata BWV538, the Six Schubler Chorales, selected Chorale Preludes from the Eighteen Chorales, Vivaldi/Bach Concerto in A minor, and the Prelude & Fugue in CBWV545.

Wind is supplied by three bellows mounted in a rack behind the organ's case. Although they are usually inflated by an electric blower, the organ sounds best when the wind is raised by foot pumping. Such a traditional wind system makes the organ's music more lifelike and endearing. This organ has a freestanding wooden case housing its pipes and playing mechanism, which in turn define its architectural form. The case and its carvings are made of solid black walnut. The most important pipes in the organ are given a place of prominence in the façade. They are made of tin and belong to the family of pipes known as the principals and produce the unique tone we associate with an organ. Other stops, belonging to flute, string, and reed families complement the principals with a variety of timbres and pitches. The interior metal pipes are made of alloys containing a higher percentage of lead than the façade principals. Several of the stops have pipes made of wood, which has been selected for its structural or tonal properties. Altogether there are 2,016 pipes disposed on two manuals and pedal. There are currently twenty-four stops, with space provided for five more. The instrument has a mechanical playing action and stop action. For the past two centuries it has been customary to tune all instruments in equal temperament, where each semitone is made an equal interval. By contrast, early music depended on unequal intervals for its vitality and beauty, exploiting varying relationships of consonance in different tonalities. The matter becomes critical when building an organ with brilliant mixture stops customary in the 18th century. For this instrument an unequal temperament particularly well suited to the works of J. S. Bach and his contemporaries was chosen.

Graham Blyth was born in 1948, began playing the piano aged 4 and received his early musical training as a Junior Exhibitioner at Trinity College of Music in London, England. Subsequently at Bristol University, where he read Electrical Engineering, he founded the University ➥

Music Society, conducting their Chamber Orchestra and Choir. He holds diplomas in Organ Performance from the Royal College of Organists, The Royal College of Music and Trinity College of Music.

In 1973 he founded Soundcraft with Phil Dudderidge, and has been Technical Director from the beginning. Soundcraft has since grown to be one of the most significant designers and manufacturers of audio mixing consoles in the world. In 1988, the company was bought by Harman, whose professional audio companies now include JBL, DBX, Lexicon, AKG and Studer.

In the late 1980s he renewed his musical studies with Sulemita Aronowsky for piano and Robert Munns for organ. He gives numerous concerts each year, principally as organist and pianist, but also as a conductor and harpsichord player. He made his international debut with an organ recital at St. Thomas Church, New York in 1993 and since then has given concerts on many of the finest organs in Europe, including the Madeleine and St. Etienne du Mont in Paris, and the Liebfrauen Dom in Munich, and in North America, including Grace Cathedral, San Francisco and St. Ignatius Loyola in New York.

He has lived in Wantage, Oxfordshire, since 1984 where he is currently Artistic Director of Wantage Chamber Concerts and Director of the Wantage Summer Festival. In 1995 he built the Challow Park Recital Hall, an 80 seat auditorium with completely variable acoustics, using the Lexicon LARES system, designed by his then Harman colleague David Griesinger. This allows for performance and recording of music ranging from String Quartets through to Organ Recitals.

Today he divides his time between audio engineering and organ design activities. In 2003 he founded the Veritas Organ Company to address the top end of the digital classical organ market, specialising in creating new organs by adding digital voices to existing pipes. He is a Fellow of the Royal Society of Arts and the Audio Engineering Society. He presented the Heyser Lecture at the Budapest Convention earlier this year.

Session P8 **Saturday, October 27**
9:00 am – 12:30 pm **Room 121**

EMERGING AUDIO TECHNOLOGIES

Chair: **Agnieszka Roginska**, New York University, New York, NY, USA

9:00 am

P8-1 A Method for Enhancement of Background Sounds in Forensic Audio Recordings—*Robert C. Maher*, Montana State University, Bozeman, MT, USA

A method for suppressing speech while retaining background sound is presented in this paper. The procedure is useful for audio forensics investigations in which a strong foreground sound source or conversation obscures subtle background sounds or utterances that may be important to the investigation. The procedure uses a sinusoidal speech model to represent the strong foreground signal and then performs a synchronous subtraction to isolate the background sounds that are not well-modeled as part of the speech signal, thereby enhancing the audibility of the background material.
Convention Paper 8731

9:30 am

P8-2 Transient Room Acoustics Using a 2.5 Dimensional Approach—*Patrick Macey*, Pacsys Ltd., Nottingham, UK

Cavity modes of a finite acoustic domain with rigid boundaries can be used to compute the transient response for a point source excitation. Previous work, considering steady state analysis, showed that for a room of constant height the 3-D modes can be computed very rapidly by computing the 2-D cross section modes. An alternative to a transient modal approach is suggested, using a trigonometric expansion of the pressure through the height. Both methods are much faster than 3-D FEM but the trigonometric series approach is more easily able to include realistic damping. The accuracy of approximating an "almost constant height" room to be constant height is investigated by example.
Convention Paper 8732

10:00 am

P8-3 Multimodal Information Management: Evaluation of Auditory and Haptic Cues for NextGen Communication Displays—
Durand Begault,[1] *Rachel M. Bittner*,[2] *Mark R. Anderson*[3]
[1]Human Systems Integration Division, NASA Ames Research Center, Moffett Field, CA, USA
[2]New York University, New York, NY, USA
[3]Dell Systems, NASA Ames Research Center, Moffett Field, CA, USA

Auditory communication displays within the NextGen data link system may use multiple synthetic speech messages replacing traditional air traffic control and company communications. The design of an interface for selecting among multiple incoming messages can impact both performance (time to select, audit, and release a message) and preference. Two design factors were evaluated: physical pressure-sensitive switches versus flat panel "virtual switches," and the presence or absence of auditory feedback from switch contact. Performance with stimuli using physical switches was 1.2 s faster than virtual switches (2.0 s vs. 3.2 s); auditory feedback provided a 0.54 s performance advantage (2.33 s vs. 2.87 s). There was no interaction between these variables. Preference data were highly correlated with performance.
Convention Paper 8733

10:30 am

P8-4 Prototype Spatial Auditory Display for Remote Planetary Exploration—*Elizabeth M. Wenzel*,[1] *Martine Godfroy*,[2] *Joel D. Miller*[3]
[1]NASA Ames Research Center, Moffett Field, CA, USA
[2]San Jose State University Foundation, NASA Ames Research Center, Moffett Field, CA, USA
[3]Dell Systems, NASA Ames Research Center, Moffett Field, CA, USA

During Extra-Vehicular Activities (EVA), astronauts must maintain situational awareness (SA) of a number of spatially distributed "entities" such as other team members (human and robotic), rovers, and a lander/habitat or other safe havens. These

entities are often outside the immediate field of view and visual resources are needed for other task demands. Recent work at NASA Ames has focused on experimental evaluation of a spatial audio augmented-reality display for tele-robotic planetary exploration on Mars. Studies compared response time and accuracy performance with different types of displays for aiding orientation during exploration: a spatial auditory orientation aid, a 2-D visual orientation aid, and a combined auditory-visual orientation aid under a number of degraded vs. nondegraded visual conditions. The data support the hypothesis that the presence of spatial auditory cueing enhances performance compared to a 2-D visual aid, particularly under degraded visual conditions.
Convention Paper 8734

11:00 am

P8-5 The Influence of 2-D and 3-D Video Playback on the Perceived Quality of Spatial Audio Rendering for Headphones—*Amir Iljazovic, Florian Leschka, Bernhard Neugebauer, Jan Plogsties*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Algorithms for processing of spatial audio are becoming more attractive for practical applications as multichannel formats and processing power on playback devices enable more advanced rendering techniques. In this study the influence of the visual context on the perceived audio quality is investigated. Three groups of 15 listeners are presented to audio-only, audio with 2-D video, and audio with 3-D video content. The 5.1 channel audio material is processed for headphones using different commercial spatial rendering techniques. Results indicate that a preference for spatial audio processing over a downmix to conventional stereo can be shown with the effect being larger in the presence of 3-D video content. Also, the influence of video on perceived audio quality is significant for 2-D and 3-D video presentation.
Convention Paper 8735

11:30 am

P8-6 An Autonomous System for Multitrack Stereo Pan Positioning—*Stuart Mansbridge*,[1] *Saorise Finn*,[1,2] *Joshua D. Reiss*[1]
[1]Queen Mary University of London, London, UK
[2]Birmingham City University, Birmingham, UK

A real-time system for automating stereo panning positions for a multitrack mix is presented. Real-time feature extraction of loudness and frequency content, constrained rules, and cross-adaptive processing are used to emulate the decisions of a sound engineer, and pan positions are updated continuously to provide spectral and spatial balance with changes in the active tracks. As such, the system is designed to be highly versatile and suitable for a wide number of applications, including both live sound and post-production. A real-time, multitrack C++ VST plug-in version has been developed. A detailed evaluation of the system is given, where formal listening tests compare the system against professional and amateur mixes from a variety of genres.
Convention Paper 8736

12:00 noon

P8-7 DReaM: A Novel System for Joint Source Separation and Multitrack Coding—*Sylvain Marchand*,[1] *Roland Badeau*,[2] *Cléo Baras*,[3] *Laurent Daudet*,[4] *Dominique Fourer*,[5] *Laurent Girin*,[3] *Stanislaw Gorlow*,[5] *Antoine Liutkus*,[2] *Jonathan Pinel*,[3] *Gaël Richard*,[2] *Nicolas Sturmel*,[3] *Shuhua Zang*[3]
[1]University of Western Brittany, Brest, France
[2]Telecom ParisTech, Paris, France
[3]GIPSA-Lab, Grenoble, France
[4]University Paris Diderot, Paris, France
[5]University Bordeaux, Talence, France

Active listening consists in interacting with the music playing, has numerous applications from pedagogy to gaming, and involves advanced remixing processes such as generalized karaoke or respatialization. To get this new freedom, one might use the individual tracks that compose the mix. While multitrack formats lose backward compatibility with popular stereo formats and increase the file size, classic source separation from the stereo mix is not of sufficient quality. We propose a coder / decoder scheme for informed source separation. The coder determines the information necessary to recover the tracks and embeds it inaudibly in the mix, which is stereo and has a size comparable to the original. The decoder enhances the source separation with this information, enabling active listening.
Convention Paper 8737

Session P9 Saturday, October 27
9:00 am – 12:00 noon Room 122

AUDITORY PERCEPTION

Chair: **Scott Norcross**, Dolby, San Francisco, CA, USA

9:00 am

P9-1 Subjective Evaluation of Personalized Equalization Curves in Music—*Weidong Shen*,[1] *Tiffany Chua*,[2] *Kelly Reavis*,[3] *Hongmei Xia*,[4] *Duo Zhang*,[5] *Gerald A. Maguire*,[2] *David Franklin*,[2] *Vincent Liu*,[6] *Wei Hou*,[7] *Hung Tran*[8]
[1]The Institute of Otolaryngology, Department of Otolaryngology, PLA General Hospital, Beijing, China
[2]University of California, Irvine, Irvine, CA, USA
[3]Portland Veterans Affairs Medical Center, Portland, OR, USA
[4]Hubei Zhong Shan Hospital, Wuhan, Hubei, China
[5]MaxLinear Inc.,- Carlsbad, CA, USA
[6]Logitech, Irvine, CA, USA
[7]Huawei Technologies Co., Ltd., Shenzhen, Guangdong, China
[8]AuralWare LLC, Rancho Santa Margarita, CA, USA

This paper investigated the subjective quality of equalized music in which equalization (EQ) curves were tailored to the individuals' preferences. Listeners subjectively rated a number of pre-selected psychoacoustic-based EQ curves over three test sessions. The personalized EQ curve was the curve that had the highest rating ➥

among the pool of pre-selected equalization curves. Listeners were instructed to rate music quality according to the ITU-R BS 1284 scale. Statistical analysis showed that listeners consistently rated music to which personalized EQ curves were applied significantly higher than the original CD-quality music.
Convention Paper 8738

9:30 am

P9-2 Thresholds for the Discrimination of Tonal and Narrowband Noise Bursts—*Armin Taghipour, Bernd Edler, Masoumeh Amirpour, Jürgen Herre*, International Audiolabs Erlangen, Erlangen, Germany

Several psychoacoustic models used in perceptual audio coding take into account the difference in masking effects of tones and narrowband noise and therefore incorporate some kind of tonality estimation. For possible optimization of these estimators for time-varying signals it is desirable to know the duration above which the auditory system is able to discriminate between tone bursts and narrowband noise bursts. There is some knowledge that this duration is frequency, bandwidth, and loudness dependent, but up to now no systematic studies have been performed. Therefore this paper presents the setup and the results of experiments for determining frequency dependent thresholds.
Convention Paper 8739

10:00 am

P9-3 Identification and Evaluation of Target Curves for Headphones—*Felix Fleischmann, Andreas Silzle, Jan Plogsties*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Generally, loudspeakers are designed to have a flat frequency response. For headphones there is no consensus about the optimal transfer function and equalization. In this study several equalization strategies were tested on commercially available headphones. The headphones were measured on an artificial head and equalization filters were designed in the frequency domain consisting of two parts: The first part of the filter is specific to each headphone, flattening the magnitude response of the headphone at the entrance of the blocked ear-canal. The second part of the filter is generic target curve for headphones. Different target curves were tested on the three headphones during a formal listening test using binaural signals. A target curve designed by expert listeners comparing loudspeaker with binaural headphone reproduction was preferred.
Convention Paper 8740

10:30 am

P9-4 Consistency of Balance Preferences in Three Musical Genres—*Richard King,[1,2] Brett Leonard,[1,2] Grzegorz Sikora[3]*
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
[3]Bang & Olufsen Deutschland GmbH, Pullach, Germany

Balancing the level of different sound sources is the most basic task performed in the process of mixing. While this task forms a basic building block of music mixing, very little research has been conducted to objectively study mixing habits and balance preferences. In this study data is collected from 15 highly-trained subjects performing simple mixing tasks on multiple musical excerpts spanning three musical genres. Balance preference is examined over musical genre, and the results exhibit more narrow variances in balance for certain genres over others.
Convention Paper 8741

11:00 am

P9-5 The Effect of Acoustic Environment on Reverberation Level Preference—*Brett Leonard,[1,2] Richard King,[1,2] Grzegorz Sikora[3]*
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
[3]Bang & Olufsen Deutschland GmbH, Pullach, Germany

Reverberation plays a very important role in modern music production. The available literature is minimal concerning the interaction between reverberation preference and the listening environment used during critical balancing tasks. Highly trained subjects are tasked with adding reverberation to a fixed stereo mix in two different environments: a standard studio control room and a highly reflective mix room. Distributions of level preference are shown to be narrower for more reflective mixing environments, and the mean level is below those set in a less reverberant environment.
Convention Paper 8742

11:30 am

P9-6 Localization of a Virtual Point Source within the Listening Area for Wave Field Synthesis—*Hagen Wierstorf,[1] Alexander Raake,[1] Sascha Spors[2]*
[1]Technische Universität Berlin, Berlin, Germany
[2]Universität Rostock, Rostock, Germany

One of the main advantages of Wave Field Synthesis (WFS) is the existence of an extended listening area contrary to the sweet spot in stereophony. At the moment there is little literature available on the actual localization properties of WFS at different points in the listening area. One reason is the difficulty to place different subjects reliably at different positions. This study systematically investigates the localization performance for WFS at many positions within the listening area. To overcome the difficulty to place subjects, the different listening positions and loudspeaker arrays were simulated by dynamic binaural synthesis. In a pre-study it was verified that this method is suitable to investigate the localization performance in WFS.
Convention Paper 8743

Tutorial 4 **Saturday, October 27**
9:00 am – 11:00 am **Room 132**

SMALL ROOM ACOUSTICS

Presenter: **Ben Kok**, BEN KOK acoustic consulting, Uden, The Netherlands

Acoustic basics of small rooms will be discussed. Specific issues related to the size of the room (room-modes) will be addressed. Absorption, reflection, diffraction, diffusion, and how to use it, as well as specific aspects regarding low frequency treatment will be discussed.

Although this will not be a studio design class, specifics and differences of recording rooms and control rooms will be identified, including considerations for loudspeaker and microphone placement.

Workshop 4 **Saturday, October 27**
9:00 am – 11:00 am **Room 133**

WHAT DOES AN OBJECT SOUND LIKE?
TOWARD A COMMON DEFINITION
OF A SPATIAL AUDIO OBJECT

Chair: **Frank Melchior**, BBC R&D, Salford, UK

Panelists: *Jürgen Herre*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Jean Marc Jot, DTS, USA
Nicolas Tsingos, Dolby Laboratories, San Francisco, CA, USA
Matte Wagner, Red Storm Entertainment
Hagen Wiersdorf, Technische Universität Berlin, Berlin, Germany

At the present time, several concepts for the storage of spatial audio data are under discussion in the research community. Besides the distribution of audio signals corresponding to a specific speaker layout or encoding a spatial audio scene in orthogonal basis functions like spherical harmonics, several solutions available on the market are applying object-based formats to store and distribute spatial audio scenes. The workshop will cover the similarities and difference between the various concepts of audio objects. This comparison will include the production and reproduction of audio objects as well as their storage. The panelists will try to find a common definition of audio objects in order to enable an object-based exchange format in the future.

Broadcast/Media Streaming Session 5
Saturday, October 27 **9:00 am – 10:30 am**
Room 131

STREAM DISTRIBUTION:
IP IN THE MOBILE ENVIRONMENT

Chair: **David Layer**, NAB

Panelists: *Mike Daskalopoulos*, Dolby Labs, San Francisco, CA, USA
Samuel Sousa, Triton Digital
Jason Thibeault, Limelight

The public has demanded the portability of stream listening, whether in a handheld or other mobile devices including the car. There are a variety of streaming technologies in the marketplace that can support portable streaming, and in this session representatives from three of the leading companies in this space will offer their insights and vision. Some of the specific topics to be covered include: Audio on mobile, the list of challenges; how mobile streaming can interact with traditional in-car listening; HTML5—savior or just more trouble?; and challenges in IP streaming and advertising interaction.

Game Audio Session 4 **Saturday, October 27**
9:00 am – 11:00 am **Room 123**

EDUCATION PANEL—NEW MODELS FOR GAME AUDIO EDUCATION IN THE 21ST CENTURY

Chair: **Steve Horowitz**, The Code International Inc., MPA

Panelists: *Matt Donner*, Pyramind, San Francisco, CA, USA
Steve Horelick, macProVideo
Scott Looney, Academy of Art University, San Francisco, CA, USA
Stephan Schütze, Composer / Sound Designer, Sound Librarian/FMOD
Michael Sweet, Berklee College of Music, Boston, MA, USA

Steve Horelick from macProVideo will run down the new ways that the internet and social media are changing the face of game audio education.

Formal Game Audio education programs are just starting to take root and sprout up all across the country and the world. From full fledged degree programs, 1 year certificate programs, to single class offerings, the word on the street is out and game audio education is becoming a hot topic and a big money-maker for schools. This panel brings together department heads from some of the country's top public and private institutions to discuss the current landscape and offerings in audio for interactive media education. Students looking to find the right institute will get a fantastic overview of what is out there and available. This is a must for students who are trying to decide what programs are right for them as they weigh their options for getting a solid education in sound and music for games and interactive media.

Product Design Session 4 **Saturday, October 27**
9:00 am – 10:00 am **Room 120**

AUDIO IN HTML 5

Presenters: **Jeff Essex**, AudioSyncrasy
Jory K. Prum, studio.jory.org, Fairfax, CA, USA

HTML 5 is coming. Many expect it to supplant Flash as an online rich media player, as Apple has made abundantly clear. But audio support is slow in coming, and there are currently marked differences between browsers. From an audio content standpoint, it's the Nineties all over again. The W3C's Audio Working Group is developing standards, but this is a fast-moving target. This talk will provide an update on what's working, what isn't.

Sound for Pictures 1 **Saturday, October 27**
9:00 am – 11:00 am **Room 130**

POST-PRODUCTION AUDIO TECHNIQUES
FOR DIGITAL CINEMA AND ANCILLARY MARKETS

Chair: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia

➥

Panelists: *Lon Bender*, Soundelux
Jason LaRocca, La-Rocc-A-Fella, Inc., Los Angeles, CA, USA
Brian A. Vessa, Sony Pictures Entertainment, Culver City, CA, USA

Cinema sound has traditionally been limited in fidelity because optical soundtracks, used until recently, were incapable of delivering full-bandwidth audio to the theaters. As Digital Cinema replaces film in distribution, sound mixers are now delivering uncompressed lossless tracks to the audience. These three top sound mixers will discuss the changes and challenges this has presented to them.

Saturday October 27 9:00 am Room 124

Technical Committee Meeting on Transmission and Broadcasting

Session EB1 Saturday, October 27
10:00 am – 11:30 am Foyer

POSTERS 1

10:00 am

EB1-1 Accuracy of ITU-R BS.1770 Algorithm in Evaluating Multitrack Material—*Pedro Duarte Pestana*,[1] *Alvaro Barbosa*[2]
[1]CITAR-Portuguese Catholic University, Porto, Portugal
[2](CEAUL)-Universidade Lisboa, Lisbon, Portugal

Loudness measurement that is computationally efficient and applicable on digital material disregarding listening level is a very important feature for automatic mixing. Recent work in broadcast specifications of loudness (ITU-R BS.1770) deserved broad acceptance and seems a likely candidate for extension to multitrack material, though the original design did not bear in mind this kind of development. Some empirical observations have suggested that certain types of individual source materials are not evaluated properly by the ITU's algorithm. In this paper a subjective test is presented that tries to shed some light on the subject.
Engineering Brief 53

10:00 am

EB1-2 An Online Resource for the Subjective Comparison of Vocal Microphones—*Bradford Swanson*, University of Massachusetts - Lowell, Lowell, MA, USA

Forty-eight microphones were gathered into small groups and tested using four vocalists (two male, two female). The recorded results are collected online so users may subjectively compare a single performance on closely related microphones.
Engineering Brief 54

10:00 am

EB1-3 Perception of Distance and the Effect on Sound Recording Distance Suitability for a 3-D or 2-D Image—*Luiz Fernando Kruszielski, Toru Kamekawa, Atsushi Marui*, Tokyo University of the Arts, Adachi-ku, Tokyo, Japan

Possible differences in the perception of the sound caused by 3-D image are still unclear. The aim of this research is to understand a possible difference in the perception of distance caused by interaction of sound and 3-D image compared to a 2-D image and also how this could affect the suitability of the sound recording distance. Using a 3-D setup, a saxophone player was recorded at five different distances. The subjects where asked to judge their subjective distance and also the suitable sound for the presented image. The results show that one group perceived 3-D to be more distant, however it did not change the sound suitability compared 3-D to 2-D.
Engineering Brief 55

10:00 am

EB1-4 Modeling Auditory-Haptic Interface Cues from an Analog Multi-line Telephone—
Durand Begault,[1] *Mark R. Anderson*,[2] *Rachel M. Bittner*[3]
[1]Human Systems Integration Division, NASA Ames Research Center, Moffett Field, CA, USA
[2]Dell Systems, NASA Ames Research Center, Moffett Field, CA, USA
[3]New York University, New York, NY, USA

The Western Electric company produced influential multi-line telephone designs during the 1940s–1970s using a six-button interface (line selection, hold button, intercom). Its simplicity was an example of a successful human factors design. Unlike touchscreen or membrane switches used in its modern equivalents, the older multi-line telephone used raised surface mechanical buttons that provided robust tactile, haptic, and auditory cues. This multi-line telephone was used as a model for a trade study comparison of two interfaces: a touchscreen interface (iPad) versus a pressure-sensitive strain gauge button interface (Phidget USB interface controllers). This engineering brief describes how the interface logic and the visual and auditory cues of the original telephone were analyzed and then synthesized using MAX-MSP. (The experiment and results are detailed in the authors' AES 133rd convention paper "Multimodal Information Management: Evaluation of Auditory and Haptic Cues for NextGen Communication Displays").
Engineering Brief 56

10:00 am

EB1-5 Tailoring Practice Room Acoustics to Student Needs—*Scott R. Burgess*, Central Michigan University, Mt. Pleasant, MI, USA

A crucial part of any music education facility is the student practice rooms. While these rooms typically vary in size, the acoustic treatments often take a "one size fits all" approach. This can lead to student dissatisfaction and a lack of rooms that are suitable to some instruments. The School of Music at Central Michigan University surveyed our students and created a variety of acoustic environments based on the results of this survey. This presentation will discuss this process and the results of the follow-up survey, which indicates an improvement in student satisfaction, along with suggestions for further study.
Engineering Brief 57

10:00 am

EB1-6 Acoustic Properties of Small Practice Rooms Where Musicians Can Practice Contentedly: Effect of Reverberation on Practice—*Ritsuko Tsuchikura*,[1] *Masataka Nakahara*,[1] *Takashi Mikami*,[1] *Toru Kamekawa*,[2] *Atsushi Marui*[2]
[1]SONA Corp., Nakano-ku, Tokyo, Japan
[2]Tokyo University of the Arts, Tokyo, Japan

This paper describes results of study on practice room acoustics regarding the level of satisfaction players feel about the acoustical conditions. Two different factors are found to be involved for musicians to evaluate the acoustics of practice rooms: "comfort in practice" and "comfort in performance." Further evaluation of the two factors shows that "comfort in practice" and "comfort in performance" have different desired reverberation times. The average absorption coefficients, therefore, are estimated. Though the experiments were carried out on several kinds of instruments, this paper describes the results of experiments involving trumpeters and violinists.
Engineering Brief 58

10:00 am

EB1-7 Bellamy Baffle Array: A Multichannel Recording Technique to Improve Listener Envelopment—*Steven Bellamy*, Humber College, Toronto, ON, Canada

The paper outlines a 6-microphone technique that makes use of a baffle between front and rear arrays. This addresses three common challenges in multichannel recording for 5.1 channel playback. First, to improve the sense of connectedness between LS/RS, L/LS and R/RS channel pairs. Second, to maintain clarity of the direct sound while allowing for strong levels of room sound in the mix. Third, to provide a flexible system that can work well with a variety of ensembles. The result is a flexible microphone technique that results in recordings of increased clarity and envelopment.
Engineering Brief 59

Audio Industry Seminar **Saturday, October 27**
10:00 am – 11:00 am **Room 114**

PMC: MASTERS OF AUDIO SERIES

The Art of Sound Design for Trailers and Games

Presenter: **Charles Deenen**, Electronic Arts

Charles will gives us a demonstration on "Sound Design for Trailers & Games." Sound design is one of most important and fun aspects of audio post and Charles will be sharing his wealth of experience and knowledge on the subject.

In a career that has spanned more than 25 years, Charles has leant his hand to over 200 games, numerous films, and almost a hundred (film) trailers. He's worked on sound design for many feature films including two Fast and Furious films, among others. This work translated into a passion for fast cars, loud sound, and the ultimate pursuit of emotionally engaging audio. In 2003 he started at Electronic Arts to oversee audio on the Racing franchises. Alongside long-format sound design, Charles continues to contribute to Hollywood's trailer advertising arm. After having been a Sr. Audio Director at Electronic Arts until 2011, he's now working on Trailer Design at EA, expanding his horizon.

