AES 139TH CONVENTION PROGRAM

OCTOBER 20 – NOVEMBER 1, 2015

JACOB JAVITS CONVENTION CENTER, NY, NY

At 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 136th Convention Peer-Reviewed Paper Award.

* * * * *

The Winner of the 139th AES Convention Best Peer-Reviewed Paper Award is:

Horns Near Reflecting Boundaries—*Bjørn Kolbrek*, Norwegian University of Science and Technology, Trondheim, Norway

Convention Paper 9412

To be presented on Friday, October 30, in Session 9—Transducers—Part 3

* * * * *

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 139th 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.

* * * * *

The Winner of the 139th AES Convention Student Paper Award is:

Low Impedance Voice Coils for Improved Loudspeaker Efficiency—*Niels Elkjær Iversen, Arnold Knott, Michael A. E. Andersen,* Technical University of Denmark, Kgs. Lyngby, Denmark

Convention Paper 9389

To be presented on Friday, October 30, in Session 6—Transducers—Part 2

Session P1 — **Thursday, October 29**
9:00 am – 12:30 pm — **Room 1A08**

SIGNAL PROCESSING

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

9:00 am

P1-1 Time-Frequency Analysis of Loudspeaker Sound Power Impulse Response—*Pascal Brunet, Allan Devantier, Adrian Celestinos*, Samsung Research America, Valencia, CA, USA

In normal conditions (e.g., a living room) the total sound power emitted by the loudspeaker plays an important role in the listening experience. Along with the direct sound and first reflections, the sound power defines the loudspeaker performance in the room. The acoustic resonances of the loudspeaker system are especially important, and thanks to spatial averaging, are more easily revealed in the sound power response. In this paper we use time-frequency analysis to study the spatially averaged impulse response and reveal the structure of its resonances. We also show that the net effect of loudspeaker equalization is not only the attenuation of the resonances but also the shortening of their duration.
Convention Paper 9354

9:30 am

P1-2 Low-Delay Transform Coding Using the MPEG-H 3D Audio Codec—*Christian R. Helmrich,*[1] *Michael Fischer*[2]
[1] International Audio Laboratories, Erlangen, Germany
[2] Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Recently the ISO/IEC MPEG-H 3D Audio standard for perceptual coding of one or more audio channels has been finalized. It is a little-known fact that, particularly for communication applications, the 3D Audio core-codec can be operated in a low-latency configuration in order to reduce the algorithmic coding/decoding delay to 44, 33, 24, or 18 ms at a sampling rate of 48 kHz. This paper introduces the essential coding tools required for high-quality low-delay coding–transform splitting, intelligent gap filling, and stereo filling–and demonstrates by means of blind listening tests that the achievable subjective performance compares favorably with, e.g., that of HE-AAC even at low bit-rates.
Convention Paper 9355

10:00 am

P1-3 Dialog Control and Enhancement in Object-Based Audio Systems—*Jean-Marc Jot,*[1] *Brandon Smith,*[2] *Jeff Thompson*[2]
[1] DTS, Inc., Los Gatos, CA, USA
[2] DTS, Inc., Bellevue, WA, USA

Dialog is often considered the most important audio element in a movie or television program. The potential for artifact-free dialog salience personalization is one of the advantages of new object-based multichannel digital audio formats, along with the ability to ensure that dialog remains comfortably audible in the presence of concurrent sound effects or music. In this paper we review some of the challenges and requirements of dialog control and enhancement methods in consumer audio systems, and their implications in the specification of object-based digital audio formats. We propose a solution incorporating audio object loudness metadata, including a simple and intuitive consumer personalization interface and a practical head-end encoder extension.
Convention Paper 9356

10:30 am

P1-4 Frequency-Domain Parametric Coding of Wideband Speech–A First Validation Model—*Aníbal Ferreira,*[1] *Deepen Sinha*[2]
[1] University of Porto, Penafiel, Portugal
[2] Audio Technologies and Codecs, Inc., Newark, NJ, USA

Narrow band parametric speech coding and wideband audio coding represent opposite coding paradigms involving audible information, namely in terms of the specificity of the audio material, target bit rates, audio quality, and application scenarios. In this paper we explore a new avenue addressing parametric coding of wideband speech using the potential and accuracy provided by frequency-domain signal analysis and modeling techniques that typically belong to the realm of high-quality audio coding. A first analysis-synthesis validation framework is described that illustrates the decomposition, parametric representation, and synthesis of perceptually and linguistically relevant speech components while preserving naturalness and speaker specific information.
Convention Paper 9357

11:00 am

P1-5 Proportional Parametric Equalizers—Application to Digital Reverberation and Environmental Audio Processing—*Jean-Marc Jot,* DTS, Inc., Los Gatos, CA, USA

Single-band shelving or presence boost/cut filters are useful building blocks for a wide range of audio signal processing functions. Digital filter coefficient formulas for elementary first- or second-order IIR parametric equalizers are reviewed and discussed. A simple modification of the classic Regalia-Mitra design yields efficient solutions for tunable digital equalizers whose dB magnitude frequency response is proportional to the value of their gain control parameter. Practical applications to the design of tone correctors, artificial reverberators and environmental audio signal processors are described.
Convention Paper 9358

11:30 am

P1-6 Comparison of Parallel Computing Approaches of a Finite-Difference Implementation of the Acoustic Diffusion Equation Model—*Juan M. Navarro,*[1] *Baldomero Imbernón,*[1] *José J. López,*[2] *José M. Cecilia*[1]
[1] Universidad Católica San Antonio - Murcia, Guadalupe (Murcia), Spain
[2] Universitat Politcnica de Valencia, Valencia, Spain

The diffusion equation model has been intensively researched as a room-acoustics simulation algorithm during last years. A 3-D finite-difference implementation of this model was proposed to evaluate the propagation over time of sound field within rooms. Despite the computational saving of this model to calculate the room energy impulse response, elapsed times are still long when high spatial resolutions and/or simulations in several frequency bands are needed. In this work several data-parallel approaches of this finite-difference solution on Graphics Processing Units are proposed using a compute unified device architecture programming model. A comparison of their performance running on different models of Nvidia GPUs is carried out. In general, 2D vertical block approach running in a Tesla K20C shows the best speed-up of more than 15 times versus CPU version.
Convention Paper 9359

12:00 noon

P1-7 An Improved and Generalized Diode Clipper Model for Wave Digital Filters—*Kurt James Werner,*[1] *Vaibhav Nangia,*[1] *Alberto Bernardini,*[2] *Julius O. Smith, III,*[1] *Augusto Sarti*[2]
[1] Center for Computer Research in Music and Acoustics (CCRMA), Stanford University, Stanford, CA, USA
[2] Politecnico di Milano, Milan, Italy

We derive a novel explicit wave-domain model for "diode clipper" circuits with an arbitrary number of diodes in each orientation, applicable, e.g., to wave digital filter emulation of guitar distortion pedals. Improving upon and generalizing the model of Paiva et al. (2012), which approximates reverse-biased diodes as open circuits, we derive a model with an approximated correction term using two Lambert W functions. We study the energetic properties of each model and clarify aspects of the original derivation. We demonstrate the model's validity by comparing a modded Tube Screamer clipping stage emulation to SPICE simulation.
Convention Paper 9360

Session P2 — **Thursday, October 29**
9:00 am – 12:00 noon — **Room 1A07**

AUDIO EDUCATION

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

9:00 am

P2-1 LabVIEW as a Music Synthesizer Laboratory Learning Environment—*Edward B. Stokes,*[1] *Ed Doering*[2]
[1] University of North Carolina at Charlotte, Charlotte, NC, USA
[2] Rose-Hulman Institute of Technology, Terre Haute, IN, USA

Most electrical engineering (EE) students are familiar with LabVIEW. This graphical programming environment is commonly used in university EE educational and research labs to facilitate data acquisition and processing using a suite of built-in mathematical, DSP, and communication functions. LabVIEW is particularly adept at emulating control panels with a variety of knobs, sliders, and gages. The audio functionality of LabVIEW, along with its "knobby" user interface, makes it ideal for exploration of music synthesis concepts by EE students. In this paper several types of music synthesis are explored in LabVIEW. Implementation of these in elective EE coursework gives EE students a unique opportunity to experience abstract concepts such as waveforms, frequency, filtering, and envelopes through their auditory cortex, reinforcing what they have learned through traditional pedagogy, and also provides EE students an introduction to some basic audio engineering (AE) concepts.
Convention Paper 9361

9:30 am

P2-2 A Model for International and Industry-Engaged Collaboration and Learning—*Mark Thorley*, Coventry University, Coventry, Warwickshire, UK

Traditional barriers of geography, organization, and culture and being broken down by emerging technology [1]. In the record-

ing industry, professionals often collaborate on projects globally, engaging in what Tapscott and Williams [2] call "peer-production." The potential in these concepts extends to those developing their expertise—they can connect with peers and industry professionals on a global scale. Despite the potential however, most Higher Education institutions fail to engage for cultural reasons. This paper outlines a model for collaborative learning explored and developed through a project funded by the UK's Higher Education Academy. The project involved Coventry University and industry organization JAMES as well as a number of other international partners. The paper looks at the pedagogical background to the project, some typical activities undertaken before summarizing the key outcomes and opportunities for further work.
Convention Paper 9362

10:00 am

P2-3 From Creativity to Science and Back Again: Supporting Audio Students Through Active Teaching Approaches—*Jason Fick*, The Art Institute of Dallas, Dallas, TX, USA

Many students enrolling in audio programs are not fully aware of the importance of science for the audio professional. Typically these students are creative but may have deficiencies in math and science. My goal as an instructor is to minimize the negative associations of these subjects through active lesson plans that stress practical audio situations in a compelling and interactive manner. As a result, students develop confidence through their ability to use science as a tool to both solve audio problems and create expressive art forms. My approaches empower them to succeed in early courses, which facilitate creative applications in later classes. Consequently, students are better prepared for the job force using skills that promote both technical and creative capacities.
Convention Paper 9363

10:30 am

P2-4 The Use of Digital Reverberation Projects to Teach Audio Signal Processing—*Benjamin D. McPheron, Kelsey M. Cintorino, Nicholas J. Benoit, Abdulrahim S. Hasan, Kevin J. Oliveira, Andrew D. Senerchia, Daniel M. Wisniewski*, Roger Williams University, Bristol, RI, USA

Hands-on application is essential to the development of practicing engineers capable of designing and implementing digital signal processing methods. The application of digital signal processing to audio applications provides students with instantly gratifying results and further develops future audio engineering professionals. In order to provide deeper understanding of audio processing techniques, students can be presented with projects that challenge them to create unique applications or methods in the field of audio processing. This work reports the project framework and outstanding student work resulting from implementing this method in a digital signal processing course, as well as the assessment strategy used to evaluate student understanding of key audio engineering techniques.
Convention Paper 9364

11:00 am

P2-5 Audio Recording and Production Education: Skills New Hires Have and Where They Reported Learning Them—*Doug Bielmeier*, Indiana University-Purdue University Indianapolis, Indianapolis, IN, USA

To understand how audio recording and production programs meet the needs of the larger entertainment industry, this study directly asked new hires what skills they have and where they were learned. In the New Hires Survey they were asked to rate the level of proficiency of their skills, where they learned these skills, and what skills they need to learn. The new hires reported learning basic technical skills during formal audio recording and production training but learned social and communication skills on their own or on the job. They requested a greater emphasis on career-critical areas of live sound and music business. Further research is recommended to understand industry needs, identify best practices for the acquisition of skills, and to determine how educational institutions can keep pace with the ever-changing entertainment industry.
Convention Paper 9365

11:30 am

P2-6 Case Study: Expanding Audio Production Facilities at Ohio University to Accommodate Student Needs—*Kyle P. Snyder*, Ohio University, School of Media Arts & Studies, Athens, OH, USA

Creating a recording facility is equal parts art and science. However, designing and adapting recording studios in higher education environments presents several challenges unseen within the commercial arena. In its third major design iteration since the formation of the College of Communication in 1968, the School of Media Arts & Studies has expanded its facilities to include a new mixing and mastering suite, an expansive 5.1 post-production and critical listening facility, and numerous classrooms and additional lab spaces, more than tripling the space available to faculty, graduate, and undergraduate students.
Convention Paper 9366

Tutorial 1 **Thursday, October 29**
9:00 am – 10:30 am **Room 1A22**

PARAMETRIC SPATIAL AUDIO PROCESSING: AN OVERVIEW AND RECENT ADVANCES*

Presenters: **Emanuël A. P. Habets**, International Audio Laboratories Erlangen, Erlangen, Germany
Oliver Thiergart, International Audio Laboratories Erlangen, Erlangen, Germany

Parametric spatial processing is a promising and emerging technique that is fundamentally different from traditional spatial processing techniques. First, a relatively simple sound field model is adopted and the parameters of the model (such as for example the direction of arrival and diffuseness), are estimated in a time-frequency domain. Second, the estimated parameters are used to process the received microphone signals. The compact and efficient representation of the sound field can be used to develop algorithms for different applications. In this tutorial different sound field models and corresponding parameter estimation techniques will be presented. We will then focus on selected applications such as speech enhancement (directional filtering and dereverberation), acoustical zooming, spatial audio communication, and spatial sound reproduction. Finally, we show how one of the parametric representations can be used to create virtual microphone signals with user-defined pickup patterns.

**This session is presented in association with the AES Technical Committee on Spatial Audio.*

Tutorial 2 **Thursday, October 29**
9:00 am – 11:00 am **Room 1A10**

MICROPHONES—CAN YOU HEAR THE SPECS?*

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

Panelists: *Jürgen Breitlow*, Georg Neumann Berlin, Berlin, Germany
David Josephson, Josephson Engineering, Inc., Santa Cruz, CA, USA
Helmut Wittek, SCHOEPS GmbH, Karlsruhe, Germany

There are lots and lots of microphones available to the audio engineer. The final choice is often made on the basis of experience or perhaps just habits. (Sometimes the mic is chosen because of the Looks…). Nevertheless, there is valuable information in the microphone specifications. This tutorial demystify the most important microphone specs and provide the attendee with up-to-date information on how these specs are obtained and understood and how the numbers relate to the perceived sound. It takes a critical look on how specs are presented to the user, what to look for, and what to expect.

**This session is presented in association with the AES Technical Committee on Microphones and Applications*

Workshop 1 **Thursday, October 29**
9:00 am – 10:00 am **Room 1A14**

HEARING SMART

Chair: **Kathy Peck**, ED and Co-founder H.E.A.R., San Francisco, CA, USA

Moderator: **Dan Beck**, Trustee of The Music Performance Trust Fund

Panelistss: *Richard Einhorn*, Hearing Impaired Musician/ Advocate
Marty Garcia, Future Sonics
S. Benjamin Kanters, Columbia College, Chicago, IL, USA; Hear Tomorrow
Joseph Montano, Chief of Audiology and Speech Language Pathology at New York Presbyterian Hospital-Weill Cornell Medical Center, New York, NY, USA

(2015) WHO—World Health Organization—sites 1.1 billion people at risk of hearing loss a serious threat posed by exposure to recreational noise due to the unsafe use of personal audio devices and exposure to damaging levels of sound at noisy entertainment venues such as nightclubs, bars, and sporting events. (1989) Nonprofit H.E.A.R, Hearing Education and Awareness for Rockers an early proponent with founding support of Pete Townshend of the Who launched worldwide grassroots initiatives. In efforts to unite the music industry and the hearing health/medical community, raise awareness and improve hearing conservation for performers, personnel, and consumers of music to insure its continued creation, performance, and enjoyment a panel of music and medical industry hearing conservation experts were brought together to discuss hearing education and prevention.

Product Development 1 **Thursday, October 29**
9:00 am – 10:30 am **Room 1A13**

ALMOST EVERYTHING YOU EVER WANTED TO KNOW ABOUT LOUDSPEAKER DESIGN

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.

Tutorial 3 **Thursday, October 29**
9:30 am – 11:00 am **Room 1A06**

VOX POP – TRANSDUCER TECHNOLOGIES AND THEIR CONNECTION TO VOCAL PERFORMANCE TECHNIQUES*

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

For the first century of music recording, the sound of the recorded voice was very much driven by the capabilities of the audio technology available at the time. We are less limited by the extraordinary gear available today, leaving us with a broad range of creative options for tracking the all-important lead vocal. In this tutorial, Alex U. Case highlights what we might learn from audio history to drive our decisions for getting the right vocal sound at our next gig.

Game Audio 1 **Thursday, October 29**
9:30 am – 10:30 am **Room 1A21**

REINVENTING THE SOUND FOR CALL OF DUTY: ADVANCED WARFARE

Presenter: **David Swenson**, Sledgehammer Games, Activision

How do you go about reinventing the audio experience for a well-established game like Call of Duty? How do you make the audio sound fresh, next-gen and "Advanced" in the 11th installment of a franchise? The answers to these questions may surprise you. Learn how Sledgehammer Games employed some unorthodox methods to create an all new sound for Call of Duty.

Spatial Audio Demo 1 **Thursday, October 29**
9:30 am – 10:30 am **Room 1A18**

OBJECT BASED IMMERSIVE / 3D AUDIO PRODUCTION

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

Some new object based formats like Dolby Atmos, MDA, MPEG-H, and ECMA 407 were introduced to the audio industry recently. But how to create content, how to master, how to deliver, and finally content applications are still a bit unclear for a lot of audio producer and service provider. Some insides, tools, strategies and listening examples are given in this event.

Archiving 1 **Thursday, October 29**
10:15 am – 11:15 am **Room 1A14**

THE MEDIA DIGITIZATION AND PRESERVATION PROJECT AT INDIANA UNIVERSITY: A CASE STUDY IN LARGE-SCALE PRESERVATION OF TIME-BASED MEDIA

Presenters: **Andrew Dapuzzo**, Memnon
Mark Hood, Indiana University, Bloomington, IN, USA
Konrad Strauss, Indiana University, Bloomington, IN, USA

Indiana University possesses an unusually rich collection of time-based media that document subjects of enduring value. They include wax cylinder sound recordings of Native American music, performances of world renowned musicians, lectures by the leading thinkers of the 20th and 21st centuries, one of the largest collections of ethnographic field recordings, the film collections of Peter Bogdanovich and David Bradley, the collection of the Kinsey Institute for Research in Sex Gender and Reproduction, and many other unique and rare items. In 2010 a task force conducted an inventory of IU collections and identified an excess of 500,000 items that were in need of preservation. As a result, the university provided $15 million in funding to build a facility to digitize and preserve the collections; and make them available to researchers worldwide. This workshop will provide an overview of the survey process, the design of the preservation facility, and an overview of the digitization workflow. We will also discuss the lessons learned from building such a large project and how processes and methodology can be scaled down to work for the small library or archive.

Spatial Audio Demo 2 **Thursday, October 29**
10:30 am – 11:30 am **Room 1A18**

IMMERSIVE / 3D AUDIO FOR GAMES

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

Today immersive/3D audio is a well know task for film as well for music production. Approaches and technologies have been discussed and presented at AES conventions for some years. But immersive / 3D audio for games that involves the gamer much more than ever before is just coming up. These sounds, strategies, and applications will be shown in this event.

This event is part of the Games Track

Tutorial 4 **Thursday, October 29**
10:45 am – 12:15 pm **Room 1A22**

INTERACTIVE MUSIC: FUTURE LISTENING EXPERIENCES…

Presenters: **Justin Paterson**, London College of Music, University of West London, London, UK
Rob Toulson, Anglia Ruskin University, Cambridge, UK

Listeners have long been inspired to interact with commercial music and create new representations of popular releases. Vinyl offered many opportunities to reappropriate chart music, from scratching and tempo manipulation to mixing multiple songs. Nowadays, artists can engage their audience to interact with the music by offering mix stems for experimentation, a trend started by Nine Inch Nails in 2005 continuing to artists such as U2 in 2014. With the extended processing power of mobile devices, the opportunities for interactive music are limitless; both Bjork and Peter Gabriel have explored these new platforms in an interactive manner. This session will offer a history and context of interactive music and demonstrate potential future listening experiences. The presenters' will also showcase their own funded research activities, which focus on a new interactive listening platform for commercial music, allowing the listener to manipulate mix stems, evaluate alternate mixes and explore additional music content.

Workshop 2 **Thursday, October 29**
10:45 am – 12:45 pm **Room 1A21**

MIXING MUSIC*

Chair: **Richard King**, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

Panelists: *Buford Jones*, Meyer Sound
George Massenburg, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Shawn Murphy, Independent Engineer

Panel discussion and presentations from award-winning expert practitioners in the industry, describing the process of mixing, actual techniques used, and proven methodologies that have yielded successful results over the years. Focus will remain on real information as opposed to anecdotes, such as different ways to approach a mix, how to improve an existing mix, how to best interpret and address mix comments from an artist or a client, or the record label (it happens). A large portion of time will be left open for questions, so that the audience will have the chance to solicit specific and meaningful information from the panelists.

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

Product Development 2 **Thursday, October 29**
10:45 am – 12:15 pm **Room 1A13**

PRACTICAL LOUDSPEAKER PROCESSING FOR THE PRACTICING ENGINEER

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

Loudspeaker signal processing is making the transition from traditional analog designs to digital processing. This is being driven by the availability of digital content, the desire to have wireless products, and the promise of improved sound through digital signal processing. We cover the main concepts behind digital audio processing for loudspeakers. We use a hands-on approach and interactively build up the signal chain using graphical tools. We discuss crossovers, equalizers, limiters, and perceptual loudness controls. Key concepts are reinforced through examples and real-time demos. The session is aimed at the practicing audio engineer and we go easy on math and theory. Instead of writing code we leverage modern design tools and you will leave ready to design your own processing chain.

Session P3 **Thursday, Oct. 29**
11:00 am – 12:30 pm **Foyer**

POSTERS: TRANSDUCERS/PERCEPTION

11:00 am

P3-1 Predicting the Acoustic Power Radiation from Loudspeaker Cabinets: A Numerically Efficient Approach—*Mattia Cobianchi, Martial Rousseau*, B&W Group Ltd., West Sussex, UK

Loudspeaker cabinets should not contribute at all to the total sound radiation but aim instead to be a perfectly rigid box that encloses the drive units. To achieve this goal, state of the art FEM software packages and Doppler vibro-meters are the tools at our disposal. The modeling steps covered in the paper are: measuring and fitting orthotropic material properties, including damping; 3D mechanical modeling with a curvilinear coordinates system and thin elastic layers to represent glue joints; scanning laser Doppler measurements and single point vibration measurements with an accelerometer. Additionally a numerically efficient post-processing approach used to extract the total radiated acoustic power and an example of what kind of improvement can be expected from a typical design optimization are presented.
Convention Paper 9367

11:00 am

P3-2 New Method to Detect Rub and Buzz of Loudspeakers Based on Psychoacoustic Sharpness—*Tingting Zhou, Ming Zhang, Chen Li*, Nanjing Normal University, Nanjing, Jiangsu, China

The distortion detection of loudspeakers has been researched for a very long time. Researchers are committed to finding an objective way to detect Rub and Buzz (R&B) in loudspeakers that is in line with human ear feelings. This paper applies the psychoacoustics to distortion detection of loudspeakers and describes a new method to detect the R&B based on the psychoacoustic sharpness. Experiments show, comparing with existing objective detection methods of R&B, detection results based on the proposed method are more consistent with subjective judgments.
Convention Paper 9368

11:00 am

P3-3 Modal Impedances and the Boundary Element Method: An Application to Horns and Ducts—*Bjørn Kolbrek*, Norwegian University of Science and Technology, Trondheim, Norway

Loudspeaker horns, waveguides, and other ducts can be simulated by general numerical methods, like the Finite Element or Boundary Element Methods (FEM or BEM), or by a method using a modal description of the sound field, called the Mode Matching Method (MMM). BEM and FEM can describe a general geometry but are often computationally expensive. MMM, on the other hand, is fast, easily scalable, requires no mesh generation and little memory but can only be applied to a limited set of geometries. This paper shows how BEM and MMM can be combined in order to efficiently simulate horns where part of the horn must be described by a general meshed geometry. Both BEM-MMM and MMM-BEM couplings are described, and examples given.
Convention Paper 9369

11:00 am

P3-4 Audibility Threshold of Auditory-Adapted Exponential Transfer-Function Smoothing (AAS) Applied to Loudspeaker Impulse Responses—*Florian Völk*,[1,2] *Yuliya Fedchenko*,[1] *Hugo Fastl*[1]
[1] Technical University of Munich, Munich, Germany
[2] WindAcoustics UG (haftungsbeschränkt), Windach, Germany

A reverberant acoustical system's transfer function may show deep notches or pronounced peaks, requiring large linear amplification in the play-back system when used, for example, in auralization or for convolution reverb. It is common practice to apply spectral smoothing, with the aim of reducing spectral fluctuation without degrading auditory-relevant information. A procedure referred to as auditory-adapted exponential smoothing (AAS) was proposed earlier, adapted to the spectral properties of the hearing system by implementing frequency-dependent smoothing bandwidths. This contribution presents listening experiments aimed at determining the audibility threshold of auditory-adapted exponential smoothing, which is the maximum amount of spectral smoothing allowed without being audible. As the results depend on the specific acoustic system, parametrization guidelines are proposed.
Convention Paper 9371

11:00 am

P3-5 Developing a Timbrometer: Perceptually-Motivated Audio Signal Metering—*Duncan Williams*, University of Plymouth, Plymouth, UK

Early experiments suggest that a universally agreed upon timbral lexicon is not possible, and nor would such a tool be intrinsically useful to musicians, composers, or audio engineers. Therefore the goal of this work is to develop perceptually-calibrated metering tools, with a similar interface and usability to that of existing loudness meters, by making use of a linear regression model to match large numbers of acoustic features to listener reported timbral descriptors. This paper presents work towards a proof-of-concept combination of acoustic measurement and human listening tests in order to explore connections between 135 acoustic features and 3 timbral descriptors, brightness, warmth, and roughness.
Convention Paper 9372

11:00 am

P3-6 A Method of Equal Loudness Compensation for Uncalibrated Listening Systems—*Oliver Hawker, Yonghao Wang*, Birmingham City University, Birmingham, UK

Equal-loudness contours represent the sound-pressure-level-dependent frequency response of the auditory system, which implies an arbitrary change in the perceived spectral balance of a sound when the sound-pressure-level is modified. The present paper postulates an approximate proportional relationship between loudness and sound-pressure-level, permitting relative loudness modification of an audio signal while maintaining a constant spectral balance without an absolute sound-pressure-level reference. A prototype implementation is presented and accessible at [1]. Preliminary listening tests are performed to demonstrate the benefits of the described method.
Convention Paper 9373

Live Sound Seminar 1 **Thursday, October 29**
11:00 am – 12:45 pm **Room 1A12**

AC POWER AND GROUNDING

Chair: **Mike Sokol**

Panelists: *Steve Lampen*
Bill Sacks

Much misinformation remains about what is needed for AC power for events—much of it potentially life-threatening advice. This panel will discuss how to provide AC power properly and safely and without causing noise problems. The 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 proper power configurations by this panel of industry veterans.

Recording & Mastering 1 **Friday, October 30**
11:00 am – 12:30 pm **Room 1A23/24**

MASTER CLASS WITH JOHN CONGLETON—RECORDING ST. VINCENT, SWANS, AND MODEST MOUSE

Moderator: **Mike Tierney**, Mike Tierney LLC, New York, NY, USA

Presenter: **John Congleton**, St. Vincent, David Byrne, Clap Your Hands Say Yeah, Modest Mouse, The Walkmen, Erykah Badu, R. Kelly, Nelly Furtado, Franz Ferdinand, Swans, Sigur Ros

Crafting a Modern Indie Record

Producer/engineer/musician John Congleton is a highly versatile sonic artist, having worked with a diverse profile of musicians including St. Vincent, David Byrne, Angel Olsen, Swans, Franz Ferdinand, The Mountain Goats, Modest Mouse, Sigur Ros, The New Pornographers, and Explosions in the Sky. His recent work with St. Vincent, on her critically acclaimed self-titled album, garnered his first Grammy for Best Alternative Music Album. In this master class, Mr. Congleton will discuss his career, methods, and experiences in producing some of the most iconic indie rock albums of the day. He will use a selection of musical examples to demonstrate what it means to be on the cutting edge of the indie music scene.

Tutorial 5 **Thursday, October 29**
11:15 am – 12:45 pm **Room 1A06**

MIC IT & RECORD IT!

Presenter: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA; off-beat-open-hats recording & sound reinforcement

Too many resources emphasize “instant” miking solutions, and tell the aspiring recording engineer to simply “mic it this way.” This often results in sounds that have to be significantly electronically processed to force them into place during mixing, degrading the integrity of the sound and making the mix process longer and more difficult than it otherwise might be. Topics discussed in this presentation will include the effects microphone technologies, mic techniques, and the recording room have on the recorded sound, and how they can be explored and exploited to capture the sound you actually need for the mix, improving your mix, and making mixing an easier and quicker process.

Archiving 2 **Thursday, October 29**
11:15 am – 12:45 pm **Room 1A14**

PRESERVING HISTORIC AUDIO WAX CYLINDERS AND EXPERT TRANSFER TECHNIQUES

Presenters: **Brad McCoy**, Library of Congress, Culpeper, VA, USA
Nadja Wallaszkovits, Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria

Audio preservation experts have decades of experience with rare and problematic media formats and modes of degradation and have developed sophisticated techniques and work flows to recover the content from a wide variety of damaged, degraded, and obsolete types of storage media. The Library of Congress staff have produced video tutorials on basic playback techniques for wax cylinders conducted by expert practitioners in the field. This presentation will showcase these videos, discuss how they came into being, and plans for disseminating them to the preservation community. Different approaches will be outlined and compared, including internationally proven methods such as improved mechanical signal pickup. Recordings will be played of transfers made of historic cylinders from both the Phonogrammarchiv in Vienna, Austria, and the Library of Congress Packard Campus in Culpepper, VA.

Broadcast/Streaming Media 1 **Thursday, October 29**
11:15 am – 12:45 pm **Room 1A10**

STREAMING FACILITIES—BROADCAST SCALED TO INTERNET FEEDS*

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

Panelists: *Renato Cipriano*, Walters Storyk Design Group
David Pentecost, Lower East Side Girls Club
Nick Squire, Boston Symphony Orchestra, Brookline, MA, USA
Jonathan Talley, Director of Production / Lighting for Le Poisson Rouge

Streaming's emergence as the de facto 21st Century programming distribution medium was confirmed earlier this year when Apple Inc. announced its exclusive "HBO Now" streaming service. Wireless device-viewers may have already eclipsed the numbers of traditional cable TV audiences. As with any new media format, the need for cutting edge production/post-production skills and facilities remains a critical element. The scale of these studios may be smaller than conventional facilities, but their technology and acoustics must be up to professional broadcast standards. This panel will explore four highly diverse and rapidly expanding streaming content producers. It will survey the similarities and disparities between broadcast/cable and streaming facility designs, acoustic requirements, and issues. And, it will provide unparalleled insights into artist (and audience) preferences and technical requirements for high quality live streaming performances.

Spatial Audio Demo 3 **Thursday, October 29**
11:45 am – 12:45 pm **Room 1A18**

RULES TO GET GREAT MULTICHANNEL 3D SOUND FOR SYNTHESIZER MUSIC

Presenter: **Lasse Nipkow**, Silent Work LLC, Zurich, Switzerland

The 3D 9.1 format and other multichannel 3D formats allow creation of incredibly immersive soundscapes in comparison to standard stereo and 5.1 surround. The listener has a spatial impression of being in the recording room. Room sound is used to realize concert-hall-like envelopment of the listener. Synthesizer music is a different case without a natural reference, consisting to a large extent of sounds, which were not recorded in recording rooms. To render such music immersively, pad sounds, low frequency bass, and other sound elements are used instead of room sound to reach a similar spatial impression. In contrast to classical music there is no need to reproduce the instruments as a body of sound on a stage in the front. Creative composition of the 3D mix enables a multifaceted result. During the presentation, relationships between mixing 3D recordings of classical music and synthesizer music will be described and demonstrated with example recordings and videos.

Session EB1 **Thursday, October 29**
12:00 noon – 12:45 pm **Room 1A07**

TRANSDUCERS—PART 1

Chair: **Michael Smithers**, Dolby australia, McMahons Point, NSW, Australia

12:00 noon

EB1-1 Wireless Speaker Synchronization: Solved—*Simon Forrest*, Imagination Technologies, Hertfordshire, UK

Many high-end stereo systems offer the opportunity to connect several speakers together wirelessly to create a multi-room audio experience. However, linking speakers wirelessly to create stereo pairs or surround sound systems is technically challenging, due to the extremely tight synchronization necessary to accurately reproduce a faithful sound stage and maintain channel separation. Imagination measures several competing technologies on the market today and illustrates how innovative application of Wi-Fi networking protocols in audio chips can deliver several orders of magnitude improvement, creating opportunity for high quality wireless audio and producing results that are indistinguishable from wired speaker systems.
Engineering Brief 202

12:15 pm

EB1-2 Multiphysical Simulation Methods for Loudspeakers—Advanced CAE-Based Simulations of Vibration Systems—
Alfred Svobodnik,[1] *Roger Shively*,[2] *Marc-Olivier Chauveau*,[3] *Tommaso Nizzoli*,[1] *Dieter Thöres*[1]
[1]Konzept-X GmbH, Karlsruhe, Germany
[2]JJR Acoustics, LLC, Seattle, WA, USA
[3]Moca Audio, Tours, France

This is the second 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 structural dynamics of loudspeaker's vibration systems will be presented. Primarily, the calculation of stiffness, or its inverse the compliance in the virtual world, will be demonstrated. Furthermore, the predictive simulation of complex vibration patterns, e.g., rocking or break-up, will be shown. Finally the simulation of coupling effects to the motor system will be discussed. Results will be presented, correlating the simulated model results to the measured physical parameters. From that, the important aspects of the modeling that determine its accuracy will be discussed.
Engineering Brief 203

12:30 pm

EB1-3 New Design Methodologies in Mark Levinson Amps—*Todd Eichenbaum*, Harman Luxury Audio, Shelton, CT, USA

The HARMAN Luxury Audio electronics engineering team has designed a completely new generation of Mark Levinson amplifiers. Combining tried-and-true technologies with innovative implementations and unique improvements has yielded products with exemplary measured and subjective performance. In this article are circuit design highlights of the No536, a fully balanced mono power amplifier rated for 400W/8 ohms and 800W/4 ohms.
Engineering Brief 204

Thursday, October 29 **12:00 noon** **Room 1A19**

Technical Committee Meeting on Recording Technology and Practices

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Thursday, October 29, 1:00 pm – 2:00 pm
Room 1A23/24

Opening Remarks: • Executive Director **Bob Moses**
• President **Andres Mayo**

Convention Chairs **Jim Anderson** and **Paul Gallo**
Program: • AES Awards Presentation by **Frank Wells,** Awards Chair
• Introduction of Keynote Speaker
• Keynote Address by **Michael Abrash**

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:

CITATION
• Linda Gedemer
• Cesar Lamschtein
• Howard Sherman

FELLOWSHIP AWARD
• William (Bill) F. Hanley, Jr.
• David Moulton
• Agnieszka Roginski
• Ulrike K. Schwarz

BOARD OF GOVERNORS AWARD
• Michael Kelly
• Bozena Kostek
• Peter Mapp
• Valeria Palomino
• Jeff M. Smith
• Wieslaw Woszczyk
• Jorge Urbano
• Umberto Zanghieri

GOLD MEDAL AWARD
• Bob Ludwig

KEYNOTE SPEAKER

This year's Keynote Speaker is **Michael Abrash**. Abrash is Chief Scientist of Oculus VR. He was the GDI development lead for the first two versions of Windows NT, joined John Carmack to write Quake at Id Software, worked on the first two versions of Xbox, co-authored the Pixomatic software renderer at Rad Game Tools, worked on Intel's Larrabee project, worked on both augmented and virtual reality at Valve, and currently leads the Oculus Research team. He is also the author of several books, including *Michael Abrash's Graphics Programming Black Book*, and has written and spoken frequently about graphics, performance programming, and virtual reality.

The title of his talk is "Virtual Reality, Audio, and the Future." Michael Abrash will talk about why virtual reality is unique and potentially world-changing, and about how and why audio research and development will play a key part in the future of VR.

Thursday, October 29 **2:00 pm** **Room 1A19**
Technical Committee Meeting on Coding of Audio Signals

Thursday, October 29 **2:00 pm** **Room 1A20**
Standards Committee Meeting SC-02-02 Digital Inout/Output Interfacing

Workshop 3 **Thursday, October 29**
2:15 pm – 4:15 pm **Room 1A22**

LOW FREQUENCY BEHAVIOR IN SMALL HIGH ACCURACY LISTENING ENVIRONMENTS*

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

Panelists: *Renato Cipriano*, WSDG SA, Belo Horizonte, Brazil
Eddie Kramer, Audio and Technical Consultant, Los Angeles, CA, USA
Richard Lenz, Principal, Director/Chief Designer Real Acoustix, Detroit, MI
Art Noxon, Physicist and Principal AcousticSciences Corp. / Tube Trap, Eugene, OR
Dirk Noy, WSDG, Basel, Switzerland
Roger Roschnik, PSI Audio

Low frequency prediction in large and medium-size venues has become a standard in the audio industry. However, acoustic modeling of small rooms has not yet evolved into a widely accepted concept mainly because of the unavailability of one accurate too set. The workshop will explore currently available software-based approaches and real world applications to low frequency prediction.

Specific studio examples will illustrate comparisons of these approaches and their success in the field. The workshop will also explore the limitations of current LF design modeling, and in specific the underlying mathematical and numerical algorithms, such as ray tracing, which are only valid in frequency ranges where lengths are small compared to the characteristic dimensions of the room. The common dividing line is often identified with the so-called Schroeder frequency

The workshop will review these theoretical prediction limits as well as hope to create a dialogue concerning future prediction and design techniques.

