AES 137th Convention Program

October 9 – 12, 2014

Los Angeles Convention Center, LA, CA, USA

At recent AES conventions, authors have had the option of submitting complete 4- to 10-page manuscripts for peer-review by subject-matter experts. The following paper has been recognized as winner of the AES 137th Convention Peer-Reviewed Paper Award.

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The Audibility of Typical Digital Audio Filters in a High-Fidelity Playback System—
Helen M. Jackson, Michael D. Capp, J. Robert Stuart,
Meridian Audio Ltd., Huntingdon, UK
Convention Paper 9174
To be presented on Saturday, October 11, in Session 14 —Perception—Part 2

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The AES has launched an opportunity to recognize student members who author technical papers. The Student Paper Award Competition is based on the preprint manuscripts accepted for the AES convention.

A number of student-authored papers were nominated. The excellent quality of the submissions has made the selection process both challenging and exhilarating.

The award-winning student paper will be honored during the Convention, and the student-authored manuscript will be considered for publication in a timely manner for the *Journal of the Audio Engineering Society*.

Nominees for the Student Paper Award were required to meet the following qualifications:

(a) The paper was accepted for presentation at the AES 137th Convention.
(b) The first author was a student when the work was conducted and the manuscript prepared.
(c) The student author's affiliation listed in the manuscript is an accredited educational institution.
(d) The student will deliver the lecture or poster presentation at the Convention.

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The Winner of the 137th AES Convention Student Paper Award is:

The Duplex Panner: Comparative Testing and Applications of an Enhanced Stereo Panning Technique for Headphone-Reproduced Commercial Music—
Samuel Nacach, New York University, New York, NY, USA
Convention Paper 9134

To be presented on Friday, Oct. 10, in Session P7 —Cinema Sound, Recording and Production

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Session P1 **Thursday, Oct. 9**
9:00 am – 12:30 pm **Room 308 AB**

SPATIAL AUDIO—PART 1

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

9:00 am

P1-1 MPEG-H Audio—The New Standard for Universal Spatial / 3D Audio Coding—
Jürgen Herre,[1] *Johannes Hilpert,*[2] *Achim Kuntz,*[1] *Jan Plogsties*[2]
[1]International Audio Laboratories Erlangen, Erlangen, Germany
[2]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Recently, a new generation of spatial audio formats were introduced that include elevated loudspeakers and surpass traditional surround sound formats, such as 5.1, in terms of spatial realism. To facilitate high-quality bitrate-efficient distribution and flexible reproduction of 3D sound, the MPEG standardization group recently started the MPEG-H Audio Coding development for the universal carriage of encoded 3D sound from channel-based, object-based, and HOA-based input. High quality reproduction is supported for many output formats from 22.2 and beyond down to 5.1, stereo, and binaural reproduction—independently of the original encoding format, thus overcoming incompatibility between various 3D formats. The paper describes the current status of the standardization project and provides an overview of the system architecture, its capabilities, and performance.
Convention Paper 9095

9:30 am

P1-2 Bit Rate of 22.2 Multichannel Sound Signal Meeting Broadcast Quality—*Takehiro Sugimoto, Yasushige Nakayama, Satoshi Oode*, NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

The bit rate of a 22.2 multichannel sound (22.2 ch) signal meeting broadcast quality was investigated by preforming several subjective evaluations. 22.2 ch is currently planned to be transmitted by MPEG-4 AAC (advanced audio coding) in 8K Super Hi-Vision broadcast. A subjective evaluation of the basic audio quality of a

coded 22.2 ch signal was carried out using 49 stimuli made from a combination of seven bit rates and seven contents. Moreover, a subjective evaluation at two different listening positions and that of a downmixed 5.1 ch signal were also carried out for comparison with that of a 22.2 ch signal at the sweet spot. A bit rate meeting broadcast quality was found from the obtained results.
Convention Paper 9096

10:00 am

P1-3 Design, Coding and Processing of Metadata for Object-Based Interactive Audio—*Simone Füg, Andreas Hölzer, Christian Borß, Christian Ertel, Michael Kratschmer, Jan Plogsties*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

For object-based audio an appropriate definition of metadata is needed to ensure flexible playback in any reproduction scenario and to allow for interactivity. Important use-cases for object-based audio and audio interactivity are described and metadata requirements are derived. A metadata scheme is defined that allows for enhanced audio rendering techniques such as content-dependent processing, automatic scene scaling and enhanced level control. Also, a metadata preprocessing logic is proposed that prepares rendering and playout and allows for user interaction with the audio content of an object-based scene. In addition, the paper points out how the metadata can be transported efficiently in a bitstream. The proposed metadata scheme has been adopted and integrated into the currently finalized MPEG-H 3D Audio standard.
Convention Paper 9097

10:30 am

P1-4 On Spatial-Aliasing-Free Sound Field Reproduction Using Finite Length Line Source Arrays—*Frank Schultz, Till Rettberg, Sascha Spors*, University of Rostock, Rostock, Germany

Concert sound reinforcement systems aim at the reproduction of homogeneous sound fields over extended audiences for the whole audio bandwidth. For the last two decades this has been mostly approached by using so called line source arrays for which Wavefront Sculpture Technology (WST) was introduced in the literature. The paper utilizes a signal processing model developed for sound field synthesis in order to analyze and expand WST criteria for straight arrays. Starting with the driving function for an infinite and continuous linear array, spatial truncation and discretization are subsequently taken into account. The role of the involved loudspeakers as a spatial lowpass filter is stressed, which can reduce undesired spatial aliasing contributions. The paper aims to give a better insight on how to interpret the synthesized sound fields.
Convention Paper 9098

11:00 am

P1-5 The Focal Shift Phenomena for Focused Source Reproduction Using Loudspeaker Arrays—*Robert Oldfield, Ian Drumm*, University of Salford, Salford, Greater Manchester, UK

The focal shift phenomenon in optics describes how the position of the focus point in a focusing system is not simply defined by geometrical ray-based models but is affected by diffraction and is consequently a function of the size of the lens and the frequency of the light. The same effect is also observed in acoustics when looking at the focused field using physical focusing reflectors. This paper describes the focal shift phenomenon applied to the reproduction of focused sources with sound field synthesis systems, and presents a formula for the prediction of the actual rendered focal point position and also a frequency dependent positional correction for the improved rendering of a focused source with a given loudspeaker setup.
Convention Paper 9099

11:30 am

P1-6 Impulse Response Upmixing Using Particle Systems—*Nuno Fonseca*, ESTG/CIIC, Polytechnic Institute of Leiria, Leiria, Portugal

With the increase of the computational power of DSP's and CPU's, impulse responses (IR) and the convolution process are becoming a very popular approach to recreate some audio effects like reverb. But although many IR repositories exist, most IR recordings consider only mono or stereo. This paper presents an approach for impulse response upmixing using particle systems. Using a reverse engineering process, a particle system is created, capable of reproducing the original impulse response. By re-rendering the obtained particle system with virtual microphones, an upmixing result can be obtained. Depending on the type of virtual microphone, several different output formats can be supported, ranging from stereo to surround, and including binaural support, Ambisonics, or even custom speaker scenarios (VBAP).
Convention Paper 9100

12:00 noon

P1-7 ECMA-407: New Approaches to 3D Audio Content Data Rate Reduction with RVC-CAL —*Junaid Jameel Ahmad*,[1] *Claudio Alberti*,[1] *Jung Wook (Jonathan) Hong*,[2] *Brett Leonard*,[2] *Marco Mattavelli*,[1] *Clemens Par*,[3] *Schuyler Quackenbush*,[4] *Wieslaw Woszczyk*[1]
[1]Swiss Federal Institute of Technology (EPFL), Lausanne, Switzerland
[2]McGill University, Montreal, QC, Canada
[3]Swiss Audec, Morges, Switzerland
[4]Audio Research Labs, Scotch Plains, NJ, USA

Inverse problems have only been known in spatial audio for a very short time; their only solution, called "inverse coding" in literature, is essentially based on time-level modeling. Inverse problems, however, unlike parametric coding, require only an initial transmission of spatial side information, and thus can achieve much lower bitrates than could be achieved with parametric coding. For instance, inversely coded NHK 22.2 multichannel signals in combination with USAC may be delivered at bitrates as low as 48kb/s and optimum performance can be achieved in combination with commercially available HE-AAC v2 at 256kb/s—without any scaling of output channel order, and with moderate complexity in the decoder. A new way to perceptually

eliminate redundant information makes use of invariant theory inside the encoder. Invariants with Gaussian processes were unknown until 2010 and have represented one major problem in non-applied mathematics for more than a century: David Hilbert's proof that these coefficient functions form a field then insinuated that their existence in random processes was very likely. As will be shown, when applied to spatial audio coding, invariants represent a numerically efficient and perceptually powerful algebraic tool. We likewise present a 3D audio codec design for signals up to NHK 22.2 with two profiles: one profile, based on co-incidence, is able to code and synthesize a full Higher Order Ambisonics soundfield, up to order 6, at 48kb/s, 64kb/s, 96kb/s, 128kb/s, and above. The second profile, which optimizes de-correlation for phantom source imaging, codes channel-based or object-basedsignals at the same bitrates. The technology has been specified as the world's first international 3D audio standard ECMA-407 and may be further extended with static models in frequency domain. A preliminary version of this technology, based on a downmix in frequency domain, was submitted to MPEG's "Phase 2" selection of low-bitrate 3D coding technologies and made use of an USAC binary, which unfortunately offered no tuning options.
Convention Paper 9218

Session P2 **Thursday, Oct. 9**
9:00 am – 11:30 am **Room 309**

EDUCATION

Chair: **Tim Ryan**, Webster University, St. Louis, MO, USA

9:00 am

P2-1 Apprenticeship Skills in Audio Education: A Comparison of Classroom and Institutional Focus as Reported by Educators—*Doug Bielmeier*, Middle Tennessee State University, Murfreesboro, TN, USA

Recent research of audio industry employers indicated that their new hires lacked communication skills, which the employers deemed valuable for their new hire's success. In this research audio engineering technology (AET) educators were surveyed about the communication skills focused on in their classrooms, focus of their departments/ institutions, and their internship programs. The quantitative data suggested that both educators and their institutions lacked a focus on apprenticeship skills. Also, fewer than half of the institutions required an internship. Further research must be conducted to understand what these educators reported and how it affects AET education as a whole.
Convention Paper 9101

9:30 am

P2-2 Partnering Approaches for Teaching Music Technology—*Jeffrey Rodgers*, University of Saint Francis, Fort Wayne, IN, USA

The use of collaborative learning techniques is rapidly becoming a popular method for teaching 21st century skills across the United States. The term "partnering" has been used to refer to different types of collaborative learning; most recently being defined by Marc Prensky in his book, *Teaching Digital Natives, Partnering for Real Learning*. This paper addresses the need for more established methods for teaching music technology skills, concepts, and theories that utilize a collaborative, partnering style of instruction. Specifically, these partnering methods are intended for students in high school and higher education.
Convention Paper 9102

10:00 am

P2-3 Pathways through Recording Analysis—*William Moylan*, University of Massachusetts–Lowell, Lowell, MA, USA

If you are in the audio industry, you analyze recordings. What do you listen for? What do you hear? Why are you listening? There are many relevant answers to these questions depending on one's role in the industry or purpose for listening. This paper will explore the process of recording analysis and the idea of "pathways" through the many elements, dimensions, and functions that it might address; pathways that can be modified to suit the material and purpose for the analysis. Music recording will then be used as an example to bring a focus to the process. "Recording analysis" will then be the study of sound qualities of recordings and the interrelationships of those qualities and the music's materials and structure and its text.
Convention Paper 9103
[Paper was not presented but is available for purchase in the E-Library]

10:30 am

P2-4 Case Study: University Recording Arts Program Seeks to Educate, Engage, and Recruit High School Students—*Leslie Gaston-Bird, Lorne Bregitzer, Lynnae Rome*, University of Colorado Denver, Denver, CO, USA

A team of researchers from the University of Colorado Denver Recording Arts program visited the Denver School of the Arts high school in an effort to discover (1) whether students could detect an audible difference between music encoded as AAC and MP3 when comparing them to the original WAV file, (2) whether a trend in music career choices would appear based on gender, and (3) whether this visit could serve as an ongoing recruitment activity. The results presented here could be useful for other universities who want to engage in these activities. We will also consider the time, cost, and impact of a day-long visit.
Convention Paper 9104

11:00 am

P2-5 Musical Chairs: From Spectator to Stage—*Mike Godwin, Leslie Linton*, University of Western Ontario, London, Ontario, Canada

This presentation demonstrates an interdisciplinary and multidisciplinary project that involves ➥

the creation of an interactive computer application involving the fields of music education, music performance, acoustical engineering, and computer science. Typical music computer programs or "apps" usually involve creative strategies to explore various techniques for teaching the elements of music theory, history or composition and often use computer generated characters and music. This "app" is quite different in that it allows users to explore (tap, touch, pinch-to-zoom) actual performances through video; they can seamlessly "walk" through the orchestra, band or choir, and as they move around the audio changes according to where they are situated. Imagine "Google street view" with seamless video and audio instead of connected still photos and no sound.
Convention Paper 9105

Workshop 0 **Thursday, October 9**
9:00 am – 10:00 am **Room 409 AB**

MICROPHONE ARRAYS FOR MOBILE SPEECH APPLICATIONS

Presenters: **Eddy Brixen**, EBB consult
J. Keith McElveen, Wave Sciences

Microphone arrays are employed to spatially filter an acoustic scene to emphasize desired source directions—and, thereby, particular sources—and de-emphasize all others. Microphone arrays are currently being widely deployed in stationary applications, such as meeting rooms, but are just beginning to be deployed in mobile applications. An introduction to spatial filtering using microphone arrays in terms particularly relevant to mobile speech applications will be given. The theory of operation of both fixed and electronically-steerable arrays will be reviewed to serve as a foundation. From this foundation the practical design and user constraints of microphone arrays in mobile applications will then be addressed. The workshop will also be complemented with a variety of practical demonstrations of mobile microphone array implementations.

Tutorial 1 **Thursday, October 9**
9:00 am – 11:00 am **Room 406 AB**

LOUDSPEAKER DESIGN, PART 1: ALMOST EVERYTHING YOU EVER WANTED TO KNOW ABOUT LOUDSPEAKER DESIGN—A MASTER CLASS

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

This tutorial will walk the audience through an entire loudspeaker design as well as introducing the basic concepts of loudspeakers. Equivalent circuits, impedance, and Thiele-Small Parameters are shown. Inherent driver nonlinearities are explained. The effects of modal behavior and cone breakup are demonstrated. Closed Box and Ported Box systems are analyzed and several design examples are meticulously worked through, both with hand calculations and using CAD. Passive Radiator, Band Pass, and Transmission Line systems are also shown. Issues with multiple drivers and cabinet construction are discussed. Directivity and diffraction effects are illustrated. Crossover network design fundamentals are presented, with a specific design example for the previously shown ported enclosure design. A brief overview of room acoustic effects on loudspeakers is also presented.

This session is presented in association with the AES Technical Committee on Loudspeakers and Headphones

Tutorial 2 **Thursday, October 9**
9:00 am – 10:30 am **Room 306 AB**

3D AUDIO

Presenter: **Schuyler Quackenbush**, Audio Research Labs, Scotch Plains, USA

MPEG-H 3D Audio is the newest MPEG Audio standard. With the move to Ultra-High-Definition video and large screens that provide an immersive visual experience, it is compelling to have an equally immersive audio experience. MPEG-H 3D Audio provides both compression and flexible rendering for such immersive audio programs. These can be, for example, 9.1 or 22.2 channel audio programs for presentation on loudspeakers or spatialized for headphones. In addition, programs can include dynamic audio objects or can be Higher Order Ambisonic recordings. The standard makes extensive use of metadata to control the audio presentation and to support user interaction. A very important aspect of 3D Audio is its rendering engine. It is not expected that all consumers would have a 22.2 loudspeaker setup, so the rendering engine is able to adapt the audio program to the loudspeaker configuration of the consumer's setup. This can include fewer loudspeakers, incorrectly place loudspeakers or non-standard loudspeaker configurations.

The tutorial will review example scenarios in which immersive audio can be enjoyed (home theatre, tablet TV, smartphone TV); give an overview of the technology; and give a look at compression and rendering performance.

The session is presented in association with the AES Technical Committee on Coding of Audio Signals

Tutorial 3 **Thursday, October 9**
9:00 am – 11:00 am **Room 408 B**

AN OVERVIEW OF AUDIO SYSTEM GROUNDING AND SIGNAL INTERFACING—A MASTER CLASS

Presenter: **Bill Whitlock**, Whitlock Consulting, Oxnard, CA, USA

Equipment makers like to pretend noise problems don't exist, but this tutorial replaces myth and hype with insight and knowledge, revealing their true causes. 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, resulting in equipment with inadequate noise rejection in real-world systems. Another widespread design error gives some equipment a built-in noise problem. Simple, no-test-equipment troubleshooting methods can not only identify this equipment but pinpoint the exact location and cause of hum and buzz. Signal-path ground isolators are generally the best solution. Optimum interfaces between unbalanced and balanced connections, RF interference, and power-line treatments are also discussed as well as why some widely-used "cures" are both illegal and deadly.

Live Sound Seminar 1 **Thursday, October 9**
9:00 am – 11:00 am **Room 402 AB**

THEATER SOUND DESIGN IN LOS ANGELES

Presenters: **John Ballinger**, Independent Sound

Designer and Composer
Martin Carrillo, Independent Sound Designer
Cory Carrillo, Resident Sound Designer for The Laguna Playhouse
Cricket Myers
Vincent Olivieri, UC Irvine, Irvine, CA, USA
Veronika Vorel, Freelance, Los Angeles, CA, USA

Los Angeles has a vibrant theater community with hundreds of high quality productions each year. The sound design for many of these productions are created by LA sound designers who also often work nationally and internationally. This session will highlight some of these designers, their designs, and what makes the Los Angeles theater sound design work unique. The designers participating in this session will talk about their techniques for creating, producing and executing their designs.

Network Audio Session 1 Thursday, October 9
9:00 am – 10:00 am Room 404 AB

A PRIMER ON FUNDAMENTAL CONCEPTS OF MEDIA NETWORKING

Presenter: **Landon Gentry**, Audinate, Portland, OR, USA; Sydney, Australia

This session will cover the OSI model and how data travels through network layers (a "networking stack"): Layers 1, 2, 3 and 4; Cables, MAC Addresses, IP Addresses, and networking protocols. An overview of some networking standards and standards organizations, including the IEEE and the IETF. An introduction to IP data networking . . . it is how everything is already wired together. Identify some of the advantages and limitations of IP data networks with respect to real-time media. A brief discussion of IP networking standards and protocols that can be leveraged for media networking.

Sound for Picture 1 Friday, October 10
9:00 am – 10:30 am Room 306 AB

CINEMA LOUDNESS SUBCOMMITTEE TEST RESULTS AND PROPOSED SOLUTIONS

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

Panelists: *Florian Camerer*, ORF - Austrian TV, Vienna, Austria; EBU - European Broadcasting Union
Eelco Grimm, Grimm Audio, Utrecht, The Netherlands

Cinema audiences worldwide have heard movies get louder and louder since the introduction of Digital Cinema Distribution Masters. Government legislation in Europe (and proposed in Connecticut) suggest incorrect technical solutions. A committee working under TC-SDCTV Chair Brian McCarty has documented the problem and come to this convention with solutions for Standards consideration.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Thursday, October 9 9:00 am Room 405
Technical Committee Meeting on Transmission and Broadcasting

Thursday, October 9 10:00 am Room 405
Technical Committee Meeting on Electro Magnetic Compatibility

Network Audio Session 2 Thursday, October 9
10:15 am – 11:45 am Room 404 AB

HISTORY OF DIGITAL AUDIO NETWORKING

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

Digital audio networking was pioneered by the telephone companies in the 1960s. Digital audio came into use in professional applications starting in the late 1970s. Standard digital interconnect for professional applications emerged in the 1980s. It wasn't until the mid-1990s that real-time network distribution as we currently know it was born. This tutorial will follow the evolution of real-time digital audio networking, examining design decisions and changes in the underlying network technology that has enabled advancement to the current state of the art. Some of the specific networking technologies to be covered include: VoIP, CobraNet, MaGIC, Ethersound, A-Net, AES47, Livewire, AES50, ACIP, Dante, AVB, Q-LAN, RAVENNA and AES67. Historical perspective on the topic should give attendees an indication of where audio networking is headed.

This session is presented in association with the AES Technical Committee on Network Audio Systems

Recording and Production Session 1
Thursday, October 9 10:30 am – 12:00 noon
Room 403 AB

RAW TRACKS: FLEETWOOD MAC—A MASTER CLASS

Moderator: **Jonathan Pines**

Panelist: **Ken Caillat**

Renowned engineer/producer Ken Caillat will discuss, analyze, and deconstruct a classic Fleetwood Mac song "Second Hand News" in the inaugural "Raw Tracks" series at the AES 137th International Convention in Los Angeles.

Tutorial 4 Thursday, October 9
10:45 am – 12:45 pm Room 306 AB

FLOYD TOOLE—MASTER CLASS

Presenter: **Floyd Toole**, Acoustical consultant to Harman, ex. Harman VP Acoustical Engineering, Oak Park, CA, USA

Sound Reproduction: The State of Our Science

Most of the music we enjoy is generated by loudspeakers of differing pedigree, and propagated to our ears through spaces that can mostly be described as acoustically arbitrary. In spite of the obvious huge variations, humans have managed to not only enjoy reproduced music, but sometimes even to exhibit enthusiasm for it. Common acoustical measurements confirm the variations. Are they wrong? Do they matter? In double-blind subjective evaluations of loudspeakers in rooms, listeners exhibit strong and remarkably consistent opinions about the sound quality from loudspeakers. The challenge has been to identify those technical measurements that correlate with the subjective ratings. What is it that these listeners are responding to? This review will examine the acoustical properties of loudspeakers (the sound source), rooms (the acoustical conveyance) and listeners (the powerfully perceptive, and adaptable receptor). In some respects, our problems began when we started to make certain kinds of simplistic measurements. Two ears and a brain do not respond to complex sound fields the way an omnidirectional microphone and analyzer do. Belief that "room equalization" is a universal cure-all has added to the confusion. Much of the applied acoustical science was developed for large, reverberant, venues, not those most used for sound reproduction. We can do better. At the present time the audio industry is significantly lacking in meaningful standards, material specifications and loudspeaker performance descriptions. As a consequence, opinions often substitute for facts.

Tutorial 5 **Thursday, October 9**
10:45 am – 11:45 am **Room 409 AB**

PROTOTYPING AUDIO ALGORITHMS AS VST PLUGINS

Presenter: **Edward Stein**, DTS, Inc., Los Gatos, CA, USA

Professional algorithm designers, hobbyist programmers with a passion for audio, and experimentalist musicians often have a common challenge–"How do I hear my idea come to life?" Forums are full of posts with subjects like "I want to…, where do I start?" Depending on your budget, very comprehensive tools are out there with various trade-offs on ease and control. This tutorial looks at a powerful open-source C++ framework, JUCE for rapidly prototyping your ideas as VSTs (and other plugin formats) with a real-time graphical user interface. The focus will be on kick-starting beginners with a limited but working knowledge of C++. Topics will include: good practice for building a highly reusable C++ audio class library, basics of real-time audio plugins, quickly setting up and working with JUCE projects, real-time parameters (GUI, MIDI control, presets, etc.), and troubleshooting tips for when things don't go as you planned. By the end, you should be comfortable building your own VST plugins and be able to move forward focusing on what you care about–how it sounds.

Broadcast/Streaming Media Session 1
Thursday, October 9 **10:45 am – 12:45 pm**
Room 408 A

FACILITY DESIGN: TO MOVE, OR NOT TO MOVE? CONTRASTING SOLUTIONS—TWO WEST COAST FM STATIONS ADDRESS SHIFTING LISTENER NEEDS

Presenters: *Eddie Kramer*, Audio and Technical Consultant for KPFK 90.7FM Studio Upgrade, Los Angeles, CA, USA
John Storyk, Walters-Storyk Design Group, Highland, NY, USA
Mark Torres, Pacifica Radio Archives/KPFK 90.7FM - Los Angeles, CA USA

The meaningful (and growing) role that radio continues to play in listener lives is clearly illustrated by the contrasting upgrade decisions of two leading west coast FM stations. A staple of Seattle's booming music scene for over 40 years, KEXP 90.3 FM is preparing to move to a 21st Century broadcast facility. Situated in the world-famous Seattle Center, in the shadow of the iconic Space Needle, KEXP's new home will feature cutting edge broadcast/recording studios, and, a live performance venue designed to showcase visiting artists. Videotaped live performances will be posted on the station's website.

Concurrent with this move, KPFK Public Radio in Hollywood, a multi-award-winning, listener-sponsored part of the Pacifica Network since 1959, is upgrading its long-term home. The project will include a major acoustical update of their live performance studio and a brand new control room. With its 110,000-watt main transmitter atop Mount Wilson, KPFK is one of the most powerful FM stations in the western U.S. The redesigned live room and new control room will enable the station to produce high quality performance, political, public affairs, and cultural programming videos for posting on their Website.

Is it purely coincidental that both these stations have commissioned professional-level recording studio/performance venues? Is this a new trend for 21st Century Internet Radio? This panel discussion will include station engineers (TBD) and, celebrated producer/engineer Eddie Kramer who is consulting on the design of the KPFK studio.

Session EB1 **Thursday, Oct. 9**
11:00 am – 12:30 pm **S-Foyer 1**

POSTER SESSION 1

11:00 am

EB1-1 Analysis of Sound Field Generated by Line Arrays with Waveguides—*Xuelei Feng, Yong Shen, Simiao Chen, Ye Zhao*, Nanjing University, Nanjing, Jiangsu Province, China

Considering that a loudspeaker line array results from assembly of separate loudspeaker enclosures, it is important to take the gaps between enclosures into account and the line array is considered as a collection of several line sources of certain height. Particularly at high frequency, since the waveguides are used as the array element to achieve the line sources, the sound field generated by the arrays of waveguides is analyzed. The results are compared with those described in previous works that assumed that the array elements were perfect line sources.
Engineering Brief 156

11:00 am

EB1-2 Active Noise, Acoustic Echo, and Audio Ducking Using TMSC5535 eZdsp USB Stick Development Kit—*Prabhanjan Kadepurkar*, California State University, Long Beach, Long Beach, CA, USA

Audio Ducking, Active Noise, and Acoustic Echo Cancellation using TMS320C5535 eZdsp USB Stick Development Kit is an application system designed for live audio broadcasting and recording purposes. This application system initializes and executes adaptive audio and acoustic signal processing algorithms using a high performance and low power DSP TMS320C5535. In this study three sub-systems are discussed in brief and finally system setup and implementation is analyzed.
Engineering Brief 157

11:00 am

EB1-3 Loudspeaker Electrical Impedance Measurements Methods: A Brief Review—*Daniele Ponteggia*, Audiomatica Srl, Firenze, Italy

There are several possible methods to measure the loudspeaker driver electrical impedance. Those methods have followed the development of measurement instruments starting from the era of the simple needle voltmeters to the PC-based instruments widely available today. This paper will go through the theory and practice of impedance measurements with a series of examples where the pros and cons of each method are highlighted using real measurements. Effect of current sensor choice, noise, vibrations, and test level will be discussed in detail.
Engineering Brief 158

11:00 am

EB1-4 Implementation of Segmented Circular-Arc Loudspeaker Arrays—*D.B. (Don) Keele, Jr.*, DBK Associates and Labs, Bloomington, IN, USA

Circular-arc loudspeaker line arrays composed of multiple loudspeaker sources are used very frequently in loudspeaker applications to provide uniform vertical coverage [1, 2]. To simplify these arrays, the arrays may be simplified by forming the array using multiple straight-line segments or individual straight-line arrays. This approximation has errors because some of the speakers are now no longer located on the circular arc and exhibit a "bulge error." This error decreases as the number of segments increase or the splay angle of an individual straight segment is decreased. The question is how small does the segment splay angle have to be so that the overall performance is not compromised compared to the non-segmented version of the array? Based on two simple spacing limitations that govern the upper operating frequency for each type of array this paper shows that the bulge deviation should be no more than about one-fourth the center-to-center spacing of the sources located on each straight segment and that surprisingly, the maximum splay angle and array radius depends only on the number (N) of equally-spaced sources that are on a straight segment. As the number of sources on a segment increases, the maximum segment splay angle decreases and the required minimum array radius of curvature increases. Design guidelines are presented that allow the segmented array to have nearly the same performance as the accurate circular arc array.
Engineering Brief 159
[This EB will not be presented.]

11:00 am

EB1-5 Sound Field Intensity Measurements and Visualization around the Human Head Model—*Bozena Kostek, Adam Kurowski, Piotr Kryger, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

The main goal of this research study was to measure and visualize sound field in the presence and without presence of the human head model. Measurements were performed in the anechoic chamber at 5 cm grid. Experimental setup consisted of a multitone generator, two loudspeakers, human head model, intensimetric probe, the Cartesian robot applied for precise positioning of the acoustic sensor, and an analyzer. Based on the collected data sound field visualization was created in the form of colored maps and arrows illustrating pressure and the intensity vector at a given point in the presence of the artificial head, without this obstacle and the difference resulted from the mentioned conditions. A thorough analysis of the results obtained and conclusions followed.
Engineering Brief 160

11:00 am

EB1-6 Stereo Aligned Saturation—*Quintino Sardo, Angelo Mangano*, SKnote - SK Cooperative, La Punta (CT), Italy

All analog devices provide a finite dynamic range. Saturation occurs when available headroom ceiling is reached. Often, saturation is a desired effect to provide punch, brightness or changes in timbre, but stereo image is compromised because stereo saturation cannot be matched. Tests and methods for stereo aligned saturation are analyzed and described.
Engineering Brief 174
[This EB was not presented.]

Thursday, October 9 11:00 am Room 405
Technical Committee Meeting on Archiving, Restoration, and Digital Libraries

Thursday, October 9 11:00 am Room 407
Standards Committee Meeting on Measurement of Sound Systems in Rooms

Product Design Sess ion 1 Thursday, October 9
11:15 am – 12:45 pm Room 402 AB

THE SECRETS OF ANALOG DESIGN

Presenters: **Dave Derr**, Empirical Labs, Parsippany, NJ, USA
Dave Hill, Crane Song, Superior, WI, USA; Dave Hill Designs, Superior, WI, USA
Hutch Hutchison, Freelancer

Navigating the secrets of analog can be both frustrating and rewarding. In the design of an analog product there are many different ways to create the design. The choices that are made determine the sound, manufacturability, and cost of the device. With clever choices one can create a product with a unique and desirable sound while controlling the undesired factors such as cost, manufacturability issues, and reliability. We will look at a number of choices to create the same function and how the choices will affect the sonics of the device. In the market these choices can result in a product the fits the studio to the Concert Tour. What we work to avoid is a product that has so-what characteristics. Unique circuits have unique sounds; in this workshop the panel will explore some of the mystery and help attendees better understand the choices the options they have in designing in the analog domain.

Tutorial 7 **Thursday, October 9**
11:30 am – 12:30 pm **Room 408 B**

ALL ABOUT: THE DECIBEL

Presenter: **Alex Case**, University of Massachusetts Lowell, Lowell, MA, USA

The decibel is defined by an equation. "Yuck!" some might say. However that equation is rich with meaning and need not be a source of confusion. Total mastery of the decibel makes interfaces far more informative, specs sheets so much clearer, and every session easier.

Game Audio Session 1 **Thursday, October 9**
11:30 am – 12:45 pm **Room 406 AB**

SOUND BUSINESS: STRATEGIES AND FUNDAMENTALS IN GAME AUDIO CONTRACTS

Presenter: **Keith Arem**, President, PCB Productions and PCB Entertainment

Understand how new technologies and delivery methods can affect ownership, residuals, and copyright. Actors, musicians, composers, sound engineers, and directors can discover new opportunities in the expanding frontier of games. PCB President Keith Arem shares his experiences and insight into how games are transforming the way the entertainment industry works with sound.

Tutorial 6 **Thursday, October 9**
11:45 am – 12:15 pm **Room 309**

PRODUCE 3D AUDIO FOR MUSIC, FILM, AND GAME APPLICATIONS

Presenter: **Tom Ammermann**, New Audio Technology GmbH, Hamburg, Germany

Beyond of formats and applications but having later distribution in mind, the session will show production strategies and tools in 3D audio. Complete sessions from different genres will be opened and setups will be explained. Furthermore current and future end customer application and distribution possibilities will be shown.

Thursday, October 9 **12:00 noon** **Room 405**
Technical Committee Meeting on Audio for Telecommunications

Tutorial 8 **Thursday, October 9**
12:15 am – 12:45 pm **Room 309**

PRODUCE 3D HEADPHONE ENTERTAINMENT

Presenter: **Tom Ammermann**, New Audio Technology GmbH, Hamburg, Germany

Over 80% of the people listen to their music with headphones. The exploding mobile entertainment is close to a 100% headphone application. Furthermore evolution seems not to give humans a third ear. So headphones are not a fashion but a huge application now and in the future at all. How to conquer this marked and how to produce new virtual 3D audio productions is the task of this session.

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Thursday, October 9, 1:00 pm – 2:00 pm
Room 403 AB

Opening Remarks:
- Executive Director Bob Moses
- President Sean Olive
- President-Elect Andres Mayo
- Convention Co-chairs Michael MacDonald, Valerie Tyler

Program:
- AES Awards Presentation
- Introduction of Keynote Speaker
- Keynote Address by Alan Parsons

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

GOLD MEDAL AWARD
- **Floyd Toole**

BRONZE MEDAL AWARD
- **Mark Gander**
- **Peter Mapp**
- **Francis Rumsey**

BOARD OF GOVERNORS AWARD
- **Jim Anderson**

FELLOWSHIP AWARD
- **Alex Case**
- **Mark F. Davis**
- **Jim Kaiser**
- **Bob Lee**
- **Bruce Swedien**
- **Edmund Welly**
- **James Yeary**

CITATION
- **Christopher Freitag**

Keynote Speaker
Alan Parsons, the acclaimed engineer, musician, and record producer, has been involved in the production of several legendary albums throughout his career. He was fortunate enough to work as assistant engineer on the final two albums by The Beatles, and after he qualified as a fully-fledged recording engineer, he went on to work with Paul McCartney and The Hollies, among many others. But it was his contribution as engineer on Pink Floyd's classic *Dark Side of the Moon* that really got him worldwide attention and earned him his first of many GRAMMY nominations. That soon led to striking successes as a producer—notably with Pilot's "Magic," John Miles' "Highfly" and Steve Harley's "Make Me Smile (Come Up And See Me)." He also produced the hugely successful *Year of the Cat* album with Al Stewart and two albums with American progressive rock band Ambrosia. With Eric Woolfson, Parsons cofounded The Alan Parsons Project, famous for its revolving group of studio musicians and vocalists and its platinum albums and singles, including "Games People Play" and "Eye in the Sky."

Additionally, he developed *The Art and Science of Sound Recording*, a definitive collection of training videos presented by Parsons, which gives viewers his exclusive insider access to legendary musicians, producers and engineers and their award-winning recording techniques.

Alan has come full circle and is back in his role as producer and engineer, both at his own Santa Barbara studio and other studios internationally. In 2012, Alan produced an album called *Grand Ukulele* with virtuoso ukulele player Jake Shimabukuro. A collaboration with Steven Wilson as engineer and associate producer resulted in the album *The Raven That Refused to Sing (And Other Stories)* in 2013, hailed as a major success and reaching Top 5 on the album charts in Germany.

In his Keynote address, Mr. Parsons will provide his unique insights on the current and future direction of the music and recording industry from his singular perspective of success and experience throughout the history of popular music.