Project Studio Expo **Saturday, October 27**
10:00 am – 11:00 am **PSE Stage**

FOCAL PRESS AUTHOR PANEL – CAN PROJECT STUDIOS REALLY GET PROFESSIONAL RESULTS?

Presenters: **Kyle P. Snyder,** Ohio University, School of Media Arts & Studies, Athens, OH, USA
David Miles Huber
Jay Kadis, CCRMA, Stanford University, Stanford, CA, USA
William Moylan, University of Massachusetts - Lowell, Lowell, MA, USA
Will Pirkle, University of Miami, Coral Gables, FL, USA
Mike Senior, Sound On Sound, Munich, Germany; Cambridge Music Technology

Join Focal Press authors for a discussion as to whether project studios really can get professional results. Panelists include Mike Senior, *Mixing Secrets for the Small Studio*; Will Pirkle, *Designing Audio Effect Plugins in C++*; William Moylan, *Understanding and Crafting the Mix*; and Jay Kadis, *The Science of Sound Recording*.

Saturday October 27 **10:00 am** **Room 124**

Technical Committee Meeting on Automotive Audio

Live Sound Seminar 4 **Saturday, October 27**
10:15 am – 11:45 am **Room 120**

TECHNICAL AND PRACTICAL CONSIDERATIONS FOR WIRELESS MICROPHONE SYSTEM DESIGNERS AND USERS

Chair: **Karl Winkler**, Lectrosonics

Panelists: *Joe Ciaudelli*, Sennheiser Electronic Corporation
Gino Sigismondii, Shure, Inc.
Tom Turkington, CoachComm LLC

Wireless microphone users and system designers encounter many of the same issues when setting up and using these systems, whether for house of worship, touring music, theater, corporate AV, or TV/video production. Channel counts from 1 to 24 are within the scope of this discussion. Topics will include RF Spectrum allocation, "safe haven" channels, TVBD database registration, frequency coordination, system design, transmitter and receiver antenna placement, and emerging wireless microphone technologies.

Audio Industry Seminar **Saturday, October 27**
10:30 am – 11:00 am **Room 112**

THE CHINA AUDIO & VIDEO ENGINEERING GROUP

Discussion on Speech Intelligibility Improved by Electroacoustic Systems.

Workshop 2 **Friday, October 26**
10:45 am – 12:30 pm **Room 133**

SPATIAL AUDIO EVALUATION

Chair: **Sean Olive**, Harman International, Northridge, CA, USA

Panelists: *Poppy Crum*, Dolby, San Francisco, CA, USA
Martin Dewhirst, University of Surrey, Guildford, Surrey, UK
Ville Pulkki, Aalto University, Espoo, Finland
Thomas Sporer, Fraunhofer IDMT, Ilmenau, Germany

While there are an increasing number of audio technologies aimed at improving the spatial rendering of live and recorded sounds in homes, performance spaces, cinemas, automobiles, and personal audio systems, there currently exists no standardized or recommended methods for evaluating their perceived spatial quality. This workshop will provide an overview of some current and future best-practices for subjective evaluation of spatial aspects of sound reproduction with the goal of helping promote methods that provide more accurate, reliable, and meaningful data.

Broadcast/Media Streaming Session 6
Saturday, October 27 10:45 am – 12:15 pm
Room 131

AUDIO FOR MOBILE TELEVISION

Chair: **Brad Dick**, Broadcast Engineering Magazine

Panelists: *Tim Carroll,* Linear Accoustics
David Layer, NAB
Robert Murch, Fox Television
Geir Skaaden, DTS, Inc.
Jim Starzynski, NBC Universal
Dave Wilson, CEA

Many TV stations recognize mobile DTV as a new and great financial opportunity. By simply simulcasting their main channel an entirely new revenue stream can be developed. But according to audio professionals, TV audio engineers should consider carefully the additional audio processing required to ensure proper loudness and intelligibility in a mobile device's typically noisy environment. The proper solution may be more complex that just reducing dynamic range or adding high-pass filtering. This panel of audio experts will provide in-depth guidance on steps that may be taken to maximize the performance of your mobile DTV channel.

Tutorial 5 Saturday, October 27
11:00 am – 1:00 pm Room 132

GETTING THE SOUND OUT OF (AND INTO) YOUR HEAD: THE PRACTICAL ACOUSTICS OF HEADSETS

Presenter: **Christopher Struck**, CJS Labs, San Francisco, CA, USA

The electroacoustics of headsets and other head-worn devices are presented. The Insertion Gain concept is reviewed and appropriate free and diffuse field target responses are detailed. The selection and use of appropriate instrumentation, including ear and mouth simulators, and test manikins appropriate for head-worn devices are described. Boom, close-talking, and noise-canceling microphone tests are shown and practical methods are discussed for obtaining consistent data. Relevant standards are reviewed, typical measurement examples are described, and applications of these methods to analog, USB, and Bluetooth wireless devices are explained.

Workshop 5 Saturday, October 27
11:00 am – 1:00 pm Room 133

LOUDNESS WARS: THE WRONG DRUG?

Chair: **Thomas Lund**, TC Electronic A/S, Risskov, Denmark

Panelists: *John Atkinson*, Stereophile Magazine, New York, NY, USA
Florian Camerer, ORF, Vienna, Austria
Bob Katz, Digital Domain, Orlando, FL, USA
George Massenburg, McGill University, Montreal, Quebec, Canada

Newly produced pop/rock music rarely sounds good on fine loudspeakers. Could it be that the wrong mastering drug has been used for decades, affecting Peak to Average Ratio instead of Loudness Range? With grim side effects all around—and years of our music heritage irreversibly harmed—the panel provides a new status on the loudness wars and sets out to investigate the difference between the two from a technical, a perceptual, and a practical point of view. In a normalized world, bad drugs will no longer be compensated by a benefit of being loud. Learn to distinguish between a cure and quack practice, and save your next album.

Game Audio Session 5 Saturday, October 27
11:00 am – 1:00 pm Room 123

CAREERS PANEL—
GETTING A JOB IN THE GAME INDUSTRY

Chair: **Steve Horowitz**, The Code International Inc., MPA

Panelists: *Charles Deenen*, Electronic Arts
Jesse Harlin, LucasArts
Adam Levenson, Levenson Artists
Richard Warp, Leapfrog

From AAA titles to social media, the game industry offers a lot of opportunity for the audio practitioner. In this event our panel will break down the current state of the industry.

Everyone wants to work in games, just check out the news. The game industry is larger then the film industry and the growth curve keeps going up and up and up. So, what is the best way to get that first gig in audio for games? How can I transfer my existing skills to interactive media? Should I go to school? What are the pros and cons of a degree program versus just getting out there on my own? Good questions! We will take a panel of today's top creative professionals from large game studios to Indie producers and ask them what they think you need to know when looking for work in the game industry. So, whether you are already working in the game industry or just thinking of the best way to transfer your skills from film, TV or general music production to interactive media or a complete newbie to the industry, this panel is a must!

Sound for Pictures 2 Saturday, October 27
11:00 am – 1:00 pm Room 130

RECONSIDERING STANDARDS FOR CINEMA SOUND—ALTERNATIVES TO ISO 2969

Chair: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia

Panelists: *Keith Holland*, University of Southampton, Southampton, UK
Floyd Toole, Acoustical consultant to Harman, ex. Harman VP Acoustical Engineering, Oak Park, CA, USA

For over eighty years ISO 2969 (aka SMPTE S202) has been a cornerstone of the film sound reproduction "B-Chain." Like the RIAA curve, it was originally implemented to compensate for defects in the delivery medium. Groundbreaking acoustical research, led by Philip Newell and Dr. Keith Holland, has exposed shortcomings in the standard as well as the testing methodology used in the standard. This panel will examine the implications of these standards as film rapidly shifts to Digital Cinema delivery, as mixing rooms being used become smaller and as full bandwidth soundtracks and newer formats like 3-D audio are now delivered to theaters for reproduction.

Audio Industry Seminar **Saturday, October 27**
11:00 am – 12:00 noon **Room 112**

TAGAI EX-PORT LTD.

Solitone 1: Introduction, World Premiere! Revolution of Audio! What Is Solitone?

Presenter: **Péter Tagai**, Tagai EX-Port Ltd.

Solitone is a revolutionary technology. If you want to be a witness of the revolution of audio, you must be there! Answers for: AC or DC, analog or digital, balanced or unbalanced, grounding or non grounding, PC, DAW or something else, software or hardware, the origin of the hum, buzz, and noise in any system. Audio-related problems will be explained and solved by the Solitone technology. Also, a brief overview of the whole audio chain through Solitone, starting with AC power, mics, and up to the speaker drivers.

Project Studio Expo **Saturday, October 27**
11:00 am – 12:00 noon **PSE Stage**

PSE1 - IT WON'T SOUND RIGHT IF YOU DON'T HEAR IT RIGHT: STUDIO ACOUSTICS, MONITORING, AND CRITICAL LISTENING

Presenter: **Hugh Robjohns**, Technical Editor, Sound on Sound, Crowle, UK

The monitoring environment acoustics and the monitoring loudspeakers are critical links in every music production chain. Any weaknesses impact negatively not only on the overall quality of mixes but also on the confidence and ability of the user to assess and process audio material efficiently and effectively. This workshop examines the theoretical requirements and practical optimization of high-quality monitoring systems for home and project studios, drawing on the author's experiences in the "Studio SOS" series published in *Sound On Sound* magazine. It will also explore choosing new monitoring loudspeakers, optimizing control room acoustics, and honing critical listening skills.

Saturday October 27 **11:00 am** **Room 124**

Technical Committee Meeting on Spatial Audio

Special Event
LEGENDARY ARTISTS: SOUNDS OF SAN FRANCISCO
Saturday, October 27, 11:15 am – 12:30 pm
Room 134

Moderator: **Mr. Bonzai**

Panelists: Peter Albin (Big Brother and the Holding Company)
Country Joe McDonald (Country Joe and the Fish)
Mario Cipollina (Huey Lewis and The News)
Joel Selvin (Bay Area historian/journalist)

Award-winning photographer, journalist, and author Mr Bonzai (David Goggin) moderates a panel of leading musical luminaries in a panel discussion looking into the world-renowned SF Sound from the early days to today.

Big Brother and the Holding Company, Country Joe and the Fish, The Sons of Champlin, Huey Lewis and The News—they defined a lifestyle and a spirit that is very much alive today. Peter Albin, Country Joe McDonald, Bill Champlin, Mario Cipollina, and Bay Area historian/journalist Joel Selvin share insights of performance, the recording craft, and keys to the creative process.

Student/Career Event
SC-2 STUDENT RECORDING CRITIQUES
Saturday, October 27, 11:30 am – 12:30 pm
Room 114

Moderator: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA

Students! Bring in your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or surround work on CD, DVD, memory-stick, or hard disc, as clearly-labeled 44.1 KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC.

Product Design Session 5 **Saturday, October 27**
12:00 noon – 1:00 pm **Room 120**

GRAPHICAL AUDIO/DSP APPLICATIONS DEVELOPMENT ENVIRONMENT FOR FIXED AND FLOATING POINT PROCESSORS

Presenter: **Miguel Chavez**, Analog Devices

Graphical development environments have been used in the audio industry for a number of years. An ideal graphical environment not only allows for algorithm development and prototyping but also facilitates the development of run-time DSP applications by producing production-ready code. This presentation will discuss how a graphic environments' real-time control and parameter tuning makes audio DSPs easy to evaluate, design, and use resulting in a shortened development time and reduced time-to-market. It will then describe and explain the software architecture decisions and design challenges that were used to develop a new and expanded development environment for audio-centric commercially available fixed and floating-point processors.

Project Studio Expo **Saturday, October 27**
12:00 noon – 1:00 pm **PSE Stage**

PSE2 - TOTAL TRACKING: GET IT RIGHT AT SOURCE: CHOOSING AND RECORDING YOUR SOUND SOURCE

Presenters: **Hugh Robjohns**, Technical Editor, Sound on Sound, Crowle, UK
Bill Gibson, Art Institute of Seattle, Seattle, WA, USA

The astonishing and ever-improving power and versatility of digital signal processing plug-ins for computer audio workstations has encouraged the widespread belief that everything can be "fixed in the mix"—and in many cases, of course, it can. However, this approach is always extremely time-consuming and the results aren't always perfect. It is often much faster, and with far more satisfying results, to get the right sound from the outset by careful selection of the source and appropriate microphone selection and positioning. This workshop will explore a wide variety of examples, analyzing the requirements and discussing practical techniques of optimizing source recordings.

Saturday October 27 **12:00 noon** **Room 124**

Technical Committee Meeting on Microphones and Applications

Historical Program
H2 -THE EVOLUTION OF ELECTRICAL RECORDING AT RCA VICTOR STUDIOS 1925–1953
Saturday, October 27, 12:15 pm – 1:15 pm
Room 122

Presenter: **Nicholas Bergh**, Endpoint Audio Labs, Burbank, CA, USA

This must-see presentation explores the first decades of electric recording by examining the technical evolution of the RCA Victor recording studios. The early commercial recording industry was comprised of only a few major studios, employing just a handful of recording personnel who used relatively secret equipment. Very little is left from the music industry to document the early electrical recording era—cryptic technical notations in recording ledgers, brief technical comments in memoirs, a few crude photos, and decades of secondhand lore. Since the few remaining Victor studio documents alone are not adequate in explaining what was going on behind the scenes, the focus of this research has been a number of alternative primary sources, for example the original engineering reports from Western Electric and RCA, manuals for the leased studio equipment, and modern tests of restored original recording equipment. Although these sources are not always definitive on their own, they become quite powerful when used together. This research helps address some of the important questions and misconceptions of this era that still face us today in the modern use and archival transfer of these recordings. Mr. Bergh will explain sound quality shifts during different years, the development of equalization standards, and session ledger notations. Exploring the evolution of RCA Victor also provides an important understanding of the evolution of the modern recording industry as a whole. The presentation will include images of the recording equipment used, as well as fascinating audio clips.

Student/Career Event
SC-3 EDUCATION AND CAREER/JOB FAIR
Saturday, October 27, 12:30 pm – 2:00 pm
Foyer

The combined AES 133 Education and Career Fair will match job seekers with companies and prospective students with schools.

Companies

Looking for the best and brightest minds in the audio world? No place will have more of them assembled than the 133rd Convention of the Audio Engineering Society. Companies are invited to participate in our Education and Career Fair, free of charge. This is the perfect chance to identify your ideal new hires!

All attendees of the convention, students and professionals alike, are welcome to come visit with representatives from participating companies to find out more about job and internship opportunities in the audio industry. Bring your resume!

Schools

One of the best reasons to attend AES conventions is the opportunity to make important connections with your fellow educators from around the globe. Academic Institutions offering studies in audio (from short courses to graduate degrees) will be represented in a "table top" session. Information on each school's respective programs will be made available through displays and academic guidance. There is no charge for schools/institutions to participate. Admission is free and open to all convention attendees.

Audio Industry Seminar **Saturday, October 27**
12:30 pm – 2:30 pm **Room 112**

LECTROSONICS

Optimizing Your Wireless Microphone Systems for the Real World

Presenters: **Suzanna Bailey**, American Conservatory Theater
Karl Winkler, Lectrosonics

This workshop aims to provide attendees with a range of tools and concepts involved with getting the most out their wireless mic systems. Experience shows that proper planning, system design, and audio basics go a long way toward success in this area. Topics covered will be the spectrum environment, transmitter placement, gain structure, band planning & frequency coordination, antenna systems, and lavaliere microphone dressing.

Broadcast/Media Streaming Session 7
Saturday, October 27 **12:45 pm – 1:45 pm**
Room 131

MAINTENANCE, REPAIR, AND TROUBLESHOOTING

Chair: **Kirk Harnack**, Telos Alliance, Nashville, TN, USA;
South Seas Broadcasting Corp., Pago Pago, American Samoa

Panelists: *Dan Mansergh*, KQED
Mike Pappas
Milos Nemcik
Bill Sacks, Optimod Refurbishing, Hollywood, MD, USA
Kimberly Sacks, Optimod Refurbishing, Hollywood, MD, USA

Secrets of Today's Best Troubleshooters

Much of today's audio equipment may be categorized as "consumer, throw-away" gear, or so complex that factory assistance is required for a board or module swap. The art of Maintenance, Repair, and Troubleshooting is actually as important as ever, even as the areas of focus may be changing. This session, open to any AES attendee (even Exhibits Only passes) brings together some of the sharpest troubleshooters in the audio business. They'll share their secrets to finding problems, fixing them, and working to ensure they don't happen again. We'll delve into troubleshooting on the systems level, module level, and the component level, and explain some guiding principles that top engineers share.

Audio Industry Seminar **Saturday, October 27**
1:00 pm – 1:30 pm **Room 110**

JBL PROFESSIONAL

"Big Sound" in Small and Medium Size Rooms: Next-Gen Transducer, Acoustic and Electronic Technologies Make High-Performance, Large Format Monitoring Accessible to a Broad Range of Rooms

Presenters: **Peter Chaikin**, Sr. Manager, Recording and Broadcast Marketing, JBL Professional
Allan Devantier, Manager of Objective Evaluation, Harman International
Charles Sprinkle, Sr. Acoustic Systems Engineer, Harman International

To date, space, acoustic and budgetary limitations have prevented smaller rooms from enjoying the high output, detailed monitoring experience of the large control room. For these rooms, near field monitors are the fall back solution. At AES, JBL Professional introduces a large-format monitoring system in a form-factor and at a price point that make large format monitoring a viable option for project studios and small mix rooms. We will discuss the enabling technologies and demonstrate the system.

** Session times on Saturday are: 1:00, 1:30, 2:00, 2:30, 3:00, 3:30, 4:00, and 4:30. Each session is 30 minutes.*

Audio Industry Seminar **Saturday, October 27**
1:00 pm – 2:00 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

The Collaborative Mix; Session Prep and Pre-Mix

Presenters: **Justin Hergett**
James Krausse
Tony Maserati

Since they can't have everyone over to their studio, multi Grammy-winning engineer/producer Tony Maserati (Mary J Blige, David Bowie, Usher, Puff Daddy, Notorious BIG, Jay-Z, R Kelly, Maxwell, Destiny's Child, Jennifer Lopez, Alicia Keys) will be joined by his team Outfit 27.

Outfit 27 consists of Ron "Spider " Entwistle (GOTYE, Knife Party), Justin Hergett (Jason Mraz, Lady Gaga) , Chris Tabron (John Legend, Santigold), and Jon Castelli (GOTYE, Lady Gaga), and Tony.

Together they will present 3 sessions (Session Prep/Premix, Additional Production, and Mix-down) in which they will take a song from concept to final print and all "in the box." During these sessions they will actually prep, tweak, and mix a track in front of the audience complete with commentary as is customary at their place.

The ultimate opportunity to see Tony and his crew at work!!

Saturday October 27 **1:00 pm** **Room 124**

Technical Committee Meeting on Studio Practices and Production

Project Studio Expo **Saturday, October 27**
1:00 pm – 2:00 pm **PSE Stage**

PSE3 - MIXING SECRETS: PRODUCTION TRICKS TO USE WITH ANY DAW

Presenter: **Mike Senior**, Sound On Sound, Munich, Germany; Cambridge Music Technology

Affordable DAW software now provides all the processing tools you need to create commercially competitive music mixes within a home, college, or project studio. As such, the overriding concern for budget-conscious engineers these days should be to develop effective habits with regard to studio monitoring, mix balancing, and quality control. Important techniques in each of these three areas are often neglected in small-scale productions, leading to mixes that don't stack up against professional releases, or that collapse on some mass-market listening systems. In this seminar, Sound On Sound magazine's "Mix Rescue" columnist Mike Senior will draw on his experience of thousands of project-studio mixes to highlight the most frequently overlooked studio tricks. In the process he'll demonstrate how these methods can powerfully upgrade your sonics without breaking the bank, no matter which DAW you're using.

Saturday October 27 **1:30 pm** **Room 125**

AES Standards Committee Meeting SC-05-02 on Audio Connectors

Session P10 **Saturday, October 27**
2:00 pm – 6:00 pm **Room 121**

TRANSDUCERS

Chair: **Alexander Voishvillo**, JBL Professional, Northridge, CA, USA

2:00 pm

P10-1 The Relationship between Perception and Measurement of Headphone Sound Quality— *Sean Olive, Todd Welti*, Harman International, Northridge, CA, USA

Double-blind listening tests were performed on six popular circumaural headphones to study the relationship between their perceived sound quality and their acoustical performance. In terms of overall sound quality, the most preferred headphones were perceived to have the most neutral spectral balance with the lowest coloration. When measured on an acoustic coupler, the most preferred headphones produced the smoothest and flattest amplitude response, a response that deviates from the current IEC recommended diffuse-field calibration. The results provide further evidence that the IEC 60268-7 headphone calibration is not optimal for achieving the best sound quality.
Convention Paper 8744

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2:30 pm

P10-2 On the Study of Ionic Microphones—
Hiroshi Akino,[1] *Hirofumi Shimokawa*,[1] *Tadashi Kikutani*,[2] *Jackie Green*[2]
[1]Kanagawa Institute of Technology, Kanagawa, Japan
[2]Audio-Technica U.S., Inc., Stow, OH, USA

Diaphragm-less ionic loudspeakers using both low-temperature and high-temperature plasma methods have already been studied and developed for practical use. This study examined using similar methods to create a diaphragm-less ionic microphone. Although the low-temperature method was not practical due to high noise levels in the discharges, the high-temperature method exhibited a useful shifting of the oscillation frequency. By performing FM detection on this oscillation frequency shift, audio signals were obtained. Accordingly, an ionic microphone was tested in which the frequency response level using high-temperature plasma increased as the sound wave frequency decreased. Maintaining performance proved difficult as discharges in the air led to wear of the needle electrode tip and adhesion of products of the discharge. Study results showed that the stability of the discharge corresponded to the non-uniform electric field that was dependent on the formation shape of the high-temperature plasma, the shape of the discharge electrode, and the use of inert gas that protected the needle electrode. This paper reviews the experimental outcome of the two ionic methods, and considerations given to resolve the tip and discharge product and stability problems.
Convention Paper 8745

3:00 pm

P10-3 Midrange Resonant Scattering in Loudspeakers—*Juha Backman*, Nokia Corporation, Espoo, Finland

One of the significant sources of midrange coloration in loudspeakers is the resonant scattering of the exterior sound field from ports, recesses, or horns. This paper discusses the qualitative behavior of the scattered sound and introduces a computationally efficient model for such scattering, based on waveguide models for the acoustical elements (ports, etc.), and mutual radiation impedance model for their coupling to the sound field generated by the drivers. In the simplest case of driver-port interaction in a direct radiating loudspeaker an approximate analytical expression can be written for the scattered sound. These methods can be applied to numerical optimization of loudspeaker layouts.
Convention Paper 8746

3:30 pm

P10-4 Long Distance Induction Drive Loud Hailer Characterization—*Marshall Buck*,[1] *David Graebener*,[2] *Ron Sauro*[3]
[1]Psychotechnology, Inc., Los Angeles, CA, USA
[2]Wisdom Audio Corporation, Carson City, NV, USA
[3]NWAA Labs, Inc., Elma, WA, USA

Further development of the high power, high efficiency induction drive compression driver when mounted on a tight pattern horn results in a high performance loud hailer. The detailed performance is tested in an independent laboratory with unique capabilities, including indoor frequency response at a distance of 4 meters. Additional characteristics tested include maximum burst output level, polar response, and directivity balloons. Outdoor tests were also performed at distances up to 220 meters and included speech transmission index and frequency response. Plane wave tube driver-phase plug tests were performed to assess incoherence, power compression, efficiency, and frequency response.
Convention Paper 8747

4:00 pm

P10-5 Optimal Configurations for Subwoofers in Rooms Considering Seat to Seat Variation and Low Frequency Efficiency—*Todd Welti*, Harman International, Northridge, CA, USA

The placement of subwoofers and listeners in small rooms and the size and shape of the room all have profound influences on the resulting low frequency response. In this study, a computer model was used to investigate a large number of room, seating, and subwoofer configurations. For each configuration simulated, metrics for seat to seat consistency and bass efficiency were calculated and combined in a newly proposed metric, which is intended as an overall figure of merit. The data presented has much practical value in small room design for new rooms, or even for modifying existing configurations.
Convention Paper 8748

4:30 pm

P10-6 Modeling the Large Signal Behavior of Micro-Speakers—*Wolfgang Klippel*, Klippel GmbH, Dresden, Germany

The mechanical and acoustical losses considered in the lumped parameter modeling of electro-dynamical transducers may become a dominant source of nonlinear distortion in micro-speakers, tweeters, headphones, and some horn compression drivers where the total quality factor Q_{ts} is not dominated by the electrical damping realized by a high force factor Bl and a low voice resistance R_e. This paper presents a nonlinear model describing the generation of the distortion and a new dynamic measurement technique for identifying the nonlinear resistance $R_{ms}(v)$ as a function of voice coil velocity v. The theory and the identification technique are verified by comparing distortion and other nonlinear symptoms measured on micro-speakers as used in cellular phones with the corresponding behavior predicted by the nonlinear model.
Convention Paper 8749

5:00 pm

P10-7 An Indirect Study of Compliance and Damping in Linear Array Transducers—
Richard Little, Far North Electroacoustics, Surrey, BC, Canada

A linear array transducer is a dual-motor, dual-coil, multi-cone, tubularly-shaped transducer whose

shape defeats many measurement techniques that can be used to examine directly the force-deflection behavior of its diaphragm suspension system. Instead, the impedance curve of the transducer is compared against theoretical linear models to determine best-fit parameter values. The variation in the value of these parameters with increasing input signal levels is also examined.
Convention Paper 8750

5:30 pm

P10-8 Bandwidth Extension for Microphone Arrays —*Benjamin Bernschütz*, Cologne University of Applied Sciences, Cologne, Germany, and Technical University of Berlin, Berlin, Germany

Microphone arrays are in the focus of interest for spatial audio recording applications or the analysis of sound fields. But one of the major problems of microphone arrays is the limited operational frequency range. Especially at high frequencies spatial aliasing artifacts tend to disturb the output signal. This severely restricts the applicability and acceptance of microphone arrays in practice. A new approach to enhance the bandwidth of microphone arrays is presented, which is based on some restrictive assumptions concerning natural sound fields, the separate acquisition and treatment of spatiotemporal and spectrotemporal sound field properties, and the subsequent synthesis of array signals for critical frequency bands. Additionally, the method can be used for spatial audio data reduction algorithms.
Convention Paper 8751

Session P11 **Saturday, October 27**
2:00 pm – 3:30 pm **Foyer**

POSTERS: SPATIAL AUDIO

2:00 pm

P11-1 Blind Upmixing for Height and Wide Channels Based on an Image Source Method —*Sunwoong Choi,*[1] *Dong-il Hyun*,[1] *Young-cheol Park*,[2] *Seokpil Lee*,[3] *Dae Hee Youn*[1]
[1]Yonsei University, Seoul, Korea
[2]Yonsei University, Wonju, Kwangwon-do, Korea
[3]Korea Electronics Technology Institute (KETI), Seoul, Korea

In this paper we present a method of synthesizing the height and wide channel signals for stereo upmix to multichannel format beyond 5.1. To provide an improved envelopment, reflections from ceiling and side walls are considered for the height and wide channel synthesis. Early reflections (ERs) corresponding to the spatial sections covered by the height and wide channel speakers are separately synthesized using the image source method, and the parameters for the ER generation are determined from the primary-to-ambient ratio (PAR) estimated from the stereo signal. Later, the synthesized ERs are mixed with decorrelated ambient signals and transmitted to the respective channels. Subjective listening tests verify that listener envelopment can be improved by using the proposed method.
Convention Paper 8752