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

Broadcast/Streaming Media 2 **Thursday, October 29**
2:15 pm – 3:45 pm **Room 1A10**

STREAMING AUDIO FROM THE CLOUD

Moderator: **John Kean**, Consultant, Washington DC, USA

Panelists: *Michael Dube*, NPR
Dan Jesselsohn, New York Public Radio
Kyle Wesloh, American Public Media
Adrian Wisbey, BBC

Delivery of audio content via the online "cloud" has become a significant consumer media enjoyed by millions of listeners each day. This panel will discuss their own form of delivery, such as streaming, podcasts, or progressive file transfer, and their system architecture. The panelists are encouraged to talk about the audio codec(s) and bit rates they use, and why. All are invited to address the forthcoming guidelines for audio loudness being developed by the AES: their likes, concerns, and suggestions for implementing the guidelines in their own system.

Product Development 3 **Thursday, October 29**
2:15 pm – 3:45 pm **Room 1A13**

DESIGNING FOR ULTRA-LOW THD+N IN ANALOG CIRCUITS, CIRCA 2015

Presenter: **Bruce E. Hofer**, Audio Precision, Inc., Beaverton, OR, USA

The performance of ultra-low THD+N in analog circuits is more often limited by component quality and component interaction, rather than the circuit design itself. Factors such as thermal modulation and the effects of component voltage coefficient are commonly over-looked, yet they may be the dominant source of non-linearity in some circuits. This presentation will focus on modeling and quantifying these non-linear effects that will hopefully enable the analog design engineer to achieve higher levels of performance. Among the many topics covered, the various types of resistors and capacitors will be compared with a particular emphasis regarding their impact on THD+N. There will also be a discussion of op-amps along with some suggested circuit "tricks" to minimize distortion contribution. Noise and its estimation will also be covered along with some examples of how to keep digital noise sources from finding their way into critical analog circuits. The sometimes surprising effects of mutual inductance between circuit components and power supply rails will also be discussed along with how these can be minimized by good circuit layout practices. This tutorial is highly recommended for the analog design engineer interested in taking their designs to a significantly higher level of performance.

Spatial Audio Demos 4 — **Thursday, October 29**
2:15 pm – 3:45 pm — **Room 1A18**

SOUNDS ACROSS THE SEA—
A JOURNEY IN 9.1 IMMERSIVE AUDIO

Presenters: **Morten Lindberg**, 2L (Lindberg Lyd AS), Oslo, Norway; Westerdals, Oslo School of Arts, Communication and Technology
Daniel Shores, Sono Luminus, Boyce, VA, USA; Shenandoah Conservatory Music Production and Recording Technology, Winchester, VA, USA

For years Morten Lindberg and Daniel Shores have inspired each other across the Atlantic to develop new techniques and implementing new recording formats; initially in surround sound and lately moving forward in immersive audio. In this workshop we'll meet together playing a wide range of 9.1 examples, discussing microphone techniques, production workflow and musical sound design.

Special Event
PLATINUM LATIN PRODUCERS & ENGINEERS
Thursday, October 29, 2:15 pm – 4:15 pm
Room 1A23/24

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

Panelists: *Carli Beguerie*, Blue Note Entertainment Group, New York, NY, USA; S.I.R, New York, NY, USA
Andres Landinez
Ariel Lavigna, Capital Federal, Buenos Aires, Argentina
Valeria Palomino
Salvador Tercero, Sala de Audio, México City, Mexico; Centro de Educación Tecnológica y Arte
Stefano Vieni, KIVA Music Inc., Woodland Hills, CA, USA

The Audio Engineering Society brings together the top-notch music producers and engineers from the Latin scene. Multiple Grammy-winning pros will discuss the current status of the industry and will open the debate for Q&A´s from the audience.

Student Event & Career Development
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Thursday, October 29, 2:15 pm – 3:30 pm
Room 1A06

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, November 1.

Session P4 — **Thursday, October 29**
2:30 pm – 5:30 pm — **Room 1A08**

TRANSDUCERS—PART 1: HEADPHONES, AMPLIFIERS, AND MICROPHONES

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

2:30 pm

P4-1 Headphone Response: Target Equalization Trade-offs and Limitations—*Christopher Struck*,[1] *Steve Temme*[2]
[1]CJS Labs, San Francisco, CA, USA
[2]Listen, Inc., Boston, MA, USA

The effects of headphone response and equalization are examined with respect to the influence on perceived sound quality. Free field, diffuse field, and hybrid real sound field targets are shown and objective response data for a number of commercially available headphones are studied and compared. Irregular responses are examined to determine the source of response anomalies, whether these can successfully be equalized and what the limitations are. The goal is to develop a robust process for evaluating and appropriately equalizing headphone responses to a psychoacoustically valid target and to understand the constraints.
Convention Paper 9374

3:00 pm

P4-2 A Headphone Measurement System Covers both Audible Frequency and beyond 20 kHz—*Naotaka Tsunoda, Takeshi Hara, Koji Nageno*, Sony Corporation, Tokyo, Japan

New headphone measurement system consisting of a 1/8" microphone and newly developed HATS (Head And Torso Simulator) with a coupler that have realistic ear canal shape is proposed to enable entire frequency response measurement from audible frequency and higher frequency area up to 140 kHz. At the same time a new frequency response evaluation scheme based on HRTF correction is proposed. Measurement results obtained by this scheme enables much better understanding by enabling direct comparison with free field loudspeaker frequency response.
Convention Paper 9375

3:30 pm

P4-3 Measurements of Acoustical Speaker Loading Impedance in Headphones and Loudspeakers—*Jason McIntosh*, McIntosh Applied Engineering, Eden Prairie, MN, USA

The acoustical design of two circumaural headphones and a desktop computer speaker have been studied by measuring the acoustical impedance of the various components in their design. The impedances were then used to build an equivalent circuit model for the devices that then predicted their pressure response. There was seen to be good correlation between the model and measurements. The impedance provides unique insight into the acoustic design that is not observed though electrical impedance or pressure response measurements that are commonly relied upon when designing such devices. By building models for each impedance structure, it is possible to obtain an accurate model of the whole system where the effects of each component upon the device's overall performance can be seen.
Convention Paper 9376

4:00 pm

P4-4 Efficiency Investigation of Switch-Mode Power Audio Amplifiers Driving Low Impedance Transducers—*Niels Elkjær Iversen, Henrik Schneider, Arnold Knott, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

The typical nominal resistance span of an electro dynamic transducer is 4 Ohms to 8 Ohms. This work examines the possibility of driving a transducer with a much lower impedance to enable the amplifier and loudspeaker to be directly driven by a low voltage source such as a battery. A method for estimating the amplifier rail voltage requirement as a function of the voice coil nominal resistance is presented. The method is based on a crest factor analysis of music signals and estimation of the electrical power requirement from a specific target of the sound pressure level. Experimental measurements confirm a huge performance leap in terms of efficiency compared to a conventional battery-driven sound system. Future optimization of low voltage, high current amplifiers for low impedance loudspeaker drivers are discussed.
Convention Paper 9377

4:30 pm

P4-5 Self-Oscillating 150 W Switch-Mode Amplifier Equipped with eGaN-FETs—*Martijn Duraij*,[1] *Niels Elkjær Iversen*,[1] *Lars Press Petersen*,[1] *Patrik Boström*[2]
[1]Technical University of Denmark, Lyngby, Denmark
[2]Bolecano Holding AB, Helsingborg, Sweden

Where high-frequency clocked system switch-mode audio power amplifiers equipped with eGaN-FETs have been introduced in the past years, a novel self-oscillating eGaN-FET equipped amplifier is presented. A 150 Wrms amplifier has been built and tested with regard to performance and efficiency with an idle switching frequency of 2 MHz. The amplifier consists of a power-stage module with a self-oscillating loop and an error-reducing global loop. It was found that an eGaN-FET based amplifier shows promising potential for building high power density audio amplifiers with excellent audio performance. However care must be taken of the effects caused by a higher switching frequency.
Convention Paper 9378

5:00 pm

P4-6 Wind Noise Measurements and Characterization Around Small Microphone Ports—*Jason McIntosh, Souray Bhunia*, Starkey Hearing Technologies, Eden Prairie, MN, USA

The physical origins of microphone wind noise is discussed and measured. The measured noise levels are shown to correlate well to theoretical estimates of non-propagating local fluid dynamic turbulence pressure variations called "convective pressure." The free stream convective pressure fluctuations may already be present in a flow independent of its interactions with a device housing a microphone. Consequently, wind noise testing should be made in turbulent air flows rather than laminar. A metric based on the Speech Intelligibility Index (SII) is proposed for characterizing wind noise effects for devices primarily designed to work with speech signals, making it possible to evaluate nonlinear processing effects on reducing wind noise on microphones.
Convention Paper 9379

Game Audio 2 — **Thursday, October 29**
2:15 pm – 3:15 pm — **Room 1A21**

VR GAME AUDIO: THE IMPORTANCE OF SOUND PROPAGATION

Presenter: **Ravish Mehra**, Redmond, WA, USA

Realistic sound propagation is extremely important for VR game audio for improving the sense of presence and immersion of the player in the virtual environment. Sound propagation cues can provide additional information about the game environment (small vs large, inside vs outside) and about events happening outside the field-of-view (such as enemy sneaking from behind). Most current games use simple techniques, such as pre-baked reverb filters, to generate the game sound. These techniques do not respond to the dynamic aspects of the game (e.g., moving sources, listeners, objects) and also do not model the various acoustic effects produced by propagation of sound (e.g., echoes, low-passing, scattering, focusing). In many cases, the sound engine is not given access to the underlying scene geometry to compute these propagation effects. In this presentation, I talk about the importance of accurate sound propagation for VR applications and how it can improve the overall experience of the player in VR games.

Networked Audio 1 — **Thursday, October 29**
2:15 pm – 3:15 pm — **Room 1A14**

BASIC NETWORKING AND LAYER 3 - PROTOCOLS: LAYERS, MODELS? A DISAMBIGUATION IN THE CONTEXT OF AUDIO OVER IP

Presenter: **Kieran Walsh**, Audinate Pty. Ltd., Ultimo, NSW, Australia

The OSI model is a great starting point to understand a structure for integrating network protocols and creating software. Topics for discussion include: • Examining the positives of a layered approach and fill in the "missing gaps" that are required to create a real implementation. • An implementation from a "solution provider" (manufacturer) is different to creating a real "on the ground" working full system – the Model and layered approach however can be valuable in converging these two challenges. • "Protocols, Standards, implementations" These terms are used interchangeably—they, however, have distinct meanings; we will examine the differences and distinctions of these terms. • Deploying core AoIP services in the context of other technologies that can be leveraged to make a fully working system function in an effective production environment. • Distinguishing between standards, implementations, transports, protocols, layers and have a better insight into what each means and how to define requirements for systems. • Understanding the IT centric approach to a network, and identify challenges and workarounds when deploying an AoIP system. • Understanding some techniques that "come for free" in an enterprise IT network environment..

Session P5 — **Thursday, October 29**
2:30 pm – 5:00 pm — **Room 1A07**

PERCEPTION—PART 1

Chair: **Jon Boley**, GN ReSound, Mundelein, IL, USA

2:30 pm

P5-1 Detection of High-Frequency Harmonics in a Complex Tone—*Wesley Bulla*, Belmont University, Nashville, TN, USA

Prior investigations have generally failed to confirm or deny the influence of high-frequency harmonics contained in musical sounds. Embedded within this experiment were two listening tests: one investigating threshold for differences in timbre, and thus, participant ability, and another seeking to find an influence of high-frequency harmonic content on timbre perception. Based on the premise that harmonics out of the range of auditory detection influence the resultant waveform and therefore may alter the percept of a sound's tonal character, this study found no evidence that capable listeners noticed an effect of high frequency harmonics.
Convention Paper 9380

3:00 pm

P5-2 Towards a Perceptual Model of "Punch" in Musical Signals—*Steven Fenton, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

This paper proposes a perceptual model for the measurement of "punch" in musical signals. Punch is an attribute that is often used to characterize music or sound sources that convey a sense of dynamic power or weight to the listener. A methodology is

explored that combines signal separation and low level parameter measurement to produce a perceptually weighted "punch" score. The parameters explored are the onset time and frequency components of the signal across octave bands. The "punch" score is determined by a weighted sum of these parameters using coefficients derived through a large scale listening test. The model may have application in music information retrieval (MIR) and music production tools. The paper concludes by evaluating the perceptual model using commercially released music.
Convention Paper 9381

3:30 pm

P5-3 Factors That Influence Listeners' Preferred Bass and Treble Levels in Headphones—*Sean Olive, Todd Welti*, Harman International Inc., Northridge, CA, USA

A listening experiment was conducted to study factors that influence listeners' preferred bass and treble balance in headphone sound reproduction. Using a method of adjustment a total of 249 listeners adjusted the relative treble and bass levels of a headphone that was first equalized at the eardrum reference point (DRP) to match the in-room steady-state response of a reference loudspeaker in a reference listening room. Listeners repeated the adjustment five times using three stereo music programs. The listeners included males and females from different age groups, listening experiences, and nationalities. The results provide evidence that the preferred bass and treble balances in headphones was influenced by several factors including program, and the listeners' age, gender, and prior listening experience. The younger and less experienced listeners on average preferred more bass and treble in their headphones compared to the older, more experienced listeners. Female listeners on average preferred less bass and treble than their male counterparts.
Convention Paper 9382

4:00 pm

P5-4 Identifying and Validating Program Material: A Hyper-Compression Perspective—*Malachy Ronan,*[1] *Nicholas Ward,*[1] *Robert Sazdov*[1,2]
[1]University of Limerick, Limerick, Ireland
[2]Currently with the Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Two listening experiments were conducted to assess: (i) the effect of program material on six sound quality dimensions and (ii) the effect of 20 dB of compression limiting on distraction. Thirty-five participants completed two experiments using a MuSHRA style interface. The experimental results demonstrate that program material significantly affected dimension and distraction ratings. Dimension ratings were influenced by prior listening experience while distraction ratings related to audible artifacts in different program material. Program material from the same artist was rated similarly for distraction in two-thirds of the dimensions suggesting a possible correlation between production aesthetics and audible artifacts. It is concluded that validating program material is a necessary precaution to avoid distracting perceptual cues generated by the process of dynamic range compression.
Convention Paper 9383
Convention Paper 9384 was withdrawn

4:30 pm

P5-5 Validation of Experimental Methods to Record Stimuli for Microphone Comparisons—*Andy Pearce, Tim Brookes, Martin Dewhirst*, University of Surrey, Guildford, Surrey, UK

Test recordings can facilitate evaluation of a microphone's characteristics but there is currently no standard or experimentally validated method for making recordings to compare the perceptual characteristics of microphones. This paper evaluates previously used recording methods, concluding that, of these, the most appropriate approach is to record multiple microphones simultaneously. However, perceived differences between recordings made with microphones in a multi-microphone array might be due to (i) the characteristics of the microphones and/or (ii) the different locations of the microphones. Listening tests determined the maximum acceptable size of a multi-microphone array to be 150 mm in diameter, but the diameter must be reduced to no more than 100 mm if the microphones to be compared are perceptually very similar.
Convention Paper 9385

Thursday, October 29 3:00 pm Room 1A19

Technical Committee Meeting on Automotive Audio

Tutorial 6 Thursday, October 29
3:30 pm – 4:45 pm Room 1A06

FROM STUDENT TO PROFESSIONAL: STRATEGIES AND BEST PRACTICES FOR MATRICULATING INTO YOUR AUDIO CAREER

Presenter: **John Krivit**, New England Institute of Art, Brookline, MA, USA; Emerson College, Boston, MA, USA

AES Education Chair John Krivit shares tried and true methods for gaining your foothold in the audio industry. From industry trends that may alter your career trajectory to small gestures that can help shape how you are perceived, come find out how his students find work and launch successful careers.

Game Audio 3 Thursday, October 29
3:30 pm – 5:00 pm Room 1A12

AUDIO SHORTS: INDIE EDITION

Presenters: **Alexis Brandow**
Damien DiFede
Garrett Nantz, Luxurious Animals, New York, NY, USA
Matt Piersall, GL33k, Austin, TX, USA

Big games from little studios. Three presenters get only 30 minutes each to serve up an in-depth look at topics in game audio tech that matter most to them. Q&A to follow.

Shorty #1: The All Music Game: Inside the Tech and Design of Cosmic DJ This talk delves in to the tech and design challenges of developing a music creation game. It's a peek behind the curtain of the custom code and tools written that allowed players the ability to create their own music within our rule-set. The talk delves into the audio side of creating the content as well as the "jam-responses" and metronome system used to drive the entire game experience from the players musical input. In addition the talk will take a look inside the generative final music creator that allowed players to export their songs and share them. Matt Piersall, Damien DiFede

Shorty #2: Adventures with Midi in Audio Game Development: When making audio games the big challenge is crunching down large audio sizes to keep file sizes manageable. The Notespace team used midi solutions on various games within our musical activity book, Notespace Beat, to address this problem. In this conversation we would like to share with you the challenges and advantages we found in implementing Midi technology. We also would like to address briefly some recommendations based on our experience for recording and sharing audio across PC and iOS mobile platforms. Alexis Brandow

Shorty #3: Breaking the 3D Sound Barrier: Until recently, recreating three dimensional audio on the web has been a difficult and limiting task. With modern browsers finally supporting more robust audio systems, web audio can finally rival sound experiences traditionally found only on desktop and mobile apps. Using the award-winning Lux Ahoy www.luxahoy.com and Feisty Galaxies www.feistygalaxies.com games, we will take a behind the scenes look at the process for creating audio for browser-based games using the Dolby Digital Plus E-AC-3 codec and the Web Audio API. Topics covered will include how to create and encode surround sound and video for the web, methods for your audience to hear 3D sound, stereo audio fallback support, tricks and techniques, creating memorable sound effects, audio loop creation, and music sourcing.

Networked Audio 2 Thursday, October 29
3:30 pm – 4:30 pm Room 1A14

AVB/TSN ETHERNET IS BUILT-IN EVERYWHERE NOW; HOW DO YOU MAKE THE MOST OF IT? A SYSTEM IMPLEMENTATION PRIMER FOR CONSULTANTS AND TECH MANAGERS

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

Panelists *Richard Bugg*
Jim Cooper
Tom Knesel
Nathan Philips
Curtis Rex Reed

This presentation will introduce technology managers, integrators, and specifiers to the basics of distributing audio, video, and control signals over an Ethernet network in ready-to-play fashion. The presentation will also focus on system implementation with Time Sensitive Networking (TSN) standards—the evolution of Audio Video Bridging (AVB).

Attendees will be provided with a system-level understanding on how to achieve networked AV success; discuss the advantages of using a network; and overview challenges and approaches and provide tips and troubleshooting for networking with AVB/TSN. Discover how easy it is to scale and upgrade TSN systems.

An overview of the methods of time synchronization will also be outlined. AVnu Alliance will start by reviewing system requirements for demanding applications such as performance venue installs, house of worship, large convention systems, conference rooms and broadcast and discuss the Ethernet capabilities needed for the network including characteristics and definitions of TSN for these applications.

The presentation will highlight the importance of certification for interoperability. Finally, AVnu Alliance will present the existing tools and resources that designers need for successful TSN system operation.

Learning Objectives: • Gain a basic understanding of distributing audio, video and control systems over an Ethernet network and the advantages to doing so. • Understand the existing tools and resources that designers need to successfully operate TSN systems. • Understand what is required from a network for applications such as performance venue installs, houses of worship, conference rooms etc.

Workshop 4 Thursday, October 29
4:00 pm – 5:30 pm Room 1A21

INTERACTION AND EVOLUTION OF RECORDING CLASSICAL MUSIC THRU THE EYES OF GRAMMY AWARD-WINNING PRODUCERS*

Chair: **David Merrill**, DMMUSIUC, Inc., Brooklyn, NY, USA

Panelists: *Steve Epstein*
David Frost
Tom Shepard
Judy Sherman

Grammy Award winning classical music producers discussing the interaction and observations they have had with technology through the decades.

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

Broadcast/Streaming Media 3 Thursday, October 29
4:00 pm – 5:30 pm Room 1A10

LOUDNESS FOR STREAMING

Moderator: **Bob Katz**, Digital Domain Mastering, Orlando, FL, USA

Panelists: *Rob Byers*, American Public Media, St. Paul, MN, USA
John Kean, Consultant, Washington DC, USA
Thomas Lund, Genelec, Risskov, Denmark
Scott Norcross, Dolby Labs, San Francisco, CA, USA
Adrian Wisbey, BBC FM Media Services, London, UK

The advent of Internet streaming services has shaken the entire audio industry. Every sector has been quickly affected: Broadcast, radio, TV, music production. One of the prime problems is that of regulating audio levels. There is already a de facto loudness war among streamers, with some using a high target level that requires them to use extreme amounts of compression and limiting that alter producer's intent and/or cause distortion, dilute the impact, etc. As a result, producers may fall into the trap of loudness envy and create their recordings at the lowest common denominator of sound quality by trying to match the level of the loudest streaming service. Enter the AES Subcommittee on Loudness in Streaming and Network Playback, which has produced a new set of Recommendations for streaming entities. Let us hope that these recommendations help civilize the wild west of streaming. Learn about the issues and the new recommendations.

Product Development 4 Thursday, October 29
4:00 pm – 5:30 pm Room 1A13

ELECTRICAL AND MECHANICAL MEASUREMENT OF SOUND SYSTEM EQUIPMENT

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

This tutorial explains the physical background and practical motivation for a new measurement standard replacing the IEC 60268-5 applicable to all kinds of transducers, loudspeakers, and other sound reproduction systems. The focus are electrical and mechanical measurements (part B) complementing the acoustical measurements (part A) presented at the 137th AES convention in LA last year. Voltage and current measured at the electrical terminals provide not only the electrical input impedance but also meaningful parameters of linear, nonlinear, and thermal models describing the behavior of the transducer in the small and large signal domain. This standard addresses long-term testing to assess power handling, heating process, product reliability, and climate impact. New mechanical characteristics are derived from laser scanning techniques that are the basis for modal analysis of cone vibration and predicting the acoustical output.

Spatial Audio Demos 5 Thursday, October 29
4:00 pm – 5:30 pm Room 1A18

PSYCHOACOUSTICS OF 3D SOUND RECORDING

Presenter: **Hyunkook Lee**, University of Huddersfield, Huddersfield, UK

This tutorial/demo session will explain psychoacoustic principles that need to be considered when making 3D sound recordings, with demonstrating a number of practical 3D recordings made using various microphone configurations. The topics to be addressed include: vertical interchannel time and level relationship, the effect of decorrelation/mic spacing in vertical stereo perception, the role of spectral cue for vertical localization and image spread, and effective microphone configurations for the capturing of height channel ambience. 3D recordings to be demonstrated will be of various types of music: orchestra, chamber choir, string ensemble, organ, piano, jazz funk, etc. This session will be useful for those who would like to learn how perceptual principles can be applied in practical 3D recording.

Thursday, October 29 4:00 pm Room 1A19

Technical Committee Meeting on Signal Processing

Archiving 3 — Thursday, October 29
4:45 pm – 5:45 pm — Room 1A14

HARD TO HANDLE: UNSTABLE FORMATS

Presenters: **Matt Barton**, Library of Congress
James Sam, Hoover Institution Archives, Stanford University, Stanford, CA, USA
Seth Winner, Seth B. Winner Sound Studios, Inc., Merrick, NY, USA

One of the most unstable formats found in audio archives, the lacquer disc requires great expertise to properly handle. Often called an "acetate" in the vernacular, its having multiple parts and potentially a glass base leaves it ripe for various breakages. James Sam will begin will cover the discs' composition, handling, cleaning, playback, and storage, including the presentation of a housing that was custom-designed by conservators and archivists at Stanford University's Hoover Institution Library and Archives for the horizontal storage of cracked, broken, and/or delaminating discs. Matthew Barton will be presenting the troubles with playing back 78s. Seth B. Winner will also present the myriad unstable formats he has encountered.

Tutorial 7 — Thursday, October 29
5:00 pm – 6:30 pm — Room 1A06

KEEP IT "REEL": THE ULTIMATE ULTRA-PORTABLE PRODUCTION /RECORDING STUDIO—FROM IDEA TO FINAL MASTER: HOW TO WRITE, SEQUENCE, RECORD, AND PRODUCE YOUR MUSIC USING ONLY YOUR iPAD

Presenter: **Andrea Pejrolo**, Berklee College of Music, Boston, MA, USA

In this highly interactive and hands on presentation you will learn the tools, techniques, tips, and tricks required to write, produce, and mix a song using only your iPad.

Through practical examples and scenarios you will learn how to: • Pick the best software for sequencing, producing and mixing your music • Pick the best iPad-compatible hardware tools (microphones, audio interface, MIDI interfaces, controller etc.) • Setup your mobile production/recording studio • Sketch your musical ideas • Use your iPad as a creative inspirational tool for music composition and sound design • Sequence and arrange your music ideas on your iPad • Use your iPad as a powerful virtual mixing console • Add audio plug-ins • Master your final mix.

Who should attend? Anyone who wants to create some great music with their iPads, from beginners to advanced. Attendees will learn: • How to assemble the ultimate ultra-portable multi-track recording rig • How to create a complete final production of a song using only the iPad • How to pick the right hardware and software available for mobile music production on the iPad • How to take advantage of the highly interactive interface of the iPad to streamline and enhance your creative process • How to share and interact with other music creators using solely the iPad and the cloud.

Special Event
PRODUCING ACROSS GENERATIONS: NEW CHALLENGES, NEW SOLUTIONS—MAKING RECORDS FOR NEXT TO NOTHING IN THE 21ST CENTURY
Thursday, October 29, 5:00 pm – 7:00 pm
Room 1A23/24

Moderator: **Jesse Lauter**, New York, NY, USA

Panelists: *Jon Altschiller*
Bryce Goggin
Hank Shocklee
Erin Tonkon, Tony Visconti Productions, Brooklyn, NY, USA

Budgets are small, retail is dying, streaming is king and studios are closing… yet devoted music professionals continue to make records for a living. How are they doing it? How are they getting paid? What type of contracts are they commanding? In a world where the "record" has become an artists' business card, how will the producer and mixer derive participatory income? Are studio professionals being left out of the so-called 360 deals? How can we expect to see any income from streaming royalties when artists aren't even seeing any? Let's get a quality bunch of young rising producers and a handful of seasoned vets in a room and finally open the discussion about empowerment and controlling our own destiny.

Thursday, October 29 — 5:00 pm — Room 1A20

Standards Committee Meeting SC-02-08 Audio file Transfer and Exchange

Live Sound Seminar 2 — Thursday, October 29
5:15 pm – 7:00 pm — Room 1A12

CANCELED

Session EB2 — Thursday, October 29
5:30 pm – 6:30 pm — Room 1A08

SPATIAL AUDIO

Chair: **Bryan Martin**, McGill University, Montreal, Quebec, Canada

5:30 pm

EB2-1 Array-Based HRTF Pattern Emulation for Auralization of 3D Outdoor Sound Environments with Direction-Based Muffling of Sources—*Pieter Thomas, Timothy Van Renterghem, Dick Botteldooren*, Ghent University, Ghent, Belgium

The use of spatial audio reproduction techniques is widely employed for the subjective analysis of concert halls and, more recently, complex outdoor sound environments. In this work a binaural reproduction technique is developed based on a 32-channel spherical microphone array, optimized for the simulation of a virtual microphone with directional characteristics that approximate the directivity of the human head. A set of weights is calculated for each microphone of the constituting array based on a regularized least-square solution. This technique allows for adaptation of the auditory scene based on source direction. The performance of variants of the technique has been evaluated by means of listening tests. Furthermore, its use for the auralization of outdoor soundscapes has been illustrated.
Engineering Brief 205

5:45 pm

EB2-2 Polar Pattern Comparisons for the Left, Center, and Right Channels in a 3-D Microphone Array—*Margaret Luthar,*[1] *Elaine Maltezos*[2]
[1]Sonovo Mastering, Stavanger, Norway
[2.]University of Stavanger, Stavanger, Norway

Standard 5.1 microphone arrays are long established and have been applied to psychoacoustic research, as well as for commercial purposes in film and music. Recent interest in the creative possibilities of "3-D audio" (a lateral layer of microphones, as well as an additional height layer) has led to research in both adapting 5.1 arrays for 3-D recordings as well as creating new methods to better capture the listener's experience. The LCR configuration in a 5.1 array is a factor that contributes to the stability and localization of the auditory image in the horizontal plane. In this experiment, two different LCR configurations have been adapted for 9.1 in a traditional concert-recording environment. They are then compared in various combinations for their ability to produce a stable, natural, and effective frontal image in a 9.1 reproduction method. Preliminary listening suggests that the polar characteristics of the L,C, and R microphones do affect the sense of envelopment, spaciousness, and localization of the frontal image, as well as cohesiveness within the entire 9.1 image. These results have led to options for further study, as suggested by the researchers.
Engineering Brief 206

6:00 pm

EB2-3 Coding Backward Compatible Audio Objects with Predictable Quality in a Very Spatial Way—*Stanislaw Gorlow*, Gorlow Brainworks, Paris, France; Laboratoire Bordelais de Recherche en Informatique, Talence, Aquitaine, France

A gradual transition from channel-based to object-based audio can currently be observed throughout the film and the broadcast industries. One paramount example of this trend is the new MPEG-H 3D Audio standard, which is under development. Other object-based standards in the market place are DTS:X and Dolby Atmos. In this engineering brief a newly developed prototype of an object-based audio coding system is introduced and discussed in terms of its technical characteristics. The codec can be of use everywhere where a given sound scene is to be rerendered according to the listener's preference or environment in a backward compatible manner. The areas of application cover not only interactive music listening or remixing, but also location-dependent, immersive, and 3D audio rendering.
Engineering Brief 207

6:15 pm

EB2-4 Decorrelated Audio Imaging in Radial Virtual Reality Environments—*Bryan Dalle Molle, James Pinkl, Mark Blewett*, University of Illinois at Chicago, Chicago, IL, USA

University of Illinois at Chicago's CAVE2 is a large-scale, 320-degree radial visualization environment with a 360-degree 20.2 channel radial speaker system. The purpose of our research is to develop solutions for spatially accurate playback of audio within a virtual reality environment, reconciling differences between the circular speaker array, the location of a user in the physical space, and the location of virtual sound objects within CAVE2's OmegaLib virtual reality software, all in real time. Previous research presented at AES 137 detailed our work on object geometry, dynamically mapping a virtual object's width and distance to the speaker array with volume and delay compensation. Our recent work improves virtual width perception using dynamic decorrelation with transient fidelity, implemented via Supercollider on the CAVE2 sound server.
Engineering Brief 208
[This eBrief was presented on Friday in session EB3]

Game Audio 4 — Thursday, October 29
5:15 pm – 6:15 pm — Foyer

AUDIO SHORTS/POSTERS

Presenters: **Alexis Brandow**
Damien Di Fede
Garrett Nantz, Luxurious Animals, New York, NY, USA
Matt Piersall, GL33k, Austin, TX, USA

Immediately following Audio Shorts: Indie Edition, walk over to the Posters area to see short demonstrations from each of the presenters. As in the Shorts format, each presenter only get 20 minutes. Think of this as a lab class where can get a hands on demo of what you just saw in class.

Shorty #1: Audio Driven Design: The Cosmic DJ Workflow: Typically audio is the last thing in the pipeline for most games. Cosmic DJ took the opposite approach. During the development of Cosmic DJ the team relied on audio to be created first to drive design. That workflow lead the team to create toolsets that were completely based on music sequencing, giving the entire experience a natural musical feel.

Shorty #2: - Notespace Beat Audio Games Demo: A Notespace team member will be demoing midi games from Notespace Beat. The Orbitron game will show how we mixed a midi track with a background soundtrack. The Synthball game will show how we determined a correct key stroke and the way we ultimately decided to time it with the background music. The Do tempo game, will further illustrate the timing solutions we used with midi.

Shorty #3: Web Sound Insights: The Audio Shorts session for Breaking the Gaming 3D Sound Barrier will give individuals an opportunity to experience the Lux Ahoy and Feisty Galaxies web games first hand while getting a chance to ask Garrett Nantz specific questions about how the 3D and Dolby audio segments were created.

Tutorial 8 — Thursday, October 29
5:45 pm – 6:45 pm — Room 1A13

POP MUSIC FROM 1000 YEARS AGO: RECORDING THROAT SINGERS AND INSTRUMENTS FROM INNER MONGOLIA

Presenter: **Richard King**, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

A case study of a modern recording session featuring centuries-old music from Inner Mongolian group Anda Union. Challenges and solutions as to how the voices were recorded in conjunction with the instruments, as each musician was simultaneously singing and playing. Horse head fiddles, calf skin drums and various strummed instruments were captured as part of the album. Throat singing, or "Hoomei," is performed in two fascinating manners—a sustained note is sung in the normal range and a second note is produced either one octave below, or in the range two octaves above where actual melodies may be produced in harmony with the held "tonic." Microphone choices, mix decisions, and production techniques will be discussed, video and audio examples will be provided.

Archiving 4 — Thursday, October 29
5:45 pm – 6:45 pm — Room 1A14

MAGNETIC TAPE AND TOMATOES: ONE DECAYS FOREVER, ONE IS FOREVER RECOVERABLE

Presenters: Marty Atias
Charles Richardson

What do tomatoes and magnetic tape have in common? They can both react with the atmosphere causing chemical changes to occur. Unlike fruit however, a very common dysfunction of magnetic tape, generally known as Sticky Shed Syndrome, is the result a natural chemical process known as hydrolysis, which interestingly, can offer the option of a naturally occurring inverse reaction known as reverse hydrolysis. Chemical science holds the answers to the complete recovery of existing sticky magnetic tapes and how to manufacture long lasting recording tape free of sticky shed type problems. An explanation by chemical science, lab work, microscopic pictures, and before and after results are provided to attain the goal of high-level restoration and preservation of magnetic tape for a very long time. High levels of mechanical and magnetic performance can only occur by first solving the chemical issues.

Special Event
50TH ANNIVERSARY OF THE MASTER ANTENNA ON THE EMPIRE STATE BUILDING
Thursday, October 29, 7:30 pm – 9:30 pm
Empire State Building

Moderators: **David Bialik**, CBS, New York, NY, USA
Scott Fybush, Northeast Radio Watch

Panelists: *Andy Lanset*, Historian, WNYC/WQXR, New York, NY, USA

Shane O'Donoghue, Director of Broadcasting, Empire State Building, New York, NY, USA
Tom Silliman, Electronics Research Inc.
Herb Squire
Robert Tarsio, Broadcast Devices Inc.

On December 9, the Alford master FM antenna that rings the 102nd floor observation gallery at the Empire State Building will mark its 50th anniversary. When the antenna went into service in 1965, it marked a revolution in FM broadcast technology: for the first time, most of a market's FM signals could share a single antenna, sharing costs and reducing the amount of space needed for FM transmission at the market's tallest broadcast site.

The Alford antenna at Empire was the model for master antenna sites in places such as Toronto, St. Louis, Houston, Minneapolis-St. Paul, and eventually back at Empire, where a new master antenna system was commissioned in the 1980s to supplant the original Alford. The original 1965 Alford antenna continues to serve as a backup at Empire, being pressed into service after 9/11 to provide emergency replacement antenna capacity for stations that were displaced from their World Trade Center sites.

This special commemorative event will be held in the 67th floor conference room at the Empire State Building.

Produced jointly with AES and SBE

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Thursday, October 29, 7:00 pm – 8:30 pm
Room 1A06

Lecturer: **Bruce Koenig**

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 Heyser Lecturer this year is former FBI agent and forensic consultant **Bruce Koenig**, who will illuminate the mysteries of acoustic gunshot analysis that first drew public attention with the examinations of a police department recording made during President John F. Kennedy's assassination in Dallas, Texas, in 1963. He'll explain the audio tests performed in those examinations and shed light on controversial cases, such as recent police shootings. Koenig will offer insights on the use of critical listening, high-resolution waveform, spectrographic, narrow-band spectrum, energy contour, and statistical analyses, as well as giving his views on the latest developments in the field. He concludes with a look at the research needed to move acoustic gunshot analysis forward in coming years. The title of his lecture is, "Acoustic Forensic Gunshot analysis—The Kennedy Assassination and Beyond."

Student Event & Career Development
AES STUDENT PARTY
Thursday, October 29, 8:00 pm – 10:00 pm
New York University

Audio Students! Join us for a fun and exciting evening at the AES Student Party to be held at NYU's James L. Dolan Music Recording Studio, a 7,500 square foot multifunctional teaching, recording, and research space. This is one of the most technologically advanced audio teaching facilities in the United States and a great place for pizza and music. The studio is located on the sixth floor of 35 West 4th Street. Space is limited so register in advance at the Student Booth (close to the registration desk).