Thursday, October 9 2:00 pm Room 405
Technical Committee Meeting on Acoustics and Sound Reinforcement

Thursday, October 9 2:00 pm Room 407
Standards Committee Meeting on Digital Input/ Output Interfacing

Tutorial 9 Thursday, October 9
2:15 pm – 3:45 pm Room 409 AB

DITHER AND NOISE SHAPING IN DIGITAL AUDIO: HOWS AND WHYS

Presenter: **Duane Wise**, Wholegrain Digital Systems LLC, Boulder, CO, USA

This tutorial investigates finite-word-length in digital audio: how it differs from analog audio, the side-effects of signal quantization, and how the adverse effects of quantization can be diminished via dither and/or noise shaping. The tutorial presents the theory of quantization along with a live interactive audio demonstration that illustrates the motivation behind the theory.

The author invites attendees to download the presentation materials at http://www.wholegrain-ds.com/DigAud_Dither.pdf.

This session is presented in association with the AES Technical Committee on Signall Processing.

Broadcast/Streaming Media Session 2
Thursday, October 9 2:15 pm – 4:15 pm
Room 408 A

LOUDNESS FOR STREAMING AND RADIO

Presenters: *Dave Casey*, DTS Inc.
Frank Foti, Telos, New York, NY, USA
John Kean, NPR Labs, Washington DC, USA; National Public Radio
Thomas Lund, TC Electronic A/S, Risskov, Denmark
Scott Norcross, Dolby Laboratories, San Francisco, CA, USA
Robert Orban, Orban, San Leandro, CA, USA

The average consumer today stands little chance of accessing audio of even just the quality of compact cassette. The loudness wars devour most new productions as well as remastered tracks, and current legislation to prevent early hearing loss from listening to mobile devices promotes “music-sausaging” further.

Results from CEA’s new study on Loudness Range for Consumers in Various Listening Modes and Ambient Noise Levels are presented, and leveling challenges in mobile and streaming are described. Finally, the panel will discuss if FM radio is a lost cause, or if a better audio quality could be delivered to consumers, e.g., using the same loudness metrics now employed in TV.

Focus will be on technical and perceptual issues without the mentioning of commercial products.

Game Audio Session 2 Thursday, October 9
2:15 pm – 3:15 pm Room 408 B

EFFECTIVE INTERACTIVE MUSIC SYSTEMS: THE NUTS AND BOLTS OF DYNAMIC MUSICAL CONTENT

Presenter: **Winifred Phillips**, Generations Productions LLC, New York City Metropolitan Area, USA

Interactive methodologies have profoundly impacted the way that music is recorded, mixed and integrated in video games. From horizontal resequencing and vertical layering techniques for the interactive implementation of music recordings, to MIDI and generative systems for the manipulation of music data, the structure of game music poses serious challenges both for the composer and for the game audio engineer. This talk will examine the procedures for designing interactive music models and implementing them effectively into video games. The talk will include comparisons between additive and interchange systems in vertical layering, the lessons that can be learned from conventional stem mixing, the use of markers for switching between segments, and how to disassemble a traditionally composed piece of music for use within an interactive system.

Product Design Session 2 Thursday, October 9
2:15 pm – 2:45 pm Room 406 AB

WELCOME TO TOP GUN FOR PRODUCT DESIGNERS

Presenter: **Scott Leslie**, Product Design Track Chairperson

Following on from the special workshop this morning on “The Secrets of Analog Design,” this session will introduce the theme and sessions for the rest of the PD Track for AES LA. The curriculum has two themes this year. Some sessions will focus on great audio design topics as AES has done in the past.

The second and new theme is about how to develop great products in a global sourcing and engineering environment. Today’s engineers and managers must develop a skill set to deal with the complex world of global availability of:

1) Off-shore manufacturing and supply chain
2) Design services
3) Engineering services
4) Off the shelf and semi-custom subassemblies

Topics in this theme will provide valuable insight for understanding how to find, specify, integrate and manage these disciplines into successful end products.

This session is to provide an overview of the current global sourcing phenomenon and direct you to the PD track sessions that can leverage your talents in new ways to get products to market that are huge successes and meet your company’s business objectives.

Welcome to Top Gun!

Historical Event H1
EVOLUTION OF STUDIO ACOUSTIC DESIGN
Thursday, October 9, 2:15 pm – 4:15 pm
Room 404 AB

Moderator: **Mark Gander**, JBL/Harman Professional, Northridge, CA, USA

Presenters: *George Augspurger*, Perception Inc., Los Angeles, CA, USA
Chips Davis, Chips Davis Designs, LLC, San Francisco, CA, USA
Richard Schrag, Russ Berger Design Group Inc., Addison, TX, USA
John Storyk, Walters-Storyk Design Group, Highland, NY, USA

A panel of four distinguished studio designers, George Augspurger, Russ Berger, John Storyk, and Chips Davis, each with career durations of over forty years, will give individual perspectives on how methods have evolved in their work and the industry from the 1960s to the present day. They will outline the development of modern studio design principles by presenting key examples of their projects, to be followed by panel interaction and questions from the audience.

Special Event
THE LATIN PANEL:
GREAT PRODUCERS FROM LATIN AMERICA
Thursday, October 9, 2:15 pm – 4:15 pm
Room 402 AB

Moderator: **Andres A. Mayo**, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina

Panelists: *Daniel Anselmi*, Música CaReta, Montevideo, Uruguay
Rafa Arcaute, Rafa Arcaute, Buenos Aires, Argentina
Armando Avila, Cosmos, Mexico
Aureo Baqueiro, Brava! Music, Los Angeles, CA, USA; Nico Plus Music Group, Los Angeles, CA, USA
Humberto Gatica, Lionshare/Gatica Music, Los Angeles, CA, USA
Tweety González, Twitin Records, Buenos Aires, Argentina
Anibal Kerpel, SK Associates, Los Angeles, CA, USA
Sebastian Krys
Guido Nisenson
Rafa Sardina, Fishbone Productions, Inc., Los Angeles, CA, USA

Por primera vez en la historia, tendremos un panel oficialmente hablado en idioma Español en una Convención Internacional de AES. El panel "Grandes Productores de América Latina! reunirá a algunos de los más grandes nombres de la región, todos ellos múltiples ganadores de Premios Grammy, para presentar sus últimas producciones y hablar acerca del estado de la industria en América Latina.

For the first time in AES history, there will be a panel officially held in Spanish in an International AES Convention. The panel "Grandes Productores de América Latina" ("Great producers from Latin America") will bring together several of the biggest names in the region, all of them multiple Grammy Award winners, to present their most recent work and discuss the state of the musical industry in Latin America.

Student Event/Career Development
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Thursday, October 9, 2:15 pm – 3:45 pm
Room 306 AB

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 Student Recording Competition categories and the 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 dialog 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 Sunday, October 12.

Session P3 **Thursday, Oct. 9**
2:30 pm – 6:30 pm **Room 308 AB**

SPATIAL AUDIO—PART 2

Chair: **Eric Benjamin**

2:30 pm

P3-1 A Polygon-Based Panning Method for 3D Loudspeaker Setups—*Christian Borß*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

In this paper we introduce the "Edge Fading Amplitude Panning" (EFAP) method for 3D loudspeaker setups. Similar to other panning methods like Vector Base Amplitude Panning (VBAP), it can be used to create phantom sources between the loudspeaker positions. The proposed method features symmetric panning gains for symmetric loudspeaker setups, N-wise panning by using polygons instead of triangles, and a better behavior for large opening angles between loudspeakers while involving a computational complexity that is in the same order of magnitude as VBAP.
Convention Paper 9106

3:00 pm

P3-2 Utilizing Contralateral ICTDs to Stabilize Lateral Imaging in 5.1 Surround Systems—*Michael Tierney, Adrian Tregonning*, New York University, New York, NY, USA

In 5.1 surround sound systems the problems of lateral image instability and a non-linear lateral panning path are well known. Alternative panning techniques have been developed in an attempt to overcome these problems, but often improved spatial imaging compromises spectral integrity. This is an undesirable tradeoff in the context of music mixing and production. The current paper examines the effect of low-level inter-

channel time differences (ICTDs) in contralateral channels with respect to lateral imaging. Subjective experiments evaluated localization perceptions with ICTDs of 1 ms in either the front or surround contralateral channel. This led to more accurate and predictable lateral image positioning with minimal spectral coloration. The results are used to propose a more effective 5.1 lateral panning mechanism.
Convention Paper 9107

3:30 pm

P3-3 Investigation into the Impact of 3D Surround Systems on Envelopment—*Paul Power, Bill Davies, Jonathan Hirst*, University of Salford, Salford, Greater Manchester, UK

This investigation assessed a number of 2D and 3D surround systems focusing on the attribute "envelopment" to determine if surround systems with height significantly enhance the perception of envelopment over current 2D systems. To assess each of the systems an objective and subjective method was used. The objective method consisted of measuring the IACC (Inter-Aural Cross Correlation) of each reproduction system by reproducing three different types of sound scenes over each system. In addition, a subjective listening test was also carried out to evaluate the perceived envelopment. The objective measure showed that the introduction of height channels did lower the IACC. Further, subjective listening test results showed that there were significant differences between the height and horizontal surround systems in terms of envelopment, however this was dependent on the audio stimulus used. Finally, a correlation between the objective and subjective measures showed a strong negative correlation.
Convention Paper 9108

4:00 pm

P3-4 An Architecture for Reverberation in High Order Ambisonics—*Fernando Lopez-Lezcano*, Stanford University, Stanford, CA, USA

This paper describes a reverberation architecture implemented within the signal chain of a periphonic HOA (High Order Ambisonics) audio stream. An HOA signal (3rd order in the example implementation) representing the dry source signal is decoded into an array of virtual sources uniformly distributed within the reverberant space being simulated. These virtual sources are convolved with independent, decorrelated impulse responses, optionally tailored to model spatial variations of the simulated reverberation. The output of each convolver is then encoded back into High Order Ambisonics and mixed with the original Ambisonics dry signal. The result is a convolution reverberation engine with a HOA input that outputs HOA and maintains the spatial characteristics of the input signal.
Convention Paper 9109
[Paper was not presented but is available for purchase in the E-Library]

4:30 pm

P3-5 Spatial Calibration of Surround Sound Systems including Listener Position Estimation—*Guangji Shi, Martin Walsh, Edward Stein*, DTS, Inc., Los Gatos, CA, USA

While most modern surround sound formats specify ideal loudspeaker placement, it is often impractical to comply with these specifications in most homes. In this paper we propose a simplified approach to spatial calibration for incorrectly set up surround sound systems. The proposed system utilizes a microphone array embedded into a component of the reproduction system whose location is predictable, such as a sound bar or a front center speaker. In addition to estimating loudspeaker positions, the proposed system is able to estimate a listener's position using voice input and calibrate the surround sound system utilizing the estimated listener position. Tests conducted in a typical living room setup show that the proposed system is able to improve the listening experience on such compromised systems with only simple user interactions such as voice commands.
Convention Paper 9110

5:00 pm

P3-6 Comparison of Pressure-Matching and Mode-Matching Beamforming for Methods for Circular Loudspeaker Arrays—*Filippo Maria Fazi,*[1] *Mincheol Shin,*[1] *Ferdinando Olivieri,*[1] *Simone Fontana,*[2] *Lang Lang Yue*[2]
[1]University of Southampton, Southampton, Hampshire, UK;
[2]Huawei European Research Center, Munich, Germany

Pressure-matching and mode-matching are two well-known strategies used for the computation of beamforming digital filters for microphone and loudspeaker arrays. A theoretical comparison is presented of these two methods when these are applied to a circular loudspeaker array mounted on a rigid cylinder. The pressure-matching method is used to generate the desired acoustic pressure at a number of control points arranged in the far field of a circular loudspeaker array, while in the case of mode-matching an attempt is made to minimize the squared error between the Fourier coefficients that represent the reproduced and tar-get radiation pattern of the array. It is shown that, in the case under consideration, the two strategies are identical if the effect of spatial aliasing is negligible.
Convention Paper 9111

5:30 pm

P3-7 MIAP: Manifold-Interface Amplitude Panning in Max/MSP and Pure Data—*Zachary Seldess*, Sonic Arts R&D, UC San Diego, San Diego, CA, USA

This paper discusses MIAP (Manifold-Interface Amplitude Panning), a new freely available implementation of Meyer Sound's SpaceMap abstract spatialization software via a collection of C externals for Max/MSP and Pure Data. SpaceMap's technical and conceptual innovations are discussed and placed within the larger context of widely available codified spatialization algorithms and approaches such as Vector-base amplitude panning. An examination of the new implementation is made along with discussion of ➥

added features resulting from the translation.
Convention Paper 9112

6:00 pm

P3-8 Aurally Aided Visual Search Performance Comparing Virtual Audio Systems—*Camilla H. Larsen, David S. Lauritsen, Jacob J. Larsen, Marc Pilgaard, Jacob B. Madsen, Rasmus Stenholt*, Aalborg University, Aalborg, Denmark

Due to increased computational power reproducing binaural hearing in real-time applications, through usage of head-related transfer functions (HRTFs), is now possible. This paper addresses the differences in aurally-aided visual search performance between an HRTF enhanced audio system (3D) and an amplitude panning audio system (panning) in a virtual environment. We present a performance study involving 33 participants locating aurally-aided visual targets placed at fixed positions, under different audio conditions. A varying amount of visual distractors were present, represented as black circles with white dots. The results indicate that 3D audio yields faster search latencies than panning audio, especially with larger amounts of distractors. The applications of this research could fit virtual environments such as video games or virtual simulations.
Convention Paper 9150

Session P4 Thursday, Oct. 9
2:30 pm – 5:30 pm Room 309

TRANSDUCERS—PART 1

Chair: **Joerg Panzer**, R&D Team Software Development, Germany

2:30 pm

P4-1 The Dynamics Detection and Processing Method for Preventing Large Displacement Transducer Damage Problem—*Yu-Ting Tsai, Jin H. Huang*, Feng Chia University, Taichung, Taiwan

The method for avoiding the large displacement damage problem of the transducer diaphragm is presented in this study. To account for the displacement and supply current, a set of transduction equations with parameters for the receiver is established. The numerical solver is used to obtain the displacement values from the transduction equations of the transducer and then limit the peak of the current and coil velocity by using the dynamics limiter. Once the peak limited process is done, the safe input voltage can be obtained satisfactorily for preventing the impulse damage of displacement. The numerical and experimental results indicate that the proposed method features high efficiency and ability applied in the wide application of transducers.
Convention Paper 9113

3:00 pm

P4-2 Specifying Xmax for Micro-Speakers and Smart Amplifiers—*Géraldine Vignon*,[1] *Shawn Scarlett*[2]
[1]NXP, Leuven, Belgium
[2]NXP, Nijmegen, The Netherlands

Smart amplifiers are a new generation of amplifiers using a real-time feedback loop to ensure the speaker stays within its physical limits, and they are capable of delivering significantly more output power than previously possible. This jump requires the definition of new performance ratings for micro-speakers. The purpose of this paper is to recommend test procedures and methods for the specification of a maximal speaker-membrane excursion (Xmax) suitable for an application that uses a smart amplifier. The proposed set of criteria to characterize this usable excursion should allow micro-speaker vendors to provide optimized and reliable solutions to their customers.
Convention Paper 9114

3:30 pm

P4-3 Comparative Study of Si and SiC MOSFET for High Voltage Class D Audio Amplifiers—*Dennis Nielsen, Arnold Knott, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

Silicon (Si) Metal–Oxide–Semiconductor Field-Effect Transistors (MOSFETs) are traditionally utilized in class D audio amplifiers. It has been proposed to replace the traditional inefficient electrodynamic transducer with the electrostatic transducer. This imposes new high voltage requirements on the MOSFETs of class D amplifiers and significantly reduces the selection of suitable MOSFETs. As a consequence it is investigated if Silicon-Carbide (SiC) MOSFETs could represent a valid alternative. The theory of pulse timing errors are revisited for the application of high voltage and capacitive loaded class D amplifiers. It is shown that SiC MOSFETs can compete with Si MOSFETs in terms of THD. Validation is done using calculations and a ± 500 V amplifier driving a 100 nF load. THD+N below 0.3% is reported.
Convention Paper 9115
[Paper will be presented by Thomas Birch]

4:00 pm

P4-4 Adaptive Stabilization of Electro-Dynamical Transducers—*Wolfgang Klippel*, Klippel GmbH, Dresden, Germany

A new control technique for electro-dynamical transducer is presented that stabilizes the voice coil position, compensates for nonlinear distortion, and generates a desired transfer response by preprocessing the electrical input signal. The control law is derived from transducer modeling using lumped elements and identifies all free parameters of the model by monitoring the electrical signals at the transducer terminals. The control system stays operative for any stimulus including music and other audio signals. The active stabilization is important for small loudspeakers generating the acoustical output at maximum efficiency.
Convention Paper 9116

4:30 pm

P4-5 Retrofitting a Complex, Safety-Critical PA System for Periodic Testing—*Gregor Schmidle*,[1] *Philipp Schwizer*,[1] *Winfried Häns*[2]

[1]NTi Audio AG, Schaan, Liechtenstein
[2]Kernkraftwerk Gundremmingen GmbH, Gundremmingen, Germany

This paper describes considerations and the implementation of retrofitting a fully-automated procedure, for testing a public address system, into a safety-critical environment (a nuclear power plant). There are over 4000 loudspeakers, about 200 amplifiers as well as various alarm-signal generators that need to be tested every day within a few minutes. Additionally, all command room microphones are checked using a semi-automated procedure. The procedures were designed and configured to not only reliably detect single defective components, but also to not produce any false alarms.
Convention Paper 9117

5:00 pm

P4-6 The Correlation between Distortion Audibility and Listener Preference in Headphones— *Steve Temme*,[1] *Sean Olive*,[2] *Steve Tatarunis*,[1] *Todd Welti*,[2] *Elisabeth McMullin*[2]
[1]Listen, Inc., Boston, MA, USA
[2]Harman International, Northridge, CA USA

It is well-known that the frequency response of loudspeakers and headphones has a dramatic impact on sound quality and listener preference, but what role does distortion have on perceived sound quality? To answer this question, five popular headphones with varying degrees of distortion were selected and equalized to the same frequency response. Trained listeners compared them subjectively using music as the test signal, and the distortion of each headphone was measured objectively using a well-known commercial audio test system. The correlation between subjective listener preference and objective distortion measurement is discussed.
Convention Paper 9118

Product Design Session 3 Thursday, October 9
2:45 pm – 4:45 pm Room 406 AB

BRINGING IT ALL TOGETHER—A PANEL SESSION

Chair: **Scott Leslie**, PD Squared, Irvine, CA, USA

Panelistss: *Stefan Feistel*, AFMG Technologies GmbH, Berlin, Germany
Steve Hutt
Denis Labrecque, Analog Devices, San Jose, CA, USA
Bill Whitlock, Whitlock Consulting, Oxnard, CA, USA

Today's engineers and managers must develop a skill set to deal with the complex world of global availability of: (1) Off-shore manufacturing and supply chain; (2) Design services; (3) Engineering services; (4) Off the shelf and semi-custom subassemblies. This panel session will kickoff the Track and open up the discussion on the key theme for this year's AES Product Design Track. The panelists will represent the following subsystem areas in bringing together all technology and supply areas required in bringing great products to market: (1) Digital Processing, (2) Amplifiers, (3) Software, (4) Loudspeaker Drivers, (5) EMI. This is a unique opportunity to engage these experts of each of their domains in a discussion at the overall system level. Each will also be conducting tutorials on their subject in depth during the Product Design Track.

Session P5 Thursday, Oct. 9
3:00 pm - 4:30 pm Foyer 1

POSTERS: AUDIO SIGNAL PROCESSING

3:00 pm

P5-1 An Evaluation of Chromagram Weightings for Automatic Chord Estimation—*Zhengshan Shi, Julius O. Smith, III*, Stanford University, Stanford, CA, USA

Automatic Chord Estimation (ACE) is a central task in Music Information Retrieval. Generally, audio files are parsed into chroma-based features for further processing in order to estimate the chord being played. Much work has been done to improve the estimation algorithm by means of statistical models for chroma vector transitions, but not as much attention has been given to the loudness model during the feature extraction stage. In this paper we evaluate the effect on chord-recognition accuracy due to the use of various nonlinear transformations and loudness weightings applied to the power spectrum that is “folded" to form the chromagram in which chords are detected. Nonlinear spectral transformations included square-root magnitude, magnitude, magnitude-squared (power spectrum), and dB magnitude. Weightings included A-weighted dB and Gaussian-weighted magnitude.
Convention Paper 9119

3:00 pm

P5-2 CUDA Accelerated Audio Digital Signal Processing for Real-Time Algorithms— *Nicholas Jillings*, *Yonghao Wang,* Birmingham City University, Birmingham, UK

This paper investigates the use of idle graphics processors to accelerate audio DSP for real-time algorithms. Several common algorithms have been identified for acceleration and were executed in multiple thread and block configurations to ascertain the desired configuration for the different algorithms. The GPU and CPU performing on the same data sizes and algorithm are compared against each other. From these results the paper discusses the importance of optimizing the code for GPU operation including the allocating shared resources, optimizing memory transfers, and forced serialization of feedback loops. It also introduces a new method for audio processing using GPU's as the default processor instead of an accelerator.
Convention Paper 9120

3:00 pm

P5-3 A Modified Variable Step Size NLMS Algorithm for Acoustic Echo Cancellation Application—*Youhong Lu, Syavosh Zadissa*, Microsoft, Redmond, WA, USA

The Variable Step Size (VSS) Normalized Least Mean-Square (NLMS) algorithm has been studied in depth. Numerous publications have cov- ➥

ered this technique from both theoretical and practical point of views. This contribution builds on the past knowledge and proposes an improvement. This is a single filter approach without any double talk detection. We will show that the proposed technique, in the context of acoustic echo cancellation, offers superior performance in terms of convergence speed, misalignment error, while offering superior resilience to low Echo to Interference Ratio, and Echo Path Change.
Convention Paper 9121
[Paper was not presented but is available for purchase in the E-Library]

3:00 pm

P5-4 Robust Artificial Bandwidth Extension Technique Using Enhanced Parameter Estimation—*Jonggeun Jeon,[1] Yaxing Li,[1] Sangwon Kang,[1] Kihyun Choo,[2] Eunmi Oh,[2] Hosang Sung[2]*
[1]Hanyang University, Korea
[2]Samsung Electronics, Suwon, Korea

We propose a robust artificial bandwidth extension (ABE) technique to improve the narrowband speech signal quality using enhanced excitation estimation and spectrum envelope. For excitation estimation, we use a whitened narrowband excitation signal, generated by passing the excitation signal through a whitening filter. An adaptive spectral double shifting method is introduced to obtain an enhanced wideband excitation signal. For envelope estimation, we propose an enhanced combined method using the codebook and linear mapping. The proposed ABE system is applied to the decoded output of an adaptive multi-rate (AMR) codec at 12.2 kbps. We evaluate its performance using spectral distortion, wideband perceptual evaluation of speech quality, and a formal listening test. The objective and subjective evaluation confirm that the proposed ABE system provides better speech quality than that of AMR at the same bit rate.
Convention Paper 9122

3:00 pm

P5-5 Audio Signal Recovery from Single-Frame Randomly Gated Fourier Magnitudes—*Dominic Fannjiang,[1] Albert Fannjiang[2]*
[1]Davis Senior High School, Davis, CA, USA
[2]University of California, Davis, Davis, CA, USA

Few-frame phase retrieval is motivated by the demand of real time audio signal processing which is severely ill-posed and fundamentally challenging. This paper is an exploratory study of single-frame phase retrieval of audio signals with two additional ingredients: a random gating function and symmetry-breaking DC component. In general, randomly phased gating and a suitably chosen DC component can bring the success rate of single-frame phase retrieval to unity and yield accurate, stable, fast convergent numerical reconstruction. The tradeoff between the diversity of the gate and the magnitude of DC component is investigated.
Convention Paper 9123

3:00 pm

P5-6 Triode Emulator: Part 2—*Dimitri Danyuk*, Consultant, Palmetto Bay, FL, USA

The paper describes method for accurate emulating of triode behavior at high input levels. Under gross overload the grid current becomes a main origin of distortion. The measurements of grid current for popular 12AX7 triode are presented. In the region of interest grid current dependence on input signal is emulated with a simple circuit. The output harmonic weighting of the emulator is examined and compared with existing solution. The results can be applied to solid-state guitar amplifiers.
Convention Paper 9124
[Paper was not presented but is available for purchase in the E-Library]

3:00 pm

P5-7 A SIMULINK Toolbox of Sigma-Delta Modulators for High Resolution Audio Conversions—*Isacco Arnaldi, Yonghao Wang***,** Birmingham City University, Birmingham, UK

Sigma–Delta modulation is the only form of analog-to-digital conversion that allows achievement of high bit resolution at relatively low costs. There is a lack of tools in academic and in industry that allow entry level engineers to familiarize with the concepts governing this conversion technique, especially for high orders, multi-bit, and different architectures of sigma-delta modulators. The goal of this paper is to present a graphical toolbox, developed in Simulink and based on the behavioral model previously presented in [2] and available in [3] that allows to simulate and theoretically evaluate ten different architectures, continuous and discrete time, as well as single- and multi-bit implementations of different orders of analog-to-digital sigma-delta modulators.
Convention Paper 9125
[Paper was not presented but is available for purchase in the E-Library]

Thursday, October 9 3:00 pm Room 405
Technical Committee Meeting on Microphones and Applications

Games Audio Session 3 Thursday, October 9
3:15 pm – 4:45 pm Room 408 B

SOUND DESIGN & MIX: CHALLENGES AND SOLUTIONS—GAMES, FILM, ADVERTISEMENT

Presenters: **Charles Deenen**, Mixer / Sound Supervisor, Source Sound Inc., Los Angeles, CA, USA
John Fasal, Sound Designer Specializing in Sound Effects Recording
Tim Gedemer, Owner / Sound Supervisor, Source Sound Inc., Los Angeles, CA, USA
Csaba Wagner, Freelance Sound Designer
Bryan Watkins, Game Sound Supervisor, Warner Brothers Game Audio

Different media in today's marketplace require different (immersive) sound. Each media format has it's own level requirements, channel distinctions, hidden technical challenges to overcome, and more. This panel will attempt to demonstrate, through example and discussion, how audio production and post production techniques can (and should) be effectively tailored to their respective visual release media formats, including Games, Mobile, Trailers, Commercials, and Film.

Thursday, October 9 4:00 pm Room 405
Technical Committee Meeting on Hearing and Hearing Loss Prevention

Workshop 1 Thursday, October 9
4:30 pm – 6:30 pm Room 306 AB

LOUDNESS WARS: GIVE PEAKS A CHANCE

Presenters: **Florian Camerer**, ORF - Austrian TV, Vienna, Austria; EBU - European Broadcasting Union
Bob Ludwig, Gateway Mastering Studios, Inc., Portland, ME, USA
Thomas Lund, TC Electronic A/S, Risskov, Denmark
George Massenburg, Schulich School of Music, McGill University, Montreal, Quebec, Canada
Susan Rogers, Berklee College of Music, Boston, MA, USA

Music production, distribution, and consumption has been caught in a vicious spiral rendering two decades of our music heritage irreversibly damaged. Today, new tracks and remastered ones typically sound worse than what could even be expected from compact cassette. As a pro society, do we just sit by and let that happen on our watch?

With us to discuss the most important pro audio topic today are Susan, George and Bob—the finest from music auditory research, production, and mastering—while Florian and Thomas are at the helm of two European initiatives to also help reverse the vicious spiral: EBU 308 AB28 and EU legislation to prevent early hearing loss from listening to personal music players. Another glimpse of hope is iTunes Radio with loudness normalization on by default.

More reasonable distribution will not and should not prevent engineers from squashing music for artistic reasons, but it will take away any “advantage” of being louder. Learn to live with it!

Tutorial 10 Thursday, October 9
4:30 pm – 6:00 pm Room 409 AB

SPEECH TRANSMISSION INDEX (STI) MEASUREMENTS IN PRACTICE

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

The Speech Transmission Index is today the most widely used international measure of potential speech intelligibility. In particular it is cited and performance requirements are incorporated in many national and international sound system and emergency sound system / voice alarm system standards and codes of practice. Few standards however state how STI performance should be measured and the equipment required to carry out such measurements. The workshop will discuss measurement techniques, data analysis, measurement equipment (including smart phone apps), equipment calibration, and the capture and logging of measurement data. The tutorial will be given by Dr. Peter Mapp, a leading authority on STI measurement and the current chairman of IEC 60268-16—the international standard relating to STI.

This session is presented in association with the AES Technical Committee on Acoustics and Sound Reinforcement

Broadcast/Streaming Media Session 3
Thursday, October 9 4:30 pm – 6:00 pm
Room 408 A

ROUTING AUDIO IN A BROADCAST FACILITY

Chair: **Mike DaSilva**, CBS Radio, Sacramento, CA, USA

Panelists: *Steve Dove*, Wheatstone Corporation, New Bern, NC, USA
Andreas Hildebrand, ALC NetworX GmbH, Munich, Germany
Herbert Lemcke, Lawo North America, Tornonto, Ontario, Canada
Al Salci, SAS Audio
Greg Shay, The Telos Alliance, Cleveland, OH, USA

What new audio routing advancements are we likely to see in a typical broadcast plant design? Over the past decade, more broadcasters have had to upgrade their facilities while having to utilize modern audio routing techniques. This panel will discuss such advancements including protocols such as Ravenna, AES67, AVB, Dante, Livewire, and other related techniques and the typical challenges broadcasters are likely to face.

Live Sound Seminar 2 Thursday, October 9
4:30 pm – 6:30 pm Room 402 AB

THE CREATIVE TECHNOLOGIST: CAREER PATHS OF THE AUDIO PROFESSIONAL

Moderator: **Paul Freudenberg**, Rat Sound Systems, Inc., Thousand Oaks, CA, USA

Panelists: *Kevin Becka*, Blackbird Academy, Nashville, TN
Mario DiCola, Audio Labs Systems, Florence,Italy
Claudio Lastrucci, Powersoft SpA
Shawn Murphy, Recording and Live Sound Engineer
Dave Rat, Rat Sound Systems, Inc., Thousand Oaks, CA, USA
Frederick Vogler, Sonitus Consulting, Los Angeles, CA, USA
Brett Valasek, ATK Audiotek, Valencia, CA

The goal of this panel discussion is primarily to inspire those people who are thinking about or just starting a career in audio but all are welcomed to hear this panel of audio expert discuss being part of an audio community. Those on the panel will address the action and attitudes that are required to make a living in music recording, live sound production, broadcast audio or other fields. The panelist, all at the top of their respective fields, will also address the creative aspects, the science of sound; how sound is controlled and processed and how to develop and maintain professional standards and repeatability. ➡

Product Design Session 4 **Thursday, October 9**
4:45 pm – 6:45 pmRoom 406 AB

D/A AND A/D DESIGN FOR TODAY'S HIGH RESOLUTION AUDIO FORMATS

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

Panelists: *Andy McHarg*, dCS Ltd, UK
Bruno Putzeys, Hypex Electronics, Rotselaar, Belgium
Daniel Weiss, Weiss Engineering Ltd., Uster, Switzerland

With new interest in "ultra" DSD and DXD, A/D and D/A converters, to be fully inclusive, now must support a wide range of PCM formats (1Fs, 2Fs, 4Fs, 8Fs) as well as DSD (64 Fs, 128 Fs, and 256 Fs). Converter design and signal processing influence the performance obtained from these various formats and can also affect how they are perceived. A panel of AD/DA design engineers highly experienced in getting the best out of formats ranging from redbook to high resolution offer their take on these formats and on implementation for optimal quality. Along the way, many of the issues, assertions, less understood areas influencing sonic quality, and perhaps some of the canards that have developed surrounding these formats are discussed.

Historical Event H2

AUDIO ARCHITECTS OF THE NASHVILLE SOUND: HIGHLIGHTS FROM THE AES NASHVILLE LIFETIME ACHIEVEMENT AWARDS

Thursday, October 9, 5:00 pm – 6:30 pm
Room 408 B

Presenter: **Michael Janas**, Belmont University, Nashville, TN, USA

"Nashville would have never become a recording center without the engineers." - Country Music Hall of Fame and Musicians Hall of Fame guitarist Harold Bradley.

As often as people talk about how great records sound from many decades back, so little attention has been given to the engineers who captured the sound for all-time. If the sounds weren't captured just right by the engineer, it would be a completely different story to all who hear them." - Eddie Stubbs, WSM 650 disc jockey and Grand Ole Opry Announcer. Audio engineers played a crucial role in the evolution of the Nashville sound through the combination of their technical prowess, their own musicianship and their ability to understand the needs of the artist, the music, and bring to fore the elements of a hit song. In 2012 the Nashville Chapter of the AES began to honor these audio engineers with the AES Nashville Lifetime Achievement Awards. Presented annually in the Ford Theater at the Country Music Hall of Fame and Museum, these awards honor the individuals with a sustained record of outstanding achievement in music and sound in the Nashville recording industry. Many of the twenty-one individuals honored as of 2014 have recorded and mastered thousands of hit recordings from many different genres while others developed technology audio engineers and musicians rely upon today, such as inline console architecture and the distortion pedal. Highlights from the past three AES Nashville Lifetime Achievement Awards presentations will be featured in this program, including audio recordings, still images, video interviews and video footage from the presentations.