2:00 pm

P11-2 Spatial Sound Design Tool for 22.2 Channel 3-D Audio Productions, with Height—*Wieslaw Woszczyk*,[1,2] *Brett Leonard*,[1,2] *David Benson*[1,2]
[1]McGill University, Montreal, QC, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

Advanced television and cinema systems utilize multiple loudspeakers distributed in three dimensions potentially allowing sound sources and ambiances to appear anywhere in the 3-D space enveloping the viewers, as is the case in 22.2 channel audio format for Ultra High Definition Television (UHDTV). The paper describes a comprehensive tool developed specifically for designing auditory spaces in 22.2 audio but adaptable to any advanced multi-speaker 3-D sound rendering system. The key design goals are the ease of generating and manipulating ambient environments in 3-D and time code automation for creating dynamic spatial narration. The system uses low-latency convolution of high-resolution room impulse responses contained in the library. User testing and evaluation show that the system's features and architecture enable fast and effective spatial design in 3-D audio.
Convention Paper 8753

2:00 pm

P11-3 Efficient Primary-Ambient Decomposition Algorithm for Audio Upmix—*Yong-Hyun Baek*,[1] *Se-Woon Jeon*,[2] *Young-cheol Park*,[1] *Seokpil Lee*[3]
[1]Yonsei University, Wonju, Kwangwon-do, Korea
[2]Yonsei University, Seoul, Korea
[3]Korea Electronics Technology Institute (KETI), Seoul, Korea

Decomposition of a stereo signal into the primary and ambient components is a key step to the stereo upmix and it is often based on the principal component analysis (PCA). However, major shortcoming of the PCA-based method is that accuracy of the decomposed components is dependent on both the primary-to-ambient power ratio (PAR) and the panning angle. Previously, a modified PCA was suggested to solve the PAR-dependent problem. However, its performance is still dependent on the panning angle of the primary signal. In this paper we proposed a new PCA-based primary-ambient decomposition algorithm whose performance is not affected by the PAR as well as the panning angle. The proposed algorithm finds scale factors based on a criterion that is set to preserve the powers of the mixed components, so that the original primary and ambient powers are correctly retrieved. Simulation results are presented to show the effectiveness of the proposed algorithm.
Convention Paper 8754

2:00 pm

P11-4 On the Use of Dynamically Varied Loudspeaker Spacing in Wave Field Synthesis—*Rishabh Ranjan, Woon-Seng Gan*, Nanyang Technological University, Singapore, Singapore

➥

Wave field synthesis (WFS) has evolved as a promising spatial audio rendering technique in recent years and has been widely accepted as the optimal way of sound reproduction technique. Suppressing spatial aliasing artifacts and accurate reproduction of sound field remain the focal points of research in WFS over the recent years. The use of optimum loudspeaker configuration is necessary to achieve perceptually correct sound field in the listening space. In this paper we analyze the performance of dynamically spaced loudspeaker arrays whose spacing varies with the audio signal frequency content. The proposed technique optimizes the usage of a prearranged set of loudspeaker arrays to avoid spatial aliasing at relatively low frequencies as compared to uniformly fixed array spacing in conventional WFS setups.
Convention Paper 8755

2:00 pm

P11-5 A Simple and Efficient Method for Real-Time Computation and Transformation of Spherical Harmonic-Based Sound Fields—*Robert E. Davis, D. Fraser Clark*, University of the West of Scotland, Paisley, Scotland, UK

The potential for higher order Ambisonics to be applied to audio applications such as virtual reality, live music, and computer games relies entirely on the real-time performance characteristics of the system, as the computational overhead determines factors of latency and, consequently, user experience. Spherical harmonic functions are used to describe the directional information in an Ambisonic sound field, and as the order of the system is increased, so too is the computational expense, due to the added number of spherical harmonic functions to be calculated. The present paper describes a method for simplified implementation and efficient computation of the spherical harmonic functions and applies the technique to the transformation of encoded sound fields. Comparisons between the new method and typical direct calculation methods are presented.
Convention Paper 8756

2:00 pm

P11-6 Headphone Virtualization: Improved Localization and Externalization of Non-Individualized HRTFs by Cluster Analysis—*Robert P. Tame,*[1,2] *Daniele Barchiese,*[2] *Anssi Klapuri*[2]
[1]DTS Licensing Limited Northern Ireland, Bangor, County Down, UK
[2]Queen Mary University of London, London, UK

Research and experimentation is described that aims to prove the hypothesis that by allowing a listener to choose a single non-individualized profile of HRTFs from a subset of maximally different best representative profiles extracted from a database improved localization, and externalization can be achieved for the listener. *k*-means cluster analysis of entire impulse responses is used to identify the subset of profiles. Experimentation in a controlled environment shows that test subjects who were offered a choice of a preferred HRTF profile were able to consistently discriminate between a front center or rear center virtualized sound source 78.6% of the time, compared with 64.3% in a second group given an arbitrary HRTF profile. Similar results were obtained from virtualizations in uncontrolled environments.
Convention Paper 8757

2:00 pm

P11-7 Searching Impulse Response Libraries Using Room Acoustic Descriptors—*David Benson,*[1,2] *Wieslaw Woszczyk*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

The ease with which Impulse Response (IR) libraries can be searched is a principal determinant of the usability of a convolution reverberation system. Popular software packages for convolution reverb typically permit searching over metadata that describe how and where an IR was measured, but this “how and where” information often fails to adequately characterize the perceptual properties of the reverberation associated with the IR. This paper explores an alternative approach to IR searching based not on “how and where” descriptors but instead on room acoustics descriptors that are thought to be more perceptually relevant. This alternative approach was compared with more traditional approaches on the basis of a simple IR search task. Results are discussed.
Convention Paper 8758

2:00 pm

P11-8 HRIR Database with Measured Actual Source Direction Data—*Javier Gómez Bolaños, Ville Pulkki*, Aalto University, Espoo, Finland

A database is presented consisting of head-related impulse responses (HRIR) of 21 subjects measured in an anechoic chamber with simultaneous measurement of head position and orientation. The HRIR data for sound sources at 1.35 m and 68 cm in 240 directions with elevations between ±45 degrees and full azimuth range were measured using the blocked ear canal method. The frequency region of the measured responses ranges from 100 Hz up to 20 kHz for a flat response (+0.1 dB/–0.5 dB). This data is accompanied with the measured azimuth and elevation of the source respect to the position and orientation of the subject's head obtained with a tracking system based on infrared cameras. The HRIR data is accessible from the Internet.
Convention Paper 8759

2:00 pm

P11-9 On the Study of Frontal-Emitter Headphone to Improve 3-D Audio Playback—*Kaushik Sunder, Ee-Leng Tan, Woon-Seng Gan*, Nanyang Technological University, Singapore, Singapore

Virtual audio synthesis and playback through headphones by its virtue have several limitations, such as the front-back confusion and in-head localization of the sound presented to the listener. Use of non-individual head related

transfer functions (HRTFs) further increases these front-back confusion and degrades the virtual auditory image. In this paper we present a method for customizing non-individual HRTFs by embedding personal cues using the distinctive morphology of the individual's ear. In this paper we study the frontal projection of sound using headphones to reduce the front-back confusion in 3-D audio playback. Additional processing blocks, such as decorrelation and front-back biasing are implemented to externalize and control the auditory depth of the frontal image. Subjective tests are conducted using these processing blocks, and its impact to localization is reported in this paper.
Convention Paper 8760

2:00 pm

P11-10 Kinect Application for a Wave Field Synthesis-Based Reproduction System—
Michele Gasparini, Stefania Cecchi, Laura Romoli, Andrea Primavera, Paolo Peretti, Francesco Piazza, Universitá Politecnica della Marche, Ancona (AN), Italy

Wave field synthesis is a reproduction technique capable to reproduce realistic acoustic image taking advantage of a large number of loudspeakers. In particular, it is possible to reproduce moving sound sources, achieving good performance in terms of sound quality and accuracy. In this context, an efficient application of a wave field synthesis reproduction system is proposed, introducing a Kinect control on the transmitting room, capable to accurately track the source movement and thus preserving the spatial representation of the acoustic scene. The proposed architecture is implemented using a real time framework considering a network connection between the receiving and transmitting room: several tests have been performed to evaluate the realism of the achieved performance.
Convention Paper 8761

Tutorial 6 — Saturday, October 27
2:00 pm – 3:00 pm — Room 120

SECRETS OF THE SILENT STUDIO

Presenter: **Jamie Fox**, The Engineering Enterprise, Alameda, CA, USA

How does one create recording studios with a low noise floor? Would it be possible to apply the same technology to music festivals and concert halls? The goal of this presentation is to explain simple and effective methods for studio and venue designers to lower the noise floor by orders of magnitude, thereby increasing the dynamic range. "The secrets," include a revolutionary installation method for power wiring. Come learn how this new concept was applied to the Stanford Bing Concert Hall, scheduled to open in 2013.

Workshop 6 — Saturday, October 27
2:00 pm – 4:00 pm — Room 133

ARE YOU READY FOR THE NEW MEDIA EXPRESS? OR DEALING WITH TODAY'S AUDIO DELIVERY FORMATS

Chair: **Jim Kaiser**, CEMB / Belmont University, Nashville, TN

Panelists: *Robert Bleidt*, Fraunhofer USA Digital Media Technologies, San Jose, CA, USA
Stefan Bock, msm-studios GmbH
David Chesky, HD Tracks/Chesky Records
Robert Katz, Digital Domain Mastering, Altamonte Springs, FL, USA

With the rapid dominance of digital delivery mediums for audio, traditional physical media and their associated standards are losing relevance. Today's audio mastering engineer is expected to provide an appropriate format for any of their client's expanding uses. The ability to do so properly is affected by the preceding production and mixing processes, as well as an understanding of what follows. This workshop will focus on detailing newer audio delivery formats and the typical processes that a digital file goes through on its way through to the end-consumer. The intention is to provide the know-how to ensure that one is not technically compromising the client's music on the way to providing for various lower and higher-resolution releases (e.g., Mastered for iTunes, HD Tracks, etc.).

Broadcast/Media Streaming Session 8
Saturday, October 27 — 2:00 pm – 4:00 pm
Room 131

LOUDNESS AND METADATA— LIVING WITH THE CALM ACT

Chair: **Joel Spector**, Freelance Television and Theater Sound Designer, Riverdale, NY, USA

Panelists: *Florian Camerer*, ORF, Vienna, Austria
Tim Carroll, Linear Acoustic Inc., Lancaster, PA, USA
Stephen Lyman, Dolby Laboratories, San Francisco, CA, USA
Robert Murch, Fox Television
Lon Neumann, Neumann Technologies, Sherman Oaks, CA, USA
Robert Seidel, CBS
Jim Starzynski, NBC Universal

The Commercial Advertising Loudness Mitigation (CALM) Act was signed by President Obama in 2011. Enforcement by the FCC will begin in December of this year.

Television broadcasters and Multichannel Video Program Distributors (MVPDs) are required to put in place procedures, software and hardware to "effectively control program-to-interstitial loudness … and loudness management at the boundaries of programs and interstitial content." Objective data must be supplied to the FCC to support compliance with the legislation as well as timely resolution of listener complaints. Similar rules have been developed in the UK and other parts of the world.

Members of our panel of experts have worked tirelessly to either create loudness control recommendations that have become the law or to bring those recommendations to implementation at the companies they represent. This session will cover the FCC's Report and Order on the CALM Act, the development of the ATSC's A/85 Recommended Practice that is now part of the U.S. legislation, and both domestic and European technical developments by major media distributors and P/LOUD.

Game Audio Session 6 **Saturday, October 27**
2:00 pm – 3:30 pm **Room 130**

**BUILDING AN AAA TITLE—
ROLES AND RESPONSIBILITIES**

Presenters: **Justin Drust**, Red Storm Entertainment, Cary, NC, USA
Fran Dyer, Red Storm Entertainment, Cary, NC, USA
Chris Groegler, Red Storm Entertainment, Cary, NC, USA
Matt McCallus, Red Storm Entertainment, Cary, NC, USA
Matte Wagner, Red Storm Entertainment, Cary, NC, USA

Look behind the curtain of an AAA title and into the world of game audio development from multiple perspectives— the Producer, Audio Director, Sound Designers, and Programmer. See the inner workings of the Red Storm Audio Team as they collaborate with multiple Ubisoft studios to create the Tom Clancy's Ghost Recon: Future Soldier Multiplayer experience. Discover the tips, tricks, and techniques of a major AAA title's audio design process from conception to completion in this postmortem.

Live Sound Seminar 5 **Saturday, October 27**
2:00 pm – 3:30 pm **Room 132**

**PLANNING A LIVE SOUND EDUCATION:
SHOULD I STUDY THE THEORY, OR PRACTICE
THE SKILLS?**

Chair: **Ted Leamy**, Promedia-Ultrasound

Panelists: *Kevin W. Becka*, Conservatory of Recording Arts and Sciences/Mix Magazine, Gilbert, AZ, USA
Michael Jackson, Independent Live Sound Engineer, Richmond, CA, USA
David Scheirman, Harman Professional

Getting the correct education to prepare for a career in live audio is a challenging proposition. Which is better—formal education or hands-on experience? The choices being marketed for "education" today seem endless. Manufacturer trainings, school programs, product user certifications, continuing "education" credits, CTS technician exams, smart training, stupid training, "learn to be a genius" seminars. … Who you gonna call? Is it better to just go on the road and "earn while you learn"? Join your peers and some industry veterans for a lively investigation of the options you face when learning about live sound career choices. Hear what works, and what doesn't. Come and discuss your own experiences … good, bad, or otherwise. This will be an open discussion with all participants. Educators, students, business owners, engineers, and anyone involved in live sound should come and be heard at this unique session.

Networked Audio Session 2 **Saturday, October 27**
2:00 pm – 3:30 pm **Room 123**

OPEN IP PROTOCOLS FOR AUDIO NETWORKING

Presenter: **Kevin Gross**, AVA Networks, Boulder, CO, USA

The networking and telecommunication industry has its own set of network protocols for carriage of audio and video over IP networks. These protocols have been widely deployed for telephony and teleconferencing applications, internet streaming, and cable television. This tutorial will acquaint attendees with these protocols and their capabilities and limitations. The relationship to AVB protocols will be discussed.

Specifically, attendees will learn about Internet protocol (IP), voice over IP (VoIP), IP television (IPTV), HTTP streaming, real-time transport protocol (RTP), real-time transport control protocol (RTCP), real-time streaming protocol (RTSP), session initiation protocol (SIP), session description protocol (SDP), Bonjour, session announcement protocol (SAP), differentiated services (DiffServ), and IEEE 1588 precision time protocol (PTP)

An overview of AES Standards work, X192, adapting these protocols to high-performance audio applications will be given.

Special Event
GRAMMY SOUNDTABLE
Saturday, October 27, 2:00 pm – 3:30 pm
Room 134

Moderator: **Ed Cherney**, GRAMMY-winning engineer/producer (Eric Clapton, Bonnie Raitt, the Rolling Stones)

Panelists: *Ryan Hewitt*, GRAMMY-winning engineer/producer/mixer (Avett Brothers, Brandi Carlile, Red Hot Chili Peppers)
Leslie Ann Jones, GRAMMY-winning engineer and Director of Music & Scoring/Skywalker Sound (Herbie Hancock, Kronos Quartet, Bobby McFerrin)
Dave Pensado, GRAMMY-winning engineer and host of Pensado's Place web series (Christina Aguilera, Beyonce, Pink)
Salaam Remi, Composer/Producer (Alicia Keys, NAS, Amy Winehouse)
Elliot Scheiner, GRAMMY-winning engineer (The Eagles, Bruce Hornsby, Steely Dan)

Sonic Imprints: Songs That Changed My Life—Part 3

Some songs are hits, some we just love, and some have changed our lives. Our panelists break down the DNA of their favorite tracks and explain what moved them, what grabbed them, and why these songs left a life-long impression. Back by popular demand, this Special Event is guaranteed to make you feel good about being in the recording business.

Project Studio Expo **Saturday, October 27**
2:00 pm – 3:00 pm **PSE Stage**

**PSE4 - MASTER YOUR TRACKS: DIY RESULTS
TO COMPETE WITH THE PROS**

Presenter: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

Mastering is the final step prior to duplication and, as such, represents the last opportunity to make any final tweaks to a piece of music for highest sonic quality—and maximum transportability among playback systems. Traditionally, musicians have used professional mastering engineers in order to take advantage of their experience and ears, but in today's tight economy—and with the advent of tools that allow for "do-it-yourself" mastering—many musicians are choosing to do their own mastering. This workshop describes the pitfalls and advantages of "project mastering" as well as the main mistakes to avoid but primarily emphasizes practical techniques that can bring out the very best in a piece of music. It also covers the process of album assembly and how to make sure the music in a collection or album provides a smooth, cohesive listening experience.

Audio Industry Seminar **Saturday, October 27**
2:15 pm – 3:00 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

Erik Zobler Presents: Music Projects from Teena Marie, Jeffrey Osborne, and More

Presenter: **Erik Zobler**

Multi Grammy winner Zobler has engineered and mixed hundreds of albums, DVDs, and TV and film scores. He is also a mastering engineer, front-of-house live sound mixer, and a writer for Mix Magazine. His credits include Miles Davis, Natalie Cole, Whitney Houston, Gladys Knight, Anita Baker, Sarah Vaughan, Al Jarreau, Stanley Clarke, Esparanza Spaulding, Bob James, Teena Marie, George Duke, and many more.

Erik will be showing some of his latest projects from Teena Marie, Jeffrey Osborne, and a 5.1 surprise from Frank Zappa.

Session P12 **Saturday, October 27**
2:30 pm – 6:00 pm **Room 122**

SOUND ANALYSIS AND SYNTHESIS

Chair: **Jean Laroche**, Audience, Inc.

2:30 pm

P12-1 Drum Synthesis via Low-Frequency Parametric Modes and Altered Residuals—*Haiying Xia, Julius O. Smith, III*, Stanford University, Stanford, CA, USA

Techniques are proposed for drum synthesis using a two-band source-filter model. A Butterworth lowpass/highpass band-split is used to separate a recorded "high tom" drum hit into low and high bands. The low band, containing the most salient modes of vibration, is downsampled and Poisson-windowed to accelerate its decay and facilitate mode extraction. A weighted equation-error method is used to fit an all-pole model—the "modal model"—to the first five modes of the low band in the case of the high tom. The modal model is removed from the low band by inverse filtering, and the resulting residual is taken as a starting point for excitation modeling in the low band. For the high band, low-order linear prediction (LP) is used to model the spectral envelope. The bands are resynthesized by feeding the residual signals to their respective all-pole forward filters, upsampling the low band, and summing. The modal model can be modulated to obtain the sound of different drums and other effects. The residuals can be altered to obtain the effects of different striking locations and striker materials.
Convention Paper 8762

3:00 pm

P12-2 Drum Pattern Humanization Using a Recursive Bayesian Framework—*Ryan Stables, Cham Athwal, Rob Cade*, Birmingham City University, Birmingham, UK

In this study we discuss some of the limitations of Gaussian humanization and consider ways in which the articulation patterns exhibited by percussionists can be emulated using a probabilistic model. Prior and likelihood functions are derived from a dataset of professional drummers to create a series of empirical distributions. These are then used to independently modulate the onset locations and amplitudes of a quantized sequence, using a recursive Bayesian framework. Finally, we evaluate the performance of the model against sequences created with a Gaussian humanizer and sequences created with a Hidden Markov Model (HMM) using paired listening tests. We are able to demonstrate that probabilistic models perform better than instantaneous Gaussian models, when evaluated using a 4/4 rock beat at 120 bpm.
Convention Paper 8763

3:30 pm

P12-3 Procedural Audio Modeling for Particle-Based Environmental Effects—*Charles Verron, George Drettakis*, REVES-INRIA, Sophia-Antipolis, France

We present a sound synthesizer dedicated to particle-based environmental effects, for use in interactive virtual environments. The synthesis engine is based on five physically-inspired basic elements (that we call sound atoms) that can be parameterized and stochastically distributed in time and space. Based on this set of atomic elements, models are presented for reproducing several environmental sound sources. Compared to pre-recorded sound samples, procedural synthesis provides extra flexibility to manipulate and control the sound source properties with physically-inspired parameters. In this paper the controls are used simultaneously to modify particle-based graphical models, resulting in synchronous audio/graphics environmental effects. The approach is illustrated with three models that are commonly used in video games: fire, wind, and rain. The physically-inspired controls simultaneously drive graphical parameters (e.g., distribution of particles, average particles velocity) and sound parameters (e.g., distribution of sound atoms, spectral modifications). The joint audio/graphics control results in a tightly-coupled interaction between the two modalities that enhances the naturalness of the scene.
Convention Paper 8764

4:00 pm

P12-4 Knowledge Representation Issues in Audio-Related Metadata Model Design—*György Fazekas, Mark B. Sandler*, Queen Mary University of London, London, UK

In order for audio applications to interoperate, some agreement on how information is structured and encoded has to be in place within developer and user communities. This agreement can take the form of an industry standard or a widely adapted open framework consisting of conceptual data models expressed using formal description languages. There are several viable approaches to conceptualize audio related metadata, and several ways to describe the conceptual models, as well as encode and ➥

exchange information. While emerging standards have already been proven invaluable in audio information management, it remains difficult to design or choose the model that is most appropriate for an application. This paper facilitates this process by providing an overview, focusing on differences in conceptual models underlying audio metadata schemata.
Convention Paper 8765

4:30 pm

P12-5 High-Level Semantic Metadata for the Control of Multitrack Adaptive Digital Audio Effects—*Thomas Wilmering, György Fazekas, Mark B. Sandler*, Queen Mary University of London, London, UK

Existing adaptive digital audio effects predominantly use low-level features in order to derive control data. These data do not typically correspond to high-level musicological or semantic information about the content. In order to apply audio transformations selectively on different musical events in a multitrack project, audio engineers and music producers have to resort to manual selection or annotation of the tracks in traditional audio production environments. We propose a new class of audio effects that uses high-level semantic audio features in order to obtain control data for multitrack effects. The metadata is expressed in RDF using several music and audio related Semantic Web ontologies and retrieved using the SPARQL query language.
Convention Paper 8766

5:00 pm

P12-6 On Accommodating Pitch Variation in Long Term Prediction of Speech and Vocals in Audio Coding—*Tejaswi Nanjundaswamy, Kenneth Rose*, University of California, Santa Barbara, Santa Barbara, CA, USA

Exploiting inter-frame redundancies is key to performance enhancement of delay constrained perceptual audio coders. The long term prediction (LTP) tool was introduced in the MPEG Advanced Audio Coding standard, especially for the low delay mode, to capitalize on the periodicity in naturally occurring sounds by identifying a segment of previously reconstructed data as prediction for the current frame. However, speech and vocal content in audio signals is well known to be quasi-periodic and involve small variations in pitch period, which compromise the LTP tool performance. The proposed approach modifies LTP by introducing a single parameter of “geometric” warping, whereby past periodicity is geometrically warped to provide an adjusted prediction for the current samples. We also propose a three-stage parameter estimation technique, where an unwarped LTP filter is first estimated to minimize the mean squared prediction error; then filter parameters are complemented with the warping parameter, and re-estimated within a small neighboring search space to retain the set of S best LTP parameters; and finally, a perceptual distortion-rate procedure is used to select from the S candidates, the parameter set that minimizes the perceptual distortion. Objective and subjective evaluations substantiate the proposed technique’s effectiveness.
Convention Paper 8767

5:30 pm

P12-7 Parametric Coding of Piano Signals—*Michael Schnabel,*[1] *Benjamin Schubert,*[2] *Gerald Schuller*[1]
[1]Ilmenau University of Technology, Ilmenau, Germany
[2]Ilmenau University of Technology, Ilmenau, Germany, now with Fraunhofer IIS, Erlangen, Germany

In this paper an audio coding procedure for piano signals is presented based on a physical model of the piano. Instead of coding the waveform of the signal, the compression is realized by extracting relevant parameters at the encoder. The signal is then re-synthesized at the decoder using the physical model. We describe the development and implementation of algorithms for parameter extraction and the combination of all the components into a coder. A formal listening test was conducted, which shows that we can obtain a high sound quality at a low bit rate, lower than conventional coders. We obtain a bitrate of 11.6 kbps for the proposed piano coder. We use HE-AAC as reference codec at a gross bitrate of 16 kbps. For low and medium chords the proposed piano coder outperforms HE-AAC in terms of subjective quality, while the quality falls below HE-AAC for high chords.
Convention Paper 8768

Audio Industry Seminar — Saturday, October 27
3:00 pm – 4:00 pm — Room 112

M-AUDIO

Empowering Musicians as Non-Technicians. Is the Role of an Audio Engineer Destined for Extinction?

Presenter: **Hiro Shimozato**

As technology redefines how musicians create and perform music, the aspirations of musicians operating without creative limitations and technical dependencies grows. The delineation between musical instrument and recording technology disappears as more music creation tools take on audio production capabilities. Akai Professional, M-Audio, and other inMusic brands along with special guest panelists will address this topic.

Project Studio Expo — Saturday, October 27
3:00 pm – 4:00 pm — PSE Stage

PSE5 - YOU ASK, WE ANSWER

Presenters:
Joe McGrath, SAE Institute
Hugh Robjohns, Technical Editor, Sound on Sound, Crowle, UK
Mike Senior, Sound On Sound, Munich, Germany; Cambridge Music Technology
Paul White, Sound On Sound, Malvern, Worcestershire, UK

Panel discussion with industry pros on topics of special interest to attendees.

Saturday October 27 3:00 pm Room 124

Technical Committee Meeting on Audio Recording and Mastering Systems

Saturday October 27 3:00 pm Room 125

AES Standards Committee Meeting SC-05-05 on EMC

Sound for Pictures 3 Saturday, October 27
3:30 pm – 6:30 pm Room 130

NEW MULTICHANNEL FORMATS FOR 3-D CINEMA AND HOME THEATER

Co-chairs: **Christof Faller**, Illusonic, Uster, Switzerland
Brian McCarty, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia

Panelists: *Kimio Hamasaki*, NHK, Chiba-ken, Japan
Jeff Levison, IOSONO GmbH, Germany
Nicolas Tsingos, Dolby Labs, San Francisco, CA, USA
Wilfried Van Baelen, Auro-3D, Mol, Belgium
Brian Vessa, Sony Pictures

Several new digital cinema formats are under active consideration for cinema soundtrack production. Each was developed to create realistic sound "motion" in parallel with 3-D pictures. This workshop presents a rare first opportunity for the proponents of these leading systems to discuss their specific technologies.