Session P6 **Friday, October 30**
9:00 am – 12:30 pm **Room 1A08**

TRANSDUCERS—PART 2: LOUDSPEAKERS

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

9:00 am

P6-1 Wideband Compression Driver Design, Part 1: A Theoretical Approach to Designing Compression Drivers with Non-Rigid Diaphragms—*Jack Oclee-Brown*, GP Acoustics (UK) Ltd., Maidstone, UK

This paper presents a theoretical approach to designing compression drivers that have non-rigid radiating diaphragms. The presented method is a generalization of the Smith "acoustic mode balancing" approach to compression driver design that also considers the modal behavior of radiating structure. It is shown that, if the mechanical diaphragm modes and acoustical cavity modes meet a certain condition, then the diaphragm non-rigidity is not a factor that limits the linear driver response. A theoretical compression driver design approximately meeting this condition is described and it's performance evaluated, using FEM models.
Convention Paper 9386

9:30 am

P6-2 Time/Phase Behavior of Constant Beamwidth Transducer (CBT) Circular-Arc Loudspeaker Line Arrays—*D.B. (Don) Keele, Jr.*, DBK Associates and Labs, Bloomington, IN, USA

This paper explores the time and phase response of circular-arc CBT arrays through simulation and measurement. Although the impulse response of the CBT array is spread out in time, it's phase response is found to be minimum phase at all locations in front of the array: up-down, side-to-side, and near-far. When the magnitude response is equalized flat with a minimum-phase filter, the resultant phase is substantially linear phase over a broad frequency range at all these diverse locations. This means that the CBT array is essentially time aligned and linear phase and as a result will accurately reproduce square waves anywhere within its coverage. Accurate reproduction of square waves is not necessarily audible but many people believe that it is an important loudspeaker characteristic. The CBT array essentially forms a virtual point-source but with the extremely-uniform broadband directional coverage of the CBT array itself. When the CBT array is implemented with discrete sources, the impulse response mimics a FIR filter but with non-linear sample spacing and with a shape that looks like a roller coaster track viewed laterally. An analysis of the constant-phase wave fronts generated by a CBT array reveals that the sound waves essentially radiate from a point that is located at the center of curvature of the array's circular arc and are essentially circular at all distances, mimicking a point source.
Convention Paper 9387

10:00 am

P6-3 Progressive Degenerate Ellipsoidal Phase Plug—*Charles Hughes*, Excelsior Audio, Gastonia, NC, USA

This paper will detail the concepts and design of a new phase plug. This device can be utilized to transform a circular planar wave front to a rectangular planar wave front. Such functionality can be very useful for line array applications as well as for feeding the input, or throat section, of a rectangular horn from the output of conventional compression drivers. The design of the phase plug allows for the exiting wave front to have either concave or convex curvature if a planar wave front is not desired. One of the novel features of this device is that there are no discontinuities within the phase plug.
Convention Paper 9388
[Paper presented by Don Keele]

10:30 am

P6-4 Low Impedance Voice Coils for Improved Loudspeaker Efficiency—*Niels Elkjær Iversen, Arnold Knott, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

In modern audio systems utilizing switch-mode amplifiers the total efficiency is dominated by the rather poor efficiency of the loudspeaker. For decades voice coils have been designed so that nominal resistances of 4 to 8 Ohms is obtained, despite modern audio amplifiers, using switch-mode technology, can be designed to much lower loads. A thorough analysis of the loudspeaker efficiency is presented and its relation to the voice coil fill factor is described. A new parameter, the drivers mass ratio, is introduced and it indicates how much a fill factor optimization will improve a driver's efficiency. Different voice coil winding layouts are described and their fill factors analyzed. It is found that by lowering the nominal resistance of a voice coil, using rectangular wire, one can increase the fill factor. Three voice coils are designed for a standard 10" woofer and corresponding frequency responses are estimated. For this woofer it is shown that the sensitivity can be improved approximately 1 dB, corresponding to a 30% efficiency improvement, just by increasing the fill factor using a low impedance voice coil with rectangular wire.
Convention Paper 9389

11:00 am

P6-5 Effectiveness of Exotic Vapor-Deposited Coatings on Improving the Performance of Hard Dome Tweeters—*Peter John Chapman*, Bang & Olufsen Automotive, Struer, Denmark

The audio industry is constantly striving for new and different methods with which to improve the sound quality and performance of components in the signal chain. In many cases however, insufficient evidence is provided for the benefit of so-called improvements. This paper presents the results of a scientific study to analyze the effectiveness of applying vapor-deposited diamond-like-carbon, chromium, and chromium nitride coatings to aluminum and titanium hard dome tweeters. Careful attention was paid during the processing, assembly, and measurement of the tweeters to ensure a control and equal influence of other factors such that a robust analysis could be made. The objective results were supplemented with listening tests between the objectively most significant change and the control.
Convention Paper 9390

11:30 am

P6-6 Wideband Compression Driver Design. Part 2, Application to a High Power Compression Driver with a Novel Diaphragm Geometry—*Mark Dodd*, GP Acoustics, Ipswich, Suffolk, UK

Performance limitations of high-power wide-bandwidth conventional and co-entrant compression drivers are briefly reviewed. An idealized co-entrant compression driver is modeled and acoustic performance limitations discussed. The beneficial effect of axisymmetry is illustrated using results from numerical models. Vibrational behavior of spherical-cap, conical, and bi-conical diaphragms are compared. Axiperiodic membrane geometries consisting of circular arrays of features are discussed. This discussion leads to the conclusion that, for a given feature size, annular axiperiodic diaphragms have vibrational properties mostly dependent on the width of the annulus rather than it's diameter. Numerically modeled and measured acoustic performance of a high-power wide-bandwidth compression driver using an annular axiperiodic membrane, with vibrational and acoustic modes optimized, is discussed.
Convention Paper 9391

12:00 noon

P6-7 Dual Diaphragm Asymmetric Compression Drivers—*Alexander Voishvillo*, JBL/Harman Professional, Northridge, CA, USA

A theory of dual compression drivers was described earlier and the design was implemented in several JBL Professional loudspeakers. This type of driver consists of two motors and two annular diaphragms connected through similar phasing plugs to the common acoustical load. The new concept is based as well on two motors and acoustically similar phasing plugs but the diaphragms are mechanically "tuned" to different frequency ranges. Summation of acoustical signals on common acoustical load provides extended frequency range compared to the design with identical diaphragms. Theoretically maximum overall SPL sensitivity is achieved by the in-phase radiation of the diaphragms. Principles of operation of the new dual asymmetric driver are explained using a combination of matrix analysis, finite elements analysis, and data obtained from a scanning vibrometer and the electroacoustic measurements are presented. Comparison of the performance of these dual drivers and the earlier fully symmetric designs is provided.
Convention Paper 9392

Session P7 **Friday, Oct. 30**
9:00 am – 12:00 noon **Room 1A07**

PERCEPTION—PART 2

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

9:00 am

P7-1 In-Vehicle Audio System Sound Quality Preference Study—*Patrick Dennis*, Nissan North America, Farmington Hills, MI, USA

In-vehicle audio systems present a unique listening environment. Listeners were asked to adjust the relative bass and treble levels as well as fade and balance levels based on preference on three music programs reproduced through a high quality in-vehicle audio system. The audio system frequency response was initially tuned to a frequency spectrum similar to that preferred for in-room loudspeakers. The fade control was initially set to give a frontal image with some rear envelopment using two different rear speaker locations, rear deck and rear door, while the balance control was set to give a center image between the center of the steering wheel and rearview mirror. Stage height was located on top of the instrument panel (head level). Results showed that on average listeners preferred +13 dB bass and –2 dB treble compared to a flat response while fade was +3.5 dB rearward for rear deck mounted speakers,+2.6 dB rearward for rear door mounted, and balance was 0 dB. Significant variations between individual listeners were observed.
Convention Paper 9393

9:30 am

P7-2 Adapting Audio Quality Assessment Procedures for Engineering Practice—*Jan Berg*, Nyssim Lefford, Luleå University of Technology, Luleå, Sweden

Audio quality is of concern up and down the production chain from content creation to distribution. The technologies employed at each step—equipment, processors like codecs, downmix algorithms, and loudspeakers—all are scrutinized for their impact. The now well-established field of audio quality research has developed robust methods for assessments. To form a basis for this work, research has investigated how perceptual dimensions are formed and expressed. The literature includes numerous sonic attributes that may be used to evaluate audio quality. All together,

these findings have provided benchmarks and guidelines for improving audio technology, setting standards in the manufacture of sound and recording equipment and furthering the design of reproduction systems and spaces. They are, however, by comparison rarely used to inform recording and mixing practice. In this paper quality evaluation and mixing practice are compared on selected counts and observations are made on what points these fields may mutually inform one another.
Convention Paper 9394

10:00 am

P7-3 Perception and Automated Assessment of Audio Quality in User Generated Content—*Bruno Fazenda,*[1] *Paul Kendrick,*[1] *Trevor Cox,*[1] *Francis Li,*[1] *Iain Jackson*[2]
[1]University of Salford, Salford, Greater Manchester, UK
[2]University of Manchester, Manchester, UK

Many of us now carry around technologies that allow us to record sound, whether that is the sound of our child's first music concert on a digital camera or a recording of a practical joke on a mobile phone. However, the production quality of the sound on user-generated content is often very poor: distorted, noisy, with garbled speech or indistinct music. This paper reports the outcomes of a three-year research project on assessment of quality from user generated recordings. Our interest lies in the causes of the poor recording, especially what happens between the sound source and the electronic signal emerging from the microphone. We have investigated typical problems: distortion; wind noise, microphone handling noise, and frequency response. From subjective tests on the perceived quality of such errors and signal features extracted from the audio files we developed perceptual models to automatically predict the perceived quality of audio streams unknown to the model. It is shown that perceived quality is more strongly associated with distortion and frequency response, with wind and handling noise being just slightly less important. The work presented here has applications in areas such as perception and measurement of audio quality, signal processing, and feature detection and machine learning.
Convention Paper 9395

10:30 am

P7-4 Compensating for Tonal Balance Effects Due to Acoustic Cross Talk Removal while Listening with Headphones—*Bob Schulein*, RBS Consultants, Schaumburg, IL, USA

With the large number of headphones now in use, a preponderance of recorded music mixed with loudspeakers is experienced while listening with headphones. It is well known that the headphone experience creates a difference in spatial perception due to the fact that the crosstalk normally associated with loudspeaker listening is eliminated, resulting in a widening of the perceived sound stage. In addition to this difference, a question arises as to changes in the perceived tonal balance that may occur with the removal of acoustic crosstalk. This paper presents a method of measuring such differences based on a series of near field binaural mannequin recordings for which the spectral influence of crosstalk is determined. Measurement data is presented as to the findings of this investigation. Results suggest that headphones designed to sound well balanced for most popular music benefit from a low frequency boost in frequency response, whereas headphones designed primarily for classical listening require less boost.
Convention Paper 9396

11:00 am

P7-5 The Use of Microphone Level Balance in Blending the Timbre of Horn and Bassoon Players—*Sven-Amin Lembke, Scott Levine, Martha de Francisco, Stephen McAdams*, McGill University, Montreal, Quebec, Canada

A common musical aim of orchestration is to achieve a blended timbre for certain instrument combinations. Its success has been shown to also depend on the timbral coordination between musicians during performance, which this study extends by adding the subsequent involvement of sound engineers. We report the results from a production experiment in which sound engineers mixed independent feeds for a main and two spot microphones to blend the timbre of pairs of bassoon and horn players in a two-channel stereo mix. The balance of microphone feeds can be shown to be affected by leadership roles between performers, the musical material, and aspects related to room acoustics and performer characteristics.
Convention Paper 9397

11:30 am

P7-6 101 Mixes: A Statistical Analysis of Mix-Variation in a Dataset of Multi-Track Music Mixes—*Alex Wilson, Bruno Fazenda*, University of Salford, Salford, Greater Manchester, UK

The act of mix-engineering is a complex combination of creative and technical processes; analysis is often performed by studying the techniques of a few expert practitioners qualitatively. We propose to study the actions of a large group of mix-engineers of varying experience, introducing quantitative methodology to investigate mix-variation and the perception of quality. This paper describes the analysis of a dataset containing 101 alternate mixes generated by human mixers as part of an on-line mix competition. A varied selection of audio signal features is obtained from each mix and subsequent principal component analysis reveals four prominent dimensions of variation: dynamics, treble, width, and bass. An ordinal logistic regression model suggests that the ranking of each mix in the competition was significantly influenced by these four dimensions. The implications for the design of intelligent music production systems are discussed.
Convention Paper 9398

Tutorial 9 **Friday, October 30**
9:00 am – 10:30 am **Room 1A22**

REAL INDUSTRY: HOW MEDIA TECHNOLOGY PRODUCTS ARE REALLY MADE

Chair: **Jay LeBoeuf**, Real Industry, San Francisco, CA, USA; Stanford University, San Francisco, CA, USA

Panelists: *Dave Hill*, iZotope
Daniel Rowland, LANDR
Charles Van Winkle, Adobe Systems Inc., Minneapolis, MN, USA

How do leading audio companies bring products from ideation through commercialization? Join us we go behind the scenes at leading companies and explore key roles in the media tech industry including marketing, software development, hardware engineering, user experience, R&D, industrial design, and more. We'll feature 16 audio companies as our case studies.

Tutorial 10 **Friday, October 30**
9:00 am – 10:30 am **Room 1A06**

80S REDUX—THE SCIENCE BEHIND THE SOUND OF GATED DRUMS, YESTERDAY AND TODAY

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

The sound of 1980s pop music includes a heavy dose of gated drums. The effect—part discovered, and part invented—is still relevant today. History teaches us how to overdo it. Alex U. Case wants you to hear it in its contemporary form. You won't mix without it.

Tutorial 11 **Friday, October 30**
9:00 am – 10:45 am **Room 1A21**

MAIN MICROPHONE TECHNIQUES FOR 2.0 AND 5.1

Presenter: **Helmut Wittek**, SCHOEPS GmbH, Karlsruhe, Germany

This tutorial will explain step-by-step, using many practical examples, what a suitable stereophonic microphone array can look like. With 2.0 stereo setups as the starting point, multichannel setups will also be introduced.

Many factors influence the choice of a stereophonic microphone setup, but the relevance of these factors can vary greatly depending on the application, such that there is never one single "correct" setup. Knowledge of various options gives a Tonmeister the ability to make optimal choices.

In this session the free iPhone App "Image Assistant" will be presented. It calculates the spatial characteristics of arbitrary stereophonic microphone arrays and auralizes the result. Moreover, the educational website "hauptmikrofon.de" is presented offering various comparative sound samples on the subject.

Workshop 5 **Friday, October 30**
9:00 am – 10:30 am **Room 1A12**

IMMERSIVE AUDIO WITH HEIGHT CHANNELS: THEORY AND PRACTICE

Chair: **David Bowles**, Swineshead Productions LLC, Berkeley, CA, USA

Panelists: *Paul Geluso*, New York University, New York, NY, USA
Juergen Peissig, Sennheiser Electronics, Wedemark, Germany
Agnieszka Roginska, New York University, New York, NY, USA

The next step in immersive audio is to move into the vertical dimension through height channel recording and reproduction. Vital to this dialogue is a clearer understanding via psychoacoustics of our hearing perception outside the horizontal planes: how this perception influences engineers' choices in microphone technique and speaker placement. Members of this panel will discuss different recording techniques to capture height channels and whether this audio information can be integrated into conventional stereo and surround-sound recordings. The growing number of 3D mix engines and Blu-ray encoding formats will also be discussed. This workshop will be divided into two parts: a technical panel discussion at Javits Center, followed by playback sessions at the James Dolan Studios at New York University.

Workshop 6 **Friday, October 30**
9:00 am – 10:15 pm **Room 1A14**

HELP, I HAVE A TAPE MACHINE! (REDUX)

Chair: **Noah Simon**, New York University, New York, NY, USA

Panelists: *John French*, JRF Magnetic Sciences Inc., Greendell, NJ, USA
Chris Mara, Welcome to 1979, Nashville, TN, USA
Bob Shuster, Shuster Sound, Smithtown, NY USA
Daniel Zellman, Zeltec Service Labs, New York, NY, USA; Zeltec Research & Development

Although analog tape continues to be a viable format for recording and mixing audio, there is no denying the decline in the manufacture of machines, media, and tools for this beloved technology. In order to use analog tape reliably, machines must be maintained and repaired by users armed with technical know-how. This panel will cover topics related to the evaluation, maintenance, repair, and every-day practices to keep this technology functioning to its fullest potential.

Broadcast/Streaming Media 4 **Friday, October 30**
9:00 am – 10:30 am **Room 1A10**

AUDIO AND IP: ARE WE THERE YET?

Moderator: **Steve Lampen**, Belden, San Francisco, CA, USA

Panelists: *Kevin Gross*, AVA Networks, Boulder, CO, USA
David Josephson, Josephson Engineering, Inc., Santa Cruz, CA, USA
Dan Mortensen, Dansound Inc., Seattle, WA, USA
Tony Peterle, Worldcast Systems
Tim Pozar, Fandor

In 2010, Reed Hundt, former head of the Federal Communications Commission, said in a speech at Columbia Business School, [We] "decided in 1994 that the Internet should be the common medium in the United States and broadcast should not be." This was twenty-one years ago. So, are we there yet? I tried to invite Mr. Hundt to participate on this panel, but he is too well protected, I couldn't even get an invitation to him.

This panel of esteemed experts will look at the "big picture" of audio in networked formats and internet delivery systems. Do we have the hardware and software we need? If not, what is missing? Can we expect the same quality, consistency, and reliability as we had in the old analog audio days? There are dozens, maybe hundreds, of companies using proprietary Layer 2/Layer 3 Ethernet for audio, and there is much work on combining or cross-fertilizing these systems, such as Dante and Ravenna. There are also new standards such as IEEE 802.1BA-2011 AVB (Audio-Video Bridging), and IEEE 802.1ASbt TSN (Time-Sensitive Networks) that use specialized Ethernet switches in a network architecture. But these do not address anything outside of the Ethernet network itself. Then we have AES IP67, specifically looking at "high performance" IP-based audio.

Mixed in with this is the question "What is a broadcaster? Do you have to have a transmitter to be a broadcaster?" Consider that next year (2016) one company claims they will be the largest broadcaster in the world, and that company is Netflix.

Product Development 5 **Friday, October 30**
9:00 am – 10:30 am **Room 1A13**

BEST PRACTICES IN PRODUCTION TEST

Presenter: **Jonathan Novick**, Audio Precision, Camarillo, CA, USAt

Production test is an integral part of bringing quality products to market. It can also be time consuming, expensive, and produce inconsistent and confusing results. There is constant pressure to increase quality, track data, reduce test times, and reduce investment in the testing—goals that often conflict with each other. Modern measurement techniques based on best practices, standards, and available technology can provide a high degree of consistency, quality with low test times. In today's diverse product organizations, it is also important to be able to communicate test specifications and results across engineering, product management, manufacturing, and supply chain. This session explores the trade-offs one must consider when implementing new test methodologies. Real-world case studies will be discussed and comparisons made between different approaches. This session will also discuss the benefits and caveats of various test approaches and how a good test strategy can become a competitive advantage.

Student Event & Career Development
STUDENT RECORDING CRITIQUES
Friday, October 30, 9:00 am – 10:00 am
Room 1A18

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

Students! Bring your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on memory-stick, or hard disk, as clearly labeled 24/44.1 KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers!

Friday, October 30 9:00 am Room 1A19

Technical Committee Meeting on Perception and Subjective Evlauation of Audio Signals

Special Event
PLATINUM PRODUCERS – BAD VIBRATIONS
Friday, October 30, 9:15 am – 10:45 am
Room 1A23/24

Moderator: **Errol Kolisine**

Panelists: *Joel Hamilton*
Nick Sansano
Hank Shocklee

We all spend plenty of time discussing the tangible technical elements of making a great record, but what about certain more etherial challenges? Our panel of world class producers discuss some of the worst distractions, impediments and energy vampires—and how to combat them.

Friday, October 30 10:00 am Room 1A19

Technical Committee Meeting on Acoustics and Sound Reinforcement

Networked Audio 3 Friday, October 30
10:30 am – 11:30 am Room 1A14

AVB/TSN IMPLEMENTATION FOR LIVE SOUND AND HOUSE OF WORSHIP

Presenter: **Tom Knesel**, Pivitec, Emmaus, PA, USA

Ethernet AVB/TSN (Time Sensitive Networking) enables precise timing and synchronization and bandwidth reservation, making it an ideal solution for several of the consistency and convenience issues musicians' face on the road and during live performances. International rock band, ACCEPT, has been touring the globe for over three decades, and for most of that time, they had to lug heavy performance equipment onto planes, trains and taxis or take the risk of using unfamiliar local equipment at each venue. They needed a solution that eliminated some gear and ensured a consistent sound and performance at each venue. That's where a compact touring system, powered by Audio Video Bridging (AVB) stepped in.

For Houses of Worship, similar solutions can be implemented. NOW Church in Ocala, FL, has always incorporated cutting-edge technology into their facility, but they were looking for a system that would take them into the future. After hearing about the benefits of AVB they made the move to an AVID VENUE system for their Front of House, as well as the Pivitec personal monitoring system that they have described as a "game-changer."

Tom Knesel, Co-Founder and President of Pivitec, will walk through the specifics of the AVB enabled systems for each install including lessons learned and how AVB was monumental in providing a powerful experience during ACCEPT's and NOW Church's performances. Knesel will present how AVB allowed these two installations to combat common sound issues on stage, create pre-sets, simplify travel, and most importantly, give them a future-proof way to take advantage of next-gen compatibility with hardware and software from other manufacturers.

Project Studio Expo 1 PSE Stage
Friday, October 30 10:30 am –11:15 am

MAKING THE MOST OF YOUR STUDIO PURCHASING BUDGET

Presenter: **Paul White**, Sound On Sound, UK

Sound On Sound Editor In Chief Paul White offers down-to-earth perspective on what really makes a difference in your studio equipment and how to get the best out of a home studio without relying on your wallet to bail you out!

Workshop 7 Friday, October 30
10:45 am – 12:45 pm Room 1A06

THE X-FACTOR IN AUDIO

Chair: **Darcy Proper**, Mastering Engineer, Wisseloord Studios, Hilversum, The Netherlands

Panelists: *Cynthia Daniels*, Recording Engineer and Producer, Monk Music Studios
Leslie Ann Jones, Recording Engineer and Producer, Director of Music Recording and Scoring, Skywalker Sound, San Rafael, CA, USA
Piper Payne, Coast Mastering, Dearborn, MI, USA
Michelle Sabolchick Pettinato, Freelance FOH Engineer
Ulrike Schwarz, Tonmeister/MBA Anderson Audio (formerly at Bayerischer Rundfunk, Germany), Munich, Germany
Leanne Ungar, Berklee College of Music, West Newton, MA, USA

"The X-Factor in Audio": What does it take for an engineer to rise to the top of his/her profession? What is the "magic" that makes a session special (and productive) for all involved? What elements combine to elevate a live concert from "a nice night out" to "a show we'll never forget"?

No, Simon Cowell will not be making an appearance here. Instead, this workshop will feature a remarkable group of audio engineers, outstanding in their respective fields—live sound, recording, mixing, and mastering—for a wide variety of musical genres including classical, pop/rock/r&b, film, Broadway, and television.

These engineers have worked with an impressive list of artists, including: Herbie Hancock, Michel Feinstein, Nile Rodgers, Leonard Cohen, Laurie Anderson, "The Producers" cast, Mumford & Sons, Florence & The Machine, Sir Simon Rattle, Yo Yo Ma, Donald Fagen, Marcus Miller, Gwen Stefani, The Goo Goo Dolls, and dozens more.

Join these award-winning panelists as they share their life-long adventures in audio and share their tips and tricks for surviving in this challenging industry, told from their unique perspective as women in a male-dominated profession.

Workshop 8 Friday, October 30
10:45 am – 12:15 pm Room 1A12

ISO/MPEG-H AUDIO—THE NEW STANDARD FOR UNIVERSAL SPATIAL / 3D AUDIO CODING*

Chair: **Jürgen Herre**, International Audio Laboratories Erlangen, Erlangen, Germany; Fraunhofer IIS, Erlangen, Germany

Panelists: *Alexander Krüger*, Technicolor
Nils Peters, Qualcomm, San Diego, CA, USA
Jan Plogsties, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Recently, the ISO/MPEG standardization group created the MPEG-H 3D Audio specification to go along with Ultra High Definition Television

(UHDT) video. The specification features several unique elements, such as handling of channel-based content, object-based content and higher order ambisonics (HOA) content or the capability of rendering encoded high-quality content on a wide range of loudspeaker setups (22.2 … 5.1 … stereo / headphones). This workshop provides an overview of the MPEG-H 3D Audio standard regarding its underlying architecture, technology, performance and how to produce immersive content for it.

**This session is presented in association with the AES Technical Committees on Coding of Audio Signals, Spatial Audio*

Broadcast/Streaming Media 5 Friday, October 30
10:45 am – 12:15 pm Room 1A10

AUDIENCE MEASUREMENT FOR STREAM AND BROADCAST

Moderator: **David Layer**, National Association of Broadcasters, Washington, DC, USA

Panelists: *Frank Foti*, Telos Systems/Omnia Audio, New York, NY, USA
Rob Green, Vice President WO Streaming, Digital and Programmatic, WideOrbit, Inc.
John Rosso, President, Market Development, Triton Digital

Radio broadcasting in the 21st Century is not just about over-the-air signals. The Internet and mobile broadband have made audio streaming a viable option to over-the-air delivery. One thing that has not changed, audience measurement is still fundamental to the business of radio no matter how the signal is being delivered. This 90-minute session will focus on some of the latest audience measurement techniques for both over-the-air and streaming and help you to understand some of the nuances of the methods and technologies employed.

Product Development 6 Friday, October 30
10:45 am – 12:15 pm Room 1A13

WHAT HAPPENS IN A PATENT LAWSUIT

Presenters: **John Strawn**, S Systems Inc., Larkspur, CA, USA
Tom Millikan, Perkins Coie LLP, San Diego, CA, USA

This session covers the mechanics of patent lawsuits and what you can expect when you are involved, whether you are an owner, manager, engineer, or employee. We will cover the basic steps including: starting a lawsuit; proving a product infringes a patent, proving a patent is invalid, using experts to show infringement or invalidity; deposing experts and company personnel; asking the judge to end the case; limiting what information is available at trial, and trying a case. There will be a detour through the recently established procedures to challenge patents at the patent office rather than in court. The presentation will involve real-world experience, including our work in what was the largest audio patent case in US history—Lucent v. Microsoft—where the MP3 standard itself was on trial for patent infringement. We will present information on how often and at what stage cases settle, as most do. And we will share insights on how to win.

Tutorial 12 Friday, October 30
11:00 am – 12:30 pm Room 1A22

AUDIO FORENSICS: OVERVIEW 1*

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

Panelists: *Catalin Grigoras*, University of Colorado at Denver, Denver, CO, USA
Gordon Reid, CEDAR Audio Ltd., Cambridge, UK

This tutorial will feature 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.

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

Session P8 Friday, Oct. 30
11:00 am – 12:30 pm Foyer

P0STERS: SIGNAL PROCESSING

11:00 am

P8-1 Robust MPEG-4 High-Efficiency AAC With Fixed- and Variable-Length Soft-Decision Decoding—*Sai Han, Tim Fingscheidt*, Technische Universität Braunschweig, Braunschweig, Germany

MPEG-4 High-Efficiency advanced audio coding (HE-AAC) is optimized for low bit rate applications, such as digital radio broadcasting and wireless music streaming. In HE-AAC, the differential scale factors and quantized spectral coefficients are variable-length coded (VLC) by Huffman codes. The common reference value of the scale factors is a fixed-length coded global gain. Due to the error propagation in VLCs, a robust source decoder is desired for HE-AAC transmission over an error-prone channel. Unlike traditional hard-decision decoding or error concealment, soft-decision decoding utilizing bit-wise channel reliability information offers improved audio quality. In this work we apply soft-decision decoding of fixed length to the global gain and of variable length to the scale factors. Simulation results show a clearly improved performance.
Convention Paper 9399
[This poster was not presented]

11:00 am

P8-2 Extension of Monaural to Stereophonic Sound Based on Deep Neural Networks—*Chan Jun Chun,[1] Seok Hee Jeong,[1] Su Yeon Park,[1] Hong Kook Kim[1,2]*
[1] Gwangju Institute of Science and Tech (GIST), Gwangju, Korea
[2] City University of New York, New York, NY, USA

In this paper we propose a method of extending monaural into stereophonic sound based on deep neural networks (DNNs). First, it is assumed that monaural signals are the mid signals for the extended stereo signals. In addition, the residual signals are obtained by performing the linear prediction (LP) analysis. The LP coefficients of monaural signals are converted into the line spectral frequency (LSF) coefficients. After that, the LSF coefficients are taken as the DNN features, and the features of the side signals are estimated from those of the mid signals. The performance of the proposed method is evaluated using a log spectral distortion (LSD) measure and a multiple stimuli with a hidden reference and anchor (MUSHRA) test. It is shown from the performance comparison that the proposed method provides lower LSD and higher MUSHRA score than a conventional method using hidden Markov model (HMM).
Convention Paper 9400

11:00 am

P8-3 Nonnegative Tensor Factorization-Based Wind Noise Reduction—*Kwang Myung Jeon,[1] Ji Hyun Park,[1] Seung Woo Yu,[1] Young Han Lee,[2] Choong Sang Cho,[2] Hong Kook Kim[1,3]*

[1] Gwangju Institute of Science and Tech (GIST), Gwangju, Korea
[2] Korea Electronics Technology Institute (KETI), Seongnam-si, Gyeonggi-do, Korea
[3] City University of New York, New York, NY, USA

In this paper a wind noise reduction method based on nonnegative tensor factorization (NTF) is proposed to enhance the audio quality recorded using an outdoor multichannel microphone array. The proposed method first prepares learned bases for NTF by training exemplar blocks of spectral magnitudes for a series of wind noises and audio contents. Then, the spectral magnitudes of wind noise to be reduced are estimated from the exemplar blocks. Finally, a wind noise reduction multichannel filter is constructed based on a minimum mean squared error (MMSE) criterion and applied to the multichannel noisy signal to obtain the signal with reduced wind noise. The performance of the proposed method is compared with those of conventional methods using minimum statistics (MS) and nonnegative matrix factorization (NMF) for wind noise reduction. As a result, it is shown that the proposed method provides a higher signal-to-distortion ratio (SDR), signal-to-interference ratio (SIR), and signal-to-artifact ratio (SAR) than the conventional methods under various signal-to-noise ratio (SNR) conditions.
Convention Paper 9401

11:00 am

P8-4 Detection and Removal of the Birdies Artifact in Low Bit-Rate Audio—*Simon Desrochers, Roch Lefebvre*, Universite de Sherbrooke, Sherbrooke, QC, Canada

Audio signals compressed at low bit rates are known to generate audible artifacts that degrade perceptual quality. These different artifacts have been documented and solutions have been proposed by many authors to modify the internal mechanisms of codecs that cause these artifacts. In this paper we propose a post-processing approach to detecting and removing the birdies artifact by modeling spectral components as partials. This approach has the advantage of being compatible with any codec as it only requires the compressed signal. Formal listening tests have shown that this prototype algorithm can increase the perceptual quality of birdies-ridden signals. Furthermore, the explicit detection of this artifact could eventually be used in an objective perceptual quality assessment algorithm.
Convention Paper 9402

11:00 am

P8-5 Using Cascaded Global Optimization for Filter Bank Design in Low Delay Audio Coding—*Stephan Preihs, Jörn Ostermann*, Leibniz Universität Hannover, Hannover, Germany

This paper demonstrates the possibility of finding suitable design parameters for a filter bank optimization procedure by the use of global optimization techniques. After the Nayebi filter bank optimization algorithm is summarized and its degrees of freedom are described, a global optimization framework for the parameters involved in the design process is presented. It includes a cost function monitoring the frequency characteristics and reconstruction properties of different filter bank designs and allows for an automatic search of suitable design parameters for a given number of bands and taps and a predefined delay. The global optimization itself is done by means of well known methods like pattern search and the genetic algorithm. Experiments show that with our method manual parameter adjustment becomes obsolete. Furthermore with our proposed cascaded optimization, compared to manually adjusted designs, a gain of up to 10 dB in stopband attenuation can be achieved without loss in reconstruction quality.
Convention Paper 9403

11:00 am

P8-6 Effect of Reverberation on Overtone Correlations in Speech and Music—*Sarah R. Smith, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

This paper explores the effect of reverberation on audio signals that possess a harmonically rich overtone spectrum such as speech and many musical instrument sounds. A proposed metric characterizes the degree of reverberation based upon the cross correlation of the instantaneous frequency tracks of the signal overtones. It is found that sounds that exhibit near perfect correlations in an anechoic acoustic environment become less correlated when passed through a reverberant channel. These results are demonstrated for a variety of music and speech tones using both natural recordings and synthetic reverberation. The proposed metric corresponds to the speech transmission index and thus may be employed as a quantitative measure of the amount of reverberation in a recording.
Convention Paper 9404

11:00 am

P8-7 Stacked Modulation in a Hall Reverberation Algorithm—*Kelsey M. Cintorino, Daniel M. Wisniewski, Benjamin D. McPheron*, Roger Williams University, Bristol, RI, USA

Reverberation is the reflection of sound caused by objects in space, similar to the way the visual world is sensed by the reflection of light. Novel reverberation algorithms are in high demand within the music industry due to changing trends and desire for unique sounds. As DSP hardware has improved, it is easier to implement multiple effects into the same algorithm. This paper presents a hall algorithm augmented with a series of chorus modulation blocks in an attempt to create new sounds. The approach is to add chorus blocks before the early decay phase of the hall algorithm as well as within the late reverb generation phase. The result is a stacked modulation reverberation algorithm.
Convention Paper 9405

11:00 am

P8-8 Efficient Multi-Band Digital Audio Graphic Equalizer with Accurate Frequency Response Control—*Richard J. Oliver*,[1] *Jean-Marc Jot*[2]
[1] DTS, Inc., Santa Ana, CA, USA
[2] DTS, Inc., Los Gatos, CA, USA

Graphic equalizers give listeners an intuitive way to modify the frequency response of an audio signal—simply set the sliders to visually represent the desired curve and the corresponding shape of audio filter frequency response will be invoked. At least, that is the implied promise of the technology. However, the actual measured response of the equalizer can reveal some surprises. Filter inaccuracy, boost/cut asymmetry and unexpected nulls can disappoint both the eye and the ear. An equalizer design is presented that uses efficient IIR filter sections tuned with a closed form algorithm to give an accurate and intuitive frequency response with low complexity and minimal processing overhead. Design parameters and implementation details are discussed.
Convention Paper 9406

Workshop 9 — Friday, October 30
11:00 am – 1:00 pm — Room 1A21

GIVE PEAKS A CHANCE

Chair: **Thomas Lund**, Genelec, Risskov, Denmark

Panelists: *Florian Camerer*, ORF - Austrian TV, Vienna, Austria; EBU - European Broadcasting Union
Bob Katz, Digital Domain Mastering, Orlando, FL, USA
Bob Ludwig, Gateway Mastering Studios, Inc., Portland, ME, USA
George Massenburg, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Susan Rogers, Berklee College of Music, Boston, MA, USA

The Tribunal is ready to present an update on the Loudness Wars. Important 2015 developments include Apple Music and WHO's sudden attention to sound exposure from Personal Media Players (PMPs). Both will have an impact on our industry at large, good or bad.

The panel will cover production, mastering, and distribution of quality audio, and we will discuss how the prevention of hearing loss fits in as a key element. While our most social sense must be preserved, cultural aspects should not be forgotten. We need something worth listening to also :-).

Live Sound Expo 1 — LSE Stage
Friday, October 30 — 11:00 am –11:45 am

THEATRICAL VOCAL MIKING

Presenters: **James Caparelle**
Jim van Bergen
Ken Travis

In theatrical vocal applications, mics should largely be heard and not seen. Our session covers the practical issues of reproducing song and voice from the stage, including body mic dressing, use of omnis vs. directional polar patterns, earset vs. hairline mics.

Friday, October 30 — 11:00 am — Room 1A19

Technical Committee Meeting on Human Factors in Audio Systems

Friday, October 30 — 11:00 am — Room 1A20

Standards Committee Meeting SC-05-02 Audio Connectors

Project Studio Expo 2 — PSE Stage
Friday, October 30 — 11:30 am – 12:30 pm

THE FIVE MOST COMMON PROJECT STUDIO RECORDING MISTAKES

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.

Workshop 10 — Friday, October 30
11:45 pm – 12:45 pm — Room 1A14

EDUCATING YOUR CLIENTS ON THE VINYL PROCESS

Chair: **Cameron Henry**, Welcome To 1979, Nashville, TN, USA

Panelists: *Bruce Dickinson*, Legacy Music
Jeffrey James, Director of A&R Sony Music Entertainment

The purpose of this workshop is to educate people in the recording industry of the unique requirements of the vinyl process. Everyone is aware that vinyl sales are increasing dramatically, this workshop will help educate on ways to better deal with vinyl and help find ways to potentially increase revenue by offering vinyl-centric services.

Session EB3 — Friday, Oct. 30
12:00 noon – 1:00 pm — Room 1A07

TRANSDUCERS—PART 2

Chair: **Michael Smithers**, Dolby Australia, McMahons Point, NSW, Australia

12:00 noon

EB3-1 Dual Filtering Technique for Speech Signal Enhancements—*Mahdi Ali*, Hyundai-Kia America Technical Center, Superior Township, MI, USA

Hands free applications in automobiles, such as voice recognition and Bluetooth communications, have been one of the great features added to vehicles' infotainment systems. However, cab-in and road noises degrade audio quality and negatively impact consumers' experience. Research has been conducted and several noise reduction techniques have been proposed. However, due to the complexity of noise environments inside and outside vehicles, the hands free sounds quality still poses an issue for consumers. This continuously existed problem requires more research in this field. This paper proposes a novel technique to reduce noise and enhance voice recognition and Bluetooth audio quality in vehicles' hands free applications. It utilizes the dual Kalman Filtering (KF) technique to suppress noise. This method has been validated using a MATLAB/SIMULINK simulation environment, which showed improvements in noise reductions in both Gaussian and non-Gaussian environments.
Engineering Brief 209
[This eBrief was not presented but is available in the E Library]

12:15 pm

EB3-2 Implementation of Segmented Circular-Arc Constant Beamwidth Transducer (CBT) 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, and 4]. To simplify these arrays, the arrays may be formed 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 210

12:30 pm

EB3-3 Speech Intelligibility Advantages Using an Acoustic Beamformer Display—*Durand Begault*,[1] *Kaushik Sunder*,[1,2] *Martine Godfroy*,[1,2] *Peter Otto*[3]
[1] NASA Ames Research Center, Moffet Field, CA, USA
[2] San Jose State University Foundation, San Jose, CA, USA
[3] Calit2, UC San Diego, La Jolla, CA, USA

A speech intelligibility test conforming to the Modified Rhyme Test of ANSI S3.2 "Method for Measuring the Intelligibility of Speech Over Communication Systems" was conducted using a prototype 12-channel acoustic beamformer system. The target speech material (signal) was identified against speech babble (noise), with calculated signal-noise ratios of 0, 5 and 10 dB. The signal was delivered at a fixed beam orientation of 135 degrees (re 90 degrees as the frontal direction of the array) and the noise at 135 (co-located) and 0 degrees (separated). A significant improvement in intelligibility from 58% to 75% was found for spatial separation for the same signal-noise ratio (0 dB). Significant effects for improved intelligibility due to spatial separation were also found for higher signal-noise ratios.
Engineering Brief 211

12:45 pm

EB3-4 Designing Near-Field MVDR Acoustic Beamfomers for Voice User Interfaces—*Andrew Stanford-Jason*, XMOS Ltd.

