AES 141ST CONVENTION PROGRAM

SEPTEMBER 29–OCTOBER 2, 2016

LOS ANGELES CONVENTION CENTER, CA, USA

The Winner of the 141st AES Convention Best Peer-Reviewed Paper Award is:

The Relationship between the Bandlimited Step Method (BLEP), Gibbs Phenomenon, and Lanczos Sigma Correction— *Akhil Singh, Will Pirkle*, University of Miami, Coral Gables, FL, USA

Convention Paper 9670

To be presented on Saturday, October 1, *in Session 22—Signal Processing—Part 3*

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

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

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

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

(a) The paper was accepted for presentation at the AES 141st Convention.

(b) The first author was a student when the work was conducted and the manuscript prepared.

(c) The student author's affiliation listed in the manuscript is an accredited educational institution.

(d) The student will deliver the lecture or poster presentation at the Convention.

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The Winner of the 141st AES Convention Student Paper Award is:

Physically Derived Synthesis Model of an Aeolian Tone—*Rod Selfridge, Joshua D. Reiss, Eldad J. Avital, Xiaolong Tang*, Queen Mary University of London, London, UK

Convention Paper 9679

To be presented on Sunday, October 2, in Session 23—Signal Processing—Part 4

Session P1 **Thursday, September 29**
9:00 am – 10:30 am **Room 403A**

TRANSDUCERS—PART 1

Chair: **Alex Voishvillo**, JBL/Harman Professional, Northridge, CA, USA

9:00 am

P1-1 Holographic Nearfield Measurement of Loudspeaker Directivity—*Wolfgang Klippel, Christian Bellmann*, Klippel GmbH, Dresden, Germany

The acoustical output of loudspeaker systems is usually measured in the far field under anechoic conditions requiring a large measurement distance and special treatment of the room (absorbing room boundaries, air condition). Also the measurements of directivity characteristics at sufficient angular resolution are also very time consuming. The measurement in the near field of the sound source provides significant benefits (dominant direct sound, higher SNR, less climate impact) but requires a scanning process and a holographic processing of the measured data. This paper describes the theoretical basis of the new measurement technique and the practical consequences for the loudspeaker diagnostics.
Convention Paper 9598

9:30 am

P1-2 Fully Coupled Time Domain Simulation of Loudspeaker Transducer Motors—*Andri Bezzola, Pascal Brunet*, Samsung Research America, DMS Audio, Valencia, CA, USA

We present a novel time-dependent simulation method to calculate the response of a loudspeaker motor. The model allows for the simulation of complex signals and predicts the large-signal behavior including motor nonlinearities using only the motor geometry and material parameters without the need to measure physical samples. The transient large-signal simulation is made possible by the implementation of a moving-mesh algorithm for the displacement of the voice coil. Two motor geometries are simulated with different input signals, ranging from simple sine to complex random signals. The method provides previously unavailable insight into effects of flux modulation. The results are validated against a lumped parameter

model and experimental measurements. The presented method can be used to compare different motor geometries before the prototyping stage, which is a useful tool for loudspeaker transducer engineers.
Convention Paper 9599

10:00 am

P1-3 Necessary Delay and Non-Causal Identification for Online Loudspeaker Modelling Considering Voice Coil Inductance—*Rong Hu, Jie Su*, Cirrus Logic, Austin, TX, USA

The authors revisit the topic of online electrical system identification with adaptive filters for dynamic loudspeakers and investigate into the causality of the plant loudspeaker systems to be identified. The effects of the non-causal portion of plant system are analyzed and simulated results establish the link between the non-causality in impulse response and the voice coil inductance. The improvements of introducing necessary delay to the desired signal are proposed to enable the characterization of such non-causality. The proposed architecture with small delay extends the working bandwidth of online loudspeaker system identification and improves the accuracy of existing adaptive identification schemes without delays, which are traditionally restricted to run at low frequency bands.
Convention Paper 9600

Session P2 Thursday, September 29
9:00 am – 11:00 am Room 409B

EDUCATION

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

9:00 am

P2-1 Understanding Project-Based Learning in the Audio Classroom: Using PBL to Facilitate Audio Storytelling—*Kyle Snyder*, Ohio University, Athens, OH, USA

One of the more prevalent buzzwords in education today, project-based learning is a natural fit for the audio engineering classroom. With students that thrive by working toward a common goal or "learning by doing," this constructivist framework is worth examining as implemented by educators. This paper discusses project-based learning as implemented in an audio engineering classroom to facilitate audio storytelling and provides recommendations for faculty looking to implement project-based learning into their curriculum.
Convention Paper 9601

9:30 am

P2-2 The Graduate Audio Database Project: A Look into Pragmatic Decision-Making in Postgraduate AE Curricular Design—*Daniel A. Walzer*, University of Massachusetts – Lowell, Lowell, MA, USA

This paper reports on the first phase of a comparative project to build a Graduate Audio Database (GAD) of North American colleges and universities (N =66) offering 86 Master's degrees. Data came from available information drawn from institutional websites, course descriptions, professional and educational organizations, and targeted keyword searches. Each credential received categorization across seven areas. Results indicate that 38% of institutions list the Master of Fine Arts (MFA) as the most common degree offering and 92% of universities emphasize the creative aspects of audio and sound. This paper explores the role of action research to build an exploratory review of graduate-level audio degrees and reflect on how decision-making affects postgraduate curricular mapping.
Convention Paper 9602

10:00 am

P2-3 Equalizing Frequencies: Gender and Audio Technology in American Higher Education—*Rosanna Tucker*, University of Southern California, Los Angeles, CA, USA

Unequal gender representation pervades audio engineering and production programs in higher education in the United States but has hitherto been the subject of limited discourse. This paper intends to corroborate survey data and observations from audio-technology professors and students with research concerning gender and academic performance in audio-technology and other disciplines displaying similar gender inequities. Research pertaining to female science, technology, engineering, arts, and math (STEAM) majors suggests a number of strategies to assist educators in affecting more inclusive, equitable classroom cultures. The author focused primarily on the dearth of female audio-technology professors, gender as a factor in classroom participation, and extracurricular student culture, and the impact of gendered expectations concerning music and audio-technology during the precollege years.
Convention Paper 9603
[This paper was not presented]

10:30 am

P2-4 Interactive Music on the Music's Terms: Creative Technical Solutions to Musically Portray the Nobel Prize—*Jan-Olof Gullö, Hans Lindetorp, Johan Ramström*, Royal College of Music, Stockholm, Sweden

This article describes an artistic music production project that was a part of two exhibitions on the Nobel Laureates in 2014 and 2015: Nobel Creations. The exhibition's aim was to creatively interpret the various Nobel prizes in design, fashion, and music and specially developed software was used to compose and design interactive music that interacted with actions from the visitors in the museum hall. The music was played out without interruption or without repeating itself during the four months each exhibition lasted.
Convention Paper 9604
[This paper was not presented]

Tutorial 1 Thursday, September 29
9:00 am – 10:30 am Room 402AB

THE ART, STUDY, AND PRACTICES OF LISTENING

Presenter: **Martha DeFrancisco**, McGill University, Montreal, QC, Canada

At McGill University an interdisciplinary seminar brought together scientific researchers as well as experts in many areas that consider "listening" as one of their fundamental activities. The course explored how learned auditory skills and fine discrimination constitute an essential requirement for the practice of various professions. While critical listening of music was the guiding motif, invited speakers lead the class in their exploration of "listening" as a main component in a variety of areas of human life, while recognizing connections. Topics include critical listening in music performance, sound recording and engineering, record production, music instrument making, as well as "listening" in psychoacoustics, perception and cognition, in the neurosciences, in education, psychiatry, media studies, urban planning, and in the preservation of oral tradition through story telling.

Broadcast/Streaming Media 1 **Thursday, September 29**
9:00 am – 10:30 am **Room 408A**

IMMERSIVE AUDIO ABSORBING RADIO AND TV AUDIENCES IN 2016 AND BEYOND

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

Panelists: *Brian Long*, Skywalker Sound
Robert Margouleff
Matt Marrin
Chris Pelonis, Musician, studio designer/owner, Santa Barbara, CA, USA

In an industry deluged by acronyms, Immersive Audio appears to have leapfrogged the trend (alt-hough I.A. or 3D Audio, may suffice). As with Surround Sound, Quad Sound, and their various 3.1 – 5.1 – 7.1…, etc., iterations, much of the noise made by these new innovations is focused on hype rather than on specific real world listener/viewer needs or actual desires. That said, I.A. sys-tems for producing, distributing and receiving this new sound experience do exist, they work and, they are proliferating.

This Panel Discussion will feature four experts in radio and TV broadcast technology, systems de-velopment /integration, and studio design. Panelists include: Grammy Award-winning engineers Robert Margouleff and Matt Marrin, and award-winning Studio Designer, Chris Pelonis. Moder-ated by WSDG Founding Partner, John Storyk, the discussion will explore studio and gear design issues both acoustic and technological. Areas to be covered will include: What needs to be done to equip, upgrade, and future proof existing studios for the production, broadcast and streaming of Immersive Audio? What creative and/or technical issues differentiate traditional speaker perfor-mance from headphone/earbud reception for I. A. What loudness issues need to be addressed, e.g., Noise and quietness, Internal room responsiveness, Speech vs. music, Reflection and Absorption.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Networked Audio 1 **Thursday, September 29**
9:00 am – 10:30 am **Room 408B**

ROLLING OUT AES67 INTO REAL-WORLD APPLICATIONS

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

The AES67 Standard on High performance Streaming Audio-over-IP Interoperability has been published in September 2013. Since then, first applications with AES67-compatible devices have been projected and put into operation. This session provides hints and insights on rolling out AES67 AoIP networking into real-world installations. The first part of the presentation will discuss applicability of AES67 and its related network requirements in general. In the second part some real world use cases based on AES67 interoperability will be presented and their operational benefits as well as the difficulties experienced during roll-out will be examined. The presentation closes with a brief status and outlook on the on-going standard maintenance work and the current industry support of AES67.

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

Recording & Production 1 **Thursday, September 29**
9:00 am – 10:30 am **Room 501ABC**

RESTORATION—TECHNICAL ARCHIVAL, FORMATS, METADATA, GRANT PROPOSALS AND RELEASE

Moderator: **Jessica Thompson**, Coast Mastering, Berkeley, CA, USA

Panelists: *Jamie Howarth*, Plangent Processes, Nantucket, MA, USA
Cheryl Pawelski, Omnivore
Steve Rosenthal, Reissue Producer

Sound archives and reissue record labels have overlapping interests in preservation of archival audio recordings. This panel brings together archivists, engineers, and record producers to explore how archives and labels can collaborate on preservation, presentation, and monetization of archival sound recordings. Topics to be discussed include how that collaboration can be leveraged to refine and promote best practices in digitization, metadata encoding, restoration ,and preparation for high resolution comercial release. Moderated by restoration and mastering engineer Jessica Thompson, this panel will also look at how libraries, archives, and record labels can form fruitful private/public relationships and the role record labels can play in initiating and potentially funding preservation of archival collections.