Student/Career Event

SC-4 SPARS SPEED COUNSELING WITH EXPERTS — MENTORING ANSWERS FOR YOUR CAREER
Saturday, October 27, 3:30 pm – 5:30 pm
Room 132

Moderator: **Kirk Imamura**, President, Society of Professional Audio Recording Services (SPARS)

Mentors: *Elise Baldwin*
Chris Bell
David Bowles
Neil Dorfsman
Chris Estes
Deanne Franklin
David Glasser
Dave Hampton
Andrew Hollis
Steve Horowitz
Scott Hull
Eric Johnson
Paul Lipson
Krysten Mate
Dren McDonald
Pat McMakin
David Hewitt
Lauretta Molitor
Leslie Mona-Mathus
Shawn Murphy
Piper Payne
Michael Romanowski
Mark Rubel
Tom Salta
Bob Skye
Chris Spahr
Richard Warp

This event is specially suited for students, recent graduates, young professionals, and those interested in career advice. Hosted by SPARS in cooperation with the AES Education Committee, Women's Audio Mission, G.A.N.G., and Manhattan Producers Alliance, career related Q&A sessions will be offered to participants in a speed group mentoring format. A dozen students will interact with 4–5 working professionals in specific audio engineering fields or categories every 20 minutes. Audio engineering fields/categories include gaming, live sound/live recording, audio manufacturer, mastering, sound for picture, and studio production. Listed mentors are subject to change.

Audio Industry Seminar Saturday, October 27
3:30 pm – 4:15 pm Room 114

PMC: MASTERS OF AUDIO SERIES

Wisseloord Studios

Presenter: **Jochen Veith**

Every (home) studio will have its fair share of acoustical problems. World renowned German acoustician Jochen Veith (JV-Acoustics) will give you some insights into acoustics through a case study of one of his latest projects: the famous Wisseloord Studios in The Netherlands.

Jochen has designed hundreds of mixing, recording, movie and broadcast studios over the last 20 years for major studios and artists throughout the world including Coldplay, Metropolis Studios London, Max Martin Stockholm, Walt Disney, Koch International, BMG Ariola, Wisseloord Studios, NBC Universal, and many more.

Saturday, October 27 3:30 pm Room 111

Historical Committee Meeting

Interested in committee activities? All are welcome.

Session P13 Saturday, October 27
4:00 pm – 5:30 pm Foyer

POSTERS: AUDITORY PERCEPTION AND EVALUATION

4:00 pm

P13-1 Real-Time Implementation of Glasberg and Moore's Loudness Model for Time-Varying Sounds—*Elvira Burdiel, Lasse Vetter, Andrew J. R. Simpson, Michael J. Terrell, Andrew McPherson, Mark B. Sandler*, Queen Mary University of London, London, UK

In this paper, a real-time implementation of the loudness model of Glasberg and Moore [*J. Audio Eng. Soc.* 50, 331–342 (2002)] for time-varying sounds is presented. This real-time implementation embodies several approximations to the model that are necessary to reduce computational costs, both in the time and frequency domains. A quantitative analysis is given that shows the effect of parametric time and frequency domain approximations by comparison to the loudness predictions of the original model. Using real-world music, both the errors introduced as a function of the optimization parameters and the corresponding reduction in computational costs are quantified. Thus, this work provides an informed, contextual approach to approximation of the loudness model for ➥

practical use.
Convention Paper 8769

4:00 pm

P13-2 Subjective Selection of Head-Related Transfer Functions (HRTF) Based on Spectral Coloration and Interaural Time Differences (ITD) Cues—*Kyla McMullen,*[1] *Agnieszka Roginska,*[2] *Gregory H. Wakefield*[1]

[1]University of Michigan, Ann Arbor, MI, USA
[2]New York University, New York, NY, USA

The present study describes an HRTF subjective individualization procedure in which a listener selects from a database those HRTFs that pass several perceptual criteria. Earlier work has demonstrated that listeners are as likely to select a database HRTF as their own when judging externalization, elevation, and front/back discriminability. The procedure employed in this original study requires individually measured ITDs. The present study modifies the original procedure so that individually measured ITDs are unnecessary. Specifically, a standardized ITD is used, in place of the listener's ITD, to identify those database minimum-phase HRTFs with desirable perceptual properties. The selection procedure is then repeated for one of the preferred minimum-phase HRTFs and searches over a database of ITDs. Consistent with the original study, listeners prefer a small subset of HRTFs; in contrast, while individual listeners show clear preferences for some ITDs over others, no small subset of ITDs appears to satisfy all listeners.
Convention Paper 8770

4:00 pm

P13-3 Does Understanding of Test Items Help or Hinder Subjective Assessment of Basic Audio Quality? —*Nadja Schinkel-Bielefeld,*[1,2] *Netaya Lotze,*[3] *Frederik Nagel*[1,2]

[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories, Erlangen, Germany
[3]Leibniz Universität Hannover, Hannover, Germany

During listening tests for subjective evaluation of intermediate audio quality, sometimes test items in various foreign languages are presented. The perception of basic audio quality thereby may vary depending on the listeners' native language. This study investigated the role of understanding in quality assessment employing regular German sentences and sentences consisting of half-automatically generated German-sounding pseudo words. Especially less experienced listeners rated pseudo words slightly higher than German sentences of matching prosody. While references were heard longer for pseudo items, for other conditions they tend to hear German items longer. Though effects of understanding in our study were small, they may play a role in foreign languages that are less understandable than our pseudo sentences and differ in phoneme inventory.
Convention Paper 8771

4:00 pm

P13-4 Subjective Assessments of Higher Order Ambisonic Sound Systems in Varying Acoustical Conditions—*Andrew J. Horsburgh, Robert E. Davis, Martyn Moffat, D. Fraser Clark*, University of the West of Scotland, Paisley, Scotland, UK

Results of subjective assessments in source perception using higher order Ambisonics are presented in this paper. Test stimuli include multiple synthetic and naturally recorded sources that have been presented in various horizontal and mixed order Ambisonic listening tests. Using a small group of trained and untrained listening participants, materials were evaluated over various Ambisonic orders, with each scrutinized for localization accuracy, apparent source width and realistic impression. The results show a general preference for 3rd order systems in each of the three test categories: speech, pure tone, and music. Localization results for 7th order show a trend of stable imagery with complex stimuli sources and pure tones in the anechoic environment providing the highest accuracy.
Convention Paper 8772

4:00 pm

P13-5 A Viewer-Centered Revision of Audiovisual Content Classifiers—*Katrien De Moor,*[1] *Ulrich Reiter*[2]

[1]Ghent University, Ghent, Belgium
[2]Norwegian University of Science and Technology, Trondheim, Norway

There is a growing interest in the potential value of content-driven requirements for increasing the perceived quality of audiovisual material and optimizing the underlying performance processes. However, the categorization of content and identification of content-driven requirements is still largely based on technical characteristics. There is a gap in the literature when it comes to including viewer and content related aspects. In this paper we go beyond purely technical features as content classifiers and contribute to the deeper understanding of viewer preferences and requirements. We present results from a qualitative study using semi-structured interviews, aimed at exploring content-driven associates from a bottom-up perspective. The results show that users' associations, requirements, and expectations differ across different content types, and that these differences should be taken into account when selecting stimulus material for subjective quality assessments. We also relate these results to previous research on content classification.
Convention Paper 8773

4:00 pm

P13-6 Perception of Time-Varying Signals: Timbre and Phonetic JND of Diphthong—*Arthi Subramaniam, Thippur V. Sreenivas*, Indian Institute of Science, Bangalore, India

In this paper we propose a linear time-varying model for diphthong synthesis based on linear interpolation of formant frequencies. We, thence, determine the timbre just-noticeable difference

(JND) for diphthong /a^{I}/ (as in "buy") with a constant pitch excitation through perception experiment involving four listeners and explore the phonetic JND of the diphthong. Their JND responses are determined using 1-up-3-down procedure. Using the experimental data, we map the timbre JND and phonetic JND onto a 2-D region of percentage change of formant glides. The timbre and phonetic JND contours for constant pitch show that the phonetic JND region encloses timbre JND region and also varies across listeners. The JND is observed to be more sensitive to ending vowel /I/ than starting vowel /a/ in some listeners and dependent on the direction of perturbation of starting and ending vowels.
Convention Paper 8774

4:00 pm

P13-7 Employing Supercomputing Cluster to Acoustic Noise Map Creation—*Andrzej Czyzewski, Jozef Kotus, Maciej Szczodrak, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

A system is presented for determining acoustic noise distribution and assessing its adverse effects in short time periods inside large urban areas owing to the employment of a supercomputing cluster. A unique feature of the system is the psychoacoustic noise dosimetry implemented to inform interested citizens about predicted auditory fatigue effects that may be caused by the exposure to excessive noise. The noise level computing is based on the engineered Noise Prediction Model (NPM) stemmed from the Harmonoise model. Sound level distribution in the urban area can be viewed by users over the prepared www service. An example of a map is presented in consecutive time periods to show the capability of the supercomputing cluster to update noise level maps frequently.
Convention Paper 8775

4:00 pm

P13-8 Objective and Subjective Evaluations of Digital Audio Workstation Summing—
Brett Leonard,[1,2] *Scott Levine*,[1,2] *Padraig Buttner-Schnirer*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

Many recording professionals attest to a perceivable difference in sound quality between different digital audio workstations (DAWs), yet there is little in the way of quantifiable evidence to support these claims. To test these assertions, the internal summing of five different DAWs is tested. Multitrack stems are recorded into each DAW and summed to a single, stereo mix. This mix is evaluated objectively in reference to a purely mathematical sum generated in Matlab to avoid any system-specific limitations in the summing process. The stereo sums are also evaluated by highly trained listeners through a three-alternative forced-choice test focusing on three different DAWs. Results indicate that when panning is excluded from the mixing process, minimal objective and subjective differences exist between workstations.
Convention Paper 8776

4:00 pm

P13-9 Hong Kong Film Score Production: A Hollywood Informed Approach—*Robert Jay Ellis-Geiger*, City University of Hong Kong, Hong Kong, SAR China

This paper represents a Hollywood-informed approach toward film score production, with special attention given to the impact of dialogue on the recording and mixing of the music score. The author reveals his process for creating a hybrid (real and MIDI) orchestral film score that was recorded, mixed, and produced in Hong Kong for the English language feature film, *New York November* (2011). The film was shot in New York, directed by Austrian filmmakers Gerhard Fillei and Joachim Krenn in collaboration with film composer and fellow Austrian, Sascha Selke. Additional instruments were remotely recorded in Singapore and the final sound track was mixed at a dubbing theater in Berlin. The author acted as score producer, conductor, co-orchestrator, MIDI arranger, musician, and composer of additional music.
Convention Paper 8777

4:00 pm

P13-10 Investigation into Electric Vehicles Exterior Noise Generation—*Stefania Cecchi*,[1] *Andrea Primavera*,[1] *Laura Romoli*,[1] *Francesco Piazza*,[1] *Ferruccio Bettarelli*,[2] *Ariano Lattanzi*[2]
[1]Universitá Politecnica della Marche, Ancona (Ancona), Italy
[2]Leaff Engineering, Ancona, Italy

Electric vehicles have been receiving increasing interest in the last years for the well-known benefit that can be derived. However, electric cars do not produce noise as does an internal combustion engine vehicle, thus leading to safety issues for pedestrians and cyclists. Therefore, it is necessary to create an external warning sound for electric cars maintaining users' sound quality expectation. In this context several sounds generated with different techniques are here proposed, taking into consideration some aspects of real engine characteristics. Furthermore, a subjective investigation is performed in order to define users' preferences in the wide range of possible synthetic sounds.
Convention Paper 8778

Broadcast/Media Streaming Session 9
Saturday, October 27 4:00 pm – 6:00 pm
Room 134

WHAT HAPPENS TO YOUR PRODUCTION WHEN PLAYED BACK ON THE VARIOUS MEDIA

Chair: **David Bialik**, CBS, New York, NY, USA

Panelists: *Karlheinz Brandenburg*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany; Ilmenau University of Technology, ➡

Ilmenau, Germany
Frank Foti, Omnia
George Massenburg, McGill University, Montreal, Quebec, Canada
Greg Ogonowski, Orban, San Leandro, CA, USA
Robert Orban, Orban, San Leandro, CA, USA

Everyone has a different perspective when producing or playing back audio. This session will look at what happens to the audio product during the stages of recording, reproduction, digital playback, radio broadcast, and streaming. Is it the same experience for everyone?

Live Sound Seminar 6 — Saturday, October 27
4:00 pm – 6:00 pm — Room 120

WIRELESS FREQUENCY WRANGLING AT LARGE EVENTS

Chair: **Bob Green**, Audio-Technica U.S., Inc.

Panelists: *Dave Belamy*, Soundtronics
Steve Caldwell, Norwest Productions, Australia
Henry Cohen, Production Radio
Chris Dodds, P.A. People
Pete Erskine, Freelancer
Jason Eskew, Professional Wireless Systems, FL
Larry Estrin, Clear-Com

This session will cover the overall approach to RF coordination, preparation, and setup for a large event. Topics that will be discussed will include microphones, in-ear monitoring, and production communications with specific focus on antenna systems, cabling, RF environment isolation, monitoring, verifying operation, and procedures.

The members of the panel will bring to the table their experience (national and international) in events as diverse as Olympic Games Ceremonies, Super Bows, Presidential Debates, Grammy Awards, and Eurovision contests.

Networked Audio Session 3 — Saturday, October 27
4:00 pm – 6:00 pm — Room 123

AUDIO NETWORKS—PARADIGM SHIFT FOR BROADCASTERS

Chair: **Stefan Ledergerber**, Lawo Group, Zurich, Switzerland; LES Switzerland GmbH

Panelists: *Kevin Gross*, AVA Networks, Boulder, CO, USA
Andreas Hildebrand, ALC Networx
Sonja Langhans, Institut für Rundfuncktechnik, Munich, Germany
Lee Minich, Lab X Technologies, Rochester, NY, USA
Greg Shay, Telos Alliance/Axia, Cleveland, OH, USA
Kieran Walsh, Audinate Pty. Ltd., Ultimo, NSW, Australia

Today a variety of audio networking technologies are emerging. However, a number of questions related to workflow in broadcasting organizations seem still unanswered. This panel will try to find possible answers to some of the hot topics, such as:

• Will traditional crosspoint matrix switches (routers) disappear and fully be replaced by networks?
• Which component will deal with signal processing, which is currently done within audio routers?
• Which department is handling audio networks: audio or IT?
• How do we educate personnel handling audio networks?

The panelists will explain their views from a technology provider point of view, but lively participation by the audience is highly appreciated.

Audio Industry Seminar — Saturday, October 27
4:00 pm – 5:00 pm — Room 112

TAGAI EX-PORT LTD.

Solitone 2: The Way of Audio's Future. How to Make a Profit with Solitone

Presenter: **Péter Tagai**

This session is mainly for investors, but everyone is welcome who wants see what the possibilities are in Solitone. The benefits of this technology and the areas where it can be used where the most profit can be made will also be explained. We have also calculated the R/R ratio. The technology is ready for launch!

Project Studio Expo — Saturday, October 27
4:00 pm – 5:00 pm — PSE Stage

PSE6 - TAKE YOUR STUDIO ON STAGE: LIVE PERFORMANCE WITH LAPTOPS, LOOPING PEDALS, & OTHER STUDIO TECHNIQUES

Presenter: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

For many musicians, as well as DJs and electronic acts, a 21st century live performance requires much more than just a mixer and a bunch of amps. This workshop takes a practical look at how to use technology on stage without being overwhelmed by it, ways to insure a smooth performance, and includes invaluable information on the "care and feeding" of laptops to insure optimum performance—and uninterrupted performances. Other topics include using controllers for a more vibrant live performance, performing with Ableton Live and dedicated control surfaces, improvisation with looping pedals and DAW software, and the evolution of DJ controller / laptop combinations into tools for a musical, complex new art form.

Tutorial 7 — Saturday, October 27
4:15 pm – 5:15 pm — Room 133

THE MAGIC OF ANALOG TAPE

Presenter: **Mark Spitz,** ATR Service Co.; ATR Magnetics

While the digital modeling market continues to expand, there continues to be a sizable demand for recording with analog tape, but what continues to make using analog tape so special? This tutorial aims to illustrate many of the techniques and strategies to leverage this age-defying medium and empower the group with new skills to maximize their returns when recording with analog tape. This class provides a unique opportunity to gain knowledge from the industry's modern leader in analog tape technology, Mike Spitz, of ATR Service Co. and

ATR Magnetics. Learn from the master some of the secrets of maximizing calibration, formulas, levels, preparing mixes for vinyl disc cutting, and other techniques that make the analog tape medium sought after.

Game Audio Session 7 **Saturday, October 27**
4:15 pm – 5:45 pm **Room 131**

LOUDNESS ISSUES IN GAMES

Chair: **Steve Martz**, THX Ltd., San Rafael, CA, USA

Panelists: *Mike Babbitt*, Dolby Labs, San Francisco, CA, USA
Richard Cabot, Qualis Audio, Lake Oswego, OR, USA
Mark Yeend, Microsoft, Redmond, WA, USA

If it's too loud ….
Loudness wars in games have been hotly debated but without significant progress. Other industries have taken steps to rein in the content delivered to consumers. Are there parallels that can be applied to games? A panel of industry experts will review the present implementation of the broadcast industries' Commercial Advertisement Loudness Mitigation Act (CALM Act) of 2012 and investigate its potential application to the games industry. The panel will also discuss current attempts to address this issue amongst Publishers and Developers.

Audio Industry Seminar **Saturday, October 27**
4:30 pm – 5:30 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

The Collaborative Mix; Additional Production

Presenters: **Jon Castelli**
Tony Maserati

Since they can't have everyone over to their studio, multi Grammy-winning engineer/producer Tony Maserati (Mary J Blige, David Bowie, Usher, Puff Daddy, Notorious BIG, Jay-Z, R Kelly, Maxwell, Destiny's Child, Jennifer Lopez, Alicia Keys) will be joined by his team Outfit 27.

Outfit27 consists of Ron "Spider " Entwistle (GOTYE, Knife Party), Justin Hergett (Jason Mraz, Lady Gaga) , Chris Tabron (John Legend, Santigold), and Jon Castelli (GOTYE, Lady Gaga), and Tony.

Together they will present 3 sessions (Session Prep/Premix, Additional Production, and Mix-down) in which they will take a song from concept to final print and all "in the box." During these sessions they will actually prep, tweak, and mix a track in front of the audience complete with commentary as is customary at their place.

The ultimate opportunity to see Tony and his crew at work!!

Project Studio Expo **Saturday, October 27**
5:00 pm – 6:00 pm **PSE Stage**

M-AUDIO/AKAI PROFESSIONAL PRESENT: YOUNG GURU

Saturday October 27 4:30 pm Room 125
AES Standards Committee Meeting SC-07-01 on Metadata for Audio

Audio Industry Seminar **Saturday, October 27**
5:00 pm – 5:30 pm **Room 112**

THE CHINA AUDIO & VIDEO ENGINEERING GROUP

Application of professional loudspeakers with ultra thin depth.

Saturday October 27 5:00 pm Room 124

Technical Committee Meeting on Electro Magnetic Compatibility

Student/Career Event
SC-5 RECORDING COMPETITION—1
Saturday, October 27, 5:30 pm – 6:30 pm
Room 133

The Student Recording Competition is a highlight at each convention. A distinguished panel of judges participates in critiquing finalists of each category in an interactive presentation and discussion. This event presents stereo and surround recordings in this category:

- Sound for Visual Media 5:30 to 6:30
  Judges: Lora Hilschberg, Shawn Murphy

The top three finalists in each category, as identified by our judges, present a short summary of their production intentions and the key recording and mix techniques used to realize their goals. They then play their projects for all who attend. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2) on Monday afternoon. The competition is a great chance to hear the work of your fellow students at other educational institutions. Everyone learns from the judges' comments even if your project isn't one of the finalists, and it's a great chance to meet other students and faculty.

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Saturday, October 27, 6:30 pm – 8:00 pm
Room 134

Lecturer: **James "JJ" Johnston**

The Heyser Series is an endowment for lectures by eminent individuals with outstanding reputations in audio engineering and its related fields. The series is featured twice annually at both the United States and European AES conventions. Established in May 1999, The Richard C. Heyser Memorial Lecture honors the memory of Richard Heyser, a scientist at the Jet Propulsion Laboratory, who was awarded nine patents in audio and communication techniques and was widely known for his ability to clearly present new and complex technical ideas. Heyser was also an AES governor and AES Silver Medal recipient.

The Richard C. Heyser distinguished lecturer for the 133rd AES Convention is James Johnston. Many years ago, when he was an undergrad at Carnegie-Mellon University, Johnston mentioned to a couple of the professors that he found audio and radio to be interesting. Their reactions were quite clear, none of them thought it was even a remotely good idea. When pressed for a reason why, the responses were quite similar, in particular, he was told that there were too many people in audio who simply didn't make sense, that there was no money in audio, and that all the problems were solved. Note that this was in the days of early cassette recorders, huge Crown tape decks (well, remembered, but not fondly, for ➥

the relay problems), as well as Scully, Ampex, and the other famous makers, and the idea of digital audio wasn't even in the cards. Speech transmission had just started to completely switch over to digital signaling with mu-Law in the USA, and a-Law most everywhere else. The computer he used to feed cards to was a very fast 1 MIPS (and filled a room) and the timesharing box he could get a terminal on sometimes was a whopping 200 KIPS.

From there, Johnston went to Bell Labs, hired in by David Goodman and Jim Flanagan to do work on speech coding. Dave reported to Jim, who reported to Max Matthews, who was at that time still at the labs, doing computer music. A long story ensued, but he wound up doing work on ADPCM at much higher sampling rates than 8 kHz, with 2 to 12 bits of resolution, switch selected. That work indicated rather clearly, as did some follow-up work, that perceptual issues were a key to audio, in more or less every fashion. This turned out not to be too interesting to upper management, who, partially as a result of various legal issues, said, "we don't do audio." It was, however, also conveyed to him very clearly that audio wasn't the place to be, it was full of people who were a bit off, at least for the most part. As fate would have it, he spent his life working on audio signal processing at Bell Labs and AT&T Research, contributing heavily to MP3 and even more so to MPEG-2 AAC, then short periods at Microsoft in the multimedia division, and even shorter periods with Neural Audio and elsewhere working on spatial sensation, room correction, and loudness modeling, at which time Johnston decided it was time to pack it in regarding the corporate world." The title of his talk is "Audio, Radio, Acoustics, and Signal Processing-the Way Forward."

In this talk, Johnston will mention some experiences he's had, discuss our present understanding of human auditory perception in very broad terms, point out how the way we actually work (meaning our ears and brains) encourage a dichotomy of knowledge that no longer exists, and then suggest some ways that education on all sides (I think there are no longer "two" sides, if there ever were) can bring people closer together, apply some of the technical things we know on the artistic side, learn what the artistic side needs and wants, and solve some of the issues like "mix buss performance" claims, the performance of various processors (which are quite nonlinear, and for good reason), and so on. It is his hope that we can continue to push the understanding of perception into all sides of the equation in a more direct fashion, which should help create the "really here" immersive understanding that is the goal of the realists, create the "you could never be here but you wish you could" sensation of the avant-garde, and encourage the delivery systems of the world to "get with the program." It might be a dream, but there is a lot to gain by exchanging information, allowing information to be tested, learning what is really going on, and taking advantage of modern science, in the service of art.

Johnston's presentation will be followed by a reception hosted by the AES Technical Council.

Special Event
POE—A LIFE AND STORIES IN SOUND
Saturday, October 27, 8:00 pm – 9:00 pm
Room 131

Starring Phil Proctor of the Firesign Theater as Edgar Allen Poe, this one hour live audio drama performance will highlight Poe's life and stories in sound. Featuring scenes from Poe's most famous short stories and "moments" in his life this presentation will feature actors, live sound effects artists, recorded effects, and original music to bring to life Poe and his works.

The evening will feature the actor Phil Proctor (Firesign Theatre / Crazy Dog Audio Theater / Animation for Disney, etc.) as Edgar sharing moments from his life which led to writing works like "The Tell Tale Heart," "The Cask of Amontillado," and many others. In addition to Phil the cast will include voice actors Melinda Peterson, Lucien Dodge, and possibly one more actor who will help to bring to life Edgar's stories.

Produced and directed by award winning audio dramatist Sue Zizza with sound design by David Shinn of Sue-Media Productions.

Student/Career Event
STUDENT PARTY
Saturday, October 27, 9:00 pm – 11:00 pm
Coast Recorders / Michael Romanowski Mastering
1340 Mission Street

Audio Students! Join us for a fun and exciting evening at the AES Student Party to be held at Coast Recorders/ Michael Romanowski Mastering, an historic recording studio with the largest tracking room in San Francisco designed and built by the legendary Bill Putnam over 40 years ago. Tickets must be purchased in advance, either at the first meeting of the Student Delegate Assembly or at the AES Student Chapters Booth outside of the Exhibit Hall. The party will provide another opportunity for students to speak with industry mentors and will feature a special SPARS Legacy award presentation to Record Plant founder and industry stalwart Chris Stone.