We present an analysis and design recommendations for a reduced computational complexity minimum variance distortionless response(MVDR) beamforming microphone. An overview of MVDR beamforming is given then decomposed into a generalized implementation to aid mapping to a microcontroller with some discussion of optimizations for real-world performance.
Engineering Brief 212

Live Sound Expo 2 — LSE Stage
Friday, October 30 — 12:00 noon –12:45 pm

WIRELESS ISSUES FOR LIVE THEATER: BROADWAY AND BEYOND

Moderator: **Karl Winkler**

Panelists: *Chris Evans*
Simon Matthews

Manhattan's Broadway represents one of the most hostile environments imaginable for wireless microphone use. How do sound designers and system engineers cope with the RF soup that fills the ether in "The Great White Way," and what lessons learned can be applied to theater applications in general? This session will offer answers.

Friday, October 30 — 12:00 noon — Room 1A19

Technical Committee Meeting on Audio for Games

Tutorial 13 — Friday, October 30
12:15 pm – 1:15 pm — Room 1A12

CANCELED

Spatial Audio Demo 6 — Friday, October 30
12:30 pm – 1:30 pm — Room 1A18

ISO/MPEG-H AUDIO—THE NEW STANDARD OR UNIVERSAL SPATIAL / 3D AUDIO CODING

Presenter: **Valentin Schilling**, Fraunhofer IIS, Erlangen, Germany

Recently, the ISO/MPEG standardization group created the MPEG-H 3D Audio specification to go along with Ultra High Definition Television (UHDT) video. The specification features several unique elements, such as handling of channel-based content, object-based content, and higher order ambisonics (HOA) content or the capability of rendering encoded high-quality content on a wide range of loudspeaker setups (22.2 ... 5.1 ... stereo / headphones). This demonstration showcases some of the features and sonic performance of the MPEG-H 3D Audio standard.

This demonstration supports the workshop of the same name given on Friday (Workshop 8).

Special Event
LUNCHTIME KEYNOTE—ED GREENE
Friday, October 30, 12:30 pm – 1:30 pm
Room 1A10

Presenter: **Ed Greene**

Come on Down

CBS – "The Price is Right": This classic American game show just started its 44th season. While the core of TPIR's success, it's host, contestants, audiences, and games remains the same, the producers are now using recent technology to enhance the show's appeal. A description with demonstrations of these techniques will reveal why listeners and studio audiences can't wait to "Come on Down."

"Whose Line Is it Anyway?": A genuinely improv comedy that originated in the UK. The UK production team came to America in the late 90s to produce this very successful show for American audiences on ABC. The same producers have now returned to record new shows for the CW. With continued success, the show has just completed production for its 3rd new season.

Ed Greene, who mixes both shows, will "Come on Down" to provide detailed insights to both shows audio anatomy.

Project Studio Expo 3 — LSE Stage
Friday, October 30 — 12:45 pm – 1:30 pm

OUTSIDE THE BOX: ALTERNATIVE OUTLETS FOR YOUR MUSIC

Presenters: **Steve Horowitz**, Game Audio Institute, San Francisco, CA, USA; Nick Digital
Christopher Kaufman, Composer/Educator/Trans Media Author
Jerome Rossen, Freshmade Music, San Francisco, CA, USA; Manhattan Producers Alliance
Brian Walker, Audio Director, Leap Frog
Richard Warp, Manhattan Producers Alliance, San Francisco, CA; Leapfrog Enterprises Inc., Emeryville, CA, USA

Careers in music don't just revolve around hit records, and never more so than in today's fragmented music business. There are many other ways of exploiting musical creativity and production skills, as this seminar and panel discussion will demonstrate.

Special Event
DTVAG AES FORUM
Friday, October 30, 1:00 pm – 5:00 pm
Room 1A14

Keynote Speaker: *Tom Sahara*, Turner Sports Vice President, Operations and Technology, Turner Sports, Atlanta, GA, USA

The Accelerating Pace of Change in Television Audio

What Just Happened? The impacts of mobile and fixed streaming services have been even greater and more far-reaching than previously predicted. Will this pace of change continue … or accelerate?

Other discussion topics will include:

Wireless Spectrum Roadmap
The FCC's recent release of rule making around the 600 MHz incentive auction and wireless microphone use provides some clarity about the future but still leaves many questions unanswered.

ATSC 3.0 Audio Update
The multi-year process of defining an audio standard for next-generation broadcast television is coming to a close. What capabilities can we expect? Will ATSC 3.0 provide a template for other future audio services?

Microphone Metadata and Network Control
Networked wireless and native IP microphones have the potential to streamline identification and control of multiple sources in complex mixing environments. How does organic microphone metadata change our approach to mixing, automation, and object audio authoring?

Console Metadata Authoring
As console mixing functions become virtualized over the production WAN, best practices and universal standards need to keep up with the demand for real-time exportable metadata. How is that going?

Higher-Order Ambisonics and Scene-Based Audio
Recent advances in audio coding and real-time processing have made the application of HOA capture and encoding techniques significantly more practical. What are the practical implications of applying HOA techniques to real-time production?

Audio Definition Modeling
Recent EBU standards-making efforts around ADM technology and Broadcast Wave File extensions encompassing ADM open the door for standardized object metadata. With the backing of Dolby and others, is universal audio file interoperable delivery around the corner?

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

Live Sound Expo 3 — LSE Stage
Friday, October 30 — 1:00 pm –1:45 pm

THEATER SOUND SYSTEM DESIGN AND OPTIMIZATION

Presenters: **Andrew Keister**
Bob McCarthy

Theater sound designers can face architectural and aesthetic concerns within a given facility, audio content that ranges from dialog heavy drama to rocking reviews and a blend of live and recorded elements. Seasoned veterans of theatrical sound design will share their experience.

Friday, October 30 — 1:00 pm — Room 1A19

Technical Committee Meeting on Fiber Optics for Audio

Networked Audio 4 — Friday, October 30
1:30 pm – 2:30 pm — Room 1A13

NAMESPACE & DIRECTORY SERVICES (I.E., ODA)

Presenter: **Jeff Berryman**, Bosch Communications, Ithaca, NY, USA

AES67 addresses interoperability in media transport over networks, AES X210 addresses command and control over networks (as does the ill fated AES64). Neither of these standards efforts tackles Namespace and Directory services, aspects such as Discovery and addressing. This session will explore requirements for these network services and looks forward to an AES standards effort for these services.

Special Event
PLATINUM ENGINEERS
Friday, October 30, 1:30 pm – 3:00 pm
Room 1A23/24

Moderator: **Justin Colletti**, SonicScoop, Brooklyn, NY, USA; Trust Me, I'm a Scientist

Panelists: *Joe Chiccarelli*
John Congleton
Chris Godbey
Chris Zane

The Platinum Engineers Panel, hosted by Justin Colletti of SonicScoop and Joe Lambert Mastering, gathers top engineers for a discussion about ideas and techniques in contemporary record making. Each of the engineers selected is a multi-talented collaborator on major music projects, able to wield the studio as an instrument to the creative advantage of each recording. Guests panelists play excerpts from their work, discuss their approach in detail, and answer questions from the audience.

Broadcast/Streaming Media 6 — Friday, October 30
1:45 pm – 3:15 pm — Room 1A10

PRODUCTION OF *A PRAIRIE HOME COMPANION*

Moderator: **John Holt**, Retired

Panelists: *Sam Hudson*, Broadcast Engineer/Talent Producer
Nick Kereakos
Thomas Scheuzger, Broadcast/Transmission Engineer

"From the control board at the Orpheum, PHC travels via underground phone lines to the tiny Satellite Control Room on the fourth floor of Minnesota Public Radio, from there by cable to MPR's transmitting dish in a junkyard on the East Side of Saint Paul, and from there 22,300 miles to Western Union's Westar IV satellite…"

A lot has changed technically since that was written over 30 years ago for the 10th anniversary of "A Prairie Home Companion." You'll hear a little history and a lot about how the technology has changed and will change the production and distribution of this iconic radio program.

Game Audio 5 — Friday, October 30
1:45 pm – 2:45 pm — Room 1A21

JUST CAUSE 3: POSTMORTEM ON THE GAME AUDIO OF A MASSIVE OPEN WORLD GAME

Presenters: **Jason Kanter**, Avalanche Studios
Dominic Vega, Avalanche Studios

Open world sandbox games (OWS) are arguably the most challenging to provide sound for. Two members from Avalanche Studios' audio department present what it was like providing game audio for one of the largest OWS's in the genre. Spanning 3½ years in development across two continents with a content creation team fluctuating in size from 1–4½ people, the audio department on JC3 had their fair share of obstacles. Jason and Dom will discuss what it was like to create the soundtrack for a world filled with fast cars, big guns and even bigger explosions in a Southern European Mediterranean island setting, all while attempting to keep the game as "Made in NY" as possible.

Project Studio Expo 4 — PSE Stage
Friday, October 30 — 1:45 pm – 2:30 pm

MIX AND MASTERING OPTIMIZED FOR STREAMING

Presenter: **Thomas Lund**, Genelec, Finland

Loudness-based normalization in distribution is a game-changer. Even the resulting loudness at the consumer now follows an inverted U curve as mastering levels are cranked up. Free tools for equal-loudness comparisons are shown, and new streaming recommendations from AES and EBU are summarized. The session includes listening examples and tips for optimized delivery that will stand the test of time.

Session P9 — Friday, October 30
2:00 pm – 5:00 pm — Room 1A08

TRANSDUCERS—PART 3: LOUDSPEAKERS

Chair: **Peter John Chapman**, Harman, Struer, Denmark

2:00 pm

P9-1 A Model for the Impulse Response of Distributed-Mode Loudspeakers and Multi-Actuator Panels—*David Anderson, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Panels driven into transverse (bending) vibrations by one or more small force drivers are a promising alternative approach in loudspeaker design. A mechanical-acoustical model is presented here that enables computation of the acoustic transient response of such loudspeakers driven by any number of force transducers at arbitrary locations on the panel and at any measurement point in the acoustic radiation field. Computation of the on- and off-axis acoustic radiation from a panel confirms that the radiated sound is spatially diffuse. Unfortunately, this favorable feature of vibrating panel loudspeakers is accompanied by significant reverberant effects and such loudspeakers are poor at reproducing signals with rapid transients.
Convention Paper 9409

2:30 pm

P9-2 Loudspeaker Rocking Modes (Part 1: Modeling)—*William Cardenas, Wolfgang Klippel*, Klippel GmbH, Dresden, Germany

The rocking of the loudspeaker diaphragm is a severe problem in headphones, micro-speakers, and other kinds of loudspeakers causing voice coil rubbing that limits the maximum acoustical output at low frequencies. The root causes of this problem are small irregularities in the circumferential distribution of the stiffness, mass, and magnetic field in the gap. A dynamic model describing the mechanism governing rocking modes is presented and a suitable structure for the separation and quantification of the three root causes exciting the rocking modes is developed. The model is validated experimentally for the three root causes and the responses are discussed conforming a basic diagnostics analysis.
Convention Paper 9410

3:00 pm

P9-3 Active Transducer Protection Part 1: Mechanical Overload—Wolfgang Klippel, Klippel GmbH, Dresden, Germany

The generation of sufficient acoustical output by smaller audio systems requires maximum exploitation of the usable working range. Digital preprocessing of audio input signals can be used to prevent a mechanical or thermal overload generating excessive distortion and eventually damaging the transducer. The first part of two related papers focuses on the mechanical protection defining useful technical terms and the theoretical framework to compare existing algorithms and to develop meaningful specifications required for the adjustment of the protection system to the particular transducer. The new concept is illustrated with a micro-speaker and the data exchange and communication between transducer manufacturer, software provider, and system integrator are discussed.
Convention Paper 9411

3:30 pm

P9-4 Horns Near Reflecting Boundaries—*Bjørn Kolbrek*, Norwegian University of Science and Technology, Trondheim, Norway

It is well known that when a sound source is placed near one or more walls, the power output increases due to the mutual coupling between the source and its image sources. This is reflected in an increase in the low frequency radiation resistance as seen by the sources. While direct radiating loudspeakers may benefit from this whenever the sources are within about a quarter wavelength of each other, horns will behave differently depending on if the increase in radiation resistance comes within the pass band of the horn or not. This has implications for the placement of corner horns. In this paper the Mode Matching Method (MMM) is used together with the modal mutual radiation impedance and the concept of image sources to compute the throat impedance and radiated sound pressure of horns placed near infinite, perpendicular reflecting boundaries. The MMM is compared with another numerical method, the Boundary Element Rayleigh Integral Method (BERIM), and with measurements and is shown to give good agreement with both. The MMM also has significantly shorter computation time than BERIM, making it attractive for use for the initial iterations of a design, or for optimization procedures.
Convention Paper 9412

4:00 pm

P9-5 State-Space Modeling of Loudspeakers Using Fractional Derivatives—*Alexander King, Finn T. Agerkvist*, Technical University of Denmark, Kgs. Lyngby, Denmark

This work investigates the use of fractional order derivatives in modeling moving-coil loudspeakers. A fractional order state-space solution is developed, leading the way towards incorporating nonlinearities into a fractional order system. The method is used to calculate the response of a fractional harmonic oscillator, representing the mechanical part of a loudspeaker, showing the effect of the fractional derivative and its relationship to viscoelasticity. Finally, a loudspeaker model with a fractional order viscoelastic suspension and fractional order voice coil is fit to measurement data. It is shown that the identified parameters can be used in a linear fractional order state-space model to simulate the loudspeakers' time domain response.
Convention Paper 9413

4:30 pm

P9-6 Comparative Static and Dynamic FEA Analysis of Single and Dual Voice Coil Midrange Transducers—*Felix Kochendörfer, Alexander Voishvillo*, JBL/Harman Professional, Northridge, CA, USA

The concept of the dual coil direct-radiating loudspeakers have been known for several decades. JBL Professional pioneered in design and application of dual coil woofers and midrange loudspeakers. There are several properties of the dual coil transducers that differentiate them from the traditional single voice coil design. First is the better heat dissipation—the dual coil may be considered as a traditional coil slit in two parts and each one is positioned into its own magnetic gap. Second is the symmetry of the force factor (Bl product) versus position of the voice coils in their gaps. It is explained by the fact that one coil leaves its gap the other one on contrary enters its gap. These two features are well researched and described in literature [1,2]. Less is known about advantage of the dual coil transducers related to the flux modulation and dependence of the alternating magnetic flux (and corresponding voice coil inductance) on frequency, current, and voice coil positions. In this work comparison of a regular single coil design and dual coil configuration is carried out through dynamic magnetic FEA modeling and measurements.
Convention Paper 9414

Session P10 **Friday, October 30**
2:00 pm – 4:30 pm **Room 1A07**

RECORDING & PRODUCTION

Chair: **Grzegorz Sikora**, Bang & Olufsen Deutschland GmbH, Pullach, Germany

2:00 pm

P10-1 Lossless Audio Checker: A Software for the Detection of Upscaling, Upsampling, and Transcoding in Lossless Musical Tracks—*Julien Lacroix,*[1] *Yann Prime,*[1] *Alexandre Remy,*[1] *Olivier Derrien*[2]
[1]Independent Developer
[2]University of Toulon / CNRS-LMA, Toulon, France

Internet music dealers currently sell "CD quality" tracks, or even better ("Studio Master"), thanks to lossless audio coding formats (FLAC, ALAC). However, a lossless format does not guarantee that the audio content is what it seems to be. The audio signal may have been upscaled (increasing the resolution), upsampled (increasing the sample rate), or even transcoded from a lossy to a lossless format. In this paper we describe a new software that analyzes lossless audio tracks and detects upsampling, upscaling, and transcoding (only for AAC in this early version). Validation tests over a large music database (with groundtruth available) show that this method is fast and accurate: 100% of success for upscaling and transcoding, 91.3% for upsampling.
Convention Paper 9416

2:30 pm

P10-2 Comparison of Audio Signals Obtained with Source Overlay (OAS) and Other Conventional Recording Methods—*Juliette Olivella, William Romo, Dario Páez*, Universidad de San Buenaventura, Bogotá, Colombia

Overlay Model of Acoustic Sources (OAS) is an unconventional recording method with a stereo microphone array. This model was proposed as a methodological alternative that allows emulating a recording single-take of a musical group. It is based on the presumption of a linear behavior in a recording system and involves doing partial captures of musical instruments that integrate the entire assembly. Experimental tests were done to corroborate the system's linearity; two speakers are used instead of musicians and audio is recorded with conventional techniques and model of Overlay of Acoustic Sources. The audios were discretized using MATLAB in order to evaluate their physical parameters and the correlation coefficients between energy, maximum values, minimum values, frequency response, the zero crossings rate, and spatiality of recordings. All the research sought to answer the question if it is possible to get an audio signal able to imitate the signal characteristics captured in real time in a recording by takes. The results showed that it is possible when the recording is performed with the method of overlay of acoustic sources (OAS).
Convention Paper 9417

3:00 pm

P10-3 Process Improvement in Audio Production from a Sociotechnical Systems Perspective—*Gerhard Roux*, Stellenbosch University, Stellenbosch, Western Cape, South Africa

Audio professionals involved in live sound reinforcement, record production, and broadcasting are continuously solving complex problems in creative ways. It is wasteful if the pragmatic methodologies used in solving these problems do not contribute towards a reusable model of process improvement. This paper suggests a systems-level engagement with audio production that strikes a balance between human creativity and technological infrastructure. A conceptual model of process improvement is developed through analysis of audio production as a complex system and subsequently implemented through an action research methodology in multiple case studies. The study found that significant quality improvements in audio production could be attained through a sociotechnical systems approach. The results imply that the application of process improvement methodologies can coexist with creative social practice, resulting in improved technical performance of production systems.
Convention Paper 9418

3:30 pm

P10-4 Listener Preference for Height Channel Microphone Polar Patterns in Three-Dimensional Recording—*Will Howie, Richard King, Matthew Boerum, David Benson, Alan Joosoo Han*, McGill University, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada

A listening experiment was conducted to determine if a preference exists among three microphone polar patterns when recording height channels for three-dimensional music production. Seven-channel 3D recordings of four different musical instruments were made using five-channel surround microphone arrays, augmented with two Sennheiser MKH 800 Twin microphones as height channels. In a double-blind listening test, subjects were asked to rate different mixes of the same recordings based on preference. The independent variable element in these mixes was the polar pattern of the height channel microphones. Analysis of the results found that the vast majority of subjects showed no statistically significant preference for any one polar pattern.
Convention Paper 9419

4:00 pm

P10-5 Listener Discrimination of High-Speed Digitization from Analog Tape Masters with Spectral Matching—*Nick Lobel, Eric Tarr, Wesley Bulla*, Belmont University, Nashville, TN, USA

This study investigated whether listeners could discriminate between real-time (RT) and double-speed (DS) digital transfers from analog tape recordings. Signals were recorded to tape at 15 inches per second (ips), then digitized at two copy rates: 15 ips (RT) and 30 ips (DS). The DS transfers were digitally time-stretched and spectrally processed to match the duration and frequency response of the RT transfers. Thirty-one listeners participated in an ABX experiment to discriminate between the RT and DS transfers. Results show discrimination between RT and DS transfers was not statistically significant. Additionally, discrimination did not vary significantly across different types of source signals.
Convention Paper 9420

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

POSTERS: SPATIAL AUDIO

2:00 pm

P11-1 Comparison of Techniques for Binaural Navigation of Higher-Order Ambisonic Soundfields—*Joseph G. Tylka, Edgar Choueiri*, Princeton University, Princeton, NJ, USA

Soundfields that have been decomposed into spherical harmonics (i.e., encoded into higher-order ambisonics—HOA) can be rendered binaurally for off-center listening positions, but doing so requires additional processing to translate the listener and necessarily leads to increased reproduction errors as the listener navigates further away from the original expansion center. Three techniques for performing this navigation (simulating HOA playback and listener movement within a virtual loudspeaker array, computing and translating along plane-waves, and re-expanding the soundfield about the listener) are compared through numerical simulations of simple incident soundfields and evaluated in terms of both overall soundfield reconstruction accuracy and predicted localization. Results show that soundfield re-expansion achieves arbitrarily low reconstruction errors (relative to the original expansion) in the vicinity of the listener, whereas errors generated by virtual-HOA and plane-wave techniques necessarily impose additional restrictions on the navigable range. Results also suggest that soundfield re-expansion is the only technique capable of accurately generating high-frequency localization cues for off-center listening positions, although the frequencies and translation distances over which this is possible are strictly limited by the original expansion order.
Convention Paper 9421

2:00 pm

P11-2 Estimation of Individual HRIRs Based on SPCA from Impulse Responses Acquired in Ordinary Sound Fields—*Shouichi Takane*, Akita Prefectural University, Yurihonjo, Akita, Japan

In this paper a method for estimation of individual Head-Related Impulse Responses (HRIRs) from impulse responses acquired in an ordinary sound field is proposed based on the Spatial Principal Components Analysis (SPCA) of the HRIRs. The average vector and the principal components matrix are assumed to be obtained by adopting the SPCA to the set of HRIRs of multiple subjects covering all directions. A part of the impulse response from sound source to an ear of a certain subject, regarded as one of his/her HRIR, is used together for estimating the weight coefficients of the principal components. Applying the method using the dataset involving the HRIRs of the multiple subjects covering all sound source directions to the estimation of the individual HRIRs showed that the acceptable estimation accuracy is obtained for the estimation of the HRIRs in an ipsilateral direction.
Convention Paper 9422

2:00 pm

P11-3 Height Perception in Ambisonic Based Binaural Decoding—*Gavin Kearney, Tony Doyle*, University of York, York, UK

This paper presents an investigation into the perception of height in Ambisonic decoding schemes for binaural reproduction. We compare the spatial resolution of first, third, and fifth order Ambisonic decoders to that of real-world monophonic sources presented in the vertical plane. Spatial preservation of the spectral cues required for rendering sources with height is investigated and cross-referenced to binaural models of the rendered systems. The results presented address the applicability of higher order Ambisonics to the rendering of sound source elevation given the high frequency distortion of pinnae cues.
Convention Paper 9423

2:00 pm

P11-4 An HRTF Database for Virtual Loudspeaker Rendering—*Gavin Kearney, Tony Doyle*, University of York, York, UK

This paper presents a database of Head Related Transfer Functions (HRTFs), collected from 20 subjects for use in virtual loudspeaker reproduction systems. The paper documents the measurement procedure and format of the HRTFs. The database accommodates Ambisonic rendering up to 5th Order and includes loudspeaker configurations derived from platonic, convex polyhedra and other spherical distributions. The datasets are also presented with matching acoustic responses to assist externalization and decode matrices for higher order Ambisonic rendering.
Convention Paper 9424

2:00 pm

P11-5 Influence of Energy Distribution on Elevation Judgments—*Taku Nagasaka, Shunsuke Nogami, Julian Villegas, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

The relative influence of spectral cues on elevation localization of virtual sources was investigated by comparing judgments of loudspeaker reproduced stimuli spatialized with three methods, two of them based on vector-based amplitude panning: 3D vector-based amplitude panning (3D-VBAP), and 2D-VBAP in conjunction with HRIR convolution; and a third method that filtered the stimuli to simulate spectral peaks and troughs naturally occurring at different angles (equalizing filters). For the last two methods a single horizontal loudspeaker array was used. Smallest absolute errors were observed for the 3D-VBAP judgments regardless of azimuth; no significant difference in the mean absolute error was found between the other two methods. However, for most presentation azimuths, the equalizing filter method yielded the least dispersed results. These results could be used for improving elevation localization in two-dimensional VBAP reproduction systems.
Convention Paper 9425

2:00 pm

P11-6 Influence of Spectral Energy Distribution on Subjective Azimuth Judgments—*Shunsuke Nogami, Taku Nagasaka, Julian Villegas, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

In this research we compare subjective judgments of azimuth obtained by three methods: Vector-Based Amplitude Panning (VBAP), VBAP mixed with binaural rendition over loudspeakers (VBAP + HRTF), and a newly proposed method based on equalizing spectral energy. In our results, significantly smaller errors were found for the stimuli treated with VBAP and HRTFs; differences between the other two treatments were not significant. Regarding spherical dispersion of the judgments, VBAP results have the greatest dispersion, whereas the dispersion on the results of the other two methods were significantly smaller, however similar between them. These results suggest that horizontal localization using VBAP methods can be improved by applying a frequency dependent panning factor a opposed to a constant scalar as commonly used.
Convention Paper 9426

2:00 pm

P11-7 Subjective Diffuseness in Layer-Based Loudspeaker Systems with Height—*Michael P. Cousins,*[1] *Filippo Maria Fazi,*[1] *Stefan Bleeck,*[1] *Frank Melchior*[2]
[1]University of Southampton, Southampton, UK
[2]BBC Research and Development, Salford, UK

Loudspeaker systems with more channels and with elevated loudspeakers are becoming more common. There is an opportunity for greater spatial impression with listeners surrounded in three dimensions. There is research showing the advantages of more loudspeakers and of 3D layouts over 2D layouts although it is not clear whether the cause of these improvements is the greater number of loudspeaker, their position, or both. In this paper two listening tests are presented that investigate the subjective diffuseness of a range of loudspeaker layouts. The first experiment was used to optimize the distribution of loudness between horizontal layers of loudspeakers to allow fair comparison between different layouts. The second experiment investigated the perceived diffuseness of a range of loudspeaker layouts chosen to critically assess parameters of layer-based loudspeaker systems as well as validate the results of the first experiment. The number of loudspeakers at head-height, the number of loudspeakers not at head-height, and the relative level between head-height and non-head-height layers were all found to be statistically significant in terms of perceived diffuseness. It was also confirmed that 3D loudspeaker layouts can have statistically greater perceived diffuseness than 2D layouts.
Convention Paper 9427

2:00 pm

P11-8 Echo Canceler for Real-Time Audio Communication with Wave Field Reconstruction—*Satoru Emura, Sachiko Kurihara*, NTT Media Intelligence Laboratories, Tokyo, Japan

For immersive sharing of a sound field between two remote sites wave field synthesis (WFS) and echo cancellation are essential. Though both technologies have been studied for more than a decade, it was not clear whether it was possible to build a real-time system for full-duplex audio communication with WFS. We show in this paper that such a system can be built.
Convention Paper 9428

Tutorial 14 **Friday, October 30**
2:00 pm – 3:30 pm **Room 1A22**

AUDIO FORENSICS: OVERVIEW 2*

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

Panelists: *Durand R. Begault, Audio Forensic Center, Charles M. Salter Associates, San* Francisco, CA, USA
Rob Maher, Montana State University, Bozeman, MT, USA
J. Keith McElveen, Wave Sciences, Charleston, SC, USA

This tutorial will feature 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, Eddy Brixen (EBB Consult) will present on auditory crime scene analysis. Rob Maher (Montana State University - Bozeman) will present on the analysis of recorded gunshots. Keith McElveen (Wavesciences, Corp.) will present on microphone applications. And 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.*

Workshop 11 **Friday, October 30**
2:00 pm – 3:00 pm **Room 1A06**

THE CHANGING AUDIO DELIVERABLES FOR BROADCAST AND MEDIA

Moderator: **Jay Yeary**, Transient Audio Labs, Windcrest, TX, USA

Panelists: *Jeff Brugger*, Turner Studios, Atlanta, GA, USA
Michael Cardillo, Creative Waves Inc., Atlanta, GA, USA
Ed Greene
Sean Richardson, Starz Entertainment, Parker, CO, USA

Television content can now be viewed almost anywhere, on screens of all sizes, and on a seemingly limitless number of devices. This multitude of viewing choices means that it is often necessary to finish content differently for each delivery platform, requiring multiple mix passes or automated processing in order to meet an expanding range of delivery specifications. This session will look at how different networks, content providers, and engineers are dealing with the hodgepodge of loudness, channel configuration, codec, and delivery requirements to provide high-quality audio for streaming, video on demand, mobile, cable, and broadcast outlets.

This event is part of the Sound for Pictures Track.

Live Sound Seminar 3 **Friday, October 30**
2:00 pm – 3:45 pm **Room 1A12**

SOUND SYSTEM DESIGN AND OPTIMIZATION

Chair: **Bob McCarthy**, Meyer Sound Labs

Panelists: *Jamie Anderson*
Andrew Keister
Dominic Sack
Nevin Steinberg

Sound system design and tuning is a multi-step process that begins long before the pink noise can be heard. What are the steps and procedures taken to ensure a successful tuning? How are clients convinced to provide the time and resources to do this vital work? Prioritizing limited resources when time is short, and what can be done ahead of time.

Live Sound Expo 4 **LSE Stage**
Friday, October 30 **2:00 pm – 2:45 pm**

THEATRICAL CONSOLE AUTOMATION

Presenters: **Jason Crystal**
Richard Ferriday
Andrew Keister
Matt Larson

Scene and snapshot storage and recall, working with time code, synchronizing with lighting and EFX—these are all among the components of modern theatrical audio production. This session examines the console automation utilized to help the show go on.

Friday, October 30 **2:00 pm** **Room 1A19**

Technical Committee Meeting on Hearing and Hearing Loss Prevention

Networked Audio 5 **Friday, October 30**
2:30 pm – 4:00 pm **Room 1A13**

DANTE CASE STUDIES

Presenters: **John C. Huntington**, NYC College of Technology, Brooklyn, NY, USA
Sam Kusnetz, Team Sound, Brooklyn, NY, USA
Joe Patten, Communications Design Associates, Carver, MA, USA

Part 1: Using an Audio Network for a Themed Attraction in an Academic Environment: Sound Designer Sam Kusnetz and Network Engineer John Huntington give an overview of the Dante network that is the backbone of the audio system for the Gravesend Inn haunted attraction at Citytech in downtown Brooklyn. Here are two learning points: • The benefits of networked audio in themed attractions • Using networked audio over managed networks.

Part 2: Cost Saving and Digital Audio Networking: The use of digitized audio networks has changed the flow of information and cost associated with it for the better. More channels of audio are available in more location with the installation cost greatly reduced. Benefits: • Savings with infrastructure such as cabling and conduit • Flexibility with routes or multiple routes/distribution of audio • Density of audio paths, 128 channels over single link via CAT6

Friday, October 30 **2:30 pm** **Room 1A20**

Standards Committee Meeting SC-02-12 Audio Applications of Networks

Project Studio Expo 5 **PSE Stage**
Friday, October 30 **2:45 pm – 4:00 pm**

THE SPECIAL SAUCE FOR MIXING A HIT RECORD

Presenter: **Fab Dupont**
Michael Brauer, Michael Brauer, New York, NY, USA

Producer Fab Dupont (Mark Ronson, Jennifer Lopez) talks with esteemed mix engineer, Michael Brauer (Coldplay, John Mayer) 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.

Workshop 12 **Friday, October 30**
3:00 pm – 5:00 pm **Room 1A21**

MIXING MEETS MASTERING: WHERE THE LINE BECOMES BLURRED

Chair: **Rob Toulson**, Anglia Ruskin University, Cambridge, UK

Panelists: *George Massenburg*, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research

in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Mandy Parnell, Black Saloon Studios, London, UK
Ronald Prent, Wisseloord Studios, Hilversum, The Netherlands
Darcy Proper, Mastering Engineer, Wisseloord Studios, Hilversum, The Netherlands
Michael Romanowski, Coast Mastering, Berkely, CA, USA; Owner Coast Recorders

This workshop will discuss the evolution of mixing and mastering, particularly with reference to scenarios when the two practices become merged into one. There are a number of arguments for and against the use of mastering techniques at the mixing stage. For example, it can be argued that mix engineers need to take a greater responsibility towards dynamics and noise cancellation. Whereas, in contrast, the use of mix bus limiting when generating draft listening copies can confuse and falsify the sign-off process. Furthermore, mastering engineers might be tempted to work from mix stems, but does that mean they are effectively mixing as well as mastering the songs? We will also explore professional and home studio practices, innovative tools and educational approaches.

Spatial Audio Demo 7 — **Friday, October 30**
3:00 pm – 6:00 pm — **New York University**

IMMERSIVE AUDIO DEMOS AND LISTENING SESSIONS AT NYU

Moderators: **Paul Geluso**
Agnieszka Roginska

The Music Technology program at NYU Steinhardt will host a multi-presentation immersive audio listening and demo session. Visitors will experience music produced specifically for 3D listening environments, binaural and transaural research and demonstrations. Listening sessions in the James L. Dolan Music Recording Studio will include recordings made by the faculty and students of NYU Music Technology and McGill University, and will be presented on a 3D loudspeaker system. Binaural and transaural demonstrations will include HRTF customization, head modeling, BACCH 3D Sound and the BACCH-HP (for headphones), binaural Ambisonics and multi-loudspeaker environments. The session will culminate with a panel discussion on immersive audio in VR for audio and education.

This event has a limited capacity. Tickets will be allocated on a first-come first-served basis. To participate, please sign up at the AES Tours Desk.

For more detailed information on Demo Sessions and Presenters see the AES Convention Web Site.

Live Sound Expo 5 — **LSE Stage**
Friday, October 30 — **3:00 pm – 3:45 pm**

NETWORKING FOR THEATER

Presenter: **Marc Brunke**

As with audio infrastructure in general, digital audio networking is permeating the theater. We examine why audio networking is finding a natural fit into theatrical applications, and discuss the details of network implementation.

Recording & Mastering 2 — **Friday, October 30**
3:15 pm – 4:45 pm — **Room 1A23/24**

MASTER CLASS WITH WARREN HUART—
RECORDING JAMES BLUNT AND THE FRAY

Moderator: **Jonathan Pines**, Rupert Neve Designs / Fingerprint Audio, Wimberly, TX, USA

Presenter: **Warren Huart**

Warren Huart, a multi-platinum producer, mixer, songwriter, and educator, engineered "The Fray," which debuted at #1 on the Billboard chart, and a range of hits for artists including James Blunt. Warren's engineering and production techniques have received accolades from Aerosmith's Joe Perry and the Fray's Isaac Slade. His web series "Produce Like a Pro" has attracted hundreds of thousands of views on YouTube. Warren will detail his techniques with live ProTools sessions and provide a look behind the scenes of alternative rock production.

Special Event
VOODOO: THE ANALOG WORLD OF RUSS ELEVADO
Friday, October 30, 3:15 pm – 4:15 pm
Room 1A06

Moderator: **Henry Weinger**

Presenter: **Russell Elevado**

Russell is well known in engineering circles for his devotion to analog recording and mixing, and for the results he's achieved as an engineer and producer with D'Angelo, The Roots, Al Green, Ledisi, Alicia Keys, Erykah Badu and more. A Grammy® winner for his work with D'Angelo, Russ will be in conversation with Harry Weinger, as they discuss his philosophy, his techniques and perhaps a few secrets.

Broadcast/Streaming Media 7 — **Friday, October 30**
3:30 pm – 5:00 pm — **Room 1A10**

AUDIO FOR ADAPTIVE STREAMING—UNDERSTANDING HLS-DASH, HTML5

Moderator: **Ray Archie**, MixLuv, New York, NY, USA; Music is My First Language, New York, NY, USA

Panelists: *Richard Doherty*, Director, Connected Technology Strategy at Dolby Laboratories, San Francisco, CA, USA
Ronny Katz, DTS, Calabasas, CA, USA
John Kean, Consultant, Washington DC, USA
Jan Nordmann, Fraunhofer USA, San Jose, CA, USA
Charles Van Winkle, Adobe Systems Incorporated, Minneapolis, MN, USA

Adaptive streaming is the process of encoding a single source stream at multiple bit rates - allowing players to switch adaptively to deliver the optimal experience to each viewer based on available bandwidth and CPU capacity. So how do we encode for this? This panel will look at HTTP Live Streaming (HLS), MPEG-DASH, Adobe Dynamic Streaming, and more!

Live Sound Seminar 4 — **Friday, October 30**
4:00 pm – 5:45 pm — **Room 1A12**

THEATRICAL MICROPHONE DRESSING

Chair: **Mary McGregor**, Freelance, Local 1, New York, NY, USA

Panelists: *John Cooper*, Local 1 IATSE
Annalee Craig

Fitting actors with wireless microphone elements and transmitters is an art form. From ensuring the actor is comfortable and the electronics are safe and secure, to getting the proper sound with minimal detrimental audio artifacts, all while maintaining the visual illusion. Two of the most widely recognized artisans in this field provide hands on demonstrations of basic technique along with some time tested "tricks of the trade."

Live Sound Expo 6 — **LSE Stage**
Friday, October 30 — **4:00 pm – 4:45 pm**

THEATRICAL SOUND DESIGN

Presenter: **Simon Matthews**

Most every theatrical production starts from scratch for its sound design,

an experimental progress developed and honed during pre-production and rehearsal. Sonic elements, textures and effects are hand-crafted throughout the process. Our presenters discuss their process working across development in DAWs and translation to the stage, including modern tools like plug-ins that can provide a time-saving predictive bridge between pre-production and a realized design.