Sound Reinforcement 1 **Thursday, September 29**
9:00 am – 10:30 am **Room 404AB**

NUMERICAL LOUDSPEAKER ARRAY OPTIMIZATION —THE NEXT GENERATION LOUDSPEAKER SYSTEM

Chair: **Jim Risgin**, OSA International, Inc., Nashville, TN, USA

Panelists:l *Paul Bauman*, MUSIC
Dave Gunness, Fulcrum Acoustics
Dom Harter, Martin Audio
John Monitto, Meyer Sound
Howard Page, Clair Global
Janko Ramuscak, d&b audiotechnik

2016 brought Numerical Optimization to the forefront. Over 50% of the Systems providing reinforcement for today's audiences either use or have the ability to use computational based optimization technologies to deliver a superior audio experience to the audience. Call it what you will but Array Processing, Adaptive Technology or Numerical Optimization- the Technology is changing how we listen and how we deliver the audience experience. Experts in design and implementation of Optimized Systems will discuss and explain the process – working to demystify and improve the everyday users understanding of what this does for them and their audience.

Game Audio 1 **Thursday, September 29**
9:30 am – 10:30 am **Room 406AB**

TALES OF AUDIO FROM THE THIRD DIMENSION!

Presenter: **Scott Selfon**, Sr. Development Lead, Microsoft Corporation

High fidelity and non-repetitive sound effects, music, dialogue, and ambience are only the beginning of a compelling in-game sound experience. Spatialization (hearing sounds from where they are perceived to occur) is increasingly critical for traditional gameplay and virtual/augmented/mixed reality experiences alike. This talk surveys real-time 3D sound manipulation techniques that marry psychoacoustic theory with practical-for-real-time applied engineering. Topics will include dynamic simulation of position, distance, interaction with game geometry, environmental reverberation, and more. We'll offer a primer on topics both technical (HRTFs and other processing; spatial formats; middleware integration) and creative (placing non-diegetic audio in a mixed-reality game; mixing techniques; evolving best practices for spatial implementations for headphone and speaker solutions).

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

Special Event
BRUNCHING WITH BONZAI
Thursday, September 29, 9:30 am – 10:30 am
Room 502AB

Moderator: **Mr Bonzai**, Communication Arcs, Hollywood, CA, USA

Panelists: **Jack** and **Blake Douglas**

Industry media mainstay Mr. Bonzai and his special guests, Jack & Blake Douglas kick-off the AES Special Events calendar with a first-morning wake-up dialog, talking music, motivation, and music industry trends and prognostications. Legendary producer/engineer Jack Douglas got his start at Record Plant in NYC as a janitor, and within a few short years was working with John Lennon, Cheap Trick, and Aerosmith. His son Blake is a successful engineer (Nas, Common, No ID), production consultant, and studio designer.

Thursday, September 29 10:00 am Room 405

Technical Committee Meeting on Acoustics and Sound Reinforcement

Project Studio Expo 1 Thursday, September 29
10:30 am – 11:15 am PSE Stage

MAKING A LIVING ROOM:
A CONVERSATION WITH FRANK FILIPETTI

Presenters: **Frank Filipetti**
Glenn Lorbecki, Glenn Sound Inc. Seattle, WA, USA

One of the most sought-after mixers on the planet works out of a personal studio. Frank Filipetti takes it home in a discussion with Glenn Lorbecki.

Session P3 Thursday, September 29
10:45 am – 12:15 pm Room 403A

TRANSDUCERS—PART 2

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

10:45 am

P3-1 Power Considerations for Distortion Reduction of Loudspeakers—*Ajay Iyer,[1] Doug Button,[2] Russ Lambert[1]*
[1]Harman International, Salt Lake City, UT, USA;
[2]Harman International, Northridge, CA, USA

Over the last 25 years, scientists and engineers have written extensively about methods to reduce distortion in loudspeakers with Digital Signal Processing (DSP). Despite the several proposed solutions, no formal product exists on the market today that employs distortion reduction. In this paper the answer to some fundamental questions about what is required to make substantial improvements in loudspeaker performance is investigated through computer simulations. This research examines the level of volume achievable while still maintaining acceptable levels of distortion. Transducer designs that are best suited for this application are studied and identified.
Convention Paper 9605
[Paper presented by Doug Button]

11:15 am

P3-2 Improving the Sound Balance with Dynamic Control of Membrane Excursion—*Mikhail Pakhomov, Ivan S. Tolokonnikov, Victor Rozhnov, Mikhail Gusev*,
LG Electronics, St. Petersburg, Russian Federation

The electrodynamic transducers' that are used in mobile devices are typically prone to voice coil overheating and excessive excursion of the membrane. The paper focuses on the second aspect. Nonlinear distortion is known to depend on membrane excursion amplitude. High sound pressure at low frequencies also requires the maximum vibration amplitude. But now the sound balance is at stake. Thus, we face the challenge of finding the optimal relation between the sound balance and the level of audible distortion to obtain the maximum subjective quality evaluation.
Convention Paper 9606

11:45 am

P3-3 Force Factor Modulation in Electro Dynamic Loudspeakers—*Lars Risbo,[1] Finn T. Agerkvist,[2] Carsten Tinggaard,[3] Morten Halvorsen,[3] Bruno Putzeys[1]*
[1]Purifi, Denmark/Belgium
[2]Technical University of Denmark, Lyngby, Denmark
[3]Pointsource Acosutics, Roskilde, Denmark

Loudspeaker horns, waveguides, and other ducts can The relationship between the non-linear phenomenon of "reluctance force" and the position dependency of the voice coil inductance was established in 1949 by Cunningham, who called it "magnetic attraction force." This paper revisits Cunningham's analysis and expands it into a generalized form that includes the frequency dependency and applies to coils with non-inductive (lossy) blocked-impedance. The paper also demonstrates that Cunningham's force can be explained physically as a modulation of the force factor that again is directly linked to modulation of the flux of the coil. A verification based on both experiments and simulations is presented along discussions of the impact of force factor modulation for various motor topologies. Finally, it is shown that the popular L2R2 coil impedance model does not correctly predict the force unless the new analysis is applied.
Convention Paper 9607

Session EB1 Thursday, September 29
10:45 am – 12:15 pm Room 403B

POSTERS: SPATIAL AUDIO & TRANSDUCERS

10:45 am

EB1-1 A Ground Plane Measurement Comparison between Two Floor-Standing Loudspeaker Systems: A Conventional Three-Way Studio Monitor vs. A Ground-Plane Constant Beamwidth Transducer (CBT) Line Array—*D.B. (Don)*

Keele, Jr., DBK Associates and Labs, Bloomington, IN, USA

This paper compares two different types of floor-standing loudspeaker systems. Both were measured over an acoustically reflective hard surface in a large space. The first is a high-performance conventional three-way 12"-woofer studio monitor and the second is a ground-plane circular-arc CBT line array loudspeaker. Measurements included direct-field frequency responses in front of the systems at 20 grid locations ranging over different distances/heights and response vs. distance at seated and standing heights. Horizontal off-axis and near-field responses were also gathered along with ceiling illumination responses at several launch angles. The measurements reveal that the CBT system has vastly more even coverage at all these locations compared to the three-way monitor and in addition eliminates the detrimental effects of floor bounce.
Engineering Brief 279

10:45 am

EB1-2 A 3D Sound Localization System Using Two Side Loudspeaker Matrices—*Ryo Kaneta, Akira Saji, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

We have been researching about 3D spatial sound localization using loudspeakers. However, loudspeaker systems using VBAP methods, usually need a large listening space and a lot of loudspeakers. In this paper we propose a loudspeaker matrix system to improve sound localization on 3D sound system for personal use. Four loudspeakers were set on all vertices of 25 centimeters regular square, 2 matrices on both sides of the listener. Then, we have held audio experiments to confirm the effect of the matrix system. As a result, the loudspeaker matrices system can improve sound localization especially with higher elevation.
Engineering Brief 280

10:45 am

EB1-3 Source-Distance Based Panning Algorithm (SDAP) for 3D Surround Sound—*Matthew Wong, Sungyoung Kim*, Rochester Institute of Technology, Rochester, NY, USA

The Source-Distance Based Amplitude Panning (SDAP) algorithm offers a new approach in determining gain amounts for distributed loudspeakers in a three-dimensional (3D) space. Similar to the 3D implementation of Vector Based Amplitude Panning (VBAP), this method is based on the use of non-overlapping triangular regions formed by the known locations of sets of three loudspeakers. Unlike VBAP, however, this method compares the location of the panning vector to the surface formed by the triangular region and uses Barycentric coordinates to determine the speakers' respective amplitudes. In addition, SDAP removes the possibility of negative amplitudes as may appear in VBAP. Informal listening test results showed that the perceived position of the sound source was perceptually well matched to the target position.
Engineering Brief 281

10:45 am

EB1-4 STEAK: Backward-Compatible Spatial Telephone Conferencing for Asterisk—*Dennis Guse, Frank Haase*, University of Technology Berlin, Berlin, Germany·

In this paper we present our implementation of a telephone conferencing system that renders a spatial representation via binaural synthesis. The implementation extends the open-source software Asterisk and complies with established Voice-over-IP standards. The implementation only requires clients to be capable of receiving and reproducing the rendered binaural signals (two channels). Furthermore, the implementation is backward-compatible as clients not fulfilling these requirements are provided with mono-rendered signals without additional spatial information. The implemented system is released as open-source software and will enable researchers to investigate the (dis-)advantages of spatial conferencing under real-world conditions. The project name is Spatial TelephonE conferencing for AsterisK (STEAK)
Engineering Brief 282

10:45 am

EB1-5 Process of HRTF Individualization by 3D Statistical Ear Model—*Slim Ghorbal,*[1,2] *Renaud Séguier,*[1] *Xavier Bonjour* [2]
[1] CentraleSupélec
[2]3D Sound Labs, Rennes, France

The use of HRTFs is of well-known importance when it comes to binaural listening. However, easily capturing accurate data is a key point that has not been solved yet. In this paper we present a process for individualizing the HRTFs of an individual using ear photographs. These are fitted to a statistical 3D ear model coupled to a statistical HRTF model. This coupling allows to instantly generate from a given parameterization of the ear model a corresponding set of HRTFs. The accuracy of the results is a direct consequence of the quality and the size of the underlying databases.
Engineering Brief 283

10:45 am

EB1-6 Evaluation of Portable Loudspeakers Using Virtual Listening Test—*Ziyun Liu, Yong Shen, Pei Yu, Hao Yin*, Nanjing Technological University, Nanjing, China

The audio quality of portable wireless speakers for consumer use has improved in recent years. In this study perceptual evaluations of several portable loudspeaker systems were performed using virtual listening test methods. A preference test was designed to assess the comparative performance of these loudspeakers. The recordings of each loudspeaker were done in a controlled situation and then the listening panel took the listening test by headphone playbacks. Considering the wide use of bass-boost effect in these products, perceptual qualities of both high playback level and low level were investigated. Several objective measures were also applied to these loudspeakers. The results of the listening test and objective measures were compared and discussed.
Engineering Brief 284

Tutorial 2 **Thursday, September 29**
10:45 am – 12:15 pm **Room 402AB**

PRODUCT MANAGEMENT 101: THE BUSINESS AND MARKETING BEHIND THE PRODUCT

Presenters: **Greg Riggs**, Guitar Center, Westlake Village, CA, USA
Scott Leslie, Ashly Audio, Webster, NY, USA; PD Squared, Irvine, CA USA

Product management has developed into an interdisciplinary field that takes a holistic view at the entire product life cycle from inception to end of life, bringing together strategy, market needs, opportunity analysis, product plans, product development and engineering, manufacturing and operations, financial management, marketing, customer support, and additional functions as needed to ensure products meet customer needs and business requirements. This tutorial will walk through the major product management implications through the product life cycle and discuss key interdisciplinary functions for each phase.