Session P6 **Friday, Oct. 10**
9:00 am – 12:00 noon **Room 308 AB**

SPATIAL AUDIO—PART 3

Chair: **Robert Schulein**, RBS Consulting, Schaumburg, IL, USA

9:00 am

P6-1 PHOnA: A Public Dataset of Measured Headphone Transfer Functions—*Braxton B. Boren,[1] Michele Geronazzo,[2] Piotr Majdak,[3] Edgar Choueiri[1]*
[1]Princeton University, Princeton, NJ, USA
[2]University of Padova, Padova, Italy
[3]Austrian Academy of Sciences, Vienna, Austria

A dataset of measured headphone transfer functions (HpTFs), the Princeton Headphone Open Archive (PHOnA), is presented. Extensive studies of HpTFs have been conducted for the past twenty years, each requiring a separate set of measurements, but this data has not yet been publicly shared. PHOnA aggregates HpTFs from different laboratories, including measurements for multiple different headphones, subjects, and repositionings of headphones for each subject. The dataset uses the spatially oriented format for acoustics (SOFA), and SOFA conventions are proposed for efficiently storing HpTFs. PHOnA is intended to provide a foundation for machine learning techniques applied to HpTF equalization. This shared data will allow optimization of equalization algorithms to provide more universal solutions to perceptually transparent headphone reproduction.
Convention Paper 9126

9:30 am

P6-2 Converting Two-Channel Stereo Signals to B-Format for Directional Audio Coding Reproduction—*Mikko-Ville Laitinen*, Aalto University, Espoo, Finland

A method for transforming two-channel stereo audio signals to B-format is proposed, which provides unaltered spatial qualities when the B-format signals are reproduced with directional audio coding (DirAC). The proposed method simulates anechoic B-format recordings of the stereo signals with two different virtual loudspeaker configurations, and the simulated B-format signals are combined according to time-frequency analysis of the stereo signals. The analysis is based on estimating the diffuseness of the generated virtual sound field and the coherence between the loudspeaker channels.
Convention Paper 9127

10:00 am

P6-3 Binaural Reproduction over Loudspeakers Using Low-Order Modeled HRTFs—
Kentaro Matsui,[1,2] Yasushige Nakayama,[1] Maho Sugaya,[2] Shuichi Adachi[2]
[1]NHK Science & Technology Research Laboratories, Setagaya, Tokyo, Japan
[2]Keio University, Yokohama-shi, Kanagawa, Japan

A method for binaural reproduction over loudspeakers using low-order modeled head-related transfer functions (HRTFs) is proposed. The low-order modeling consists of two steps: high-order model estimation using a prediction error method and subsequent model reduction based on asymptotic theory. Binaural processing over loudspeakers using the low-order modeled HRTFs is done in the time domain. In general, the directly derived controller for crosstalk cancellation is unstable, and so a method for approximating the unstable components in the controller as stable ones with processing delays is proposed. Results of computer simulation indicated that the designed controller worked well for producing equalization and crosstalk cancellation.
Convention Paper 9128

10:30 am

P6-4 Assessment of Ambisonic System Performance Using Binaural Measurements—*Eric M. Benjamin*,[1] *Aaron Helle*[2]
[1]Surround Research, Pacifica, CA, USA
[2]SRI International, Menlo Park, CA, USA

The phenomenon described by Solvang as spectral impairment in Ambisonic reproduction is examined. The timbre of reproduced sounds is arguably the most important aspect of an audio system. In multichannel systems audio is almost always reproduced through two or more loudspeakers simultaneously. The combination of those audio signals produces variable localization, but interference between them also causes comb filtering that then causes a reduction in output at high frequencies. The present work reports on measurements, including binaural measurements, of the spectral changes encountered in Ambisonic systems. In the case where a system has more loudspeakers than the minimum required the amount of interference is increased. What is the best choice for the use of an array designed for higher-order reproduction when used to reproduce lower-order program?
Convention Paper 9129

11:00 am

P6-5 The Design, Calibration, and Validation of a Binaural Recording and Playback System for Headphone and Two-Speaker 3D-Audio Reproduction—*Bob Schulein*,[1] *Dan Mapes-Riordan*[2]
[1]RBS Consultants, Schaumburg, IL, USA
[2]DMR Consultants, Evanston, IL, USA

The evolution of iOS, Android, Windows Mobile, and other operating systems has fueled a rapid growth in personal entertainment products and has revolutionized the way consumers receive, control, and listen to audio content. Headphones or earphones and two-speaker stereo have become the dominant means of listening. Multichannel /speaker audio systems in contrast are primarily a part of the motion picture and home theater experience and can create audio content with richer spatial content. This 3D or immersive audio experience is desired by consumers but is not a part of the typical listening experience. Binaural sound, reproduced by headphones or two speakers, using cross-talk cancellation techniques, has been shown to provide significant spatial audio benefits when properly implemented. This paper presents a detailed look as to how these technologies are being refined and applied today to create entertainment content with significantly improved spatial qualities.
Convention Paper 9130

11:30 am

P6-6 Analytical Interaural Time Difference Model for the Individualization of Arbitrary Head-Related Impulse Responses—*Ramona Bomhardt, Janina Fels*, RWTH Aachen University, Aachen, Germany

If dummy head or individual Head-Related Impulse Responses (HRIR) are used for binaural reproduction, either it can result in an incorrect perception of virtual sound sources or poses an enormous measurement effort. Therefore, in this paper a model is presented that helps to calculate the Interaural Time Difference by anthropometric head-data. By means of this Interaural Time Difference, the time of arrival of arbitrary HRIRs can be individualized. This model is compared with 13 individual measured HRIRs and the subjects' anthropometric head-data. The result of comparison leads to the conclusion that the model works well to individualize the Time of Arrival of an arbitrary HRIR.
Convention Paper 9131

Session P7 — Friday, Oct. 10
9:00 am – 11:30 am — Room 309

CINEMA SOUND, RECORDING AND PRODUCTION

Chair: **Scott Levine**, Skywalker Sound, San Francisco, CA, USA

9:00 am

P7-1 Particle Systems for Creating Highly Complex Sound Design Content—*Nuno Fonseca*, ESTG/CIIC, Polytechnic Institute of Leiria, Leiria, Portugal

Even with current audio technology, many sound design tasks present practical constraints in terms of layering sounds, creating sound variations, fragmenting sound, and ensuring space distribution especially when trying to handle highly complex scenarios with a significant number of audio sources. This paper presents the use of particles systems and virtual microphones, as a new approach to sound design, allowing the mixing of thousands or even millions of sound sources, without requiring laborious work and providing a true coherence between sound and space, with support for several surround formats, Ambisonics, Binaural, and even partial Dolby Atmos support. By controlling a particle system, instead of individual sound sources, a high number of sounds can be easily spread over a virtual space. By adding movement or random audio effects, even complex scenarios can be created.
Convention Paper 9132

9:30 am

P7-2 Stage Metaphor Mixing on a Multi-Touch Tablet Device—*Steven Gelineck, Dannie Korsgaard*, Aalborg University, Copenhagen, Denmark

This paper presents a tablet based interface (the Music Mixing Surface) for supporting a more natural user experience while mixing music. It focuses on the so-called stage metaphor control scheme where audio channels are represented by virtual widgets on a virtual stage. Through previous research the interface has been developed iteratively with several evaluation sessions with professional users on different platforms. The iteration presented here has been developed especially for the mobile tablet platform and explores this format for music mixing both in a professional and casual setting. The paper first discusses various contexts in which the tablet platform might be optimal for music mixing. It then describes the overall design of the mixing interface (especially focused on the stage metaphor), after which the iOS implementation is briefly described. Finally, the interface is evaluated in a qualitative user study comparing it to two alternative existing tablet solutions. Results are presented and discussed focusing on how the evaluated interfaces invite four different forms of exploration of the mix and on what consequences this has in a mobile mixing context.
Convention Paper 9133

10:00 am

P7-3 The Duplex Panner: Comparative Testing and Applications of an Enhanced Stereo Panning Technique for Headphone-Reproduced Commercial Music—*Samuel Nacach*, New York University, New York, NY, USA

As a result of new technology advances consumers primarily interact with recorded music on-the-go through headphones. Yet, music is primarily mixed using stereo loudspeaker systems consisting of crosstalk signals, which are absent in headphone reproduction. Consequently, the audio engineer's intended sound image collapses with headphones. To solve this, the work presented in this paper examines existing 3D audio techniques—primarily Binaural Audio and Ambiophonics—and enhances them to develop a novel and improved mixing technique, the Duplex Panner, for headphone-reproduced commercial music. Through subjective experiments designed for two groups, the Duplex Panner is compared to conventional Stereo panning to determine what the advantages are, if any.
Convention Paper 9134

10:30 am

P7-4 The Role of Acoustic Condition on High Frequency Preference—*Richard King*,[1,2] *Brett Leonard*,[1,2,3] *Stuart Bremner*,[1,2] *Grzegorz Sikora*[4]
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
[3]University of Nebraska at Omaha, Omaha, NE, USA
[4]Bang & Olufsen Deutschland GmbH, Pullach, Germany

Subjective preference for high frequency content in music program has shown a wide variance in baseline testing involving expert listeners. The same well-trained subjects are retested for consistency in setting a high frequency shelf equalizer to a preferred level under varying acoustic conditions. Double-blind testing indicates that lateral energy significantly influences high frequency preference. Furthermore, subject polling indicates that blind preference of acoustic condition is inversely related to optimal consistency when performing high frequency equalization tasks.
Convention Paper 9135

11:00 am

P7-5 Listener Preferences for Analog and Digital Summing Based on Music Genre—*Eric Tarr, Jane Howard, Benjamin Stager*, Belmont University, Nashville, TN, USA

The summation of multiple audio signals can be accomplished using digital or analog technologies. Digital summing and analog summing are not identical processes and, therefore, produce different results. In this study digital summing and analog summing were performed separately on the audio signals of three different recordings of music. These recordings represented three genres of music: classical, pop/country, and heavy rock. Twenty-one listeners participated in a preference test comparing digital summing to analog summing. Results indicated that listeners preferred one type of summing to the other; this preference was dependent on the genre of music.
Convention Paper 9136

Workshop 2 — **Friday, October 10**
9:00 am – 10:30 am — **Room 409 AB**

AUDIO LEGAL: INTELLECTUAL PROPERTY AND AUDIO

Chair: **Andrea Yankovsky**, Kilpatrick, Townsend & Stockton LLP, New York, NY, USA

Panelists: *Scott Hull*, Masterdisk
Danny "Sage" McKinney, Requisite Audio
James Trigg, Kilpatrick, Townsend & Stockton LLP, New York, NY, USA
Jason Vogel, Kilpatrick, Townsend & Stockton LLP, New York, NY, USA
Erik Zobler

Working in a creative industry, our livelihoods are continually influenced by IP law. Protecting our creative assets is important, and yet the issues we encounter are becoming increasingly difficult to navigate without a solid knowledge base and expert input. Focusing on current developments and practical applications, this panel of legal and music industry experts will explore intellectual property issues and other legal matters affecting audio engineers, producers, studios, equipment makers.

Tutorial 11 — **Friday, October 10**
9:00 am – 10:00 am — **Room 404 AB**

ALL ABOUT: PHASE VS. POLARITY

Presenter: **Alex Case**, University of Massachusetts Lowell, Lowell, MA, USA

The word "phase" is misused almost as often as it is properly applied. Meantime, its close cousin "polarity" is consistently neglected. Worst of all, the two terms are frequently, mistakenly, interchanged. Phase and polarity are essential but distinct concepts that, when fully understood, help the engineer get better sounds, quicker.

Broadcast/Streaming Media Session 4
Friday, October 10 9:00 am – 11:00 am
Room 408 A

AUDIO ISSUES FOR 4K AND 8K TELEVISION

Chair: **Fred Willard**, Univision, Washington, DC, USA

Panelists: *Robert Bleidt*, Fraunhoffer USA, San Jose, CA, USA
Tim Carroll, Telos Alliance, Lancaster, PA, USA
Thomas Lund, TC Electronic A/S, Risskov, Denmark
David McIntyre, DTS Inc., Calabasas, CA, USA
Skip Pizzi, NAB, Washington DC, USA
Jeff Riedmiller, Dolby Laboratories, San Francisco, CA USA

The session will touch on the tools and practical experience of realizing the promise of the ultra resolution visual-aural experience and the standardization considerations of bonding the two in a single production-to-consumer format. 2014 has so far seen practical production, encoding, and transmission of 4k and 8k and its equivalent quality audio complement with the Sochi Olympics and the World Cup games in Brazil. Implementation of compression formats, auto scalability, and backwards compatibility are topics of discussion. Re-imagined production techniques and workflows; transmission, interface, and playback standards are still in development. The extended localization, multi-directivity, three dimensional acoustic space generation, and increased optimal listening zones for audio for 4K and 8K will likely provoke end user acceptance with creation of a unified reproduction of reality immersion experience in a single format.

Live Sound Seminar 3 Friday, October 10
9:00 am – 11:00 am Room 406 AB

USING PLUG-INS FOR LIVE SOUND

Chair: **Mick Olesh**, Waves

Panelists: *Dave Aron*, FOH with Snoop Dogg
Fabrizio Piazzini, FOH with Amy MacDonald
Ken "Pooch" Van Druten, FOH with Linkin Park

Plug-in enabled digital consoles are now standard in many live sound applications. New loudspeaker designs that are able to deliver higher fidelity than ever have raised the bar for listeners who expect album-quality sound experiences at a concert. Whether this is the artists on stage with in-ear monitors or the audience, they all expect an exceptional mix every day. Using plug-ins vs. outboard gear means lower transit costs, maintenance, and less physical space at venue or in storage. Today's analog emulations also bring back some of the qualities lost in the purely digital age providing as much harmonic distortion and noise as you want. In this workshop expert mixers will demonstrate which plugins they use in live situations and explain how a few of the plugins help them get a better sound, more control, easier and faster set up time.

Game Audio Session 4 Friday, October 10
9:00 am – 11:00 am Room 408 B

NEXT GEN GAME AUDIO EDUCATION

Chair: **Steve Horowitz**, Game Audio Institute, San Francisco, CA, USA; Nickelodeon Digital

Panelists: *Dale Everingham*, Video Symphony Director of Audio Program, Burbank, CA, USA
Scott Looney, Academy of Art University, San Francisco, CA, USA
Leonard J. Paul, School of Video Game Audio, Vancouver, Canada
Stephan Schütze, Sound Librarian, Melbourne, Australia
Michael Sweet, Berklee College of Music, Boston, MA, USA

Game Audio education programs are starting to take root and sprout up all over the world. Game audio education is becoming a hot topic. What are some of the latest training programs out here? What are the pros and cons of a degree program versus just getting out there on my own? I am already a teacher, how can I start a game audio program at my current school? Good questions! This panel brings together entrepreneurs from some of the top private instructional institutions to discuss the latest and greatest educational models in audio for interactive media. Attendees will get a fantastic overview of what is being offered inside and outside of the traditional education system. This is a must for students and teachers alike, who are trying to navigate the waters and steer a path toward programs that are right for them in the shifting tides of audio for games and interactive media.

Product Design Session 5 Friday, October 10
9:00 am – 10:30 am Room 402 AB

HOW TO CHOOSE AMPLIFIERS AND POWER SUPPLIES

Presenter: **Gordon Wanlass**, PowerPhysics

How to select a power supply or amplifier for OEM speaker design using basic measurements and simple calculations. There is an ever increasing supply of off the shelf audio electronics. The aim of this tutorial is to walk you through how to evaluate them and select the right power supply or amplifier for your powered speaker design.

Friday, October 10 9:00 am Room 405
Technical Committee Meeting on Network Audio Signals

Recording and Production Session 2
Friday, October 10 10:00 am – 11:30 am
Room 403 AB

RAW TRACKS: DAVID BOWIE—A MASTER CLASS

Moderator: **Mark Rubel**, The Blackbird Academy, Nashville, TN, USA; Pogo Studio, Nashville, TN, USA

Panelist: **Ken Scott**

Legendary recording engineer and producer Ken Scott will discuss, analyze, and deconstruct a classic David Bowie recording track by track, in the inaugural Raw Tracks series at the AES 137th International Convention in Los Angeles. Ken Scott has had a hand in many of the most important recordings of the 20th Century: The Beatles, Supertramp, Elton John, Pink Floyd, The Mahavishnu Orchestra, Lou Reed, America, Procol Harum, The Jeff Beck Group, Devo, and many others.

Special Event
DIGITAL ENTERTAINMENT GROUP PRESENTS HIGH RESOLUTION AUDIO SESSIONS
Friday, October 10, 10:00 am – 4:00 pm
Room 304 AB

10:00 am – 10:50 am

Hi-Res Audio Devices for Every Lifestyle: Learn more details about the growing number of hi-res compatible devices available today from some of the biggest names in hi-res devices, including Astell & Kern, dCS, DTS, Kimber Kable, Meridian, Mytek, and Sony. Subjects will include how to demonstrate hi-res audio at retail; the latest options for enjoying hi-res music on-the-go; and how to educate and engage young music enthusiasts. Moderated by *Marc Finer*, the panel includes *Owen Kwon, John Quick, Fred Maher, Ray Kimber, Bob Stuart, Michal Jurewicz,* and *Aaron Levine.*

11:30 am – 12:20 pm

The New Business of Hi-Res Music: Get an inside look at the opportunities and challenges associated with hi-res music from *Mark Piibe* at Sony Music, *Howie Singer* at Warner Music, and *Jim Belcher* at Universal Music. Topics will include licensing hi-res files; the latest distribution partners; ingesting and archiving digital assets; new subscription models; and the best ways to promote hi-res music.

1:00 pm – 1:50 pm

Hi-Res Audio Production Workshop: [co-sponsored by the Recording Academy P&E Wing] Join top producers and engineers as they discuss the music creation process and best practices when recording, mixing and mastering in high resolution. The panel moderated by *Leslie Ann Jones* features *Chuck Ainlay, John Burk, Bob Clearmountain*, and *Ryan Ulyate* who will review the key aspects of various audio formats in context with their latest music projects.

3:00 pm – 3:50 pm

High Resolution Audio—Super Session: Meet and mix with some of the brightest minds in the business, including *Bruce Botnick, George Massenburg*, *Bill Schnee*, and *Andrew Scheps*, as they explore a number of the most challenging issues facing the recording industry today concerning the adoption of high resolution audio. Don't miss this rare opportunity to hear from these opinion makers!

Friday, October 10 10:00 am Room 405
Technical Committee Meeting on Human Factors in Audio Systems

Session EB2 Friday, Oct. 10
10:30 am – 12:00 noon S-Foyer 1

POSTER SESSION 2

10:30 am

EB2-1 Android Framework Implementation of 3D Audio with Range Control—*Phyo Ko Ko, Kaushik Sunder, Woon-Seng Gan*, Nanyang Technological University, Singapore, Singapore

Due to the rapid improvement in processing power of mobile devices, real-time 3D audio rendering is becoming a reality on these devices. Using 3D audio rendering, many interesting applications, such as teleconferencing and immersive gaming can be developed. In this paper an Android framework is developed for rendering 3D audio in real-time with additional range control. Range-dependent head related transfer functions (HRTFs) are used in order to render the spatial audio. Distance dependent HRTFs are extremely tedious to be measured since these measurements have to be carried out for several distances in the near-field. In this work, the range-dependent HRTFs used by the Android framework are experimentally measured in the horizontal plane and made available online for the researchers to use. This 3D audio framework will serve as a useful platform to deliver new audio processing applications, such as teleconferencing over headphones, personalized hearing in gaming and virtual reality etc.
Engineering Brief 161

10:30 am

EB2-2 Flexible Audio Rendering for Arbitrary Input and Output Layouts—*Hyunjoo Chung, Sang Bae Chon, Sunmin Kim*, Samsung Electronics Co. Ltd., Suwon, Korea

This engineering brief introduces a rendering method compatible with various input audio formats and output reproduction layouts. By using audio scene analysis, *N*-channel audio input signals are converted to channel-independent spatial parameters then sound fields are rendered based on the *M*-channel output loudspeaker layout with maintaining spatial information of original audio formats. Therefore, proposed method enabled *N*-to-*M* flexible audio rendering with immersive sound perception.
Engineering Brief 163

10:30 am

EB2-3 A Touchpad-Based Method for Inducing Attentional Tunneling—*Durand R. Begault,*[1] *Bonny R. Christopher,*[1] *Charlotte Zeamer,*[1] *Mark R. Anderson,*[1] *Kirstianna Burns*[2]
[1]NASA Ames Research Center, Moffett Field, CA, USA
[2]City College of San Francisco, San Francisco, CA, USA

Attentional tunneling is a recognized problem for aviation safety in the flight deck. A prototype system (touchpad and associated application and experimental software) was developed and evaluated for its success in inducing attentional tunneling in a reliable and predictable manner in

training and experimental contexts. An experiment with ten participants using the system examined baseline performance for visual memory of a color or number sequence, simultaneous with performing a competing auditory detection task. Spatial auditory separation of the auditory stimuli was also evaluated. Data are provided for various aspects of touchpad entry (accuracy, speed) as well as hit and false alarm rates for the auditory task. The results will help determine means of inducing attentional tunneling in more complex flight simulator experiments and for developing an inexpensive prototype for pilots to measure cognitive fixation and develop mitigation strategies.
Engineering Brief 164

10:30 am

EB2-4 The Open Multitrack Testbed—*Brecht De Man, Mariano Mora-Mcginity, György Fazekas, Joshua D. Reiss*, Queen Mary University of London, London, UK

We introduce the Open Multitrack Testbed, an online repository of multitrack audio, mixes or processed versions thereof and corresponding mix settings or process parameters such as DAW files. Multitrack audio is a much sought after resource for audio researchers, students, and content producers, and while some online resources exist few are large and reusable and none allow querying audio fulfilling specific criteria. The test bed we present contains a semantic database of metadata corresponding with the songs and individual tracks, enabling users to retrieve all pop songs featuring an accordion, or all tracks recorded in reverberant spaces. The open character is made possible by requiring the contributions, mainly from educational institutions and individuals, to have a Creative Commons license.
Engineering Brief 165

10:30 am

EB2-5 Maintenance Considerations in Higher Education Facilities—*Daniel Gonko*, Western Carolina University, Cullowhee, NC, USA

Running and maintaining a recording studio in a higher education environment poses various benefits and challenges that are not often encountered in commercial facilities. This presentation will examine the pros and cons of working in such a facility, including availability, funding considerations, and training needs. The Center for Applied Technology and Commercial and Electronic Music degree program at Western Carolina University will be utilized as a case study.
Engineering Brief 166

Tutorial 12 Friday, October 10
10:30 am – 12:30 pm Room 404 AB

ACOUSTICS—SORTING IT OUT AND GETTING IT RIGHT

Presenter: **Tony Hoover**

This tutorial on architectural acoustics covers three distinct issues: sound isolation (airborne and structure-borne), HVAC noise control, and surface treatments (absorption, reflection, and diffusion). The format has been widely used, including for over 25 years at Berklee College of Music. The objective is to provide the foundation for optimized audio design and recording decisions, and the confidence to better navigate the oceans of information and propaganda about "acoustical" products and practices.

This session is presented in association with the AES Technical Committee on Acoustics and Sound Reinforcement

Workshop 3 Friday, October 10
10:45 am – 12:15 pm Room 409 AB

MAKING THE TRANSITION FROM STUDENT TO PROFESSIONAL

Chair: **Jonathan Wyner**, Berklee College of Music, Boston, MA, USA; M Works Mastering

Panelists: *Adam Ayan*, Gateway Mastering Studios, Portland, ME USA
Rob Jaczko, Berklee College of Music, Boston, MA, USA
John Krivit, New England Institute of Art, Brookline, MA, USA; Emerson College, Boston, MA, USA
Jay LeBoeuf, Real Industry, San Francisco, CA, USA; Stanford University, San Francisco, CA, USA*Henry Moyerman*

There are many paths that lead to a fulfilling career as an audio engineer. This workshop will include tales from those who have successfully navigated a version of this path and suggestions of recommended approaches along with tools and skills worth acquiring to make the young engineer an attractive hire.

Product Design Session 6 Friday, October 10
10:45 am – 12:15 pm Room 402 AB

HOW TO LEVERAGE DSP FOR PRODUCT GREATNESS

Presenter: **Denis Labrecque**, Analog Devices, San Jose, CA, USA

How to deal with the “more” law. While many are familiar with Moore's Law of computing, those of us in Pro Audio are also painfully aware of the “more” law—designing todays' DSP based products require “more”—more performance, more precision, more complex processes, more channels, more power efficiency, more connectivity etc. This tutorial will explore the design considerations, decisions and trade-offs necessary to develop modern audio systems with advanced DSP capabilities. Overall design methodology discussed will include: fixed vs. floating point processing, internal vs. external memory, how requirements change for low power applications, Ethernet/USB connectivity, and peripheral (I/O) selection. In addition, this tutorial will touch on software tools and modules available for the creation of DSP based products. Specific examples will include system block diagrams of SHARC, Blackfin, and SigmaDSP processors.

Live Sound Seminar 4 Friday, October 10
11:00 am – 1:00 pm Room 406 AB

MULTICELLULAR LOUDSPEAKERS—

NOT YOUR FATHER'S LINE ARRAY

Chair: **Jim Risgin**, OSA International, Inc., Chelsea, MI, USA

Panelists: *Bernie Broderick*, System Support/Trainer EAW Tom Petty Tour 2014
Martyn "Ferrit" Rowe, System Tech for Steely Dan 2014; Director Engineering Services, OSA

In this presentation some of the current experts in the field of deploying large line arrays for live events will discuss their techniques, successes, and challenges in deploying new state-of-the-art, digitally-steered line array systems such as Martin MLA and EAW Anya. A panel of industry experts will reveal their real-world experiences with these hi-tech loudspeakers. What are their experienced with driving the software that is required for using these systems? How do these system perform in the field, how are they different from conventional line arrays? The panel will also discuss how these platforms are evolving and what they see happening with this format in the future.

Game Audio Session 5 Friday, October 10
11:00 am – 1:00 pm Room 408 B

AUDIO MIDDLEWARE FOR THE NEXT GENERATION

Presenters: **Scott Looney**, Academy of Art University, San Francisco, CA, USA
Steve Horowitz, Game Audio Institute, San Francisco, CA, USA; Nickelodeon Digital

Until quite recently, there was very little audio middleware capabilities available in popular game engines without the need for scripting. Now, there are at least four hearty middleware solutions to choose from. In this presentation and demonstration, we will bring you up to speed on the latest and greatest audio middleware solutions. We will get an overview of FMOD, WWISE, FABRIC, and MASTER AUDIO. We will compare and contrast their different features, strengths and weaknesses, as well as discuss elements of audio workflow and how sound designers can go about deploying these full featured solutions in a number of different game engines.

Sound for Picture 2 Friday, October 10
11:00 am – 1:00 pm Room 306 AB

MUSIC PRODUCTION FOR FILM—A MASTER CLASS

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

Moderator: *Tom Salta*, Persist Music, Norwalk, CT, USA

Panelists: *Chris Boardman*
Simon Franglen, Class1 Media, Los Angeles, CA, USA; London
Laura Karpman, Art Farm West, Playa Del Rey, CA, USA
Trevor Morris
Steven Saltzman, Music Visions, Los Angeles, CA USA

Film soundtracks contain three elements: dialog, music, and sound effects. The creation of a music soundtrack is far more complex than previously, now encompassing "temp music" for preview screenings, synthesizer-enhanced orchestra tracks, and other special techniques. This Master Class with one of Hollywood's leading professionals puts the process under the microscope.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Friday, October 10 11:00 am Room 405
Technical Committee Meeting on Loudspeakers and Headphones

Friday, October 10 11:00 am Room 407
Standards Committee Meeting on Audio Connectors

Live Sound Expo LSE1 LSE STAGE

NETWORKS AND IT—THE BASICS
Friday, October 10, 11:00 am – 11:50 am

Presenter: **Landon Gentry**, Director of Global Support Services, Audinate

Live sound is increasingly embracing networking for audio distribution and system control. You don't have to be an IT professional to set up such networks, but you do need the core knowledge covered here.

Project Studio Expo PSE1 PSE STAGE
VOCAL RECORDING IN THE PROJECT STUDIO
Friday, October 10, 11:15 am – 12:15 pm

Presenter: **Mike Senoir**, Sound on Sound

Recording vocals is easy, right? Just put up a large-diaphragm condenser mic (with popshield) about six inches away from your mouth, sling up a couple of old quilts to soak up the room sound, grab a pair of headphones for monitoring, and you're away! Sadly, however, this approach frequently achieves poor or inappropriate results in real-world project studios. To find out why, join *Sound On Sound* magazine's "Session Notes" and "Mix Rescue" columnist Mike Senior who'll be using live demonstrations and audio examples to show you how to get the best out of the stereotypical project-studio vocal tracking setup—as well as exploring a variety of common situations where you're actually better off abandoning it completely.

Broadcast/Streaming Media Session 5
Friday, October 10 11:15 am – 12:45 pm
Room 408 A

THE STREAMING EXPERIENCE

Chair: **Dave Wilson**, CEA, Arlington, VA, USA

Panelists: *Don Backus*, Broadcast Electronics, Quincy, IL USA; Farmington Hills, MI USA
Frank Foti, Telos, New York, NY, USA
Philippe Generali, RCS, White Plains, NY USA
*Greg Ogonowsk*i, Orban, San Leandro, CA, USA
Geir Skaaden, DTS, Inc.

Pandora, iHeartRadio, Radio.com, SiriusXM, and thousands of others use streaming technology as the foundation for their businesses. This session will cover the latest advancements in streaming technology and help content creators and distributors learn how best to use this tool to reach, capture, and retain an audience. The expert panelists have extensive experience with codecs, audio processing, and audio distribution. They will discuss the latest advancements in bringing quality audio to the Internet audience.

Tutorial 13 **Friday, October 10**
11:45 am – 12:45 pm **Room 309**

LET YOUR EYES HELP YOUR EARS—TECHNIQUES AND NEW CHALLENGES IN AUDIO METERING

Presenter: **Paul Tapper**

Why use audio analysis or metering at all? Why loudness metering is relevant to engineers working in the music industry and the implications for music mastering practice. If you think that your music might ever get played out on TV, radio, or a music streaming service, you need to have an awareness of loudness normalization. The importance of true-peak metering for music to avoid distortion and fizzing caused by codec conversions. The importance of mono compatibility for music that might ever be listened to on iPod docks, flat-screen TVs, DAB radios, or club PAs. The difficulty of auditioning this and pitfalls to avoid when trying to meter it will be covered.

Special Event
LUNCHTIME KEYNOTE: NEIL PORTNOW
Friday, October 10, 12:00 noon – 1:00 pm
Room 403 AB

Presenter: **Neil Portnow**, President/CEO The Recording Academy, The GRAMMY Foundation, and MusiCares

Neil Portnow, President/CEO of The Recording Academy® (internationally known for the GRAMMY Awards®) in his keynote address, will discuss the challenges and opportunities currently facing recording professionals, as well as targeted advocacy initiatives The Academy is developing to address some of these concerns.

Friday, October 10 **12:00 noon** **Room 405**
Technical Committee Meeting on Perception and Subjective Evlauation of Audio Signals

Live Sound Expo LSE2 **LSE STAGE**
LOUDSPEAKER SET-UP AND CONFIGURATION
Friday, October 10, 12:00 noon – 12:50 pm

Moderator: **Mark Frink**

Presenters: *Scott Sugden*
Vic Wagner

FOH and system engineers share their strategies for successful touring system deployment in a cluster of case-studies.

Network Audio Session 4 **Friday, October 10**
12:15 pm – 1:15 pm **Room 308 AB**

THE BENEFITS OF ETHERNET AVB

Presenter: **Rick Kreifeldt,** Harman International

As, what have been termed, third generation media networking technologies and protocols have emerged over the past few years the benefits have been poorly understood by integrators and systems designers. This session will focus on the practical benefits of Ethernet AVB. Where it's benefits can best be exploited will be explored as regards systems scale, environment, performance requirements such as latency and ease of deployment.

Student Event/Career Development
SPARS SPEED COUNSELING WITH EXPERTS —MENTORING ANSWERS FOR YOUR CAREER
Friday, October 10, 12:30 pm – 2:00 pm
Room 409 AB

Mentors: *Tom Salta*
Dren McDonald
Chanel Summers
Danny Leake
Juan R Garza
Craig Doubet
TW Blackmon
Lorita de la Cerna
Eric Johnson
Jeri Palumbo
Rick Senechal
Geoff Gray
David Rideau
Erik Zobler
Chuck Zwicky
Sylvia Massy
Mark Rubel
Anthony Schultz
Pat McMakin
Lisa Chamblee
David Glasser
Bruce Maddocks
Andrew Mendelson

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 and G.A.N.G., 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.

Project Studio Expo PSE2 **PSE STAGE**
IT WON'T SOUND RIGHT IF YOU DON'T HEAR IT RIGHT: STUDIO ACOUSTICS, MONITORING, AND CRITICAL LISTENING
Friday, October 10, 12:30 pm – 1:30 pm

Presenter: **Hugh Robjohns**

The monitoring environment acoustics and the monitoring chain are critical to the music production process. 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 the options for monitoring controllers and loudspeakers, optimizing control room acoustics, and honing critical listening skills

This event replaces original PSE2 and PSE3 which have been canceled.

Friday, October 10 12:30 pm Room 407
Standards Committee Meeting on Microphone Measurement and Characterization

Special Event
DTV AUDIO GROUP/AES FORUM: THE IMPLICATIONS OF STREAMED CONTENT DELIVERY ON THE EVOLUTION OF TELEVISION AUDIO SERVICES
Friday, October 10, 1:00 pm – 6:00 pm
Room 404 AB

Moderator: **Roger Charlesworth,** , Executive Director, DTV Audio Group

Program Advisors, Panelists, and Contributors:
Tim Carroll, Chief Technology Officer, Telos Alliance, President and Founder, Liner Acoustic
David Colantuoni, Senior Director, Broadcast, Storage, and Editor Product Development, Avid
Kevin Collier, Director of Engineering, Post Production, Warner Bros Studio Facilities
Craig Cuttner, Senior Vice President, Technology Development & Standards, HBO
Thomas Edwards, Vice President Engineering and Development, Fox Networks
Michael Englehaupt, Chief Technology Officer, KQED Television
Tino Fibaek, Chief Technical Officer, Fairlight AU
Will Files, Sound Designer, Re-Recording Mixer, Skywalker Sound
Mark Francisco, Fellow, Premises Technology, Office of the CTO, Comcast
Richard Friedel, Executive Vice President and General Manager, Fox Networks Engineering and Operations
Steve Harvey, West Coast Editor, Pro Sound News
Chris Jenkins, Sound Designer and Re-RecordingMixer
John Kellogg, Senior Director, Corporate Strategy and Development, DTS
Karl Malone, Director of Sound Design at NBC Sports and Olympics
Steve Morris, Director of Engineering, Skywalker Sound
Michael Novitch, Chief Engineer, Technicolor Sound Services
Nathan Oishi, Chief Engineer, Sony Pictures Digital
Sean Richardson, Executive Director and Principal Audio Engineer, Starz Entertainment
Jeffery Riedmiller, Senior Director, Sound Group, Office of the CTO, Dolby Laboratories
Tom Sahara, Turner Sports Vice President, Operations and Technology, Turner Sports
Steve Silva, Vice President of Technology and Strategy, Fox Networks
Jim Starzynski, Director and Principal Audio Engineer, NBCUniversal
Jeff Willis, Coordinating Technical Manager at ESPN

Content delivery is converging on a streamed model whether for mobile, over the public internet, within the walled garden of the MVPD, or over next generation broadcast services. Virtualized delivery infrastructure for streamed content allows new services to be deployed with very short lead times. The rapidity of this transition to streaming has significantly accelerated the time frame for adoption of advanced object-based audio services offering spatially immersive sound, enhanced personalization, greater bandwidth efficiency and improved audio quality. It is likely we will begin seeing these new features in our homes and mobile devices in months rather than years. This forum explores some of the tools and workflow approaches required to manage and exploit the capabilities of next-generation audio standards. It takes a look at the convergence around streamed content delivery and transition to IP distribution and contribution that makes this rapid deployment possible.

"*The explosive growth in streamed content delivery over mobile and fixed devices, has re-written the playbook for television media distribution. Streaming delivery models are also inexorably finding their way into existing cable services and are being written into next-generation advanced television standards. The migration from traditional broadcasting to a IP stream-based model greatly simplifies implementation, and combined with tablet-empowered UI, frees distributors to pursue a range of formats and encoding solutions with sophisticated interactive and object-oriented audio services*"—Roger Charlesworth, Executive Director, DTV Audio Group

Discussion topics will include:

This Is Not Your Father's MVPD: A look into how the transition to IP infrastructure and streamed content is transforming cable and how this facilitate advanced audio codecs

So Long SDI and MADI: IP infrastructure for audio and video contribution within the broadcast plant and in the field

Interactivity, Objects, and Spatially Immersive Audio: Mixers and technologist on the migration of object audio from cinema to the small screen and mobile

Object Audio Toolbox: What are the essential monitoring and authoring tools required for object audio production?

It's All in Your Head: The state of headphone virtualization for immersive audio formats

The DTV Audio Group Forum at AES is produced in association with the Sports Video Group and is sponsored by: Calrec, Dolby Laboratories, DTS, Lawo, Linear Acoustic

Friday, October 10 1:00 pm Room 405
Technical Committee Meeting on Recording Technology and Practices

Live Sound Expo LSE3 LSE STAGE
RF SYSTEMS—PRACTICAL CONSIDERATIONS
Friday, October 10, 1:00 pm – 1:50 pm

Presenter: **Ike Zimbel**, Zimbel Audio Productions

Wireless microphones and monitors are now a standard part of live performance, though not without challenges as spectrum constraints take their toll. This RF Systems session explores frequency coordination and onsite spectrum analysis. Antenna design and placement and managing RF interference (including musician systems) are examined.

Project Studio Expo PSE4 PSE STAGE
THE FIVE MOST COMMON PROJECT STUDIO RECORDING MISTAKES
Friday, October 10, 1:45 pm – 2:45 pm

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

Tearing your hair out at mixdown? Then you've probably fallen into one of the classic project-studio traps during tracking. Learn what they are—and, more importantly, how to avoid them—in this down-to-earth workshop with *Sound On Sound* magazine's "Session Notes" and "Mix Rescue" columnist Mike Senior. The seminar will be specifically tailored to those working on a budget and will feature insider tips on equipment choice, mic technique, and session psychology, with plenty of supporting audio examples so you can judge the results with your own ears.