Friday, October 30 — **4:00 pm** — **Room 1A19**

Technical Committee Meeting on Audio Forensics

Tutorial 15 — **Friday, October 30**
4:15 pm – 5:45 pm — **Room 1A22**

BRAND YOURSELF: MARKETING, COMMUNITY, AND FINDING YOUR NICHE AS AN AUDIO ENGINEER

Chair: **Margaret Luthar**, Sonovo Mastering, Stavanger, Norway

Panelists: *Erik Braund*, Braund Sound
Adam Gonsalves, Telegraph Mastering
Jesse Lewis, Jesse's Bakery
Jon Lurie, The Wild Honey Pie
Piper Payne*, Coast Mastering, Dearborn, MI, USA*

"Brand yourself" is a panel discussion on marketing and PR for engineers who are trying to further establish a regional presence, or a niche market within a larger geographical framework. You already have a market, and solid goals for your career. But how can you lock down a market for yourself? What troubles do you encounter in selling yourself to clients? How do you talk about rates? What is your role in helping establish a strong audio community in your region? The panel discussion will address these issues and more, as we navigate the often complicated world of being a "brand."

Archiving 5 — **Friday, October 30**
4:15 pm – 5:15 pm — **Room 1A13**

BITS IS BITS, RIGHT? CHECK AGAIN!

Presenter: **George Blood**, George Blood Audio/Video/Film, Philadelphia, PA, USA

In our increasingly digital world the reliability of digital transmission and storage is paramount. From CRCs to public key encryption, systems abound to establish trust in digital information. Preservationists are concerned with bit rot, authenticity and fixity. But what if the bits coming out of the analog-to-digital converter don't make it to the file? How would you know? What can you do about it? This presentation shares information on data loss gathered during a major systems upgrade, the sordid tale of exploring many common hardware and software tools, the sad reality, and a simple test users can perform to search for this problem. Now commonly called interstitial errors, we'll look at loss in both the time and amplitude domains.

Project Studio Expo 6 — **PSE Stage**
Friday, October 30 — **4:15 pm – 5:00 pm**

SONGWRITING / RECORDING ON THE FAST TRACK

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

Capture inspiration in your DAW before it goes away, thanks to these tips on how to develop an efficient, fast, and productive workflow.

Session P12 — **Friday, October 30**
4:30 pm – 5:30 pm — **Room 1A07**

GAME AUDIO

Chair: **Michael Kelly**, DTS, Inc., London, UK

4:30 pm

P12-1 **Real-Time Morphing of Impact Sounds—***Sadjad Siddiq*, Square Enix Co., Ltd., Tokyo, Japan

This paper introduces an algorithm to morph between two or more sounds, which can be used to synthesize new sounds in real-time whose features lie between the tone color, amplitude envelope, pitch, and length of the source sounds. It is used to increase variation of commonly used impact sounds in video games, but the algorithm can also be applied to other sound types like instrument sounds. Morphing of the tone color is achieved by shifting formants in the frequency spectrum of one sound toward the frequencies of the corresponding formants in the other sounds. Corresponding formants are found automatically by pairing frequency regions of equal normalized cumulative energy. Morphing of the temporal structure is achieved by aligning those frames of all sounds that have equal normalized cumulative amplitude. A link to samples is provided.
Convention Paper 9407

5:00 pm

P12-2 **Using Pure Data as a Game Audio Engine—***Leonard J. Paul*, School of Video Game Audio, Vancouver, Canada

Recent improvements in the Pure Data (Pd) library library code (libpd) and significant run-time improvements using the Heavy compiler have made Pd more viable as a free audio engine for use in video games. Open source projects are now available to help speed the process of integrating Pd into the popular Unity game and create new possibilities for the use of Pd by game studios with limited budgets and for educational purposes as well. Details on best practices on the use of Pd for audio in video games are outlined in this paper.
Convention Paper 9408

Networked Audio 6 — **Friday, October 30**
4:30 pm – 6:30 pm — **Room 1A06**

NETWORK PERFORMANCE REQUIREMENTS FOR AUDIO APPLICATIONS

Chair: **Jim Meyer**, Clair Global, Lititz, PA, USA

Panelists: *Greg Gerhiser,* NPR
Kevin Gross, AVA Networks, Boulder, CO, USA
Andreas Hildebrand, ALC NetworX GmbH, Munich, Germany
Greg Shay, The Telos Alliance, Cleveland, OH, USA
Dave Taht

Networks are now regularly used for many classes of audio applications. Some applications have higher performance requirements than others. For file-based workflow in post-production, for instance, the emphasis is on high throughput to minimize the time required to move files across the network between workstations and storage systems. For real-time applications—such as Dante, Q-LAN, and AES67—the emphasis is on latency and network stability. Older systems such as CobraNet, Ethersound, and AES50 have other specific requirements. When video and other network applications share the network with audio applications, network design considerations are potentially significantly more complex. This workshop will outline requirements for specific audio networking applications and technologies. The audience will learn about the design issues that must be considered to support these applications and technologies.

Friday, October 30 — **4:30 pm** — **Room 1A20**

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

Special Event
THE GREAT BRITISH RECORDING STUDIOS
Friday, October 30, 5:00 pm – 6:30 pm
Room 1A23/24

Moderator: **Howard Massey**, OTRW, New York, NY, USA

Panelists: *Bill Foster*
John Smith
Jules Standen
Tony Visconti

Some of the most important and influential recordings of all time were created in British studios during the 1960s and 1970s—iconic places like Abbey Road, Olympic, Trident, Decca, Pye, IBC, Advision, AIR, and Apple. This presentation will unravel the origins of the so-called "British Sound" and celebrate the people, equipment, and innovative recording techniques that came out of those hallowed halls, including rare photographs, videos, and musical examples.

Tutorial 16 **Friday, October 30**
5:15 pm – 6:15 pm **Room 1A08**

CONTROL SYSTEMS AND ELECTROACOUSTICAL CONSIDERATIONS FOR LARGE-SCALE LOUDSPEAKER ARRAYS: PAST, PRESENT, AND FUTURE

Presenter: **David Scheirman**, Audio Engineering Society, Carlsbad, CA USA

System design evolution over four decades will be highlighted in a timeline for control and electroacoustical domains. In addition to historical review of control & monitoring processes, presentation bridges the gap from control-only networks to network digital audio, noting migration paths to beam-steerable line array elements now described as network endpoint devices. Tutorial also presents various enclosure and multi-box array topologies in use, as a broad-spectrum overview of technical developments taking place since the 6th International Conference (Sound Reinforcement, Nashville, 1988) and the 13th International Conference (Computer-controlled sound systems, Dallas, 1994). Each of these landmark AES events included content foreshadowing the development of today's modern high-powered loudspeaker arrays that incorporate beam-steering technology. Recently-emerging trends will be examined, and potential future developments contemplated. Of potential interest to sound reinforcement technicians and system operators, installed-system designers, rental sound service company providers, and live-sound equipment product development engineers.

Broadcast/Streaming Media 8 **Friday, October 30**
5:15 pm – 6:45 pm **Room 1A10**

MIXING FOR TELEMEDIA IN 21ST

Moderator: **Ed Greene**

Presenter: **Bob Katz**, Digital Domain

A frank discussion from the mixers chair of the present challenges in crafting program audio for broadcast and the www. There will be a discussion of realistic monitoring situations important for mixers who work in different CR's. The presentation will also include a review of the circumstances leading us to where we are today and the possible effect of pending new broadcast guidelines from ATSC 3.0 and pending recommended streaming practices for the www. from the AES.

Project Studio Expo 7 **PSE Stage**
Friday, October 30 **5:15 pm – 6:00 pm**

PERSONAL NETWORKING FOR THE AUDIO PROFESSIONAL

Moderator: **Joe Carroll**, Manhattan Producers Alliance, New York, NY, USA

Panelists: *Harold Chambers*, Principal Recording Engineer, Pittsburgh Symphony
John Kiehl, Manhattan Producers Alliance, New York, NY, USA; Soundtrack Studios
Mike Sayre, Independent Film Composer

"It's not what you know, it's who you know" they say. We say, it's both, because however good you are at what you do, you can always smooth the path to success with a bit of contact cultivation and networking.

Session EB4 **Friday, October 31**
5:30 pm – 6:45 pm **Room 1A07**

LISTENING, HEARING, & PRODUCTION

Chair: **Bruno Fazenda**, University of Salford, Salford, Greater Manchester, UK

5:30 pm

EB4-1 Why Do My Ears Hurt after a Show (And What Can I Do to Prevent It)—*Dennis Rauschmayer*, REVx Technologies/REV33

In this brief we review the traditional methods of preventing ear fatigue, short-term ear damage, and long term ear damage. A new method to prevent ear fatigue, focused on performing musicians is then presented. This method, which reduces noise and distortion in the artist's mix, is discussed. Qualitative and quantitative results from a series of trials and experiments is presented. Qualitative results from artist feedback indicate less ear fatigue, less ringing in the ears, and a better ability to have normal conversations after a performance when noise and distortion in their mix is reduced. Quantitative results are consistent with the qualitative results and show a reduction in the change in otoacoustic emissions measured for a set of musicians when noise and distortion are reduced. The result of the study suggests that there is an important new tool for musicians to use to combat ear fatigue and short term hearing loss.
Engineering Brief 213

5:45 pm

EB4-2 Classical Recording with Custom Equipment in South Brazil—*Marcelo Johann,*[1] *Andrei Yefinczuk,*[1] *Marcio Chiaramonte,*[2] *Hique Gomez*
[1]UFRGS, Porto Alegre, RS, Brazil
[2]Meber Metais, Bento Gonçalves, Brazil

This paper describes the process developed by Marcello Sfoggia for recording acoustic and classical music in south of Brazil, making intensive use of custom equipment. Sfoggia spent most of his lifetime building dedicated circuits to optimize sound reproduction and recording. He took the task of registering major performances in the city of Porto Alegre, using his home-developed equipment, what became a reference process. We describe the system employed for both sound capture and mixdown. Key components of the signal flow include preamplifiers with precision op-amps, short signal paths, modified A/D/A converters and the mixing desk with pure vacuum tube circuitry. Finally, we address our current efforts to continue his activities and improve upon his system with updated circuits and techniques.
Engineering Brief 214

6:00 pm

EB4-3 Techniques For Mixing Sample-Based Music—*Paul "Willie Green" Womack*, Willie Green Music, Brooklyn, NY, USA

Samples are a great way to add impact, vibe, and texture to a song and can often be the primary component of a new work. From a production standpoint, audio that is already mixed and mastered can add to a producer's sonic palette. From a mixing perspective, however, these same bonuses also provide a number of challenges. Looking more closely at each of the common issues an engineer often faces with sample-based music, I will illustrate techniques that can enable an engineer to better manipulate a sample, allowing it to sit more naturally inside the mix as a whole.
Engineering Brief 215

6:15 pm

EB4-4 Case Studies of Inflatable Low- and Mid-Frequency Sound Absorption Technology—*Niels Adelman-Larsen*, Flex Acoustics, Copenhagen, Denmark

Surveys among professional musicians and sound engineers reveal that a long reverberation time at low frequencies in halls during concerts of reinforced music is a common cause for an unacceptable sounding event. Mid- and high-frequency sound is seldom a reason for lack of clarity and definition due to a 6 times higher absorption by audience compared to low frequencies, and a higher directivity of speakers at these frequencies. Lower frequency sounds are, within the genre of popular music however, rhythmically very active and loud, and a long reverberation leads to a situation where the various notes and sounds cannot be clearly distinguished. This reverberant bass sound rumble often partially masks even the direct higher pitched sounds. A new technology of inflated, thin plastic membranes seems to solve this challenge of needed low-frequency control. It is equally suitable for multipurpose halls that need to adjust their acoustics by the push of a button and for halls and arenas that only occasionally present amplified music and need to be treated just for the event. This paper presents the authors' research as well as the technology showing applications in dissimilarly sized venues, including before and after measurements of reverberation time versus frequency.
Engineering Brief 216

6:30 pm

EB4-5 Advanced Technical Ear Training: Development of an Innovative Set of Exercises for Audio Engineers—*Denis Martin, George Massenburg*, Schulich School of Music, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada

There are currently many automated software solutions to tackle the issue of timbral/EQ training for audio engineers but only limited offerings for developing other skills needed in the production process. We have developed and implemented a set of matching exercises in Pro Tools that fill this need. Presented with a reference track, users are trained in matching inter-instrument levels/gain, lead instrument volume automation, instrument spatial positioning/panning, reverberation level, and compression settings on a lead element within a full mix. The goal of these exercises is to refine the listener's degree of perception along these production parameters and to train the listener to associate these perceived variations to objective parameters they can control. We also discuss possible future directions for exercises.
Engineering Brief 217

Friday, October 30 **5:00 pm** **Room 1A19**

Technical Committee Meeting on Audio for Telecommunications

Workshop 13 **Friday, October 30**
5:30 pm – 7:00 pm **Room 1A14**

AUTOMOTIVE AUDIO—THE MAKING OF THE SOUND SYSTEM IN A CAR

Chair: **Grzegorz Sikora**, Harman, Pullach, Germany

Panelists: *Thomas Bachmann*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Peter John Chapman, Harman, Denmark

In this workshop we look behind the scenes of developing an OEM sound system, "the making of." It's a unique opportunity to understand the key fundamentals of the Automotive Audio perspective, from the project team behind many successful OEM sound systems. This workshop will cover a range of topics from the initial idea to the finished product. The experts from the respective fields will discuss car cabin acoustics, loudspeaker selection and placement, signal flow and amplifier characteristics, sound tuning, audio design philosophy, and creation of the 3D sound algorithm. Each topic will be discussed in general and in the context of actual projects.

Workshop 14 **Friday, October 30**
5:30 pm – 7:00 pm **Room 1A13**

INTELLECTUAL PROPERTY BASICS FOR AUDIO PROFESSIONALS

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

Panelists: *TBA*

Workshop on copyright, trademark, patent, trade secrets, and other IP topics.

Stedent Event & Career Development
RECORDING COMPETITION—PART 1
Friday, October 30, 5:30 pm – 7:30 pm
Room 1A21

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.

5:30 pm: Category 2– Traditional Studio Recording
Judges: Richard King, Sean McLauglin, Brandie Lane, Frank Filipetti

6:30 pm: Category 3—Modern Studio Recording & Electronic Music
Judges: Jonathan Wyner, Piper Payne, Mandy Parnell, William Crabtree

Workshop 15 **Friday, October 30**
6:00 pm – 7:00 pm **Room 1A22**

AUDIO ENHANCEMENT: HOW TO IMPROVE THE EFFECTIVENESS OF CROSS-CHANNEL FILTERS BY USING TIME COMPRESSION / EXPANSION TO MAINTAIN SIGNAL ALIGNMENT*

Chair: **Daniel Rappaport**

Panelists: *Eddy B. Brixen*
Gordon Reid

A common forensic audio enhancement request is to attenuate music or a television/radio broadcast from a recording to unmask underlying speech. Traditional equalization and expansion techniques affect the desired target signal as well as the unwanted noise. When a clean reference recording is available, cross-channel adaptive and lattice filters or dynamic spectral subtraction may be used to cancel out the offending signal. Various tools and filters are available for this type of processing, but they all rely on precise alignment of the reference signal to the recording being enhanced. Even with the aid of auto-alignment tools, word clock drift negatively impacts the effectiveness of the filter. This workshop will present a technique to calculate the linear offset and apply time compression/expansion to the reference track, which increases the effectiveness of the filter.

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

Archiving 6 **Friday, October 30**
6:00 pm – 7:00 pm **Room 1A12**

ANATOMY OF A BOOTLEG: ARCHIVING ONE PART POETRY, ONE PART PUNK

Presenter: **Allie Whalen**, NYU MIAP Grad

Just as finding the right speed to play back a record, or finessing the sound from a wax cylinder can be challenging, recordings on magnetic media such as videotape or audio-cassette bring their own unique challenges. These challenges are especially complex when dealing with the homemade and DIY nature of punk collections and are well represented by recordings found in UCLA Library's S.A. Griffin Collection, currently being digitized by UCLA Library's AV Preservation Unit. Griffin is an LA-based poet and actor who was closely involved in LA's neo- beat poetry, punk and performance art scene during the 1980s and 1990s. Content includes recordings of poetry readings and various performances held at iconic, now vanished, LA locations. Performers and groups represent a cross section of the LA poetry, performance, and punk scene including the Lost Tribe and their later incarnation the Carma Bums, Charles Bukowski, Wanda Coleman, and William Burroughs reading Naked Lunch. The non-standard and "rough" nature of the recordings mean dealing with problems like mono/stereo switching mid-recording, inconsistent levels, analog noise (incorrect speed, clicks, pops, dropout, distortion, crackle, buzz) and post-digitization clicks and clipping, all of which require thoughtful assessment. Along these lines, the Griffin project also highlights the importance of thorough documentation and metadata when it comes to preserving such a unique niche collection, as well as the value of being in communication with the donor to provide additional background and context on everything from contributors to recording locations.

Special Event
FROM THE ETHER: A DISTRIBUTED PERFORMANCE CONCERT AND PANEL DISCUSSION
Friday, October 30, 7:00 pm – 9:00 pm
Frederick Loewe Theatre at NYU

Producers: **Tim Shuttleworth**
Michael Palumbo
Tom Beyer

A special joint presentation by members of the AES Technical Committee on Network Audio Standards, in the Frederick Loewe Theater, at the Music and Performing Arts Professions Department's Music Technology Program at New York University, and the Distributed Performance and Sensorial Immersion Lab at York University, Toronto. "From The Ether" is an international concert with performers located in New York, Toronto, Montreal, Stanford, CA; Tromsø, Norway; Buenos Aries, Argentina, and Belfast, Ireland. Audio and video will be transmitted between all locations over high-bandwidth internet links. The AES audience in New York will hear the performers from all locations and be treated to a multiple-screen projection of the concert: an audiovisual unification of spectators and performers. At each remote location, all global musicians' contributions will be mixed locally together—each location will experience a complete orchestra.

In order to inform the audience as to how the underlying technology is being utilized, how they can use it, and how it serves the music, the producers have arranged for brief talks to precede the performances, each addressing a particular aspect of distributed performance, including: compositional considerations, equipment, techniques, and production affordances and constraints.

Session P13 **Saturday, Oct. 31**
9:00 am – 12:30 pm **Room 1A08**

SPATIAL AUDIO—PART 1

Chair: **Francis Rumsey**, Logophon Ltd. ,Oxfordshire, UK

9:00 am

P13-1 On the Performance of Acoustic Intensity-Based Source Localization with an Open Spherical Microphone Array—*Mert Burkay Côteli,*[1,2] *Hüseyin Hacihabiboglu*[1]
[1]Middle East Technical University (METU), Ankara, Turkey
[2]ASELSAN A.S., Ankara, Turkey

Sound source localization is important in a variety of contexts. A notable example is acoustic scene analysis for parametric spatial audio where not only recording the sound source but also deducing its direction is necessary. Sound source localization methods based on acoustic intensity provide a viable alternative to more traditional, delay-based techniques. However, special sound intensity probes or microphone arrays need to be used. This paper presents the evaluation of the sound source localization performance of an icosahedral open spherical microphone array using a method based on intensity vector distributions in time-frequency domain.
Convention Paper 9429

9:30 am

P13-2 A Microphone Array for Recording Music in Surround-Sound with Height Channels—*David Bowles*, Swineshead Productions LLC, Berkeley, CA, USA

In the past few years, sound recordings with spatial audio have moved from the realm of theoretical research to the actuality of physical and digital releases in the market. At present three Blu-ray disc formats utilize a traditional 5.1 surround-sound recording, with an added 4-channel layer of height channels. The topic of this paper is how to capture vertical localization effectively within this release format, utilizing existing research on hearing localization and techniques learned in the field. The proposed microphone array has time-of-arrival differences between all microphones, yet mixes down to 5.1 and stereo without excessive comb-filtering or other artifacts.
Convention Paper 9430

10:00 am

P13-3 Exploring 3D: A Subjective Evaluation of Surround Microphone Arrays Catered for Auro-3D Reproduction Systems—*Alex Ryaboy*, New York University, New York, NY, USA

As multichannel systems grow in popularity, audio professionals must make an informed decision when choosing a correct capturing method to deliver their vision. Many of today's microphone arrays that are catered for surround sound with height, employ traditional spaced surround techniques that are aided by an additional array in the upper plane and are widely used to capture a performance in large spaces. This paper uses a perceptual study to evaluate a fully coincident microphone array Double-MSZ and a semi-coincident array Twins Square on Envelopment, Localization and Spatial Impression in a small recording studio environment. The study revealed overall lower widths, better localization, and a more stable vertical imaging for Double-MSZ, while the Twins Square technique exhibited higher ensemble envelopment and a more spacious perceived environment.
Convention Paper 9431

10:30 am

P13-4 Three Dimensional Spatial Techniques in 22.2 Multichannel Surround Sound for Popular Music Mixing—*Bryan Martin, Richard King*, McGill University, Montreal, Quebec, Canada;
The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

Current multichannel spatial mixing practices are largely limited to the construction of three-dimensional space using two dimensional panning tools (meant for 5.1, 7.1, etc.) and those designed for common stereo production. A great deal of research is currently underway in spatial sound reproduction through computer modeling and signal processing, with little focus on actual recording and mixing practices. This investigation examines the design and implementation of early and late reflections and reverberant fields in 22.2 multichannel sound system mixing based upon research in listener envelopment. The techniques discussed will include the expansion of spatial elements into three dimensions using conventional tools and the implementation of multichannel impulse responses for reverberant fields. Listening tests were conducted upon the final music mix with positive results reported for listener immersion.
Convention Paper 9432

11:00 am

P13-5 On the Use of a Lebedev Grid for Ambisonics—*Pierre Lecomte,*[1,2,3] *Philippe-Aubert Gauthier,*[2,3] *Christophe Langrenne,*[1] *Alexandre Garcia,*[1] *Alain Berry*[2,3]
[1]Conservatoire National des Arts et Métiers, Paris, France
[2]Université de Sherbrooke, Sherbrooke, Quebec, Canada
[3]McGill University, Montreal, Quebec, Canada

Ambisonics provide tools for three-dimensional sound field analysis and synthesis. The theory is based on sound field decomposition using a truncated basis of spherical harmonics. For the three-dimensional problem the decomposition of the sound field as well as the synthesis imply an integration over the sphere that respects the orthonormality of the spherical harmonics. This integration is practically achieved with discrete angular samples over the sphere. This paper investigates spherical sampling using a Lebedev grid for practical applications of Ambisonics. The paper presents underlying theory, simulations of reconstructed sound fields, and examples of actual prototypes using a 50 nodes grid able to perform recording and reconstruction up to order 5. Orthonormality errors are provided up to sixth order and compared for two grids: (1) the Lebedev grid with 50 nodes and (2) the Pentakis-Dodecahedron with 32 nodes. Finally, the paper presents some practical advantages using Lebedev grids for Ambisonics, in particular the use of sub-grids working up to order 1 or 3 and sharing common nodes with the 50 nodes grid.
Convention Paper 9433

11:30 am

P13-6 ISO/MPEG-H 3D Audio: SAOC 3D Decoding and Rendering—*Adrian Murtaza,*[1] *Jürgen Herre,*[2] *Jouni Paulus,*[2] *Leon Terentiv,*[1] *Harald Fuchs,*[1] *Sascha Disch*[2]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories Erlangen, Erlangen, Germany

The ISO/MPEG standardization group recently finalized the MPEG-H 3D Audio standard for the universal carriage of encoded 3D-sound from channel-based, object-based, and HOA-based input. To achieve efficient low-bitrate coding of a high number of channels and objects, an advanced version of the well-known MPEG-D Spatial Audio Object Coding (SAOC) has been developed under the name SAOC 3D. The new SAOC 3D system supports direct reproduction to any output format from 22.2 and beyond down to 5.1 and stereo. This paper describes the SAOC-3D technology as it is part of the MPEG-H 3D Audio (phase one) International Standard and provides an overview of its features, capabilities, and performance.
Convention Paper 9434

12:00 noon

P13-7 Auditory Distance Rendering Using a Standard 5.1 Loudspeaker Layout—*Mikko-Ville Laitinen,*[1] *Andreas Walther,*[2] *Jan Plogsties,*[2] *Ville Pulkki*[1]
[1]Aalto University, Espoo, Finland
[2]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Human hearing is known to be sensitive to the distances of sound sources. However, spatial-sound rendering systems typically do not allow controlling the distance of the auditory objects. This paper proposes a distance-rendering method that uses standard 5.1 loudspeaker layouts. The proposed method applies an input signal to multiple loudspeakers and controls the gains and the coherence of the loudspeaker signals. In addition, the method is combined with amplitude panning, thus allowing to continuously control both the distance and the direction of the auditory objects. Based on listening tests, the proposed method was found to provide the ability to realistically manipulate the perception of both direction and distance.
Convention Paper 9435

Session P14 **Saturday, October 31**
9:00 am – 12:00 noon **Room 1A07**

PERCEPTION—PART 3

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

9:00 am

P14-1 Spatial Sound Attributes—Development of a Common Lexicon—*Nick Zacharov, Torben Holm Pedersen*, DELTA SenseLab, Hørsholm, Denmark

Sound quality and spatial sound have been topics of research for decades in relation to loudspeakers and headphones as well as performance spaces (e.g., concert halls). Attributes may be used as a means to characterize sound quality through listening tests. Attribute development protocols are well reported and have been employed to a wide range of spatial sound applications. However, the usage of attribute often leads to researchers discussing the merits of the attributes as opposed to focusing upon the object of the research. Over the last few decades a large number of articles have included the development of a spatial sound attribute. This paper describes the collection of many known research articles on spatial sound attributes from a wide range of domains. As opposed to repeating traditional word elicitation and group discussion, we have chosen to use a semantic text data mining approach to find common attribute meanings, which is then followed by a sorting and refinement process with expert assessors. This process is defined in detail and the results of the semantic text mining are presented as part of the further development of a sound wheel for sound reproduction.
Convention Paper 9436

9:30 am

P14-2 Towards a MATLAB Toolbox for Imposing Speech Signal Impairments Following the P.TCA Schema—*Friedemann Köster, Falk Schiffner, Dennis Guse, Jens Ahrens, Janto Skowronek, Sebastian Möller*, University of Technology Berlin, Berlin, Germany

In this paper we present and validate a freely available MATLAB Toolbox for imposing speech signal impairments similar to those occurring in real-world telecommunication systems. The purpose of the toolbox is to facilitate research on the perception of different dimensions of speech quality and their relation to technical system properties. In that context the International Telecommu-

nication Union (ITU) is working on the annotation method P.TCA, which enables expert listeners to identify the technical cause for an observed speech signal impairment. Our contribution addresses one current challenge of P.TCA: it was found out that providing written definitions of speech degradations without exemplary listening material is not sufficient to be reliably understood by annotators. To address this issue and make the schema accessible for a wide range of users, this paper describes a systematic approach to generate and validate such exemplary listening material. A validation experiment shows that experts can identify more than half of the processed examples correctly and it encourages further research towards improving the P.TCA procedure as well as the processing algorithms.
Convention Paper 9437

10:00 am

P14-3 The Influence of Dumping Bias on Timbral Clarity Ratings—
Kirsten Hermes, Tim Brookes, Christopher Hummersone, University of Surrey, Guildford, Surrey, UK

When listening test subjects are required to rate changes in a single attribute, but also hear changes in other attributes, their ratings can become skewed by "dumping bias." To assess the influence of dumping bias on timbral "clarity" ratings, listeners were asked to rate stimuli: (i) in terms of clarity only; and (ii) in terms of clarity, warmth, fullness, and brightness. Clarity ratings of type (i) showed (up to 20%) larger interquartile ranges than those of type (ii). It is concluded that in single-attribute timbral rating experiments, statistical noise—potentially resulting from dumping bias—can be reduced by allowing listeners to rate additional attributes either simultaneously or beforehand.
Convention Paper 9438

10:30 am

P14-4 Method for Objective Evaluation of Nonlinear Distortion—
Mikhail Pahomov,[1] *Yong Kyuk Na*[2]
[1]LG Electronics, Inc., St. Petersburg, Russia
[2]LG Electronics, Inc., Seoul, South Korea

A perceptual method is presented for assessing nonlinear distortion audibility in sound systems with high distortion levels of the original signals (mobile devices). The method is based on the Perceptual Evaluation of Audio Quality (PEAQ) standard [1]. To estimate the audibility of non-linear distortion, generating a content-dependent multitone signal and extracting a distortion signal from it is proposed. Then, the distortion signal's properties are measured. Next, a regression analysis is applied to combine the properties to derive a metric that denotes the overall audible harmonic distortion. Experimental results on mobile handsets are provided to verify the high accuracy of the method.
Convention Paper 9439

11:00 am

P14-5 Subjective and Objective Measurements of Speech Loudness in Hands-Free Telephony—Toward an Extended Loudness Model for Telephonometry—*Idir Edjekouane,*[1,2] *Cyril Plapous,*[1] *Catherine Quinquis,*[1] *Sabine Meunier*[2]
[1]Orange Labs, Lannion Cedex, France
[2]Aix-Marseille Université, Marseille Cedex, France

The loudness rating technique is widely used in telephony. This technique shows some limitations with the recent advances in telecommunications. This paper proposes a new alternative for the loudness rating technique using an extension of Zwicker's loudness model. We first investigated the loudness of speech transmitted via a telephone system and the ability of Zwicker's model to predict the perceived loudness. The model predicts the main trends observed in perceptual data. However, a bias exists between the prediction and the measure that depends on sound pressure level. Based on our perceptual data and on recent studies, we proposed a modification of the model at the specific loudness calculation stage. This modification brought a significant improvement on the predictions.
Convention Paper 9440

11:30 am

P14-6 Investigation on the Phantom Image Elevation Effect—
Hyunkook Lee, University of Huddersfield, Huddersfield, UK

Listening tests have been carried out in order to evaluate the phantom image elevation effect depending on horizontal stereophonic base angle. Seven ecologically valid sound sources as well as four noise sources were tested. Subjects judged the perceived image positions of phantom center image created with seven loudspeaker base angles. Results generally showed that perceived images were elevated from front to above as the loudspeaker base angle increased up to around 180°. This tendency depended on the spectral characteristics of sound source. The perceived results are explained from both physical and cognitive points of view.
Convention Paper 9441

Tutorial 17 **Saturday, October 31**
9:00 am – 10:30 am **Room 1A06**

COUNTER CLOCKWISE—THE AESTHETIC WONDER AND TECHNICAL MERIT OF SOUNDS PLAYED BACKWARDS

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

We can learn from pop music's many examples of artistic expression through the presentation of sounds While such sounds can't happen in nature, they are a common studio creation. Alex U. Case demonstrates and discusses their strong musical value, and intriguing technical and perceptual advantages.

Workshop 16 **Saturday, October 31**
9:00 am – 10:30 am **Room 1A21**

RECORDING THE MODERN BIG BAND

Chair: **Bradford Swanson**, University of Massachusetts – Lowell, Lowell, MA, USA

Moderator: **Ryan Hewitt**, Nice Rack Audio Services, Venice, CA, USA

Panelists: *Jim Anderson*, New York University, New York, NY, USA
Bob Dawson, Bias Studios, Springfield, VA, USA
James Farber
Leslie Ann Jones, Recording Engineer and Producer, Director of Music Recording and Scoring, Skywalker Sound, San Rafael, CA, USA
Al Schmitt, Los Angeles, CA, USA

The Jazz Orchestra and it's repertoire has evolved significantly over the past 75 years. How are modern engineers recording the next great masters of the genre?

Archiving 7 **Saturday, October 31**
9:00 am – 10:30 am **Room 1A22**

SPECIAL CHALLENGE METADATA: MULTIMEDIA-BASED PRESERVATION OF THE COLLECTION OSKAR SALA OR … HOW TO SAFEGUARD HITCHCOCK'S *BIRDS*

Presenter: **Nadja Wallaszkovits**, Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria

Oskar Sala (1910 – 2002) was a German musician, scientist, and a pioneer of electronic music. He played and further developed the trautonium, a predecessor of the synthesizer. By enhancing and modifying this instrument Sala was able to create totally new sounds and effects. 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, among others, a collection of about 1200 analog magnetic audio tapes which are stored in the archives of Deutsches Museum in Munich. Oskar Sala fully exploited all the possibilities of the analog tape technology, using impressive experimental approaches. His tapes have become artworks themselves, as they comprise a unique richness of very special and specific metadata: most of the tapes are cut up to 200 times per reel, and he used them as a (more or less readable) notebook. Such and many more surprises made the adequate safeguarding and digitization of the collection a unique undertaking. This 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.

Broadcast/Streaming Media 9 **Saturday, October 31**
9:00 am – 10:30 am **Room 1A10**

AUDIO FOR OVER THE TOP TELEVISION (OTT)

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

Panelists: *Richard Galvan*, Senior Technical Marketing Manager for OTT, Dolby Labs
Tom McCarthy, Owner, AM-DVD
Sean Richardson, Starz Entertainment, Parker, CO, USA

Like many other industries, television has been disrupted by the Internet, with a growing amount of content delivered to audiences via streaming. The mechanisms involved are substantially different from traditional television delivery, yet the two forms will continue to coexist for some time. Meanwhile, OTT has greater agility to adopt new formats than traditional delivery schemes, and therefore content suppliers are being asked to deliver new components to OTT providers at a rapid pace. This session will address how audio is changing for OTT delivery, presented from the perspectives of technology providers, content producers and service operators.

Game Audio 6 **Saturday, October 31**
9:00 am – 9:45 am **Room 1A14**

3D AUDIO FOR VIRTUAL REALITY

Presenter: **Frederick Umminger**, Sony Computer Entertainment America

Overview of some of the engineering problems and challenges in developing a 3D audio solution suitable for widespread use in virtual reality and ordinary gaming and some of the impacts on mixing workflows.

Live Sound Seminar 5 **Saturday, October 31**
9:00 am – 10:45 am **Room 1A12**

WIRELESS MATTERS, PART I: THEORY AND PRACTICAL APPLICATIONS

Moderator: **James Stoffo**, Radio Active Designs, Key West, FL, USA

Panelists: *TBA*

The average production today is utilizing more and more channels of wireless microphones, in-ear monitors, intercom, and interruptible foldback (IFB). This panel will discuss how to outline the strategy, tactics, and practices of implementing multiple wireless systems in RF challenging environments, from the first phone call for the production through the event itself.
The session will be immediately followed by a session on the upcoming spectrum changes.

Product Development 7 **Saturday, October 31**
9:00 am – 10:30 am **Room 1A13**

MODERN DIGITAL PROCESSING OF MICROPHONE SIGNALS

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

Microphones have been in use for decades in professional audio applications. Recently they are also being incorporated into consumer and automotive products and their use is exploding. And although they are ubiquitous they are usually the weakest link in the audio signal chain. Common problems include dynamic range issues (too loud or too soft) and noise (electrical noise, background noise, wind, and plosives and sybillants). This session covers modern digital approaches to microphone processing. We use an interactive approach and build up the signal chain using graphical tools. We design single and multiband automatic gain controls, noise gates, and dynamics processors for reducing plosives and handling noise ("de-poppers"). We show how these algorithms are designed and tuned in practice.

Student Event & Career Development
STUDENT DESIGN COMPETITION
Saturday, October 31, 9:00 am – 10:30 am
Foyer

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.
Judges: Jay LeBoeuf, Charlie DeVane, Scott Dorsey, Doug Fearn, Dave Hill, Steve Green

Student Event & Career Development
STUDENT RECORDING CRITIQUES
Saturday, October 31, 9:00 am – 10:00 am
Room 1A18

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

Students! Bring your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on memory-stick, or hard disk, as clearly labeled 24/44.1 KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers!

Saturday, October 31 **9:00 am** **Room 1A19**

Technical Committee Meeting on Microphones and Applications

Saturday, October 31 **9:00 am** **Room 1A20**

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

Special Event
PLATINUM MASTERING
Saturday, October 31, 9:15 am – 10:45 am
Room 1A23/24

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

Panelists: *Adam Ayan*, Gateway Mastering Studios, Portland, ME USA
Tom Coyne, Sterling Sound, New York, NY, USA
Stephen Marcussen, Marcussen Mastering, Hollywood, CA, USA
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA

A platinum panel of renowned mastering engineers will discuss the creative process behind some of the most famous albums in the world. We will speak about the creative elements of the mastering process that made these recordings so special and perhaps tell some stories about them as well. We will be playing examples and taking questions from the audience.

Workshop 17 **Saturday, October 31**
10:00 am – 11:30 am **Room 1A14**

IMMERSIVE AUDIO SIGNAL PROCESSING AND EFFECTS*

Chair: **Christof Faller**, Illusonic GmbH, Zurich, Switzerland; EPFL, Lausanne, Switzerland

Panelists: *Jean-Marc Jot*, DTS, Inc., Los Gatos, CA, USA
Itai Neoran, Waves Audio Ltd., Tel Aviv, Israel
Sebastian Schlecht, International Audio Labs, Erlangen, Germany
Andreas Silzle, Fraunhofer Institute forIntegrated Circuits IIS, Erlangen, Germany
Nicolas Tsingos, Dolby Labs, San Francisco, CA, USA

Multichannel surround in 3D and immersive audio open new possibilities for rendering of sound objects and acoustic environments. Experienced experts in the field discuss about audio signal processing and effects for immersive audio, such as rendering of objects in 3D, generation of height channels, reverberators for immersive audio, etc.

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

Saturday, October 31 **10:00 am** **Room 1A19**

Technical Committee Meeting on Archiving, Restoration, and Digital Libraries

Project Studio Expo 8 **PSE Stage**
Saturday, October 31 **10:30 am – 12:00 noon**

THE PROJECT STUDIO IN THE COMMERCIAL WORLD

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

Panelists: *Christian Cooley*, Leitvox Studios, Miami, FL, USA
Sergio Molho, Walters Storyk Design Group, Highland, NY, USA; Dream Asylum Studios, Miami, FL, USA
Alex Santilli, Spice House Studios, Philadelphia, PA, USA
David Shinn, National Audio Theatre Festivals, New York, NY, USA
Carl Tatz, Carl Tatz Design, Nashville, TN, USA

Most people think of project studios as a "personal workplace," with little or no regard for how these environments and systems will deal with real world commercial pressures. Over the years the Project Studio has come to mean many things to the studio world, often defined by terms such as budget, size, commercial status, personality, residential setting, etc. In fact, the lines of recording studio status that divide the commercial and non-commercial studio world; the lines that differentiate between large and small and that define residential and non-residential environments have been blurred for quite some time. This panel will explore this new frontier by presenting four "Project Studios" – each of which vary dramatically in size budget, acoustic solution and purpose. The panelists (either owners or designers) will describe the studio's individual goals, strengths, design / installation tips and significant issues encountered during the design/construction process. Most importantly, they will also reveal the tale of the studio's success or failure after opening.