Broadcast/Streaming Media 2 **Thursday, September 29**
10:45 am – 12:15 pm **Room 408A**

LISTENER FATIGUE AND RETENTION

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

Panelists: *Rob Arbittier*
Bruce Botnick, Music Producer, Recording and Mastering Engineer, VP Content Acquisition for Pono Music, Los Angeles, CA, USA
John Gavin, University of California Los Angeles, Los Angeles, CA, USA
Scott Gershin, Technicolor
Thomas Lund, Genelec Oy, Iisalmi, Finland
Thomas Sporer, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Listener fatigue is directly related to the epidemic of hearing loss among both the producing and consuming population. It is essential for the producers of audio content to understand what listener fatigue is and what causes it. The panel will discuss the physiological and psychological aspects of the phenomenon. Each panelists has his own perspective on what can be done to reduce the phenomenon.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Game Audio 2 **Thursday, September 29**
10:45 am – 11:45 am **Room 406AB**

DESIGNING, PLANNING AND CREATING A DYNAMIC MUSIC SYSTEM

Presenter: **Stephan Schütze**, Sound Librarian, Melbourne, Australia

Dynamic music is becoming more common in video game projects. At its most basic it can highlight a variety of events and states within the gameplay, but at its most complex it can underscore the narrative and action of a game and dramatically improve the player's experience. A Dynamic or interactive score, however, can take on many different forms, and the process of designing and creating a dynamic score can become very complex before a single note of music is even written. This session will discuss some of the variety of formats that dynamic music can adopt and how some of these formats may suit a specific project better. This session is not about writing the music itself, but about how to analyze a project and work out what dynamic elements may work best under different circumstances.

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

Networked Audio 2 **Thursday, September 29**
10:45 am – 12:15 pm **Room 408B**

UNIVERSITY OF NORTH TEXAS COLLEGE OF MUSIC —RECORDING WITH NETWORKED AUDIO

Presenter: **Blair Liikala**, University of North Texas College of Music, Texas, USA

The University of North Texas College of Music recently installed a Dante network within their main music building, and performing arts center consisting of +70 Dante devices in multiple studios and concert halls for one integrated recording, and reinforcement system. Learn about the challenges, rewards and insights from a deeply integrated system at one of the largest enrolled music colleges in the nation.

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

Sound Reinforcement 2 **Thursday, September 29**
10:45 am – 12:15 pm **Room 404AB**

AC POWER & GROUNDING

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

Panelists: *Ken Fause*, Auerbach Consultants
Bruce Olson, AFMG Services North America

Nearly all sound reinforcement, or SR, systems must have utility AC power. To keep them safe and legal, regulatory rules such as "Code" must be strictly observed. However, these compliant power distribution systems inherently create harmlessly small voltage differences, called GVD, between physical locations in the safety ground network. This session will explain:

1. Exactly how GVD is created in premises wiring;
2. New-construction wiring techniques that can reduce GVD by several orders of magnitude; and
3. How GVD couples into signal paths, causing hum, buzz, and myriad digital network problems.

Related topics such as electrician mistakes, earth grounding, power conditioning (including "balanced power"), and choosing proper signal cables will also be discussed.

Student Event and Career Development
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING—PART 1
Thursday, September 29, 10:45 am – 12:15 pm
Room 501ABC

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 audio students.

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

Live Sound Expo 1 **Thursday, September 29**
11:00 am – 11:45 am **LSE Stage**

AES67: AUDIO NETWORKING TODAY & TOMORROW

Presenters: **TJ Adams**, QSC Audio Products, LLC., Boulder, CO, USA
Jeff Berryman, Bosch Communications,

Ithaca, NY, USA
Patrick Killianey, Yamaha Professional Audio, Buena Park, CA, USA
Morten Lave, TC Applied Technologies, Toronto, ON, Canada
Ethan Wetzell, Bosch Communications Systems, Burnsville, MN USA; OCA Alliance

A discussion of current and future audio networking for live sound. The average sound guy wants to know what all the fuss is about Dante, AES67, and AVB for next week's gig or next year's installation.

Thursday, September 29 11:00 am Room 405

Technical Committee Meeting on Archiving Restoration, and Digital Libraries

Project Studio Expo 2 Thursday, September 29
11:30 am – 12:15 pm PSE Stage

WHAT REALLY MAKES A DIFFERENCE? GETTING GREAT RESULTS FROM A BUDGET STUDIO

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

This session will cover the essentials of getting the best from a typical home studio along with some hints on choosing appropriate equipment and improvising acoustics when recording vocals and common instruments. The main difference between a high-end pro studio and a home studio these days comes down mainly to experience, acoustic spaces, and accurate monitoring. Experience comes with time and practice but there's a lot you can do to combat the inevitable acoustic compromises when recording and mixing in a small space. There are also some fairly simple things you can do to help you get a more accurate picture of what you've recorded. Along the way I'll pass on some easy-to-apply tips and tricks for mixing and will include, where appropriate, anecdotes from our popular Studio SOS series. The session will conclude with a question and answer session.

Live Sound Expo 2 Thursday, September 29
12:00 noon – 12:45 pm LSE Stage

FLEXIBLE FOO: DIGITAL SYSTEM DRIVE

Presenter: **Philip Reynolds**, System Tech, Foo Fighters, Gilbert, AZ, USA

Digital drive signal flow from FOH is today's professional tour standard for sheds, arenas, and stadiums and the holy grail of system engineers. This session looks at one solution to keep the mix in the digital domain all the way to the amplifiers, avoiding DA/AD conversions and signal degradation.

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Thursday, September 29, 12:30 pm – 2:00 pm
Room 502AB

Opening Remarks: • Executive Director Bob Moses
• President John Krivit
Convention Chairs Michael MacDonald & Valerie Tyler
Program: • AES Awards Presentation by Sean Olive, Awards Chair
• Introduction of Keynote Speaker
• Keynote Address by Ron Jones

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
- Juan San Martin
- Natalia Sotelo

BOARD OF GOVERNORS AWARD
- Jim Anderson
- David Bialik
- Filippo Fazi
- Paul Gallo
- Patrick Hegarty
- Michael Kelly
- Andres Felipe Millan
- Fabio Nicholls
- Sean Olive
- Neil Shaw
- Valerie Tyler

FELLOWSHIP AWARD
- Mauricio Ardila
- David Josephson
- Bob Katz

SILVER MEDAL AWARD
- Don Puluse

GOLD MEDAL AWARD
- Diana Deutsch
- D.B. (Don) Keele, Jr.

Keynote Speaker

This year's Keynote Speaker is **Ron Jones**. Ron Jones is a professional composer with over 40,000 compositions to his credit and growing. Ron has scored for film, television, games and live performance. His credits include "Star Trek: The Next Generation," "Family Guy," and many more. With a Grammy nomination and five Emmy nominations (as well as many other awards) Jones has been recognized as a composer of many of the most popular series in television. After 37 years living and working in L.A. Ron and his wife Laree moved to the beautiful Northwest. Ron has built SkyMuse Studios where he is involved not only in composing, recording, and producing his own work but providing a state-of-the-art studio doing recording for all sorts of groups and artists. SkyMuse presents a regular Master Class Series offering lectures and seminars on Audio Production, Scoring, Orchestration, Composition and many other aspects on music featuring experts and special guests. Ron founded the educational nonprofit The Academy of Scoring Arts based in Los Angeles. Last November Ron conducted his music with the London Philharmonic at Royal Albert Hall, and is a member of the PNW Chapter of the AES. The title of his talk is "Remember the Human Receptor on the Road to the Future."

I support the new directions that audio (and video) are heading, but caution everyone involved in this rapidly changing time to not lose sight of the Human Receptor. In my work as a composer of a vast number of scores for all the major studios, my job is not to tell the story. The picture, the characters and the dialogue, as well as the language of the camera and editing, do that. My job is to engage emotions using the art and craft of music. In other words, I manipulate emotional perception and response within the viewer and from experience I know with a large degree of accuracy how to push emotional buttons in sync with picture and story. Audio is processed very differently by the brain than visual stimuli. Audio has deep connections to the neural network, connecting directly with neural electrical-chemical transmitters. Without a firm understanding of the Human Receptor, new technologies and inventions won't achieve the desired effect with consumers, content creators and the manufacturers and markets for these products. Computers easily exceed humans in computing bandwidth, and with quantum computing coming in the near future, we will all be left in the dust. Common sense, and a solid regard for how and why the Human Receptor works serves as a compass, a map, a wonderful guide as things rocket forward.

Project Studio Expo 3 **Thursday, September 29**
12:30 pm – 1:15 pm **PSE Stage**

MIXING IN THE COUNTRY: HOW I MOVED TO RURAL WORCESTERSHIRE, UK AND NOBODY NOTICED

Presenter: **Andrew Scheps**, Waves

Live Sound Expo 3 **Thursday, September 29**
1:00 pm – 1:45 pm **LSE Stage**

MONO, STEREO OR LCR IN HOW, THEATER, AND PACS

Presenter: **Dan Palmer**, L-Acoustics Inc., Oxnard, CA, USA

Learn the advantages and dilemmas of choosing a one, two or three channel sound reinforcement system for House of Worship, theater or multi-use performing arts center. What are the trade-offs: is stereo good enough, is a center channel worth the added expense and what are the pitfalls of LCR.

Live Sound Expo 4 **Thursday, September 29**
2:00 pm – 2:45 pm **LSE Stage**

MIKING THE SYMPHONY: VARIATIONS ON A THEME

Presenters: **Shawn Murphy**, Recording and Live Sound Engineer, Berkeley, CA, USA
George Relles, Relles Sound, Inc., Eugene

Symphony sound reinforcement has grown from modified Decca trees with outriggers and spot mics borrowed from recording to individually miking every stand or even every musician. Approaches vary from matched pencil condensers to various types of mics for each orchestra section with interesting similarities and differences from one engineer to another, depending, of course on available inventory.