Session P8 **Friday, Oct. 10**
2:00 pm – 5:00 pm **Room 308 AB**

PERCEPTION—PART 1

Chair: **Dan Mapes-Riordan**, Etymotic Research, Evanston, IL, USA

2:00 pm

P8-1 Effect of Phase on the Perceived Level of Bass—*Mikko-Ville Laitinen, Kai Jussila, Ville Pulkki*, Aalto University, Espoo, Finland

The perceived level of bass is typically considered to be related to the level of the magnitude spectrum at the corresponding frequencies. However, recently it has been found that, in the case of harmonic complex signals, also the phase spectrum can affect it. This paper studies this effect further using formal listening tests. It is found out that the phase spectrum that produces the perception of the loudest bass depends on the individual. Furthermore, the loudness of the bass appears to be affected by the phase characteristics of the tone in a relatively wide band.
Convention Paper 9137

2:30 pm

P8-2 Auditory Compensation for Spectral Coloration—*Cleopatra Pike, Russell Mason, Tim Brookes*, University of Surrey, Guildford, Surrey, UK

The "spectral compensation effect" (Watkins, 1991) describes a decrease in perceptual sensitivity to spectral modifications caused by the transmission channel (e.g., loudspeakers, listening rooms). Few studies have examined this effect: its extent and perceptual mechanisms are not confirmed. The extent to which compensation affects the perception of sounds colored by loudspeakers and other channels should be determined. This compensation has been mainly studied with speech. Evidence suggests that speech engages special perceptual mechanisms, so compensation might not occur with non-speech sounds. The current study provides evidence of compensation for spectrum in non-speech tests: channel coloration was reduced by approximately 20%.
Convention Paper 9138

3:00 pm

P8-3 The Importance of Onset Features in Listeners' Perception of Vocal Modes in Singing—*Eddy B. Brixen,*[1] *Cathrine Sadolin,*[2] *Henrik Kjelin*[2]
[1]EBB-consult, Smorum, Denmark
[2]Complete Vocal Institute, Copenhagen, Denmark

The Complete Vocal Technique defines four vocal modes: Neutral, Curbing, Overdrive, and Edge. This paper reports the result of a listening test involving 59 subjects. The goal has been to find the importance of onset and decay features when identifying the vocal modes. The conclusion is that the onset only to a minor degree is responsible for the aural detection of vocal modes.
Convention Paper 9139

3:30 pm

P8-4 The Influence of Up- and Down-mixes on the Overall Listening Experience—*Michael Schoeffler, Alexander Adami, Jürgen Herre*, International Audio Laboratories Erlangen, Erlangen, Germany; Fraunhofer IIS, Erlangen, Germany

Former studies have shown that up- and down-mix algorithms have a significant effect on ratings of audio quality. The question arises whether this significant effect is also verifiable when it comes to rating the overall listening experience of music. When listeners rate the overall listening experience, they are allowed to take everything into account that is important to them for enjoying a listening experience. An experiment was conducted where 25 participants rated the overall listening experience while listening to music that was artistically mixed and up- and down-mixed by six algorithms. The results show that there are no significant differences between the artistic mixes and the up- and down-mix algorithms except for two mixing algorithms which served as "lower anchors" and had a significant negative effect on the ratings.
Convention Paper 9140

4:00 pm

P8-5 Measures of Microdynamics—*Esben Skovenborg*, TC Electronic, Risskov, Denmark

Overall loudness variations such as the distance between soft and loud scenes of a movie are known as macrodynamics and can be quantified with the Loudness Range measure. Microdynamics, in contrast, concern variations on a (much) finer time-scale. In this study six types of objective measures—some based on loudness level, some based on peak-to-average ratio—were evaluated against perceived microdynamics. A novel measure LDR, based on the maximum difference between a "fast" and a "slow" loudness level, had the strongest perceptual correlation. Peak-to-average ratio (or crest factor) type of measures had little or no correlation. The ratings of perceived microdynamics were obtained in a listening experiment, with stimuli consisting of music and speech of different dynamical properties.
Convention Paper 9141

4:30 pm

P8-6 Real-Time Infant Cry Detection in Diverse Environments: A Novel Approach—*Anant* ➥

Baijal, Jinsung Kim, Jae-hoon Jeong, Inwoo Hwang, JungEun Park, Byeong-Seob Ko, Samsung Electronics Co. Ltd., Suwon, Korea

We present a novel approach to detect infant cry in actual outdoor and indoor settings. Using computationally inexpensive features like Mel Frequency Cepstral Coefficients (MFCCs) and timbre-related features, the proposed algorithm yields very high recall rates for detecting infant cry in challenging settings such as café, street, playground, office, and home environments, even when Signal to Noise Ratio (SNR) is as low as 6 dB, while maintaining high precision. The results indicate that our approach is highly accurate, robust and, works in real-time.
Convention Paper 9142

Session P9 Friday, Oct. 10
2:00 pm – 4:30 pm Room 309

TRANSDUCERS—PART 2

Chair: **Mario DiCola**, Audio Labs Systems

2:00 pm

P9-1 The Implementation of MEMS Microphones for Urban Sound Sensing—*Charlie Mydlarz, Samuel Nacach, Agnieszka Roginska, Tae Hong Park, Eric Rosenthal, Michelle Temple*, New York University, New York, NY, USA

The urban sound environment of New York City (NYC) is notoriously loud and dynamic. The current project aims to deploy a large number of remote sensing devices (RSDs) throughout the city, to accurately monitor and ultimately understand this environment. To achieve this goal, a process of long-term and continual acoustic measurement is required, due to the complex and transient nature of the urban soundscape. Urban sound recording requires the use of robust and resilient microphone technologies, where unpredictable external conditions can have a negative impact on acoustic data quality. For the presented study, a large-scale deployment is necessary to accurately capture the geospatial and temporal characteristics of urban sound. As such, an implementation of this nature requires a high-quality, low-power and low-cost solution that can scale viably. This paper details the microphone selection process, involving the comparison between a range of consumer and custom made MEMS microphone solutions in terms of their environmental durability, frequency response, dynamic range and directivity. Ultimately a MEMS solution is proposed based on its superior resilience to varying environmental conditions and preferred acoustic characteristics.
Convention Paper 9143

2:30 pm

P9-2 Graphene Oxide Based Materials as Acoustic Transducers: A Ribbon Microphone Application Case Study—*Peter Gaskell,*[1,2] *Robert-Eric Gaskell,*[1,2] *Jung Wook (Jonathan) Hong,*[1,2] *Thomas Szkopek*[1]
[1]University, Montreal, Quebec, Canada
[2]GKL Audio Inc., Montreal, Quebec, Canada

Materials used in acoustic transducer membranes need very specific qualities that in any real system require many tradeoffs to be made. Graphene and graphene related materials are a newly discovered class of materials with some exceptional properties that has the potential to make significant contributions to the performance of many acoustical transduction systems. The properties of graphene relevant to transducer applications are discussed and two graphene based films, an aluminum coated Graphene Oxide film and an aluminum coated reduced Graphene Oxide film, are tested in a ribbon microphone application. Physical and acoustical measurements of the films indicate that with minor improvements, ribbon transducers could significantly benefit from graphene-based materials.
Convention Paper 9144

3:00 pm

P9-3 Subwoofers in Rooms: Stereophonic Reproduction—*Juha Backman*, Microsoft, Espoo, Finland

A study based on computational model of interaural level and time differences at the lowest audio frequencies, often reproduced through subwoofers, is presented. This work studies whether interaural differences can exist, and if they do, what kind of relationship there is between the loudspeaker direction and the interaural differences when monophonic and stereophonic subwoofer arrangements are considered. The calculations are made for both simple amplitude panned signals and for simulated microphone signals. The results indicate that strong narrow-band differences can exist, especially near room eigenfrequencies when the listener is close to nodes of the room modes and that the modes of the recording room can have an effect on the sound field of the listening room.
Convention Paper 9145
[Paper not presented but available for purchace]

3:30 pm

P9-4 Subwoofer Design with Moving Magnet Linear Motor—*Mario Di Cola,*[1] *Claudio Lastrucci,*[2] *Lorenzo Lombardi*[2]
[1]Audio Labs Systems, Casoli (CH), Italy
[2]Powersoft S.p.a., Scandicci, FL, Italy

A new electro-dynamic transducer has been studied, based on a moving magnet linear motor instead of a traditional moving coil, and it has been carefully described into a recently presented paper from Claudio Lastrucci. This moving magnet motor could considerably improve the conversion efficiency and the sound quality at the lowest frequency range. It has been developed around a fully balanced and symmetrical moving magnet motor geometry and it can significantly reduce the distortion, in the lowest range, to a fraction if compared to that of a conventional moving coil loudspeaker in the same range. It also offers a considerably higher power handling and overall robustness thus being able of reproducing the lowest range on bass spectrum with an unprecedented level of quality and output. The novel motor design also shows a considerable high acceleration that makes it suit-

able for the application also in the upper bass region. This paper proposes a review of the methodology that can be pursued in subwoofer design while using this motor technology. The new motor technology will require a different approach to subwoofer design. Several aspects that are in common with conventional loudspeakers will be outlined while also described those characteristics that differs significantly. The application of the technology and relative results will be shown through examples of practical applications and with measurement results.
Convention Paper 9146

4:00 pm

P9-5 A Direct Driver for Electrostatic Transducers —*Dennis Nielsen, Arnold Knott, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

Electrostatic transducers represent a very interesting alternative to the traditional inefficient electrodynamic transducers. In order to establish the full potential of these transducers, power amplifiers that fulfill the strict requirements imposed by such loads (high impedance, frequency depended, nonlinear, and high bias voltage for linearization) must be developed. This paper analyzes a power stage suitable for driving an electrostatic transducer under biasing. Measurement results of a plus/minus 400 V prototype amplifier are shown. THD below 1 % is reported.
Convention Paper 9147
[Paper presented by Henrik Schneider]

Workshop 4 — Friday, October 10
2:00 pm – 3:30 pm — Room 4

THE END IS NEAR! THE PRESSING NEED TO PRESERVE HISTORIC ANALOG SOUND RECORDINGS: AN OVERVIEW OF STRATEGIES AND BEST PRACTICES

Chair: **Konrad Strauss**, Indiana University, Bloomington, IN, USA

Panelists: *George Massenburg*, McGill University, Montreal, Quebec, Canada
Brad McCoy, Library of Congress, Charlottesville, VA, USA
John Spencer, BMS Chace LLC, Nashville, TN, USA
Nadja Wallaszkovits, Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria

The National Recording Preservation Plan, published in December 2012, identifies in excess of 46 million audio recordings held by the nation's libraries, archives, museums, record companies, and private collectors that are in need of preservation. The window of time to preserve these recordings is rapidly closing; some experts believe that we have as little as 15 years left. Archivists, librarians, engineers, and studio owners will encounter increased demand for preservation and digitization services in the coming years. This workshop, presented by the Association for Recorded Sound Collections http://www.arsc-audio.org, will focus on best practices for preserving and migrating analog audio recordings. Experts in the field will discuss strategies for extending the lifetime of the original media; building and equipping a studio for audio preservation; proper handling, cleaning, and restoration of original media; maintaining and optimizing playback machines; and building a digital infrastructure for preserving digital files.

Broadcast/Streaming Media Session 6
Friday, October 10 — 2:00 pm – 3:30 pm
Room 408 A

LISTENER FATIGUE AND RETENTION

Chair: **Marvin Caesar**, Founder and Former President Aphex Systems, Sherman Oaks, CA, USA

Panelists: *John Galvin, III*, Department of Head and Neck Surgery, David Geffen School of Medicine, UCLA, Los Angeles, CA, USA
JJ Johnston, Retired, Redmond, WA, USA
Jeff Levison, Iosono GmbH, Erfurt, Germany
Sean Olive, Harman International, Northridge, CA, USA
Greg Ogonowski, Orban, San Leandro, CA, USA
Robert Reams, Psyx Research, Santa Clara, CA, USA

A discussion of the psychological and physiological aspects of listener fatigue and its causes, the short and long term impact on people who produce, amplify, broadcast, stream as well as consumer audio.

Live Sound Seminar 5 — Friday, October 10
2:00 pm – 4:00 pm — Room 406 AB

RF SPECTRUM UPDATE: HOW MUCH LOSS AND WHEN?

Chair: **Karl Winkler**, Lectrosonics, Rio Rancho, NM, USA

Panelists: *Mark Bruner*, Shure Inc., Niles, IL USA
Joe Ciaudeli, Sennheiser, Old Lyme, CT USA
James Stoffo, Radio Active Designs, Key West, FL, USA

Recent rulings by the FCC on RF spectrum as applies to wireless microphones will have a profound effect on our industry in the years to come. Added to the loss of the 700 MHz band just a few short years ago, now the 600 MHz band is about to be auctioned. Due to this pending loss of spectrum and the resulting crowding in the remaining UHF bands, the panel will discuss strategies for success going forward, along with the potential for new frequency bands to become available for wireless mic use on a shared basis. An additional topic will be the expanded license eligibility for sound companies and venues that routinely use 50 or more wireless mic channels (including IEMs and intercoms).

Game Audio Session 6 — Friday, October 10
2:00 pm – 3:30 pm — Room 306 AB

DIABLO III: REAPER OF SOULS THE DEVIL IS IN THE DETAILS

Presenters: **Seph Lawrence**, Blizzard Entertainment, Lake Forest, CA, USA
Andrea Toyias, Blizzard Entertainment

Derek Duke, Blizzard Entertainment
Kris Giampa, Blizzard Entertainment
David Rovin, Blizzard Entertainment

Look, listen, and learn from the audio team behind Diablo III: Reaper of Souls as they show us the world of game audio development from multiple perspectives—Sound Design, Music, and VO. Hear from some the audio team members how they approached the task of bringing sound to the next installment in the Diablo franchise and how it has evolved from the original Diablo III release. Attendees will also get a peek at the Reaper of Souls cinematic.

Product Design Session 7 Friday, October 10
2:00 pm – 3:30 pm Room 402 AB

SUCCEEDING WITH EXTERNAL AND INTERNAL PARTNERS

Presenter: **Rob Chadderdon**, Sound Resolution

As audio reproduction hardware becomes increasingly complex and corporate budgets shrink, product development resources become more and more strained. Options like supplier design support, engineering consulting firms, and contract engineers sound intriguing, but can also be “more trouble than they’re worth” if not utilized well. Not so successful example: A company contracted a very qualified project management consultant at the very beginning of a design project. While he was an excellent program manager, the culture in the company needed strong technical leadership while the design architecture was being defined. The team became frustrated and progress was slow. In the end, even though the product was a success, the project was late, over budget, and the company lost some engineers. The team dynamic wasn’t considered in the decisions about outside help. Successful example: A regulatory compliance project was repeatedly re prioritized by management in favor of new product development. The changes needed to be completed before a new standard became effective to continue to sell the product. The contract-manufacturer had in-house engineering, had experience with the product, and was motivated to help since the alternative was to stop production. This previously untapped resource allowed the company to finish the project with little impact to their core engineering team. Striking a balance between internal engineering teams and external resources can optimize the cost to benefit ratio without alienating your core team or compromising your intellectual property. The presenter will present best practices and methods and provide an open forum to discuss the benefits, pitfalls, and process of finding that balance between internal and external resources.

Special Event
PLATINUM ENGINEERS
Friday, October 10, 2:00 pm – 4:00 pm
Room 403 AB

Moderator: **Michael MacDonald**, ATK Audiotek, Valencia, CA, USA

Presenters: *Dave Pensado*
Herb Trawick

Dave Pensado & Herb Trawick:
THE SCRIPT IS FLIPPED

Dave Pensado and Herb Trawick, hosts of wildly popular weekly show *Pensados Place*, are interviewed for the first time ever. See what they have learned from their superstar guests; from studio technique and engineering, to the pressures of creating online television for 180 straight weeks (and counting). Plus it just wouldn’t be Pensadisan without a couple of surprises to boot.

Live Sound Expo LSE4 LSE STAGE
INSTALLED AUDIO—SOUND-CENTRIC SPACES
Friday, October 10, 2:00 pm – 2:50 pm

Moderator: **Scott Sugden**, Head of Applications, L-Acoustics

Presenters: *Marcus Ross*, Head of Audio, Blue Man Group
Fred Vogler, FOH LA Philharmonic / System Engineer Hollywood Bowl / Sound Designer

High performance sound systems are essential to the success of high profile performance venues. World-class system designers detail application specific component selection, installation and control.

Network Audio Session 5 Friday, October 10
2:15 pm – 3:00 pm Room 409 AB

HOW MICROSOFT UPGRADED THEIR REDMOND, WA, PRODUCTION FACILITIES WITH AUDIO-OVER-IP

Presenter: **John L. Ball**, Microsoft Corp., Redmond, WA, USA

How Microsoft upgraded their production facilities and parts of the Redmond, WA campus with Audio-over-IP using a Dante network with a multi-vendor system.

Student Event/Career Development
STUDENT RECORDING CRITIQUES
Friday, October 10, 2:15 pm – 3:15 pm
Room 305

Moderator: **Ian Corbett**

Students can bring stereo or surround mixes to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Sign-up for time slots is immediately after the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on CD, DVD, memory-stick, or hard disc, as clearly labeled 44.1 KHz WAVE or AIFF files. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers! (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.)

Session P10 Friday, Oct. 10
2:30 pm – 4:00 pm Foyer 1

POSTERS: SPATIAL AUDIO

2:30 pm

P10-1 An Object-Based Audio System for Interactive Broadcasting—*Robert Oldfield*,[1] *Ben Shirley*,[1] *Jens Spille*[2]

[1]University of Salford, Salford, Greater Manchester, UK
[2]Technicolor, Research, and Innovation, Hannover, Germany

This paper describes audio recording, delivery, and rendering for an end-to-end broadcast system allowing users free navigation of panoramic video content with matching interactive audio. The system is based on one developed as part of the EU FP7 funded project, FascinatE. The premise of the system was to allow users free navigation of an ultra-high definition 180 degree video panorama for a customizable viewing experience. From an audio perspective the complete audio scene is recorded and broadcast so the rendered sound scene at the user end may be customized to match the view point. The approach described here uses an object-based audio paradigm. This paper presents an overview of the system and describes how such a system is useful for facilitating an interactive broadcast.
Convention Paper 9148
Paper presented by Ben Shirley

2:30 pm

P10-2 Effect of Headphone Equalization on Auditory Distance Perception—*Kaushik Sunder, Ee-Leng Tan, Woon-Seng Gan*,
Nanyang Technological University, Singapore, Singapore

Headphones are not acoustically transparent and thus it affects both the timbral as well as the spatial quality of the input sound source. The effect of the headphones has to be compensated by calculating the inverse of the headphones transfer function and convolving it with the binaurally synthesized audio. Headphone transfer function (HPTF) also depends on the headphone-ear coupling and thus displays high spectral variation between individuals. It has been found that the type of equalization (individual or non-individual) affects the directional perception of the virtual audio reproduced using headphones. However, little investigation has been carried out on the effect of headphone equalization on auditory distance perception. In this paper, we study in detail the perceptual effects of equalization on the auditory distance perception in the proximal region in anechoic conditions. It was found that the equalization of the headphone is critical for good distance perception. The type of equalization (individual or non-individual) did not have a significant effect on the auditory distance perception indicating that the distance perception does not depend on the idiosyncratic features. The effect of repositioning of the headphone on auditory depth perception is also studied in this work.
Convention Paper 9149

2:30 pm

P10-3 Perceptual Evaluation of Loudspeaker Binaural Rendering Using a Linear Array—
Ismael Nawfal, Joshua Atkins, Stephen Nimick,
Beats Electronics, LLC, Culver City, CA, USA

In this paper we evaluate two different techniques for spatial rendering using various linear array arrangements and filter lengths in the context of their perceived ability to render a given sound event. The two techniques explored are a recently introduced numerical technique and a conventional crosstalk cancellation system. Extensive perceptual evaluations were conducted in order to evaluate the perceived quality of the proposed and conventional synthesis methods using a binaural representation over headphones. The data were compiled to show the relationship between linear array loudspeaker arrangement, reproduction angle, filter length, and subjective mean opinion scores.
Convention Paper 9151

2:30 pm

P10-4 Uncorrelated Input Signals Design and Identification with Low-Complexity for Simultaneous Estimation of Head-Related Transfer Functions—*Sekitoshi Kanai*,[1] *Kentaro Matsui*,[1,2] *Yasushige Nakayama*,[2] *Shuichi Adachi*[1]
[1]Keio University, Yokohama-shi, Kanagawa, Japan
[2]NHK Science & Technology Research Laboratories, Setagaya, Tokyo, Japan;

In our previous study, we verified that a set of head-related transfer functions (HRTFs) can simultaneously be estimated by treating it as a multi-input single-output (MISO) system. However, this leads to a lack of accuracy if appropriate input signals are not chosen and high computational cost is required to estimate. To improve the accuracy, a novel input design method is proposed. Moreover, we also propose a system identifiation method that reduces the space complexity even when the number of measuring directions increases. The effectiveness of the proposed methods was demonstrated through simultaneous estimation experiments of HRTFs.
Convention Paper 9152

2:30 pm

P10-5 An Evolutionary Algorithm Approach to Customization of Non-Individualized Head Related Transfer Functions—*Eric S. Schwenker*,[1,2] *Griffin D. Romigh*[2]
[1]Carnegie Mellon University, Pittsburgh, PA, USA
[2]Air Force Research Labs, Wright Patterson Air Force Base, OH, USA

Currently, the commercialization of high-quality virtual auditory display technology is limited by the costly and time-consuming methods required for obtaining listener-specific head-related transfer functions (HRTFs), directionally-dependent filters that encode spatial information. As such, there is an increased interest in the estimation of individualized HRTFs based on non-acoustic data. This study highlights the capabilities of an evolutionary algorithm method applied to the complex parameter optimization problem that arises when HRTFs are fit to individuals (or populations), rather than acoustically measured. Results suggest the algorithm may be capable of providing HRTFs that improve localization through both personalization of generic HRTFs and the generation of an optimized set of generic HRTFs.
Convention Paper 9153

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2:30 pm

P10-6 Multichannel-Reproduced Music with Height Ambiences: Investigating Physical and Perceptual Factors for Comprehensive 3D Experience—*Antonios Karampourniotis*,[1] *Mark J. Indelicato*,[1] *Sungyoung Kim*,[1] *Richard King*[2]
[1]Rochester Institute of Technology, Rochester, NY, USA;
[2]McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

This study investigated the influence of the height loudspeaker positions and their signals on perceived overall sound quality. Two layers and a total of seventeen loudspeakers were used in a horizontal and height layer. Twelve participants were asked to subjectively rank and describe eight randomly presented configurations that consisted of four height loudspeakers. A set of inverse filters was generated and applied to remove the room's acoustic influence and a new set of listeners were asked to evaluate sound quality. The experimental results indicate the significance of the height loudspeaker positioning for a perceived 3D sound field. These results show that the room's acoustic influence affects desired perceptual characteristics of the sound field and influences subjective preferences.
Convention Paper 9154

Network Audio Session 6 Friday, October 10
3:00 pm – 3:45 pm Room 409 AB

DEPLOYMENT OF LARGE SCALE NETWORKING BASED ANNOUNCEMENT SYSTEM FOR SYDNEY RAIL

Presenter: **Mark Lowndes**, Stagetec

Large deployment of Dante Networking used to upgrade the mass transit rail systems. Sydney Trains, the operator of all railway services in New South Wales. Stagetec presents a look at this system rollout to equip the platforms with new announcement technology.

Friday, October 10 3:00 pm Room 405
Technical Committee Meeting on Fiber Optics for Audio

Project Studio Expo PSE5 PSE STAGE
LISTEN UP, AND LEARN!
Friday, October 10, 3:00 pm – 4:00 pm

Presenters: **Alex Case**, University of Massachusetts Lowell, Lowell, MA, USA
Stephen Webber, Berklee College of Music, Valencia, Spain

Bring your ears, your artistry, and your opinions for an hour dedicated to the art of listening. Guided by your hosts, Stephen Webber and Alex U. Case, you'll focus on an iconic record that is a proven success—artistically and commercially—and glean useful aural insights. We'll listen as producers, engineers, composers, performers, and music fans analyzing the elements that contribute to the work's success. You'll gain a deeper appreciation of this recording. More importantly, you'll be inspired to approach your own work in new ways. Most importantly, you'll get an up-close view into how experienced audio engineers break down what they hear, empowering you to keep learning whenever you listen.

Live Sound Expo LSE5 LSE STAGE
MIXING PRIMER
Friday, October 10, 3:00 pm – 3:50 pm

Presenter: **Chap Cooper**

Aimed at the club, live event and House Of Worship mixer, this session offers a practical, systematic approach to building a mix for smaller venues, including tips for controlling feedback and excessive stage volumes.

Broadcast/Streaming Media Session 7
Friday, October 10 3:30 pm – 5:00 pm
Room 306 AB

MPEG-DASH—WHAT ABOUT AUDIO?

Chair: **Jan Nordmann**, Fraunhofer USA, San Jose, CA, USA

Panelists: *Rupert Brun*, BBC Audio & Music, London, UK
Richard Doherty, Dolby Laboratories, San Francisco, CA, USA
Ronny Katz, DTS, Calabasas, CA, USA
Greg Ogonowski, Orban, San Leandro, CA, USA

MPEG-DASH is the emerging adaptive streaming standard that's designed to replace proprietary transport mechanisms such as HDS, Smooth Streaming or HLS. Already adopted or considered by many TV standards around the world, deployed by YouTube and Netflix and supported in Google Chrome, Chromecast, and Microsoft Internet Explorer among others, MPEG-DASH is taking the video world by storm. Even though adaptive streaming is often considered a pure video topic, the panel will discuss its implications on the audio side from an engineering and commercial perspective.

Product Design Session 8 Friday, October 10
3:30 pm – 5:00 pm Room 402 AB

CHOOSING THE BEST PROCESSOR FOR YOUR AUDIO DSP APPLICATION

Presenter: **Paul Beckmann**, DSP Concepts, Sunnyvale, CA, USA

This tutorial focuses on one of the key design decisions for audio product makers today - which processor to use for audio signal processing. While digital signal processors (DSPs) have been the processor of choice for many years, recent trends such as connectivity, integration, and low power are pushing designs to microcontrollers. This tutorial reviews the state of the art for both DSPs and microcontrollers and their applicability to audio products. We provide extensive benchmarking results and real world examples allowing you to make the right choice for your next design.

Game Audio Session 7 Friday, October 10
3:45 pm – 5:15 pm Room 408 B

DYNAMIC MIXING FOR GAMES

Presenter **Simon Ashby**, Audiokinetic

Given the linear nature of film and music, audio mixing is easily controlled and predictable. Mixing game audio brings with it many challenges, including performance constraints and the non-linear event based triggering of in-game sounds. Using real-game practical audio examples, this session will demonstrate the many positive benefits that dynamic audio mixing can have on modern sound design.

Network Audio Session 7 Friday, October 10
4:00 pm – 5:00 pm Room 409 AB

HOW THE NEWLY FORMED MEDIA NETWORKING ALLIANCE WILL SUPPORT AES67 ADOPTERS

Chair: **Bill Scott**, Bosch Communications Systems, Burnsville, MN, USA

Panelists: *Terry Holton*, Yamaha R&D Centre, London, UK
Stefan Lederberger, Lawo Group, Zurich, Switzerland; LES Switzerland GmbH
Marty Sacks, Telos Alliance
Rich Zweibel, QSC Audio

IT technology has a lot to offer, far more than our industry could ever afford to develop on its own. A wide installed base and huge functionality at an unbeaten price point are just two examples. We need to adopt what is already there, carefully rethink our traditional requirements, and adapt the commodity technology to our needs. AES67 is the essential recipe how to do that, making sure all manufacturers have a common denominator while still being able to maintain healthy competitive advantages. AES67 (after AES/MADI) is to the audio industry what VoIP (after ISDN) is to the telephone industry.

This panel will summarize AES67, discuss why we need it within the industry, and the purpose of the Media Networking Alliance whose mission will be to encourage adoption and interoperability of the standard.

Friday, October 10 4:00 pm Room 407
Standards Committee Meeting on Audio File Transfer and Exchange

Live Sound Expo LSE6 LSE STAGE
THE ART OF THE SOUND CHECK
Friday, October 10, 4:00 pm – 4:50 pm

Moderator: **Mark Frink**

Presenters: *Thomas Pesa*, Super Bowl, Grammy Awards, Academy Awards – ATK
Ken "Pooch" Van Druten, FOH - Linkin Park

Even the first song can sound good with pre-show planning and an organized approach to the oft rushed sound check.

Project Studio Expo PSE6 PSE STAGE
WHERE TO FOCUS YOUR STUDIO GEAR PURCHASING WITH A LIMITED BUDGET
Friday, October 10, 4:15 pm – 5:00 pm

Presenter: **Larry Crane**, Tape Op Magazine, Portland, OR, USA; Jackpot! Recording Studio

Musicians and home recordists on lower budgets are always on the prowl for recording equipment deals and looking to upgrade what they currently have. *Tape Op* magazine founder and editor, Larry Crane, was once a musician recording at home and for 18 years has owned a commercial studio, Jackpot! recording in Portland, Oregon. In this presentation he will cover what recording equipment is truly important, how to never regret your purchases, and where to focus your spending.

Live Sound Seminar 6 Friday, October 10
4:30 pm – 6:30 pm Room 406 AB

UNDERSTANDING THE MYSTERY OF MIXING FOR IN-EAR MONITORS

Moderator: **Mike Pirich**, VER

Panelists: *Beau Alexander*, Monitor engineer for Green Day, Tool, Nashville, TN, USA
Michael Parker, Parker Audio Inc., Los Angeles, CA, USA
Michael Santucci, Sensaphonics

This seminar is geared toward the music engineer who wants to make the transition to mixing with in-ear monitors. Although in-ear monitors seem ubiquitous on television and stage, many sound engineers still rely on the stage monitor due to the cost of in-ears or the lack of experience. As the cost of wireless comes down, there will be more opportunities for less experienced mixers to use in-ear monitors, but what are some of the things to look out for as you make this transition. Professional monitor mixers will discuss some of the ways a sound engineer can prepare for mixing in-ears including important hearing safety issues as well as some of the minimum equipment requirements.

Special Event
PLATINUM MASTERING: HIGH RESOLUTION AUDIO
Friday, October 10, 4:30 pm – 6:30 pm
Room 403 AB

Moderator: **Bob Ludwig**, Gateway Mastering Studios, Inc., Portland, ME, USA

Panelists: *Bruce Botnick*, Music Producer, Engineer, Mastering Engineer / VP Content Acquisition for Pono Music, Los Angeles, CA, USA
Mark Donahue, Recording and Mastering Engineer, Soundmirror, Boston, MA, USA
Andrés Mayo, President, Andrés Mayo Mastering & Audio Post, Buenos Aires, Argentina
Barak Moffitt, Head of Strategic Operations at Universal Music Group, Los Angeles, CA, USA

There is a unified push to make High Resolution Audio the next big thing for the consumer. The Digital Entertainment Group in cooperation with the Consumer Electronics Association, The Recording Academy, and the major record labels have recently defined High Resolution Audio as “lossless audio that is capable of reproducing the full range of sound from recordings that have been mastered from better than CD quality music sources.” We will discuss the ramifications of this initiative with experts who have worked with High Resolution Audio for a long time as well as play many examples from the participants so the audience can hear for themselves how High Resolution Audio can allow the listener to hear exactly what the artist has been hearing in the studio while creating their music.

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Session EB3 **Friday, Oct. 10**
5:00 pm – 6:15 pm **Room 308 AB**

PAPERS SESSION 1

Chair: **Christoph Musialik**, Sennheiser

5:00 pm

EB3-1 "It Has to Work With the Picture": Audio Education for Film and Media Students— *Ufuk Onen*, Bilkent University, Ankara, Turkey

Audio education for students who major in filmmaking, video production, visual media practice, or visual communication design in universities' bachelor-degree programs usually starts with creating and improving awareness for sound both in general and, also, in relation to visual media as well. In addition to that, since these students utilize recording and mixing only as a part of their professional practice, not as their main field of specialty, teaching the technical concepts and aesthetics to them requires making use of different content and approach than to those who intend to become audio specialists. This paper discusses these issues by using COMD 361 Sound Design course at Bilkent University, Department of Communication and Design, as a case study.
Engineering Brief 167

5:15 pm

EB3-2 Film Production Sound in Secondary Markets— The Value of Networking—*Tom Hauser*, Hooz Audio, Winston-Salem, NC, USA

I came back home to North Carolina after a negative experience interning in Nashville and eventually navigated my way into sound for picture in corporate and commercial work. I have built my network over several years, gone to grad school for film scoring, moved away from the slightly more competitive area of Raleigh, and now have a small 5.1 mixing studio in Winston-Salem, NC. I want to highlight some of the things a young person needs to know to get work as a new comer among veterans where ever they go, emphasize continual skill development beyond school, the value of personal relationships, and navigating the ups and downs of being a freelancer.
Engineering Brief 168

5:30 pm

EB3-3 "Object" Panning for Film: Challenges and Solutions—*Ben Loftis*, Harrison Consoles, Nashville, TN, USA

Object-based panning allows film mixers to break free of the limits imposed by traditional surround formats. Harrison Consoles has developed software panning that allows sounds to be treated as objects in a 3D space that can then be rendered in various surround formats. This enables users to mix in immersive sound formats and transition to traditional surround formats without having to start a new mix; it was most recently implemented at Sony Pictures for the mix of *The Amazing Spider-Man 2*. Explained in this paper are the obstacles that we faced, and how we overcame them, making technologies such as Dolby Atmos and Barco Auro3D part of the original mix rather than an afterthought.
Engineering Brief 169

6:45 pm

EB3-4 Orchestral Recording and Live Webcasting at McGill University—*Alejandro Aspinwall*, McGill University, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada

Recording and broadcasting live audio and video of a large student orchestra performance can be a tremendous challenge but can also be a great educational experience when organized properly. The most important aspects to succeed in this task are assembling a team of students with the adequate technical skills to resolve potential problems and putting together an efficient system that can handle potential problems such as power outages or digital audio workstation failure. This kind of event prepares the students for real life scenarios where they will encounter similar workflows and will be able to spot and prevent technical flaws that could compromise the success of future events. Finally, having a group of students with assigned responsibilities will improve their teamwork skills and allow them to communicate with peers outside their area of expertise.
Engineering Brief 170

6:00 pm

EB3-5 Stereo Bluetooth and Low Latency Applications —*Jonny McClintock*, CSR, Belfast, Northern Ireland, UK

The A2DP Bluetooth protocol is used to transport stereo audio over a non-synchronous packetized structure. Using a frame-based codec, i.e., SBC or AAC, results in system latencies between 150 and 800 milliseconds with wide drifts up to +/– 200ms. A2DP can be used for music but because of the problems associated with the frame-based codecs, Bluetooth is not suitable for audio for video or gaming applications. The aptX codec offers an alternative. It uses a fundamentally different coding architecture and is sample-based delivering a system latency of 40ms with minimal drift, i.e., +/– 1 ms. With the use of aptX, Bluetooth can now be used to wireless connect TV's to soundbars, gaming consoles to headsets, and PC's to speakers.
Engineering Brief 171

Tutorial 14 **Friday, October 10**
5:00 pm – 6:30 pm **Room 309**

LOUDSPEAKER DESIGN PART 2: HORN DRIVERS— HISTORY, THEORY, AND TECHNOLOGY— A MASTER CLASS

Presenter: **Alexander Voishvillo**, JBL/Harman Professional, Northridge, CA, USA

Horn drivers are the oldest "electrical" transducers and their invention preceded the development of the direct-radiating loudspeaker. Historically, several major inventions in the late 1800s and early 1900s triggered the invention and development of horn drivers; it was the invention of the telephone, radio, phonograph, and triode. Principles of operation for the compression driver, based on matching the output mechanical impedance of the vibrating diaphragm and the loading impedance of the phasing plug and horn, were understood since the beginning of the 1900s. The motors of the early compression drivers were based on a moving armature rather than on a moving coil.