Session P14 — Sunday, October 2
9:00 am – 11:30 am — Room 121

SPATIAL AUDIO OVER HEADPHONES

Chair: **David McGrath**, Dolby Australia, McMahons Point, NSW, Australia

9:00 am

P14-1 Preferred Spatial Post-Processing of Popular Stereophonic Music for Headphone Reproduction—*Ella Manor, William L. Martens, Densil A. Cabrera*, Sydney, NSW, Australia

The spatial imagery experienced when listening to conventional stereophonic music via headphones is considerably different from that experienced in loudspeaker reproduction. While the difference might be reduced when stereophonic program material is spatially processed in order to simulate loudspeaker crosstalk for headphone reproduction, previous listening tests have shown that such processing typically produces results that are not preferred by listeners in comparisons with the original (unprocessed) version of a music program. In this study a double blind test was conducted in which listeners compared five versions of eight programs from a variety of music genres and gave both preference ratings and ensemble stage width (ESW) ratings. Out of four alternative postprocessing algorithms, the outputs that were most preferred resulted from a nearfield crosstalk simulation mimicking low-frequency interaural level differences typical for close-range sources.
Convention Paper 8779

9:30 am

P14-2 Interactive 3-D Audio: Enhancing Awareness of Details in Immersive Soundscapes?—
Mikkel Schmidt,[1] *Stephen Schwartz*,[2] *Jan Larsen*[1]
[1]Technical University of Denmark, Kgs. Lyngby, Denmark
[2]SoundTales, Helsingør, Denmark

Spatial audio and the possibility of interacting with the audio environment is thought to increase listeners' attention to details in a soundscape. This work examines if interactive 3-D audio enhances listeners' ability to recall details in a soundscape. Nine different soundscapes were constructed and presented in either mono, stereo, 3-D, or interactive 3-D, and performance was evaluated by asking factual questions about details in the audio. Results show that spatial cues can increase attention to background sounds while reducing attention to narrated text, indicating that spatial audio can be constructed to guide listeners' attention.
Convention Paper 8780

10:00 am

P14-3 Simulating Autophony with Auralized Oral-Binaural Room Impulse Responses—*Manuj Yadav, Luis Miranda, Densil A. Cabrera, William L. Martens*, University of Sydney, Sydney, NSW, Australia

This paper presents a method for simulating the sound that one hears from one's own voice in a room acoustic environment. Impulse responses from the mouth to the two ears of the same head are auralized within a computer-modeled room in ODEON; using higher-order ambisonics for modeling the directivity pattern of an anthropomorphic head and torso. These binaural room impulse responses, which can be measured for all possible head movements, are input into a mixed-reality room acoustic simulation system for talking-listeners. With the system, "presence" in a room environment different from the one in which one is physically present is created in real-time for voice related tasks.
Convention Paper 8781

10:30 am

P14-4 Head-Tracking Techniques for Virtual Acoustics Applications—*Wolfgang Hess*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Synthesis of auditory virtual scenes often requires the use of a head-tracker. Virtual sound fields benefit from continuous adaptation to a listener's position while presented through headphones or loudspeakers. For this task position- and time-accurate, continuous robust capturing of the position of the listener's outer ears is necessary. Current head-tracker technologies allow solving this task by cheap and reliable electronic techniques. Environmental conditions have to be considered to find an optimal tracking solution for each surrounding and for each field of application. A categorization of head-tracking systems is presented. Inside-out describes tracking stationary sensors from inside a scene, whereas outside-in is the term for capturing from outside a scene. Marker-based and marker-less approaches are described and evaluated by means of commercially available products, e.g., the MS Kinect, and proprietary developed systems.
Convention Paper 8782

11:00 am

P14-5 Scalable Binaural Synthesis on Mobile Devices—*Christian Sander*,[1] *Frank Wefers*,[2] *Dieter Leckschat*[1]
[1]University of Applied Sciences Düsseldorf, Düsseldorf, Germany
[2]RWTH Aachen University, Aachen, Germany

The binaural reproduction of sound sources through headphones in mobile applications is becoming a promising opportunity to create an immersive three-dimensional listening experience without the need for extensive equipment. Many ideas for outstanding applications in teleconferencing, multichannel rendering for headphones, gaming, or auditory interfaces implementing binaural audio have been proposed. However, the diversity of applications calls for scalability of quality and performance costs so as to use and share hardware resources economically. For this approach, scalable real-time binaural synthesis on mobile platforms was developed and implemented in a test application in order to evaluate what current mobile devices are capable of in terms of binaural technology, both qualitatively and quantitatively. In addition, the audio part of three application scenarios was simulated.
Convention Paper 8783

Tutorial 8 **Sunday, October 28**
9:00 am – 11:00 am **Room 132**

AN OVERVIEW OF AUDIO SYSTEM GROUNDING AND INTERFACING

Presenter: **Bill Whitlock**, Jensen Transformers, Inc., Chatsworth, CA, USA; Whitlock Consulting, Oxnard, CA, USA

Equipment makers like to pretend the problems don't exist, but this tutorial replaces hype and myth with insight and knowledge, revealing the true causes of system noise and ground loops. Unbalanced interfaces are exquisitely vulnerable to noise due to an intrinsic problem. Although balanced interfaces are theoretically noise-free, they're widely misunderstood by equipment designers, which often results in inadequate noise rejection in real-world systems. Because of a widespread design error, some equipment has a built-in noise problem. Simple, no-test-equipment, troubleshooting methods can pinpoint the location and cause of system noise. Ground isolators in the signal path solve the fundamental noise coupling problems. Also discussed are unbalanced to balanced connections, RF interference, and power line treatments. Some widely used "cures" are both illegal and deadly.

Workshop 7 **Sunday, October 28**
9:00 am – 10:30 am **Room 130**

MUSHRA RELOADED

Chair: **Thomas Sporer**, Fraunhofer IDMT, Ilmenau, Germany

Panelists: *Poppy Crum*, Dolby, San Francisco, CA, USA
Frederik Nagel, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany; International Audio Laboratories, Erlangen, Germany

Since its finalization in 2001 Recommendation ITU-R BS.1534, nicknamed MUSHRA (multi stimulus with hidden reference and anchors), for testing audio quality has become very popular. It is widely used and to some extent misused. Many recent studies have also identified limitations of the current Recommendation. These limitations can have a direct impact on the interpretation of data collected using ITU-R BS.1534. ITU-R WP 6C is currently working on a revision of MUSHRA. The modification should preserve the general multiple-stimulus-hidden-reference (MUSHRA) design characteristic of ITU-R BS.1534 while improving the robustness and testing reliability. This workshop will present current discussions and results from the autumn 2012 meeting of ITU-R.

Workshop 8 **Sunday, October 28**
9:00 am – 11:00 am **Room 132**

THE CONTROVERSY OVER UPSAMPLING —BOON OR SCAM?

Chair: **Vicki R. Melchior**, Independent Technical Consultant, Audio DSP, Lynn, MA, USA

Panelists: *Poppy Crum*, Dolby Labs., San Francisco, CA, USA
Robert Katz, Digital Domain Mastering, Altamonte Springs, FL, USA
Bruno Putzeys, Hypex Electronics, Rotselaar, Belgium; Grimm Audio
Rhonda Wilson, Dolby Laboratories, San Francisco, CA, USA

Many “high resolution” releases on Blu-ray, DVD, and also some HD downloads are created by upsampling Redbook or 48-kHz data, a practice that draws frequent and quite vehement outcries of fraud. Yet at the same time upsamplers, both hardware and software, are commonly marketed to consumers and professionals now with the promise of boosting redbook sonics to near-equality with high resolution. What's going on? A panel of experienced mastering engineers, DAC, and DSP designers looks at the long-standing controversies and claims behind upsampling, as well as its uses in audio. The issues relating to down/up conversion go well beyond headroom and relate directly to hardware and software implementation, the design of low pass filters, consequences of signal processing, and mastering considerations related to the production and release of modern high quality discs and music files.

Workshop with Height 1 **Sunday, October 28**
9:00 am – 11:00 am **Pyramind**
832 Folsom St.

SOUND DESIGN TOOLS FOR MULTICHANNEL AUDIO WITH HEIGHT

Chair: **Wieslaw Woszczyk**, McGill University, Montreal, QC, Canada

Panelists: *Kimio Hamasaki*, NHK, Chiba-ken, Japan
Richard King, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
Brett Leonard, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
Frank Melchior, BBC R&D, Salford, UK
Nicolas Tsingos, Dolby Labs, San Francisco, CA, USA

Multichannel audio systems in advanced cinema and television utilize many loudspeaker channels distributed horizontally and vertically in the listening space. Comprehensive tools have been developed to control sound design and mixing in such environments as traditional stereophonic tools are inadequate. The panelists will present different channel, object based, and hybrid approaches and demonstrate how the respective systems are used in post-production to achieve artistic flexibility and save time. Discussion on merits of these solutions will follow. [Workshop will be held at Pyramind, 832 Folsom Street (a 10-minute walk from Moscone Center). Space is limited and free tickets can be obtained at the *AES Tours Desk* [full program badge required].]

Broadcast/Media Streaming Session 10
Sunday, October 28 **9:00 am – 10:30 am**
Room 131

SOUND DESIGN: HOW DOES THAT "THING" GO BUMP IN THE NIGHT?

Presenters: **David Shinn**, SueMedia Productions
Sue Zizza, SueMedia Productions

Whether you are working with props, recording sounds on location, or using pre-recorded sounds from a library, the sound design elements you choose will impact the aesthetics of the stories you tell. Working with scenes from the AES Performance, *Poe A Life and Stories in Sound* (see Special Events), this session will examine working with sound effects props in the studio and recording elements on location. Recording and performance techniques will be discussed. A brief overview of sound effects libraries will also be included.

Networked Audio Session 4 **Sunday, October 28**
9:00 am – 10:30 am **Room 123**

AVNU – THE UNIFIED AV NETWORK: OVERVIEW AND PANEL DISCUSSION

Chair: **Rob Silfvast**, Avid

Panelists: *Ellen Juhlin*, Meyer Sound
Denis Labrecque, Analog Devices
Lee Minich, LabX Technologies; Chair of AVnu Marketing Workgroup
Bill Murphy, Extreme Networks
Michael Johas Teener, Broadcom

This session will provide an overview of the AVnu Alliance, a consortium of audio and video product makers and core technology companies committed to delivering an interoperable open standard for audio/video networked connectivity built upon IEEE Audio Video

Bridging standards. AVnu offers a logo-testing program that allows products to become certified for interoperability, much like the Wi-Fi Alliance provides for the IEEE 802.11 family of standards. Representatives from several different member companies will speak in this panel discussion and provide insights about AVB technology and participation in the AVnu Alliance.

Product Design Session 6 **Sunday, October 28**
9:00 am – 10:30 am **Room 120**

MULTIMEDIA DEVICE AUDIO ARCHITECTURE

Presenter: **Laurent Le Faucheur**, Texas Instruments

A hardware audio architecture solves several mobile low power multimedia application processor constraints related to: legacy software reuse, signal processing performance, power optimization, multiple data format interfaces, low latency voice and tones, low system costs. The presented audio architecture is optimized for solving those constraints with several assets: a powerful DSP, a low-power Audio Back-End processor, and a high-performance mixed-signal audio device. The DSP audio framework enables the integration of multiple audio features developed by third-parties.

Student/Career Event
SC-6 STUDENT DESIGN COMPETITION
Sunday, October 28, 9:00 am – 10:30 am
Room 122

Moderator: **Scott Dorsey**, Williamsburg, VA, USA

The three graduate level and three undergraduate level finalists of the AES Student Design Competition will present and defend their designs in front of a panel of expert judges. This is an opportunity for aspiring student hardware and software engineers to have their projects reviewed by the best minds in the business. It's also an invaluable career-building event and a great place for companies to identify their next employees.

Students from both audio and non-audio backgrounds are encouraged to submit entries. Few restrictions are placed on the nature of the projects, but designs must be for use with audio. Examples include loudspeaker design, DSP plug-ins, analog hardware, signal analysis tools, mobile applications, and sound synthesis devices. Products should represent new, original ideas implemented in working-model prototypes.

Judges: John La Grou, Dave Collins, Bill Whitlock, W.C. (Hutch) Hutchison

Sunday October 28 **9:00 am** **Room 124**

Technical Committee Meeting on Sound for Digital Cinema and Television (formative)

Sunday October 28 **9:00 am** **Room 125**

AES Standards Committee Meeting SC-04-04 on Microphone Measurement and Characterization

Game Audio Session 8 **Sunday, October 28**
9:30 am – 10:30 am **Room 133**

AUDIO SHORTS: RESOURCES

Presenters: **Charles Deenen**, Electronic Arts
Tom Salta
Stephan Schütze, Sound Librarian

This hour long session will be split into three twenty-minute segments. Each segment will go in depth into a subject that is near and dear to the presenter. Audio Shorts is designed to pack in as much usable information in as short of period of time as possible. It's like the Reader's Digest of game audio tutorials. You won't want to miss this one.

Shorty #1: Tools, Tips, and Techniques, Tom Salta, presenter

Shorty #2: Sound Libraries, Stephan Schutze, presenter

Shorty #3: My Favorite Plug-in!, Charles Deenen, presenter

Sunday October 28 **10:00 am** **Room 124**

Technical Committee Meeting on Fiber Optics for Audio

Student/Career Event
SC-7 STUDENT DESIGN EXHIBITION
Sunday, October 28, 10:30 am – 12:00 noon
Foyer

All accepted student entries to the AES Student Design Competition will have the opportunity to show off their designs at this poster/tabletop exhibition. This audio "science fair" is free and open to all convention attendees and is an opportunity for aspiring student hardware and software engineers to have their projects seen by the AES design community. It's also an invaluable career-building event and a great place for companies to identify their next employees.

Students from both audio and non-audio backgrounds are encouraged to submit entries. Few restrictions are placed on the nature of the projects, but designs must be for use with audio. Examples include loudspeaker design, DSP plug-ins, analog hardware, signal analysis tools, mobile applications, and sound synthesis devices. Products should represent new, original ideas implemented in working-model prototypes.

Audio Industry Seminar **Sunday, October 28**
10:15 am – 11:00 am **Room 114**

PMC: MASTERS OF AUDIO SERIES

PMC Playback Sessions

PMC will play a selection of extraordinary recordings AND give you the chance to listen to your own music and projects

Audio Industry Seminar **Sunday, October 28**
10:30 am – 12:00 noon **Room 112**

THE METALLIANCE

The METAlliance Presented by Sanken Studio Mics

Presenters: **Chuck Ainlay**
Ed Cherney
Frank Filipetti
George Massenburg
Phil Ramone
Elliot Scheiner
Al Schmitt

The METAlliance will discuss the current opportunities for improvement in audio quality. All attendees are welcome to join us in this informal meet-and-greet atmosphere and participate in this industry forum.

Broadcast/Media Streaming Session 11
Sunday, October 28 10:45 am – 12:15 pm
Room 131

LISTENER FATIGUE AND RETENTION

Chair: **Dave Wilson**, CEA

Panelists: *Stephen Ambrose*, Asius Technologies
J. Todd Baker, DTS
Sean Olive, Harman International, Northridge, CA, USA
Robert Reams, Streaming Appliances/DSP Concepts, Mill Creek, WA, USA
Bill Sacks, Optimod Refurbishing, Hollywood, MD, USA

This panel will discuss listener fatigue and its impact on listener retention. While listener fatigue is an issue of interest to broadcasters, it is also an issue of interest to telecommunications service providers, consumer electronics manufacturers, music producers, and others. Fatigued listeners to a broadcast program may tune out, while fatigued listeners to a cell phone conversation may switch to another carrier, and fatigued listeners to a portable media player may purchase another company's product. The experts on this panel will discuss their research and experiences with listener fatigue and its impact on listener retention.

Game Audio Session 9 Sunday, October 28
10:45 am – 12:45 pm Room 133

DEMO DERBY

Presenters: **Paul Gorman**, Electronic Arts
Paul Lipson, Microsoft
Jonathan Mayer, Sony Computer Entertainment America
Dren McDonald, Loot Drop
Don Veca, Sledgehammer Games at Activision

The Demo Derby is now at AES. Bring your best demo material and have it reviewed by the Pros. Let's see you have what it takes to make it in games.

Music:

Attendees submit 60 seconds of their best work for a detailed critique and feedback from a team of leading audio directors and professionals and participate in an active discussion with fellow panelists and audience members. The Derby facilitates game audio practitioners of all levels and is suited for producers, composers, audio directors, and anyone interested in music for games and interactive entertainment.

Sound Design:

Attendees submit 120 seconds of their best work for a detailed critique and feedback from a team of leading audio directors and professionals and participate in an active discussion with fellow panelists and audience members. The Derby facilitates game audio practitioners of all levels and is suited for producers, composers, audio directors, and anyone interested in music for games and interactive entertainment.

Bring a CD/DVD with only 1 demo track on it. This will be played on a disc player, not a computer. Please author the disc so that your demo auto plays immediately.

Submissions will be collected 15 minutes before the session begins.

Historical Program
H3 - LEE DE FOREST: THE MAN WHO INVENTED THE AMPLIFIER
Sunday, October 28, 10:45 am – 11:45 am
Room 122

Presenter: **Mike Adams**, San Jose State University, San Jose, CA, USA

After Lee de Forest received his PhD in physics and electricity from Yale University in 1899, he spent the next 30 years turning the 19th century science he learned into the popular audio media of the 20th century. First he added sound to the wireless telegraph of Marconi and created a radiotelephone system. Next, he invented the triode vacuum tube by adding a control grid to Fleming's two-element diode tube, creating the three-element vacuum tube used as an audio amplifier and oscillator for radio wave generation. Using his tube and building on the earlier work of Ruhmer and Bell, he created a variable density sound-on-film process, patented it, and began working with fellow inventor Theodore Case. In order to promote and demonstrate his process he made hundreds of short sound films, found theaters for their showing, and issued publicity to gain audiences for his invention. While de Forest did not profit from sound-on-film, it was his earlier invention of the three-element vacuum tube that allowed amplification of audio through loudspeakers for radio and the movies that finally helped create their large public audiences.

Special Event
DOUBLE YER MONEY ! ... AN HOUR THAT COULD CHANGE YOUR EARNING PROSPECTS FOREVER
Sunday, October 28, 10:45 am – 12:15 pm
Room 120

Moderator: **Peter Filleul**, Producer Rights Activist and Chairman of the European Sound Directors' Association

Presentations and discussion that looks at the implications of recording in countries that enjoy neighbouring rights and revenues. Explore how recording in the UK can entitle performers (and producers) to significant streams of revenue. Hear the latest developments at PPL (the collection society in the UK) that smooth the way for claims from "Eligible Studio Producers" who record in the UK and have the criteria that PPL have identified that will enable a studio producer to make a claim explained. How will this effect U.S. and UK recording studios? Will UK studios, keen to attract work from the USA, have a special competitive edge on their U.S. counterparts? How will the new WIPO Audio Visual Performances Treaty impact on performer revenues in the USA—and when? Will these economic realities add leverage to efforts to acquire similar terrestrial rights in the USA.?

This Special Event will be conducted by Producer Rights activist and Chairman of the European Sound Directors' Association, Peter Filleul and attended by key P&E Wing council members and will include an A/V presentation and Q&A session.

Live Sound Seminar 7 Sunday, October 28
11:00 am – 12:30 pm Room 130

THE WOMEN OF PROFESSIONAL CONCERT SOUND

Chair: **Terri Winston,** Women's Audio Mission, San Francisco, CA, USA

Panelists: *Claudia Engelhart*, FOH engineer for Bill Frisell, Herbie Hancock, Waye Shorter
Deanne Franklin, FOH engineer for Tom Waits, David Byrne, Pink
Karrie Keyes, Monitor Engineer for Pearl Jam, Red Hot Chili Peppers, Sonic Youth
Jeri Palumbo, Live Television Production Engineer, Super Bowl, The Oscars, Tonight Show w/Jay Leno
Michelle Sabolchick Pettinato, FOH engineer for Gwen Stefani, Jewel, Melissa Etheridge

This all-star panel of live sound engineers averages over 250 days a year mixing the biggest name acts in the business. Drawing from their experience running sound in arenas and large venues all over the world, these women will share their tips and tricks, from using EQ and learning the problematic frequencies of instruments to choosing the best outboard gear and the systems typically used. This panel will explore what a day on a major world tour looks like, how to adjust to the acoustics of different venues, the difference between the positions of FOH and monitors, and how to successfully manage a life of constant touring.

Networked Audio Session 5 — Sunday, October 28
11:00 am – 12:30 pm — Room 123

INTEROPERABILITY ISSUES IN AUDIO TRANSPORT OVER IP-BASED NETWORKS

Chair: **Tim Shuttleworth**, Oceanside, CA, USA

Panelists: *Kevin Gross*, AVA Networks, Boulder, CO, USA
Sonja Langhans, Institut für Rundfuncktechnik, Munich, Germany
Lee Minich, Lab X Technologies, Rochester, NY, USA
Greg Shay, Telos Alliance/Axia, Cleveland, OH, USA

This workshop will focus on interoperability issues in two areas of audio/media transport over IP based networks. These are:

• Multichannel Audio distribution over Ethernet LANs for low latency, high reliability interconnections in home, automobile, and commercial environments. Interoperability standards and methods based on the Ethernet AVB suite of IEEE standards as well as the AES X-192 interoperability project shall be discussed.

• Audio Contribution over Internet Protocol (ACIP and ACIP2) interoperability issues will be discussed from both a European and U.S. perspective with presenters discussing activities within the EBU community and the U.S. broadcasting market. Audio over IP methods are being widely used in remote broadcast situations. The challenges and solutions in achieving reliable content distribution shall be examined.

Cross-vendor operability is becoming increasingly demanded in all audio applications markets. This topic will be of interest to audio systems designers and users across the gamut of market segments. Two presenters will provide their overview within each of the three topic areas.

Audio Industry Seminar — Sunday, October 28
11:00 am – 11:30 am — Room 110

JBL PROFESSIONAL

"Big Sound" in Small and Medium Size Rooms: Next-Gen Transducer, Acoustic and Electronic Technologies Make High-Performance, Large Format Monitoring Accessible to a Broad Range of Rooms

Presenters: **Peter Chaikin**, Sr. Manager, Recording and Broadcast Marketing, JBL Professional
Allan Devantier, Manager of Objective Evaluation, Harman International
Charles Sprinkle, Sr. Acoustic Systems Engineer, Harman International

To date, space, acoustic and budgetary limitations have prevented smaller rooms from enjoying the high output, detailed monitoring experience of the large control room. For these rooms, near field monitors are the fall back solution. At AES, JBL Professional introduces a large-format monitoring system in a form-factor and at a price point that make large format monitoring a viable option for project studios and small mix rooms. We will discuss the enabling technologies and demonstrate the system.

** Session times on Sunday are: 11:00, 11:30, 12:00, 12:30, 3:00, 3:30, 4:00, and 4:30. Each session is 30 minutes.*

Project Studio Expo — Sunday, October 28
11:00 am – 12:00 noon — PSE Stage

PSE7 - KEEPING THE HUMAN ELEMENT IN THE DIGITAL AGE

Presenter: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

Ways to keep music sounding alive and interesting.

Sunday October 28 — 11:00 am — Room 124

Technical Committee Meeting on Signal Processing

Sunday October 28 — 11:00 am — Room 125

AES Standards Committee Meeting SC-04-03 on Loudspeaker Modeling and Measurement

Workshop with Height 2 — Sunday, October 28
11:15 am – 12:45 pm — Pyramind
832 Folsom Street

HEIGHT CHANNELS: ADDING THE VERTICAL DIMENSION TO SURROUND SOUND

Chair: **Paul Geluso**, New York University, New York, NY, USA

Panelists: *Tom Ammermann*, New Audio Technology GmbH, Hamburg, Germany
David Bowles, Swineshead Productions LLC, Berkeley, CA, USA
Gregor Zielinsky, Sennheiser Electronic GmbH & Co. KG, Germany

While a great deal of development has gone into reproducing audio in the horizontal plane, the next step in immersive audio moves into the vertical dimension. How does one capture and create height information for a true 3-D listening experience? How does the listener perceive the addition of height loudspeakers? New recording techniques, microphone configurations, processing methods, and transmission formats are needed to fill this demand. At this workshop a panel of music producers and engi- ➥

neers will present different methods to capture and process vertical ("Z axis") information for later reproduction in surround playback environments with height channels. Post-production methods to create 3-D sonic imagery for recordings not originating in 3-D will be discussed as well. Recordings made using these technologies will be demonstrated. [Workshop will be held at Pyramind, 832 Folsom Street (a 10-minute walk from Moscone Center). Space is limited and free tickets can be obtained at the *AES Tours Desk* [full program badge required].]

Audio Industry Seminar **Sunday, October 28**
11:15 am – 12:30 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

The Collaborative Mix: Mixdown with Tony Maserati

Presenter: **Tony Maserati**

Since they can't have everyone over to their studio, multi Grammy-winning engineer/producer Tony Maserati (Mary J Blige, David Bowie, Usher, Puff Daddy, Notorious BIG, Jay-Z, R Kelly, Maxwell, Destiny's Child, Jennifer Lopez, Alicia Keys) will be joined by his team Outfit 27.

Outfit 27 consists of Ron "Spider" Entwistle (GOTYE, Knife Party), Justin Hergett (Jason Mraz, Lady Gaga), Chris Tabron (John Legend, Santigold), and Jon Castelli (GOTYE, Lady Gaga), and Tony.

Together they will present 3 sessions (Session Prep/Premix, Additional Production, and Mix-down) in which they will take a song from concept to final print and all "in the box." During these sessions they will actually prep, tweak, and mix a track in front of the audience complete with commentary as is customary at their place.

<u>The ultimate opportunity to see Tony and his crew at work!!</u>

Product Design Session 7 **Sunday, October 28**
11:45 am – 12:45 pm **Room 121**

A NEXT GENERATION AUDIO PROCESSING SUITE FOR THE ENHANCEMENT OF ACOUSTICALLY CHALLENGED DEVICES

Presenter: **Alan Seefeldt**, Dolby Laboratories, San Francisco, CA, USA

This tutorial will describe the design principles and algorithms behind a recently released commercial audio processing suite intended to enhance the sound of acoustically challenged devices such as laptops, tablets, and mobile phones. The suite consists of numerous algorithms, all operating within a common frequency domain framework, with several of these algorithms tuned specifically for the acoustics of the device on which it operates.

Workshop 9 **Sunday, October 28**
12:00 noon – 1:00 pm **Room 122**

ACOUSTIC AND AUDIO IPHONE APPS

Chair: **Peter Mapp**, Peter Mapp Associates, Colchester, Essex, UK

A range of audio and acoustic apps are available for the iPhone, iPad, and other smartphones. Both Measurement and Calculation apps are available. The workshop will review current apps and discuss some of their uses and limitations.

Audio Industry Seminar **Sunday, October 28**
12:00 noon – 1:00 pm **Room 112**

DAN DUGAN SOUND DESIGN

Features and Applications of Dugan Automatic Mixers

Presenter: **Dan Dugan**

Dan will present a brief tutorial on the history and theory of automatic mic mixing. He will explain the application of the various features of his current Model E-1A, Model D-2, Model E-3, and Dugan-MY16 automatic mixing controllers.

Project Studio Expo **Sunday, October 28**
12:00 noon – 1:00 pm **PSE Stage**

PSE8 - TOTAL TRACKING: GET IT RIGHT AT SOURCE—CHOOSING & RECORDING YOUR SOUND SOURCE

Presenters: **Bill Gibson**, Art Institute of Seattle, Seattle, WA, USA
Hugh Robjohns, Technical Editor, Sound on Sound, Crowle, UK

The astonishing and ever-improving power and versatility of digital signal processing plug-ins for computer audio workstations has encouraged the widespread belief that everything can be "fixed in the mix"—and in many cases, of course, it can. However, this approach is always extremely time-consuming and the results aren't always perfect. It is often much faster, and with far more satisfying results, to get the right sound from the outset by careful selection of the source and appropriate microphone selection and positioning. This workshop will explore a wide variety of examples, analyzing the requirements and discussing practical techniques of optimizing source recordings.

Sunday October 28 **12:00 noon** **Room 124**

Technical Committee Meeting on Audio for Telecommunications

Historical Program
H4 - THE EGG SHOW: A DEMONSTRATION OF THE ARTISTIC USES OF SOUND ON FILM
Sunday, October 28, 1:30 pm – 4:00 pm
Dolby Laboratories Theater

Presenter: **Ioan Allen**, Dolby Laboratories Inc., San Francisco, CA, USA

The Egg Show is a two-hour entertaining and evocative look at the influences of sound mixing on motion picture presentation. This educational and exquisitely entertaining survey of sound design and acoustical properties in film has been presented at many industry events, film festivals, technology seminars, and film schools with great success. With pictures of eggs ingeniously illustrating his points, and an array of 35-mm clips, Mr. Allen explains mixing and editing techniques that sound designers and filmmakers have developed over the years. Many film examples are used, each of which has been chosen to demonstrate a different mixing issue. You will see how film sound mixers highlight a conversation in a crowd scene, inter-cut music to film and create artificial sound effects that sound more real on film than on a live recording. Numerous slides enhance the lecture, most of which use photographs of eggs to illustrate the artistic uses of sound on film.

Location: This event will be held at Dolby Laboratories' Theater, a multi-format projection room/recording studio complex within Dolby's South-of-Market Dolby headquarters. The facility was designed as an ideal listening and viewing environment for film, recorded sound, and live presentations.

A limited number of $10 tickets will be available exclusively to registered convention attendees at the tours counter in the main lobby at Moscone. The marked bus will depart at 1:30 for the short ride to Dolby.

Student/Career Event
SC-8 STUDENT RECORDING CRITIQUES
Sunday, October 28, 12:30 pm – 1:30 pm
Room 114

Moderator: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA

Students! Bring in your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or surround work on CD, DVD, memory-stick, or hard disc, as clearly-labeled 44.1KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC.