Thursday, September 29 **2:00 pm** **Room 405**

Technical Committee Meeting on Coding of Audio Signals

Session P4 **Thursday, September 29**
2:15 pm – 3:45 p **Room 409B**

SPATIAL AUDIO—PART 1: PRODUCTION

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

2:15 pm

P4-1 **A Three-Dimensional Orchestral Music Recording Technique, Optimized for 22.2 Multichannel Sound—***Will Howie, Richard King, Denis Martin*, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music, Media, and Technology, Montreal, Quebec, Canada

Based on results from previous research, as well as a new series of experimental recordings, a technique for three-dimensional orchestral music recording is introduced. This technique has been optimized for 22.2 Multichannel Sound, a playback format ideal for orchestral music reproduction. A novel component of the recording technique is the use of dedicated microphones for the bottom channels, which vertically extend and anchor the sonic image of the orchestra. Within the context of highly dynamic orchestral music, an ABX listening test confirmed that subjects could successfully differentiate between playback conditions with and without bottom channels.
Convention Paper 9612

2:45 pm

P4-2 **Subjective Graphical Representation of Microphone Arrays for Vertical Imaging and Three-Dimensional Capture of Acoustic Instruments, Part I—***Bryan Martin*,[1,2,3] *Richard King*,[1,2] *Wieslaw Woszczyk*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
[3]Enhanced Reality Audio Group (ERA), Montreal, Quebec, Canada

This investigation employs a simple graphical method in an effort to represent the perceived spatial attributes of three microphone arrays designed to create vertical and three-dimensional audio images. Three separate arrays were investigated in this study: 1. Coincident, 2. M/S-XYZ, and 3. Non-coincident. Instruments of the orchestral string, woodwind, and brass sections were recorded. Test subjects were asked to represent the spatial attributes of the perceived audio image on a horizontal/vertical grid via a pencil drawing. It can be seen in the subjects' representations that these techniques clearly capture much more information than a single microphone and exhibit vertical as well as horizontal aspects of the audio image.
Convention Paper 9613

3:15 pm

P4-3 **Grateful Live: Mixing Multiple Recordings of a Dead Performance into an Immersive Experience—***Thomas Wilmering, Florian Thalmann, Mark B. Sandler*, Queen Mary University of London, London, UK

Recordings of historical live music performances often exist in several versions, either recorded from the mixing desk, on stage, or by audience members. These recordings highlight different aspects of the performance, but they also typically vary in recording quality, playback speed, and segmentation. We present a system that automatically aligns and clusters live music recordings based on various audio characteristics and editorial metadata. The system creates an immersive virtual space that can be imported into a multichannel web or mobile application allowing listeners to navigate the space using interface controls or mobile device sensors. We evaluate our system with recordings of different lineages from the Live Music Archive's Grateful Dead collection.
Convention Paper 9614

Session P5 **Thursday, September 29**
2:15 pm – 3:45 pm **Room 403B**

POSTERS: ACOUSTICS, TRANSDUCERS, AND AUDIO

2:15 pm

P5-1 Combined Inverse Filtering and Feedback Control for Robust Equalization and Distortion Reduction in Loudspeaker Systems—*Yusuke Kadowaki, Toshiya Samejima*, Kyushu University, Fukuoka, Japan

A method for the robust equalization and distortion reduction of loudspeakers is proposed. The proposed method adopts both an IIR-type inverse filter and a feedback control. The feedback control based on model-following control theory is used to force a loudspeaker to move as a linear time-invariant (LTI) system. Accordingly, we expect the inverse filter that is specifically designed for the LTI system to work correctly. Furthermore, nonlinear distortion of a loudspeaker is expected to be reduced. Computer simulation shows that the proposed method achieves more robust equalization of a loudspeaker than inverse filtering alone. In addition, the proposed method simultaneously reduces nonlinear distortion of the loudspeaker.
Convention Paper 9608

2:15 pm

P5-2 Investigation of Impulse Response Recording Techniques in Binaural Rendering of Virtual Acoustics—*Kaushik Sunder, Wieslaw Woszczyk*, McGill University, Montreal, Quebec, Canada; Center for Interdisciplinary Research in Music Media and Technology (CIRRMT), Montreal, Quebec, Canada

With the advent of virtual reality headsets, accurate rendering of the acoustics of the real space is critical to deliver a truly immersive experience. To ensure the veracity of immersion, there is a need to obtain high quality impulse responses that captures all the relevant acoustical features of the space. In this work we investigate and compare the perception of virtual acoustics rendered over headphones using impulse responses captured with (a) binaural dummy-head, and (b) multichannel (8-channel) microphone array. A downmixing algorithm is developed that converts the free-field 8-channel impulse responses to binaural for rendering over headphones. Subjective experiments suggest higher quality of immersion with reconstructed binaural from multichannel room impulse responses compared to the measured binaural room impulse responses. This investigation provides important information in understanding the essential elements in creating a convincing perception of an acoustic space.
Convention Paper 9609

2:15 pm

P5-3 New Recording Application for Software Defined Media—*Masahiro Ikeda*,[1] Takuro Sone,[1] *Kenta Niwa*,[2] *Shoichiro Saito*,[2] *Manabu Tsukada*,[3] *Hiroshi Esaki*[3]
[1]Yamaha Corporation, Shizuoka, Japan
[2]NTT Media Intelligence Laboratories, Tokyo, Japan
[3]University of Tokyo, Tokyo, Japan

In recent years, hardware-based systems are becoming software-based and networked. From IP based media networks, the notion of Software Defined Media (SDM) has arisen. SDM is an architectural approach to media as a service by virtualization and abstraction of networked infrastructure. With this approach, it would be possible to provide more flexible and versatile systems. To test this concept, a baroque orchestra was recorded by various methods with 82 channels of microphones in total. All the data was organized based on the object-based concept and we applied advanced signal processing to the data based on array signal processing technology to produce a content matching various purposes of possible applications. Through this study, the value of SDM concept is verified.
Convention Paper 9610

2:15 pm

P5-4 Interference Evaluation of Parametric Loudspeakers on Digital Hearing Aids—*Santi Peksi*,[1] *Woon-Seng Gan*,[1] *Dong-Yuan Shi*,[1] *Satya Vijay Reddy Medapati*,[2] *Eu-Chin Ho*[2]
[1] Nanyang Technological University, Singapore
[2]Tan Tock Seng Hospital, Singapore

Parametric loudspeakers are able to generate a highly-directional sound, and recently it has also been used to help the hearing impaired to hear TV programs better. However, there are incidents that particular hearing aid users have reported audible interferences in the path of directional sound beams during the clinical trials. The interference varies from buzzing noise to static noise for various commercialized behind-the-ear (BTE) hearing aids. To investigate the audible interference, hearing aid output measurements were carried out using B&K Head and Torso Simulators (HATS) inside an anechoic room at various distances for four types of parametric loudspeakers. This paper also investigates its possible cause of interference and raises awareness to professionals on potential audible interference on hearing aids using parametric loudspeakers.
Convention Paper 9611

Workshop 1 **Thursday, September 29**
2:15 pm – 3:45 pm **Room 501ABC**

Q VS Q

Chair: **Robert Bristow-Johnson**, audioImagination, Burlington, VT, USA

Panelistss: *Alex Case*, University of Massachusetts Lowell, Lowell, MA, USA
Jason Corey, University of Michigan, Ann Arbor, MI, USA
Jean-Marc Jot, DTS, Inc., Los Gatos, CA, USA
George Massenburg, Schulich School of Music, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Christoph M. Musialik, Sennheiser Audio Labs, Waldshut-Tiengen, Germany

This is maybe about continuing a discussion that I might have tried starting 20 years ago ("The Equivalence of Various Methods of Computing Biquad Coefficients for Audio Parametric Equalizers"). I'm sure that longer ago than that, practitioners in the studio had discussions regarding the setting of the Q knob (or bandwidth) for the bell parametric EQ, but this is about settling the issue of what the biquad filters should be, from an input/output or transfer function POV, given the positions of the knobs on an EQ whether it's a good-old-fashioned analog mix board or an EQ plugin or built-in EQ in a DAW. There has been some discussions about why a seemingly identical function implemented in different products sound so different, and to the extent possible, this proposed discussion is there to try to remove "different settings" as the cause.

Perhaps another reason to have this discussion perhaps a little late, but better than never, is because of automation. At present, as best as I know from what I am told, automation in Pro Tools and other DAWs is unitless and is essentially a zero to full-scale parameter that is interpreted solely by the floor and ceiling attributes of the slider the built-in EQ or the plugin control. I dunno, but it seems like to me that the AES is involved in standardization, when possible, of parameters and performance of audio functions that are common between gear from various manufacturers. There is a Standards Subcommittee on Metadata and a Standards Working Group on Audio-File Transfer and Exchange. Perhaps there is not yet a group defining automation standards, but eventually a user should be able to move the raw sound files of a song or session running on one platform, a specific DAW with plugins or even an analog mixer with motorized faders, and move that to another platform and compare how they like the sound. It might be nice if the platforms could interpret the recorded metadata, which is essentially what automation is, in a compatible manner, at least with parameters that are universally understood.

So before too much of this is carved into stone in a standard somewhere, it might be nice if we could, as an industry, discuss among each other exactly what we mean by Q.

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

Workshop 2 **Thursday, September 29**
2:15 pm – 3:45 pm **Room 404AB**

IRONS IN THE FIRE: CAREER AND BUSINESS DEVELOPMENT MENTORING

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

Panelists: *Rafaël Leloup*
Ashwin Subramanian
Carl Tatz
Rick Torres
Richard Warp

In cooperation with the Manhattan Producers Alliance, this event will offer fresh insights into career development, networking, and entrepreneurship in an ever-changing industry. Come participate in open discussions, discuss your personal career goals, and get a chance to meet some ManhatPro members.

Broadcast/Streaming Media 4 **Thursday, September 29**
2:15 pm – 3:45 pm **Room 408A**

TELEVISION AND MOVIES—THE EVOLUTION OF AUDIO AND SOUND: A LOOK BACK TO THE MOMENTS THAT MATTER

Moderator: **Brian Vessa**, Sony Pictures

Panelists: *Corey Bailey*
Ed Greene
J. Mark King

There is no doubt that audio and sound is essential to every storyline. It enhances the visual experience and adds depth, dimension, and emotion. It is sound that engages the viewer and drives the storyline. The creativity of the audio design and balancing of multiple audio elements produces the key "sonic signature" of the sound and brings a unique life to the content. In the spirit of SMPTE's centennial year, this panel will look back at key developments in audio and sound technologies for both broadcast and cinema and how these have affected production for live events, movie shoots and post-production sound. The panel will discuss important milestones such as digital recording, editing and distribution, immersive audio and, perhaps, if we're lucky, share a few personal experiences and memories as they travel along with the evolution.

Co-produced with Society of Motion Picture and Television Engineers

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Game Audio 3 **Thursday, September 29**
2:15 pm – 3:15 pm **Room 406AB**

DIALOGUE RECORDING WORKFLOW

Presenter: **Tom Hays**, Rocket Sound, North Hollywood, CA, USA

The human voice is a core part of most large video games. We use voice to tell stories, add dynamism, and create worlds and situations that players care about. The sheer scale and complexity of dialogue in most AAA games dwarfs that of feature films, with typically 10 to 100 times as many lines as a movie, with non-linear implementation supporting player experiences that can last dozens of hours. This session will look at the challenges this presents: technical, artistic, and logistical. We'll look at issues ranging from getting good performances from actors who can't see what they're reacting to, to some of the nuts-and-bolts rigor required to deliver dialogue in the form of data that a piece of game software can digest. We'll touch on micing and signal flow techniques used by many games, some specialized toolsets developed for game workflows, and how some general-purpose tools are often used in games. We'll also look at ways to tie voice to body motion and facial movement, bringing full performance capture to games.

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

Product Development 1 **Thursday, September 29**
2:15 pm – 3:45 pm **Room 402AB**

HEADPHONES, HEADSETS, AND EARPHONES: ELECTROACOUSTIC DESIGN AND VERIFICATION

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

This presentation reviews the basic electroacoustic concepts of gain, sensitivity, sound fields, linear and non-linear systems, and test signals for ear-worn devices. The Insertion Gain concept is explained and free and diffuse field target responses are shown. Equivalent volume and acoustic impedance are defined. Ear simulators and test manikins appropriate for Circum-, Supra-, and Intra-aural and insert earphones are presented. The salient portions of the ANSI/ASA S3.7 and IEC 60268-4 standards are reviewed. Examples of frequency response, left-right tracking, insertion gain, distortion, and impedance are shown. The basic concepts of noise cancelling devices are also presented.