Workshop 18 **Saturday, October 31**
10:45 am – 12:15 pm **Room 1A06**

WHAT HAPPENED TO MY MASTER? HIDDEN SIDE EFFECTS OF COMMON SIGNAL PROCESSING THAT CAN RUIN YOUR AWESOME SOUND

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

Panelists: *George Massenburg*, McGill University, Montreal, Quebec, Canada
Aaron Wishnick, iZotope Inc, Cambridge, MA, USA

Some common types of signal processing carry with them significant artifacts and if we don't know to look for them we might miss the problems they create. Signal processing tools addressed will be: brickwall limiters, multiband compressors, linear phase filters, compressor time constants, pass band filters.

Broadcast/Streaming Media 10 **Saturday, October 31**
10:45 am – 12:15 pm **Room 1A10**

LISTENER FATIGUE AND RETENTION

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

Panelists: *Rob Arbittier*
Robert Reams, Consultant, Santa Clara, CA, USA
Bill Sacks, Orban / Optimod Refurbishing, Hollywood, MD, USA
Andrew Scheps, Tonequake Records, Van Nuys, CA, USA
Larry Zinn, Monitor Engineer

Listener fatigue is present in every audio professional's and music consumer's life. It affects the quality of recorded, live, and broadcast audio. It affects the ability of listeners to enjoy and/or understand the program. And, ultimately, can impact long term health.

The session will seek to define the psychological and physiological aspects of listener fatigue, the causes and possible solutions.

The panel is made up of top audio professionals with varied backgrounds and decades of experience.

Live Sound Seminar 6 **Saturday, October 31**
10:45 am – 12:00 noon **Room 1A12**

WIRELESS MATTERS, PART II: THE COMING SPECTRUM CHANGES AND THE NEW PARADIGM

Presenter: **James Stoffo**, Radio Active Designs, Key West, FL, USA

The 600 MHz spectrum auction is tentatively scheduled for first quarter 2016. This session will discuss how much spectrum is likely to be taken away and where, how the new FCC rules are shaping up, and what the future holds for new spectrum bands for wireless microphone operations and new operational practices.

Product Development 8 **Saturday, October 31**
10:45 am – 12:15 pm **Room 1A13**

ADAPTIVE LOUDSPEAKER CONTROL—AN APPLICATION TUTORIAL

Presenter: **Gregor Hoehne**, Klippel GmbH, Dresden, Germany

Adaptive control provides new potential for speaker design. Digital preprocessing of the electrical input signal can used to equalize, linearize, stabilize, and actively protect the transducer against overload. The control technology allows the exploitation of all hardware resources, which makes the transducer smaller, lighter, and more cost effective. This workshop gives an insight into adaptive loudspeaker control by presenting the Klippel Control Technology and demonstrates how it can be utilized by transducer and system engineers. An introduction into the theory of adaptive nonlinear control is given, but the focus lies on how the technique can be integrated into a system design. This comprises the selection of suitable transducers, the derivation of protection limits and the adjustment of the system alignment. A strong emphasis lies on the evaluation of an adaptive controlled loudspeaker system. Various methods for evaluating the transducer protection system, the reduction of nonlinear distortion and the compensation of ageing-effects are discussed and demonstrated.

Special Event
YOUR CREDITS, YOUR MONEY, THE NEW DATA STANDARDS AND DDEX —WHAT YOU NEED TO KNOW!
Saturday, October 31, 10:45 am – 11:45 am
Room 1A22

Moderator: **Paul Jessop**, County Analytics Ltd., Dunstable, Bedfordshire, UK

Panelists: *Jonathan Bender*, SoundExchange
Maureen Droney, The Recording Academy, Los Angeles, CA, USA
Niels Rump, Digital Data Exchange (DDEX), Global
John Spencer, BMS Chace LLC, Nashville, TN, USA

In the digital world, capturing data in the studio is more important than ever before. This includes your credits, contributor names, technical information, and all versions of recording titles. Without this data, payments can get missed, at home and abroad! The challenges to collecting this data and ensuring it flows throughout the whole supply chain are being addressed by a consortium of media companies working with DDEX (www.ddex.net), an international organization standardizing the music supply chain through the creation of common formats for information communication and the support of metadata improvement initiatives.

This workshop, in collaboration with The Recording Academy Producers & Engineers Wing, will introduce you to new data standards, a metadata collection application, and information about how these will impact you in the future. Please join us for this important discussion. Don't be left behind!

Workshop 19 **Saturday, October 31**
11:00 am – 12:30 pm **Room 1A21**

"THE AMERICANS"—MEET THE MIXERS

Presenters: **Ken Hahn**, Sync Sound Inc. / Digital Cinema, LLC, New York, NY, USA
James David Redding, Dial and Music

The Fx series, "The Americans" follows embedded Russian spies (Matthew Rhys and Keri Russell) during in the 1980s. Set in the Washington DC suburbs, but shot in the boroughs of New York, their double lives expose them to both domestic and worldly conflicts. Join Lead Re-recording Mixer (Dial and Music) Ken Hahn and (Sfx) Mixer James David Redding as they look back at some of the challenges they encounter during the sound design and mix of the critically acclaimed "Cold War" drama.

This event is part of the Sound for Pictures Track.

Recording & Mastering 3 **Saturday, October 31**
11:00 am – 12:30 pm **Room 1A23/24**

MASTER CLASS WITH VAL GARAY—RECORDING KIM CARNES, LINDA RONSTADT, AND JAMES TAYLOR

Moderator: **Glenn Lorbecki**, Glenn Sound Inc., Seattle, WA, USA

Presenter: **Val Garay**, Ranleigh Music LLC, Topanga, CA, USA

Val Garay (recording Kim Carnes, Linda Ronstadt, and James Taylor) is a legendary Producer/Engineer with a career spanning 4 decades. He's garnered over 100 Gold & Platinum records, 9 Grammy Nominations, and an Emmy Nomination, with total record sales over 125,000,000 worldwide. In this Master Class, Mr. Garay takes you behind the console for an in-depth look at making hit recordings and working with top-name artists, while discussing his revolutionary recording techniques in detail. Some of the classic recordings to be presented include "Bette Davis Eyes," for which Mr. Garay won the "Record of The Year" GRAMMY as a Producer/Engineer, Linda Ronstadt's "You're No Good," and tracks from James Taylor's acclaimed "JT" album.

Student Event & Career Development
EDUCATION AND CAREER/JOB FAIR
Saturday, October 31, 11:00 am – 12:30 pm
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.

Live Sound Expo 7 **LSE Stage**
Saturday, October 31 **11:00 am – 11:45 am**

SPEECH INTELLIGIBILITY: CONTRIBUTING FACTORS

Presenter: **Dan Palmer**

The cliché installed sound system is hampered by poor reproduction fidelity and reflected sound—hardly desirable when the message is delivered by spoken word, be it a sermon, a reading, an informational announcement or an evacuation warning. Through cases studies of problems solved, this session will demonstrate how systems can live in harmony with their environment.

Saturday, October 31 **11:00 am** **Room 1A19**

Technical Committee Meeting on Loudspeakers and Headphones

Game Audio 7 **Saturday, October 31**
11:45 am – 12:45 pm **Room 1A14**

THE AUDIO IMPLEMENTATION ARMS RACE: IMPLEMENTATION AS A WEAPON

Presenter: **Sally-anne Kellaway**, Technical Sound Designer, FMOD

Overview of some of the engineering problems and challenges in developing a 3D audio solution suitable for widespread use in virtual reality and ordinary gaming and some of the impacts on mixing workflows.

In the realm of game audio, the race to make interactive and adaptive audio implementation accessible to the entire game development team is on. Developers of game engines and audio middleware solutions are making their products more accessible to wider markets by including more tools and UI improvements for absolute beginners and audio professionals alike. Sally Kellaway will discuss the current challenges this movement poses to game audio professionals, and, using FMOD Studio as a lens, illustrate the value in extending a sound design skill base to take command of this element of the audio pipeline. Sound implementation will be explored for the potential it holds, focussing on the value of new tools and workflows that are on offer to sound designers and project teams. This discussion will enable sound designers to argue the value of upgrading and taking ownership of the audio pipeline to include advanced implementation tools.

Special Event
BRIDGING THE GAP BETWEEN CREATIVITY AND TECHNOLOGY: WORKING WITH COMPOSERS ON FILM AND MEDIA PROJECTS
Saturday, October 31, 12:00 noon – 1:00 pm
Room 1A22

Presenter: **Frank Ferrucci**, NYC Film and TV composer

NYC Film and TV composer Frank Ferrucci has presented his seminars at professional, educational, arts and cultural organizations in New York and Brazil to enthusiastic and capacity audiences. This seminar gives a behind the scenes look into the technological challenges composers and engineers face when collaborating on film, television, and other visual media projects. It addresses some less obvious but no less important ways that Music Engineers and Film Mixers can best work with composers. Frank delves into the history and evolution of setting up film cues; working with audio & video formats; working with timings and tempo; working with stems; the use of past and current technology; how computers are used in the creative process and how technology can be used to help collaboration be as seamless as possible. Frank is vice president of Manhattan Producers Alliance, the premier networking organization for audio producers, engineers, sound designers, and composers in NYC.

Live Sound Expo 8 **LSE Stage**
Saturday, October 31 **12:00 noon – 12:45 pm**

MODERN DIGITAL MIXING CONSOLE FUNDAMENTALS: A PRACTICAL AND ERGONOMIC APPROACH

Presenters: **Stephen Bailey**
Richard Ferriday
Matt Larson
Marc Lopez
Robert Scovill

Instant recall of settings and configurations, consistent and predictable performance, increased performance and options in smaller footprints, affordability: for all those benefits and more, digital consoles are dominating in installed sound. While the ergonomics are designed to somewhat emulate analog console signal flow, it doesn't take long for the commonalities of the paradigms to diverge. Today's digital consoles offer refinements in operation networking, along with a broad array of processing through sophisticated plug-in environments. This session offers a ground up, logical approach to digital mixing.

Saturday, October 31 **12:00 noon** **Room 1A19**

Technical Committee Meeting on Network Audio Signals

Project Studio Expo 9 **PSE Stage**
Saturday, October 31 **12:15 pm – 1:00 pm**

BRICKWALL LIMITING: THE INFINITE OPTIONS IN MODERN MUSIC PRODUCTION

Presenter: **Brian Jackson**

Music producers have more options than ever before. It is commonly assumed that this reality is always desirable. Yet, unlimited options can inhibit productivity and creativity. Beginners that are self-producing in the home or project studio are most likely to experience such snags, though the majority of producers will bump into them at some point in their career. This talk explores immediately applicable strategies for maximizing productivity and creativity by avoiding common "option overload" pitfalls.

Spatial Audio Demo 8 **Saturday, October 31**
12:30 pm – 1:30 pm **Room 1A04**

MUSICAL PHENOMENOLOGY: HERBERT VON KARAJAN AND IMMERSIVE AUDIO

Presenter: **Gregor Zielinsky**, Sennheiser Electronic GmbH & Co. KG, Germany

"Musical Phenomenology" refers to what happens in our brains when we hear music. Sergiu Celibidache wrote about it, while Herbert von Karajan practised it his whole musical life. Von Karajan was a sound guru: He loved the real, perfect musical sound. What would have happened if he had learned about 3D sound? The workshop describes von Karajan's way of interpreting music and how it relates to 3D. Many audio examples will be presented. These examples describe how 3D sound actually helps to make music performance much better in 3D than in 2D.

Special Event
SAUL WALKER—THE ROCKET SCIENTIST IN THE RECORDING STUDIO
Saturday, October 31, 12:30 pm – 2:00 pm
Room 1A06

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

Presenter: **Saul Walker**, New York University, New York, NY, USA

Meet Saul Walker, co-founder and former Chief Engineer of API, creator of the 500 Series, including the 512 Mic Preamp, the 550A EQ, and the 500 Lunch Box. His 2520 Op-Amp paved the way for modern audio technology, and his console system designs have remained highly sought-after throughout the past four decades. His designs include digitally controlled spectrum analyzers for NASA and the US Navy, and automated film post-production consoles for major film studios worldwide. Alex U. Case helps navigate the conversation with the inventor, engineer, and educator whose career has had such far-reaching influence, from missiles to mixers, from outer space to rack space.

Student Event & Career Development
SPARS SPEED COUNSELING WITH EXPERTS —MENTORING ANSWERS FOR YOUR CAREER
Saturday, October 31, 12:30 pm – 2:00 pm
Room 1A07

Mentors: **Scott Adamson**, Live
Jamie Baker, Post
Jeff Greenberg, Studio Operations
Bill Higley, Post
Kirk Imamura, Studio Operations
Eric Johnson, Studio Operations
Chris Mara, Mixing
Leslie l. Mona-Mathus, Broadcast
Jun Mizumachi, Post
John O'Mahoney, Mixing
Barry Rudolph, Mixing
Tom Salta & GANG, Gaming
Tony Schultz, Production

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. Mentors are subject to change.

Saturday, October 31 **12:30 pm** **Room 1A20**

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

Live Sound Expo 9 **LSE Stage**
Saturday, October 31 **1:00 pm – 1:45 pm**

MONO VS STEREO VS LCR IN HOW AND FIXED-INSTALL

Presenter: **Jeff Taylor**

Architectural issues, acoustic concerns, audience point-of-view, style of music—all these elements come to play in a decision as to whether to configure a fixed installation system in mono, to attempt stereo, or combine the two in an LCR configuration. We examine the practical considerations in making a decision and in mixing for the chosen configuration.

Saturday, October 31 **1:00 pm** **Room 1A19**

Technical Committee Meeting on Spatial Audio

Project Studio Expo 10 **PSE Stage**
Saturday, October 31 **1:15 pm – 2:15 pm**

MIXING AN ENSEMBLE RECORDING

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

Recording a bunch of musicians in the same room is usually the quickest and most fun way to get results, but too many project-studio engineers still painstakingly isolate and/or overdub everything while tracking because of worries about spill at mixdown. In this nuts-and-bolts seminar, *Sound On Sound* magazine's "Session Notes" and "Mix Rescue" columnist Mike Senior will use real-world small-studio recordings to demonstrate a wide variety of methods for dealing with spill—and how it's frequently a blessing, not a curse.

Tutorial 18 **Saturday, October 31**
1:30 pm – 3:00 pm **Room 1A14**

HORNS AND WAVEGUIDES—HISTORY, THEORY, AND TECHNOLOGY*

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

Horns are undoubtedly the oldest audio equipment. There are two major functions of horns: providing high-efficiency of horn-loaded transducers (often in combination with phasing plugs in compression drivers,) and providing desirable SPL coverage and directivity control. The tutorial will consider such aspects of horns as derivation of basic horn wave equation (Webster Equation), analysis of directivity control, the role of the wavefront at the throat of horn on directivity of horns at high frequencies, influence of the high-order modes on performance of horns, influence of the mouth diffraction on performance, comparison of axisymmetric, elliptical, and rectangular horns. Also nonlinear propagation effects due are analyzed and explained. Retrospective review of patents is carried out as well as the review of the technical achievements of JBL Professional in horns and waveguides technologies.

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

Broadcast/Streaming Media 11 **Saturday, October 31**
1:30 pm – 3:00 pm **Room 1A10**

AUDIO FOR BROADCAST VIDEO—IMMERSIVE, PERSONALIZED, 4K, AND 8K

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

Panelists: *Tim Carroll*, Telos Alliance, Lancaster, PA, USA
Kazuho Ono, NHK Engineering System Inc., Setagaya-ku, Tokyo, Japan
Skip Pizzi, NAB, Washington DC, USA
Robert Reams, Consultant, Santa Clara, CA, USA
Jeff Riedmiller, Dolby Laboratories, San Francisco, CA USA
James Moore, DTS, Inc.

Each of our panelists is at the forefront of harnessing technology to augment the human experience of immersive and personalized audio in 4K, 8K, and ATSC 3.0 broadcast and streaming. Come to share the latest and greatest discoveries and standards proposals forged in the past year. Work in this sphere progresses at a feverish pace as we close in on final standardization and the next magnitude of consumer psycho-acoustic involvement. Don't miss this chance to present your questions to our experts after they share their most recent stories and accomplishments.

Networked Audio 7 **Saturday, October 31**
1:30 pm – 2:30 pm **Room 1A13**

BENEFITS OF AES67 TO THE END USER

Presenter: **Rich Zwiebell**

Presenter Rich Zwiebel has a long history in audio networking. He was a founder of Peak Audio, the company that developed CobraNet, the first widely used audio network for Professional applications. As a VP at QSC he continues to be very active in the field and is currently the Chairman of the Media Networking Alliance.

This presentation reviews the history of professional studio networking, where we are today, and what the future may hold. A clear explanation of what AES67 is, as well as what it is not, along with how it will benefit those who choose to use it will be included. Attendees will understand it's relationship to existing audio network technologies in the market.

Additionally, an explanation of who the Media Networking Alliance is, who it's members are, and what it's goals are will be presented.

A discussion of the advantages of a single facility network will close out the session.

Archiving 8 **Saturday, October 31**
1:45 pm – 3:45 pm **Room 1A12**

THE 78 PROJECT: SPECIAL SCREENING AND Q&A WITH CREATORS FOLLOWED BY LIVE RECORD CUTTING

Presenters: **Alex Steyermark**, Filmmaker
Lavinia Jones Wright, Filmmaker

The 78 Project is on a journey across America to make one-of a-kind 78rpm records with musicians in their hometowns using a 1930s Presto direct-to-disc recorder. With one microphone. With one blank disc. In one 3-minute take. Along the way, a kaleidoscope of technologists, historians, and craftsmen from every facet of field recording—Grammy-winning producers, 78 collectors, curators from the Library of Congress and Smithsonian—provide insights and history. In Tennessee, Mississippi, California, Louisiana the folk singers, punk rockers, Gospel and Cajun singers in the film share their lives through intimate performances, and find in that adventure a new connection to our cultural legacy.

Recording & Mastering 4 **Saturday, October 31**
1:45 pm – 3:15 pm **Room 1A21**

RAW TRACKS 2.0—ANATOMY OF: A LATIN JAZZ CLASSIC—THE AFRO LATIN JAZZ ORCHESTRA IN HAVANA

Moderator: **Daniel Freiberg**, Daniel Freiberg Music, Westchester, NY, USA

Panelists: *Orestes Aquila*
Katherine Miller, Annandale Recording, North Plainfield, NJ, USA

Continuing the momentum of their multiple-Grammy winning "Offense of the Drum," the ALJO's new album, *Cuba: The Conversation Continues*, recorded this year in Havana, sets a high bar for depth, power, and dynamism. Brilliant music, stellar performances, and top-notch engineering make this set a one-of-kind masterpiece. Buena Vista Social Club engineer, Orestes Aguilar, and Grammy-winning mixer, Katherine Miller, show how it was done with photos and live ProTools sessions.

Session P15 **Saturday, October 31**
2:00 pm – 5:30 pm **Room 1A08**

SPATIAL AUDIO—PART 2

Chair: **Filippo Maria Fazi**, University of Southampton, Southampton, Hampshire, UK

2:00 pm

P15-1 Capturing the Elevation Dependence of Interaural Time Difference with an Extension of the Spherical-Head Model—*Rahulram Sridhar, Edgar Choueiri*, Princeton University, Princeton, NJ, USA

An extension of the spherical-head model (SHM) is developed to incorporate the elevation dependence observed in measured interaural time differences (ITDs). The model aims to address the inability of the SHM to capture this elevation dependence, thereby improving ITD estimation accuracy while retaining the simplicity of the SHM. To do so, the proposed model uses an elevation-dependent head radius that is individualized from anthropometry. Calculations of ITD for 12 listeners show that the proposed model is able to capture this elevation dependence and, for high frequencies and at large azimuths, yields a reduction in mean ITD error of up to 13 microseconds (3% of the measured ITD value), compared to the SHM. For low-frequency ITDs, this reduction is up to 160 microseconds (23%).
Convention Paper 9447

2:30 pm

P15-2 Temporal Reliability of Subjectively Selected Head-Related Transfer Functions (HRTFs) in a Non-Eliminating Discrimination Task—*Yunhao Wan, Ziqi Fan, Kyla McMullen*, University of Florida, Gainesville, FL, USA

The emergence of commercial virtual reality devices has reinvigorated the need for research in realistic audio for virtual environments. Realistic virtual audio is often realized through the use of head-related transfer functions (HRTFs) that are costly to measure and individualistic to each listener, thus making their use unscalable. Subjective selection allows a listener to pick their own HRTF from a database of premeasured HRTFs. While this is a more scalable option further research is needed to examine listeners' consistency in choosing their own HRTFs. The present study extends the current subjective selection research by quantifying the reliability of subjectively selected HRTFs by 12 participants over time in a non-eliminating perceptual discrimination task.
Convention Paper 9448

3:00 pm

P15-3 Plane-Wave Decomposition with Aliasing Cancellation for Binaural Sound Reproduction—*David L. Alon, Jonathan Sheaffer, Boaz Rafaely*, Ben-Gurion University of the Negev, Beer Sheva, Israel

Spherical microphone arrays are used for capturing three-dimensional sound fields, from which binaural signals can be obtained. Plane-wave decomposition of the sound field is typically employed in the first stage of the processing. However, with practical arrays the upper operating frequency is limited by spatial aliasing. In this paper a measure of plane-wave decomposition error is formulated to highlight the problem of spatial aliasing. A novel method for plane-wave decomposition at frequencies that are typically considered above the maximal operating frequency is then presented, based on the minimization of aliasing error. The mathematical analysis is complemented by a simulation study and by a preliminary listening experiment. Results show a clear perceptual improvement when aliasing-cancellation is applied to aliased binaural signals, indicating that the proposed method can be used to extend the bandwidth of binaural signals rendered from microphone array recordings.
Convention Paper 9449

3:30 pm

P15-4 Modeling ITDs Based on Photographic Head Information—*Jordan Juras, Christian Miller, Agnieszka Roginska*, New York University, New York, NY, USA

Research has shown that personalized spatial cues used in 3D sound simulation lead to an improved perception and quality of the sound image. This paper introduces a simple method for photographically extracting the size of the head, and proposes a fitted spherical head model to more accurately predict Interaural Time Differences (ITD). Head-Related Impulse Responses (HRIR) were measured on eleven subjects, and ITDs were extracted from the measurements. Based on a photograph taken of each subject's face, the distance between the ears was measured and used to model a subject's personal ITDs. A head model is proposed that adjusts the spherical head model to more accurately model ITDs. Acoustic measurements of ITDs are then compared to the modeled ITDs demonstrating the effectiveness of the proposed method for photographically extracting personalized ITDs.
Convention Paper 9450

4:00 pm

P15-5 Recalibration of Virtual Sound Localization Using Audiovisual Interactive Training—*Xiaoli Zhong, Jie Zhang, Guangzheng Yu*, South China University of Technology, Guangzhou, Guangdong, China

In virtual auditory display, non-individual head-related transfer functions (HRTF) of KEMAR result in localization degradation. This work investigates the efficacy of audiovisual interactive training as to the recalibration of such localization degradation. First, an audiovisual interactive training system consisting of control module, binaural virtual sound module, and vision module, was constructed. Then, ten subjects were divided into a control group and a training group, and underwent three-day training and localization tests. Results indicate that in the horizontal plane, the localization accuracy of azimuth is significantly improved with training and the front-back confusion is also reduced; however, in the median plane a three-day short-term training has no significant improvement on the localization accuracy of elevation.
Convention Paper 9451

4:30 pm

P15-6 Analysis and Experiment on Summing Localization of Two Loudspeakers in the Median Plane—*Bosun Xie, Dan Rao*, South China University of Technology, Guanzhou, Guangdong, China

Based on the hypothesis that the change of interaural time difference caused by head rotation and tilting provides dynamic cues for front-back and vertical localization, low-frequency localization equations or panning laws for multiple loudspeakers in the median plane were derived in our previous work. In present work we further supplement some psychoacoustic explanation of these equations and utilize them to analyze the summing localization of two loudspeakers with various configurations and pair-wise amplitude panning in the median plane. Relationship between current method and other localization theorems is also analyzed. Results indicate that for some configurations, pair-wise amplitude panning is able to create virtual sources between loudspeakers. However, it is unable to do so for some other loudspeaker configurations. A virtual source localization experiment yields consistent results with those of analysis, and therefore validates the proposed method.
Convention Paper 9452

5:00 pm

P15-7 Immersive Audio Content Creation Using Mobile Devices and Ethernet AVB—*Richard Foss,[1] Antoine Rouget[2]*
[1]Rhodes University, Grahamstown, Eastern Cape, South Africa
[2]DSP4YOU Ltd., Kowloon, Hong Kong

The goal of immersive sound systems is to localize multiple sound sources such that listeners are enveloped in sound. This paper describes an immersive sound system that allows for the creation of immersive sound content and real time control over sound source localization. It is a client/server system where the client is a mobile device. The server receives localization control messages from the client and uses an Ethernet AVB network to distribute appropriate mix levels to speakers with in-built signal processing.
Convention Paper 9453

Game Audio 8 **Saturday, October 31**
2:00 pm – 3:30 pm **Room 1A22**

GAME AUDIO CAREERS—BLAZING A PATH TO YOUR FUTURE

Chair: **Stephen Harwood, Jr.**, Education Working Group Chair; IASIG, New York, NY, USA

Panelists: *Bonnie Bogovich*, Schell Games, Pittsburgh, PA, USA
Jacques Deveau, Audiokinetic, Montreal, QC, Canada
Jason Kanter, Avalanche Studios
Tom Salta, Persist Music, Norwalk, CT, USA
Brian Walker, Audio Director, Leap Frog

Games are big business. From social and mobile games to consoles the field is diverse and growing. 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!.

Special Event
GRAMMY SOUNDTABLE: AFTER HOURS—
MIXING FOR LATE NIGHT NEW YORK
Saturday, October 31, 2:00 pm – 3:30 pm
Room 1A23/24

Moderator: **Will Lee**

Presenters: *Josiah Gluck*
Harvey Goldberg
Lawrence Manchester, Late Night with Jimmy Fallon, New York, NY, USA

It's a new era on late night television, but one thing hasn't changed: Side by side with political satire and cutting edge comedy, these programs continue to be the go-to showcase for the best in music, from buzz-building newcomers to the established cream of the crop—not to mention killer house bands! Join us for a personal conversation with the music mixers for "The Late Show With Stephen Colbert," "The Tonight Show Starring Jimmy Fallon," and "Saturday Night Live" as we delve into the logistics, challenges and technical expertise required to make these groundbreaking shows happen.

Live Sound Expo 10 **LSE Stage**
Saturday, October 31 **2:00 pm – 2:45 pm**

IEM FUNDAMENTALS AND HEARING CONSERVATION

Presenter: **Mark Frink**

Drawing on his decades of road experience, our LSE host, Mark Frink, explains the logic behind moving live performers to personal In-Ear-Monitoring solutions. Topics will include the selection of IEMs (universal vs. custom), mixing monitors for IEMs, personal mixing by performers and protecting performers hearing.

Saturday, October 31 **2:00 pm** **Room 1A19**

Technical Committee Meeting on Sound for Digital Cinema and Television

Session P16 **Saturday, Oct. 31**
2:15 pm – 3:45 pm **Room 1A07**

ROOM ACOUSTICS

Chair: **Remi Audfray**, Dolby Laboratories, Inc., San Francisco, CA, USA

2:15 pm

P16-1 Environments for Evaluation: The Development of Two New Rooms for Subjective Evaluation—*Elisabeth McMullin, Adrian Celestinos, Allan Devantier,* Samsung Research America, Valencia, CA, USA

An overview of the optimization, features, and design of two new critical listening rooms developed for subjective evaluation of a wide-array of audio products. Features include a rotating wall for comparing flat-panel televisions, an all-digital audio switching system, custom tablet-based testing software for running a variety of listening experiments, and modular acoustic paneling for customizing room acoustics. Using simulations and acoustic measurements, a study of each of the rooms was performed to analyze the acoustics and optimize the listening environment for different listening situations.
Convention Paper 9460

2:45 pm

P16-2 Low Frequency Behavior of Small Rooms—*Renato Cipriano, Robi Hersberger, Gabriel Hauser, Dirk Noy, John Storyk*, Walters-Storyk Design Group, Highland, NY, USA

Modeling of sound reinforcement systems and room acoustics in large- and medium-size venues has become a standard in the audio industry. However, acoustic modeling of small rooms has not yet evolved into a widely accepted concept, mainly because of the unavailable tool set. This work introduces a practical and accurate software-based approach for simulating the acoustic properties of studio rooms based on BEM. A detailed case study is presented and modeling results are compared with measurements. It is shown that results match within given uncertainties. Also, it is indicated how the simulation software can be enhanced to optimize loudspeaker locations, room geometry, and place absorbers in order to improve the acoustic quality of the space and thus the listening experience.
Convention Paper 9461

3:15 pm

P16-3 Measuring Sound Field Diffusion: SFDC—*Alejandro Bidondo, Mariano Arouxet, Sergio Vazquez, Javier Vazquez, Germán Heinze*, Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina

This research addresses the usefulness of an absolute descriptor to quantify the degree of diffusion in a third octave band basis of

a sound field. The degree of sound field diffuseness in one point is related with the reflection's energy control multiplied by the temporal distribution uniformity of reflections. All this information is extracted from a monaural, broadband, omnidirectional, high S/N impulse response. The coefficient range varies between 0 and 1, evaluates the early, late, and total sound field for frequencies above Schroeder's and in the far field from diffusive surfaces, zero being "no diffuseness" at all. This coefficient allows the comparison of different rooms, different places inside rooms, measurement of the effects of different sound diffusers coatings, and the resulting spatial uniformity variation, among other applications.
Convention Paper 9462

Session P17 **Saturday, October 31**
2:15 pm – 3:45 pm **Foyer**

POSTERS: APPLICATIONS IN AUDIO

2:15 pm

P17-1 Application of Object-Based Audio for Automated Mixing of Live Football Broadcast—*Robert Oldfield, Ben Shirley, Darius Satongar*, University of Salford, Salford, Greater Manchester, UK

The challenge of creating a live sound mix for a sports event such as a football/soccer match cannot be underestimated. The mixing engineer needs to constantly raise and lower the levels of the faders corresponding to the pitch-side microphones that cover the area of the pitch containing the action at that point in time such that the on-pitch sounds can be heard over the crowd noise. This paper presents an automation of this process based on the detection of audio objects in the microphone feeds and then controls the levels of the faders on the mixing console accordingly. This paper includes a brief description of the underlying algorithms for the detection of ball-kicks and whistle-blows and describes how such a system can be integrated into current broadcast workflows.
Convention Paper 9454

2:15 pm

P17-2 Personal Adaptive Tuning of Mobile Computer Audio—*Kuba Lopatka, Jozef Kotus, Piotr Suchomski, Andrzej Czyzewski, Bozena Kostek,* Gdansk University of Technology, Gdansk, Poland

An integrated methodology for enhancing audio quality in mobile computers is presented. The key features are adaptation of the characteristics of the acoustic track to the changing conditions and to the user's individual preferences. Original signal processing algorithms are introduced, which concern linearization of frequency response, dialogue intelligibility enhancement, and dynamics processing tuned up to the user's preferences. The principles of the algorithm implemented in the C++ programming language are provided. The processing is performed utilizing custom Audio Processing Objects (APO) installed in Windows sound system. The sound enhancement bundle is managed with a User Interface enabling control over the sound system. The results of subjective evaluation of the introduced methods are discussed.
Convention Paper 9455

2:15 pm

P17-3 Audio Effects Data on the Semantic Web—*Thomas Wilmering, György Fazekas, Alo Allik, Mark B. Sandler*, Queen Mary University of London, London, UK

We discuss the development of a linked data service exposing metadata about audio effect implementations. The data is collected automatically from Web sources as well as by extracting information from effect plugin binaries, and by manual data entry and correction using a Web service. Automatically generated RDF data is represented using vocabulary terms defined by the Audio Effects Ontology. A SPARQL endpoint allows for the integration of this data resource in novel audio production software and services for the classification, comparison, and recommendation of effects, taking advantage of semantic descriptors.
Convention Paper 9456

2:15 pm

P17-4 Speech Music Discrimination Using an Ensemble of Biased Classifiers—*Kibeom Kim, Anant Baijal, Byeong-Seob Ko, Sangmoon Lee, Inwoo Hwang, Youngtae Kim*, Samsung Electronics Co. Ltd., Suwon, Gyeonggi-do, Korea

In this paper we present a novel framework for real-time speech/music discrimination (SMD). The proposed method improves the overall accuracy of automatically classifying the signals into speech, singing, or instrumental categories. In our work, first, we design several groups of classifiers such that each group's classification decision is biased towards a certain class of sounds; the bias is induced by training different groups of classifiers on perceptual features extracted at different temporal resolutions. Then, we build our system using an ensemble of these biased classifiers organized in a parallel classification fashion. Last, these ensembles are combined with a weighting scheme, which can be tuned in either forward-weighting or inverse-weighting modes, to provide accurate results in real-time. We show, through extensive experimental evaluations, that the proposed ensemble of biased classifiers framework yields superior performance compared to the baseline approach.
Convention Paper 9457
[This paper was not presented but is available in the E-Library]

2:15 pm

P17-5 Multi-Criteria Decision Aid Analysis of a Musification Approach to the Auditory Display of Micro-Organism Movement—
Duncan Williams,[1] *Laurence Wilson*[2]
[1]University of Plymouth, Devon, UK
[2]University of York, Heslington, York, UK

We evaluate a musification approach to the auditory display of P. berghei flagella movement (a micro-organism that is commonly used in laboratory analysis of malaria transmission). High resolution 3D holography techniques provide the source data. The ultimate goal of this work is to develop an auditory display that could successfully augment existing visual analysis of bacteria motility in-field. The requirement for musification as opposed to sonification, and methods for evaluating the success of this implementation, are explored. An evenly weighted multi-criteria decision aid analysis was undertaken of amenity, immersion, intuitivity, efficiency, and congruency of the musification. Listeners consistently rated the amenity, intuitivity, and congruency of the musification above that of the visual only display and that of a randomized audio accompaniment.
Convention Paper 9458

2:15 pm

P17-6 An Overview of an Online Audio Electronics Curriculum Offered at the Indiana University Jacobs School of Music—
Michael Stucker, Indiana University, Bloomington, IN, USA

An overview will be given of both the pedagogical and technical design of an online curriculum to teach electronics, specifically analog audio electronics. This approach worked to create enhanced engagement in students and allow students to work on their own schedule while still having instructional support. Engagement is particularly difficult with courses taught online and extra effort must be taken to create activities that will increase student participation, focus, and engagement. A great deal of the engagement in an in-person course comes from the interaction of the people involved in the course, whether instructor or student. Creating methods and compelling reasons for student-student and student-instructor interactions is critical to the success of an online course. One of the benefits of online courses is the ability for students to work according to their own schedule. For an online course to be effective, instructional support must be available during whatever hours the student chooses to work on course materials. It is certainly not possible for an instructor to be always available, but course materials can be designed to provide interactive instructional support. This paper will provide an overview of the course design created to solve the aforementioned problems. This will include both the technical details as well as the pedagogy behind the design.
Convention Paper 9459

2:15 pm

P17-7 A Connection Management System to Enable the Wireless Transmission of MIDI Messages—*Brent Shaw, Richard Foss*, Rhodes University, Grahamstown, Eastern Cape, South Africa

This paper examines the design and implementation of a wireless system for the distribution of MIDI messages for show control and studio environments. The system makes use of the MIDI and MIDINet protocols, creating wireless nodes that will enable the transmission of MIDI between devices on a wireless network with connection management capabilities through the use of embedded web servers. The paper describes the current state of the art, configuration of the system, hardware architectures, software design, and implementation.
Convention Paper 9474

Saturday, October 31 **2:30 pm** **Room 1A20**

Standards Committee Meeting SC-07-01 Metadate for Audio

Product Development 9 **Saturday, October 31**
2:45 pm – 4:15 pm **Room 1A13**

CREATING HIGH-RESOLUTION MODELING DATA FOR LOUDSPEAKERS AND LINE ARRAYS

Presenter: **Stefan Feistel**, AFMG Technologies GmbH, Berlin, Germany

This tutorial will discuss all important aspects of developing and publishing loudspeaker data, e.g., GLL files, for sound system modeling and optimization software. Particular focus will be put on the business aspects and technical implications of data acquisition, loudspeaker data formats, certification, and distribution to end users. It is demonstrated how mechanical information, acoustic measurements, and DSP configuration merge into one encompassing data set that describes a loudspeaker system as a whole. It is also explained how DSP control software and other tools can directly interface with modeling and optimization software in order to simplify workflows and improve user experience. Finally, typical application scenarios and best practices are discussed. Examples will be given using EASE, Focus, and Evac software.

Project Studio Expo 11 **PSE Stage**
Saturday, October 31 **2:30 pm – 3:15 pm**

MASTERING INSIDE YOUR DAW

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

You don't always need a dedicated editor to do mastering—find out how to use the tools already in your DAW to add that final professional polish to your mixes.

Live Sound Expo 11 **LSE Stage**
Saturday, October 31 **3:00 pm – 3:45 pm**

THE FUTURE OF WIRELESS: NOW WHAT?

Presenters: **Marc Brunner**
Joe Ciaudelli
Howard Kaufman

There has been dramatic erosion in the television band spaces available for wireless microphone and monitor use. How can a facility find available bandwidth and stay legal? What can be done to future-proof a system? Do 2.4 Ghz and like systems offer a solution, and if so, for whom? What can digital wireless bring to the equation? All these questions and more will be addressed.