Broadcast/Media Streaming Session 12
Sunday, October 28 12:45 pm – 1:45 pm
Room 131

TROUBLESHOOTING SOFTWARE ISSUES

Chair: **Jonathan Abrams**, Nutmeg Post, Jersey City, NJ, USA

Panelists: *Connor Sexton*, Avid Technology
Charles Van Winkle, Adobe, Minneapolis, MN, USA

What should you do before contacting support? How do you make the most of online support resources? What kind of audio driver are you using and how does that interact with the rest of your system? What role does your plug-in platform play when troubleshooting? How can permissions wreak havoc on your system or workflow?

Get the answers to these questions and bring your own for Mac OS X, Windows, Adobe Audition, and AVID Pro Tools.

Audio Industry Seminar Sunday, October 28
1:00 pm – 3:00 pm Room 112

AUDIO PRECISION (3 Sessions)

Choose Wisely: Selecting the Right Audio Test Technique for the Job

Presenters: **Dan Foley**
Christopher Gill
Jonathan Novick

In three open sessions, three of Audio Precision's FAEs will share best practices gained through nearly 60 years of cumulative audio test experience. Questions welcome.

- Proper grounding principles for audio test and measurement: 1:00 pm – 1:30 pm.
- Test methods—when to use sine sweeps vs two-tone/multitone vs noise: 1:30 pm – 2:10 pm
- Measurement of ASIO devices: 2:10 pm – 2:45 pm.

Project Studio Expo Sunday, October 28
1:00 pm – 2:00 pm PSE Stage

PSE9 - MIXING SECRETS: PRODUCTION TRICKS TO USE WITH ANY DAW

Presenter: **Mike Senior**, Sound On Sound, Munich, Germany; Cambridge Music Technology

Affordable DAW software now provides all the processing tools you need to create commercially competitive music mixes within a home, college, or project studio. As such, the overriding concern for budget-conscious engineers these days should be to develop effective habits with regard to studio monitoring, mix balancing, and quality control. Important techniques in each of these three areas are often neglected in small-scale productions, leading to mixes that don't stack up against professional releases, or that collapse on some mass-market listening systems. In this seminar, Sound On Sound magazine's "Mix Rescue" columnist Mike Senior will draw on his experience of thousands of project-studio mixes to highlight the most frequently overlooked studio tricks. In the process he'll demonstrate how these methods can powerfully upgrade your sonics without breaking the bank, no matter which DAW you're using.

Sunday October 28 1:00 pm Room 124

Technical Committee Meeting on Audio for Games

Sunday October 28 1:00 pm Room 125

AES Standards Committee Meeting SC-02-01 on Digital Audio Measurement Techniques

Special Event
PLATINUM PRODUCERS AND ENGINEERS
Sunday, October 28, 1:30 pm – 3:00 pm
Room 134

Moderator: **Michael Romanowski**, NARAS

Panelists: *Jack Douglas* (John Lennon, Miles Davis, Aerosmith, Alice Cooper)
Narada Michael Walden (Aretha Franklin, Whitney Houston, Jeff Beck, Mariah Carey, The Bodyguard Soundtrack)
Young Guru (Jay-Z, Beyonce, Rihanna, Ludacris)

President of the NARAS Board of Governors and mastering ace Michael Romanowski moderates a superstar panel of world-renowned producers and engineers who have inspired and produced some of today's most memorable and iconic recordings. Participants will discuss the creative recording and mixing techniques they've developed, playing samples of their work to illustrate their most successful collaborations. They will reminisce about their most memorable moments in the studio and talk about what it's like to work with artists over an impressively wide variety of genres including rock, jazz, pop, hip hop, R&B and fusion. Finally, they will reflect on how the recording process has changed over the years and how it has affected their creativity in the studio.

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Student/Career Event
SC-9 EDUCATION FORUM PANEL
Sunday, October 28, 1:30 pm – 3:30 pm
Room 130

Moderator: **John Krivit**, New England Institute of Art, Brookline, MA, USA; Emerson College, Bay State College

Presenters: *Bill Crabtree*, Middle Tennessee State University, Murfreesboro, TN, USA
David Greenspan, University of Michigan, Ann Arbor, MI, USA
David MacLaughlin, New England School of Communications
John Scanlon, Expression College for Digital Arts, Emeryville, CA, USA
Terri Winston, Women's Audio Mission, San Francisco, CA, USA

What Kinds of Studios Are Built for Education?

What kinds of studios and facilities are built for education? Five noted educators from diverse programs will share the decision-making process that informs the construction of various college level facilities. How do curriculums and pedagogy prescribe construction and equipment choices? Or does this too-often work in reverse when existing space and budget limitations dictate how and what will be taught? Quick presentations will be followed by a moderated open discussion among educators, manufacturers, students, employers, and anyone with a stake in the next generation of audio professionals.

Audio Industry Seminar **Sunday, October 28**
1:30 pm – 2:30 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

Mastering with Impact

Presenter: **Maor Appelbaum**, Maor Appelbaum Mastering, Los Angeles, CA, USA

Maor Appelbaum is a mastering engineer and musician. After working as staff engineer (mixing, recording, and mastering) for famed record producer Sylvia Massy Shivy at Radiostar Studios in Weed, California, Maor moved to Los Angeles where he opened his private mastering suite.

To Maor, being a mastering engineer is the best way possible to combine his love and passion for music, with his various skills- objectivity, subjectivity, and technical & artistic prowess. He finds pleasure in his job, more than anything, thanks to the variety of music and sounds he gets to master from all over the world. It is a profession he takes pride in, and masters. He will share this passion with you!

Workshop with Height 3 **Sunday, October 28**
1:45 pm – 3:45 pm **Pyramind**
832 Folsom Street

RECORDING MUSIC IN 9.1 HEIGHT SURROUND

Chair: **Morten Lindberg**, 2L (Lindberg Lyd AS), Oslo, Norway

Panelists: *Stefan Bock*, msm-studios GmbH, Munich, Germany
Wilfried Van Baelen, Auro-3D, Mol, Belgium

Balance Engineer and Recording producer Morten Lindberg will present Norwegian 2L's approach to recording music in 9.1 surround sound. Playing original navigation for high-resolution audio on Blu-ray Disc. 352.8 kHz/24 bit masters, showing photos and stage layout from recording sessions and discussing the resources and considerations involved in the process of venue recording, editing, mix, and mastering. The workshop will also discuss how 9.1 can be encoded to the Blu-ray format and how the disc can be programmed for a CD-player-style screen-less navigation as Pure Audio Blu-ray. Morten Lindberg has produced twelve GRAMMY nominations since 2006 (seven of these for "Best Engineered Album" and "Best Surround Sound Album"). Wilfried Van Baelen is the founder of Galaxy Studios in Belgium and the creative force behind the new Auro-3D codec. Stefan Bock is the founder of msm-studios in Germany and the author of the AES-21id, Screen-less navigation for high-resolution audio on Blu-ray Disc. [Workshop will be held at Pyramind, 832 Folsom Street (a 10-minute walk from Moscone Center). Space is limited and free tickets can be obtained at the *AES Tours Desk* [full program badge required].]

Product Design Session 8 **Sunday, October 28**
1:45 pm – 2:45 pm **Room 123**

IMPLEMENTING APPLICATION PROCESSOR AGNOSTIC AUDIO SYSTEMS FOR PORTABLE CONSUMER DEVICES

Presenter: **Jess Brown**, Wolfson Micro Ltd.

This tutorial will outline the audio future trends of the portable consumer device and demonstrate how this is achieved with advanced audio solutions. Covering areas, such as HD audio voice, capture, playback, and share, this tutorial will outline the total "mouth to ear" audio solution, from a technology and device standpoint.

Session P15 **Sunday, October 21**
2:00 pm – 5:30 pm **Room 121**

SIGNAL PROCESSING FUNDAMENTALS

Chair: **Lars Villemoes**, Dolby Sweden, Stockholm, Sweden

2:00 pm

P15-1 Frequency-Domain Implementation of Time-Varying FIR Filters—*Earl Vickers*, ST Microelectronics, Inc., Santa Clara, CA, USA

Finite impulse response filters can be implemented efficiently by means of fast convolution in the frequency domain. However, in applications such as speech enhancement or channel upmix, where the filter is a time-varying function of the input signal, standard approaches can suffer from artifacts and distortion due to circular convolution and the resulting time-domain aliasing. Existing solutions can be computationally prohibitive. This paper compares a number of previous algorithms and presents an alternate method based on the equivalence between frequency-domain convolution and time domain windowing. Additional computational efficiency can be attained by careful choice of the analysis window.
Convention Paper 8784

2:30 pm

P15-2 Estimating a Signal from a Magnitude Spectrogram via Convex Optimization—*Dennis L. Sun, Julius O. Smith, III*, Stanford University, Stanford, CA, USA

The problem of recovering a signal from the magnitude of its short-time Fourier transform (STFT) is a longstanding one in audio signal processing. Existing approaches rely on heuristics that often perform poorly because of the nonconvexity of the problem. We introduce a formulation of the problem that lends itself to a tractable convex program. We observe that our method yields better reconstructions than the standard Griffin-Lim algorithm. We provide an algorithm and discuss practical implementation details, including how the method can be scaled up to larger examples.
Convention Paper 8785

3:00 pm

P15-3 Distance-Based Automatic Gain Control with Continuous Proximity-Effect Compensation —*Walter Etter*, Bell Labs, Alcatel-Lucent, Murray Hill, NJ, USA

This paper presents a method of Automatic Gain Control (AGC) that derives the gain from the sound source to microphone distance, utilizing a distance sensor. The concept makes use of the fact that microphone output levels vary inversely with the distance to a spherical sound source. It is applicable to frequently arising situations in which a speaker does not maintain a constant microphone distance. In addition, we address undesired bass response variations caused by the proximity effect. Knowledge of the sound-source to microphone distance permits accurate compensation for both frequency response changes and distance-related signal level changes. In particular, a distance-based AGC can normalize these signal level changes without deteriorating signal quality, as opposed to conventional AGCs, which introduce distortion, pumping, and breathing. Provided an accurate distance sensor, gain changes can take effect instantaneously and do not need to be gated by attack and release time. Likewise, frequency response changes due to undesired proximity-effect variations can be corrected adaptively using precise inverse filtering derived from continuous distance measurements, sound arrival angles, and microphone directivity no longer requiring inadequate static settings on the microphone for proximity-effect compensation.
Convention Paper 8786

3:30 pm

P15-4 Subband Comfort Noise Insertion for an Acoustic Echo Suppressor—*Guangji Shi, Changxue Ma*, DTS, Inc., Los Gatos, CA, USA

This paper presents an efficient approach for comfort noise insertion for an acoustic echo suppressor. Acoustic echo suppression causes frequent noise level change in noisy environments. The proposed algorithm estimates the noise level for each frequency band using a minimum variance based noise estimator, and generates comfort noise based on the estimated noise level and a random phase generator. Tests show that the proposed comfort noise insertion algorithm is able to insert an appropriate level of comfort noise that matches the background noise characteristics in an efficient manner.
Convention Paper 8787

4:00 pm

P15-5 Potential of Non-uniformly Partitioned Convolution with Freely Adaptable FFT Sizes—*Frank Wefers, Michael Vorländer*, RWTH Aachen University, Aachen, Germany

The standard class of algorithms used for FIR filtering with long impulse responses and short input-to-output latencies are non-uniformly partitioned fast convolution methods. Here a filter impulse response is split into several smaller sub filters of different sizes. Small sub filters are needed for a low latency, whereas long filter parts allow for more computational efficiency. Finding an optimal filter partition that minimizes the computational cost is not trivial, however optimization algorithms are known. Mostly the Fast Fourier Transform (FFT) is used for implementing the fast convolution of the sub filters. Usually the FFT transform sizes are chosen to be powers of two, which has a direct effect on the partitioning of filters. Recent studies reveal that the use of FFT transform sizes that are not powers two has a strong potential to lower the computational costs of the convolution even more. This paper presents a new real-time low-latency convolution algorithm, which performs non-uniformly partitioned convolution with freely adaptable FFT sizes. Alongside, an optimization technique is presented that allows adjusting the FFT sizes in order to minimize the computational complexity for this new framework of non-uniform filter partitions. Finally the performance of the algorithm is compared to conventional methods.
Convention Paper 8788

4:30 pm

P15-6 Comparison of Filter Bank Design Algorithms for Use in Low Delay Audio Coding—*Stephan Preihs, Thomas Krause, Jörn Ostermann*, Leibniz Universität Hannover, Hannover, Germany

This paper is concerned with the comparison of filter bank design algorithms for use in audio coding applications with a very low coding delay of less than 1ms. Different methods for numerical optimization of low delay filter banks are analyzed and compared. In addition, the use of the designed filter banks in combination with a delay-free ADPCM coding scheme is evaluated. Design properties and results of PEAQ (Perceptual Evaluation of Audio Quality) based objective audio-quality evaluation as well as a listening test are given. The results show that in our coding scheme a significant improvement of audio-quality, especially for critical signals, can be achieved by the use of filter banks designed with alternative filter bank design algorithms.
Convention Paper 8789

5:00 pm

P15-7 Balanced Phase Equalization; IIR Filters with Independent Frequency Response and Identical Phase Response—*Peter Eastty*, Oxford Digital Limited, Oxford, UK

It has long been assumed that in order to provide sets of filters with arbitrary frequency responses but matching phase responses, symmetrical, finite impulse response filters must be used. A method is given for the construction of sets of infinite impulse response (recursive) filters that can achieve this aim with lower complexity, power, and delay. The zeros of each filter in a set are rearranged to provide linear phase while the phase shift due to the poles of each filter is counteracted by all-pass compensation filters added to other members of the set.
Convention Paper 8790

Session P16 **Sunday, October 28**
2:00 pm – 3:30 pm **Foyer**

POSTERS: ANALYSIS AND SYNTHESIS OF SOUND

2:00 pm

P16-1 Envelope-Based Spatial Parameter Estimation in Directional Audio Coding—
Michael Kratschmer,[1] *Oliver Thiergart*,[2] *Ville Pulkki*[3]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories Erlangen, Erlangen, Germany
[3]Aalto University, Espoo, Finland

Directional Audio Coding provides an efficient description of spatial sound in terms of few audio downmix signals and parametric side information, namely the direction-of-arrival (DOA) and diffuseness of the sound. This representation allows an accurate reproduction of the recorded spatial sound with almost arbitrary loudspeaker setups. The DOA information can be efficiently estimated with linear microphone arrays by considering the phase information between the sensors. Due to the microphone spacing, the DOA estimates are corrupted by spatial aliasing at higher frequencies affecting the sound reproduction quality. In this paper we propose to consider the signal envelope for estimating the DOA at higher frequencies to avoid the spatial aliasing problem. Experimental results show that the presented approach has great potential in improving the estimation accuracy and rendering quality.
Convention Paper 8791

2:00 pm

P16-2 Approximation of Dynamic Convolution Exploiting Principal Component Analysis: Objective and Subjective Quality Evaluation —*Andrea Primavera, Stefania Cecchi, Laura Romoli, Michele Gasparini, Francesco Piazza*, Universitá Politecnica della Marche, Ancona (AN), Italy

In recent years, several techniques have been proposed in the literature in order to attempt the emulation of nonlinear electro-acoustic devices, such as compressors, distortions, and preamplifiers. Among them, the dynamic convolution technique is one of the most common approaches used to perform this task. In this paper an exhaustive objective and subjective analysis of a dynamic convolution operation based on principal components analysis has been performed. Taking into consideration real nonlinear systems, such as bass preamplifier, distortion, and compressor, comparisons with the existing techniques of the state of the art have been carried out in order to prove the effectiveness of the proposed approach.
Convention Paper 8792

2:00 pm

P16-3 Optimized Implementation of an Innovative Digital Audio Equalizer—
Marco Virgulti,[1] *Stefania Cecchi*,[1] *Andrea Primavera*,[1] *Laura Romoli*,[1] *Emanuele Ciavattini*,[2] *Ferruccio Bettarelli*,[2] *Francesco Piazza*[1]
[1]Universitá Politecnica della Marche, Ancona, Italy
[2]Leaff Engineering, Ancona, Italy

Digital audio equalization is one of the most common operations in the acoustic field, but its performance depends on computational complexity and filter design techniques. Starting from a previous FIR implementation based on multirate systems and filterbanks theory, an optimized digital audio equalizer is derived. The proposed approach employs all-pass IIR filters to improve the filterbanks structure developed to avoid ripple between adjacent bands. The effectiveness of the optimized implementation is shown comparing it with the FIR approach. The solution presented here has several advantages increasing the equalization performance in terms of low computational complexity, low delay, and uniform frequency response.
Convention Paper 8793

2:00 pm

P16-4 Automatic Mode Estimation of Persian Musical Signals—*Peyman Heydarian, Lewis Jones, Allan Seago*, London Metropolitan University, London, UK

Musical mode is central to maqamic musical traditions that span from Western China to Southern Europe. A mode usually represents the scale and is to some extent an indication of the emotional content of a piece. Knowledge of the mode is useful in searching multicultural archives of maqamic musical signals. Thus, the modal information is worth inclusion in metadata of a file. An automatic mode classification algorithm will have potential applications in music recommendation and play list generation, where the pieces can be ordered based on a perceptually accepted criterion such as the mode. It has the possibility of being used as a framework for music composition and synthesis. This paper presents an algorithm for classification of Persian audio musical signals, based on a generative approach, i.e., Gaussian Mixture Models (GMM), where chroma is used as the feature. The results will be compared with a chroma-based

method with a Manhattan distance measure that was previously developed by ourselves.
Convention Paper 8794

2:00 pm

P16-5 Generating Matrix Coefficients for Feedback Delay Networks Using Genetic Algorithm—*Michael Chemistruck, Kyle Marcolini, Will Pirkle*, University of Miami, Coral Gables, FL, USA

The following paper analyzes the use of the Genetic Algorithm (GA) in conjunction with a length-4 feedback delay network for audio reverberation applications. While it is possible to manually assign coefficient values to the feedback network, our goal was to automate the generation of these coefficients to help produce a reverb with characteristics as similar to those of a real room reverberation as possible. To do this we designed a GA to be used in conjunction with a delay-based reverb that would be more desirable in the use of real-time applications than the more computationally expensive convolution reverb.
Convention Paper 8795

2:00 pm

P16-6 Low Complexity Transient Detection in Audio Coding Using an Image Edge Detection Approach—*Julien Capobianco,*[1] *Grégory Pallone,*[1] *Laurent Daudet*[2]
[1]France Telecom Orange Labs/TECH/OPERA, Lannion Cedex, France
[2]University Paris Diderot, Paris, France

In this paper we propose a new low complexity method of transient detection using an image edge detection approach. In this method, the time-frequency spectrum of an audio signal is considered as an image. Using appropriate mapping function for converting energy bins into pixels, audio transients correspond to rectilinear edges in the image. Then, the transient detection problem is equivalent to an edge detection problem. Inspired by standard image methods of edge detection, we derive a detection function specific to rectilinear edges that can be implemented with a very low complexity. Our method is evaluated in two practical audio coding applications, in replacement of the SBR transient detector in HEAAC+ V2 and in the stereo parametric tool of MPEG USAC.
Convention Paper 8796

2:00 pm

P16-7 Temporal Coherence-Based Howling Detection for Speech Applications—*Chengshi Zheng, Hao Liu, Renhua Peng, Xiaodong Li*, Chinese Academy of Sciences, Beijing, China

This paper proposes a novel howling detection criterion for speech applications, which is based on temporal coherence (will be referred as TCHD). The proposed TCHD criterion is based on the fact that the speech only has a relatively short coherence time, while the coherence times of the true howling components are nearly infinite since the howling components are perfectly correlated with themselves for large delays. The proposed TCHD criterion is computationally efficient for two reasons. First, the fast Fourier transform (FFT) can be applied directly to compute the temporal coherence. Second, the proposed TCHD criterion does not need to identify spectral peaks from the raw periodogram of the microphone signal. Simulation and experimental results show the validity of the proposed TCHD criterion.
Convention Paper 8797

2:00 pm

P16-8 A Mixing Matrix Estimation Method for Blind Source Separation of Underdetermined Audio Mixture—*Mingu Lee,*[1] *Keong-Mo Sung*[2]
[1]Samsung Electronics Co., Suwon-si, Gyeonggi-do, Korea
[2]Seoul National University, Seoul, Korea

A new mixing matrix estimation method for under-determined blind source separation of audio signals is proposed. By statistically modeling the local features, i.e., the magnitude ratio and phase difference of the mixtures, in a time-frequency region, a region can have information of the mixing angle of a source with reliability amounted to its likelihood. Regional data are then clustered with statistical tests based on their likelihood to produce estimates for the mixing angle of the sources as well as the number of them. Experimental results show that the proposed mixing matrix estimation algorithm outperform the existing methods.
Convention Paper 8798

2:00 pm

P16-9 Speech Separation with Microphone Arrays Using the Mean Shift Algorithm—*David Ayllón, Roberto Gil-Pita, Manuel Rosa-Zurera*, University of Alcala, Alcalá de Henares, Spain

Microphone arrays provide spatial resolution that is useful for speech source separation due to the fact that sources located in different positions cause different time and level differences in the elements of the array. This feature can be combined with time-frequency masking in order to separate speech mixtures by means of clustering techniques, such as the so-called DUET algorithm, which uses only two microphones. However, there are applications where larger arrays are available, and the separation can be performed using all these microphones. A speech separation algorithm based on mean shift clustering technique has been recently proposed using only two microphones. In this work the aforementioned algorithm is generalized for arrays of any number of microphones, testing its performance with echoic speech mixtures. The results obtained show that the generalized mean shift algorithm notably outperforms the results obtained by the original DUET algorithm.
Convention Paper 8799

2:00 pm

P16-10 A Study on Correlation Between Tempo and Mood of Music—*Magdalena Plewa, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

In this paper a study is carried out to identify a ➡

relationship between mood description and combinations of various tempos and rhythms. First, a short review of music recommendation systems along with music mood recognition studies is presented. In addition, some details on tempo and rhythm perception and detection are included. Then, the experiment layout is explained in which a song is first recorded and then its rhythm and tempo are changed. This constitutes the basis for a mood tagging test. Six labels are chosen for mood description. The results show a significant dependence between the tempo and mood of the music.
Convention Paper 8800

Tutorial 9 — Sunday, October 28
2:00 pm – 4:00 pm — Room 132

LARGE ROOM ACOUSTICS

Presenter: **Diemer de Vries**, RWTH Aachen University, Aachen, Germany; TU Delft, Delft, Netherlands

In this tutorial, the traditional and modern ways to describe the acoustical properties of "large" rooms—having dimensions large in comparison to the average wavelength of the relevant frequencies of the speech or music to be (re-)produced—will be discussed. Theoretical models, measurement techniques, the link between objective data and the human perception will be discussed. Is it the reverberation time, or the impulse response, or is there more to take into account to come to a good assessment?

Broadcast/Media Streaming Session 13
Sunday, October 28 — 2:00 pm – 3:30 pm
Room 131

LIP SYNC ISSUE

Chair: **Jonathan Abrams**, Nutmeg Post, Jersey City, NJ, USA

Panelists: *Paul Briscoe,* Harris Corporation
Bob Brown, AVID
Bram Desmet, Flanders Scientific, Inc.
Matthieu Parmentier, France Television, Paris, France

Lip sync remains a complex problem, with several causes and few solutions. From production through transmission and reception, there are many points where lip sync can either be properly corrected or made even worse. This session's panel will discuss several key issues. What is the perspective of the EBU and SMPTE regarding lip sync? Are things being done in production that create this problem? Who is responsible for implementing the mechanisms that ensure lip sync is maintained when the signal reaches your television? Where do the latency issues exist? How can the latency be measured? What are the recommended tolerances? What correction techniques exist? How does video display design affect lip sync? What factors need to be accounted for in Digital Audio Workstations when working with external video monitors in a post environment? Join us as our panel addresses these questions and yours.

Networked Audio Session 6 — Sunday, October 28
2:00 pm – 3:00 pm — Room 133

HIGH PERFORMANCE OVER IP

Presenter: **Rupert Brun**, BBC Audio and Music, London, UK

In the summer of 2010 Brun conducted a simple experiment, making one week of classical music concerts available on-line in high quality and with wide dynamic range. He will explain how he used Twitter and a blog to get real time feedback from the audience and the overwhelming response from the public and press to this simple idea. In the autumn of 2010 it was decided that we would make the "HD Sound" feed permanently available and eventually made it the default for delivery of classical music over IP. This session will explore the future for delivery of audio to the audience over IP and some of the opportunities it presents, while acknowledging why it is seen as a threat to traditional broadcast.

Details of the experiment can be found here: http://www.bbc.co.uk/blogs/bbcinternet/2010/09/bbc_proms_extra_high_quality_audio.html

Project Studio Expo — Sunday, October 28
2:00 pm – 3:00 pm — PSE Stage

PSE10 - MASTER YOUR TRACKS: DIY RESULTS TO COMPETE WITH THE PROS

Presenter: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

Mastering is the final step prior to duplication and, as such, represents the last opportunity to make any final tweaks to a piece of music for highest sonic quality—and maximum transportability among playback systems. Traditionally, musicians have used professional mastering engineers in order to take advantage of their experience and ears, but in today's tight economy—and with the advent of tools that allow for "do-it-yourself" mastering—many musicians are choosing to do their own mastering. This workshop describes the pitfalls and advantages of "project mastering" as well as the main mistakes to avoid but primarily emphasizes practical techniques that can bring out the very best in a piece of music. It also covers the process of album assembly and how to make sure the music in a collection or album provides a smooth, cohesive listening experience.

Sunday October 28 — 2:00 pm — Room 124

Technical Committee Meeting on Coding of Audio Signals

Game Audio Session 10 — Sunday, October 28
2:15 pm – 3:45 pm — Room 122

GAME AUDIO IN A WEB BROWSER

Presenters: **Owen Grace**, Electronic Arts
Roger Powell, Electronic Arts
Chris Rogers, Google, Inc.
Guy Whitmore, PopCap Games

Web browser-based computer games are popular because they do not require client application installation, can be played by single or multiple players over the internet, and are generally capable of being played across different browsers and on multiple devices. Audio tools support for developers is varied, with sound engine software typically employing the Adobe Flash plug-in for rendering audio, or the simplistic HTML5 <audio> element tag. This session will focus on a research project to create a game sound engine in Javascript based on the W3C WebAudio API draft proposal. The sound engine was used to generate 3D spatialized rich audio content within a WebGL-based graphics game framework. The result, a networked multi-player arena combat-style game, rivals the experience of playing on a dedicated console gaming device.

Live Sound Seminar 8 — Sunday, October 28
2:30 pm – 4:30 pm — Room 120

TUNING A LOUDSPEAKER INSTALLATION

Chair: **Jamie Anderson**, Rational Acoustics, Putnam, CT, UDSA
Panelists: *David Gunness*, Fulcrum Acoustic
Deward Timothy, Poll Sound

Loudspeaker systems are installed to achieve functional and aesthetic goals. Therefore, the act of tuning (aligning) those systems are the process of / attempt at achieving those aims. While often equated with simply the adjustment of a system's drive EQ / DSP, loudspeaker system alignment truly encompasses the sum total of the series of decisions (or non-decisions) made throughout the design, installation, drive adjustment, and usage processes. This session gathers a panel of audio professionals with extensive experience in sound system alignment over a diverse variety of system types and applications to discuss their processes, priorities, and the critical elements that make their alignment goals achievable (or not). Given finite, and often extremely limited, resources (equipment, time, money, labor, space, access, authority) this session asks its panelists what is necessary to successfully tune a loudspeaker installation.