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

Sound Reinforcement 3 **Thursday, September 29**
2:15 pm – 3:45 pm **Room 408B**

POLITICAL AUDIO: SOUND FOR PAST AND RECENT PRESIDENTIAL DEBATES AND CONVENTIONS

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

Panelists: *Pat Baltzell*, Baltzell Audio Design, Sherman Oaks, CA USA

Dave Brand, Intracom Systems, LLC, Los Angeles, CA, USA
Stanley R. Miller, Sound Motions Inc., Big Bear Lake, CA, USA

A panel with first-hand experience on these specialized and currently topical events has being assembled. Consultant Ken Fause and his panel will discuss how sound aspects of the Republican and Democratic debates and conventions, the Presidential debates, and the inaugurations were handled in the distant and recent past. Sound reinforcement, broadcast sound, and audio communications will all be discussed.

Student Event and Career Development
REAL INDUSTRY & AES: LIVE!
Thursday, September 29, 2:15 pm – 3:45 pm
Room 403A

This exceptional event in partnership with Real Industry features a mentor-led workshop for top students and early career professionals to immerse themselves into the media technology industry. The event will feature mentors from top music, video, and silicon valley industry professionals. Details about the exciting Real Industry / AES Student Party will be announced during the workshop. Students interested in attending are encouraged to RSVP to ensure adequate mentors will be on-hand - http://bit.ly/aes-ri

Project Studio Expo 4 — Thursday, September 29
2:15 pm – 3:15 pm — PSE Stage

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. The seminar will be specifically tailored to those working on a budget, and will feature insider tips on equipment choice, mic technique, and session psychology, with plenty of supporting audio examples so you can judge the results with your own ears.

Recording & Production 2 — Thursday, September 29
2:30 pm – 4:00 pm — Room 502AB

LATIN PRODUCERS PANEL (SPECIAL EVENT)

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

Panelists: *Carlos "El Loco" Bedoya*, Composer/Producer/Engineer, Miami Beach, FL, USA
Gustavo Borner, Burbank, CA, USA
Juan Cana San Martin
Martha DeFrancisco, McGill University, Montreal, QC, Canada
Tweety González, Twitin Records, Buenos Aires, Argentina
Cesar Lamschtein, Auditorio Nacional del Sodre, Montevideo, Uruguay; Mixymaster, Montevideo, Uruguay
Rafa Sardina

The Audio Engineering Society brings together the top-notch music producers and engineers from the Latin scene. Multiple Grammy-winning pros will present an in-depth look at their latest recordings and will open the debate for Q&A from the audience.

Live Sound Expo 5 — Thursday, September 29
3:00 pm – 3:45 pm — LSE Stage

LIVE MIXING ACADEMY—CONSOLE LAYOUT & WORKFLOW

Presenter: **Robert Scovill**, Avid Technologies, Scottsdale, AZ, USA; Eldon's Boy Productions Inc.

Taking a band from concept to rehearsal stage, starting with a roster of musicians, building an input list, laying it out not only for easy operation, but to fit into festivals and fly-in situations. Building mute groups and VCAs, starting an automation scene list from a song list. Preparing for Virtual Soundcheck.

Thursday, September 29 — 3:00 pm — Room 405

Technical Committee Meeting on Audio for Telecommunications

Project Studio Expo 5 — Thursday, September 29
3:30 pm – 4:15 pm — PSE Stage

A CONVERSATION WITH DAVE AUDÉ

Presenter: **Dave Audé**, Waves

The acclaimed remixer's musings on: Remixing, Producing, Writing, Plug-Ins, Publishing, Recording. And whatever else comes to mind.

Session P6 — Thursday, September 29
4:00 pm – 6:00 pm — Room 403A

TRANSDUCERS—PART 3

Chair: **Mark Gander**, JBL Professional, Northridge, CA, USA

4:00 pm

P6-1 Measurement of the Frequency and Angular Responses of Loudspeaker Systems Using Radiation Modes—
Maryna Sanalatii,[1,2,3] *Philippe Herzog*,[1] *Manuel Melon*,[2] *Régine Guillermin*,[1] *Jean-Christophe Le Roux*,[3] *Nicolas Poulain*[3]
[1]Laboratoire de Mécanique et d'Acoustique UPR CNRS, Marseille, France
[2]Université du Maine, UMR CNRS, Le Mans cedex 9, France
[3]Centre de Transfert de Technologie du Mans, Le Mans, France

In this paper the "radiation mode" (RM) method is applied to the measurement of the frequency response and directivity pattern of two loudspeaker systems. This approach is based on solving the discretized Helmholtz equation on the source boundaries to obtain an efficient expansion suitable to represent any field radiated by a source. Bookshelf and column systems have been tested. Results obtained with the proposed method are then compared to the ones given by two other methods: measurement in an anechoic room and boundary element computation based on the scanning of the membrane velocity. Results show a good agreement between the different methods. Pros and cons of the different approaches are then discussed as well as the possibility to use the "radiation mode" method in non-anechoic rooms.
Convention Paper 9615

4:30 pm

P6-2 Vandermonde Method for Separation of Nonlinear Orders and Measurement of Linear Response— *Russell H. Lambert*, Harman International, Salt Lake City, UT, USA

The Vandermonde matrix method for separation of nonlinear components and full-power linear response measurement is analyzed in this paper. This technique involves making several measurements of a nonlinear system (a woofer or horn system for example) at different gains and applying the inverse Vandermonde gain matrix to the vector of outputs. The Vandermonde matrix method does more than just return the linear response, as it more generally separates all of the nonlinear orders breaking difficult nonlinear system estimation tasks into more tractable problems, one for each Volterra kernel. Quantitative measures for the degree of nonlinear order separation are proposed. The Vandermonde matrix order separation method is analyzed for noise robustness and gain spacing sensitivity and found to be a useful and practical tool for audio measurements.
Convention Paper 9616

5:00 pm

P6-3 Fluid Dynamics Analysis of Ported Loudspeakers— *Juha Backman*, Genelec Oy, Iisalmi, Finland

The small-signal performance of ported loudspeakers is described in an excellent way by traditional models, such as lumped parameters, waveguide models, or numerical solutions of the acoustic wave equation. However, the acoustic models are clearly insufficient to predict the nonlinear behavior of ported enclosures. This paper presents the results of a computational fluid dynamics analysis of an unlined ported enclosure, focusing on the behavior around the tuning frequency. The results indicate that the vortex formation around the port ends has a significant effect already at a relatively low flow velocities and that the transient behavior of the vortex field can differ from that predicted by the acoustical solution.
Convention Paper 9617

5:30 pm

P6-4 Compression Drivers' Phasing Plugs—*Alexander Voishvillo*, JBL/Harman Professional, Northridge, CA, USA

Most of compression drivers have phasing plugs with annular slots. Existing theories give recommendations for positioning of annular slots to suppress air resonances in compression chamber. However, interaction of diaphragm's mechanical resonances with the compression chamber's air resonances makes the problem very complex and a general theoretical solution hardly exists. New approach, based on certain empirical assumptions, is proposed and explained. New phasing plugs have slots of a "meandering" shape that provide effective "averaging" of high-frequency acoustical signal received from different parts of compression chamber. The method is applicable to drivers having domes, cones, and annular diaphragms. Other aspects of the design such as efficiency, compression ratio, and difference between air resonances in dome and annular compression chambers are discussed.
Convention Paper 9618

Session P7 Thursday, Sep. 29
4:00 pm – 6:00 pm Room 409B

SPATIAL AUDIO—PART 2

Chair: **Durand Begault**, NASA Ames Research Center, Moffet Field, CA, USA

4:00 pm

P7-1 Minimum-Audible Angles in Wave-Field Synthesis: A Case Study—*Florian Völk*, Tech-nische Universität München, München, Germany; WindAcoustics UG (haftungsbeschränkt), Windach, Germany

Wave-field synthesis aims at creating a predefined sound field within a restricted listening area. Implementing and maintaining a wave-field-synthesis system is rather costly, as a high number of loudspeakers must be set up meticulously and driven individually. Despite this effort, a physically perfect synthesis is not possible. This contribution addresses a critical and relevant benchmark of synthesis quality: perceptual directional resolution. The study was conducted with a typical living-room-scale system by measuring minimum-audible angles in the horizontal plane with different stimuli. The results indicate that the procedure provides a directional resolution close to that of real sound sources.
Convention Paper 9619

4:30 pm

P7-2 Accurate Timbre and Frontal Localization without Head Tracking through Individual Eardrum Equalization of Headphones—*David Griesinger*, David Griesinger Acoustics, Cambridge, MA, USA

The ear and brain perceive the vertical position of sounds by matching the timbre detected *at the eardrum* of a listener to timbre patterns built up by that individual over a long period of time. But the eardrum timbre depends dramatically on ear canal resonances between 1000 Hz and 6000 Hz that boost the pressure at the eardrum as much as 20 dB. These resonances are highly individual and are either eliminated or altered by headphones. In-head localization is the result. We have de-veloped an app that uses an equal-loudness procedure to measure and restore the natural timbre. Accurate timbre and frontal localization are then perceived without head-tracking, and binaural re-cordings can be stunningly realistic.
Convention Paper 9620

5:00 pm

P7-3 The Room-in-Room Effect and its Influence on Perceived Room Size in Spatial Audio Reproduction— *Richard J. Hughes, Trevor Cox, Ben Shirley, Paul Power*, University of Salford, Salford, Greater Manchester, UK

In spatial audio it can be desirable to give the impression of a target space (e.g., a church). Often the reproduction environment is assumed acoustically dead; in practice most listening spaces (e.g., domestic living rooms) introduce significant reflections. The result is a room-in-room effect: a complex interaction of target and reproduction environments. This study investigates the influence on perceived room size. A number of target spaces were measured and rendered for loudspeaker playback. Reproduction rooms were measured, with variations produced via impulse response ad-justment. Dynamic binaural playback allow ed different target and reproduction room combinations,

with participants judging the size of environment being reproduced. Results indicate the more re-verberant of the target and reproduction rooms is most commonly heard.
Convention Paper 9621

5:30 pm

P7-4 Compressing Higher Order Ambisonics of a Personal Stereo Soundfield—*Panji Setiawan, Wenyu Jin,* Member IEEE

In this work we propose an approach to encode the multizone soundfield within the desired region that features a so-called bright zone with stereo sound effects based on higher order ambisonics (HOA) formats. We decompose the B-format signals for the complex multizone soundfield into the coefficients of a formulated planewave expansion. The multizone soundfield B-format signals are then directly compressed using state-of-the-art audio codecs. The results confirm the effectiveness of this HOA based multizone soundfield encoding. A significant reduction on the compression rate of the desired multizone soundfield with sufficient accuracy can be achieved by quantitatively analyzing the reproduction performance.
Convention Paper 9622
[This paper was not presented]

Workshop 3 Thursday, September 29
4:00 pm – 5:30 pm Room 406AB

SUBJECTIVE VERSUS OBJECTIVE MEASUREMENTS IN SOUND REPRODUCTION: HOW ACCURATELY CAN WE PREDICT PERCEIVED SOUND QUALITY BASED ON OBJECTIVE MEASUREMENTS?