Saturday, October 31 **3:00 pm** **Room 1A19**

Technical Committee Meeting on Semantic Audio Analysis

Broadcast/Streaming Media 12 **Saturday, October 31**
3:15 pm – 4:45 pm **Room 1A10**

INTEGRATING MOBILE TELEPHONY AND IP IN BROADCAST

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

Panelists: *Mitch Glider*, iHeart Media
Dave immer, DIGIFON/Walnut Studio
Tony Peterle, Worldcast Systems, Inc.
Paul Shulins, Greater Media, Boston, MA, USA
Chris Tobin, Technology Consultant, IPCodecs.com
Andrew Zarian, GFQ Network

Engineers are tasked to get broadcasts on-air from some very out-of-the-way places. From car dealers to tire shops, from park concerts to canoe races, and even the occasional balloon glow miles from anywhere, broadcast engineers strive to bring audio and video back to the station. The once-reliable telephone companies have already reduced their connection offerings and service footprint, leaving engineers to be creative in tying the studio to the off-site event. Radio engineers need workable alternatives to ISDN, and many television engineers are restricted from using the satellite truck unless absolutely necessary, leaving them to seek high-bandwidth connectivity, too. Today's connectivity solutions typically involve IP—both wired and wireless. Often times a clever mix of IP connection services and technologies are required to bring a broadcast in from the field.
Leading broadcast engineers and vendor technical representatives will discuss the latest practices and workable solutions to "get there from anywhere."

Networked Audio 8 **Saturday, October 31**
3:15 pm – 4:45 pm **Room 1A14**

HOW TO GET AES67 INTO YOUR SYSTEMS/PRODUCTS

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

Panelists: *Michael Dosch*, Lawo AG
Nathan Phillips, Coveloz
Greg Shay, The Telos Alliance, Cleveland, OH, USA
Nicolas Sturmel, Digigram S.A., Montbonnot Saint Martin, France
Arie van den Broek, Archwave AG, Meisterschwanden, Switzerland
Kieran Walsh, Audinate

This workshop introduces several options to implement AES67 networking capabilities into existing or newly designed products. The session starts with a quick recap on the technical ingredients of AES67 and points out the principal options on implementing AES67 into new or existing products. After providing an overview on commercially available building blocks (modules, software libraries and reference designs), the workshop commences in a discussion on the value of providing AES67 compatibility from the perspective of providers of existing AoIP networking solutions. The workshop is targeted towards product manufacturers seeking ways to implement AES67 into their products, but should also provide valuable insight to those with general technical interest in AES67.

Workshop 20 **Saturday, October 31**
3:30 pm – 5:30 pm **Room 1A21**

PERCEPTUAL EVALUATION OF HIGH RESOLUTION AUDIO

Chair: **Joshua D. Reiss**, Queen Mary University of London, London, UK

Panelists: *Bob Katz*, Digital Domain Mastering, Orlando, FL, USA
George Massenburg, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Bob Schulein, RBS Consultants, Schaumburg, IL, USA

This workshop focuses on past measurements and future potential in perceptual evaluation of high resolution audio. Past attempts to assess the audibility of higher resolutions (beyond 44.1 kHz, 16-bit) will be summarized with an overview of results, but the focus of the workshop is on testing and methodology itself. Discussion will include the problems and pitfalls of listening tests and demos and how they might be overcome. We shed some light on the psychoacoustic justifications behind the results of previous experiments, including what is known and what is not. We discuss issues in evaluating quality and perception, the structuring of tests, configuration of the testing environment, and analysis of results. Attention will be paid to the choice of test material. The talks will be supplemented with a demonstration, and we intend to engage the audience with lively discussion.

Game Audio 9 **Saturday, October 31**
3:30 pm – 5:00 pm **Room 1A22**

GAME AUDIO EDUCATION—NEW OPPORTUNITIES FOR STUDENTS

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

Panelists: *Scott Looney*, Academy of Art University, San Francisco, CA, USA
Leonard J. Paul, School of Video Game Audio, Vancouver, Canada
Winifred Phillips, Composer, LittleBigPlanet franchise, Assasins Creed Liberation, God of War, author A Composer's Guide to Game Music, New York City Metropolitan Area
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 and teachers from some growing programs 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.

Project Studio Expo 12 **PSE Stage**
Saturday, October 31 **3:30 pm – 4:15 pm**

THE REAL SKILLS YOU NEED TO RECORD PROFESSIONALLY

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

It's easy to think that recording equipment is all one needs to run a successful studio, but it's really a small part of the equation. Join *Tape Op Magazine*'s founder/editor, Larry Crane, as he uncovers the true skills and mindsets that all successful and busy audio professionals really utilize.

Session EB5 **Saturday, Oct. 31**
3:45 pm – 4:15 pm **Room 1A07**

ACOUSTICS

Chair: **Jung Wook (Jonathan) Hong**, McGill University, Montreal, QC, Canada; GKL Audio Inc., Montreal, QC, Canada

3:45 pm

EB5-1 Visualization of Compact Microphone Array Room Impulse Responses—*Luca Remaggi, Philip Jackson, Philip Coleman, Jon Francombe*, University of Surrey, Guildford, Surrey, UK

For many audio applications, availability of recorded multichannel room impulse responses (MC-RIRs) is fundamental. They enable development and testing of acoustic systems for reflective rooms. We present multiple MC-RIR datasets recorded in diverse rooms, using up to 60 loudspeaker positions and various uniform compact microphone arrays. These datasets complement existing RIR libraries and have dense spatial sampling of a listening position. To reveal the encapsulated spatial information, several state of the art room visualization methods are presented. Results confirm the measurement fidelity and graphically depict the geometry of the recorded rooms. Further investigation of these recordings and visualization methods will facilitate object-based RIR encoding, integration of audio with other forms of spatial information, and meaningful extrapolation and manipulation of recorded compact microphone array RIRs.
Engineering Brief 218

4:00 pm

EB5-2 Sensible 21st Century Saxophone Selection—*Thomas Mitchell*, University of Miami ,Coral Gables, FL, USA

This paper presents a method for selecting a saxophone, using data mining techniques with both subjective and objective data as criteria. Immediate, subjective personal impressions are given equal weight with more-objective observations made after the fact, and with hard data distilled from audio data using MIR Toolbox. Offshoots and directions for future research are considered.
Engineering Brief 219

Recording & Mastering 5 **Saturday, October 31**
3:45 pm – 5:15 pm **Room 1A23/24**

RAW TRACKS 2.0—ANATOMY OF: ON-SET RECORDING "NASHVILLE"—KEEPING IT REAL

Moderator: **Jim Kaiser**, CEMB / Belmont University, Nashville, TN, USA

Panelists: *Matt Andrews*
Michael Columby
Fred Paragano
Mike Poole, Mike Poole, Nashville, TN, USA
Glen Trew, Trew Audio, Nashville, TN, USA
Richard Weingart

The critically acclaimed "Nashville" series (now in its fourth season) is notable for its reliance on "live music performance" as key to its storyline. Join our panel of experienced professionals as they reveal the intricacies required to maintain continuity and reality in each music scene, while supporting the acting talent and providing a compelling (seamless) viewing experience. Some of these features include the capture of the "musical performance," along with preparation to make sure all pre-production music will fit these requirements. Examples from previous season's shows will demonstrate the way in which the music is developed and recorded in order to best suit the scene and all format releases (including iTunes).

This event is part of the Sound for Pictures Track.

Live Sound Expo 12 **LSE Stage**
Saturday, October 31 **4:00 pm – 4:45 pm**

MIKING GRAND PIANO AND CHOIRS

Presenters: **Daryl Bornstein**
Mark Frink
Jeremiah Slovarp

In Houses Of Worship, regardless of worship styles, acoustic grand piano and choirs are the most consistent sound sources to have fixed mics employed for sound reinforcement. This session covers the selection of mics, placement and tips for keeping a set-up consistent.

Saturday, October 31 **4:00 pm** **Room 1A20**

Standards Committee Meeting SC-04-08 Measurement of Sound Systems in Rooms

Session P18 **Saturday, Oct. 31**
4:15 pm – 5:45 pm **Foyer**

POSTERS: RECORDING AND PRODUCTION

4:15 pm

P18-1 The Impact of Subgrouping Practices on the Perception of Multitrack Music Mixes—*David M. Ronan, Brecht De Man, Hatice Gunes, Joshua D. Reiss*, Queen Mary University of London, London, UK

Subgrouping is an important part of the mix engineering workflow that facilitates the process of manipulating a number of audio tracks simultaneously. We statistically analyze the subgrouping practices of mix engineers in order to establish the relationship between subgrouping and mix preference. We investigate the number of subgroups (relative and absolute), the type of audio processing, and the subgrouping strategy in 72 mixes of 9 songs, by 16 mix engineers. We analyze the subgrouping setup for each mix of a particular song and also each mix by a particular mixing engineer. We show that subjective preference for a mix strongly correlates with the number of subgroups and, to a lesser extent, which types of audio processing are applied to the subgroups.
Convention Paper 9442

4:15 pm

P18-2 MixViz: A Tool to Visualize Masking in Audio Mixes —*Jon Ford, Mark Cartwright, Bryan Pardo*, Northwestern University, Evanston, IL, USA

This paper presents MixViz, a real-time audio production tool that helps users visually detect and eliminate masking in audio mixes. This work adapts the Glasberg and Moore time-varying Model of Loudness and Partial Loudness to analyze multiple audio tracks for instances of masking. We extend the Glasberg and Moore model to allow it to account for spatial release from masking effects. Each audio track is assigned a hue and visualized in a 2-dimensional display where the horizontal dimension is spatial location (left to right) and the vertical dimension is frequency. Masking between tracks is indicated via a change of color. The user can quickly drag and drop tracks into and out of the mix visualization to observe the effects on masking. This lets the user intuitively see which tracks are masked in which frequency ranges and take action accordingly. This tool has the potential to both make mixing easier for novices and improve the efficiency of expert mixers.
Convention Paper 9443

4:15 pm

P18-3 Sound Capture Technical Parameters of Colombian Folk Music Instruments for Virtual Sound Banks Use—*Carlos Andrés Caballero, J. Mauricio Moreno*, Instituto Tecnológico Metropolitano, Medellín, Antioquia, Colombia

This paper describes the appropriate and correct way of dealing with the technical conceptualizations required for the digital sound capture of Colombian folk music instruments, taking into account the particular para-meters of each instrument and the current audio file formats used in virtual sound banks. This paper does not pose either new capture techniques or microphone placements. Instead, the task carried out herein uses well known methods in order to get precise and clear audio takes that will allow a significant number of audio samples for the configuration of sound banks that can be used in music software and also as virtual instruments. The different tests and analysis carried out showed that a broad sound capture is required (covering the overall instrument range), using plain frequency response microphones, with high-resolution digital conversion formats (96 kHz/24 bits), and near and distant stereo recordings, all these in acoustically-controlled and well-conditioned ambiences.
Convention Paper 9444

4:15 pm

P18-4 Vocal Clarity in the Mix: Techniques to Improve the Intelligibility of Vocals—*Yuval Ronen*, New York University, New York, NY, USA

From interviewing leading professional mixing engineers and from research of known literature in the field common mixing techniques to improve the intelligibility of vocals were gathered. An experiment to test these techniques has been conducted on randomly selected participants with normal hearing and with no mixing or recording expertise. The results showed statistically significant differences between processed audio clips using these techniques versus unprocessed audio clips. To the author's knowledge, this is the first study of its kind, which proved that certain common mixing techniques statistically improve intelligibility of vocals in popular music as perceived by human subjects.
Convention Paper 9445

4:15 pm

P18-5 Affective Potential in Vocal Production—*Duncan Williams*, University of Plymouth, Devon, UK

The study of affect in music psychology—broadly construed as emotional responses communicated to, or induced in, the listener—increasingly concludes that voice processing can provide a powerful vector for emotional communication in the music production chain. The audio engineer has the ability to create a "definitive article" in the studio that gives listeners an opportunity to engage with the recorded voice in a manner that is quite distinct from everyday speech or the effect that might be achieved in a typical live performance. This paper examines the affective potential of the voice in a number of examples from popular music where the production chain has been exploited to provide

a technological mediation to the listener's emotional response.
Convention Paper 9446

4:15 pm

P18-6 Sample-Rate Variance across Portable Digital Audio Recorders —*Robert Oldfield, Paul Kendrick*, University of Salford, Salford, UK

In recent years there has been an increase in the use of portable digital recording devices such as, smart phones, tablets, dictaphones, and other portable hand-held recorders for making informal or in-situ recordings. Often it is not possible to connect a clocking signal to these devices as such recordings are affected by the deviations of the actual clocking rate of the device from the expected rate. This variation causes problems in the synchronization of signals from multiple recording devices and can prevent the use of some signal processing algorithms. This paper presents a novel methodology for determining the actual clock rate of digital recording devices based upon optimizing the correlation between a recording and a ground truth signal with varying degrees of temporal stretching. The paper further discusses the effects of sample frequency variation on typical applications. The sampling rates of a range of commonly used mobile audio recording devices was found to deviate from the nominal 48 kHz, with a standard deviation of 0.8172 Hz. The standard deviation of sampling rates for a single device type, used for long term logging of bio-acoustic signals, was found to be 0.1983 Hz (at a sampling rate of 48 kHz).
Convention Paper 9470

4:15 pm

P18-7 Comparison of Loudness Features for Automatic Level Adjustment in Mixing—*Gordon Wichern, Aaron Wishnick, Alexey Lukin, Hannah Robertson*, iZotope, Cambridge, MA, USA

Manually setting the level of each track of a multitrack recording is often the first step in the mixing process. In order to automate this process, loudness features are computed for each track and gains are algorithmically adjusted to achieve target loudness values. In this paper we first examine human mixes from a multitrack dataset to determine instrument-dependent target loudness templates. We then use these templates to develop three different automatic level-based mixing algorithms. The first is based on a simple energy-based loudness model, the second uses a more sophisticated psychoacoustic model, and the third incorporates masking effects into the psychoacoustic model. The three automatic mixing approaches are compared to human mixes using a subjective listening test. Results show that subjects preferred the automatic mixes created from the simple energy-based model, indicating that the complex psychoacoustic model may not be necessary in an automated level setting application.
Convention Paper 9370

Live Sound Seminar 7 — Saturday, October 31
4:15 pm – 6:00 pm — Room 1A12

SOUND DESIGN FOR THEATER: PRACTICAL AND ARTISTIC CONSIDERATIONS

Presenter: **Nevin Steinberg**, Nevin Steinberg Sound Design, New York, NY, USA

Whether a musical or straight play various time tested elements, as well as emerging technologies, are crucial to a successful theatrical sound design, and those elements are as much artistic and visceral as they are technical. One of Broadway's leading sound designers will discuss many of the considerations and practices of the design process from beginning to completion.

Workshop 21 — Saturday, October 31
4:30 pm – 6:00 pm — Room 1A06

FIBER OPTIC CONNECTOR CHOICES FOR AUDIO*

Chair: **Ronald Ajemian**, Owl Fiber Optics, Flushing, NY, USA

Panelists: *Marc Brunke*, Optocore GmbH, Grafelfing, Germany
Fred Morgenstern, Neutrik USA
Harry (Buddy) Oliver, FiberPlex Technologies, LLC, Elkridge, MD, USA
Warren Osse, Applications/Senior Design Engineer, Vistacom, Inc., Allentown, PA, USA

This AES workshop is designed to educate the user, engineer, technician, and student on the most popular Fiber Optic Connector types that are currently available in the audio/video market place. A brief review will be presented about these connectors. The panel will then discuss user preferences for various audio/video applications. A demo will also be shown on how to terminate (put together) a typical LC fiber optic connector to a glass optical fiber.

**This session is presented in association with the AES Technical Committee on Fiber Optics for Audio*

Product Development 10 — Saturday, October 31
4:30 pm – 6:00 pm — Room 1A13

OPTIMIZING THE POWERED LOUDSPEAKER SYSTEM

Presenter: **Scott Leslie**, Ashly Audio, Webster, NY, USA; PD Squared, Irvine, 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.

Project Studio Expo 13 — PSE Stage
Saturday, October 31 — 4:30 pm – 5:45 pm

AUDIO AS A BUSINESS: BUILDING AND DEVELOPING A CAREER

Presenters: **John Kiehl**, Manhattan Producers Alliance, New York, NY, USA; Soundtrack Studios
Jerome Rossen, Freshmade Music, San Francisco, CA, USA; Manhattan Producers Alliance
Mike Sayre, Independent Film Composer
Carl Tatz, Carl Tatz Design, Nashville, TN, USA
Brian Walker, Audio Director, Leapfrog Enterprises Inc., Emeryville, CA, USA
Richard Warp, Manhattan Producers Alliance, San Francisco, CA; Leapfrog Enterprises Inc., Emeryville, CA, USA

Irons in the Fire: Career and Business Development Mentoring with the Manhattan Producers Alliance

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.

Networked Audio 9 — Saturday, October 31
5:00 pm – 6:30 pm — Room 1A14

HOW WILL AES67 AFFECT THE INDUSTRY?

Chair: **Rich Zwiebel**

Panelists: *Claude Cellier,* Merging Technologies
Andreas Hildebrand, ALC NetworkX GmbH, Munich, Germany
Patrick Killianey
Phil Wagner, Focusrite Novation, Manhattan Beach, CA, USA
Ethan Wetzell, Bosch Communications Systems, Burnsville, MN, USA

There are many audio networking standards available today. Unfortunately, equipment designers and facility engineers have been forced to choose between them to adopt a single platform for an entire operation, or link disparate network pools by traditional cabling (analog, AES/EBU or MADI). AES67 solves this dilemma, providing a common interchange format for various network platforms to exchange audio without sacrificing proprietary advantages. Published in 2013, manufacturers are already showing products with AES67 connectivity this year. Join our panel of six industry experts for an open discussion on how AES67 will impact our industry.

Saturday, October 31 — 5:00 pm — Room 1A19

Technical Committee Meeting on Transmission and Broadcasting

Workshop 22 — Saturday, October 31
5:15 pm – 7:00 pm — Room 1A10

COMPOSING MUSIC FOR SHORT-FORM, INTERSTITIAL, AND EPISODIC TELEVISION

Chair: **Jay Yeary**, Transient Audio Labs, San Antonio, TX, USA

Panelists: *Timo Elliston*, Composer, Bang
Eric Hachikian, Composer, Soundcat Productions
Reid Hall, Turner Studios, Atlanta, GA, USA
Roy Hendrickson
Scott Hull, Masterdisk, New York, NY, USA
Jeff McSpadden, Composer, Jeff McSpadden
Peter Nashel, Composer, Duotone Music Group
Andrea Yankovsky, Kilpatrick, Townsend & Stockton LLP, New York, NY, USA

Composing music for commercials, advertising, interstitial content, and short-form television is an often demanding way to practice the craft because there is never enough time or enough music to meet client demand. This workshop will look at what it takes to be a successful composer in this high-stress, high-demand environment by examining the workflow, tools, rights management, and client management skills of composers who thrive in this world.

This event is part of the Sound for Pictures Track.

Tutorial 19 — Saturday, October 31
5:30 pm – 7:00 pm — Room 1A07

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. The goal is to create an understanding of the basic concepts used in experimental design in order to enable audio professionals to appreciate the possibilities of listening tests.

**This session is presented in association with the AES Technical Committee on Perception and Subjective Evaluation of Audio Signals*

Workshop 23 — Saturday, October 31
5:30 pm – 7:00 pm — Room 1A22

IMMERSIVE AUDIO CODING*

Chair: **Christof Faller**, Illusonic GmbH, Zurich, Switzerland; EPFL, Lausanne, Switzerland

Panelists: *Sascha Disch*, International Audio Laboratories Erlangen
Toni Hirvonen, Dolby
Ton Kalker, DTS

Immersive and object based audio has had a lot of traction recently, by the launches of different formats in cinema and consumer domain. This workshop will focus on the audio coding techniques used for immersive audio. Panelists from the major players in the field will explain their different approaches.

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

Workshop 24 — Saturday, October 31
5:30 pm – 7:00 pm — Room 1A08

MERGED WITH WORKSHOP 22

Recording & Mastering 6 — Saturday, October 31
5:30 pm – 7:00 pm — Room 1A23/24

MASTER CLASS WITH JACK DOUGLAS— RECORDING AEROSMITH, ALICE COOPER, AND JOHN LENNON

Moderator: **Jonathan Pines**, Rupert Neve Designs / Fingerprint Audio, Wimberly, TX, USA

Presenter: **Jack Douglas**

Jack Douglas has produced, engineered, mixed, and written for many of the most influential records in rock history, including John Lennon's *Double Fantasy* (Album of the Year Grammy Winner) and Aerosmith's *Toys in the Attic*, which made Rolling Stone's list of the "500 Greatest Albums." He has worked with Miles Davis, The James Gang, Alice Cooper, Cheap Trick, Patti Smith, Blue Oyster Cult and the New York Dolls. Jack will be playing tracks including some rare recordings by John Lennon, and detailing his techniques on the engineering side as well as the human side of making great records.

Special Event & Career Development
RECORDING COMPETITION—PART 2
Saturday, October 31, 5:30 pm—7:30 pm
Room 1A21

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.

5:30 pm: Category 1—Traditional Acoustic Recording
Judges: Morten Lindberg, David Bowles, Tim Martyn

6:30 pm: Category 4—Sound for Visual Media
Judges: Kris Górski, Elizabeth Fausak, Scott Levine, Shawn Murphy

Saturday, October 31 6:00 pm Room 1A19

Technical Committee Meeting on High Resolution Audio

Special Event
STORIES FOR THE EARS: LIVE AUDIO DRAMA AND NARRATION
Saturday, October 31, 8:00 pm – 10:00 pm
Dolby Laboratories NY Screening Room
1350 Ave of the Americas Main Floor
(Doors open at 7:30 pm – show begins at 8:00 pm)
Limited seating, tickets required.

Fantasy, Fiction, and Fun!

The HEAR Now Festival and SueMedia Productions in conjunction with the Audio Engineering Society (AES) presents an evening of live audio/radio drama along with narrative readings celebrating the art of sonic storytelling.

Hosted by Simon Jones (*Hitchhiker's Guide to the Galaxy*) featuring performances by Audie Award winning and Golden Voice narrators Robin Miles and Barbara Rosenblat, and the award winning NY-based audio drama troupe VoiceScapes Audio Theater.

Sponsored by Hear Now Festival, Walters Storyk Design Group, Dolby Labs, and the AES

Special Event
ORGAN CONCERT BY GRAHAM BLYTH
Saturday, October 31, 8:00 pm – 9:00 pm
Central Synagogue
652 Lexington Avenue at 55th Street, New York

CANCELED

Session P19 Sunday, Nov. 1
9:00 am – 12:30 pm Room 1A08

SPATIAL AUDIO—PART 3

Chair: **Jean-Marc Jot**, DTS, Inc., Los Gatos, CA, USA

9:00 am

P19-1 Estimating the Total Sound Power of Loudspeakers—*Adrian Celestinos, Allan Devantier, Andri Bezzola, Ritesh Banka, Pascal Brunet*, Samsung Research America, Valencia, CA USA

When designing loudspeakers, a number of parameters have to be known. The total radiated sound power is one of these measures. Typically performed in anechoic conditions a large number of measurements are needed for this estimation. It is of interest to know how accurate this estimation is related to the actual radiated power. Two coherent point sound sources separated by 30 cm are simulated in three scenarios. The sound pressure is calculated over discrete points at a distance around a sphere covering the two point sources. The error between estimated and analytical sound power solution is computed. A number of different microphone arrangements are tested. Results suggest that spatial distribution over the sphere and the number of measurements is critical.
Convention Paper 9463

9:30 am

P19-2 Loudness Matching Multichannel Audio Program Material with Listeners and Predictive Models—*Jon Francombe,*[1] *Tim Brookes,*[1] *Russell Mason,*[1] *Frank Melchior*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]BBC Research and Development, Salford, UK

Loudness measurements are often necessary in psychoacoustic research and legally required in broadcasting. However, existing loudness models have not been widely tested with new multichannel audio systems. A trained listening panel used the method of adjustment to balance the loudness of eight reproduction methods: low-quality mono, mono, stereo, 5-channel, 9-channel, 22-channel, ambisonic cuboid, and headphones. Seven program items were used, including music, sport, and a film soundtrack. The results were used to test loudness models including simple energy-based metrics, variants of ITU-R BS.1770, and complex psychoacoustically motivated models. The mean differences between the perceptual results and model predictions were statistically insignificant for all but the simplest model. However, some weaknesses in the model predictions were highlighted.
Convention Paper 9464

10:00 am

P19-3 Dynamic Range and Loudness Control in MPEG-H 3D Audio—*Fabian Kuech,*[1] *Michael Kratschmer,*[1] *Bernhard Neugebauer,*[1] *Michael Meier,*[1] *Frank Baumgarte*[2]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]Apple Inc., Cupertino, CA, USA

Recently the new MPEG-H 3D Audio standard has been finalized. It has been designed for delivery of next generation audio content to the user. In addition to highly efficient immersive audio transmission, MPEG-H 3D Audio allows new capabilities such as personalization and adaptation of the audio content to different use scenarios. It also provides an enhanced concept for loudness and dynamic range control (DRC) to adapt the characteristics of the audio content to the requirements of different playback scenarios and listening conditions. This paper gives a detailed overview of the loudness control and DRC functionality of MPEG-H 3D Audio. Relevant use cases are discussed to exemplify the application of the enhanced DRC and loudness management features.
Convention Paper 9465

10:30 am

P19-4 Implementing the Radiation Characteristics of Musical Instruments in a Psychoacoustic Sound Field Synthesis System—*Tim Ziemer, Rolf Bader*, Universität Hamburg, Hamburg, Germany

A method is introduced to measure the radiation characteristics of musical instruments and to calculate the sound field radiated to an extended listening area. This sound field is synthesized by means of a loudspeaker system to create a natural, spatial instrumental sound. All instruments are considered as complex point sources, which makes it easy to measure, analyze, and compare their radiation characteristics as well as to propagate the radiated sound to discrete listening points. The sound field at these listening points as well as the loudspeaker driving signals to synthesize them are calculated in frequency domain. This makes spatial windowing superfluous and allows for all loudspeakers to be active for any virtual source position. However, this procedure introduces synthesis errors that are compensated for the listener by implementing psychoacoustic methods. The synthesis principle works already with low-order loudspeaker systems such as discrete quadraphonic and 5.1 systems as well as with existing ambisonics and wave field synthesis setups with dozens to hundreds of loudspeakers. Aliasing frequency and synthesis precision are dependent on the number of loudspeakers and the extent of the listening area, not on the distance of adjacent loudspeakers. A listening test demonstrates that the approach creates a listening experience comparable with mono and stereo concerning localization and naturalness of the sound and an increased spaciousness.
Convention Paper 9466

11:00 am

P19-5 New Techniques for Sound Motion and Display in a 52.1 Surround Sound Hall—*Tomás Henriques*, SUNY College at Buffalo, Buffalo, NY, USA

The creation of a 52.1 surround sound system is described with a focus on new strategies for sound motion and localization. Innovative artistic, technical, and research approaches to multichannel electronic music composition, spatial sound design, and sound-localization solutions for the study of auditory perception are introduced. A set of software applications is discussed to illustrate the scope of creative possibilities offered by the surround system as a singular performance and research venue.
Convention Paper 9467

11:30 am

P19-6 Physical Properties of Modal Beamforming in the Context of Data-Based Sound Reproduction—*Nara Hahn, Sascha Spors*, University of Rostock, Rostock, Germany

A sound field captured by a microphone array can be decomposed into plane waves, and auralized by means of sound field synthesis or binaural synthesis. The achievable performance is limited by the spatial resolution of the plane wave decomposition. Typically, the plane wave decomposition is performed with respect to an expansion center. If the expansion center is translated, the accuracy of the plane wave representation decreases. It is thus likely that the reproduced sound field also suffers form artifacts at off-center listening positions. The aim of this paper is to investigate the physical properties of a sound field represented as plane wave decomposition. The sound field is re-expanded with respect to different positions, and the corresponding modal spectra are investigated. This analysis successfully explains the spectral and temporal properties of spatially continuous and discrete modal beamforming.
Convention Paper 9468

12:00 noon

P19-7 The Vertical Precedence Effect: Utilizing Delay Panning for Height Channel Mixing in 3D Audio—*Adrian Tregonning,*[1] *Bryan Martin*[2]
[1]New York University, New York, NY, USA
[2]McGill University, Montreal, QC, Canada

A strong understanding of psychoacoustic cues is necessary for effective 3D sound reproduction, and the vertical aspects of acoustics and psychoacoustics become even more important than for stereo. This study investigated vertical inter-channel time differences (ICTDs) for frontal imaging in the Auro-3D 9.1 loudspeaker configuration. It was found that vertical ICTDs had a significant effect on perceived images, indicating the operation of the precedence effect in the vertical direction. In particular, 5 ms was found to be a threshold for maximal source elevation. Above this threshold, elevation effects were less prominent but ICTDs significantly increased both phantom image width and vertical spread. The techniques established in this study can assist in the creation of effective immersive content.
Convention Paper 9469

Session P20 Sunday, Nov. 1
9:00 am – 10:00 am Room 1A07

FORENSIC AUDIO

Chair: **Rob Maher**, Montana State University, Bozeman, MT, USA

9:00 am

P20-1 Advancing Forensic Analysis of Gunshot Acoustics—*Rob Maher, Tushar Routh*, Montana State University, Bozeman, MT, USA

This paper describes our current work to create the apparatus and methodology for scientific and repeatable collection of firearm acoustical properties, including the important direction-dependence of each firearm's sound field. Gunshot acoustical data is collected for a wide range of firearms using an elevated shooting platform and an elevated spatial array of microphones to allow echo-free directional recordings of each firearm's muzzle blast. The results of this proposed methodology include a standard procedure for cataloging firearm acoustical characteristics and a database of acoustical signatures as a function of azimuth for a variety of common firearms and types of ammunition.
Convention Paper 9471

9:30 am

P20-2 Forensic Sound Analyses of Cellular Telephone Recordings—*Durand R. Begault, Adrian L. Lu, Philip Perry*, Audio Forensics Center, Charles M. Salter Associates, San Francisco, CA, USA

Recordings involving cellular telephones or personal digital assistants ("PDAs") are increasingly the source evidence in audio forensic examinations, compared to recordings originating with other devices such as hand-held digital recorders. On modern PDA cellular telephones recordings can be made either directly to the telephone or transmitted as voice mail messages. The current investigation focuses on differences in the two types of recordings in terms of dynamic range and linearity of levels. Such information can be important for characterizing the distance of sound sources relative to the microphone and are important for understanding transformation of recorded speech and non-speech sounds.
Convention Paper 9472

Tutorial 20 Sunday, November 1
9:00 am – 10:30 am Room 1A06

RIGHTING A WRONG –DISTORTION, FROM RECORDING ACCIDENT TO ROCK AND ROLL REQUIREMENT

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

Engineers obsess – as well we should – about proper gain staging and careful equipment maintenance and calibration, lest distortion should accidentally corrupt our audio. Engineers also embrace—we can't help it—the deliberate use of distortion to express feelings not easily conveyed without it. A distortion accident became a distortion product, and almost overnight, distortion became de rigueur for so many styles of pop music. Alex U. Case details the creation of distortion and the technical and artistic ways to make it most effective.

Tutorial 21 Sunday, November 1
9:00 am – 10:30 am Room 1A10

ADVANCES IN SEMANTIC AUDIO AND INTELLIGENT MUSIC PRODUCTION

Chair: **Ryan Stables**, Birmingham City University, Birmingham, UK

Panelists: *Brecht De Man*, Queen Mary University of London, London, UK
Joshua D. Reiss, Queen Mary University of London, London, UK
Thomas Wilmering, Queen Mary University of London, London, UK

Music Production can be a technically demanding process in which semantic and perceptual characteristics of a mix can have a nontrivial and nonlinear relationship to the parameters available to the producer. In order to improve access to these tools, and to encourage creativity, we can utilize semantic technologies in order to develop intelligent interfaces and production systems. In this tutorial, we present the tools and techniques currently being employed in the field of intelligent music production, along with an overview of the ways in which semantic audio data can be used to make the audio engineer's workflow more intuitive. In addition to this, we introduce the Semantic Audio Feature Extraction (SAFE) DAW plug-ins for the collection, visualization and application of musical semantics data.

Workshop 25 **Sunday, November 1**
9:00 am – 10:30 am **Room 1A21**

LOUDNESS REGULATION: NEW TOOLS TO KEEP THE SPIRIT OF DYNAMICS*

Chair: **Florian Camerer**, ORF, Vienna, Austria

Panelists: *Antoine Hurtado*, Isostem, Paris, France
Thomas Lund
Michael Kashnitz, RTW
Matthieu Parmentier, France TV

Thanks to various laws and recommendations, the loudness regulation is now well spread among TV broadcasters. Now sound mixers and broadcast engineers need for new tools to raise the quality while keeping the dynamics of good programs and gently process others. This workshop will offer an overview of the latest developments concerning loudness measures and process, such as modifying the loudness range, compensating the loudness shift of upmix/downmix, measuring the loudness through an IP network and integrating a smart loudness fader within a web player.

*This session is presented in association with the AES Technical Committee on Transmission and Broadcasting

Archiving 9 **Sunday, November 1**
9:00 am – 10:00 am **Room 1A14**

SETH B. WINNER SOUND STUDIOS: THE FIRST 25 YEARS

Presenter: **Seth Winner**, Seth B. Winner Sound Studios, Inc., Merrick, NY, USA

Seth Winner has been the president and its only employee of SBWSS since April of 1990. His talk will focus on the various projects he has worked on in the commercial industry as well as in the archival community for last two and a half decades. Among the companies he was hired by were SONY, BMG, The New York Philharmonic, The Metropolitan Opera & Guild, The Minnesota Orchestra, Pavilion Records, The Vitaphone Project/Warner Brothers, and many smaller independent labels. As a result, he has garnered one Honorable Mention and three nominations from NARAS/Grammy Foundation. Institutional work includes preservation and studio setups for UMKC/Marr Sound Archives, The Thomas Edison National Historical Park and Harvard University. Mr. Winner was one of two engineers that preserved the vast holdings of the Benny Goodman & Benny Carter Collections housed at the Institute of Jazz Studies, Rutgers University; this project contained over 700 reels of unpublished performances. One of his recent endeavors was the unearthing of the original set of lacquers made during the historic 1938 Benny Goodman Carnegie Hall Concert; he will discuss, among other things, its discovery, digital preservation and restoration. Oh yes, he has been a member in good standing with AES since 1989, and had given talks at its 1990, 1991, and 1999 conventions.

Game Audio 10 **Sunday, November 1**
9:00 am – 10:00 am **Room 1A22**

VIRTUAL REALITY 3D AUDIO—STATE OF THE ART AND VISION OF THE NEAR FUTURE

Presenter: **Edgar Choueiri**, Princeton University, Princeton, NJ, USA

Starting with a brief review of the three main methods for 3-D sound reproduction over loudspeakers: (1) Wave Field Synthesis, (2) Ambisonics, and (3) Binaural audio through two loudspeakers (BAL), we focus on recent advances with the third method. I will show that crosstalk cancellation (XTC) allows BAL to deliver to the listener the necessary cues for real 3-D audio but that it inherently imposes an intolerably high spectral coloration on the audio. I will describe recent breakthroughs, which allow producing optimized XTC filters that impose no spectral coloration. I will then discuss the two other problems that have retarded the commercialization of XTC: the fixed and single sweet spot problems. I will show how the first problem is solved through advanced head tracking; and the second problem is solved using head tracking and phased array speakers, allowing the delivery of high-spatial-fidelity 3D audio to multiple moving listeners in real listening rooms. Following the talk, there will be a demo with playback of recorded music and natural sounds.

Live Sound Seminar 8 **Sunday, November 1**
9:00 am – 10:45 am **Room 1A12**

SOUND DESIGN MEETS REALITY

Presenter: **Andrew Keister**

The best intentions of the sound designer don't always fit in with the venue's design or infrastructure, other departments' needs, or other changes as a production is loaded in and set up for the first time. How the designer's designated representative on site addresses these issues is critical to keeping the overall vision of the sound design and production aesthetics intact while keeping an eye on the budget and schedule.

Product Development 11 **Sunday, November 1**
9:00 am – 10:30 am **Room 1A13**

LOUDSPEAKER MEASUREMENTS

Presenter: **Charles 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: (1) Averaging and S/N; (2) Windowing (both signal acquisition and impulse response windowing); (3) Ground plane measurement techniques; (4) Directivity measurements; (5) Maximum input voltage measurements; (6) Impedance; (7) Alignment of pass bands.

Student Event & Career Development
STUDENT RECORDING CRITIQUES
Sunday, November 1, 9:00 am – 10:00 am
Room 1A18

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

Students! Bring your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on memory-stick, or hard disk, as clearly labeled 24/44.1 KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers!