Audio Industry Seminar — Sunday, October 28
2:30 pm – 3:30 pm — Room 114

PMC: MASTERS OF AUDIO SERIES

Advanced Music Production

Presenter: **Ross Hogarth**

Join Ross in the Audio Masters room for an in-depth discussion and demonstration of some of Ross' favorite tools. He will be sharing his 30+ years of experience and knowledge in music and pro audio, so please come pick his brain!

Gov't Mule, Roger Waters, The Black Crowes, Shawn Colvin, John Mellencamp, R.E.M., and Jewel are some of the artists this double Grammy-winning producer/engineer/mixer has had the joy of working with over his two-plus decades in the recording studio. Ross' wide varied experience and taste opened up all creative doors and usual and unusual possibilities. Versatility, absolute belief in the artists' vision, and his complete commitment to bringing that vision to the final master are all representative of Ross' strengths and abilities. In the ever changing world of the arts, Ross's goal is to let creativity be the place of certainty that allows for constant change.

Networked Audio Session 7 — Sunday, October 28
3:00 pm – 4:30 pm — Room 123

AUDIO NETWORK DEVICE CONNECTION AND CONTROL

Chair: **Richard Foss**, Rhodes University, Grahamstown, Eastern Cape, South Africa
Panelists: *Jeff Berryman*, Bosch
Andreas Hildebrand, ALC Networx
Jeff Koftinoff, MeyerSound
Kieran Walsh, Audinate Pty. Ltd., Ultimo, NSW, Australia

In this session a number of industry experts will describe and demonstrate how they have enabled the discovery of audio devices on local area networks, their subsequent connection management, and also control over their various parameters. The workshop will start with a panel discussion that introduces issues related to streaming audio, such as bandwidth management and synchronization, as well as protocols that enable connection management and control. The panelists will have demonstrations of their particular audio network solutions. They will describe these solutions as part of the panel discussion, and will provide closer demonstrations following the panel discussion.

Audio Industry Seminar — Sunday, October 28
3:00 pm – 4:00 pm — Room 112

THAT CORP.

Interfacing Digitally Controlled Microphone Preamplifiers to A/D Converters

Presenters: **Gary Hebert**
Joe Lemanski

This seminar examines practical designs for a digitally controlled microphone preamplifier driving typical modern audio A/D converters. Topics covered include maximizing the available A/D converter dynamic range, achieving the desired preamp gain range, and implementing digitally controlled switching of pads and phantom power.

Project Studio Expo — Sunday, October 28
3:00 pm – 4:00 pm — PSE Stage

PSE11 - MAKE MUSIC WITH YOUR IPAD: HOT APPS, GREAT GADGETS & ALL THINGS IOS

Presenter: **Mike Metlay**, RECORDING Magazine, Boulder, CO, USA

In this seminar we will look at hardware and software for music production on iOS devices, with an emphasis on the iPad. A number of worthwhile interface products and apps will be demonstrated for the audience, and there will be an in-depth discussion of the advantages and pitfalls of working with iOS in the studio. There will be material for beginners curious about the iPad as well as some advanced tips and tricks for experienced users.

Sunday October 28 — 3:00 pm — Room 124

Technical Committee Meeting on Loudspeakers and Headphones

Sunday October 28 3:00 pm Room 125

AES Standards Committee Meeting SC-04-01 on Acoustics and Sound Source Modeling

Broadcast/Media Streaming Session 14
Sunday, October 28 3:30 pm – 5:00 pm
Room 131

UNDERSTANDING AND WORKING WITH CODECS

Chair: **Kimberly Sacks**, Optimod Refurbishing, Hollywood, MD, USA

Panelists: *Kirk Harnack*, Telos Alliance, Nashville, TN, USA; South Seas Broadcasting Corp., Pago Pago, American Samoa
James Johnston, Retired, Redmond, WA, USA
Jeffrey Riedmiller, Dolby Laboratories, Penngrove CA, USA
Chris Tobin, Musicam USA

In the age of smart phones and internet ready devices, audio transport, and distribution has evolved from sharing low quality MP3 files to providing high quality mobile device audio streams, click to play content, over the air broadcasting, audio distribution in large facilities, and more. Each medium has several methods of compressing content by means of a codec. This session will explain which codecs are appropriate for which purposes, common misuse of audio codecs, and how to maintain audio quality by implementing codecs professionally.

Special Event
PLATINUM MASTERING: MASTERED FOR iTUNES
Sunday, October 28, 3:30 pm – 5:00 pm
Room 134

Moderator: **Bob Ludwig**, Gateway Mastering Studios, Portland, ME, USA

Panelists: *Eric Boulanger*, Engineer, The Mastering Lab, Ojai, CA, USA
Bob Katz, President and Mastering Engineer of Digital Domain, Altamonte Springs, FL, USA

Mastering Apple Inc.'s "Mastered for iTunes" initiative, the Science, the Procedures, and the Results

Apple's iTunes Store is the largest music vendor in the world. Last year, with the introduction of Colbie Caillat's All of You download, they began selling higher fidelity AAC encodes for no additional cost to the consumer. Some mastering and recording engineers have misunderstood what it is about and there has been a surprising amount of negative and incorrect press about what it does and does not do. Observing Apple's market share, the new procedure should be understood by every mastering engineer. The Apple "Mastered for iTunes" initiative is based on 100% solid science and it can sometimes yield AAC encodes so close to the 44-kHz/24 bit PCM encode masters that many have failed an A-B-X listening test. Please come and find out all there is to know and ask questions.

Student/Career Event
SC-10 RECORDING COMPETITION—2
Sunday, October 28, 3:30 pm – 6:30 pm
Room 133

The Student Recording Competition is a highlight at each convention. A distinguished panel of judges participates in critiquing finalists of each category in an interactive presentation and discussion. This event presents stereo and surround recordings in these categories:

- Traditional Acoustic Recording 3:30 pm to 4:30 pm
  Judges: David Bowles, Martha de Francisco, Richard King
- Traditional Multitrack Studio Recording 4:30 pm to 5:30 pm
  Judges: Jim Anderson, Brandie Lane, Glenn Lorbecki
- Modern Multitrack Studio Recording 5:30 pm to 6:30 pm
  Judges: Mark Rubel, Stephen Webber

The top three finalists in each category, as identified by our judges, present a short summary of their production intentions and the key recording and mix techniques used to realize their goals. They then play their projects for all who attend. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2) on Monday afternoon. The competition is a great chance to hear the work of your fellow students at other educational institutions. Everyone learns from the judges' comments even if your project isn't one of the finalists, and it's a great chance to meet other students and faculty.

Session EB2 Sunday, October 28
4:00 pm – 5:45 pm Room 122

LECTURES 1

Chair: **Lance Reichert**, Sennheiser Electronic Corp.

4:00 pm

EB2-1 Comparison of Highly Configurable CPU- and GPU-Based Convolution Engines—
Michael Schoeffler,[1] *Wolfgang Hess*[2]
[1]International Audio Laboratories Erlangen, Erlangen, Germany
[2]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

In this work the performance of real-time audio signal processing convolution engines is evaluated. A CPU-based implementation using the Integrated Performance Primitives Library and two GPU-based implementations using CUDA and OpenCL are compared. The purpose of these convolution engines is auralization, e.g., the binaural rendering of virtual multichannel configurations. Any multichannel input and output configuration is supported, e.g., 22.2 to 5.1, 7.1 to 2.0, vice versa, etc. This ability results in a tradeoff between configurability and performance. Using a 5.1-to-binaural setup with continuous filter changes due to simulated head-tracking, GPU processing is more efficient when 24 filters of more than 1.92 seconds duration each @ 48 kHz sampling rate are convolved. The GPU is capable of convolving longer filters in real-time than a CPU-based processing. By comparing both GPU-based implementations, negligible performance differences between OpenCL and CUDA were measured.
Engineering Brief 60

4:15 pm

EB2-2 Multichannel Audio Processor which Adapts

to 2-D and 3-D Loudspeaker Setups—*Christof Faller*, Illusonic GmbH, Uster, Switzerland

A general audio format conversion concept is described for reproducing stereo and surround audio content on loudspeaker setups with any number of channels. The goal is to improve localization and to generate a recording-related spatial impression of depth and immersion. It is explained how with these goals signals are processed using a strategy that is independent of a specific loudspeaker setup. The implementation of this general audio format conversion concept, in the Illusonic Immersive Audio Processor, is described.
Engineering Brief 61

4:30 pm

EB2-3 A Comparison of Recording, Rendering, and Reproduction Techniques for Multichannel Spatial Audio—*David Romblom*,[1,2,3] *Catherine Guastavino*,[1,2] *Richard King*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
[3]Sennheiser Technology and Innovation, San Francisco, CA, USA

The objective of this project is to compare the relative merits of two different spatial audio recording and rendering techniques within the context of two different multichannel reproduction systems. The two recordings and rendering techniques are "natural," using main microphone arrays, and "virtual," using spot microphones, panning, and simulated acoustic delay. The two reproduction systems are the 3/2 system (5.1 surround), and a 12/2 system, where the frontal L/C/R triplet is replaced by a 12 loudspeaker linear array. Additionally, the project seeks to know if standard surround techniques can be used in combination with wavefront reconstruction techniques such as Wave Field Synthesis. The Hamasaki Square was used for the room effect in all cases, exhibiting the startling quality of increasing the depth of the frontal image.
Engineering Brief 62

4:45 pm

EB2-4 The Reactive Source: A Reproduction Format Agnostic and Adaptive Spatial Audio Effect—
Frank Melchior, BBC R&D, Salford, UK

Spatial audio has become a more and more active field of research and various systems are currently under investigation on different scales of effort and complexity. Given the potential of 3-D audio systems, spatial effects beyond source positioning and room simulation are desirable to enhance the creative flexibility. This paper describes a new adaptive spatial audio effect called reactive source. The reactive source uses low-level features of the incoming audio signal to dynamically adapt the spatial behavior of a sound source. Furthermore, the concept is format agnostic so that the effect could easily be applied to different 3-D audio reproduction methods using the same interaction method. To verify the basic concept, a prototype system for multichannel reproduction has been developed.
Engineering Brief 63

5:00 pm

EB2-5 Teaching Critical Thinking in an Audio Production Curriculum—*Jason Corey*, University of Michigan, Ann Arbor, MI, USA

The practice of sound recording and production can be characterized as a series of decisions based primarily on subjective impressions of sound. These subjective impressions lead to equipment choices and use, not only for artistic effect but also to accomplish technical objectives. Nonetheless, the ability to think critically about recording techniques, equipment specifications, and sound quality is vitally important to equipment choice and use. The goal of this paper is to propose methods to encourage critical thinking among students in an audio production curriculum and to consider topics that might be included in coursework to help aspiring audio engineers evaluate audio equipment and processing.
Engineering Brief 64

5:15 pm

EB2-6 Sync-AV—Workflow Tool for File-Based Video Shootings—*Andreas Fitza*, University of Applied Science Mainz, Mainz, Germany

The Sync-AV workflow eases the sorting and synchronization of video and audio footage without the needs of expensive special hardware. It supports the preproduction and the shooting as well as the post-production. It consists of three elements: A script-information- and metadata-gathering iOS app that is synchronized with a server-back-end. It can be used on different devices at once to exchange information onset. A server database with a web-front-end that can sort files by their metadata and show dailies as well. It can also be used to distribute and manage information during the preproduction. A local client that can synchronize and rename the files and that implements the metadata.
Engineering Brief 65

5:30 pm

EB2-7 Audio Over IP—*Kieran Walsh*, Audinate Pty. Ltd., Ultimo, NSW, Australia

Developments in both IP networking and the attitude of professional audio to emerging technologies have presented the opportunity to consider a more abstract and all-encompassing approach to the ways that we manage data. We will examine this paradigm shift and discuss the benefits presented both in practical terms and creatively.
Engineering Brief 66

Workshop 10 — Sunday, October 28
4:00 pm – 6:00 pm — Room 130

SHOW ME THE MONEY!
FINDING SUCCESS IN AN EVOLVING AUDIO

Chair: **Bill Gibson**, Hal Leonard, Art Institute of Seattle, and Trustee for the Recording Academy

Panelists: *Moses Avalon*, Moses Avalon Company
Dave Hampton
Mixerman
Steve Turnidge, Ars Divina/Ultra Violet Studios

Music Business Panel: Show Me the Money!

Finding Success in an Evolving Audio Industry includes a panel discussion about the new music business. The mission of this workshop is to provide insightful tactics and techniques for the modern audio professional—ways to make money in the current business reality. Topics include proven ways to increase profitability, nuts and bolts explanations of setting up your business to receive payments for licensing, sync fees, Internet and other new media delivery systems, and more. Panel consists of noted authors and music business experts, Moses Avalon (*Confessions of a Record Producer, Million Dollar Mistakes*), Dave Hampton (*The Business of Audio Engineering*), Mixerman, and Steve Turnidge (UltraViolet Studios).

Workshop with Height 4 **Sunday, October 28**
4:00 pm – 6:00 pm **Pyramind**
832 Folsom Street

AUDIO FORMATS—MULTICHANNEL SOUND WITH HEIGHT: CHANNEL- OR OBJECT-BASED?

Chair: **Florian Camerer**, ORF, Vienna, Austria

Panelists: *Kimio Hamasaki*, NHK, Chiba-ken, Japan
Jeff Levison, IOSONO GmbH, Germany
Toni Mateos, imm sound, Barcelona, Spain
Wilfried Van Baelen, Auro-3D, Mol, Belgium

Several proposals for multichannel sound with height have been brought to the market in the past year(s). One of the differences of these propositions is the use of channel vs. object-based mixing and distribution methods. Some proposals also introduce a hybrid method, combining both methods into one format. This workshop aims to discuss these methods and their strengths and weaknesses. Topics that will be addressed are production methods and tools, how to fold down that 3-D Audio mix into standard distribution formats, consequences for reproduction in the professional (e.g., digital cinema), as well as consumer markets. [Workshop will be held at Pyramind, 832 Folsom Street (a 10-minute walk from Moscone Center). Space is limited and free tickets can be obtained at the *AES Tours Desk* [full program badge required].]

Audio Industry Seminar **Sunday, October 28**
4:00 pm – 5:00 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

PMC Playback Sessions

PMC will play a selection of extraordinary recordings AND give you the chance to listen to your own music and projects.

Audio Industry Seminar **Sunday, October 28**
4:00 pm – 5:00 pm **Room 112**

STUDIO SIX

Using iOS Devices for Audio Test and Measurement

Presenter: **Andrew Smith**

This seminar explores the current state-of-the-art measurement tools in audio acoustics: using the AudioTools app, iAudioInterface2, and iTestMic from Studio Six Digital. AudioTools includes a wide variety of acoustical analysis and audio test measurement modules, including SPL, RTA, FFT, Smaart Tools single channel, the Studio Six Transfer Function, and line level test functions.

Project Studio Expo **Sunday, October 28**
4:00 pm – 5:00 pm **PSE Stage**

PSE12 - ASK THE EDITORS: Q&A SESSION

Presenters: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA
E Strother Bullins, Westfield, NC, USA
Larry Crane
Sarah Jones, Electronic Musician Magazine, San Bruno, CA, USA
Tom Kenny
Mike Metlay, RECORDING Magazine, Boulder, CO, USA
Hugh Robjohns, Technical Editor, Sound on Sound, Crowle, UK
Frank Wells, Pro Sound News, Murfreesboro, TN, USA
Paul White, Sound On Sound, Malvern, Worcestershire, UK

Panel discussion with industry pros on topics of special interest to attendees.

Sunday October 28 **4:00 pm** **Room 124**

Technical Committee Meeting on High Resolution Audio

Game Audio Session 11 **Sunday, October 28**
4:15 pm – 5:45 pm **Room 132**

GETTING INTO SOUND DESIGN

Presenters: **Elise Baldwin**, Electronic Arts/Maxis, Redwood City, CA, USA
Shaun Farley, Teleproductions International, Chantilly, VA, USA
Ann Kroeber, Sound Mountain Sound Effects Service
Kyrsten Mate, Skywalker Sound, Marin County, CA, USA
Nathan Moody, Stimulant

A cross-section of industry experts (games, film, TV) discuss entering the effects editing and sound design field. In addition to Game Audio, the panel will discuss the broader industry as a whole, different mediums where work can be found, and how they got their start. “Things no one told me,” skills development, continuing education, and personality all contribute to a successful career. Have you been doing everything you should be?

Live Sound Seminar 9 **Sunday, October 28**
4:30 pm – 6:00 pm **Room 120**

ACOUSTICS FOR SMALL LIVE SOUND VENUES—CREATING (& FINE TUNING) THE CONSUMMATE PERFORMING / LISTENING ENVIRONMENT

Presenter: **John Storyk**, Walters-Storyk Design Group, Highland, NY, USA

John Storyk has designed acoustics for a number of live performance venues. These range from the Jazz at Lincoln Center Complex to the successful NYC club Le Poisson Rouge and to the Fenix Club, a brand new venue in San Rafael, which will be on the Tech Tour Schedule. John will give an illuminating presentation on improving acoustics in existing performance venues AND designing acoustics for new venues that would address potential absorption and reflection and other sound-related issues prior to construction. He also just completed the acoustics for a new NYC club called 54 Below (below the Studio 54 theater). Some VERY innovative acoustic solutions were developed for that venue.

Networked Audio Session 8 Sunday, October 28
4:30 pm – 6:00 pm Foyer

AUDIO NETWORK DEVICE CONNECTION AND CONTROL—DEMOS

Chair: **Richard Foss**, Rhodes University, Grahamstown, Eastern Cape, South Africa

Panelists: *Jeff Berryman*, Bosch
Andreas Hildebrand, ALC Networx
Jeff Koftinoff, MeyerSound
Kieran Walsh, Audinate Pty. Ltd., Ultimo, NSW, Australia

In this session a number of industry experts will describe and demonstrate how they have enabled the discovery of audio devices on local area networks, their subsequent connection management, and also control over their various parameters. The workshop will start with a panel discussion that introduces issues related to streaming audio, such as bandwidth management and synchronization, as well as protocols that enable connection management and control. The panelists will have demonstrations of their particular audio network solutions. They will describe these solutions as part of the panel discussion, and will provide closer demonstrations following the panel discussion.

Broadcast/Media Streaming Session 15
Sunday, October 28 5:00 pm – 6:30 pm
Room 131

AUDIO PROCESSING BASICS

Chair: **Richard Burden**, Richard W. Burden Associates, Canoga Park, CA, USA

Panelists: *Tim Carroll*, Linear Acoustic Inc., Lancaster, PA, USA
Frank Foti, Omnia
James Johnston, Retired, Redmond, WA, USA
Robert Orban, Orban, San Leandro, CA, USA

Limiting peak excursions to prevent over modulation and increasing the average level through compression to improve signal to noise are worthwhile objectives. Just as we can all agree that a little salt and pepper in the stew enhances the flavor, the argument is how much salt and pepper becomes too much.

It is a given that the louder signal is interpreted by the listener as sounding better. However, there are misuses of the available tools and display a lack of leadership at the point of origin. The variation in energy levels within program and commercial content, as well as, the excessive use of compression on many news interviews are annoying to the listener.

The presentation will cover the fundamentals, the history, and the philosophy of audio processing. An open discussion, with audience participation, on the subject and its practices follow.

Product Design Session 9 Sunday, October 28
5:00 pm – 6:00 pm Room 123

AUDIO FOR iPAD PUBLISHERS

Presenter: **Jeff Essex**, AudioSyncrasy

Book publishers are running to the iPad, and not just for iBooks or one-off apps. They're building storefronts and creating subscription models, and the children's book publishers are leading the way. Through two case studies, this talk will explore how to build the audio creation and content management systems needed to produce multiple apps in high-volume environments, including VO production, concatenation schemes, file-naming conventions, audio file types for iOS, and perhaps most important, helping book publishers make the leap from the printed page to interactive publishing.

Project Studio Expo Sunday, October 28
5:00 pm – 6:00 pm PSE Stage

THE MIDI MANUFACTURERS ASSOCIATION PRESENTS "MIDI MAKES MUSIC: CELEBRATING 30 YEARS OF MIDI"

In 2013, the MIDI Manufacturers Association will bring together the world's top digital music companies to recognize and celebrate the 30th anniversary of the invention of MIDI technology. MIDI dramatically changed music-making 30 years ago—and it remains a key technology for music-making today. Come and get a sneak-preview of how the MMA will celebrate this important milestone and hear from some of the people who were there from the beginning. Also learn how the standard evolved over the past three decades and see what's next for MIDI technology.

Sunday October 28 5:00 pm Room 124

Technical Committee Meeting on Semantic Audio Analysis

Special Event
WRECKING CREW FILM & Q&A
Sunday, October 28, 6:30 pm – 9:00 pm
Room 134

Presenters: **Don Randi**
Denny Tedesco

The Wrecking Crew were a group of Los Angeles-based studio musicians in the 1960s who played on hits for the Beach Boys, Frank Sinatra, Nancy Sinatra, Sonny and Cher, Jan & Dean, The Monkees, Gary Lewis and the Playboys, Mamas and Papas, Tijuana Brass, Ricky Nelson, Johnny Rivers, and were Phil Spector's Wall of Sound. Behind every recording was an elite group of engineers, using groundbreaking and innovative techniques to capture the distinct sound of the musicians. With more Gold and Platinum credits than The Beatles, The Rolling Stones, Elvis, and Elton combined, "The Wrecking Crew," is the most storied, yet anonymous group in recording history.

Produced and directed by Denny Tedesco, son of leg- ➥

endary late Wrecking Crew guitarist Tommy Tedesco, this elusive yet highly praised documentary film features interviews with Brian Wilson, Cher, Roger McGuinn, and many Wrecking Crew members, including Glen Campbell, a durable superstar, in his own right. Tedesco and special guest Don Randi will hold a Q&A following the screening.

Sunday October 28 6:30 pm Room 125

AES Standards Committee Meeting SC-02-12 (1) on Audio Applications of Networks (Session 1—Main Agenda)

Session P17 Monday, October 29
9:00 am – 1:00 pm Room 121

SPATIAL AUDIO PROCESSING

Chair: **Jean-Marc Jot**, DTS

9:00 am

P17-1 Comparing Separation Quality of Nonnegative Matrix Factorization and Nonnegative Matrix Factor 2-D Deconvolution in Audio Source Separation Tasks—*Julian M. Becker, Volker Gnann*, RWTH Aachen University, Aachen, Germany

The Nonnegative Matrix Factorization (NMF) is widely used in audio source separation tasks. However, the separation quality of NMF varies a lot depending on the mixture. In this paper we analyze the use of NMF in source separation tasks and show how separation results can be significantly improved by using the Nonnegative Matrix Factor 2D Deconvolution (NMF2D). NMF2D was originally proposed as an extension to the NMF to circumvent the problem of grouping notes, but it is used differently in this paper to improve the separation quality, without taking the problem of grouping notes into account.
Convention Paper 8801

9:30 am

P17-2 Aspects of Microphone Array Source Separation Performance—*Bjoern Erlach*,[1] *Rob Bullen*,[2] *Jonathan S. Abel*[1]
[1]Stanford University, Stanford, CA, USA
[2]SoundScience P/L/, Australia

The performance of a blind source separation system based on a custom microphone array is explored. The system prioritizes artifact-free processing over source separation effectiveness and extracts source signals using a quadratically constrained least-squares fit based on estimated source arrival directions. The level of additive noise present in extracted source signals is computed empirically for various numbers of microphones used and different degrees of uncertainty in knowledge of microphone locations. The results are presented in comparison to analytical predictions. The source signal estimate variance is roughly inversely proportional to the number of sensors and roughly proportional to both the additive noise variance and microphone position error variance. Beyond a threshold the advantages of increased channel count and precise knowledge of the sensor locations are outweighed by other limitations.
Convention Paper 8802

10:00 am

P17-3 A New Algorithm for Generating Realistic Three-Dimensional Reverberation Based on Image Sound Source Distribution in Consideration of Room Shape Complexity— *Toshiki Hanyu*, Nihon University, Funabashi, Chiba, Japan

A new algorithm for generating realistic three-dimensional reverberation based on statistical room acoustics is proposed. The author has clarified the relationship between reflected sound density and mean free path in consideration of room shape complexity [Hanyu et al., Acoust. Sci. & Tech. 33, 3 (2012), 197–199]. Using this relationship the new algorithm can statistically create image sound source distributions that reflect the room shape complexity, room's absorption, and room volume by using random numbers of three-dimensional orthogonal coordinates. The image sound source distribution represents characteristics of three-dimensional reverberation of the room. Details of this algorithm and how to apply this to multichannel audio, binaural audio, game sound, and so on are introduced.
Convention Paper 8803

10:30 am

P17-4 Audio Signal Decorrelation Based on Reciprocal-Maximal Length Sequence Filters and Its Applications to Spatial Sound— *Bo-sun Xie*,[1] *Bei Shi*,[1] *Ning Xiang*[2]
[1]South China University of Technology, Guangzhou, China
[2]Rensselaer Polytechnic Institute, Troy, NY, USA

An algorithm of audio signal decorrelation is proposed. The algorithm is based on a pair of all-pass filters whose responses match with a pair of reciprocal maximal length sequences (MLSs). Taking advantage of the characters of uniform power spectrum and low-valued cross-correlation of reciprocal MLSs, the filters create a pair of output signals with low cross-correlation but almost identical magnitude spectra to the input signal. The proposed algorithm is applied to broadening the auditory source-width and enhancing subjective envelopment in multichannel sound reproduction. Preliminary psychoacoustic experiments validate the performance of the proposed algorithm.
Convention Paper 8805

11:00 am

P17-5 Utilizing Instantaneous Direct-to-Reverberant Ratio in Parametric Spatial Audio Coding— *Mikko-Ville Laitinen, Ville Pulkki*, Aalto University, Espoo, Finland

Scenarios with multiple simultaneous sources in an acoustically dry room may be challenging for parametric spatial sound reproduction techniques, such as directional audio coding (DirAC). It has been found that the decorrelation process used in the reproduction causes a perception of added reverberation. A model for DirAC repro-

duction is suggested in this paper, which utilizes estimation of instantaneous direct-to-reverberant ratio. The sound is divided into reverberant and non-reverberant parts using this ratio, and decorrelation is applied only for the reverberant part. The results of formal listening tests are presented that show that perceptually good audio quality can be obtained using this approach for both dry and reverberant scenarios.
Convention Paper 8804

11:30 am

P17-6 A Downmix Approach with Acoustic Shadow, Low Frequency Effects, and Loudness Control—*Regis Rossi A. Faria,*[1] *José Augusto Mannis*[2]
[1]University of São Paulo, Ribeirão Preto, Brazil
[2]University of Campinas, Campinas, Brazil

Conventional algorithms for converting 5.1 sound fields into 2.0 have systematically suppressed the low frequency information and neglected spatial auditory effects produced by the real position of the loudspeakers in 3/2/1 arrangements. We designed a downmix variation in which the listener's head acoustic shadow and the LFE information are considered in the conversion in order to investigate how the use of such models can contribute to improve the surround experience in two-channel modes and to customize the downmix conversion so to achieve the particular balance required by individual surround programs. A test implementation with integrated loudness control was carried out. Preliminary results show the potential benefits in using this approach and point to critical parameters involved in the downmixing task.
Convention Paper 8806

12:00 noon

P17-7 Direct-Diffuse Decomposition of Multichannel Signals Using a System of Pairwise Correlations—*Jeffrey Thompson, Brandon Smith, Aaron Warner, Jean-Marc Jot*, DTS, Inc., Calabasas, CA, USA

Decomposing an arbitrary audio signal into direct and diffuse components is useful for applications such as spatial audio coding, spatial format conversion, binaural rendering, and spatial audio enhancement. This paper describes direct-diffuse decomposition methods for multichannel signals using a linear system of pairwise correlation estimates. The expected value of a correlation coefficient is analytically derived from a signal model with known direct and diffuse energy levels. It is shown that a linear system can be constructed from pairwise correlation coefficients to derive estimates of the Direct Energy Fraction (DEF) for each channel of a multichannel signal. Two direct-diffuse decomposition methods are described that utilize the DEF estimates within a time-frequency analysis-synthesis framework.
Convention Paper 8807

12:30 pm

P17-8 A New Method of Multichannel Surround Sound Coding Utilizing Head Dynamic Cue—*Qinghua Ye, Lingling Zhang, Hefei Yang, Xiaodong Li*, Chinese Academy of Sciences, Beijing, China

Considering the disadvantages of conventional matrix surround systems, a new method of multichannel surround sound coding is proposed. At the encoder, multichannel signals are converted into two pairs of virtual stereo signals, which are modulated to make corresponding changes by simulating dynamic cue of the head's slight rotation. Then the left and right channels of the virtual stereos are added respectively into two-channel stereo for transmission. At the decoder, the front and back stereos are separated by extracting the dynamic cue and then redistributed to the multichannel system. This new method can realize better surround sound reproduction without affecting sound effects of the downmixed stereo.
Convention Paper 8808

Tutorial 10 — **Monday, October 29**
9:00 am – 10:30 am — **Room 131**

BINAURAL AUDITORY MODELS

Presenter: **Ville Pulkki**, Aalto University, Espoo, Finland

The working principles of brain mechanisms of binaural hearing have been debated during the last decade. In the 1990s the common thinking was that human binaural decoding is based on delay lines and coincidence counters, as proposed by the Jeffress model. Later, some neurophysiological findings questioned the existence of such delay lines, and some evidence was found bolstering the count-comparison model proposed by Bekesy. In a count-comparison model, the binaural differences are rate-coded between the left and brain right hemispheres. This tutorial will introduce the basic principles of most common binaural auditory models and review some latest improvements in the models.