Broadcast/Streaming Media Session 8
Friday, October 10 5:00 pm – 6:30 pm
Room 408 A

AUDIO ISSUES AND HTML5

Chair: **Valerie Tyler**, College of San Mateo, San Mateo, CA, USA

Panelists: *Dale Curtis*, Google, Seattle, WA, USA
Greg Ogonowski, Orban, San Leandro, CA, USA
Alex Schoepel, DTS Inc., Calabasas, CA, USA
Jerry Smith, Microsoft
Charles Van Winkle, Adobe, 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 9 Friday, October 10
5:00 pm – 6:30 pm Room 402 AB

AUDIO PROCESSING PLATFORMS AND ARCHITECTURE

Presenter: **Danny Olesh**, Gateway Audio

Today's audio product design engineer and product manager are faced with a dizzying number of choices in putting together digital platforms for their end product. This session by Danny Olesh of Gateway Audio will discuss best practices in architecting and designing great products around the power of DSP, Networking, Intelligence, Firmware and Software. Topics will include platform and architecture development, buy vs. build, software building blocks, and managing a diverse engineering team that includes internal and external engineers, manufacturing partners, design partners and others.

Student Event/Career Development
RECORDING COMPETITION—PART 1
Friday, October 10, 5:00 pm – 7:00 pm
Room 306 AB

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. 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 Sunday 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 those who don't make it to the finals, and it's a great chance to meet other students and faculty.

Traditional Acoustic Recording
Judges: David Bowles, Scott Levine, Ulrike Schwarz, Jonathan Wyner

Sound for Visual Media
Judges: Tim Edwards, John Rodd, Scott Stambler

Project Studio Expo PSE7 PSE STAGE
IRONS IN THE FIRE: CAREER AND BUSINESS DEVELOPMENT MENTORING WITH THE MANHATTAN PRODUCERS ALLIANCE
Friday, October 10, 5:15 pm – 6:00 pm

Presenters: **Bassy Bob Brockman**
Joe Carroll, Manhattan Producers Alliance, New York, NY, USA
Steve Horowitz

Bring your energy, enthusiasm, business ideas, and questions. At this event the focus is on YOU! Succeeding in music today is, more than ever, challenging. Members of the Manhattan Producers Alliance will give a brief talk about developing your brand and your business, and functioning as a creative talent in an ever-changing music business. Take this unique opportunity to meet some ManhatPro members and spend some time learning some tips and tricks for business development. You'll participate in our open discussions, discuss your personal career goals one on one, and get a chance to meet some ManhatPro members.

Game Audio Session 8 Friday, October 10
5:30 pm – 6:30 pm Room 408 B

NEW TECHNIQUES FOR ZERO-LATENCY CONVOLUTION

Presenter: **Frederick Umminger**, Senior Manager Software Engineering, Sony Computer Entertainment America

Low latency in audio is critical for virtual reality applications. This creates a need for low-latency but highly efficient convolutions for HRIR and reverberation. Ordinary, FFT-accelerated block-based convolution requires increasing latency in order to lower the CPU load with larger FFT block sizes. In 1993, William Gardner proposed a method of using non-uniform block sizes to simultaneously attain low latency and high efficiency. In 2008, Jeffrey Hurchalla proposed another method to accomplish this goal. This talk introduces two recent techniques for performing efficient zero-latency convolution.

Historical Event H3
BENJAMIN BAUER AND THE SHURE UNIDYNE MICROPHONE: 75 YEARS OF AUDIO LEGEND
Friday, October 10, 5:30 pm – 6:30 pm
Room 409 AB

Presenter: **Michael Pettersen**, Shure Incorporated, Niles, IL, USA

2014 marks the 75th anniversary of the Shure model 55 Unidyne Microphone and the Uniphase acoustical network. This historical presentation provides an overview of the life and career of Benjamin Bauer who developed the ➥

Uniphase network. Bauer (born Baumzweiger), an immigrant from Ukraine, developed the network at age 24 and it earned him the first of over 100 patents in audio technology. After a twenty year career at Shure Incorporated, Bauer headed audio research for twenty years at CBS Laboratories. Included in the presentation are photos and details of the first Unidyne mic elements constructed by Bauer. These prototypes were uncovered December 2013 in the Shure Archived and have not been seen since the early 1940s. Also included are historical photos of Unidyne microphones from the past 75 years.

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Friday, October 10, 6:45 pm – 8:30 pm
Room 403 AB

Lecturer: **Marty O'Donnell**, Marty O'Donnell Music, Seattle, WA, USA

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 137th AES Convention is Marty O'Donnell. O'Donnell received a Bachelor of Music Composition from Wheaton College Conservatory and a Masters of Music Composition from USC in the early 1980s. He started an original music and audio production company with Michael Salvatori. From their studio in Chicago O'Donnell/Salvatori wrote and produced music and audio for hundreds of TV and radio commercials, as well as movie scores. In 1997 they began working on games and did the audio design for Cyan's Riven: The Sequel to Myst, and all the music and audio for Bungie's Myth: The Fallen Lords. Marty joined Bungie as full time Audio Director in May of 2000 ten days before they were purchased by Microsoft and subsequently wrote and produced award winning music and audio for the Halo series. In 2007 he helped establish Bungie as an independent game company and built an audio team to work on the upcoming game Destiny. Recently, in collaboration with Salvatori and Sir Paul McCartney he completed an orchestral/choral suite titled Music of the Spheres scheduled to be released this August. In April of 2014 he started his own company Marty O'Donnell Music.

Marty is the famed audio director behind the award-winning Halo game series and is responsible for the biggest selling game soundtrack of all time. In his talk entitled "The Ear Doesn't Blink: Creating Culture With Adaptive Audio," O'Donnell will draw on his unique perspective from games, film and jingle-writing to share the creative challenges of working in non-linear media such as games.

Session P11 — Saturday, Oct. 11
9:00 am – 11:30 am — Room 309

ROOM ACOUSTICS

Chair: **Doyuen Ko**, Belmont University, Nashville, TN, USA

9:00 am

P11-1 Development of a Sound Field Diffusion Coefficient—*Alejandro Bidondo, Mariano Arouxet, Sergio Vazquez, Javier Vazquez, Germán Heinze, Adrián Saavedra*, Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina

This research addresses the development of an absolute descriptor and its associated calculation software algorithm with user interface, which quantifies the degree of diffusion, in a third octave band basis and globally, of a sound field from a monaural impulse response. The degree of sound field diffuseness is related with the probability of getting accumulated energy in discrete reflections compared to the total energy contained in all the reflections of an impulse response, after extracting its decay and normalizing it in respect to its reverberation time. The coefficient range varies between 0 and 1, zero being "no diffuseness" and one being a maximum absolute reference obtained from analyzing different types of rooms. The challenge has been not only to develop this coefficient theoretically but converting its theory in a mathematical numerical calculation through a dedicated software. This coefficient may be used both to study the effects of sound diffusers coatings as well as coated surfaces and the degree of perception of different values which may appear within sound fields.
Convention Paper 9155

9:30 am

P11-2 Simulating Talker Directivity for Speech Intelligibility Measurements—*Peter Mapp*, Peter Mapp Associates, Colchester, Essex, UK

The research investigated how both the frequency response and directivity of a talker or voice simulator can affect the measured and predicted speech intelligibility within a given situation. Current sound system and acoustic standards provide little guidance as to the required acoustic characteristics of a simulator or the effects that its directivity and frequency response parameters may have. It is shown that the both driver size and format as well as the overall frequency response can have a marked effect on speech intelligibility measurements. A range of talker loudspeaker simulators was investigated in both real and simulated environments. The research shows that the characteristics of several commonly used simulators varied significantly which markedly affected the resultant intelligibility measurements. The results of the work are used to formulate a number of recommendations for talker and voice simulator electroacoustic characteristics and standardization of measurement methods.
Convention Paper 9156

10:00 am

P11-3 Visualization of Early Reflections in Control Rooms—*Malcolm Dunn, Daniel Protheroe*, Marshall Day Acoustics, Auckland, New Zealand

Measurements were undertaken in a variety of control rooms with a system utilizing a compact microphone array and sound intensity technique

to estimate the direction of early reflections. This paper presents the results of these measurements including 3D intensity plots that provide a visual representation of sound arrivals at the listener position. The effectiveness of this type of system for the detection of problematic reflections and the evaluation of the listening environment is discussed.
Convention Paper 9157

10:30 am

P11-4 Holistic Acoustic Absorber Design: From Modeling and Simulation to Laboratory Testing and Practical Realization—*Rob Toulson, Silvia Cirstea*, Anglia Ruskin University, Cambridge, UK

In developing a new acoustic absorber, a number of practical design challenges are experienced. Complex mathematical models for many acoustic absorbing methods have previously been developed, however there is very little accessible data describing how those models perform in a practical implementation of the design. This project describes a holistic approach to the development of a novel slotted film sound absorber and presents the results at each design iteration. Initially a number of mathematical models are considered, in order to optimize the design geometry for a maximum sound absorbing effect. Second, the modeled designs are laboratory tested with an impedance tube system. Finally, the practical acoustic absorber design, including framing and mounting methods, is finalized and tested in an ISO accredited reverberation chamber. The results of the modeling, impedance tube testing, and the room testing are all considered. It is seen that the simulation and impedance tube results match very closely, whereas the practical implementation performance is lower in terms of acoustic absorption. This research therefore presents a valuable case study for acoustic absorber designers in helping to better predict the final performance of their designs.
Convention Paper 9158

11:00 am

P11-5 Oscillating Measurement Motion—Myth or Magic? —*Wolfgang Heß, Stefan Varga*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Acoustical reproduction in cars is different to acoustical reproduction in rooms or other larger environments. This is caused by the size of the car cabin, by surrounding materials as well as non-ideal loudspeaker enclosures, and reproduction positions. In order to achieve a well balanced sound system, for most sound system tuning engineers, acoustical measurements are essential in the process of designing and optimizing an audio system. This paper analyses and describes different methods of measuring the frequency-gain behavior of single or multiple loudspeakers. When measured at a single position, often dips and notches in the form of comb-filters can be observed in the frequency response. This work focuses on practical aspects: In which way is it possible to measure a frequency response that describes the sound at the listening area as correct as possible, how can we reduce comb-filter effects? In which way is a fast and adequate measurement possible? Results showed a significant reduction by movement of the measurement microphone. An evaluation by listening tests showed that frequency response averaging by microphone movements led not only to smoother magnitude responses but also to better sound experience through less equalization.
Convention Paper 9159

Session P12 **Saturday, Oct. 11**
9:00 am – 12:00 noon **Room 308 AB**

TRANSDUCERS—PART 3

Chair: **Eric Gaskell**, GKL Audio Inc., McGill University, Montreal, Quebec, Canada

9:00 am

P12-1 Compensation of the Flux Modulation Distortion Using an Additional Coil in a Loudspeaker Unit —*Niccolo Antonello, Finn Agerkvist,* Technical University of Denmark, Kgs. Lyngby, Denmark

Flux modulation is one of the main causes of distortion in electrodynamic loudspeaker units. A new compensation technique that eliminates this type of non-linearity using an additional compensation coil in the speaker unit is presented. An equivalent circuit model of the device including the compensation coil is derived. The compensation technique consists on feeding the compensation coil and voice coil with filtered versions of the wanted audio signal. Simulations show that a significant reduction in flux modulation distortion can be achieved with this technique. A simple magnetic circuit has been constructed to test the method on a real device, and the measurements show the method works, also when eddy currents are present.
Convention Paper 9160

9:30 am

P12-2 Physical Requirements of New Acoustic Transducers to Replace Existing Moving-Coil Loudspeakers—*Gyeong-Tae Lee, Jong-Bae Kim, Dong-Hyun Jung*, Samsung Electronics, Suwon-si, Gyeonggi-do, Korea

New types of acoustic transducers have recently emerged as a possible alternative to moving-coil loudspeakers. However, they have not met the performance for commercialization enough to replace existing loudspeakers. In this paper to identify the requirements for high performance, we derived an analytical model of a moving-coil loudspeaker based on physical acoustics and electroacoustics, and evaluated the simulated results of the model in terms of acoustic performance. Finally, we discussed the physical requirements of new acoustic transducers from the perspective of bass performance, tonal balance, decay time, and spatial directivity and then made some suggestions.
Convention Paper 9161

10:00 am

P12-3 Nonlinear Flux Modulation Effects in Moving Coil Transducers—*Felix Kochendörfer, Alexander Voishvillo*, JBL/Harman Professional, Northridge, CA, USA

Adverse effects in transducer motors produced by the nonlinear force factor on performance of loudspeakers are well understood. Nonlinear effects produced by dependence of the voice coil inductance and resistance on the coil's position and on the current are less obvious. Previous work of the authors showed the nonlinear behavior of the voice coil inductive component presented as a function of the voice coil position, which was obtained by FEA simulation tools. This work is an attempt to obtain variation of the aforementioned parameters experimentally as functions of the voice coil position, current, and frequency.
Convention Paper 9162

10:30 am

P12-4 Design and Construction of a Circular AMT Speaker of 360° Radiation—*Rodrigo Fernández Arcani, Alejandro Sánchez Caparrós*, Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina

Among the electro-mechano-acoustical dynamic transducers there is one whose operating principle is not widely publicized. This transducer is known as Air Motion Transformer (AMT). Using this operation principle, a prototype of an AMT transducer type was designed and built with the particularity of making a cylindrical diaphragm. The behavior of the prototype was assessed and noted that it achieves a 360 degree of quasi uniform high frequency sound pressure level radiation in the horizontal plane. In order to improve the efficiency, several configurations are proposed.
Convention Paper 9163

11:00 am

P12-5 Identification Compression Driver Parameters Based on a Concept of Diaphragm's Frequency-Dependent Area— *Alexander Voishvillo*, JBL/Harman Professional, Northridge, CA, USA

In the previous work, matrix analysis was applied the derivation of the transfer matrix of a compression driver's diaphragm. Its mechanical impedance consisted of lumped parameters and a part corresponding to the high-frequency breakups. In the current work the mechanical impedance is based on lumped parameters whereas the area of the diaphragm is presented as a function of frequency. The transfer function of compression driver is derived from the overall matrix that includes the frequency-dependent area of the diaphragm. The area is included in the transformation matrix that links the mechanical and acoustical parts. By equating the measured SPL for a given input voltage to SPL derived from the model, the expression for the frequency-dependent area of the diaphragm can be derived. This information is used in modeling and design of different drivers using the identified diaphragm.
Convention Paper 9164

1:30 am

P12-6 A General Approach for the Acoustic Design of Compression Drivers with "Narrow" Channels and Rigid Diaphragms—*Jack Oclee-Brown*, GP Acoustics (UK) Ltd., Maidstone, UK

A generalized approach to determining the position and size of "narrow" phase-plug channels is presented that is applicable to compression drivers with radiating diaphragms of arbitrary geometry. In addition, it is demonstrated that by carefully shaping the compression cavity according to the diaphragm motion then optimal behavior results. FEM computed examples are presented demonstrating the method for the case of a conical diaphragm geometry.
Convention Paper 9165

Tutorial 15 **Saturday, October 11**
9:00 am – 10:15 am **Room 408 A**

ALL ABOUT: TIMBRE

Presenter: **Alex Case**, University of Massachusetts—Lowell, Lowell, MA, USA

One of the most important properties of audio, timbre's definition may be broader than you think. Distilling timbre into its essential elements reveals its rich meaning and lets us make a direct connection between it and the decisions, actions, and devices that drive it. Much is in the hands of the performer—their technique and their instrument. Engineers have obvious impact through microphone choice and placement. But we also focus on how to use compression, delay, reverb, and distortion effects to refine timbre.

Tutorial 16 **Saturday, October 11**
9:00 am – 11:00 am **Room 409 AB**

AUDIO FORENSICS: AN OVERVIEW

Chair: **Jeff M. Smith**, National Center for Media Forensics, Denver, CO, USA; University of Colorado Denver, Denver, CO, USA

Presenters: *Durand Begault*, Audio Forensic Center
Eddy B. Brixen, ebb-consult, Smørum, Denmark
Catalin Grigoras, University of Colorado Denver, Denver, CO, USA
Gordon Reid, CEDAR Audio Ltd.

This tutorial will feature several presenters engaged in various areas of audio forensics in lively discussion geared toward experts learning from one another and to benefit the introductory attendee. Tutorial Chair, Jeff M. Smith (National Center for Media Forensics, CU Denver) and chair of the Technical Committee on Audio Forensics, will present on Speaker Analysis and the application of Bayesian likelihood. Catalin Grigoras (NCMF, CU Denver) will present on the best practices and future challenges in forensic audio authentication. Gordon Reid (CEDAR Audio Ltd.) will present on noise reduction and speech enhancement techniques. Eddy Brixen (EBB consult) will present on auditory crime scene analysis. Finally, Durand Begault (Audio Forensic Center) will wrap up some loose ends with a discussion of forensic audio miscellany including musicological forensics, warning signal audibility, acoustics, and more.

This session is presented in association with the AES Technical Committee on Audio Forensics

Broadcast/Streaming Media Session 9
Saturday, October 11 9:00 am – 10:30 am
Room 306 AB

SOUND DESIGN AND STORYTELLING: HOW TO CREATE THE ENVIRONMENTS AND SOUNDS NEEDED TO ENHANCE ANY TALE

Presenters: **David Shinn**, SueMedia Productions, Carle Place, NY, USA
Sue Zizza, SueMedia Productions, Carle Place, NY, USA

No matter what the medium (film, TV, games, radio, etc.) creating a fully realized sound scape helps to bring your story to 'life' and engages your audience. Blending recorded effects captured specifically for your story, along with recorded effects from the many SFX libraries available, and live 'in-studio' effects, helps to create a sound scape tailored to your story's environments and worlds. This session will take you through the different ways (in mono, stereo, and surround sound) to:

- Capture recordings in the field
- Showcase live SFX performance techniques (foley) and review
- Microphone choices for sound effect recordings.

As part of the demonstration a short skit will be performed showcasing the different performance and SFX techniques.

Live Sound Seminar 7 Saturday, October 11
9:00 am – 11:00 am Room 406 AB

CHOOSING THE RIGHT LOUDSPEAKER FOR THE APPLICATION

Presenter: **Steve Bush**, Meyer Sound Labs, Inc., Berkeley, CA USA

A wide variety of loudspeakers are designed for use in different modern sound reinforcement applications. Examples include line arrays, point-source loudspeakers, loudspeakers with narrow coverage, wide coverage, short-throw and long-throw, and numerous other variations. The primary goal of sound system design is to evenly distribute the sound to the listening audience at the appropriate volume, but it can quickly become overwhelming to know how to best implement the tools available to effectively achieve that goal. This presentation will step participants through a general overview of common loudspeaker types and how they are implemented in different sound system design applications.

Game Audio Session 9 Saturday, October 11
9:00 am – 11:00 am Room 408 B

GAME AUDIO CAREERS 101— HOW TO JUMP START YOUR CAREER

Chair: **Steve Horowitz**, Game Audio Institute, San Francisco, CA, USA; Nickelodeon Digital

Panelists: *Brennan J. Anderson*, Disney Interactive
Stephan Schütze, Sound Librarian, Melbourne, Australia
Richard Warp, Manhattan Producers Alliance, New York, NY, USA; Leapfrog Enterprises Inc., Emeryville, CA, USA
Guy Whitmore, PopCap Games

Everyone wants to work in games, just check out the news. The game industry is on the rise 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? 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!

Network Audio Session 8 Saturday, October 11
9:00 am – 10:30 am Room 404 AB

IMPLEMENTATION OF A LARGE SCALE ETHERNET AVB AUDIO NETWORK AT ESPN

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

Panelists: *Brian Ames*
Warren Belkin
Christian Diehl
Jonathan Pannaman

ESPN has installed a large scale Ethernet AVB audio distribution network at its Bristol, CT Digital Center 2 (DC2). Panelists involved in the installation will discuss the business case for this major project along with the audio requirements and the IT infrastructure required to accomplish this project. The presentation will address the challenges in implementing an AVB based infrastructure on such a vast scale and the in-service experience since the facility opened in June, 2014.

Questions that this presentation will answer:

- What were the audio requirements that ESPN and how did AVB provide a solution?
- What were your other technology considerations and why did you chose AVB?
- What role did open standards play in your decision to choose AVB?
- What were some of the challenges for the installation and how did you arrive at a solution?
- What was the biggest benefit to utilizing AVB for this installation? What are the implications of the install for today’s workflow?

Product Design Session 10 Saturday, October 11
9:00 am – 10:30 am Room 402 AB

SELECTING AND WINNING WITH GLOBAL PARTNERS

Presenter: **Chris von Hellman**, Tymphany

Today's global economy presents a myriad of options in developing and delivering audio products. Most companies are engaged with some combination of: 1) Contract Manufacturers, 2) Original Equipment Manufacturers (OEMs), 3) Original Design Manufacturers (ODMs), 4) Design Engineering Companies, 5) Industrial Design Firms, 6) Market Research Firms.

Market acceptance, time to market, and cost margin are ➥

the critical measures of a product?s success. To achieve this, product management and engineering today must select and partner with the right combination of these organizations and the ones that best fit both partners' goals. Simply sending bids out to prospective suppliers will not suffice in today's highly competitive world. There are many dimensions beyond product cost that must be considered some more tangible than others. A spreadsheet should not be making such a critical decision.

The presenter will lead a discussion about the details of how to select and work with these partners that leads to a long term and highly successful relationship. This will include the following topics: 1) What type of partner do you need? 2) How to we find the right partner? 3) How do we structure the relationship? 4) What are the keys to minimizing risk? 5) Can we work without finger pointing? 6) How do we protect our intellectual property, uniqueness and brand equity? 7) What needs to change about our company to ensure success with development partners?

The presenter will discuss how a company can leverage its core competencies by selecting partners whose capabilities and competencies are complimentary. The resulting partnership enables the company's products to maximize the value of internal and external assets, resources and skill sets.

Student Event/Career Development
STUDENT DESIGN EXHIBITION
Saturday, October 11, 9:00 am – 10:30 am
S-Foyer 1

All accepted entries to the AES Student Design Competition are given the opportunity to show off their designs at this poster/ tabletop exhibition. The session 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 is 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 participate. Few restrictions are placed on the nature of the projects, which may include loudspeaker designs, DSP plug-ins, analog hardware, signal analysis tools, mobile applications, and sound synthesis devices. Attendees will observe new, original ideas implemented in working-model prototypes.

Saturday, October 11 9:00 am Room 405
Technical Committee Meeting on Sound for Digital Cinema & Television

Saturday, October 11 9:00 am Room 407
Standards Committee Meeting on Digital Audio Measurement Techniques

Recording and Production Session 3
Saturday, October 11 9:30 am – 11:00 am
Room 403 AB

RAW TRACKS: PET SOUNDS—A MASTER CLASS

Moderator: **Mark Rubel**, The Blackbird Academy, Nashville, TN, USA; Pogo Studio, Nashville, TN, USA

Panelist: **Mark Linett**

Renowned engineer/producer and three-time Grammy winner Mark Linett will discuss, analyze, and deconstruct track by track two songs from the Beach Boys seminal *Pet Sounds* album: “God Only Knows” and “Wouldn’t It Be Nice,” in the inaugural Raw Tracks series at the AES 137th International Convention in Los Angeles. Linett has worked extensively with The Beach Boys and Brian Wilson for over twenty-five years, including the first true stereo and 5.1 mix of *Pet Sounds* and has also worked with an array of artists including Jimi Hendrix, Randy Newman, Rickie Lee Jones, Jane’s Addiction, Eric Clapton, and The Red Hot Chili Peppers.

Saturday, October 11 10:00 am Room 405
Technical Committee Meeting on Coding of Audio Signals

Session P13 Saturday, Oct. 11
10:30 am – 12:00 noon Foyer 1

POSTERS: APPLICATIONS IN AUDIO—PART 1

10:30 am

P13-1 Automated Sound Optimization of Car Audio Systems Using Binaural Measurements and Parametric IIR Filters—*Friedrich von Tuerckheim, Tobias Münch*, Visteon Electronics Germany GmbH, Straubenhardt, Germany

Sound tuning is an important step towards improved listening conditions in car interiors. In most cases it is done manually by sound engineers. This paper presents an approach for fully automated sound optimization. In a first step, loudspeaker and interior responses are captured by averaged binaural measurements. Then, the resulting frequency response is matched to a given reference curve. As automotive head units often provide limited capacity for audio filters, a small set of second order recursive filters is used for equalization. Numerical optimization leads to a minimum error response while maintaining psychoacoustic specifications. The presented method is used for fast and efficient frequency response correction as well as for copying sound characteristics of different car interiors.
Convention Paper 9166

10:30 am

P13-2 Study of TV Sound Level Adjustment System for the Elderly with Speech Rate Conversion Function—*Tomoyasu Komori*,[1,2] *Atsushi Imai*,[3] *Nobumasa Seiyama*,[3] *Reiko Takou*,[3] *Tohru Takagi*,[3] *Yasuhiro Oikawa*[2]
[1]NHK Engineering System, Inc., Setagaya-ku, Tokyo, Japan
[2]Waseda University, Shinjuku-ku, Tokyo, Japan
[3]NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

Elderly viewers sometimes feel that background sound (music and sound effects) in TV programs is too loud, or that narration or speech is too fast to understand. That is why we have constructed a prototype system that compensates for both of these problems with sound on the receiver side. The results of evaluation experiments targeting elderly viewers showed that the use of this system could make it significantly easier to listen to TV sound. These results also showed that elderly viewers exhibit the “recruitment phenomenon.” They tend to select processing with a slowed speech rate that is easy to hear.
Convention Paper 9167

10:30 am

P13-3 Investigation of Gain Adjustment in a Personal Assistive Listening System Using Parametric Array Loudspeakers—*Santi Peksi*,[1] *Woon-Seng Gan*,[1] *Ee-Leng Tan*,[1] *Eu-Chin Ho*,[2] *Satya Vijay Reddy Medapati*[2]
[1]Nanyang Technological University, Singapore
[2]Tan Tock Seng Hospital, Singapore

Human hearing degrades with ages, which leads to difficulties in viewers of different age groups enjoying television together as they required different audio volumes. To address the problem Simon et al. [1] proposed loudspeaker arrays that boost 10 dB at all frequencies in a narrow spatial zone where hearing-impaired listener is located. This paper presents a different approach using a personal assistive listening (PAL) system that aims to deliver a highly directional sound beam with the required gain amplification through a parametric array loudspeaker to match the hearing profile of a hearing-impaired listener, while delivering normal sound loudness to the rest of normal listeners using conventional electro-dynamic loudspeakers. This paper investigates the gain adjustment of two commercially-available parametric loudspeakers over the frequency range for audiometry testing and relates the gain adjustments to the sound pressure level (SPL) at various positions away from the sound system.
Convention Paper 9168

10:30 am

P13-4 Cinema Sound Facility Design for Higher Education—*Robert Jay Ellis-Geiger*, City University of Hong Kong, Hong Kong, China

This paper is a narrative of the trials and tribulations that the author went through from design through to the commissioning of probably the most advanced higher education cinema sound facilities within the Asia-Pacific region. The facilities include a 7.1 THX and Dolby certified dubbing theatre, audio recording studio integrated into a 30-workstation audio/music technology lab, multiple 5.1 surround screening rooms, color correction, multi-format home entertainment environment and a large sound stage that can accommodate a full symphonic orchestra. The main purpose for the facilities were to support the delivery of undergraduate and post-graduate courses in sound, music, and audio within the academic studios of cinematic arts and animation and to establish a research center for cinema sound and music technology applications.
Convention Paper 9169

10:30 am

P13-5 A General-Purpose Decorrelator Algorithm with Transient Fidelity—*Ross Penniman*, University of Miami, Coral Gables, FL, USA

In a multichannel spatial audio presentation, a decorrelator is a signal-processing algorithm that helps to create a diffuse sensation for the listener by defeating any localization cues. In this paper the relevant psychoacoustic and signal processing principles are reviewed, and a new decorrelator algorithm is proposed that operates blindly on a single-channel input signal and creates a 5-channel decorrelated presentation. This algorithm uses transient extraction to achieve better fidelity when decorrelating a wide range of input signals. A subjective listening test compares the performance of the proposed algorithm in relation to two existing algorithms drawn from the literature. Results of the test are discussed as well as suggested improvements to the test methodology.
Convention Paper 9170

10:30 am

P13-6 Applicability of Perceptual Evaluation of Speech Quality in Evaluating Heavily Distorted Speech—*Mitsunori Mizumachi*, Kyushu Institute of Technology, Kitakyushu, Fukuoka, Japan

Speech quality assessment is indispensable to properly design a speech enhancement algorithm. The perceptual evaluation of speech quality (PESQ) is frequently employed as an objective speech distortion measure. The PESQ is a methodology for estimating subjective assessment of speech quality assuming a slight distortion caused by speech codecs for telephony systems. In case of noise reduction, however, a degree of speech distortion is heavier than those caused by the speech codecs. In this paper applicability of the PESQ is investigated for noisy and noise-reduced speech signals under severe noisy conditions. A relationship between PESQ scores and subjective mean opinion scores reveals that the PESQ can be applicable for heavily distorted speech only under non-stationary noisy conditions.
Convention Paper 9171

Broadcast/Streaming Media Session 10
Saturday, October 11 10:30 am – 12:00 noon
Room 408 A

COMPLIANCE WITH CALM ACT/PLOUD

Moderator: **Skip Pizzi**, NAB, Washington DC, USA

Presenters: *Florian Camerer*, ORF, Austrian TV, Vienna, Austria; EBU, European Broadcasting Union
Tim Carroll, Telos Alliance, Lancaster, PA, USA
Fadi Malek, DTS Inc.
Scott Norcross, Dolby Laboratories, San Francisco, CA, USA

Regulatory regimes or recommendations for control of television audio loudness are now well in place in the U.S. and Europe. Find out what these procedures entail, and learn the latest on implementation methods from top experts in the field, including some who were instrumental in creation of the governing documents.

Network Audio Session 9 Saturday, October 11
10:30 am – 12:00 noon Room 404 AB

USING AUDIO CONTENT OVER IP TECHNOLOGY IN PUBLIC RADIO

Chair: **Sherri Hendrickson**, Director Broadcast Media and Operations, American Public Media (APM)

Panelists: *Ian Adams,* System Administrator, American Public Media (APM), Los Angeles, CA, USA
Tom Nelson, Director Broadcast Engineering, American Public Media (APM), St. Paul, MN, USA

A recently completed refresh of American Public Media's Marketplace studios is focused on real-time collaboration via IP based signal distribution for audio, display and control. Legacy analog and digital cabling has been replaced by a multicast IP focused IT infrastructure connecting the main Los Angeles studios to APM's New York and Washington DC bureaus. *Marketplace Morning Report* and *Marketplace Money* are both hosted from New York but produced, directed and engineered from the control rooms in Los Angeles. This workshop will explore how audio over IP enables real time collaboration across locations for live broadcast.

Our three presenters will discuss

• The business case for this major investment and its impact on work flow and the user experience from the host, director and producers' perspective.

• The audio technologies employed and systems design from an audio signal routing and control perspective

• The underlying IT infrastructure which supports all this and how off-the-shelf network equipment is configured to support multicast distribution and provide transmission redundancy

Product Design Session 11 Saturday, October 11
10:30 am – 12:00 noon Room 402 AB

HOW TO DESIGN A ROBUST AUDIO PRODUCT FOR THE REAL WORLD

Presenter: **Bill Whitlock**, Whitlock Consulting, Oxnard, CA, USA

Testing the performance of audio products usually begins and ends in the manufacturer's test lab. Unfortunately, this testing doesn't always reveal problems that occur only when the product is installed in a real-world system. Such problems include hum, buzz, clicks, pops, and other unexplained misbehavior. Most can be traced to design issues, such as internal grounding schemes or conceptual misunderstandings of signal interface circuitry, that cost little or nothing to cure. These practical aspects of design are rarely taught in engineering schools but are precisely the focus of this tutorial. Learn what challenges your product will face in real-world systems and how to avoid the problems.

Project Studio Expo PSE8 PSE STAGE
CREATING A PROJECT STUDIO
Saturday, October 11, 10:45 am – 12:15 pm

Moderator: **John Storyk**, Architect, Studio Designer and Principal, Walters-Storyk Design Group, Highland, NY, USA

Panelists: *Nellie Barnett*, Singer, VO artist, actress, Los Angeles, CA, USA
PK Pandey, Director, GC Pro, Boston, MA, USA
Chris Pelonis, Musician, studio designer/owner, Santa Barbara, CA, USA

Part 1: How to create the best sounding, most ergonomically functional and aesthetically pleasing project studio possible on a given a budget (with questions from the audience).

Part 2: Addressing the needs and goals for low, medium, and high-budget studios. The discussion will encompass site selection, construction, acoustics, technology, neighbors, clients, and other issues encountered and overcome.

Live Sound Seminar 8 Saturday, October 11
11:00 am – 1:00 pm Room 406 AB

THE CUBE SPATIAL AUDIO RENDERER: THE NEXUS OF COLLABORATIVE PERFORMANCE AND LIVE SOUND FOR MULTIDISCIPLINARY RESEARCH

Presenters **Mike Roan**, Virginia Tech, Blacksburg, VA, USA
Tanner Upthegrove, Virginia Tech, Blacksburg, VA, USA

The Institute for Creativity, Arts, and Technology (ICAT) introduces a world-class collaborative research and performance sound instrument, The Cube Spatial Audio Renderer (CSAR). Housed in Virginia Tech's new Moss Arts Center, CSAR layers immersive sound technologies to create a unique, multi-user experience. CSAR is specifically designed for the Cube, a five-story, hybrid theater and research lab outfitted with hundreds of patchable AV and data panels. The primary 124.4 system produces Ambisonics, Wave Field Synthesis, and other spatial audio techniques. Nine ultrasonic beam speakers complement the primary system, four of which are on motorized pan-tilt arms. Eight relocatable, high-fidelity loudspeakers complete the system.

This unique facility will provide researchers with astonishing new capabilities in the fields of science, engineering, art, and design including Sound Field Synthesis, Auralization of Big Data, Human Perception, and Psychoacoustics.

The primary aesthetic goal for the Cube is to pursue advances in multichannel music and sound art for 3D audio. The methodologies will include the composition of new works for the Cube; the development of new software to facilitate work in the Cube; dissemination of research results through technical publications, software distribution, and concerts open to the public; and finally collaborations with other institutions that are working in 3D audio.

Game Audio Session 10 Saturday, October 11
11:00 am – 12:30 pm Room 408 B

BACK TO THE FUTURE: INTERACTIVE AUDIO IMPLEMENTATION TRENDS

Presenter: **Scott Selfon**, Microsoft, Redmond, WA, USA

"Next gen" has arrived, with ever increasing technical capabilities in both hardware and software processing. But are these new consoles really just offering the same thing, only more of it? What are the audio frontiers and barriers when so many of the restrictions of the past have been eliminated? And with maturing and increasingly sophisticated audio engine solutions, is game sound programming really now a "solved" problem? Scott reflects on the current state-of-the-art for areas ranging from spatial simulation and acoustic modeling to evolving and dynamic mixing, audio as a feedback mechanism, and highly personalized audio experiences. He'll use examples from both past and present to highlight the technical achievements that implementers are striving for in making audio not only compelling and realistic but an equal-footing contributor to immersive, engaging, and rewarding gameplay experiences.

Student Event/Career Development
EDUCATION AND CAREER/JOB FAIR
Saturday, October 11, 11:00 am – 12:30 pm
S-Foyer 2

The combined AES 135th 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 135th 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.

Saturday, October 11 11:00 am Room 407
Standards Committee Meeting on Metadata for Audio

Live Sound Expo LSE7 LSE STAGE
SOUND IN SPACE—THE BASICS REVISITED
Saturday, October 11, 11:00 am – 11:50 am

Presenter: **Bernie Broderick**, Technical Training Manager, EAW

Knowledge of the physics of sound—reflection, attenuation in air, the inverse square law—are as practical a part of the sound engineer's repertoire as are signal flow and a musical ear.