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

Panelists: *Thomas Sporer*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
Todd Welti, Harman International Inc., Northridge, CA, USA

Subjective measurements (i.e., listening tests) on audio components and systems are time-consuming, expensive and challenging to conduct in a controlled way that produces meaningful results. The alternative is to rely entirely on objective measurements that are then interpreted by the engineer based on his/her experience in order to estimate perceived sound quality. In recent years, there have been several attempts to predict the sound quality ratings of listening tests based on objective measurements. Various models based on objective measurements have been developed to predict the perceived the perceived sound quality of audio components and systems including audio codecs, loudspeakers, headphones, automotive audio systems, and listening rooms. This workshop will give an overview of some of the challenges and limitations researchers face in developing models, discuss how accurate they are with some specific examples given from experts in the field. The workshop should be of interest to any professional or enthusiast interested in the perception and measurement of sound quality.

This session is presented in association with the AES Technical Committee on Perception and Subjective Evaluation of Audio Signals

Broadcast/Streaming Media 3 Thursday, September 29
4:00 pm – 5:30 pm Room 408A

IMMERSIVE AND OBJECT ORIENTED AUDIO

PLAYBACK IN THE HOME

Moderator: **Dave Wilson**, Consumer Technology Association

Panelists: *Brett Crockett*, Dolby Laboratories, San Francisco, CA, USA
Pei-Lun Hsieh, Ambidio, Los Angeles, CA, USA
Jean-Marc Jot, DTS, Inc., Los Gatos, CA, USA
Dave Pedigo, CEDIA
Bert Van Daele, Auro 3D & Galaxy Studios, Mol, Belgium

Immersive audio in the home is now way beyond 5.1 channel surround. The most advanced sys-tems have up to 22.2 channels and place speakers at different elevations, including the ceiling. Be-yond the number of speakers, object oriented audio now allows people to customize the way they consume content like never before. It lets them increase the volume of the voice track or decrease the volume of the background sounds to better understand voices. It lets them silence voices, like the announcers at a ballgame, if they simply want to experience the background noise in their pro-gram. And it lets them customize their experience in many other ways. This panel will cover the latest developments in bringing more advanced and more flexible audio into the home.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Product Development 2 Thursday, September 29
4:00 pm – 5:30 pm Room 402AB

TO ANECHOIC OR NOT TOO ANECHOIC: ADVANCING SPEAKER MEASUREMENT IN THE NEARFIELD

Moderator: **Kent Peterson**, Warkwyn

Panelists: *Wolfgang Klippel*, Klippel GmbH, Dresden, Germany
Manuel Melon
Evert Start

Line arrays, sound bars, and other loudspeaker systems using multiple transducers in large enclosures require a big anechoic measurement room with proper wall treatment and climate conditioning to ensure acoustical measurements under far field condition. Scanning the near field of the loudspeaker output and holographic processing of the measured data is an interesting alternative to far field measurements to improve the accuracy and the angular resolution of the directivity pattern within a shorter measurement time. Scanning the near field also has the added benefit of measurement under less than anechoic conditions. This tutorial explains the theoretical basis of the new technique, shows the practical application to professional and consumer loudspeaker, and discusses the consequences for the development of active loudspeaker measurements with active control of the directional properties (beam steering).

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

Recording & Production 3 Thursday, September 29
4:00 pm – 5:30 pm Room 404AB

ALL STAR ALBUM PRODUCTION

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

Panelists: *Marcella Araica*, Recording & Mix Engineer (Britney Spears, Madonna, Pink), Miami, FL,

USA; Los Angeles, CA, USA
Laura Escude
Piper Payne, Coast Mastering, San Francisco Bay Area, CA,USA
Laura Sisk, Los Angeles, CA, USA

The world's top producers, engineers and artists discuss the latest work flow and production tips that most effectively move projects from pre-production to tracking and editing to mixing and mastering. Panel will explore the recording process from all angles of traditional acoustic production, beat-making, and topline production.

Sound Reinforcement 4 **Thursday, September 29**
4:00 pm – 5:30 pm **Room408B**

MIXING FOR IMMERSIVE LIVE SOUND EVENTS

Moderator: **Shane Myrbeck**, ARUP, Emeryville, CA, USA

Panelists: *Paul Chavez*, Harman Professional Solutions
Danny Echevarria, Independent Sound Designer / Four Larks Junkyard Opera
Ryan Ingebritsen, Independent Composer, Sound Designer, Sound Artist, Electronic Musician, and Sound Engineer

Spatial audio production relies on a combination of psychoacoustic cues, contextual expectations of sonic space, and when done well, a sense of surprise and wonder at being immersed in a new soundscape. The means to achieve this involves a variety of multichannel loudspeaker and panning formats including ambisonics, vector-based panning systems, WFS, and bespoke arrays of nearly any description. This panel will include sound designers, artists, technical production staff and sound system/room acoustic designers discussing techniques for live production across a variety of immersive audio formats. Discussions will range from methods for creating and manipulating content for sonic space to local monitoring techniques to 3D production interfaces.

Student Event and Career Development
STUDENT RECORDING CRITIQUES
Thursday, September 29, 4:00 pm – 5:00 pm
Room 515B

Moderators: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA; off-beat-open-hats recording & sound reinforcement
David Greenspan, University of Michigan, Ann Arbor, MI, USA

Students! Come and get pointers, tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system and then receive feedback from a panel of renowned industry professionals. Students should sign up at the student (SDA) booth immediately following the first SDA meeting, and bring mix files on USB memory stick or hard drive to the SDA booth at least two hours prior to the session they sign up for. Files should be AIFF or WAVE, 44.1 Khz, 24 bit. Stereo mixes should be a single interleaved file, up to 5.1 surround mixes should be a set of clearly labeled discrete mono files. Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by PMC and Genelec.

Live Sound Expo 6 **Thursday, September 29**
4:00 pm – 4:45 pm **LSE Stage**

POINT SOURCE OPTIMIZATION: 8 THINGS TO GET RIGHT

Presenter: **David Croxton**

Eight Things to Get Right with Point Source Systems: Height, Splay, Coupling, Fills, Delays, Amplification, Equalization and Rigging.

Recording & Production 4 **Thursday, September 29**
4:15 pm – 5:45 pm **Room 502AB**

IMPLEMENTATION & MIXING FOR VR GAMES AS BOTH ART & SCIENCE (SPECIAL EVENT)

Moderator: **Greg Gordon**, Pyramind, San Francisco, CA, USA

Panelists: *Brennan Anderson*,Senior Producer, Disney Interactive/Pyramind Studios
Sally-Anne Kellaway, Senior Sound Designer, Zero Latency PTY LTD
Viktor Phoenix, Technical Sound Designer, Soundelux
Marc Senasac, Music Engineering Manager, Sony Interactive Entertainment America

Implementing and mixing music and sound for VR games raises many technical and artistic questions. Realism is not always the end-goal as the sonic aesthetics must match the game play in order for the experience to feel cohesive. There is no one correct way to go about this just as there are various tools that can achieve the same or similar results. This panel will explore both the variable aesthetics at play as well as discuss some of the latest platform, middleware and plug in developments being used to achieve them. Topics of discussion will range from implementation of ambisonic and binaural spatialized audio to non-spatialized audio placement, appropriate soundscapes and ambiences, room and environmental effects, suitable volume levels for long term listening, diegetic use of music, and effects such as occlusion through filtering, equalization, and distortion.

Project Studio Expo 6 **Thursday, September 29**
4:30 pm – 5:15 pm **PSE Stage**

LIVING THE DREAM: PROJECT STUDIOS AT BOTH ENDS OF THE SPECTRUM. WHY WE STILL WANT THEM AND WHY WE STILL NEED THEM

Presenters: **Renato Cipriano**, Walters Storyk Design Group, Belo Horizonte, Brazil
John Storyk, Architect, Studio Designer and Principal, Walters-Storyk Design Group, Highland, NY, USA

Synthesizers, laptops, iPhones, Plug-Ins, VR, AR, 3D Audio … Where will this all end up? One thing appears certain—the Project Studio continues to be in demand; continues to be re-defined; and continues to be the Studio of Choice for a large percentage of the audio creation community. The need for small, comfortable, acoustically flexible audio production environments has not diminished in the face of proliferating digital/laptop recording options. But certain basics and certain concerns are still with us and we want to be reminded of them.

This presentation will address:

1. Studio Acoustic Basics for small rooms. This quick refresher will have an eye on isolation, internal room treatments and monitor tuning. Large or small, 2 speakers or 22 speakers—these principles apply.

2. Construction guidelines—the most important concerns in planning and execution.

3. What does all of this cost?

4. Survey of Project Studio case studies, including studios of a wide variety of sizes (and budgets). And, studios that have distinct recording rooms vs. "all in one" mix/recording rooms—some large, some small—one built inside a 140 sq. ft. Airstream Trailer (circa 1958), another built inside a luxurious log cabin deep in a Minnesota forest; another 22 feet below a luxurious new suburban home; and many more.

Thursday, September 29 4:30 pm Room 405

Technical Committee Meeting on High Resolution Audio

Special Event
THE RICHARD C. HEYSER MEMORIAL LECTURE
Thursday, September 29, 6:00 pm – 7:30 pm
Room 502AB

Lecturer: **Dave Smith**, Dave Smith Instruments, Saint Helena, CA, USA

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

The Heyser Lecturer this year is instrument designer, AES Fellow, and Grammy-winner **Dave Smith**—considered to be the "Father of MIDI."

Musical synthesizers first appeared in the 1960s—large modular beasts covered with cables. In the early '70s, portable monophonic instruments became available, leading to a gradual acceptance by musicians in popular music. In the late '70s, fu lly programmable polyphonic analog synths came out, and the synthesizer went mainstream. Things changed dramatically in the '80s as digital synths appeared: first the FM-based DX-7 and eventually the M-1 sample playback synth. From that point onward, digital was the norm. In the '90s, digital synths continued and were implemented in software as computers gained enough power for native signal processing. For 25 years, analog synths were generally not available. Things have changed in the last 10 years, though. Musicians started searching for old analog synths and began using them again. New analog synths became available. Modular synths are back, and very popular. Throughout this 50-year history of the synthesizer its impact on music of all genres has been very significant.

Student Event and Career Development
REAL INDUSTRY / AES STUDENT PARTY
Thursday, September 29, 8:00 pm – 10:00 pm
The Charleston Haus

Audio Students! Join us for a fun and exciting evening at the AES Student Party sponsored by Real Industry. The event will be hosted at The Charleston Haus - 1024 Santee St. 3rd Floor, Los Angeles. Additional details will be announced at SDA1 and the Real Industry mentoring panel. You must RSVP to attend - http://bit.ly/Real-141Party

Special Event
ORGAN CONCERT BY GRAHAM BLYTH
Thursday, September 29, 8:15 pm – 10:00 pm
First Congregational Church of Los Angeles
540 South Commonwealth Ave.
Los Angeles

Organist Graham Blyth's concerts are a highlight of every AES convention at which he performs. This year's recital will be held at First Congregational Church of Los Angeles. He will play •Buxtehude: Prelude and Fugue in D; •Bach: Trio Sonata No. 6; •Bach: Passacaglia and Fugue; •Franck: Fantasie in A; •Vierne: Prelude from Three Pieces; •Boellmann: Suite Gothique.

The Great Organs of First Church, situated in the enormous vaulted Sanctuary of Los Angeles' oldest Protestant Church, together constitute perhaps the largest musical instrument existing in any church in the world today. Now, with approximately 346 ranks, 265 stops, 233 voices, 18 divisions and more than 20,000 pipes, the Great Organs speak down the Nave and Chancel and from the South and North Transept Galleries with the music of the ages.