Tutorial 11 — **Monday, October 29**
9:00 am – 10:45 am — **Room 132**

BUILDING HEARING AWARENESS INTO AUDIO CURRICULA

Presenter: **S. Benjamin Kanters,** Columbia College Chicago, IL, USA

With the increasing incidence of hearing disorders, particularly from entertainment media, it is incumbent on audio programs to begin addressing hearing awareness with students (and professionals) in audio.

Working from materials developed for the Hearing Conservation Workshop, this tutorial will give students, professors and professionals, tools to teach and learn hearing physiology, disorders and conservation. This will include teaching techniques, graphics and animations that illustrate principles of hearing physiology. The success of The Workshop has proven that, once taught how their hearing works (and how it breaks), engineers develop a sense of “ownership” of their ears, and become concerned about their hearing health.

They are then in a unique position to act as role models of hearing awareness to their clients and the music-listening public.

Live Sound Seminar 10 **Monday, October 29**
9:00 am – 10:30 am **Room 120**

LIVE SOUND FOR CORPORATE EVENTS: IT'S BUSINESS TIME

Chair: **Michael (Bink) Knowles**, Freelance Engineer, Oakland, CA, USA

Panelists: *Steve Ratcliff*, Freelance Engineer
Scott Urie, Pinnacle Consulting Group

Corporate events demand audio perfection, even when the video screens and lighting plot take precedence over loudspeaker placement. Signal flow and mixing can be very complex, with distant parties and panel discussions creating feedback challenges. Mixing, routing, and arraying strategies will be discussed using example cases.

Product Design Session 10 **Monday, October 29**
9:00 am – 10:00 am **Room 130**

ETHERNET STANDARD AUDIO

Presenter: **Stephen Lampen**, Belden, San Francisco, CA, USA

Ethernet has been around since 1973, and with the rise of twisted-pair-based Ethernet there have been many companies who played around to get Ethernet to work for multichannel audio. The problem is that all of their solutions were proprietary and not always compatible between manufacturers. This was the impetus behind IEEE 802.1BA AVB, a re-write of the Ethernet standard to include many bells and whistles for audio and video applications. This presentation will show AVB switches, how they are different, and what is in this new standard.

Historical Program
H5 - THE REPLAY OF HISTORICAL MAGNETIC TAPE—MORE THAN PRESSING THE PLAY BUTTON
Monday, October 29, 9:00 am – 10:30 am
Room 133

Presenter: **Nadja Wallaszkovits**, Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria

This fascinating presentation discusses the various process challenges involved with the transfer of historical magnetic audio tapes. Focusing on the digitization of collections from the very early years of magnetic recording, for example the earliest tapes from Portugal and Norway, Nadja Wallaszkovits shows the various stages of preservation, handling, and reproduction of important original tapes from the early days of magnetic recording. Beginning with an overview of early magnetic tape developments, the original tape recorders and their specific parameters are introduced. Next, the practical handling of critical tapes is outlined. Based on an analysis of the physical and chemical preservation status of individual tapes, the choice and adjustment of the replay equipment and parameters are covered. The problems of tape carrier handling and physical as well as chemical restoration are outlined, along with discussions on optimizing head azimuth, tape-to-head contact, tape speed, and tension. All are important for optimum signal recovery.

Monday October 29 **9:00 am** **Room 125**

AES Standards Committee Meeting SC-02-12 (2) on Audio Applications of Networks (Session 2—AES-X192)

Session EB3 **Monday, October 29**
9:30 am – 11:00 am **Foyer**

POSTERS 2

9:30 am

EB3-1 Implementation of an Interactive 3-D Reverberator for Video Games Using Statistical Acoustics—*Masataka Nakahara*,[1] *Tomoya Kishi*,[2] *Kenji Kojima*,[2] *Toshiki Hanyu*,[3] *Kazuma Hoshi*[3]
[1]ONFUTURE Ltd., Tokyo, Japan
[2]CAPCOM Co., Ltd., Osaka, Japan
[3]Junior College, Nihon University, Chiba, Japan

An interactive reverberator, which applies realistic computed acoustic responses interactively to video game scenes, is a very important technology for the processing of in-game sounds. The mainframe of an interactive reverberator, which the authors developed, is designed based on statistical acoustics theory, so that it is possible to compute fast enough to realize real-time processing in fast-changing game scenes. Though statistical reverbs generally do not provide a high level of reality, the authors have achieved a quantum leap of sound quality by applying Hanyu's algorithm to conventional theories. The reverberator features: (1) No pre-computing jobs including room modeling are required. (2) Three-dimensional responses are generated automatically. (3) Complex factor of a room's shape, open-air areas, and effects of neighboring reverberations are expressed. The authors implemented the reverberator into a Capcom's middleware experimentally and have verified it can run effectively. In this paper the algorithm, background theories, and implementation techniques are introduced.
Engineering Brief 67

9:30 am

EB3-2 Printable Loudspeaker Arrays for Flexible Substrates and Interactive Surfaces—
Jess Rowland, Adrian Freed, University of California, Berkeley, Berkeley, CA, USA

Although planar loudspeaker drivers have been well explored for many years, a flat speaker array system that may flex or fold freely remains a current challenge to engineer. We will demonstrate a viable technique for building large loudspeaker arrays that allow for diffused fields of sound transduction on flexible membranes. Planar voice coils are made from machine-cut copper sheets, or by inkjet printing and electroless copper plating, on paper, thin plastic, or similar lightweight material. We will present various ways of attaching thin magnets to these membranes, including a novel alternative strategy of mounting magnets in gloves worn by the listener. This creates an engaging experience for listeners in which gestures can control sounds from the speaker array interactively.
Engineering Brief 68

9:30 am

EB3-3 Nonlinear Distortion Measurement in Audio Amplifiers: The Perceptual Nonlinear

Distortion Response—*Phillip Minnick*, University of Miami, Coral Gables, FL, USA

A new metric for measuring nonlinear distortion is introduced called the Perceptual Nonlinear Distortion Response (PNDR) to measure nonlinear distortion in audio amplifiers. This metric accounts for the auditory system's masking effects. Salient features of previously developed nonlinear distortion measurements are considered in the development of the PNDR. A small group of solid-state and valve audio amplifiers were subjected to various benchmark tests. A listening test was created to test perceptibility of nonlinear distortions generated in the amplifiers. These test results were analyzed and the Perceptual Nonlinear Distortion Response was more successful than traditionally used distortion metrics. This cognitive tool could provide the audio industry with more accurate test methods, facilitating product research and development.
Engineering Brief 69

9:30 am

EB3-4 EspGrid: A Protocol for Participatory Electronic Ensemble Performance—*David Ogborn*, McMaster University, Hamilton, ON, Canada

EspGrid is a protocol developed to streamline the sharing of timing, code, audio, and video in participatory electronic ensembles, such as laptop orchestras. An application implementing the protocol runs on every machine in the ensemble, and a series of "thin" helper objects connect the shared data to the diverse languages that live electronic musicians use during performance (Max, ChucK, SuperCollider, PD, etc.). The protocol/application has been developed and tested in the busy rehearsal and performance environment of McMaster University's Cybernetic Orchestra, during the project "Scalable, Collective Traditions of Electronic Sound Performance" supported by Canada's Social Sciences and Humanities Research Council (SSHRC), and the Arts Research Board of McMaster University.
Engineering Brief 70

9:30 am

EB3-5 A Microphone Technique for Improved Stereo Image, Spatial Realism, and Mixing Flexibility: STAGG (Stereo Technique for Augmented Ambience Gradient)—*Jamie Tagg*, McGill University, Montreal, Quebec, Canada

While working on location, recording engineers are often challenged by insufficient monitoring. Poor (temporary control room) acoustics or headphone monitoring can make judgments regarding microphone choice and placement difficult. These choices often lead to timbral, phase, and stereo image problems. We are often forced to choose between the improved spatial imaging of near-coincident techniques and the acoustic envelopment from spaced omni-directional mics. This poster proposes a new technique: STAAG (Stereo Technique for Augmented Ambience Gradient), which aims to improve stereo image, acoustic realism, and flexibility in the mix. The STAAG technique allows for adjustment of the acoustic envelopment once in a proper monitoring environment.
Engineering Brief 71

Audio Industry Seminar **Monday, October 29**
9:45 am – 10:45 am **Room 114**

PMC: MASTERS OF AUDIO SERIES

Mixing and Recording the San Francisco Symphony

Presenter: **Jack Vad**

Grammy nominated producer/engineer Jack Vad has over 200 commercial classical audio releases for the biggest labels including following companies: BMG Classics, Harmonia Mundi, Koch International and many others. Since 1989 Jack is the Media Producer/Chief Engineer for the San Francisco Symphony and is responsible for both the music content and audio presentation of all the PBS/DVD/BD series including "Keeping Score" and any other high resolution audio-only productions.

In addition to these activities, Mr. Vad is involved in the research and development of multichannel audio recording techniques, audio-for-video methodology, and practical listening room acoustic enhancements. Jack will be sharing some of the latest high resolution recordings with us, both in stereo and surround.

Monday October 29 **10:00 am** **Room 125**

AES Standards Committee Meeting AESSC Plenary

Product Design Session 11 **Monday, October 29**
10:15 am – 12:15 pm **Room 130**

RUB & BUZZ AND OTHER IRREGULAR LOUDSPEAKER DISTORTION

Presenter: **Wolfgang Klippel**, Klippel GmbH, Dresden, Germany

Loudspeaker defects caused by manufacturing, aging, overload, or climate impact generate a special kind of irregular distortion commonly known as rub & buzz that are highly audible and intolerable for the human ear. Contrary to regular loudspeaker distortions defined in the design process the irregular distortions are hardly predictable and are generated by an independent process triggered by the input signal. Traditional distortion measurements such as THD fail in the reliable detection of those defects. The Tutorial discusses the most important defect classes, new measurement techniques, audibility, and the impact on perceived sound quality.

Tutorial 12 **Monday, October 29**
10:45 am – 12:15 pm **Room 131**

SOUND SYSTEM INTELLIGIBILITY

Presenters: **Ben Kok**, BEN KOK acoustic consulting, Uden, The Netherlands
Peter Mapp, Peter Mapp Associates, Colchester, Essex, UK

The tutorial will discuss the background to speech intelligibility and its measurement, how room acoustics can potentially affect intelligibility, and what measures can be taken to optimize the intelligibility of a sound system. ➥

Practical examples of real world problems and solutions will be given based upon the wide experience of the two presenters.

Game Audio Session 12 **Monday, October 29**
11:00 am – 12:30 pm **Room 132**

DOING MORE WITH LESS: HOW GAMES IMMERSIVELY SIMULATE AUDIO ON A BUDGET

Presenter: **Scott Selfon**, Microsoft, Redmond, WA, USA

How do games pack tens or hundreds of hours of experience onto a disc, hard drive, or the web? This talk covers some of the many techniques used (and the tradeoffs incurred) to make seemingly infinite, unique, and dynamic sounds and music—often with only a single content creator and a part-time programmer. Topics will include 3-D spatial simulation, compression, basic and advanced variation, and physical modeling techniques as applied to interactive media, with focus on topics that broadly apply to the full spectrum from mobile to console.

Live Sound Seminar 11 **Monday, October 29**
11:00 am – 12:30 pm **Room 120**

AUDIO DSP IN UNREAL-TIME, REAL-TIME, AND LIVE SETTINGS

Chair: **Robert Bristow-Johnson**, audioImagination, Burlington, VT, USA

Panelists: *Kevin Gross*, AVA Networks, Boulder, CO, USA
TBA

In audio DSP we generally worry about two problem areas: (1) the Algorithm: what we're trying to accomplish with the sound and the mathematics for doing it; and (2) Housekeeping: the “guzzintas” and the “guzzoutas,” and other overhead. On the other hand is the audio processing (or synthesis) setting which might be divided into three classes: (1) Non-real-time processing of sound files; (2) Real-time processing of a stream of samples; (3) Live processing of audio. The latter is more restrictive than the former. We’ll get a handle on defining what is real-time and what is not, what is live and what is not. What are the essential differences? We’ll discuss how the setting affects how the algorithm and housekeeping might be done. And we’ll look into some common techniques and less common tricks that might assist in getting non-real-time algorithms to act in a real-time context and to get “parts” of a non-live real-time algorithm to work in a live setting.

Special Event
DELIVERING AUDIO INNOVATION TO THE AUDIENCE
Monday, October 29, 11:00 am – 12:00 noon
Room 133

Moderator: **Rupert Brun**, BBC Audio & Music, London, UK

Since 1960 TV has gone color, widescreen, HD, and even 3-D but the audio experience of listening to the radio hasn't changed much since stereo was introduced in the late 1950s. Head of Technology for BBC Audio & Music Rupert Brun will explain how and why he has delivered audience facing innovation for radio listeners in the UK over the last 3 years. He will focus on 3 examples: Wide dynamic range (by radio standards) classical music, dialog enhancement for those with a hearing disability, and rendered binaural music and drama including surround sound derived from archive (co-incident pair) stereo recordings. Rupert will discuss the importance of working across organizational boundaries and of partnerships with other organizations and touch on the use of Twitter to gain real-time feedback, allowing an idea to be adjusted to meet the needs of the audience and turned from experiment to permanent service in just a few months.

Audio Industry Seminar **Monday, October 29**
11:00 am – 12:00 noon **Room 112**

SPECTRA SONICS

Spectra Sonics: Analog History, Digital Future

Presenters: **William C. Cheney**
James J. Romney

A presentation that will review the history of Spectra Sonics, including discrete analog innovation of the 60s and the early product designs that are still considered leading edge technology when measured by the standards of today. A brief overview of several new discrete analog products that are designed to readily interface with current digital technology will be presented. This will be an informal seminar in which a question/answer format is encouraged.

Audio Industry Seminar **Monday, October 29**
11:15 am – 12:15 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

Erik Zobler Presents:
5.1 Medley by the George Duke Band

Presenter: **Erik Zobler**

Multi Grammy winner Zobler has engineered and mixed hundreds of albums, DVDs, and TV and film scores. He is also a mastering engineer, front-of-house live sound mixer, and a writer for Mix Magazine. His credits include Miles Davis, Natalie Cole, Whitney Houston, Gladys Knight, Anita Baker, Sarah Vaughan, Al Jarreau, Stanley Clarke, Esparanza Spaulding, Bob James, Teena Marie, George Duke, and many more.

Erik will be showing some of his latest projects from Teena Marie, Jeffrey Osborne, and a 5.1 surprise from Frank Zappa.

Student/Career Event
SC-11 STUDENT RECORDING CRITIQUES
Monday, October 29, 12:30 pm – 1:30 pm
Room 114

Moderator: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA

Students! Bring in your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or surround work on CD, DVD, memory-stick, or hard disc, as clearly-labeled 44.1KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work.

The Student Recording Critiques are generously sponsored by PMC.

Tutorial 13 **Monday, October 29**
1:30 pm – 3:30 pm **Room 133**

MASTERING FOR VINYL—TODAY'S CHALLENGES

Presenter: **Scott Hull**, Masterdisk, New York, NY, USA

Scott Hull has been mastering for vinyl and digital for 28 years, seeing formats come, and go, and come back again. In the last few years there has been a renewed interest in producing vinyl among modern artists.

What has to be considered when you mix / master your music for vinyl? Scott will dig deep into the quality control issues and discuss several sure ways to sound great on your first pressing. Topics will include:

• Why contemporary CD mastering techniques do not produce the best sounding vinyl records.
• Long Sides—The relationship between Volume, Duration, and Quality.
• The Turntable—what does yours sound like?
• The quality control process: mixing – mastering – plating – pressing.
• The realities of the current vinyl market.
• Modern trends in record making / record sales.

Workshop 11 **Monday, October 29**
1:30 pm – 3:30 pm **Room 130**

MULTI-MICROPHONE APPLICATIONS AND TESTING IN TELECOMMUNICATIONS SYSTEMS

Chair: **Bob Zurek**, Motorola, USA

Panelists: *Samir Gupta*, Qualcomm
Plamen Ivanov, Motorola Mobility
Eric Skup, Audience

Over the last few years mobile communication devices have gone from single channel speech only devices to wide-band multichannel audio recording devices that can also be used for communication. The inclusion of multiple microphones into phones has improved voice quality, reduced background noise, and allowed for the capture of quality multichannel audio. This workshop is geared toward the designers of communications devices, the developers of multichannel audio algorithms, the programmers of applications utilizing these new multichannel acquisition devices, and the engineers tasked with testing these devices. The workshop will begin with an introduction to the subject, including presentations of various aspect of design, test, and standardization of multichannel microphone systems for telecommunications by each of the panelists. The panel will then discuss the current technology and testing landscape centered around questions from the audience.

Broadcast/Media Streaming Session 16
Monday, October 29 **1:30 pm – 3:00 pm**
Room 131

THE STREAMING EXPERIENCE

Chair: **Rusty Hodge**, SomaFM

Panelists: *Mike Daskalopoulos*, Dolby Labs, San Francisco, CA, USA
Jason Thibeault, Limelight
Leigh Newsome, Targetspot, New York, NY, USA
Robert Reams, Streaming Appliances/DSP Concepts, Mill Creek, WA, USAl

How are consumers listening to streaming audio today, and how can broadcasters improve that experience? What is the future direction that streaming audio should be taking?

Home streaming on hardware devices often suffers from limitations of the hardware UI in terms of selecting between tens of thousands of available channels. How can we improve that? Mobile streaming still doesn't match the convenience of turning on your car and having the AM/FM radio start right up. What can and is being done to change that?

What does the future of CODECs hold and what formats should broadcasters be streaming in to create the best quality while achieving universal accessibility? How do we improve the continuity between the home and mobile environment, especially in regard to customized streams? What are listeaners unhappy with now and what can be done about it?

We will also talk about the future of customized content and the integration of live broadcast content with customized streams.

Audio Industry Seminar **Monday, October 29**
1:30 pm – 2:30 pm **Room 114**

PMC: MASTERS OF AUDIO SERIES

Patricia Barber—The 5.1 Mixes!

Presenter: **Jim Anderson**, New York University, New York, NY, USA

Nine time Grammy winner Jim Anderson is a New York-based recording engineer and producer. Specializing in jazz and acoustic music, he has worked with producers like Bob Belden, Michael Cuscuna, Dr. Mark Feldman, Tommy LiPuma, Delfayo Marsalis, Kazunori Sugiyama, Akira Taguchi, Creed Taylor, among others and has recorded projects for Toshiko Akiyoshi, Patricia Barber, Terence Blanchard, James Carter, Ron Carter, Joe Henderson, J.J. Johnson, Branford Marsalis, Christian McBride, Gonzalo Rubalcaba, Maria Schneider, McCoy Tyner, Phil Woods. In television he has worked with the Muppets and has been the audio producer for "In Performance at the White House" and "The Kennedy Center Gala with the National Symphony" for PBS, among others.

Jim has chaired conventions for the Audio Engineering Society (AES) in New York, received the AES Fellowship Award, and is a Past-President of the AES. Still active in recording, Jim is also a Professor at NYU's Clive Davis Department of Recorded Music and the department's former chair.

Jim will be sharing with us the 5.1 remix of Patricia Barbers' " Modern Cool," released on Blu Ray Audio Only (Pure Audio).

Session EB4 **Monday, Oct. 29**
2:00 pm – 3:30 pm **Room 121**

LECTURES 2

Chair: **Tad Rollow**, Sennheiser Electronic Corporation

2:00 pm

EB4-1 Parametric Horn Design—*Ambrose Thompson*, Martin Audio, High Wycombe, UK

The principle barrier to more widespread use of numerical techniques for horn design is not a shortage of advanced computational libraries, rather the difficulty in defining the problem to solve in a suitable format. The traditional approach of creating horn geometry in large commercial CAD programs then exporting, meshing, assigning conditions, solving, and inspecting results denies the engineer the ability to easily iterate the design. We introduce an object-orientated parametric description of a horn that enables the engineer to modify meaningful parameters to generate geometry of appropriate complexity. The entire process is performed in one environment and an early implementation has been shown to allow nearly 100 iterations of one horn in one week. Results for this and another multi-driver HF horn are given.
Engineering Brief 72

2:15 pm

EB4-2 Integration of Touch Pressure and Position Sensing with Speaker Diaphragms—*Adrian Freed*, University of California, Berkeley, Berkeley, CA, USA

Speaker cones and other driver diaphragms are usually too fragile to be good sites for touch interaction. This can be solved by employing new, lightweight piezoresistive e-textiles with flat, rectangular, stiff surfaces used in full-range drivers from HiWave. Good low-frequency performance of piezoresistive fabric has an advantage over piezoelectric sensing for this situation. Applications of these integrated sensor/actuators include haptic feedback user interfaces and responsive electronic percussion instruments.
Engineering Brief 73

2:30 pm

EB4-3 Power Entry—Where High Performance Design Begins—*Christopher Peters, Diane Cupples*, Schurter, Santa Rosa, CA, USA

What is EMC/EMI? Where does it come from? What steps do I need to take to insure compatibility in today's world? Power quality as it relates to electro-magnetic compatibility is a growing topic among audio equipment manufacturers in the advent of the digital age. This abstract looks at equipment emissions and susceptibility and how to remedy noise problems with effective EMC design in the early stages. The paper and presentation will also offer ideas for integrating voltage selection, overcurrent protection and on/off switching into the overall aspects of compact, high performance design. Cord connection and retaining systems will also be covered.
Engineering Brief 74

2:45 pm

EB4-4 Design and Construction of the Tri-Baffle: A Modular Acoustic Modification System for Task-Based Mixing Experiments—*Scott Levine*,[1,2] *Brett Leonard*,[1,2] *Richard King*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

The Tri-Baffle is a modular system capable of providing multiple acoustic conditions within a space through the use of absorptive, reflective, and diffusive materials. Each baffle is constructed in a triangular frame, and capable of rotation via a ground-positioned motor. The system was designed and constructed to fit multiple experimental requirements such as acoustic characteristics, time requirements, installation concerns, and portability. As constructed, the Tri-Baffle is fully portable and is capable of installation in any space where task-based experimentation is desired.
Engineering Brief 75

3:00 pm

EB4-5 Another View of Distortion Perception— *John Vanderkooy, Kevin B. Krauel*, University of Waterloo, Waterloo, ON, Canada

Perception of distortion is difficult to determine since it relies critically on signal level. We study a distortion characteristic possessing a relative distortion independent of signal level—a simple change in slope between positive and negative signal excursion. A mathematical analysis is presented, showing the resulting distortion to be mainly even harmonic but with some rectification effects, which need consideration. Various signals are evaluated by informal A/B listening tests, including pure tones and music. Judiciously-chosen signals have distortions that are detectable only if they are above 1%, in keeping with psychoacoustic masking data, while real music signals are considerably more tolerant of distortion up to levels of 5% or more! This implies that, except for crossover distortion, present-day electronic systems are all sufficiently linear.
Engineering Brief 76

3:15 pm

EB4-6 How Can Sample Rates Be Properly Compared in Terms of Audio Quality?— *Richard King*,[1,2] *Daniel Levitin*,[1,2] *Brett Leonard*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

A listening test was designed to compare audio quality using varying sample rates. A Yamaha Disklavier player piano was used as a repeatable acoustic source, and the signal from the microphone preamps was sent to three identical analog to digital converters of the same make and model. The digitized signals were then re-converted to analog and compared to the original "live" signal through the use of a four-way switcher. Sample rates were 44.1, 96, and 192 kHz. Technical setup and the "somewhat inconclusive" results are discussed, along with some possible options for future testing.
Engineering Brief 77

Live Sound Seminar 12 **Monday, October 29**
2:00 pm – 4:00 pm **Room 120**

THE ART OF SOUND FOR LIVE JAZZ PERFORMANCES

Chair: **Lee Brenkman**

Panelists: *Mitch Grant*, Sound Engineer for jazz events in the San Diego area
Nick Malgieri, Mixed the past several Monterey Jazz Festivals

A discussion of the sonic presentation of America's native musical art form in settings ranging from the smallest cafés and night clubs to the largest outdoor festivals. In particular the panelists will focus on how sound reinforcement for this musical form can differ from other genres.

Student/Career Event
SC-12 STUDENT DELEGATE ASSEMBLY MEETING —PART 2
Monday, October 29, 2:00 pm – 3:30 pm
Room 132

At this meeting the SDA will elect a new vice chair. One vote will be cast by the designated representative from each recognized AES student section in the North & Latin American Regions. Judges' comments and awards will be presented for the Recording Competitions and Design Competitions. Plans for future student activities at local, regional, and international levels will be summarized.

Audio Industry Seminar **Monday, October 29**
2:30 pm – 4:30 pm **Room 114**

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