Tutorial 17 Saturday, October 11
11:15 am – 1:15 pm Room 409 AB

CONTEMPORARY APPROACHES TO PROGRAMMING DRUMS

Presenter: **Justin Paterson**, London College of Music, University of West London, London, UK

Drum programming has often faced boundaries in terms of how effectively it could address the complexities of certain genres. This tutorial will explore and push some of these boundaries as implemented in contemporary professional practice, showing contrasting techniques used in the creation of both human emulation and the unashamedly synthetic, across genres from Swing to Glitch. The session will include numerous live demonstrations covering a range of approaches using different DAWs and bespoke software using multi-touch techniques. Although introducing all key concepts from scratch, its range and hybridization should provide inspiration even for experienced practitioners.

This session is presented in association with the AES Technical Committee on Recording Technology and Practices

Session EB4 Saturday, Oct. 11
11:30 am – 1:00 pm Room 309

PAPER SESSION 2

Chair: **Eric Benjamin**, Surround Research, Pacifica, CA, USA

11:30 am

EB4-1 SoundWire: A New MIPI Standard Audio Interface—*Pierre-Louis Bossart*,[1] *Juha Backman*,[2] Jens Kristian Poulsen[3]
[1]Intel Corporation, Austin, TX, USA
[2]Microsoft, Espoo, Finland
[3]BlackBerry

This paper presents the key features of the upcoming SoundWire hardware interface and protocol. SoundWire is a robust, scalable, low complexity, low power, low latency, two-pin (clock and data) multi-drop bus that allows for the transfer of multiple audio streams and embedded control/commands. SoundWire provides synchronization capabilities and supports both PCM and PDM, multichannel data, isochronous and asynchronous modes. The development of this interface has generated a lot of interest and contributions from providers of audio peripherals (microphones, amplifiers and audio codecs), silicon vendors and OEMs. It will become a ratified standard by MIPI (Mobile Industry Processor Interface consortium) at the end of 2014. The first products should appear in 2015 and enable new usage models or design choices for audio applications.
Engineering Brief 172

11:45 am

EB4-2 Performance and Response: A Framework to Discuss the Quality of Audio Systems—*Ron Bakker*,[1] *Masahiro Ikeda*,[2] *Sungyoung Kim*[3]
[1]Yamaha Music Europe, Vianen, Netherlands
[2]Yamaha Corporation, Hamamatsu, Shizuoka, Japan
[3]Rochester Institute of Technology, Rochester, NY, USA

Recently we have designed a new networked digital audio system that is far more flexible yet complicated than before. A main design parameter was sound quality, focusing on cognitive judgment as well as physical quality since they both significantly influence the final sound quality evaluation. To adopt a human centric based approach for the design of a flexible networked digital audio system, the authors redefine the concept of sound quality assessment that classifies a system's overall quality into two attributes: a physical part as performance and a cognitive part as response.
Engineering Brief 173

12:00 noon

EB4-3 Analog Mixer Digitally Controlled via Plugin —*Vasin Limsukhawat*, Max Sound Contour, Denver, CO, USA; University of Colorado at Denver, Denver, CO, USA

The purpose of this project is to turn an old analog mixer to be more controllable which enhances the workflow to be more efficient. It consists of two major components: hardware and software. A microcontroller has been used to control the motorized potentiometer as well as transfer all the data from the mixer to the plugin or vice versa. This provides an ability to control the mixer via the AU plugin or manually adjust as well as save and recall presets.
Engineering Brief 175

12:15 pm

EB4-4 Performance of the Microphone-Preamp Interface—*Eric M. Benjamin*,[1] *Andrew Kimpel*[2]
[1]Surround Research, Pacifica, CA, USA
[2]Auralab, San Francisco, CA, USA

The microphone/preamp interface is very important but not as well understood as its importance would suggest. The signal level between the microphone and the preamp is at the lowest level it will ever be. Furthermore, any degradation or loss of quality can't be restored by later processing. The microphone, microphone cable, and preamplifier are made by different companies. The power for the microphone comes from the preamplifier. To optimally design the preamplifier it would be desirable to know the signal levels and the output impedance of the microphone. Also, the output impedance of the microphone affects the intrusion of external electrical noise into the system. A survey and study of the interface yields new insights into system performance.
Engineering Brief 176

12:30 pm

EB4-5 Removal of Partially Correlated Noise to Improve Signal to Noise Ratio—A Theoretical Study—*Simon Busbridge, Chris Garrett*, University of Brighton, Brighton, UK

Time correlation and decorrelation are well established tools to improve the signal to noise ratio of a system, yet they are often poorly understood. When several unwanted signals are correlated they are much easier to remove from uncorrelated wanted signals than vice versa, where a much poorer improvement is possible. A case in point is the removal of wind noise from microphone signals. The situation is further compounded when one or other of the signals is only partially correlated or different spectral content is differently correlated. This paper looks at the theoretical improvement in signal to noise ratio when either the signal or the noise are correlated to different degrees. Application to real signals and noise is discussed.
Engineering Brief 177

12:45 pm

EB4-6 Dynamic Audio Imaging In Radial Virtual Reality Environments—*Mark Blewett, James Pinkle; Bryan Dalle Molle*, University of Illinois at Chicago, Chicago, IL, USA

The CAVE2 is a large scale, 320 degree, 3D / 2D virtual reality environment featuring motion capture technology and located in the Electronic Visualization Laboratory at the University of Illinois at Chicago. The environment includes a 20.2 channel sound system controlled by a SuperCollider based audio server that is, in turn, controlled via the proprietary Omicron SoundAPI. Audio imaging for the system had operated with the assumption that the listener was stationary in the center of the CAVE2 and spatial extent was not dynamically altered. Our team added functionality to maintain audio imaging, including source position and width, as a tracked user and sound object move relative to one another within the environment.
Engineering Brief 162

Special Event
MUSIC AND AUDIO FOR THE SMALLER SCREEN
Saturday, October 11, 11:30 am – 12:30 am
Room 304 AB

Moderator: **Jerome Rossen**, Freshmade Music, San Francisco, CA, USA

Presenters: *Steve Horowitz*, Game Audio Institute, San Francisco, CA, USA
Richard Warp, Manhattan Producers Alliance, San Francisco, CA, USA-Leapfrog Enterprises Inc., Emeryville, CA, USA

What are the important issues to take into account when you're composing, compiling, and refining your masterpiece for the small screen? What should you prioritize during preproduction? How does the smaller screen affect your creative decision making? How can you mix for success? What do you need to know if you're creating for iOS, Android and the Web? Join members of the Manhattan Producers Alliance as they conduct this panel addressing how to make the best possible audio for the "smaller screen."

Broadcast/Streaming Media Session 11
Saturday, October 11 12:00 noon – 1:30 pm
Room 408 A

SBE/TROUBLESHOOTING AND MAINTENANCE OF EQUIPMENT

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

Panelists: *John Bisset*, Telos Alliance
Bill Sacks, Orban / Optimod Refurbishing, Hollywood, MD, USA
Kimberly Sacks, CBS Radio, Hollywood, MD, USA
Joe Talbot, Telos Alliance

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 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.

This is jointly presented by the Society of Broadcast Engineers.

This session is open to all badges.

Recording and Production Session 4
Saturday, October 11 12:00 noon – 1:30 pm
Room 404 AB

MASTER MISSION: SPREADING THE WORD ON MENTORING, FORMATS, AND A CHANGING INDUSTRY

Moderator: **Tom Kenny**

Panelists: *Gavin Lurssen*, Lurssen Mastering, Los Angeles, CA, USA
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA
Joe Palmaccio, The Place . . . For Mastering, Nashville, TN, USA
Michael Romanowski, Michael Romanowski Mastering, San Francisco, CA, USA; Owner Coast Recorders

A discussion with multiple award-winning Mastering engineers on the state of the music industry from a mastering engineers position. They will be discussing the changes in the industry and how the model is changing with manufacturing and online file distribution. Topics also included will be Meta-Data, High resolution, streaming, Singles vs. Albums, LPs.

Saturday, October 11 12:00 noon Room 405
Technical Committee Meeting on High Resolution Audio

Live Sound Expo LSE8 LSE STAGE
DIGITAL CONSOLES—WHAT'S INSIDE?
Saturday, October 11, 12:00 noon – 12:50 pm

Moderator: **Mark Frink**

Presenters: *Antony David*, Managing Director Solid State Logic
Tom Der, US Brand Manager-Soundcraft, Live Sound Product Specialist-Studer
Matt Larson, National Sales Manager, DiGiCo, XTA, MC2
Marc Lopez, Marketing Manager, Yamaha Commercial Audio Systems
Ray Tantzen, Product Manager StudioLive RM-series mixers - PreSonus

What differentiates digital consoles in terms of sonic signatures? Our panel will discuss analog inputs, processing algorithms and the available approaches to DSP hardware.

Network Audio Session 10 Saturday, October 11
12:15 pm – 1:15 pm Room 308 AB

CANCELED

Network Audio Session 3 Saturday, October 11
12:15 pm – 1:45 pm Room 308 AB

USING AES67 NETWORKING—PRACTICAL ISSUES IN AES67 DEPLOYMENT

Chair: **Andreas Hildebrand**, ALC NetworX GmbH, Munich, Germany

Panelists: *Landon Gentry*, Audinate, Portland, OR, USA Sydney, Australia
Kevin Gross, AVA Networks, Boulder, CO, USA
Gints Linis, University of Latvia – IMCS, Riga, Latvia
Greg Shay, The Telos Alliance, Cleveland, OH, USA

The AES67 standard provides comprehensive interoperability recommendations for professional audio over IP networks in the areas of synchronization, media clock identification, network transport, encoding and streaming, session description, and connection management.

This workshop will discuss the practical issues that will arise when AES67 is deployed in small- and large-scale installations, including physical media, switches and routers, and timing and latency.

Historical Event H4
HISTORY OF AUDIO MEASUREMENT TECHNOLOGY
Saturday, October 11, 12:15 pm – 1:15 pm
Room 402 AB

Presenter: **John Murray**, Optimum System Solutions, Woodland Park, CO, USA

Starting with early CRT real-time analyzers and strip-chart recorders, moving through LED RTAs to time-gated direct sound and windowing methodologies, to the full dual FFT era, including both dedicated boxes as well as software cards and programs, this presentation will track the historical development of audio measurement systems for sound reinforcement. Systems to be mentioned in the timeline will include HP, Altec, Amber, Arta, Audio Control, Audio Precision (Tektronix), Barclay Badap, B&K, Crown BDP-2 & TEF, General Radio, Goldline, Ivie, Meyer SIM, Neptune (NEI), SIA SMAART, Spectra Foo, UREI Sonipulse, UREI 200/2000 w/HP 7010B, and White Instruments.

Perspective on the practical application and technical progression of these types of measurement systems will be shared. Insight will be presented as to how the advancement of the technology fostered the expansion of knowledge for both the practitioners of system design and tuning, and for the developers of new loudspeaker/processing products. The presenter, John Murray, has made a life-long study of sound-system equalization and the tools used for it. Having experience in all sides of the sound reinforcement industry, his career has been greatly influenced by these measurement systems.

Saturday, October 11 12:30 pm Room 407
Standards Committee Meeting on Loudspeaker Modeling and Measurement

Project Studio Expo PSE9 PSE STAGE
MIXING SECRETS: PRODUCTION TRICKS TO USE WITH ANY DAW
Saturday, October 11, 12:30 pm – 1:15 pm

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.

Game Audio Session 11 Saturday, October 11
1:00 pm – 3:00 pm Room 306 AB

ADVENTURES IN MUSIC AND SOUND DESIGN —THE WORLD OF HOHOKUM

Presenters: **Daniel Birczynski**, Sony Computer Entertainment America
David Collins, Sony Computer Entertainment America
Mike Niederquell, Sony Computer Entertainment America

Accompanying the vibrant visuals in Hohokum is a lush soundtrack and highly interactive audio environment that brings audio to the forefront of this title. In this workshop, the senior staff of the audio team talks about storytelling through their creative use of sound design and music.

Special Event
CHICKS IN THE MIX
Saturday, October 11, 1:00 pm – 2:30 pm
Room 403 AB

Moderator: **Chris Lord-Alge**, Multi-GRAMMY Award-winning producer/mixer

Panelists: *Marcella Araica*, Recording & Mix engineer (Britney Spears, Madonna, Pink)
Amy Burr, Larrabee Sound Studios, Studio Manager
Emily Lazar, September Mourning (Lead Vocalist for the band)
Lisa Loeb, GRAMMY-nominated Singer/Songwriter
Brenda Russell, BMPR, Los Angeles, CA, USA

Historically the audio and recording industries has been a male dominated workplace. Whether you are a female engineer, product designer, producer, mixer, manager, songwriter or artist, there are challenges you face on a day-to-day basis that your male counterparts do not. Our panel of well-known female industry professionals discuss what it takes to survive and thrive in what James Brown called “It’s A Man’sWorld.” Moderating this panel is the industry’s most testosterone driven, turn it up to 11, egotistical person AES could find—multi-GRAMMY Award-winning producer/mixer, Chris Lord-Alge whose resume includes recordings for Muse, Pink, Foo Fighters, Avril Lavigne, Green Day, Daughtry, Paramore, and Black Eyed Peas. Whether you are just starting out in the industry or are a seasoned professional, Chicks In The Mix brings together some of the most powerful and successful women to discuss what and how they do what they do. This dynamic panel, debuting at the AES137 Convention, is set to offer lively debate and discussion across all boundaries. No matter your gender, you will want to hear what these Chicks in the Mix have to say.

Special Event
THE FUTURE IS NOW: MIND CONTROLLED INTERACTIVE MUSIC
Saturday, October 11, 1:00 pm – 2:30 pm
Room 304 AB

Presenters: **Scott Looney**
Tim Mullen
Richard Warp, Manhattan Producers Alliance, San Francisco, CA; Leapfrog Enterprises Inc., Emeryville, CA, USA

If one thing is clear from the music industry over the last 20 years, it is that consumers are seeking an ever-more immersive experiences, and in many ways bio feedback is the "final frontier," where music can be made in reaction to emotions, mood and more. 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, are here to stay. These technologies are already robust enough to be integrated into everything from single interfaces to complete systems.

Student Event/Career Development
STUDENT RECORDING CRITIQUES
Saturday, October 11, 1:00 pm – 2:00 pm
Room 305

Moderator: **Ian Corbett**

Students can bring stereo or surround mixes to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Sign-up for time slots is immediately after the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on CD, DVD, memory-stick, or hard disc, as clearly labeled 44.1 KHz WAVE or AIFF files. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers! (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.)

Saturday, October 11 1:00 pm Room 405
Technical Committee Meeting on Automotive Audio

Live Sound Expo LSE9 LSE STAGE
THE SMALL VENUE MONITOR MIX
Saturday, October 11, 1:00 pm – 1:50 pm

Presenter: **Jason Spence**, President-J Sound Services, Monitor Engineer (Keith Urban, Megadeth, CMT and CMA Awards

Amps on stage, wedges in front of every musician, a drummer who could fill a domed stadium with sound without amplification—what’s a sound engineer to do to keep the musicians happy without destroying the house mix? A practical approach will be presented, including the diplomacy involved in telling musicians to “turn it down” and the alternative of personal monitoring.

Broadcast/Streaming Media Session 12
Saturday, October 11 1:30 pm – 3:00 pm
Room 408 A

UNDERSTANDING AUDIO PROCESSING—
HOW TO USE THE AUDIO PROCESSOR

Chair: **Tracy Teagarden**, CBS Radio

Panelists: *Sunil G. Bharitkar*, Dolby Laboratories, San Francisco, CA, USA
Tim Carroll, Telos Alliance, Lancaster, PA, USA
Frank Foti, Telos, New York, NY, USA
Jean-Marc Jot, DTS, Inc., Los Gatos, CA, USA
Jeff Keith, Wheatstone Corporation, New Bern, NC, USA
Greg Ogonowski, Orban, San Leandro, CA, USA
Robert Orban, Orban, San Leandro, CA, USA

What is the audio processor? What is it used for? How do we use it? Is it the solution for bad levels, bad mixes, bad audio? Will it save the world?

Saturday, October 11 1:30 pm Room 401
Historical Committee Meeting

Project Studio Expo PSE13 PSE STAGE
LISTEN UP, AND LEARN!—TRACK 2
Saturday, October 11, 1:30 pm – 2:15 pm

Presenters: **Alex Case**, University of Massachusetts Lowell, Lowell, MA, USA
Stephen Webber, Berklee College of Music, Valencia, Spain

Bring your ears, your artistry, and your opinions for an hour dedicated to the art of listening. Guided by your hosts, Stephen Webber and Alex U. Case, you'll focus on another iconic record that is a proven success—artistically and commercially—and glean useful aural insights. We'll listen as producers, engineers, composers, performers, and music fans, analyzing the elements that contribute to the work's success. You'll gain a deeper appreciation of this recording. More importantly, you'll be inspired to approach your own work in new ways. Most importantly, you'll get an up-close view into how experienced audio engineers break down what they hear, empowering you to keep learning whenever you listen.

Session P14 Saturday, Oct. 11
2:00 pm – 5:30 pm Room 308 AB

PERCEPTION—PART 2

Chair: **Sungyoung Kim**, Rochester Institute of Technology, Rochester, NY, USA

2:00 pm

P14-1 Revision of Rec. ITU-R BS.1534—*Judith Liebetrau,[1] Frederik Nagel,[2] Nick Zacharov,[3] Kaoru Watanabe,[4] Catherine Colomes,[5] Poppy Crum,[6] Thomas Sporer,[1] Andrew Mason[7]*
[1]Fraunhofer IDMT, Ilmenau, Germany
[2]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[3]DELTA SenseLab, Iisalmi, Finland
[4]NHK Broadcasting Center, Setagaya-ku, Tokyo, Japan
[5]Orange Labs, Cesson Sevigné, France
[6]Dolby Laboratories, San Francisco, CA, USA
[7]BBC Research and Development, London, UK

In audio quality evaluation, ITU-R BS.1534-1, commonly known as MUSHRA, is widely used for the subjective assessment of intermediate audio quality. Studies have identified limitations of the MUSHRA methodology [1][2], which can influence the robustness to biases and errors introduced during the testing process. Therefore ITU-R BS.1534 was revised to reduce the potential for introduction of systematic errors and biases in the resulting data. These modifications improve the validity and the reliability of data collected with the MUSHRA method. The main changes affect the post screening of listeners, the inclusion of a mandatory mid-range anchor, the number and length of test items as well as statistical analysis. In this paper the changes and reasons for modification are given.
Convention Paper 9172

2:30 pm

P14-2 Movement Perception of Risset Tones with and without Artificial Spatialization—*Julian Villegas*, University of Aizu, Aizu Wakamatsu, Fukushima, Japan

The apparent radial movement (approaching or receding) of Risset tones was studied for sources in front, above, and to the right of listeners. Besides regular Risset tones, two kinds of spatialization were included: global (regarding the tone as a whole) and individual (spatializing each of its spectral components). The results suggest that regardless of the direction of the glissando, subjects tend to judge them as approaching. The effect of spatialization type was complex: For upward Risset tones, judgments were, in general, aligned with the direction of the spatialization, but this was not observed in the downward Risset tones. Furthermore, individual spatialization yielded judgments comparable to those of non-spatialized stimuli, whereas spatializing the stimuli as a whole yielded judgments more aligned with the treatment.
Convention Paper 9173

3:00 pm

P14-3 The Audibility of Typical Digital Audio Filters in a High-Fidelity Playback System—*Helen M. Jackson, Michael D. Capp, J. Robert Stuart*, Meridian Audio Ltd., Huntingdon, UK

This paper describes listening tests investigating the audibility of various filters applied in high-resolution wideband digital playback systems. Discrimination between filtered and unfiltered signals was compared directly in the same subjects using a double-blind psychophysical test. Filter responses tested were representative of anti-alias filters used in A/D (analog-to-digital) converters or mastering processes. Further tests probed the audibility of 16-bit quantization with or without a rectangular dither. Results suggest that listeners are sensitive to the small signal alterations introduced by these filters and quantization. Two main conclusions are offered: first, there exist audible signals that cannot be encod- ➥

ed transparently by a standard CD; and second, an audio chain used for such experiments must be capable of high-fidelity reproduction.
Convention Paper 9174

3:30 pm

P14-4 Evaluation Criteria for Live Loudness Meters —*Jon Allan, Jan Berg*, Luleå University of Technology, Piteå, Sweden

As a response to discrepancies in loudness levels in broadcast, the recommendations of the International Telecommunication Union and the European Broadcasting Union state that audio levels should be regulated based on loudness measurement. These recommendations differ regarding the definition of meter ballistics for live loudness meters, and this paper seeks to identify possible additional information needed to attain a higher conformity between the recommendations. This work suggests that the qualities we seek in a live loudness meter could be more differentiated for different time scales (i.e., momentary and short-term that is defined by two different integration times), and therefore also should be evaluated by different evaluation criteria.
Convention Paper 9175

4:00 pm

P14-5 Factors Influencing Listener Preference for Dynamic Range Compression—*Malachy Ronan, Robert Sazdov, Nicholas Ward*, University of Limerick, Limerick, Ireland

The introduction of loudness normalization has led some commentators to declare that the loudness wars are over. However, factors contributing to a preference for dynamic range compression have not been removed. The research presented here investigates the role of long-term memory in sound quality judgments. Factors influencing preference judgments of dynamic range compression are discussed along with suggestions of further research areas. Research is presented that indicates that an objective measure of dynamic range will facilitate a greater understanding of how dynamic range compression affects individual sound quality attributes.
Convention Paper 9176

4:30 pm

P14-6 The Influence of Listeners' Experience, Age, and Culture on Headphone Sound Quality Preferences—*Sean Olive, Todd Welti, Elisabeth McMullin*, Harman International, Northridge, CA USA

Double-blind headphone listening tests were conducted in four different countries (Canada, USA, China, and Germany) involving 238 listeners of different ages, gender, and listening experiences. Listeners gave comparative preference ratings for three popular headphones and a new reference headphone that were all virtually presented through a common replicator headphone equalized to match their measured frequency responses. In this way, biases related to headphone brand, price, visual appearance, and comfort were removed from listeners' judgment of sound quality. On average, listeners preferred the reference headphone that was based on the in-room frequency response of an accurate loudspeaker in a reference listening room. This was generally true regardless of the listeners' experience, age, gender, and culture. This new evidence suggests a headphone standard based on this new target response would satisfy the tastes of most listeners.
Convention Paper 9177

5:00 pm

P14-7 A Hierarchical Approach to Archiving and Distribution—*J. Robert Stuart*,[1] *Peter Craven*[2]
[1]Meridian Audio Ltd., Huntingdon, UK
[2]Algol Applications Ltd., London, UK

When recording, the ideal is to capture a performance so that the highest possible sound quality can be recovered from the archive. While an archive has no hard limit on the quantity of data assignable to that information, in distribution the data deliverable depends on application-specific factors such as storage, bandwidth or legacy compatibility. Recent interest in high-resolution digital audio has been accompanied by a trend to higher and higher sampling rates and bit depths, yet the sound quality improvements show diminishing returns and so fail to reconcile human auditory capability with the information capacity of the channel. By bringing together advances in sampling theory with recent findings in human auditory science, our approach aims to deliver extremely high sound quality through a hierarchical distribution chain where sample rate and bit depth can vary at each link but where the overall system is managed from end-to-end, including the converters. Our aim is an improved time/frequency balance in a high-performance chain whose errors, from the perspective of the human listener, are equivalent to no more than those introduced by sound traveling a short distance through air.
Convention Paper 9178

Session P15 **Saturday, Oct. 11**
2:00 pm – 5:00 pm **Room 309**

SIGNAL PROCESSING—PART 1

Chair: **Jayant Datta**, THX Ltd., San Francisco, CA, USA

2:00 pm

P15-1 MATLAB Program for Calculating the Parameters of Autocorrelation and Interaural Cross-Correlation Functions Based on a Model of the Signal Processing Performed in the Auditory Pathways —*Shin-ichi Sato*,[1] *Alejandro Bidondo*,[1] *Yoshiharu Soeta*[2]
[1]Universidad Nacional de Tres de Febrero, Caseros, Buenos Aires, Argentina
[2]National Institute of Advanced Industrial Science and Technology (AIST), Ikeda, Japan

This paper describes a MATLAB program with a graphical user interface (GUI) for a signal processing based on the Auditory Image Model [S. Bleeck et al., Acta Acustica united with Acustica, 90 (2004) 781–787], followed by the summary autocorrelation function (SACF) and the summa-

ry interaural cross-correlation function (SIACF) analyses, and the calculation of the SACF and SIACF parameters. The effects of the number of the channels and the frequency range of the filterbanks on the SACF parameters are investigated.
Convention Paper 9179

2:30 pm

P15-2 An Investigation of Temporal Feature Integration for a Low-Latency Classification with Application to Speech/Music/Mix Classification—*Joachim Flocon-Cholet*,[1] *Julien Faure*,[1] *Alexandre Guérin*,[1] *Pascal Scalart*[2]
[1]Orange Labs, Lannion, France
[2]INRIA/IRISA, Université de Rennes, Rennes, France

In this paper we propose several methodologies for the use of feature integration and evaluate them in a low-latency classification framework. These general methodologies are based on three key aspects that will be assessed in this study: the selection of the features that have to be temporally integrated, the choice of the integration techniques, i.e., how the temporal information is extracted, and the size of the integration window. The experiments carried out for the speech/music/mix classification task show that the different methodologies have a significant impact on the global performance. Compared to the state of the art procedures, the methodologies we proposed achieved the best performance, even with the low-latency constraints.
Convention Paper 9180

3:00 pm

P15-3 MATLAB Program for Calculating the Parameters of the Autocorrelation and Interaural Cross-Correlation Functions Based on Ando's Auditory-Brain Model—
Shin-ichi Sato, Universidad Nacional de Tres de Febrero, Caseros, Buenos Aires, Argentina

This paper describes a MATLAB program with a graphical user interface (GUI) to calculate the parameters of the autocorrelation and the interaural cross-correlation functions of a binaural signal based on the auditory-brain model proposed by Ando [Y. Ando. (1998) Architectural Acoustics: Sound Source, Sound Fields, and Listeners, Springer-Verlag, New York, Chap. 5], which can describe the various subjective attributes such as pitch, timbre, and spatial impression.
Convention Paper 9181

3:30 pm

P15-4 Perceptual Quality of Audio Separated Using Sigmoidal Masks—*Toby Stokes*,[1] *Christopher Hummersone*,[1] *Tim Brookes*,[1] *Andrew Mason*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]BBC Research and Development, London, UK

Separation of underdetermined audio mixtures is often performed in the Time-Frequency (TF) domain by masking each TF element according to its target-to-mixture ratio. This work uses sigmoidal functions to map the target-to-mixture ratio to mask values. The series of functions used encompasses the ratio mask and an approximation of the binary mask. Mixtures are chosen to represent a range of different amounts of TF overlap, then separated and evaluated using objective measures. PEASS results show improved interferer suppression and artifact scores can be achieved using softer masking than that applied by binary or ratio masks. The improvement in these scores gives an improved overall perceptual score; this observation is repeated at multiple TF resolutions.
Convention Paper 9182

4:00 pm

P15-5 A New Approach to Impulse Response Measurements at High Sampling Rates—
Joseph G. Tylka, Rahulram Sridhar, Braxton B. Boren, Edgar Choueiri, Princeton University, Princeton, NJ, USA

High sampling rates are required to fully characterize some acoustical systems, but capturing the system's high-frequency roll-off decreases the signal-to-noise ratio (SNR). Band-pass filtering can improve the SNR but may create an undesirable pre-response. An iterative procedure is developed to measure impulse responses (IRs) with an improved SNR and a constrained pre-response. First, a quick measurement provides information about the system and ambient noise. A second, longer measurement is then performed, and a suitable band-pass filter is applied to the recorded signal. Experimental results show that the proposed procedure achieves an SNR of 37 dB with a peak pre-response amplitude of <0.2% of the IR peak, whereas a conventional technique achieves an SNR of 32 dB with a peak pre-response amplitude of 16%.
Convention Paper 9183

4:30 pm

P15-6 IIR Filters for Audio Test and Measurement: Design, Implementation, and Optimization—
Thomas Kite, Audio Precision, Inc., Beaverton, OR, USA

Audio analyzers use filters for many reasons: to define the measurement bandwidth, to isolate tones for measurement, to remove fundamental signals, and so on. In modern instruments, the majority of this filtering is done digitally, following analog-to-digital conversion if the signal is not already digital. Digital filter design is a mature field that encompasses a broad range of techniques, from classical analog filter design to advanced iterative design methods. However, the filter design considerations and techniques unique to audio analyzers do not seem to occupy much space in the published literature. This paper aims to correct this with a discussion of filter design, implementation, and optimization for modern Intel x86 architectures.
Convention Paper 9184

Workshop 5 **Saturday, October 11**
2:00 pm – 3:30 pm **Room 409 AB**

HOW ARE WE LEARNING MASTERING, TEACHING MASTERING—THE NEXT WAVE

Chair: **Jonathan Wyner**, Berklee College of Music, ➥

Boston, MA, USA; M Works Mastering

Panelists: *Scott Hull*, Masterdisk, New York, NY, USA
Eric Boulanger, The Mastering Lab, Ojai, CA, USA
Mike Wells, Mike Wells Mastering, Los Angeles, CA, USA

Traditionally mastering has been learned by apprenticing. Now with the proliferation of educational resources and the evolution of affordable hi quality in-the-box processing, more people are practicing mastering in more places than ever before. Teaching a young engineer to become a top flight mastering engineer can be challenging. Have you wondered: What does “Experienced Mastering Engineer” mean? What's the secret of mastering? In this workshop, seasoned mastering engineers and educators discuss how the craft is being taught and learned and how the next generation of mastering engineers will learn from their contemporaries. Topics will include what time tested practices remain essential and what is new in the discipline of mastering. Attendees of this workshop will walk away with a clearer understanding of what it takes to thrive in today's mastering market, how to assess internship/mentorship over “going solo” early in a mastering career, and how to grow/build your mastering skills in today's market.

Tutorial 18 — Saturday, October 11
2:00 pm – 4:00 pm — Room 406 AB

ACOUSTICAL OUTPUT-BASED EVALUATION OF SOUND SYSTEM EQUIPMENT

Presenter: **Wolfgang Klippel**, Klippel GmbH, Dresden, Germany

This tutorial session will cover best practices for loudspeaker measurements. It is critical for product development and component selection to know the response of loudspeaker systems and components with reasonable accuracy in order to make informed decisions based on comparisons of data. In this session we will briefly cover the basics of FFT-based measurement systems before moving on to additional topics.

- Averaging and S/N Windowing (both signal acquisition and impulse response windowing)
- Ground plane measurement techniques
- Directivity measurements
- Maximum input voltage measurements
- Impedance
- Alignment of pass bands

Traditional loudspeaker measurements as described in the IEC standard 60268-5 require access to the loudspeaker terminals and use the voltage and electrical input power for assessing the sensitivity, power handling, frequency response and other meaningful characteristics. These standards cannot be applied to active sound systems where signal processing, amplification and passive transducers are combined to one physical unit and the audio signal is supplied by wave-file, wireless transmission or in any other digital format.

This tutorial gives a review on the current standard activities in AES, IEC and other committees to develop output-based evaluation techniques applicable to all kinds of loudspeakers and sound reproduction systems. The maximum sound pressure level SPLmax as rated by the manufacturer is not only a meaningful characteristic for the end user but also the basis for calibrating the stimulus provided by any input channel. Sinusoidal chirp signals, multi-tone, burst and other modern test signal require standardization to ensure comparability of the measurement data. Acoustic simulation of loudspeaker-room interaction, the progress in 3D sound reproduction systems and growing importance of personal audio requires comprehensive data describing the near and far field of the source. There is also a need to develop standardized measurement techniques for detecting impulsive distortion generated by voice coil rubbing, loose particles, air leakage and other loudspeaker defects which have a high impact on sound quality.

Network Audio Session 11 — Saturday, October 11
2:00 pm – 3:00 pm — Room 404 AB

HOW STANDARDIZATION HAS BENEFITED OUR INDUSTRY AND HOW A COMMAND AND CONTROL STANDARD CAN GENERATE GROWTH AND INNOVATION

Presenter: **Ethan Wetzell**

This session will look at the impact that standards have had on the industry in shaping technological and application innovations. We will take a look back on how industries have evolved due to the benefits and proliferation of standards that have enabled interoperability and look forward to how emerging standards will create new innovations and opportunities.

Product Design Session 12 — Saturday, October 11
2:00 pm – 3:30 pm — Room 402 AB

PRODUCT DESIGN FOR STUDENTS

Presenters: **Alex Ruthmann**, New York University, New York, NY, USA
Jay LeBoeuf, RealIndustry.org, San Francisco, CA, USA

The process of bringing an audio product or service idea from ideation to mass commercialization is a complex one whether you’re working as an entrepreneur in the context of a start-up, or as a member of a development team within a large company. This session brings together leaders and mentors from across the audio product design and development process who will share their insights into what they wished they had known as students. We’ll focus on the multiple roles, day to day responsibilities, skillsets, and inner working of how today's leading companies develop products for the audio market.

Sound for Picture 3 — Saturday, October 11
2:00 pm – 4:00 pm — Room 408 B

CINEMA STANDARDS REVIEW AND THE ROAD TO IMPLEMENTING CHANGES (A JOINT PRESENTATION WITH THE SMPTE)

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

Panelists: *David Murphy*, Krix Loudspeakers, Hackham, South Australia
Neil Shaw, Menlo Scientific Acoustics Inc., Topanga, CA, USA
Brian Vessa, Sony Pictures Entertainment, Culver City, CA, USA; Chair SMPTE 25CSS standards committee

The AES and SMPTE are pleased to present this joint workshop on audio standards issues in the cinema.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Saturday, October 11 2:00 pm Room 405
Technical Committee Meeting on Audio Forensics

Live Sound Expo LSE10 LSE STAGE
HOW'D THEY GET THAT SOUND?
— INSTRUMENT SPECIFIC PROCESSING
Saturday, October 11, 2:00 pm – 2:50 pm

Presenters: **Derek Brener**, Bruno Mars
Matt Larson, National Sales Manager, DiGiCo, XTA, MC2
Kevin Madigan, FOH for Crosby, Stills and Nash

With digital consoles, a wealth of processing is available for even the smallest gig. Getting that big acoustic guitar sound, a tight bass attack, drums with punch—specific approaches to reach common goals will be shared, along with an audience Q&A.

Project Studio Expo PSE11 PSE STAGE
THE SPECIAL SAUCE FOR MIXING A HIT RECORD
Saturday, October 11, 2:30 pm – 3:45 pm

Presenters: **Fab Dupont**
Mick Guzauski

Producer Fab Dupont (Mark Ronson, Jennifer Lopez) talks with esteemed mix engineer, Mick Guzauski (Daft Punk, Pharrell) as they walk through one of today's hottest tracks. Hear how the pros approach crafting a hit with the same tools available to you and what that "special sauce" is too.