Since its founding in 1867, First Congregational Church of Los Angeles has played an important role in the musical and cultural life the city. In chambers high on both sides of the Chancel, the **Seeley Wintersmith Mudd Memorial Organ** was constructed and installed by the noted American organ builder, Ernest M. Skinner. Voiced in the style of what came to be known as the "American Classic" school of organ building, the five divisions of that organ served as the church's principal instrument until 1969, when it was greatly enlarged from its original 58 ranks. Unaltered in the 1969 expansion were the sturdy diapasons, lush strings, and the Skinner hallmarks: the romantic flute and reed stops of the Solo division.

The nationally known James W. Fifield, Jr., Senior Minister of First Church for 32 years, and Lloyd Holzgraf, the brilliant Organist in Residence at First Church from 1959 until 1998, envisioned a grand new instrument in the West Gallery of First Church, more than 200 feet from the Main Altar in the Chancel. Thus, the **Frank C. Noon Memorial Organ**, named for the distinguished banker and devout churchman who guided the project to completion, was built by Herman Schlicker, with Clarence Mader and Mr. Holzgraf as consultants. Set in a free-standing case with towering copper pedal pipes on either side of the rose window, the Gallery Organ, with its clean voicing, brilliant ensembles and grand basses in its five divisions, enables the organist to capture the spirit and inspiration of the North German tradition of the 17th century.

The duplicate consoles that grace the Chancel and the West Gallery of First Church are the largest draw-knob consoles ever built in the Western Hemisphere. The Chancel console, which can be moved out into the Chancel for performances, was installed in November, 1992, and was the last masterpiece designed by the venerable Moller firm, which soon closed its doors as a result of financial problems. (Moller knowingly underbid the actual cost of these gigantic consoles so as to have the prestige of designing/building them.) The twin Gallery console, completed by former Moller craftsmen at the Hagerstown Organ Company, was installed a few months later.

Graham Blyth began playing the piano aged 4 and received his early musical training as a Junior Exhibitioner at Trinity College of Music in London, England. Subsequently at Bristol University, where he read Electrical Engineering, he founded the University Music Society, conducting their Chamber Orchestra and Choir. He holds diplomas in Organ Performance from the Royal College of Organists, The Royal College of Music and Trinity College of Music.

In 1973 he founded Soundcraft with Phil Dudderidge, and has been Technical Director from the beginning. In the late 1980s he renewed his musical studies with Sulemita Aronowsky for piano and Robert Munns for organ. He gives numerous concerts each year, principally as organist and pianist, but also as a conductor and harpsichord player. He made his international debut with an organ recital at St. Thomas Church, New York in 1993 and since then has given concerts on many of the finest organs in Europe, including the Madeleine and St. Etienne du Mont in Paris, and the Liebfrauen Dom in Munich, and in North America, including Grace Cathedral, San Francisco and St. Ignatius Loyola in New York.

Graham is currently Artistic Director of Wantage Chamber Concerts and Director of the Wantage Summer Festival. In 1995 he built the Challow Park Recital Hall, an 80 seat auditorium with completely variable acoustics. Today he divides his time between audio engineering and organ design activities. In 2003 he founded the Veritas Organ Company to address the top end of the digital classical organ market, specializing in creating new organs by adding digital voices to existing pipes. He is a Fellow of the Royal Society of Arts and the Audio Engineering Society.

Session P8 — Friday, September 30
9:00 am – 10:30 am — Room 403A

TRANSDUCERS—PART 4

Chair: **D.B. (Don) Keele, Jr.**, DBK Associates and Labs, Bloomington, IN, USA

9:00 am

P8-1 Use of Ground-Plane Constant Beamwidth Transducer (CBT) Loudspeaker Line Arrays for Sound Reinforcement—*D.B. (Don) Keele, Jr.*, DBK Associates and Labs, Bloomington, IN, USA

Ground-plane circular-arc CBT line arrays with wide horizontal coverage offer a very viable, high performance, simple, and thrifty alternative to the usual sound reinforcement setup where loud-speakers are elevated or hung overhead. Due to the broadband constant beam-width / directivity / coverage characteristics and narrow vertical coverage of the CBT array, the ground-plane version offers a number of strong performance and operational advantages even when they are located on stage behind the performers. Among these are: even coverage, minimal front-back variation in sound level, flat-energy response, less energy directed upwards towards ceiling, improved intelligibility, less prone to feedback, and greater performer freedom to move around on stage. In addition, these arrays minimize the use of stage monitors, require minimal in-stallation voicing and on-site equalization adjustments, and result in a much simpler system, i.e., fewer speakers, fewer power amps, and fewer processing channels.
Convention Paper 9623

9:30 am

P8-2 Design of Free-Standing Constant Beamwidth Transducer (CBT) Loudspeaker Line Arrays for Sound Reinforcement—*D.B. (Don) Keele, Jr.*, DBK Associates and Labs, Bloomington, IN, USA

This paper presents design guidelines for choosing the parameters of a free-standing CBT line ar-ray including its physical height, circular arc angle, location, and downward pitch angle to appro-priately cover a single 2D straight-line audience sound-reinforcement listening region with direct sound. These parameters and conditions include: (1) array circular-arc angle and its associated beamwidth, (2) array height and low-frequency beamwidth control limit, (3) array mounting loca-tion that includes its height and setback from the front of the seating plane, and (4) the array's on-axis aiming location and associated downward pitch angle. These parameters are particularly easy to determine in advance for a CBT line array because of the extreme uniformity of its sound field with both frequency and distance, and its inherent constant-directivity characteristics. This paper describes a design scenario that allows the designer to easily choose these system parameters to optimize the direct-field coverage in the prescribed straight-line seating region while minimizing the use of sound-system design and prediction software. The design technique forces the SPL at the front and rear of the listening region to be equal by aiming the array at the rear of the listening re-gion and then choosing its beamwidth (and its associated off-axis rolloff) to provide this front-rear SPL equality. The SPL and frequency response at intermediate points of the covered region are then set by the inherent well-behaved off-axis rolloff of the CBT array.
Convention Paper 9624

10:00 am

P8-3 Constant Coverage Line Arrays Using Passive Components for Beamforming—*Douglas J. Button*, Harman International, Simi Valley, CA, USA

The work here within describes a cost effective method for beamforming utilizing passive compo-nents in a transmission line architecture to provide successive amounts of group delay from the middle to the ends of the array. The method also provides amplitude shading and some frequency shading that works to form a hybrid of several traditional methods. The method also provides a simple and straight forward way to change the shape (width) of the beam with changes in the pas-sive network. The resulting network is very cost effective and rivals the performance of a multi-channel DSP-based beamformer.
Convention Paper 9625

Session P9 — Friday, September 30
9:00 am – 10:30 am — Room 409B

SEMANTIC AUDIO & SONIFICATION

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

9:00 am

P9-1 The Audio Definition Model—A Flexible Standardized Representation for Next Generation Audio Content in Broadcasting and Beyond—*Simone Füg,[1] David Marston,[2] Scott Norcross[3]*
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]BBC R&D, London, UK
[3]Dolby Laboratories, San Francisco, CA, USA

Audio for broadcasting around the world is evolving towards new immersive and personalized audio formats, which require accompanying metadata along with the audio essence. This paper introduces the Audio Definition Model (ADM), as specified in Recommendation ITU-R BS.2076 (and EBU Tech 3364), which is an XML-based generalized metadata model that can be used to define the required metadata. The ADM is able to describe channel-based, object-based, and scene-based audio in a formalized way. The ADM can be

incorporated into RIFF/WAV-based files, such as BW64 (Recommendation ITU-R BS.2088), and can therefore be deployed in RIFF/WAV-based applications, handling immersive content, while maintaining compatibility to legacy content. This allows for program production and exchange of audio programs in these new audio formats.
Convention Paper 9626

9:30 am

P9-2 Development of Semantic Scales for Music Mastering— *Esben Skovenborg*, TC Electronic, Risskov, Denmark

Mastering is the last stage in a music production, and entails modifications of the music's spectral and dynamic properties. This paper presents the development of a set of semantic scales to characterize the (change in) perceptual properties of the sound associated with mastering. An experiment was conducted with audio engineers as subjects. Verbal elicitation and refinement procedures resulted in a list of 30 attributes. Next, 70 unmastered music segments were rated on scales corresponding to the attributes. Based on clustering and statistics of the responses, groups of similar and opposite attributes were formed. The outcome was a set of seven bipolar semantic scales. These scales could be used in semantic differentiation, to rate typical alterations of sound caused or de-sired by mastering.
Convention Paper 9627

10:00 am

P9-3 A Hierarchical Sonification Framework Based on Convolutional Neural Network Modeling of Musical Genre—*Shijia Geng, Gang Ren, Mitsunori Ogihara*, University of Miami, Coral Gables, FL, USA

Convolutional neural networks have satisfactory discriminative performances for various music-related tasks. However, the models are implemented as "black boxes" and thus their processed representations are non-transparent for manual interactions. In this paper, a hierarchical sonification framework with a musical genre modeling module and a sample-level sonification module has been implemented for aural interaction. The modeling module trains a convolutional neural network from musical signal segments with genre labels. Then the sonification module performs sample-level modification according to each convolutional layer, where lower sonification levels produce auralized pulses and higher sonification levels produce audio signals similar to the input musical signal. The usage of the proposed sonification framework is demonstrated using a musical stylistic morphing example.
Convention Paper 9628

Tutorial 3 — Friday, September 30
9:00 am – 10:30 am — Room 404AB

LISTENING TESTS—UNDERSTANDING THE BASIC CONCEPT

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

Broadcast/Streaming Media 5 — Friday, September 30
9:00 am – 10:30 am — Room 408A

AUDIO CONSIDERATIONS FOR 4K AND 8K TELEVISION

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

Panelists: *Roger Charlesworth*, DTV Audio Group, New York, NY, USA
Kazuho Ono, NHK Science & Tech. Research Laboratories, Tokyo, Japan
Skip Pizzi, NAB, Washington, D.C., USA
Peter Poers, Junger Audio GmbH, Berlin, Germany

4k and 8k UHD broadcasting and streaming is edging beyond standards creation and quickly be-coming ubiquitous, with a more than tenfold increase in consumer product over the past year. NHK and U.S. broadcasters are on the air with ATSC 3.0 and Super Hi-Vision experimental transmitters, while cable and satellite providers are already delivering movies and sports; from Su-per bowl 50, The Master's Tournament, and the Summer Games. Immersive 3D and object orient-ed audio adds a striking polish to wider visual resolution, gamut, and dynamic range. This session has been very popular in past years and we have another group of impressive speakers from the myriad facets of audio production and delivery.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Product Development 3 — Friday, September 30
9:00 am – 10:30 am — Room 408B

WHAT HAPPENS IN A PATENT LAWSUIT

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

Recording & Production 5 — Friday, September 30
9:00 am – 10:30 am — Room 501ABC

MASTERING FOR VINYL

Moderator: **Jim Kaiser**, CEMB / Belmont University, Nashville, TN, USA

Panelists: *Eric Boulanger*, The Bakery
Bernie Grundman, Bernie Grundman Mastering
Cameron Henry, Welcome To 1979, Nashville, TN, USA
Chris Mara, Welcome to 1979 - Nashville, TN, USA
Ron McMaster, Capitol Mastering
Jeff Powell, Memphis, TN, USA

Mastering engineers elaborate on the specific requirements for preparing a music project for release on vinyl, with tips for making a great sounding disc.