Sunday, November 1 **9:00 am** **Room 1A20**

AESSC Plenary Meeting

Session P21 **Sunday, November 1**
10:00 am – 11:00 am **Room 1A07**

APPLICATIONS IN AUDIO

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

10:00 am

P21-1 Loudness: A Function of Peak, RMS, and Mean Values of a Sound Signal—*Hoda Nasereddin, Ayoub Banoushi*, IRIB University, Tehran, Iran

Every sound has a loudness recognized by hearing mechanism. Although loudness is a sensation measure, it is a function of sound signal properties. However, the function is not completely clear. In this paper we show that loudness determination as a function of effective mean square (RMS), peak, and average values of a sound signal is possible with an artificial neural network (ANN). We did not access to experimental data, so we produced required data using ITU-R BS.1770 model to train the network. The results show that the loudness can be simply estimated using sound signal physical features and without referring to complex hearing mechanism.
Convention Paper 9473

10:30 am

P21-2 Robust Audio Fingerprinting for Multimedia Recognition Applications—*Sangmoon Lee, Inwoo Hwang, Byeong-Seob Ko, Kibeom Kim, Anant Baijal, Youngtae Kim*, Samsung Electronics Co. Ltd., Suwon, Gyeonggi-do, Korea

For a reliable audio fingerprinting (AFP) system for multimedia service, it is essential to make fingerprints robust to the time mismatch between live audio stream and prior recordings, as well as they should be sensitive to changes in contents for accurate discrimination. This paper presents a new AFP method using line spectral frequencies (LSFs), which are a kind of parameter that capture the underlying spectral shape: the proposed AFP method includes a new systematic scheme for the robust and discriminative fingerprint generation based on the inter-frame LSF difference and an efficient matching algorithm using the frame concentration measure based on the frame continuity property. The tests on databases containing a variety of advertisements are carried out to compare the performances of Phillips Robust Hash (PRH) and the proposed AFP. The test results demonstrate that the proposed AFP can maintain its true matched rate at over 98% even when the overlap ratio is as low as 87.5%. It can be concluded that the proposed AFP algorithm is more robust to time mismatch conditions when compared to PRH method.
Convention Paper 9475
[This paper was not presented]

Session EB6 **Sunday, November 1**
10:00 am – 11:30 am **Foyer**

POSTERS—PART 1

10:00 am

EB6-1 Duplex Panner: Spatial Source Panning for Commercial Music Applications—*Samuel Nacach*, New York University, New York, NY, USA; Abu Dhabi, UAE

The Duplex Panner, introduced at the 137th AES Convention, combines elements from binaural processing, Ambiophonics, the Haas effect, and other widening techniques, to develop a tool, that when listening on headphones, renders preferred stereo imagery over the unprocessed version of the same musical content. To understand if this algorithm can translate to loudspeaker systems without distortion, this paper examines the methodologies employed to achieve spatial panning and how the algorithm was built, how the processing affects the signal, and accordingly what its psychoacoustic implications may be. Through this detailed analysis, we conclude that, unlike other spatial panning techniques, the Duplex Panner is unlikely to be constrained by physical or psychoacoustic limitations in both headphone and loudspeaker systems.
Engineering Brief 220

10:00 am

EB6-2 Development of the Sound Field 3D Intensity Probe Based on Miniature Microphones—*Jozef Kotus, W. Moskwa, Andrzej Czyzewski, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

The engineered measuring probe uses three pairs of miniature microphones coupled. The signals from the microphones after an initial amplification are fed to differential circuits. Due to the required symmetry of the circuit it was necessary to select electronic components very carefully. Moreover, additional digital signal processing techniques were applied to avoid amplitude and phase mismatch. The view of the engineered probe is presented in photographs. Characteristics of the probe measured in an anechoic chamber are attached followed by a discussion of achieved results. The obtained results were compared with the reference USP probe, produced by the Microflown company.
Engineering Brief 221

10:00 am

EB6-3 GaME: Game for Music Education—*Raphaël Marczak,[1] Pierre Hanna,[2] Matthias Robine,[2] Elodie Duru[1]*
[1] Aquitaine Science Transfert, Pessac, France
[2] LaBRI - University of Bordeaux, Talence, France

Music teachers wish that their students spend as much time as possible with their instrument in hands between lessons. By using methods derived from game studies and computer science, GaME offers a ludo-pedagogical solution for keeping young audiences motivated. The motivation is sustained through the use of well-designed involvement mechanisms and real-time feedback about the performances. GaME relies on signal processing algorithms for extracting and comparing musical information, thus enabling an automatic recognition of the chords and notes as actually played by the musician. GaME can be played on computer, tablet, smartphone and even online. GaME includes a score editor and a gameplay metric system to provide feedback helping teachers and parents to create new levels based on specific musical concepts.
Engineering Brief 222

10:00 am

EB6-4 User-Interactive Binaural Rendering Algorithm Using Head-Related Transfer Function and Reverberation—*Hyun Jo, Jaeha Park, Sangmo Son, Sunmin Kim*, DMC R&D Center, Samsung Electronics Co., Suwon, Gyeonggi-do, Korea

This paper introduces an adaptive binaural rendering algorithm that renders sound image into a desired location for user-interactive headphone listening. The proposed algorithm provides steady sound localization during the listener's head movement by minimizing both localization error and timbral degradation caused by filtering HRTF. It is achieved by direct-ambient separation of input channel signal and the corresponding HRTF filtering with desired reverberation to the listener's head position.

By a set of experiments, it is shown that the proposed algorithm provides precise localization.
Engineering Brief 223

10:00 am

EB6-5 Computational "Drop" Detection in Modern Dance Music— *Andrew Ortiz, Colby N. Leider*, University of Miami, Coral Gables, FL, USA

Many of today's popular dance music records are identifiable by a "drop"—a section of the song that is commonly the highest in both listener-perceived and actual signal energy. In this paper we examine several computational methods for locating the exact time at which the drop occurs in a given audio sample. Various metrics are compared and contrasted based on relevant audio signal features. This technology has potential applications within automated DJ software, online music streaming services, computational ethnomusicology research, and more.
Engineering Brief 224

10:00 am

EB6-6 RECUERDAME—*Jorge Sierra Aguilar, Eduard Ramirez Garcia, Sr., Juan David Valencia*, Universidad San Buenaventura, Bogota, Cundinamarca, Colombia

The RECUERDAME is a portable practical tool intended for patients diagnosed with Alzheimer's disease in stages (GDS) 2, 3, 4, 5 (according to Global deterioration scale). This tool provides rehabilitation intervention and cognitive stimulation through a game that the patient plays so that, through the interaction of visual and auditory stimuli, s/he can recognize their family agents through the name requested by the device. This seeks to help reduce the cognitive impairment of a person with Alzheimer's disease. The device will be developed and pilot tested with patients for a generalized projection of treatments for this disease.
Engineering Brief 225

10:00 am

EB6-7 Measurements of Spherical Microphone Array Characteristics in an Anechoic Room—*Tomasz Zernicki, Lukasz Januszkiewicz Marcin Chryszczanowicz, Piotr Makaruk, Jakub Zamojski,* Zylia Sp. z o.o., Poznan, Poland

This paper describes a measurements methodology of spherical microphone array designed and developed for the purpose of soundfield recording. Presented work mainly focuses on the practical aspects of the microphone array impulse response measurements in anechoic environment. The main assumption is that proper acquisition of impulse response coefficients provides crucial information about characteristics of microphones and acoustic shadow of the sphere. Registered impulse responses are further used for generating beam patterns and building a Higher Order Ambisonics microphone.
Engineering Brief 226

10:00 am

EB6-8 Sound Field Recording Using Wireless Digital Distributed Microphone Array—*Marzena Malczewska, Andrzej Ruminski, Piotr Szczechowiak, Tomasz Zernicki*, Zylia sp. z o.o., Poznan, Poland

This paper presents development challenges when building a Wireless Acoustic Sensor Network (WASN) using common IoT devices (Beagle Bone Black). Such system can be used for sound field recording, audio object separation, tracking, etc. In our scenario we focus on recording multiple sound sources in case of a mobile recording studio. Major challenges are related to audio streaming from multiple sensors. Therefore, this paper is focused on analyzing a set of parameters including synchronization accuracy, end to end latency, packet loss, and audio compression efficiency. Experimental results have shown that it is possible to achieve synchronization at the level of micro seconds as well as end-to-end latency below 10 ms using the Opus codec.
Engineering Brief 227

Archiving 10 — Sunday, November 1
10:15 am – 11:15 pm — Room 1A22

SIMPLE. SECURE. SAFE: PARTNERING INITIATIVES WITH THE INTERNET ARCHIVE

Presenter: **B. George**

The Music Locker has been created to save millions of dollars, free up millions of man hours and avoid years of wasted energy. The simple idea is to pool the known digitized copies of every sound recording available, then allow institutions who own a physical copy, access to the digital version. All texts that are a part of any packaging will also be available digitally, OCR searchable. In this way any institution having a copy of a recording can avoid the work of creating a listening copy necessary for in-house use or study. You control access. You can save locally. Simple, secure, safe.

Workshop 26 — Sunday, November 1
10:30 am – 12:30 pm — Room 1A10

APPLICATION OF SEMANTIC AUDIO ANALYSIS TO THE MUSIC PRODUCTION WORKFLOW

Co-chairs: **György Fazekas**, Queen Mary University of London, London, UK
Ryan Stables, Birmingham City University, Birmingham, UK

Panelists: *Jay LeBoeuf*
Bryan PardA

Semantic audio technologies are becoming increasingly more common at various stages in the production chain, with commercial music systems benefiting from the added value imposed by features such as music recommendation and archiving. These technologies generally involve the inference of high-level attributes from an audio signal, based on either statistical features or user-defined labels, with the aim of providing interactions with data. In the field of music production, semantic audio analysis is able to provide abstractions for complex decisions and processes, leading to more intuitive and immersive platforms for audio processing. This paves the way for a wide range of novel systems such automated mixing tools, adaptive audio effects and reduced dimensionality interfaces. In this workshop we review cutting edge semantic audio technologies, in the context of intelligent music production. We aim to provide guidance for deploying semantic audio techniques throughout the music production workflow, while discussing current limitations and future directions.

Recording & Mastering 7 — Sunday, November 1
10:30 am – 11:30 am — Room 1A14

THE GAME HAS CHANGED BUT YOU DON'T KNOW IT: HOW TO MAKE RECORDINGS SOUND GREAT ON STREAMING

Presenter: **Alan Silverman**, Arf! Mastering, New York, NY, USA; NYU/Steinhardt Dept. of Music Technology

Records are engineered to sound their best in the real-world. To accomplish this on services like YouTube, Spotify, Apple Music, Tidal, and Pandora requires a different approach to mixing and mastering because of the way today's streaming services treat audio. Few producers are aware of the game-changing technology under the hood. Recorded music can sound bigger and better than it has in the last decade, ironically, on audiophile systems as well, by applying an understanding of the new technology. Grammy-winning mastering engineer Alan Silverman demonstrates how to harness this potential to the fullest.

Product Development 12 — Sunday, November 1
10:45 am – 12:15 pm — Room 1A13

THE PAST, PRESENT, AND FUTURE OF COAXIAL AND RELATED TRANSDUCERS

Presenters: **Steven Hutt**, Equity Sound Investments, Bloomington, IN, USA
Scott Leslie, Ashly Audio, Webster, NY, USA; PD Squared, Irvine, CA USA

Coaxial transducers have been a part of the audio world since the invention of the two way loudspeaker. They offer a unique performance envelope and compact design that even today present significant advantages over other designs. Variants such as tri-axial, coincident, co-entrant, and others have brought even more advancements to the approach. Over the past few years there has been a resurgence of coaxial products and there have never been more choices available to customers. In this session the presenters will cover the history, the present state, and the future direction. In addition the presenters will discuss the physics behind the approach and why it is a compelling solution to developing great loudspeaker systems.

Broadcast/Streaming Media 13 — Sunday, November 1
11:00 am – 12:30 pm — Room 1A06

TECHNOLOGY AND STORYTELLING: HOW CAN WE BEST USE THE TOOLS AVAILABLE TO TELL OUR STORIES*

Moderators: **David Shinn**, National Audio Theatre Festivals, New York, NY, USA
Sue Zizza, SueMedia Productions, Carle Place, NY, USA

This session will showcase three examples of how the choices we make around technology and the way we use it effect the storytelling process for all entertainment media. With on-site demonstrations by Sue Zizza and David Shinn of SueMedia Productions.

(1) *Microphones and the Voice in Storytelling*. Whether producing an audiobook or narration for a film or game, you want your talent to sound right for the story. This session will begin by looking at how we select microphones for voice talent. Two voice actors will demonstrate how working with different microphones effect their performance abilities.

(2) *Sound Effects: Studio vs. On Location Recordings*. Sound Effects enhance the storytelling process by helping to create location, specific action, emotion and more. Do you have to create every sound effect needed for your project, or can you work with a combination of already recorded elements, alongside studio produced sound effects (foley), or on-location effects, and what are some tips and tricks to recording sound design elements?

(3) *Digital Editing and Mixing*. How can you better manage multiple voice, sound effect, and music elements into "stems," or sub-mixes for better control over final mixing as well as integrating plug-ins for mastering.

Networked Audio 10 — Sunday, November 1
11:00 am – 12:30 pm — Room 1A12

INTEROPERABILITY TESTING

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

Panelists: *Andreas Hildebrand*, ALC NetworX GmbH, Munich, Germany
Peter Stevens
Nicolas Sturmel, Digigram S.A., Montbonnot Saint Martin, France

The AES has now planned two plugfests for AES67 implementers and users. The first plugfest was held in October 2014 at IRT in Munich. A report describing this event was produced and published by the AESSC as AES-R12-2014. The second plugfest is planned for early November in Washington D.C. at NPR headquarters. This workshop will summarize the testing performed and will present results. A panel comprising plugfest participants will answer audience questions and the audience should get a good feel for where AES67 implementation stands.

Live Sound Expo 13 — LSE Stage
Sunday, November 1 — 11:00 am – 11:45 am

VIRTUAL SOUND CHECKS AND PROCESSING IN A NETWORKED ENVIRONMENT

Presenter: **Peter Keppler**
Kevin Madigan
Robert Scovill
Taidus Vallandi

Digital consoles and digital networking offer a natural pathway to simple recording through a single connection, making virtual sound checks an equally simple tool. Further, network appliances are now offering universally applicable virtual effects racks with benefits in pre-production, in enhanced portability, in migrating a studio sound to the stage (including providing recording engineers familiar tools at FOH) and in producing enhance monitor mixes. This session examines the fundamentals of effectively deploying such tools.

Project Studio Expo 14 — PSE Stage
Sunday, November 1 — 11:00 am – 4:00 pm

MIC TO MONITOR

So, you care about your sound, but don't know why you can't quite get the results you strive for!

Learn from our panel of experts from acoustics, high-end audio product design, music recording and production. Supercharge your music or recording career!

Attend the Prism Sound Mic to Monitor seminars at AES New York's Project Studio Expo and discover tips, techniques, ideas, and solutions you can start using right away!

The Mic to Monitor seminars will cover practical aspects of room treatment, loudspeaker placement, loudspeaker technology, microphone technology and microphone selection and positioning, A/D & D/A converters, clocking strategies and some fascinating insights into psycho-acoustics.

The Mic to Monitor Seminar day always ends with a VIP guest speaker. You'll be treated to a talk about their career and professional approach with some exciting playback examples from recent projects. Recent presenters have worked with such luminaries as Paul McCartney, Mary J Blige, Van Morrison, AC/DC and Jay-Z to name but a few.

Make the journey from Mic to Monitor and open your ears to a whole new way of creating your own hit sound! We hope to see you at AES!

11:00 am – 11:40 am
Mastering and Recording with High Performance Analog Electronics

11:40 am – 12:20 pm
Converter Technology and Clocking Issues

12:20 pm – 1:00 pm
Practical Room Acoustics and Treatment

1:00 pm – 1:40 pm
Loudspeaker Technology and Setup

1:40 pm – 2:20 pm
Software and DSP/Plug-In Technology

2:20 pm – 3:00 pm
Microphone Technology and Usage

3:00 pm – 3:40 pm
VIP Guest Speaker on Career Success and Their Secret Sauce!

3:40 pm – 4:00 pm
Q&A

Game Audio 11 — Sunday, November 1
11:30 am – 12:30 pm — Room 1A22

INTERACTIVE MUSIC OF THE LITTLEBIGPLANET FRANCHISE: DISSECTING A COMPLEX, MULTI-COMPONENT SYSTEM

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

This talk will explore the structure and deployment strategies for multiple music tracks composed in a system of dynamic layers for six LittleBigPlanet games. Composer Winifred Phillips has over 11 years of game industry experience, including six games in the famous LittleBigPlanet franchise: LittleBigPlanet 2, LittleBigPlanet Toy Story, LittleBigPlanet Cross Controller, LittleBigPlanet PS Vita, LittleBigPlanet Karting, and LittleBigPlanet 3. This talk will cover issues of importance to composers, audio engineers and sound designers when working with a highly layered music system. Attendees will be alerted to common problems associated with a layered system, and Phillips will reveal useful tips that she learned along the way, and common sense strategies that can be employed for any layered music system, whether it's designed to be modest or large-scale.

Recording & Mastering 8 **Sunday, November 1**
11:30 am – 1:00 pm **Room 1A21**

MASTER CLASS WITH DAVE O'DONNELL —RECORDING JAMES TAYLOR'S *BEFORE THIS WORLD*

Moderator: **Glenn Lorbecki**, Glenn Sound Inc., Seattle, WA, USA

Presenter: **Dave O'Donnell**, James Taylor "Before This World"

Recording James Taylor's *Before This World*

Dave O'Donnell has recorded and mixed many top artists including Keith Richards, Eric Clapton, John Mayer, Keb' Mo', Lyle Lovett, Milton Nascimento, and Ray Charles. His credits include the critically acclaimed GRAMMY-winning *October Road* by James Taylor.

In this Master Class Mr. O'Donnell gives us a detailed look at the making of James Taylor's #1 album, *Before This World*—James' first recording of all-original material in some 13 years. Featuring raw tracks and original session notes, Dave explains the fine points of recording world-class musicians in (sometimes) unconventional surroundings ranging from hotel rooms to James' personal studio "barn."

Student Event & Career Development
STUDENT DELEGATE ASSEMBLY MEETING—PART 2
Sunday, November 1, 11:30 am – 1:30 pm
Room 1A07

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.

Session EB7 **Sunday, November 1**
12:00 noon – 1:30 pm **Foyer**

POSTERS—PART 2

12:00 noon

EB7-1 Measuring Speech Intelligibility Loss in Single-Driver Panel Loudspeakers—*David Anderson, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

The impulse response of a panel loudspeaker with a single moving-coil driver contains ringing due to the resonant frequencies, but the implication of this type of response for intelligible reproduction of speech signals is the subject of some debate. The impulse responses of three examples of such loudspeakers of various sizes and materials were measured in an anechoic environment and compared to that of a conventional speaker. Reverberation effects are clear and calculation of the Speech Transmission Index (STI) confirms a loss of intelligibility; the STI values of the plate loudspeakers are 6% to 13% lower than that of the conventional speaker. Spectrograms of reproduced speech by each plate also show a considerable loss of detail.
Engineering Brief 228

12:00 noon

EB7-2 Vibrational Analysis of Vintage Planar Loudspeakers—*Michael Heilemann, David Anderson, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Recently, there has been a strong interest in the development of flat-panel loudspeakers. The Yamaha JA4001 and the Poly-planar P20 represent two early attempts at commercializing the technology. The responses of both loudspeakers were analyzed using a laser vibrometer. The scans for each panel depict sharp peaks in the frequency response, which correspond to resonant modes. The presence of additional modes is similar to the effect of cone breakup in traditional loudspeakers. Impulse response measurements show that low-frequency modes are highly reverberant. Studying these early planar loudspeakers can provide valuable insight for the further development of such technology.
Engineering Brief 229

12:00 noon

EB7-3 A Database of Loudspeaker Polar Radiation Measurements—*Joseph G. Tylka, Rahulram Sridhar, Edgar Choueiri*, Princeton University, Princeton, NJ, USA

Anechoic directivity data for a variety of loudspeakers have been measured and compiled into a freely available online database, which may be used to evaluate these loudspeakers based on their directivities. The measurements are illustrated through four types of plots (frequency response, polar, contour, and waterfall) and are also given as raw impulse responses. Two sets of directivity metrics are defined and are used to rank the loudspeakers. The first set consists of full and partial directivity indices that isolate sections of the loudspeaker's radiation pattern (e.g., forward radiation alone) and quantify its directivity over those sections. The second set quantifies the extent to which the loudspeaker exhibits constant directivity. Measurements are taken, in an anechoic chamber, along horizontal and vertical orbits with a (nominal) radius of 1.6 m and an angular resolution of five degrees.
Engineering Brief 230

12:00 noon

EB7-4 Single-Channel Sound Source Separation Using NMF with Sparseness Constraints—*Shijia Geng, Colby N. Leider*, University of Miami, Coral Gables, FL, USA

While challenging, sound source separation is a task that has many practical applications in audio signal processing. In this paper three sound files with two sources in each were separated using the non-negative matrix factorization (NMF) approach, with and without sparseness constraints. The results showed that adding sparseness constraints had no effect when separating drums and bass guitar but had better performances when separating piano and drums, and piano and bass guitar.
Engineering Brief 231

12:00 noon

EB7-5 Speech Enhancement Using Characteristics of Periodic Noise for MRI Environment—*Sangchul Ko, Aran Cha, Youngsang Lee, Yoon Jae Lee, Hyun-Woo Kim*, DMC R&D Center, Samsung Electronics Co., Republic of Korea

This study describes noise reduction method using characteristic of periodic noise for speech communication between operators and patients during magnetic resonance imaging (MRI) scan. After the periodicity of MRI noise is acquired in intervals of non-speech activity, spectral subtraction method is applied to each frame using obtained periodic noise candidate. Simulations are conducted by using three scan sequences, experiments in a MRI device show similar performances to simulation results. As a result, the MRI noise is reduced successfully under minus signal to noise ratio (SNR) with qualitative recovered speech. In addition, the proposed method using a single channel input has competitive performances compared to others using two channel inputs.
Engineering Brief 232
[This eBrief was not presented but is available in the E-Library]

12:00 noon

EB7-6 Noise Robust End-Point Detection Algorithm Using Human Auditory and Pronunciation Characteristics —*Jae-Hoon Jeong, Min Seok Kwon, Seungyeol Lee, Young Woo Lee, Haruyuki Mori, Namgook Cho, Jae Won Lee*, DMC R&D Center, Samsung Electronics Co., Suwon, Gyeonggi-do, Korea

A noise robust end point detection algorithm is proposed that could be used in real environment speech recognition. Inaccurate end point detection brings not only speech recognition performance reduction but also users' tiredness. EPD algorithms based on energy level change or speech presence probability are vulnerable to high energy noises. After reducing much noise by auditory filter, one of human speech pronunciation characteristic, syllabic rate is used for checking if there is still speech component or not. The proposed algorithm shows much better performance in real environments like TV sound noise, café noise, etc.
Engineering Brief 233

12:00 noon

EB7-7 Evaluation of Separation Techniques for Musical Instrument Recordings Using Microphone Array in a Rehearsal Room—*Tomasz Zernicki, Lukasz Januszkiewicz, Marcin Chryszczanowicz, Piotr Makaruk, Jakub Zamojski*, Zylia sp. z.o.o., Poznan, Poland

This paper describes the comparison of two different approaches for fast and simple sound tracks separation of multiple musical instrument records. An uniform circular microphone array is used in a rehearsal room for recording of musical instruments being played simultaneously. A beamforming algorithm and additional signal post-processing is used to separate the individual instrument tracks. The separated tracks are compared to tracks recorded with dedicated highly directive microphone (shotgun). The objective evaluation of results is made by calculation of signal-to-interference ratio (SIR). Additionally the subjective test are performed where listeners had to asses the quality in terms of level of interference signals.
Engineering Brief 234

12:00 noon

EB7-8 Monitoring and Authoring of 3D Immersive Next Generation Audio Formats—*Peter Pörs*, Junger Audio GmbH, Berlin, Germany

The next generation immersive audio formats will require changes in the audio production workflow. Monitoring the audio along with authoring and verifying of dynamic metadata will become a new challenge. New procedures for managing object based encoded content the same way as for personalization of services through the selection of alternative audio objects (such as commentator languages) needs to be established. Loudness control during production and the loudness definition for the final output formats are other topics to consider. A Monitoring & Authoring Unit must be compatible with upcoming immersive multichannel 3D audio formats and should offer a platform to host all the emerging immersive 3D audio encoding formats from different vendors.
Engineering Brief 235

Live Sound Expo 14 **LSE Stage**
Sunday, November 1 **12:00 noon – 12:45 pm**

SHED AND ARENA LOUDSPEAKER OPTIMIZATION: PULLING BIG SHOWS TOGETHER

Presenter: **Bernie Broderick**

Beginning with off-line prep and carrying on through the loudspeaker hang and on to sound check, this end-user focused session uses a case study approach to walk through the steps of configuring and optimizing a rig for large audiences in amphitheaters and arenas.

Workshop 27 **Sunday, November 1**
12:30 pm – 2:30 pm **Room 1A14**

ANALYZING AND RECORDING SOUNDSCAPES: THEORY AND APPLICATIONS

Co-chairs: **Durand R. Begault**, Audio Forensic Center, Charles M. Salter Associates, San Francisco, CA, USA
Agnieszka Roginska, New York University, New York, NY, USA

Panelists: *Alex Case*, University of Massachusetts Lowell, Lowell, MA, USA
Charlie Mydlarz, New York University, New York, NY, USA
Tae Hong Park, New York University, New York, NY, USA
Jessica Schwartz, UCLA, Los Angeles, CA, USA

The total palette of sounds of the environment at a particular location that bring meaning and identity is known as its soundscape. The particular immersive quality of soundscapes results from humans, animals, and nature and is of great interest from musical, naturalistic, community, legal, and political perspectives. This event will bring together experts in the field to address their particular work and its implications for the audio engineering world and the world community at large.

Spatial Audio Demo 9 **Sunday, November 1**
12:30 pm – 1:30 pm **Room 1A18**

KRAFTWERK—HOW TO CREATE AN IMMERSIVE/3D AUDIO POP MIX IN DOLBY ATMOS FOR A COMMON BLU-RAY RELEASE

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

It³s one thing to move audio objects around the listener like a spaceship in a movie application. It's a totally different approach to create musical landscapes that involve the listener emotionally more than ever before with immersive/3D audio. An example of this is the new Kraftwerk Blu-ray production "Kraftwerk 3D." Production strategies for a common pop Blu-ray release in Dolby Atmos will be shown and explained.

Archiving 11 **Sunday, November 1**
1:00 pm – 2:30 pm **Room 1A10**

AUDIO PRESERVATION THROUGH IMAGING: IRENE & THE TALKING DOLL TRANSFERS

Presenters: **Peter Alyea**, Library of Congress
Jerry Fabris, Thomas Edison National Historical Park
Carl Haber, Lawrence Berkeley Labs
Mason Vander Lugt, Library of Congress

Sound recordings that were lost to history are now accessible again. This panel will take a look at the development of non-invasive technology, its use, and applications for special collections. This panel will begin with IRENE's creator, Carl Haber, who will provide an overview of minimally invasive and automated approaches to recorded sound preservation and access. Peter Alyea leads the IRENE project at Library of Congress in its Preservation Reformatting Division. Mason Vander Lugt is a preservationist who has worked extensively with IRENE. Then Jerry Fabris will present how IRENE was used to capture the audio of the Edison Talking Doll cylinders, the world's earliest commercial recordings.

Live Sound Seminar 9 **Sunday, November 1**
1:00 pm – 2:45 pm **Room 1A12**

LIVE SOUND DESIGN FOR TV: "THE TONIGHT SHOW WITH JIMMY FALLON"

Chair: **Duncan Edwards**

Panelists: *Mac Kerr*
Matt Kraus
Simon Matthews

House sound reinforcement for live broadcast has its own set of unique requirements where one of the primary goals is that it must not interfere with the audio for broadcast. Duncan Edwards is the in-studio sound consultant for NBC and along with his staff will discuss the primary considerations, subtleties and design for the "Tonight Show with Jimmy Fallon" among others.

Live Sound Expo 15 **LSE Stage**
Sunday, November 1 **1:00 pm – 1:45 pm**

CHOOSING THE RIGHT VOCAL MIC

Presenters: **Mark Frink**
Peter Keppler
Kevin Madigan

While there are tried and true mics clinched by singers across the world, selecting the best mic for a vocalist involves more than snatching the most familiar mic off the shelf. This session talks microphone fundamentals (including polar-patterns and capsule construction), matching performance with a given voice and singing style, as well as tips for working with vocalists.

Tutorial 22 **Sunday, November 1**
1:30 pm – 2:30 pm **Room 1A22**

RETHINKING AUDIO PRODUCTION: COMMON MISTAKES THAT HINDER PROGRESS*

Presenter: **Peter Dowsett**, Audio Production Tips, London, UK

This tutorial will focus around educating students or amateurs, so that they re-evaluate what is really important when recording music. Discussion topics include: • It's about how you use the gear, not the gear itself • Proper gain-staging (both analogue and digital scales) • Cultivating vision • The importance of monitoring environment • Do everything for a reason (rather than because "that is what should be done") • Good recordings start with good songs and arrangements • Fix problems as early as possible in the chain.

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

Session P22 **Sunday, Nov. 1**
2:00 pm – 4:00 pm **Room 1A08**

SOUND REINFORCEMENT

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

2:00 pm

P22-1 **From Studio to Stage—***Guillaume Le Hénaff,* Conservatory of Paris, Paris, France

To convert studio produced music into a live concert is a key issue for a lot of artists. Studio work is often a long-term undertaking during which everything is subject to attentive decisions, e.g., instruments, performers, recording venues, microphones. When performing songs from a record in concert, all these decisions have to be reviewed or at least questioned. Indeed, studio and stage are two really different production contexts and differ on so many points that artists often change their arrangements, line-up or even the form of their songs. However, live sound engineers may be expected to reproduce the sound quality and aesthetics of the record. In this paper we propose solutions regarding the switchover from studio to stage to provide artists and engineers with useful tools when designing the sound of a studio album-inspired live show. Specifically, we explain why and how performing music is different in concert than in studio, we detail types of microphones that are suited to both recording and sound reinforcement applications and we take an inventory of miking tricks and mixing techniques like Virtual Soundcheck that offer a studio workflow to Front of House engineers.
Convention Paper 9476

2:30 pm

P22-2 **Some Effects of Speech Signal Characteristics on PA System Performance and Design—***Peter Mapp*, Peter Mapp Associates, Colchester, Essex, UK

Although the characteristics of speech signals have been extensively studied for more than 90 years, going back to the work of Harvey Fletcher and Bell Labs pioneering research, the characteristics of speech are not as well understood by the PA and sound reinforcement industries as they perhaps should be. Significant differences occur in both the literature and between international standards concerning such basic parameters as speech spectra and level. The paper reviews the primary characteristics of speech of relevance to sound systems design and shows how differences within the data or misapplication of it can lead to impairment of system performance and potential loss of intelligibility. The implications for compliance with various National and International Life Safety standards are discussed.
Convention Paper 9477

3:00 pm

P22-3 **Directivity-Customizable Loudspeaker Arrays Using Constant-Beamwidth Transducer (CBT) Overlapped Shading—** *Xuelei Feng,*[1] *Yong Shen,*[1] *D.B. (Don) Keele, Jr.,*[2] *Jie Xia*[1]
[1]Nanjing University, Nanjing, Jiangsu, China
[2]DBK Associates and Labs, Bloomington, IN, USA

In this work a multiple constant-beamwidth transducer (Multi-CBT) loudspeaker array is proposed that is constructed by applying multiple overlapping CBT Legendre shadings to a circular-arc or straight-line delay-curved multi-acoustic-source array. Because it has been proved theoretically and experimentally that the CBT array provides constant broadband directivity behavior with nearly no side lobes, the Multi-CBT array can provide a directivity-customizable sound field with frequency-independent element weights by sampling and reconstructing the targeted directivity pattern. Various circularly curved Multi-CBT arrays and straight-line, delay-curved Multi-CBT arrays are analyzed in several application examples that are based on providing constant Sound Pressure Level (SPL) on a seating plane, and their performance capabilities are verified. The power of the method lies in the fact only a few easily-adjustable real-valued element weights completely control the shape of the polar pattern that makes matching the polar shape to a specific seating plane very easy. The results indicate that the desired directivity patterns can indeed be achieved.
Convention Paper 9478

3:30 pm

P22-4 **A Novel Approach to Large-Scale Sound Reinforcement Systems—***Mario Di Cola,*[1] *Alessandro Tatini*[2]
[1]Audio Labs Systems, Casoli (CH), Italy
[2] K-Array S.r.l., Florence, Italy

An innovative approach to vertical array technology in large-scale sound reinforcement is presented. The innovation introduced consists in mechanical arrangement of the array as well as DSP processing for computer assisted coverage optimization. Beyond these innovations, a different form factor of the vertical array elements and the unusual acoustic principle of dipole are also involved as well as an alternative mechanical aiming method. The paper presents a synthesis of this innovative concept supported by detailed descriptions, test measurement results and proven results from real world applications that have been done.
Convention Paper 9479

Broadcast/Streaming Media 14 **Sunday, November 1**
2:00 pm – 5:00 pm **Room 1A19**

SBE CERTIFICATION EXAMS

The Society of Broadcast Engineers Certification Program, established in 1975, is a service contributing to the professional development of the broadcast engineer and advancement of the field of broadcast engineering. The SBE Program of Certification is administered by the SBE National Certification Committee, which continually develops exam questions based on changes in the industry and technology. Levels of SBE certification vary based on an individual's experience. Membership in the SBE is not required to hold SBE certification. Advance registration by Oct. 2 is required to take an SBE certification exam at the convention.

Recording & Mastering 9 **Sunday, November 1**
2:00 pm – 3:30 pm **Room 1A21**

RAW TRACKS 2.0—ANATOMY OF: A COUNTRY HIT— KACEY MUSGRAVES' "SAME TRAILER DIFFERENT PARK"

Moderator: **Jim Kaiser**, CEMB / Belmont University, Nashville, TN, USA

Panelists: *Charlie Brocco*
Ryan Gore
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA

The first record by new artist Kacey Musgraves on Mercury Nashville (Same Trailer Different Park) has been certified Gold and received critical acclaim, including four 2014 Grammy nominations, with wins for Best Country Song and Best Country Album. It is NOT your "typical country record," and the engineering team behind it will share the unique story about its creation from the studio through to finished product. Join the Nashville team of Charlie Brocco (tracking), Ryan Gore(recording/mixing), and Andrew Mendelson (mastering) as they detail how this artist-driven project unfolded, with behind-the-scenes pictures, video, and playback of tracks & songs at various stages of their journey.

Special Event
CHRISTIAN MCBRIDE - LEADER - SIDEMAN: THE LIFE OF A JAZZ MUSICIAN
Sunday, November 1, 2:00 pm – 3:30 pm
Room 1A06

Moderator: **Harry Weinger**, Universal Music Enterprises (UMe), New York, NY, USA; New York University, New York, NY, USA

Presenter: **Christian McBride**

Accustomed to hosting gigs, radio shows, and podcasts Christian McBride, a four-time GRAMMY winner and one of the great bass players and bandleaders of our time, will be in conversation with Harry Weinger. Sit in for an exploration of Mr. McBride's dual careers—as bandleader and sideman, composer and interpreter, lifelong student and educator – as well as Live At The Village Vanguard, the Christian McBride Trio's latest album, among other recordings.

Mr. McBride has recorded 13 albums as a leader, and hundreds of additional recordings in several genres, collaborating with everyone from Chick Corea, Pat Metheny and Bruce Hornsby, to James Brown, Sting, The Roots, et al. He is co-founder and Artistic Chair of Jazz House Kids; artistic director of the Montclair Jazz Festival, produced by Jazz House Kids; artistic advisor of the James Moody Democracy of Jazz Festival, which debuted in 2012 at the New Jersey Performing Arts Center; and he currently serves as NJPAC's Jazz Advisor.

Mr. Weinger, as a producer and executive with Universal Music Enterprises has overseen hundreds of reissues and compilations. He is a two-time GRAMMY(R) winner, was recently a consultant for both Get On Up, the James Brown biopic, and Mr. Dynamite: The Rise Of James Brown, a documentary directed by Alex Gibney, and is an adjunct professor at NYU's Clive Davis Institute for Recorded Music

Live Sound Expo 16 **LSE Stage**
Sunday, November 1 **2:00 pm – 2:45 pm**

RF COORDINATION ON THE ROAD

Presenters: **Ike Zimbel**

Get a look into the working life of a touring RF engineer. Our guest engineer, just off a five-month road haul, compares the RF environments in North American arenas, shares a practical approach to working with wireless microphones, instruments and monitors in those environments, and discusses wireless best practices.

Workshop 28 **Sunday, November 1**
2:30 pm – 4:00 pm **Room 1A13**

CREATING SPATIAL AUDIO CONTENT FOR HEADPHONE LISTENING

Presenters: **Tom Ammermann**, New Audio Technology GmbH, Hamburg, Germany
Bob Schulein, RBS Consultants, Schaumburg, IL, USA

A large amount of music program material is being produced with two-channel loudspeaker listening in mind, which by its nature does not create the same tonal balance and spatial perspective as headphone listening. Conversely program material produced for headphone listening does not produce the same tonal and spatial perspective as for loudspeaker listening. Dealing with this conflict of purpose represents a production challenge for those who appreciate spatial audio fidelity. Sounds reproduced by means of headphones have the unique capability of replicating the wide variability of sounds heard by humans in real-word situations. This fact allows the possibility of creating the auditory portion of any immersive experience with just two audio channels. This is not the case for sound fields created by traditional loudspeakers where left ear/right ear cross talk inhibits independent control of the sound field for each ear. In addition experience has shown that sound fields with too many competing spatial elements can reduce the spatial impact of a production. This is due to the fact that humans have difficulty separating the positions of simultaneous sounds coming from different locations. This has motivated production techniques that separate differing spatial sound elements in time, so as to increases the richness of the experience. This workshop will focus on creating spatial audio musical experiences by both acoustic capture and synthesis techniques for headphone listening. A group wireless headphone system will be used for a variety of demonstrations supporting the workshop.

Live Sound Seminar 10 **Sunday, November 1**
2:45 pm – 4:30 pm **Room 1A12**

LOUDSPEAKER DEVELOPMENTS AND THEIR IMPACT ON THE INDUSTRY

Chair: **Dave Rat**

Panelists: *Dave Gunness*
Ralph Heinz
Dave Natale

Three Daves and a Ralph lend their experience to the discussion. Participants, Dave Rat, owner of Rat Sound and FOH mixer for the Red Hot Chile Peppers, Dave Gunness, speaker designer formerly with EV and EAW now partner in Fulcrum Acoustics, Dave Natate, FOH mixer for The Rolling Stones, Ralph Heinz speaker designer at Renkus-Heinz.