Session P16 Saturday, Oct. 11
3:00 pm – 4:30 pm Foyer 1

POSTERS: APPLICATIONS IN AUDIO—PART 2

3:00 pm

P16-1 General Volterra and Swept-Sine Diagonal System Estimation and Modeling Performance **—***Russell H. Lambert*, Harman International, South Jordan, UT, USA

Volterra system modeling performance results are given for various scenarios using both fully-determined and under-determined models. Three nonlinear system estimation methods are presented and compared including a novel and efficient Farina-type Hammerstein algorithm. The diagonal-only Hammerstein methods will not model off-diagonal nonlinear energy for general uncorrelated inputs but will model correlated inputs to some degree. The data matrix estimation methods work for generic input signal types. The nonlinear system must be fully determined to yield best results, but the diagonal-only models are more practical for applications having significantly long memory channels.
Convention Paper 9185

3:00 pm

P16-2 Downward Compatibility Configurations when Using a Univalent 12 Channel 3D Microphone Array Design as a Master Recording Array— *Michael Williams*, Sounds of Scotland, Le Perreux sur Marne, France

It can be shown that Microphone Array Design applied to a 12-channel 3D microphone array can create a master recording array design that will generate downward compatible signals that satisfy most of the present-day univalent lower order channel/loudspeaker configurations. The implementation of this compatibility oriented array design requires no matrixing or processing of the channel signals, while still maintaining the integrity of the overall sound field architecture. This compatibility approach to 3D array design produces a master recording system that can be adopted for an overall production, eventually to be distributed using several different media formats (stereo, DVD, Blu-ray, 3D, etc.). However this approach can also be used as a consumer choice function within a global master recording or file downloading facility.
Convention Paper 9186

3:00 pm

P16-3 Relative Influence of Spectral Bands in Horizontal Front Localization of White Noise **—***Tomomi Sugasawa, Jie Huang, Julian Villegas*, University of Aizu, Aizu Wakamatsu, Fukushima, Japan

The relationship between horizontal-front localization and the energy in different spectral bands is investigated in this paper. Specifically, we tried to identify which spectral regions produced changes in the judgments of the position of a white noise, when each band was removed from the noise presented through a front loudspeaker and presented via side loudspeakers. These loudspeakers were set at left and right from the front-midsagittal plane of the listener. Participants were asked to assess whether the noise was coming from the front loudspeaker as bands were moved from front to side loudspeakers. Results from a pilot study suggested differences in the relative importance of spectral bands for horizontal-front localization.
Convention Paper 9187

3:00 pm

P16-4 Acoustic Digital Communication for Identification Systems—*Sergio Vazquez*, Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina

In this paper a secure, low cost, energy efficient, robust digital communication system for identification purposes is presented. An alternative to magnetic stripe cards, Bluetooth LE, and NFC (Near Field Communication) that requires no specific hardware, making it compatible with almost every smartphone or portable device that has a working loudspeaker and is capable of reproducing audio. While previous works on the subject established the possibility to transmit digital data over the air using acoustic waves, this paper focuses on its implementation.
Convention Paper 9188
[Paper was not presented but is available for purchase in the E-Library]

3:00 pm

P16-5 How Critical Listening Exercises Complement Technical Courses to Effectively Provide Audio Education for Engineering Technology Students—*Mark J. Indelicato, Clark Hochgraf, Sungyoung Kim*, Rochester Institute of Technology, Rochester, NY, USA

Music is important to many aspects of our lives including student life at an institution of higher education. Combining music with academic coursework and programs therefore can be an effective way of engaging students to embrace academic programs and be successful in higher education. Some institutions have purposely incorporated audio engineering into technical programs as a way to not only create interest but to increase retention. Others have created music and technology programs and options, leveraging the student passion for music and keen interest in engineering. This paper discusses the benefit of combining music, technology, and engineering into higher education and, in particular, how the development of critical listening skills is key to the success of such a curriculum.
Convention Paper 9189

3:00 pm

P16-6 On the Acoustics of Alleyways—*Regina E. Collecchia,*[1] *Jonathan S. Abel,*[1] *Sean Coffin,*[1] *Eoin Callery,*[1] *Yoo Hsiu Yeh,*[1] *Kyle Spratt,*[2] *Julius O. Smith, III*[1]
[1]Stanford University, Stanford, CA, USA
[2]University of Texas, Austin, Austin, TX, USA

Alleyways bounded by flat, reflective, parallel walls and smooth concrete floors can produce impulse responses that are surprisingly rich in texture, featuring a long-lasting modulated tone and a changing timbre, much like the sound of a didgeridoo. This work explores alleyway acoustics with acoustic measurements and presents a computational model based on the image method. Alleyway response spectrograms show spectral zeros rising in frequency with time and a modulated tone lasting noticeably longer than the harmonic series associated with the distance between the walls. With slight canting of the walls and floors to produce the long lasting modulated tone, the image method model captures much of this behavior.
Convention Paper 9190

Broadcast/Streaming Media Session 13
Saturday, October 11 3:00 pm – 5:00 pm
Room 408 A

SMPTE: AUDIO ISSUES FOR LIVE TELEVISION—OVERCOMING THE CHALLENGES OF LIVE TELEVISION BROADCAST IN TODAY'S WILD, WILD WORLD

Chair: **Roger Charlesworth**, DTV Audio Group

Panelist: *Michael Abbott*, All Ears Inc.
Bruce Arledge Jr.
Kevin R. Cleary, ESPN, Belle Isle, FL, USA
Ed Greene
Hugh Healey

With the technology shifts experienced by the industry over the past several years, broadcasters have been challenged to keep up and evolve. Throughout it all, live broadcast has continued to present challenges for audio engineers. Listen as our renowned panelists share their thoughts on the most significant issues, what has changed, what is the same and how they overcome production challenges of live programming.

Special Event
GRAMMY SOUNDTABLE
Saturday, October 11, 3:00 pm – 4:30 pm
Room 403 AB

Moderator: **Ed Cherney**, Edward Cherney Company, Venice, CA, USA

Panelists: *Michael Brauer*
Alex Da Kid
No I.D.
Don Was

Songs That Move The Needle

Record production is a hybrid art encompassing vision, musicianship, well-honed instincts, and the bottom-line ability to get a project over the finish line. When these elements combine (and the stars align!), the result can be a milestone recording. At this GRAMMY SoundTable event, presented by The Recording Academy Producers & Engineers Wing, five multi-talented, cross-genre hitmakers will debate the who, what, when, where, and why of songs that have left an indelible imprint.

Live Sound Expo LSE11 LSE STAGE
INSTALLED SOUND—REFLECTION CONTROL, FIDELITY, AND INTELLIGIBILITY
Saturday, October 11, 3:00 pm – 3:50 pm

Presenter: **Jay Fullmer**, Applications Engineer, JBL

With applications for both fixed installation venues, and for performance venues and spaces that host touring music acts, applied physics offers tools and techniques to aid in reaching the goals of higher fidelity music and greater intelligibility for the spoken word.

Game Audio Session 12 Saturday, October 11
3:15 pm – 4:45 pm Room 404 AB

YES, YOUR MOBILE GAME CAN HAVE AWESOME AUDIO!

Presenter: **Stephan Schütze**, Sound Librarian, Melbourne, Australia

Many developers work under the fallacy that because they are working on a game for a mobile that high quality audio is not an option. Lack of resources, the need to stay within tight size constraints due to download requirements, and just a general idea that mobile means lower quality. Simply put these ideas are completely wrong. There are a range of tools available that, when utilized properly, support the creation of dynamic, high quality audio for nearly every platform available.

This presentation will cover two main aspects on this topic.

1. An overview of how audio is a sophisticated tool for communicating with your customers in regards to narrative, tactical information, and feedback.

2 Practical examples of how current tool sets allow for the creation of effective and dynamic audio elements that

are incredibly resource efficient and high quality.

Product Design Session 13 Saturday, October 11
3:30 pm – 5:00 pm Room 402 AB

OPTIMIZING THE POWERED LOUDSPEAKER SYSTEM

Presenters: **Scott Leslie**, PD Squared, Irvine, CA, USA
Brian Oppegaard, SpeakerPower Inc., Santa Ana, CA, USA

Historically amplifiers and loudspeakers have been interfaced using a simplified interface of 4/8 Ohm nominal speakers impedance. With a general market trend towards self-powered speakers, greater optimization in the interface between speaker and amplifier becomes possible. This tutorial aligns to the product design track, theme 2 and will provide a forum for discussing of required amplifier performance for self-powered speakers as well as optimization techniques between the amplifier section and speaker drivers and provide better understanding of the complex interfaces of the signal processing, power amplification and the acoustic domain in a self-powered speaker in order for speaker designers to optimize self-powered speaker designs and achieve higher SPL levels at lower cost.

Workshop 6 Saturday, October 11
3:45 pm – 5:15 pm Room 409 AB

FINDING A GOOD ACOUSTIC SPACE

Chair: **Mike Wells**, Mike Wells Mastering, Los Angeles, CA, USA

Panelists: *Bob Hodas*, Bob Hodas Acoustics, Berkeley, CA, USA
Steven Klein, Studio City, CA, USA
Ellis Sorkin

New audio studios open often, and new people entering the market are always looking for qualified, usable studio space. But where should you start? Build your own room? Find an existing room to rent? Remodel an existing space? What issues should you be concerned with acoustically? How can you find out what available spaces are on the market? What is a "reasonable" rental price range for existing studio space? This workshop aims to address common questions such a these and also to empower the audience with useful information on the subject of finding great acoustical spaces to run an audio business.

Network Audio Session 12 Saturday, October 11
4:00 pm – 5:00 pm Room 406 AB

AVB FOR AUDIO DISTRIBUTION IN CARS, SOME REAL WORLD IMPLEMENTATION

Presenter: **Robert Boatright**, Harman

One arena where Ethernet AVB has already seen strong adoption is in infotainment systems for cars. This session will discuss some of the implementations already designed or in advanced development. Systems requirements and how these drive the network architecture and the end point specifications will be explored. Potential adopters in other application spaces such as commercial and residential systems will be interested to learn where the development of AVB based products for cars is driving the integrated circuit development road maps.

Project Studio Expo PSE12 PSE STAGE
THE IMPORTANCE OF A REFERENCE MONITORING LEVEL
Saturday, October 11, 4:00 pm – 4:45 pm

Presenter: **Hugh Robjohns**, Technical Editor, Sound On Sound, Cambridge, UK

If you're serious about recording and mixing you need to set a consistent reference level to which you can always return. *SOS* Technical Editor Hugh Robjohns explains the concept of a reference operating level in the DAW and how that relates to the common metering formats, before showing how to extend that reference level into the acoustic domain with a simple seven-step process resulting in an appropriate calibrated loudspeaker monitoring level.

Live Sound Expo LSE12 LSE STAGE
MIKING FUNDAMENTALS FOR THE STAGE
Saturday, October 11, 4:00 pm – 4:50 pm

Presenters: **J. Mark King**, Production & Music Mixer, Primetime Emmy Awards, Dancing With The Stars, Grammy Nominations Concert, ACMs, CMAs
Dave Mendez, Systems Support Engineer, Shure

A wide array of microphone types, from the common dynamic mic to the condenser and even ribbon models, can find a home in the live sound toolbox. We will take a closer look at how these microphones work, including guidelines for effective use and the applications for which they might best be suited.

Sound for Picture 4 Saturday, October 11
4:30 pm – 6:30 pm Room 408 B

IMMERSIVE SOUND AND ONE-MASTER MIXING

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

Panelists: *Ton Kalker*, DTS, Mountain View, CA, USA
Stephan Mauer, IOSONO GmbH, Erfurt, Germany
Bert Van Daele, Auro Technologies, Mol, Belgium

With “Immersive Sound” and "Object-Based Mixing” the new buzzwords in Sound for Picture production, some are wondering if the systems can be designed to streamline the growing requirements for “special mixes.” The methods and tools currently available and the changes that might be necessary going forward to eliminate specialized mixing for different devices and markets are discussed by this panel.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Saturday, October 11 4:00 pm Room 407
Standards Committee Meeting on Audio Applications of Networks

Student Event/Career Development
RECORDING COMPETITION—PART 2
Saturday, October 11, 4:30 pm – 6:30 pm
Room 306 AB

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. 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 Sunday 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 those who don't make it to the finals, and it's a great chance to meet other students and faculty.

Traditional Studio Recording
Judges: Jim Anderson, Jim Kaiser, Richard King, Mandy Parnell

Modern Studio Recording
Judges: Piper Payne, Ron Prent, Darcey Proper, Michael Romanowski

Session EB5 — Saturday, Oct. 11
5:00 pm – 6:15 pm — Room 309

PAPERS SESSION 3

Chair: **Ville Pulkki**, Aalto University, Espoo, Finland; Technical University of Denmark, Denmark

5:00 pm

EB5-1 Comparison and Contrast of Reverberation Measurements in Grace Cathedral San Francisco—*Wieslaw Woszczyk*,[1] *Durand R. Begault*,[2] *Amanda G. Higbie*[3]
[1]McGill University, Montreal, QC, Canada
[2]Human Systems Integration Division, NASA Ames Research Center, Moffett Field, CA, USA
[3]Oxford Acoustics, Oxford, MS, USA

In 2001 and 2013 the authors made separate and contrasting acoustical measurements of reverberation within Grace Cathedral, a gothic-style church located in San Francisco notable for its long reverberation time and immersive sound quality for musical performance. Both measurements used different, non-standard methods for generating and capturing the reverberant sound field. This paper explores the synergy between both sets of measurements for both recording applications and for understanding spatial hearing in large spaces. Applications include manipulation of the perceived size of relatively smaller enclosures having multiple coupled spaces and for improving the long term habitability of space stations, lunar outposts, or other confined spaces.
Engineering Brief 178

5:15 pm

EB5-2 The Design of Dolby ATMOS Post Production Theaters—*Andrew Munro*, Munro Acoustics Ltd., London, UK

The introduction of Dolby ATMOS has raised several questions concerning the optimum design of the room acoustics for smaller dubbing theaters where the number of speakers and the associated hardware has some bearing on the room as a whole. It is also apparent that compact speaker systems and a flexible approach to installation opens the scope for multi-channel sound installations. The author compares several installation and also compares these with an alternative approach using the IOSONO system.
Engineering Brief 179

5:30pm

EB5-3 Mobile and Variable Absorption Product that Includes Low Frequencies—*Niels Adelman-Larsen*, Flex Acoustics, Copenhagen, Denmark

At amplified music concerts, from medium sized venues to the biggest arenas, low frequency reverberation is known to be the primary source for an undefined sound with low clarity, even close to the loud speakers. Therefore, means for providing additional low frequency absorption is always a concern. Several layers of fabric at various distances from reflecting surfaces has usually been the best option. Still this method provides a relatively modest absorption coefficient in the important 63 and 125 Hz octave bands, while damping the high frequencies that the audience absorbs well, also due to the high Q of loud speakers at higher frequencies. A new, patented technology of inflated, ultra thin plastic membranes seems to solve this challenged in both multipurpose venues that need to adjust their acoustics at the push of a button, or in halls and arenas that only occasionally present amplified music and need to be treated for the event. This paper presents briefly the technology with cases from differently sized halls and arenas.
Engineering Brief 180

5:45 pm

EB5-4 Ellipsoidal Reflector for Measuring Oto-Acoustic Emissions—*Ville Pulkki,*[1] *Vesa Heiskanen*,[1] *Bastian Epp*[2]
[1]Aalto University, Espoo, Finland
[2]Hearing Systems, Technical University of Denmark, Denmark

A truncated prolate ellipsoidal reflector having the ear canal of a listener at one focal point and large-diaphragm low-noise microphone at the other focal point is proposed for free-field recordings of oto-acoustic emissions. A prototype reflector consisting of three pieces is presented, which enables measuring the response of the system with different truncations. The response of the system is measured with a miniature loudspeaker, and proof-of-concept measurements of oto-acoustic emissions are presented. The effect of truncation and other physical parameters to the performance of the system are discussed.
Engineering Brief 181

6:00 pm

EB5-5 Key Sonic Characteristics Voice Identification Analysis and the Courts—*Thomas Guzman-Sanchez*, GS Media Lab, Northridge, CA, USA

The paper presents a voice identification analysis technique based on the analysis and comparison of four key sonic/acoustic characteristics. The proposed solution provides scientific method of analyzing voice or audio recordings

and accessing similarities or differences for uses of identification. The technique utilizes five different types of preparation and analysis generating multiple results that can be applied for comparison. The function is similar to a police line-up except this is done with audio recordings. The proposed analysis will provide an accurate process in accessing voice identification.
Engineering Brief 182

Tutorial 19 — Saturday, October 11
5:00 pm – 6:00 pm — Room 404 AB

PSYCHOACOUSTICS FOR SOUND DESIGNERS

Presenter: **Shaun Farley**, Dynamic Interference, Berkeley, CA, USA

This session will explore some of the mechanical and psychological oddities that affect our perception of sound. This will be focused on the sound designer's perspective. As such, it will be more about identifying end behaviors of the human hearing system than the underlying reasons for those behaviors. Some we develop awareness of through experience in our work, while others remain sub-conscious until they're presented to us. This is meant to be a starting place for us to begin talking about how we can use these behaviors as tools for sonic storytelling.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Broadcast/Streaming Media Session 14
Saturday, October 11 — 5:00 pm – 6:30 pm
Room 408 A

TELEPHONY AND IP CODECS:
HOW TO CHOOSE WHAT IS BEST FOR YOU

Chair: **David Bialik**

Panelist: *Kevin Campbell*, Worldcast Systems
Chris Crump, Comrex
Kirk Harnack, Telos Alliance, Nashville, TN, USA;
South Seas Broadcasting Corp., Pago Pago, American Samoa
Tom Harnett
Doug Irwin, Clear Channel, Los Angeles, CA, USA
Joe Talbot

Evolving technology has made great strides in audio transport versatility, connectivity, availability, and reliability. Whether wired or wireless, this discussion will provide real-time remote program solution options for broadcasters trying to make ends meet.

Live Sound Seminar 9 — Saturday, October 11
5:00 pm – 6:30 pm — Room 406 AB

SOUND FOR LIVE CLASSICAL MUSIC

Presenter: **Frederick Vogler**, Sonitus Consulting, Los Angeles, CA, USA

Fred Vogler is the sound designer for the LA Philharmonic overseeing both their recordings and live reinforcement in two of the most iconic venues in Los Angeles and the world: the Walt Disney Concert Hall and the Hollywood Bowl. In this seminar he will discuss techniques, challenges and tricks that he practices in reinforcing the most complex of musical ensemble—the live orchestra.

Product Design Session 14 — Saturday, October 11
5:00 pm – 6:30 pm — Room 402 AB

SPECIFYING AND SELECTING
LOUDSPEAKER DRIVERS

Presenter: **Steve Hutt**, Eighteen Sound, Reggio Emilia, Italy; Equity Sound Investments, Bloomington, IN, USA

When designing loudspeaker systems, design engineers and product managers have a seemingly infinite number of loudspeaker drivers from which to choose. Understanding LF, MF and HF driver specifications is critical to this task if value of the overall product is to be maximized. This tutorial will focus on achieving a desired sonic envelope in the overall system design by specifying the correct drivers to meet those goals and will help answer the following (and other) questions: 1) What is a loudspeaker specification and how is specification data determined? 2) How does loudspeaker selection affect overall system design? 3) Should loudspeakers be selected from a catalog or custom designed? 4) How does loudspeaker selection and budget affect marketability of a loudspeaker system? 6) How do you compare the performance of different options? 7) What matters most, electrical power in, or sound power out?

Recording and Production Session 5
Saturday, October 11 — 5:00 pm – 7:00 pm
Room 403 AB

AN AFTERNOON WITH GEOFF EMERICK

Presenters: **Geoff Emerick**
Howard Massey, OTRW, New York, NY, USA

The Beatles and Beyond

Geoff Emerick is, of course, best known for his work with The Beatles. Yes, he's the man who engineered *Revolver* and *Sgt. Pepper's Lonely Hearts Club Band*, as well as many of the tracks on *Magical Mystery Tour, The White Album*, and *Abbey Road*.

But there's much more to his career than just those milestones. Join us for a fascinating conversation with Geoff Emerick as he discusses not only his audio adventures with the Fab Four and each of the Beatles individually (including the making of Paul McCartney's *Band On The Run* in Lagos, despite band desertion and monsoons, not to mention spiders and cockroaches as big as dinner plates) but provides a glimpse behind the scenes of the records he crafted with artists such as Elvis Costello, Jeff Beck, The Zombies, Robin Trower, Badfinger, America, Art Garfunkel and Nellie McKay. This very special presentation will be enhanced with rarely seen photos and video footage as well as rarely heard audio clips. Time will be allotted for questions from the audience.

This event is open to all badges.

Special Event
AES67 TA MEETING
Saturday, October 11, 5:00 pm – 6:30 pm
Room 303-4AB

Chair: **Bill Scott**, Bosch Communications Systems, Burnsville, MN, USA

Panelists: *Terry Holton*, Yamaha R&D Centre, London, UK
Stefan Lederberger, Lawo Group, Zurich, Switzerland; LES Switzerland GmbH
Marty Sacks, Telos Alliance
Rich Zweibel, QSC Audio

The Media Networking Alliance (MNA) has been formed to actively promote the adoption and standardization of AES67 as an audio interoperability standard through marketing, education, and training. The Media Networking Alliance steering committee will discuss the formation of the Alliance, their mission and goals for the upcoming year, and provide information on membership. This meeting is open to all individuals and companies who are interested in promoting the adoption of AES67.

Saturday, October 11 5:00 pm Room 405
Technical Committee Meeting on Signal Processing

Historical Event H5
HISTORY OF LINE ARRAYS
Saturday, October 11, 5:30 pm – 6:30 pm
Room 409 AB

Presenter: **Mark Ureda**, Harman Professional, Northridge, CA, USA

A presentation on the history of line arrays from the earliest column speakers through to modern methods and configurations of fixed and portable line array implementations. The presenter, Mark Ureda, has worked for Altec-Lansing and Electro-Voice as well as JBL/Harman, and is the author of a number of AES line array papers.

Session P17 Sunday, Oct.12
9:00 am – 12:00 noon Room 308 AB

SIGNAL PROCESSING—PART 2

Chair: **J. Keith McElveen**, Wavesciences Corp.

9:00 am

P17-1 A Practical Approach to Robust Speech Recognition Using Two Microphones in Driving Environments—*Jaeyoun Cho, Seungyeol Lee, Inwoo Hwang*, Samsung Electronics Co. Ltd., Suwon-si, Gyeonggi-do, Korea

Now that the technologies related to the automatic speech recognition have been mature enough and applicable to our everyday life, people have started considering speech as the most desirable human-device interaction means and utilized speech recognition in vehicles. Nonetheless, it is still challenging to recognize speech correctly in driving environments for at least two reasons. One is that the speech signal is corrupted by innumerable noise sources such as the engine sound, road friction, music from the radio, even worse the mixture of spoken words by passengers, etc. Another is that the recognition device may be put at any place like cup holder, passenger seat or dashboard. In this paper we propose a robust speech recognition front-end that removes the probable ambient noise in a driving car regardless of where the recognition device is. The proposed method finds the direction of speech and enhances the speech signal by first detecting the existence of speech utterance using only two microphones. This front-end is designed with practical consideration so that its implementation in the mobile device showed higher recognition accuracy, shorter processing latency and lower computing power consumption than any other top-tier methods.
Convention Paper 9191

9:30 am

P17-2 Predistortion of a Bidirectional Cuk Audio Amplifier—*Thomas Haagen Birch, Dennis Nielsen, Arnold Knott, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

Some non-linear amplifier topologies are capable of providing a larger voltage gain than one from a DC source, which could make them suitable for various applications. However, the non-linearities introduce a significant amount of harmonic distortion (THD). Some of this distortion could be reduced using predistortion. This paper suggests linearizing a nonlinear bidirectional Cuk audio amplifier using an analog predistortion approach. A prototype power stage was built and results show that a voltage gain of up to 9 dB and reduction in THD from 6% down to 3% was obtainable using this approach.
Convention Paper 9192

10:00 am

P17-3 Frequency Dependent Loss Analysis and Minimization of System Losses in Switch-Mode Audio Power Amplifiers—*Akira Yamauchi, Arnold Knott, Ivan H. H. Jørgensen, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

In this paper the frequency dependent losses in switch-mode audio power amplifiers are analyzed and the loss model is improved by taking the voltage dependence of the parasitic capacitance of MOSFETs into account. The estimated power losses are compared to the measurement and great accuracy is achieved. By choosing the optimal switching frequency based on the proposed analysis, the experimental results show that the system power losses of the reference design are minimized and an efficiency improvement of 8% in maximum is achieved without compromising audio performances.
Convention Paper 9193

10:30 am

P17-4 Resolving Delay-Free Loops in Recursive Filters Using the Modified Härmä Method—*Will Pirkle*, University of Miami - Coral Gables, FL, USA

Resolving delay-free loops in recursive filter structures has been a longstanding problem approached in several different ways including signal flow graph manipulation [1], [2] and more recently with Zavalishin's instantaneous response technique [3],[4]. Härmä demonstrates a method for resolving delay-less loops in recursive filter structures [5] but the technique is limit-

ed to a specific generic loop topology in which the feedforward branch does not implement signal processing; all processing is implemented in one or more delay-less feedback loops. We modify Härmä's method to accommodate filter processing in the feedforward branch and provide a step-by-step method to resolve delay-less loops in recursive filter structures. We conclude with examples including a new method of synthesizing fourth order filters.
Convention Paper 9194

11:00 am

P17-5 Novel Hybrid Virtual Analog Filters Based on the Sallen-Key Architecture—*Will Pirkle*, University of Miami, Coral Gables, FL, USA

The Sallen-Key filter structure is a revered analog filter design topology. In Sallen-Key lowpass and highpass filters, the cutoff frequency and resonance (Q) controls are decoupled though the cutoff and resonant frequencies are not. In this paper we demonstrate novel variations on the Sallen-Key architecture and we decouple the resonant and cutoff frequencies. This produces multiple hybrid filter designs including resonant quasi-first order lowpass and highpass filters, resonant quasi-first order low and high shelving filters, decoupled resonant second order filters and doubly resonant quasi-second order lowpass and highpass filters. In the doubly-resonant filters all three frequencies may be decoupled and independently adjustable; they also self-oscillate at both resonant frequencies.
Convention Paper 9195

11:30 am

P17-6 Timbre Imitation and Adaptation for Experimental Music Instruments: An interactive Approach Using Real-Time Digital Signal Processing Framework —*Mingfeng Zhang*,[1] *John Granzow*,[2] *Gang Ren*,[1] *Mark F. Bocko*[1]
[1]University of Rochester, Rochester, NY, USA
[2]Stanford University, Stanford, CA, USA

We propose a real-time digital signal processing framework to extend the timbre control capability of experimental musical instruments. We focus on two music cognition concepts of timbre imitation and adaption to enable experimental musical instruments to be integrated into existing ensemble works. In timbre imitation, we aim to simulate known timbre patterns to enhance the musical coherence during an ensemble performance. In timbre adaptation, we explore extended timbre manipulation settings such as complementary timbre and contrasting timbre. Our proposed framework is implemented on a low cost real-time digital signal processing system to ensure easy adaptability. Our study is based on saxophone and violin but can be readily generalized to other instrument categories.
Convention Paper 9196

Session P18 Sunday, Oct. 12
9:00 am – 12:30 pm Room 309

APPLICATIONS IN AUDIO—PART 1

Chair: **Jonathan Hong**, McGill University/GKL Audio, Montreal, Quebec, Canada

9:00 am

P18-1 Measuring Time Varying or Offset Voltage Dependent Harmonic and Intermodulation Distortion via Filter Banks Including a Stairstep Signal and Measuring FM Distortion in IM Distortion Signals—*Ronald Quan*, Ron Quan Designs, Cupertino, CA, USA

This paper will present methods of measuring dynamic harmonic distortion using filter banks and a staircase signal. The harmonic distortion is measured in real time at each level of the staircase signal. For measuring dynamic time varying 2nd and 3rd order intermodulation distortion, a low frequency sinewave signal is combined with a higher frequency signal. Also, the FM distortion of the intermodulation distortion signals is measured. FM distortion in current mode op amps is measured. Also FM distortion is measured in an op amp with conventional Miller compensation then measured later in an op amp with two-pole compensation. Finally, Volterra Series distortion analysis is included as part of an equation that describes phase or frequency modulation effects in a nonlinear system.
Convention Paper 9197

9:30 am

P18-2 DC Servos and Digitally-Controlled Microphone Preamplifiers—*Gary Hebert*, That Corp., Milford, MA, USA

Microphone preamplifiers for professional audio applications require a very wide range of gain and low noise in order to provide a high-quality interface with the vast number of available microphones. In many modern systems the preamplifier gain is controlled indirectly via a digital interface in discrete steps. Often dc servo amplifiers are employed as a means of keeping the dc gain fixed to avoid large changes in output offset voltage while the audio band gain is varied. The resulting highpass filter response varies substantially as a function of the preamplifier gain. We investigate the frequency and time-domain effects of this. We also investigate several approaches to minimize these effects.
Convention Paper 9198

10:00 am

P18-3 The Design of Urban Sound Monitoring Devices—*Charlie Mydlarz, Samuel Nacach, Tae Hong Park*, *Agnieszka Roginska,* New York University, New York, NY, USA

The urban sound environment of New York City is notoriously loud and dynamic. As such, scientists, recording engineers, and soundscape researchers continuously explore methods to capture and monitor such urban sound environments. One method to accurately monitor and ultimately understand this dynamic environment involves a process of long-term sound capture, measurement and analysis. Urban sound recording requires the use of robust and resilient acoustic sensors, where unpredictable external ➥

conditions can have a negative impact on acoustic data quality. Accordingly, this paper describes the design and build of a self-contained urban acoustic sensing device to capture, analyze, and transmit high quality sound from any given urban environment. This forms part of a collaborative effort between New York University's (NYU) Center for Urban Science and Progress (CUSP) and the NYU Steinhardt School's Citygram Project. The presented acoustic sensing device prototype incorporates a quad core Android based mini PC with Wi-Fi capabilities, a custom MEMS microphone and a USB audio device. The design considerations, materials used, noise mitigation strategies and the associated measurements are detailed in the following paper.
Convention Paper 9199

10:30 am

P18-4 A Comparison of Real-Time Pitch Detection Algorithms in SuperCollider—*Elliot Kermit-Canfield*, Stanford University, Stanford, CA, USA

Three readily-available pitch detection algorithms implemented as unit generators in the SuperCollider programming language are evaluated and compared with regard to their accuracy and latency for a variety of test signals consisting of both harmonic and non-harmonic content. Suggestions are made for the type of signal on which each algorithm performs well.
Convention Paper 9200

11:00 am

P18-5 Performance Comparison Between Nested Differentiating Feedback Loops and Classic Three Stage Operational Amplifier Architectures: A SPICE-Based Simulation Approach—*Ariel Muszkat, David Kadener*, Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina

Since 1970, the three stage operational amplifier with dominant pole compensation has become the standard basis in amplifier's architectures. However, during the 1980s and following years the nested differentiating feedback loops (NDFL) concept was introduced by Edward M. Cherry as an attempt to improve the classic power amp performance, mainly distortion caused by class B output stages. The proposal of this work lies on the first part of a comparison between both topologies' performance in a SPICE based simulator. Most important analyzed parameters are open-loop gain, distortion, transient response, and, of course, stability. In addition, modern semi-conductor devices and improved inner stages are used to make the comparison circuits based on small signal devices such as discrete operational amplifiers.
Convention Paper 9201

11:30 am

P18-6 Making Audio Sound Better One Square Wave at a Time (Or How an Algorithm Called "Undo" Fixes Audio)—*Leif Claesson*, Omnia Audio, Cleveland, OH, USA

Audio mastering engineers have felt increasing pressure over the years to master recordings at ever increasing loudness levels as compared to other contemporary recordings, by way of dynamic compression, peak limiting, and hard clipping. This pursuit of loudness adds distortion, and reduces fidelity. When radio stations play the compromised audio through their FM processing chains, this confluence of degradation causes serious audio quality issues on air. This paper shall examine what the music endures when broadcast on FM, and how that led to the invention of the "undo" algorithm, which repairs damage caused by these mastering techniques by adaptively de-clipping and de-compressing the mastered recordings.
Convention Paper 9202

12:00 noon

P18-7 Acoustic Surveillance of Hazardous Situations Using Nonnegative Matrix Factorization and Hidden Markov Model—
Kwang Myung Jeon,[1] *Dong Yun Lee*,[1] *Myung J. Lee*[2]
[1]Gwangju Institute of Science and Technology (GIST), Gwangju, Korea
[2]City University of New York, New York, NY, USA

In this paper an acoustic surveillance method is proposed for accurately detecting hazardous situations under noisy conditions. In order to improve detection accuracy, the proposed method first tries to separate each atypical event from the input noisy audio signal. Next, maximum likelihood classification using multiple hidden Markov models (HMMs) is carried out to decide whether or not an atypical event occurs. Performance evaluation shows that the proposed method achieves higher detection accuracy under various signal-to-noise ratio (SNR) conditions than a conventional HMM-based method.
Convention Paper 9203

Workshop 7 **Sunday, October 12**
9:00 am – 10:30 am **Room 408 A**

21ST CENTURY VINYL

Moderator: **Jonathan Wyner**, Berklee College of Music, Boston, MA, USA; M Works Mastering

Panelists: *JJ Golden*, Golden Mastering
Robert Hadley, The Mastering Lab
Peter Lyman, Infrasonic Sound Recording Co., Los Angeles, CA, USA
Doug Sax, The Mastering Lab, Ojai, CA, USA

Vinyl—the cornerstone format of the music industry. Vinyl—despite the array of audio media formats that have come out since the LP, vinyl remains a vibrant format with a large (and growing!) fan base. Now that we find ourselves almost 15 years into the 21st century, how is the process of creating vinyl records changed? How is the equipment holding up after all these years? How hard is it to find media/supplies/parts for lathes still in use today? How has the approach to cutting vinyl masters changed (if at all) with the changes to delivered mixes (e.g., shrinking dynamic range, demands for longer side lengths, etc.)? This workshop will discuss the current state of the vinyl mastering industry, how it's adapting to the changing mix landscape, and thoughts on the future of vinyl.

Tutorial 20 **Sunday, October 12**
9:00 am – 10:30 am **Room 404 AB**

AUDIO GOES VIDEO: MULTIMEDIA-BASED PRESERVATION OF THE COLLECTION OSKAR SALA OR "HOW TO SAFEGUARD HITCHCOCK'S *THE BIRDS*"

Presenter: **Nadja Wallaszkovits**, Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria

Oskar Sala was a German musician, scientist and a pioneer of electronic music. He played and further developed the trautonium, a predecessor of the synthesizer. He composed the scores for more than 300 films and created the effect soundtrack for Alfred Hitchcock's film 'The Birds', receiving many awards for his works. After his death he left a collection of about 1200 analogue magnetic audio tapes, stored in the archives of Deutsches Museum in Munich. Oskar Sala fully exploited all the possibilities of analogue tape technology, using impressive experimental approaches. His tapes have become artworks themselves, as they comprise a unique richness of very special metadata: cut up to 200 times per reel, Sala used the tapes as a (more or less readable) notebook. Such and many more surprises made the adequate safeguarding and digitization of the collection a unique undertaking. The collection has been successfully digitized under the consultancy of the Phonogrammarchiv Vienna. The tutorial outlines the various challenges of this project and discusses the parameters and practical problems of the audio transfer, as well as the strategy of safeguarding the richness of metadata by use of multimedia-based documentation, such as photographic capturing and high definition video recording.

This session is presented in association with the AES Technical Committee on Archiving Restoration and Digital Libraries

Live Sound Seminar 10 **Sunday, October 12**
9:00 am – 11:00 am **Room 406 AB**

CORPORATE EVENTS: COMPLEX SETUP, ONE CHANCE TO GET IT RIGHT, WHAT COULD GO WRONG?

Presenters: **Harold Blumberg**, Blumberg Sound Design
Robyn Gerry-Rose, Freelance, Local 695, Los Angeles, CA
Mike Kahrs, Sound-Management
Michael (Bink) Knowles, Freelance Engineer,
Oakland, CA, USA
Ken Newman, Newman Audio, Inc., Canyon Country, CA, USA
Eric Stahlhammer, Greater Than Designs, San Diego, CA, USA

When you step up to audio for corporate events, you will find that the clients are more demanding, with a higher expectation that the spoken word is heard very clearly. Corporate sound can be rewarding but it can also be difficult to accomplish, since "the look" is often more important than the optimal loudspeaker placement. The routing of signals can get very complicated, with multiple loudspeaker zones and teleconferencing. Our panel will describe years of frustration and success in the trenches at corporate events.

Game Audio Session 13 **Sunday, October 12**
9:00 am – 10:00 am **Room 408 B**

MIDI: STILL STRONG AFTER 30 YEARS – NEW ADVANCES WITH WEB BROWSERS, BLUETOOTH, AND MORE

Presenters: **Athan Billias**, Executive Board Member, MIDI Manufacturers Association; Director of Strategic Product Planning, Yamaha
Pete Brown, DX Engineering Engagement and Evangelism, Creative Media Apps, Microsoft
Pat Scandalis, CTO & Acting CEO, moForte.com
Torrey Walker, Core Audio Software Engineering Team, Apple

Mobile platforms from Google, Apple, and Microsoft are starting to catch up to desktop/console platforms when it comes to audio capabilities. This session will provide an overview of new Audio/MIDI capabilities in the Chrome web browser, Chrome OS, Android OS, and Windows RT OS, plus a progress report on the "MIDI over Bluetooth" standard involving the major OS developers and MIDI hardware/software makers.

Network Audio Session 13 **Sunday, October 12**
9:00 am – 10:00 am **Room 409 AB**

IMPLEMENTING THE COMMAND AND CONTROL CAPABILITIES IN AVDECC (IEEE1722.1)

Presenter: **Jeff Koftinoff**, Meyer Sound Laboratories, Vernon, BC, Canada

An overview of the major capabilities of the AVB Control Protocol IEEE 1722.1-2013, known as the AVDECC (Audio/Video Discovery, Enumeration, Connection management, and Control) Protocol and details of the pieces you need to implement the protocol for an Talker, Listener, or Responder embedded AVB device.

Product Design Session 15 **Sunday, October 12**
9:00 am – 10:30 am **Room 402 AB**

SOFTWARE FOR LOUDSPEAKER AND APPLICATION ENGINEERING AND RELATED PRODUCTS

Presenter: **Stefan Feistel**, AFMG Technologies GmbH, Berlin, Germany

This tutorial will give an introduction into the capabilities of modern acoustic modeling and measurement software as it relates to loudspeaker design, marketing, and application support. Focus is put on: 1. The integration of end-user software with hardware products to create superior user experience, state-of-the-art control features, and high brand recognition. 2. The use of software during the engineering process in order to develop and refine the final product. 3. To leverage the synergies of end-user software with engineering design software.

Key areas of the presentation include: 1. Creating and offering tailored prediction software to support end users in loudspeaker system design and setup. 2. Establishing sound system and room-acoustic modeling software as tools for product marketing, case studies, and application support. 3. Implementing cross-platform communication interfaces, e.g., for web access, DSP-based control and monitoring, system EQ, or beam steering. 4. Acquiring, validating, and publishing 3D high-resolution modeling ➥

data for loudspeakers, loudspeaker systems and arrays. 5. Integrating numerical algorithms for loudspeaker array systems in order to optimize their mechanical configuration as well as their IIR/FIR filter settings. 6. Using CDPS modeling for feasibility studies and proof of concept investigations. 7. Analyzing and improving loudspeaker performance parameters by means of advanced modeling and measurement tools.

Recording and Production Session 6
Sunday, October 12 9:00 am – 10:00 am
Room 403 AB

INSIDE THE MIX BY DAVE REITZAS

Presenter: **Dave Reitzas**

Join multi-Grammy winning Engineer/Mixer Dave Reitzas (Barbara Steisand, Madonna, Whitney Houston, Josh Groban, Michael Buble), as he dissects the multi-tracks of some of the hit songs that he has worked on as an engineer over the last 30 years. Dave will share the tips and techniques that he uses to record everything from rhythm sections to orchestras, and synthesizers to vocals.

Sound for Picture 5 Sunday, October 12
9:00 am – 11:00 am Room 306 AB

WORLD-CLASS CINEMA SOUND MIXERS DISCUSS THEIR CRAFT

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

Panelists: *Sherry Klein*,
Mark Linden,
Gregory Watkins,

The pinnacle of Sound for Picture recording is the sound mixing for Hollywood-made motion pictures. This workshop features the top members of this group.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Sunday, October 12 9:00 am Room 407
AESSC Plenary Meeting

Tutorial 21 Sunday, October 12
10:15 am – 11:45 am Room 409 AB

LISTENING TESTS—UNDERSTANDING THE BASIC CONCEPTS

Presenter: **Jan Berg**, Luleå University of Technology, Piteå, Sweden

Listening tests and other forms of data collection methods that rely on human responses are important tools for audio professionals, as these methods assist our understanding of audio quality. There are numerous examples of tests, either formally recommended and widely used, or specially devised for a single occasion. In order to understand listening tests and related methods, and also to potentially design and fully benefit from their results, some basic knowledge is required. This tutorial aims to address audio professionals without prior knowledge of listening test design and evaluation. The fundamentals of what to ask for, how to do it, whom to engage as listeners, what sort of results that may be expected and similar issues will be covered, preferable in co-operation with the audience. The goal is to create an understanding of the basic concepts used in experimental design, in order to enable audio professionals to appreciate the feasibility of listening tests.

Game Audio Session 14 Sunday, October 12
10:15 am – 11:45 am Room 408 B

NEW DAW RISING: SCORING AND MIXING YOUR GAME, IN THE GAME

Presenter: **Guy Whitmore**, PopCap Games

A common practice for game composers and sound designers today is to compose and arrange fully mixed music and sound cues in Pro Tools or Logic, then have an implementation specialist drop those files into the game. Sound integration, in this case, is seen as a basic technical task. But in order to score and mix a game with greater nuance, the composer would want to see the game in action while composing; the sound designer would want to watch the interactivity of the visuals, working in a DAW that includes robust adaptive features. This new DAW exists; it is your game audio engine and its authoring tools. In this scenario, sound integration is a highly creative endeavor, where music arranging, mixing, mastering, and even composing takes place.

Recording and Production Session 7
Sunday, October 12 10:15 am – 11:45 am
Room 403 AB

RAW TRACKS: RED HOT CHILI PEPPERS —A MASTER CLASS

Moderator: **Mark Rubel**, The Blackbird Academy, Nashville, TN, USA; Pogo Studio, Nashville, TN, USA

Panelist: **Andrew Scheps**

Renowned engineer/producer Andrew Scheps will discuss, analyze, and deconstruct a classic Red Hot Chili Peppers song “Pink Like Floyd” in the inaugural Raw Tracks series at the AES 137th International Convention in Los Angeles.

Broadcast/Streaming Media Session 15
Sunday, October 12 10:30 am – 12:00 noon
Room 408 A

TROUBLESHOOTING SOFTWARE ISSUES

Chair: **Jonathan Abrams**, Nutmeg Post, New York, NY, USA

Panelist: *Mark Fassler*, Avid
Charles Van Winkle, Adobe, Minneapolis, MN, USA

Your application has either unexpectedly quit or brought you to the blue screen of death. Now what? How can you use a crash report to start the troubleshooting process? What should you do before contacting support or posting on a forum? How can plug-ins make or break the stability of your system? How can permissions wreak havoc on your system or workflow? What can you do to rule out hardware issues? At what point do you need to switch your thinking from software to hardware as the source of the problem? What problems have you been encountering that you need assistance with in order to stop troubleshooting and get back to work?

Please bring your own questions and get answers to the others for Mac OS X, Windows, Adobe Audition, and Avid Pro Tools.

This session is open to all badges.

Network Audio Session 14 Sunday, October 12
10:45 am – 11:45 am Room 404 AB

LARGE SCALE IMPLEMENTATIONS OF AVB NETWORKS

Presenter: **Jeff Koftinoff**, Meyer Sound Laboratories, Vernon, BC, Canada

Audio Video Bridging makes it easy to build medium scale audio networks. Learn aspects of how to plan, deploy, and manage AVB audio and video networks. Learning objectives include:

- Understand the important points to consider when choosing AVB devices.
- Estimate AVB network infrastructure scale and requirements.
- Learn methods to manage deployed AVB networks.

Product Design Session 16 Sunday, October 12
10:45 am – 12:15 pm Room 402 AB

MEASURING LOUDSPEAKER SYSTEMS

Presenter: **Charlie Hughes**, Excelsior Audio, Gastonia, NC, USA; AFMG, Berlin, Germany

This tutorial session will cover best practices for loudspeaker measurements. It is critical for product development and component selection to know the response of loudspeaker systems and components with reasonable accuracy in order to make informed decisions based on comparisons of data. In this session we will briefly cover the basics of FFT-based measurement systems before moving on to additional topics. Averaging and S/N Windowing (both signal acquisition and impulse response windowing) Ground plane measurement techniques Directivity measurements Maximum input voltage measurements Impedance Alignment of pass bands.

Live Sound Seminar 11 Sunday, October 12
11:00 am – 1:00 pm Room 406 AB

DIGITAL CONSOLE USER INTERFACE EVOLUTION

Moderator: **Louis Adamo**, Hi-Tech Audio

Panelists: *Pat Baltzall*, Baltzall Audio Design
Harold Blumberg, Blumberg Sound Designs
David Morgan, FOH Engineer

Analog mixing consoles have had roughly 40 years to evolve into a familiar physical interface that allows users to move from one brand to another with relative ease. Digital consoles, though, are still developing and experimenting with interface paradigms that make it difficult to exchange one model for another. Most engineers hope the that future will bring a more unified digital user interface. In this seminar expert sound engineers will discuss the current state of the digital mixing console user interface; what works, what doesn't work and what might be interesting ideas to explore in the future. Users of any specific console will benefit from hearing a general overview of the best user experiences there are in the field today.

Sunday, October 12 11:00 am Room 405
Technical Committee Meeting on Spatial Audio

Project Studio Expo PSE14 PSE STAGE
FOCAL PRESS
Sunday, October 12, 11:00 am – 11:45 am

Moderator: **Kyle P. Snyder**

Presenters: *Alex Case*
Jason Corey
David Miles Huber
Mike Senior

Mixing Perspectives: Tales of achieving big studio results on a project studio budget.

Live Sound Expo LSE13 LSE STAGE
TOURING SYSTEM OPTIMIZATION: CASE STUDIES
Sunday, October 12, 11:00 am – 11:50 am

Presenters: **Jason Decter**, FOH for Avril, Blink182, Weezer
Robert Scovill, FOH for Prince, Rush, Tom Petty

Veteran FOH pros share perspectives on prepping a large venue rig for a show.

Workshop 8 Sunday, October 12
11:15 am – 12:45 pm Room 306 AB

ANALOG TAPE IN A DIGITAL WORLD

Chair: **Mike Wells**, Mike Wells Mastering, Los Angeles, CA, USA

Panelists: *Charlie Boulis*, Vertigo Recording Services
Larry Crane, Tape Op Magazine, Portland, OR, USA; Jackpot! Recording Studio
Dan Labrie, ATR Sterices/ATR Magnetics

Despite 20+ years of digital recording advancements, recording to analog tape remains a hot topic in the world of audio engineering. Software modeling companies work hard to create the best emulations while hardware manufacturers pour over designs to also emulate the "magic" of the tape process, and yet "the real thing" remains a no-option for many artists and engineers still today. This workshop will discuss the current state of analog tape manufacturing, availability, and its usage in the recording, mixing, and mastering stages of the music production cycle.

Student Event/Career Development
STUDENT RECORDING CRITIQUES
Sunday, October 12, 11:30am – 12:30 pm
Room 305

Moderator: **Ian Corbett**

Students can bring stereo or surround mixes to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Sign-up for time slots is immediately after the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on CD, DVD, memory-stick, or hard disc, as clearly labeled 44.1 KHz WAVE or AIFF files. The Student Recording Critiques are generously spon- ➥

sored by PMC, and you get to hear your work on some amazing loudspeakers! (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.)

Workshop 9 **Sunday, October 12**
12:00 noon – 1:30 pm **Room 409 AB**

CANCELED.

Sunday, October 12 **12:00 noon** **Room 405**
TC Plenary Meeting

Project Studio Expo PSE15 **PSE STAGE**
AUDIONAMIX
Sunday, October 12, 12:00 noon – 12:45 pm

Live Sound Expo√LSE14 **LSE STAGE**
LOUDSPEAKERS—SMALL SYSTEM ANALYSIS
Sunday, October 12, 12:00 noon – 12:50 pm

Presenter: **Sam Berkow**, Principal and Founding Partner, SIA Acoustics

System performance monitoring software and hardware provide sophisticated assistance to the human ear for sound-system set-up. These tools, their capabilities and application in small venues are explored, along with system optimization using commonly available tools.

Network Audio Session 15 **Sunday, October 12**
12:15 pm – 1:15 pm **Room 404 AB**

INTEROPERABILITY TESTING FOR AUDIO NETWORK APPLIANCES

Chair: **Mark Yonge**, AES Standards Secretary, UK

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

End users, systems integrators, and design consultants need assurance that a networked device will interoperate with other networked devices thath implement the same protocol. How is this interoperability best guaranteed? Interoperability compliance testing is an essential requisite to meaningful claims by manufacturers that their product provides assured interoperability. How is this testing best achieved? This session will explore the various methods for compliance testing with particular reference to verifying compliance with the AES67 interoperability standard.

Student Event/Career Development
STUDENT DELEGATE ASSEMBLY MEETING —PART 2
Sunday, October 12, 12:30 pm – 2:00 pm
Room 408 A

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.

Workshop 10 **Sunday, October 12**
1:00 pm – 2:30 pm **Room 306 AB**

DSD AND DXD: EXTREME RESOLUTION PRODUCTIONS DISCUSSED

Chair: **Dominique Brulhart**, Merging Technologies, Puidoux, Sweitzerland

Presenters: *Robert Friedrich*, Five Four Productions
Morten Lindberg, 2L
John Newton, Soundmirror
Jared Sachs, Channel Classics

With the growing interest in high resolution by the audiophile and general audience and the rapidly growing availability of high resolution content for download on the market, and in relation with the 137th AES convention HRA special event organized with the DEG, we thought mandatory to reiterate our DSD/DXD panel.

As the reality of most extreme resolution productions today is not anymore to ideologically focus on either DSD or PCM, but to choose the most appropriate format based on the production and post-production requirements, as it is been established during the three former workshops, we found more interesting to open this time the discussion to a wider number of panelists to share their experience in this domain, and base the comparison between DSD and DXD on production prerogatives instead of purely sound quality considerations.

Morten Lindberg from 2L, John Newton from Soundmirror, Michael Bishop from Five Four Productions, and Jared Sachs from Channel Classics will be presenting some of their productions and answer to questions about their technical and artistic decisions.

Sound for Picture 6 **Sunday, October 12**
1:00 pm – 3:00 pm **Room 408 B**

SUPERSTARS OF PRODUCTION SOUND RECORDING

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

Panelists: *Peter Kurland*
Jim Tananbaum, Sound Recording Services, Los Angeles, CA, USA
Mark Ulano

The recording of Production Sound dialog for films is a unique area of sound recording, and done to a high standard by the Hollywood production sound community. This type of recording requires an understanding of the film-making process, a strong relationship with the various crafts, and a wide variety of "tricks" not usually employed in other areas of sound recording. This workshop features several of the top Production Sound Mixers prepared to discuss their craft and recording philosophies.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Project Studio Expo PSE16 **PSE STAGE**
FOCUSRITE NOVATION
Sunday, October 12, 1:00 pm – 1:45 pm

Live Sound Expo LSE15 **LSE STAGE**
INSTALLED AUDIO—MODERN AUDIO INSTALL
Sunday, October 12, 1:00 pm – 1:50 pm

Presenters: **Chuck Mitchell**, System Design Consultant, Technology of Arts

Matt Wilkinson, ARUP, designers of the 3D audio system for "The Cube" at Virginia Tech

The A in A/V installations is ever increasing in flexibility and sophistication. Guest systems designers share case studies that illustrate the trends and potential of such installation with profiles of a number of houses of worship and a unique performance space/laboratory.

Session P19 **Sunday, Oct. 12**
1:30 pm – 5:00 pm **Room 308 AB**

SIGNAL PROCESSING—PART 3

Chair: **Duane Wise**, Wholegrain Digital Systems LLC, Boulder, CO, USA

1:30 pm

P18-1 Eliminating Transducer Distortion in Acoustic Measurements—*Finn Agerkvist*,[1] *Antoni Torras-Rosell*,[2] *Richard McWalte*[1]
[1]Technical University of Denmark, Lyngby, Denmark
[2]Danish National Metrology Institute, Lyngby, Denmark

This paper investigates the influence of nonlinear components that contaminate the linear response of acoustic transducers and presents a method for eliminating the influence of nonlinearities in acoustic measurements. The method is evaluated on simulated as well as experimental data and is shown to perform well even in noisy conditions. The limitations of the Total Harmonic Distortion, THD, measure is discussed and a new distortion measure, Total Distortion Ratio, TDR, which more accurately describes the amount of nonlinear power in the measured signal, is proposed.
Convention Paper 9204

2:00 pm

P19-2 Uniformly-Partitioned Convolution with Independent Partitions in Signal and Filter—*Frank Wefers, Michael Vorländer*, RWTH Aachen University, Aachen, Germany

Low-latency real-time FIR filtering is often realized using partitioned convolution algorithms, which split the filter impulse responses into a sequence of sub filters and process these sub filters efficiently using frequency-domain methods (e.g., FFT-based convolution). Methods that split both, the signal and the filter, into uniformly-sized sub filters define a fundamental class of algorithms known as uniformly-partitioned convolution techniques. In these methods both operands, signal and filter, are usually partitioned with the same granularity. This contribution introduces uniformly-partitioned algorithms with independent partitions (block lengths) in both operands and regards viable transform sizes resulting from these. The relations of the algorithmic parameters are derived and the performance of the approach is evaluated.
Convention Paper 9205

2:30 pm

P19-3 Modeling the Nonlinear Behavior of Operational Amplifiers—*Robert-Eric Gaskell*, McGill University, Montreal, QC, Canada; GKL Audio Inc., Montreal, QC, Canada

Due to the gain-bandwidth characteristics of operational amplifiers, their nonlinearities are frequency dependent, showing a rise in distortion at higher frequencies. Depending on the circuit and system implementations, this distortion can be significant to listener perception of sonic character and quality and is therefore relevant to models of op amp-based analog equipment. Power-series models of the harmonic signature of various op amp nonlinearities are developed with and without this frequency dependence. Listening tests are performed to determine the extent to which the distortion characteristic of the model must match that of the real component to create a perceptually similar result.
Convention Paper 9206

3:00 pm

P19-4 More Cowbell: A Physically-Informed, Circuit-Bendable, Digital Model of the TR-808 Cowbell—*Kurt James Werner, Jonathan S. Abel, Julius O. Smith, III*, Stanford University, Stanford, CA, USA

We present an analysis of the cowbell voice circuit from the Roland TR-808 Rhythm Composer. A digital model based on this analysis accurately emulates the original. Through the use of physical and behavioral models of each sub-circuit, this model supports accurate emulation of circuit-bent extensions to the voice's original behavior (including architecture-level alterations and component substitution). Some of this behavior is very complicated and is inconvenient or impossible to capture accurately through black box modeling or structured sampling. The band pass filter sub-circuit is treated as a case study of how to apply Mason's gain formula to finding the continuous-time transfer function of an analog circuit.
Convention Paper 9207

3:30 pm

P19-5 A Modal Architecture for Artificial Reverberation with Application to Room Acoustics Modeling —*Jonathan S. Abel*,[1] *Sean Coffin*,[1] *Kyle Spratt*[2]
[1]Stanford University, Stanford, CA, USA
[2]University of Texas, Austin, Austin, TX, USA

The modal analysis of a room response is considered, and a computational structure employing a modal decomposition is introduced for synthesizing artificial reverberation. The structure employs a collection of resonant filters, each driven by the source signal and their outputs summed. With filter resonance frequencies and dampings tuned to the modal frequencies and decay times of the space, and filter gains set according to the source and listener positions, any number of acoustic spaces and resonant objects may be simulated. Issues of sufficient modal density, computational efficiency and memory use are discussed. Finally, models of measured and analytically derived reverberant systems are presented, including a medium-sized acoustic room and an electro-mechanical ➡

spring reverberator.
Convention Paper 9208

4:00 pm

P19-6 The Procedural Sound Design of Sim Cell— *Leonard J. Paul*, School of Video Game Audio, Vancouver, Canada

Synthesis was used to generate all of the audio for the sound design of educational game Sim Cell using the open source language Pure Data [1]. A primary advantage of using Pure Data is that it can be easily embedded into games for iOS, Android, and other platforms. This paper illustrates different examples of how synthesis can be effectively used in video games in contrast to more conventional contemporary audio production methods such as sampling. Synthesis allows for the accurate rendering of high resolution audio easily in addition to very high rates of data compression when compared to sampling.
Convention Paper 9209

4:30 pm

P19-7 OBRAMUS: A System for Object-Based Retouch of Amateur Music—*Jordi Janer*,[1] *Stanislaw Gorlow*,[1] *Keita Arimoto*[2]
[1]Universitat Pompeu Fabra, Barcelona, Catalunya, Spain
[2]Yamaha Corporation, Iwata, Shizuoka, Japan

In the more recent past, the area of semantic audio has become an object of special attention due to the increase in attractiveness of signal representations that allow manipulations of audio on a symbolic level. The semantics usually refer to audio objects, such as instruments, or musical entities, such as chords or notes. In this paper we present a system for making minor corrections to amateur piano recordings based on a nonnegative matrix factorization. Acting as middleman between the signal and the user, the system enables a simple form of musical recomposition by altering pitch, timbre, onset, and offset of distinct notes. The workflow is iterative, that is the result improves stepwise through user intervention.
Convention Paper 9210
[Paper not presented but available for purchase]

Session P20 Sunday, Oct. 12
1:30 pm—5:00 pm Room 309

APPLICATIONS IN AUDIO—PART 1

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

1:30 pm

P18-1 Producing Interactive Immersive Sound for MPEG-H: A Field Test for Sports Broadcasting—*Hanne Stenzel, Ulli Scuda*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

The present paper gives a practical example how broadcast content can be produced for MPEG-H. Existing production workflows are investigated with the question in mind, what needs to be adapted in order to make use of audio objects and immersive 3D-Audio provided by the new broadcast standard. After a short introduction to the features of MPEG-H, practical use cases are presented, such as immersive mixes and interactive personalized audio. In a field test, two sports events were accompanied and original audio material was gathered. Recording methods were tested to see how much additional effort is needed to make use of the mentioned features. The results show that already existing TV productions techniques can be used to provide enough audio material for interactive TV mixes. With little additional effort immersive 3D-audio environments can be created.
Convention Paper 9211

2:00 pm

P20-2 Headstock Resonances in the Electric Bass Guitar —*Bryan Martin*,[1,2] *Goran Petrovic*[1]
[1]McGill University, Montreal, QC, Canada
[2]Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, QC, Canada

The solid-body electric bass has long been established as a staple in jazz and popular music. This investigation examines the resonant characteristics of the headstock. Sine sweep techniques are employed to extract resonant characteristics from the headstock and compares these with those of the plucked open strings. It was found that there appears to be a correlation between the integrity of the open string resonances in the headstock with the output sound quality of the instrument.
Convention Paper 9212

2:30 pm

P20-3 Requirements Specification for Amplifiers and Power Supplies in Active Loudspeakers —*Henrik Schneider, Lasse C. Jensen, Lars Press Petersen, Arnold Knott, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

This work aims to provide designers with a method to develop a requirements specification for power supplies and amplifiers in active loudspeakers. The motivation is to avoid over-sizing and unnecessary cost. A realistic estimation of the power supplied during playback of audio in a given loudspeaker is obtained by considering a wide range of audio source material, loudness normalization of the source material, crossover filtering, driver characteristics as well as a perceived maximum loudness/volume level. The results from analyzing a sub-woofer and a woofer reveals the peak power, peak voltage, peak current, and apparent power—thus providing a solid foundation for a requirement specification.
Convention Paper 9213

3:00 pm

P20-4 Multiphysical Simulation Methods for Loudspeakers—Advanced CAE-Based Simulations of Motor Systems—*Alfred Svobodnik*,[1] *Roger Shively*,[2] *Marc-Olivier Chauveau*[3]
[1]Konzept-X GmbH, Karlsruhe, Germany
[2]JJR Acoustics, LLC, Seattle, WA, USA
[3]Moca Audio, Tours, France

This is the first in a series of papers on the details of loudspeaker design using multiphysical computer aided engineering simulation methods. In this paper, the simulation methodology for accurately modeling the electromagnetics of loudspeakers will be presented. Primarily, the creation of a useful impedance curve in the virtual world will be demonstrated. The influences of the mechanical mounting will also be illustrated, as well as the inherent non-linearities of the loudspeaker motor. Those non-linearities will be illustrated through the correct simulation of the electromagnetic driving force, which has an influence on all loudspeakers, and the voice coil inductance, which can have a profound influence on midrange and high frequency loudspeakers. Results will be presented, correlating the simulated model results to the measured physical parameters and to the impedance curve. From that, the important aspects of the modeling that determine its accuracy will be discussed.
Convention Paper 9214

3:30 pm

P20-5 MotionMix: A Gestural Audio Mixing Controller—*Jarrod Ratcliffe*, New York University, New York, NY, USA

This paper presents a control interface for stereo mixing using real time computer vision. The user's sense of depth and panorama are improved over the traditional channel strip, while broad accessibility is maintained by integrating the system with Digital Audio Workstation (DAW) software and implementing a system that is portable and affordable. To provide the user with a heightened sense of sound spatialization over the traditional channel strip, the concept of depth is addressed directly using the stage metaphor. Sound sources are represented as colored spheres in a graphical user interface to provide the user with visual feedback. Moving sources back and forward controls volume, while left to right controls panning. Preliminary evaluation is conducted through a pilot study, and user feedback is considered regarding future applications of the interface.
Convention Paper 9215

4:00 pm

P20-6 An Associative Shared Memory Approach to Audio Connection Management—*Andrew Eales*,[1] *Richard Foss*[2]
[1]Wellington Institute of Technology, Wellington, New Zealand;
[2]Rhodes University, Grahamstown, Eastern Cape, South Africa

A distributed, associative memory that advertises audio streams and represents audio connections between networked audio devices is described. Characteristic features of a shared, associative memory are discussed, and three parameter-based models that represent audio signals and audio connections are introduced. Connection management is then discussed with reference to a distributed, associative memory environment. This environment allows changes made to audio connections to be automatically propagated to all networked devices, while also eliminating potential race conditions between connection requests. Additionally, connection management applications can be shared between different networked devices and controllers.
Convention Paper 9216

4:30 pm

P20-7 Utilizing Gesture Recognition and Ethernet AVB for Distributed Surround Sound Control —*Mitchell Hedges, Richard Foss*, Rhodes University, Grahamstown, Eastern Cape, South Africa

Gesture recognition has become a preferred approach to the control of various systems. This allows users of the system to interact without having to use any controls or equipment. This paper investigates the use of gesture recognition in order to select and transport audio tracks over an Ethernet AVB network to speaker endpoints. The research uses equipment that is commercially available and relatively cost efficient. The endpoints receive audio samples that are encapsulated within network packets and processes them. The audio tracks are mixed at the endpoints according to gain ratios that will change and be different for each endpoint.
Convention Paper 9217

Product Design Session 17 — Sunday, October 12
1:30 pm – 3:00 pm — Room 402 AB

TESTING IN THE AGE OF GLOBAL PRODUCTION

Presenter: **Paul Messick**, Avermetrics, Los Angeles, CA, USA

Good design is hard. Good testing should be easy. After designing a product, making it work, and making sure it meets all the specs, often little time is left to design the right tests to make sure the product is being built correctly at the factory. Instead, already overworked design engineers are asked at the last minute to come up with the factory production test regimen. Its no wonder that production testing is too often done poorly and commonly ends up looking eerily like R&D—costs and all.

Through this tutorial you will learn more about:

• How to do the "right amount" of testing, without false positives or false negatives
• Different ways to implement product testing in the factory
• How to make the most cost effective, yet thorough, end-to-end test solution for your products
• What to look out for when setting up production line tests both domestically and overseas
• How to manage production test remotely and better manage testing costs
• The difference between product engineering verification and production test
• How test fixtures can greatly speed up production testing
• Subassembly testing vs. Board-level testing vs. Finished Goods testing
• Design for Testing and Built-in test
• Testing high-volume products quickly and speed up testing low-volume products
• The non-linear relationship between test time and test expense
• Making testing count: collecting metrics and improving delivered products

➥

Special Event
SOUND IS THE CONDUIT TO THE ARTIST HEART
Sunday, October 12, 1:30 pm – 3:00 pm
Room 404 AB

Moderator: **Jack Joseph Puig**, Multi-GRAMMY Award winning Recording Engineer

Panelists: *Dean Bolte*, Managing Director, Americas at Omnifone
Joel Clarke, Motorola Mobile Devices
Chris Dorian, Sr. Area Director/Business Sales at T-Mobile USA
Aja Schmit, Vice Bullitt Group Ltd.
Devon Worrell, Mobile Audio Architect, Intel

Ever since the consumer-branded Yamaha NS-10 speaker became ubiquitous in almost every professional recording studio, the music industry has seen the worlds of professional and consumer audio collide with the goal of creating an ecosystem that allows the professional to make the right creative choices, in order for the translation of the artists' intent to be fully realized and understood by the consumer. Now that "mobile" is how most consumers are listening to music, what is the future, and how do manufacturers in the consumer space view the future of mobile audio and pro audio intersecting? This provocative panel will address and discuss this paradigm shift while exploring the ramifications that will affect everyone in the industry, from audio manufacturer to end user.

Broadcast/Streaming Media Session 16
Sunday, October 12 2:00 pm – 5:00 pm
Room 407

SOCIETY OF BROADCAST ENGINEERS EXAMS

The Society of Broadcast Engineers (SBE) established a certification program almost 40 years ago to recognize and raise the professional status of broadcast engineers by providing standards of professional competence. It has become recognized in the industry as the primary method of verifying the attainment of knowledge and experience. With the industry constantly changing, the broadcast engineer, certified by the SBE, must keep up with those changes by recertifying every five years. From the certified operator to the Certified Professional Broadcast Engineer, SBE has a certification for every broadcast engineer, technician and operator. SBE exams will take place on October 12 at 2:00 pm during the annual AES convention. Applicants are encouraged to apply before the exams by going to www.sbe.org and accessing the certification applications. You may apply on-site for the Certified Broadcast Technologist or the Certified Broadcast Networking Technologist exams. If you wish to apply for the broadcast engineer, senior engineer or specialist certifications then you would need to pre-register by September 19. To learn more about the SBE Certification Program, visit the SBE website, www.sbe.org.

This session is open to all badges.

Live Sound Seminar 12 Sunday, October 12
2:00 pm – 4:00 pm Room 406 AB

AC POWER AND GROUNDING

Chair: **Bruce Olson**, Olson Sound Design, LLC, Minneapolis, MN, USA; AFMG Services North America, LLC, Minneapolis, MN, USA

Panelists: *Ken Fause*, Auerbach Pollock Friedlander, San Francisco, CA, USA
Jaime Fox, The Engineering Enterprise, Alameda, CA, USA
Bill Whitlock, Whitlock Consulting, Oxnard, CA, USA

There is a lot of misinformation about what is needed for AC power for events. Much of it has to do with life-threatening advice. This panel will discuss how to provide AC power properly and safely and without causing noise problems. This session will cover power for small to large systems, from a couple boxes on sticks up to multiple stages in ballrooms, road houses, and event centers; large scale installed systems, including multiple transformers and company switches, service types, generator sets, 1ph, 3ph, 240/120 208/120. Get the latest information on grounding and typical configurations by this panel of industry veterans.

Sunday, October 12 2:00 pm Room 405
Technical Committee Meeting on Audio for Games

Project Studio Expo PSE16 PSE STAGE
MOTU
Sunday, October 12, 2:00 pm – 2:45 pm

Live Sound Expo LSE16 LSE STAGE
STUMP THE PANEL
Sunday, October 12, 2:00 pm – 2:50 pm

Presenters: **Dave Rat**, Rat Sound; Sound System Designer, Consultant, Audio Engineer - Red Hot Chili Peppers, Soundgarden, Blink 182
Robert Scovill, FOH for Prince, Rush, Tom Petty
Dave Shadoan, President and Co-founder, Sound Image

Live sound audio professionals gather for a lively Q&A covering the gamut of sound reinforcement topics. They'll come equipped with entertaining tales from the trenches (names may be withheld to protect the guilty) and to take your questions.

Recording and Production Session 8
Sunday, October 12 2:30 pm – 4:00 pm
Room 409 AB

MASTERING FOR ENGINEERS

Presenters: **Andres Mayo**, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina
Darcy Proper, Wisseloord Studios, Hilversum, The Netherlands
Ronald Prent, Wisseloord Studios, Eemnes, The Netherlands

The presenters will "role-play" the real situation with a client in a mastering session. All the issues that may come up in real life will be addressed in order to demonstrate how important it is as engineers (and/or facility managers) to make clients feel confident and at ease but still keeping control of the session. In other words, how to develop interpersonal skills that become crucial to the music production business.

Historical Event H6
AUTHENTIC REPLICATION AND MODELING OF VINTAGE AUDIO GEAR
Sunday, October 12, 2:30 pm – 4:30 pm
Room 408 A

Moderator: **Dennis Fink**, Crestron Electronics, Rockleigh, NJ, USA

Presenters: *Dave Berners*, Universal Audio, Scotts Valley, CA, USA
Dave Derr, Empirical Labs, Parsippany, NJ, USA
Steve Jackson, Pulse Techniques, LLC, Fort Collins, CO, USA
Dan Kennedy, Great River electronics, St. Paul, MN, USA

A panel of manufacturers who are engaged in manufacturing modern versions of classic audio products will each profile one or more products, describing some of the methods which they use to duplicate the classic performance of the pieces. Dennis Fink, formerly with UREI, MSD consulting to JBL and others, Universal Audio and Fink Audio, will moderate and begin with an outline of the different categories of product types. The participants will include representatives of manufacturers who do both hardware as well as software plug-ins. Each manufacturer will present products, methods and techniques, with discussion of measured and sonically-judged faithfulness and/or improvements, and then panel interaction, and questions from the audience.

Sound for Picture 7 Sunday, October 12
3:00 pm – 5:00 pmRoom 408 B

RECORDING PRODUCTION SOUND—
A MASTER CLASS

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

Panelists: *Peggy Names*
Jay Patterson, Engineering For Production, Venice

The craft of recording live dialog on the set is one of the most unique areas of sound recording. In many cases the other technicians on the set, including the director, conspire to make the process of getting quality recordings very difficult. And with this work often done outdoors, day and night, sunshine, pouring rain, or drifting snow, the production sound team need to be clever and find solutions in real time. This workshop with a leading Production Sound mixer and top boom operator will shed some light on this diverse recording craft.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Product Design Session 18 Sunday, October 12
3:15 pm – 4:45 pm Room 402 AB

SYSTEM ENGINEERING AND GLOBAL SOURCING

Presenter: **Mike Klasco**, Menlo Scientific Ltd., Richmond, CA, USA

In this interactive Tutorial, Mike Klasco will lead a discussion about working with OEM and ODM product development partners and product managers in how to successfully bring products to market. Projects range from designing from a clean sheet of paper to brands that are in a hurry and are looking for how to pick an global vendor and grab something off-the-shelf to fill out a gap in their product line. This will include discussions about best practices in working with: (1) Off-shore manufacturing and supply chain, (2) Design services, (3) Engineering services, (4) Off the shelf and semi-custom subassemblies.

Upcoming AES Conventions and Conferences

56th International Conference
“Audio for Games”
London, UK
2015 February 11–13

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57th International Conference
“The Future of Audio Entertainment Technology: Cinema, Television, and the Internet”
Hollywood, CA, USA
2015 March 6–8

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138th CONVENTION
Warsaw, Poland
2015 May 7–10

•

139th CONVENTION
New York, NY, USA
2015 October 29–November 1

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For the latest information on AES conventions and conferences, visit the AES Web site at www.aes.org