Sound Reinforcement 5 Friday, September 30
9:00 am – 10:30 am Room 402AB

SOUND SYSTEMS FOR REVERBERANT AND CHALLENGING ACOUSTIC SPACES

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

This tutorial discusses what constitutes a difficult acoustic space and the effects of reverberation, echo, and noise on speech intelligibility. Sound system Intelligibility measurement methods and techniques will be reviewed, while system optimization techniques are also examined. The tutorial includes numerous case histories and examples illustrating the techniques discussed. A fundamental question that will be raised will be "how intelligible do sound systems need to be and does this change with application or is intelligibility universal." The case histories and examples will include churches, cathedrals and houses of worship, concert halls, ceremonial halls / banqueting halls, shopping malls, sports facilities and arenas, railway & metro stations, road tunnels and airports. The challenges associated with each type of venue will be discussed, as will the thorny subject of aesthetics (i.e., architects) and loudspeaker size and placement.

Special Event
OK, YOU DID NOT GET THE GIG AT THE STUDIO. WHERE ARE THE JOBS?
Friday, September 30, 9:00 am – 10:30 am
Room 502AB

Moderator: **Lisa Nigris**, New England Conservatory, Boston, MA, USA; Aspen Music Festival and School, Aspen, CO, USA

Panelists: *George Adjieff*, Westlake Professional
Scott Esterson, Adam Speaker
Steve Harvey, Pro Sound News
George Horton, Solid State Logic
Ted Leamy, Ultra Sound

You have all this great training and developing skillsets. Where are the jobs? There are plenty out there. You need to think a little differently. These presenters will talk about different related fields.

Game Audio 4 Friday, September 30
9:30 am – 10:30 am Room 406AB

IMPACT OF IMMERSIVE AUDIO FOR TODAY'S GAMES

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

Games are a massive headphone application. Beyond common stereo is virtualization (binaural) a huge new issue. More and more gamers understand that current virtualization technologies offer a fantastic new experience far away from creepy low quality "surround simulations" in the past. But game adventures can also be played on new immersive home theater environments. The presentation shows current strategies and applications for multichannel game adventures today and in the future.

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

Friday, September 30 9:30 am Room 405

Technical Committee Meeting on Sound for Digital Cinema and Television

Project Studio Expo 7 Friday, September 30
10:30 am – 11:15 am PSE Stage

USING NOISE REDUCTION SOFTWARE FOR TRACKING AND MIXING MUSIC

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

Noise Reduction and Spectral Editing software has come a long way in performance and price in recent years, and are no longer only tools for post production and audio forensics. In this presentation, *Tape Op* editor Larry Crane will show you how to use this software to repair and prep sources for mixing, including vocals, acoustic guitars, piano, drums, bass, amplifiers, organs, and more. Larry has been integrating these techniques into his tracking and mixing workflow in recent years at his studio, Jackpot! Recording, in Portland, OR, and has featured these concepts in his Lynda.com courses as well.

Session P10 Friday, September 30
10:45 am – 12:15 pm Room 409B

CINEMA SOUND AND FORENSIC AUDIO

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

10:45 am

P10-1 Wideband Audio Recordings of Gunshots: Waveforms and Repeatability—*Rob Maher, Tushar Routh*, Montana State University, Bozeman, MT, USA

For the purposes of audio forensics research we have obtained multichannel acoustical recordings of gunshots under controlled conditions for several firearms. The recordings are made using an elevated platform and an elevated spatial array of microphones to provide quasi-anechoic direction-al recordings of the muzzle blast. The consistency and repeatability of gunshot sounds is relevant to many areas of forensic analysis. This paper includes a description of the recording process and a summary comparison of the acoustical waveforms obtained from ten successive shots by the same firearm by an experienced marksman. Practical examples and applications are presented.
Convention Paper 9634

11:15 am

P10-2 Integration of CGI Information on Audio Post

Production—*Nuno Fonseca*, ESTG/Polytechnic Institute of Leiria, Leiria, Portugal; Sound Particles, Leiria, Portugal

Although CGI is a common tool in cinema, for both VFX shots and animation features, all that 3D information is disregarded on audio post production, which usually only uses the final image as reference. This paper presents a workflow that uses CGI information to help audio post-production work. Working on top of "Sound Particles" software, a 3D CGI-like software for audio applica-tions currently used at major Hollywood studios, CGI information is used to automatically control several audio parameters (volume, 3D position, Doppler, etc.), while maintaining full creativity freedom.
Convention Paper 9633

11:45 am

P10-3 Intelligibility of Cinema & TV Sound Dialogue—
Peter Mapp, Peter Mapp Associates, Colchester, Essex, UK

In recent years there has been a significant increase in the number of complaints concerning the dialogue intelligibility of both movie sound tracks and TV productions—both in Europe and the USA. The paper reviews the background to dialogue intelligibility and looks at a number of mechanisms that may be responsible for the growing trend of dissatisfaction. The transmission chain is re-viewed and new measurements and data concerning domestic listening conditions are presented. The results of a pilot measurement program show that in-situ frequency response of the TV sys-tems, operated by many domestic listeners, is far from ideal with response variations of 10–15 dB being common. Unique Speech Transmission Index (STI) and Clarity data are presented that sug-gest that the room acoustic conditions of the listening environment should not, in themselves, sig-nificantly degrade the received signal.
Convention Paper 9632

Session P11 Friday, September 30
10:45 am – 12:15 pm Room 403A

ACOUSTICS

Chair: **Alejandro Bidondo**, Universidad Nacional de Tres de Febrero - UNTREF, Caseros, Buenos Aires, Argentina

10:45 am

P11-1 The Influence of Discrete Arriving Reflections on Perceived Intelligibility and Speech Transmission Index Measurements—*Ross Hammond*,[1] Peter Mapp,[2] *Adam J. Hill*[1]
[1] University of Derby, Derby, Derbyshire, UK
[2]Peter Mapp Associates, Colchester, UK

The most widely used objective intelligibility measurement method, the Speech Transmission Index (STI), does not completely match the highly complex auditory perception and human hearing system. Investigations were made into the impact of discrete reflections (with varying arrival times and amplitudes) on STI scores, subjective intelligibility, and the subjective "annoyance factor." This allows the effect of comb filtering on the modulation transfer function matrix to be displayed, as well as demonstrates how the perceptual effects of a discrete delay cause subjective "annoyance," that is not necessarily mirrored by STI. This work provides evidence showing why STI should not be the sole verification method within public address and emergency announcement systems, where temporal properties also need thoughtful consideration.
Convention Paper 9629

11:15 am

P11-2 Spatial Stability of the Frequency Response Estimate and the Benefit of Spatial Averaging—*Aki Mäkivirta, Thomas Lund*, Genelec Oy, Iisalmi, Finland

In-room estimates of loudspeaker responses at the listening location are typically taken either at one microphone location, replacing the listener with a microphone, or averaging in space, at multiple microphone locations at and relatively close to the listening location. In-frequency averaging can attenuate the locality of the frequency response features in mid and high frequencies. In-space averaging extracts the common frequency response features visible in all the measurement positions. Spatial weighting combined with frequency domain averaging can increase the stability of the frequency response estimate for the features relevant for the subjective compensation of the sound color at the listening location. Spacing out the spatial average measurement points affects the nature of the spatial average and the focus on the frequency response features common to the measurement points. The spatial averaging points used in taking a measurement should be chosen based on the intention of the room equalization.
Convention Paper 9630

11:45 am

P11-3 A New and Simple Method to Define the Time Limit between the Early and Late Sound Fields—
Alejandro Bidondo, Javier Vazquez, Sergio Vazquez, Mariano Arouxet, Germán Heinze, Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina

In room acoustics the crossover time is defined as the transition period between a clearly deterministic regime (early sound field), to a stochastic, memoryless one (reverberant tale or late sound field). Several studies presented different calculation methods applied to impulse responses like running gaussianity test, running kurtosis, eXtensible Fourier Transform, and Matching Pursuit. No clear or "binary moment" was found analyzing the room's responses, so another new, simple, and robust method is proposed to determine the limiting instant between both sound fields using the autocorrelation function. Conclusions also include the analysis of several rooms and comments on the progressive change of the room's system behavior.
Convention Paper 9631

Tutorial 4 Friday, September 30
10:45 am – 12:15 pm Room 404AB

PODCASTING—TELLING YOUR STORY WITH SOUND

Presenter: **Jim Anderson**, New York University, New York, NY, USA

A podcast can be more than a monologue or an interview; it can be a rich environment for using sound to tell your story. One can use sound in many ways, whether it's to set the scene, illustrate a concept, or enliven a journalistic endeavor. The talk will take the audience on an international aural journey from the 'hollars' of Kentucky to the streets of Grenada in search of sounds. With Jim Anderson's deep background in broadcasting, he will demonstrate the power of sound to illustrate and enrich a podcast.

Workshop 4 **Friday, September 30**
10:45 am – 12:15 pm **Room408A**

CREATIVE WAYS TO USE AN ANALOG TAPE MACHINE WITH A DAW

Presenters: **Chris Mara**, Mara Machine,. Nashville, TN, USA
Dan Labrie, ATR Services/ATR Magnetics, York, PA, USA

We all are aware of the basic ways to use analog tape machines in traditional recording studios. Spend some time with Chris Mara (Mara Machines) and Dan Labrie (ATR Services) to explore creative uses of analog tape machines with DAWs without the hassles of synchronization. They'll be focusing on the future of tape machines & their continued relevance as the technology around us changes. Expect tons of Q&A.

Game Audio 5 **Friday, September 30**
10:45 am – 12:15 pm **Room 406AB**

ADAPTING TRADITIONAL GAME AUDIO FOR VR EXPERIENCES: A BOUND POST-MORTEM

Presenter: **Daniel Birczynski**, Senior Sound Designer, Sony Computer Entertainment America - Santa Monica Studio, Santa Monica, CA, USA

Bound transports you into a beautiful, fantastical world that exists in the mind of a woman revisiting the memories of her childhood. The game's style and mechanics presented audio with unique challenges and led the team to learn valuable lessons. This session will present the findings in a post-mortem format as well as share techniques used while developing a game to be release in both VR and traditional formats.

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

Networked Audio 3 **Friday, September 30**
10:45 am – 12:15 pm **Room 402AB**

AES67 AND THE AUDIO INDUSTRY

Moderator: **Rich Zwiebel**, QSC, Boulder, CO, USA; K2

Panelists: *TBA*

A panel discussion of representatives from manufacturers. 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.

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

Product Development 4 **Friday, September 30**
10:45 am – 12:15 pm **Room 408B**

PRACTICAL LOUDSPEAKER PROCESSING FOR THE PRACTICING ENGINEER

Presenter: **Paul Beckmann**, DSP Concepts, Inc., Santa Clara, 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.

Recording & Production 6 **Friday, September 30**
10:45 am – 12:15 pm **Room 501ABC**

RAW TRACKS: BECK "MORNING PHASE"

Moderator: **Jonathan Pines**

Presenter: **Darrell F. Thorp**

Multi Grammy winning Producer / Engineer Darrell F. Thorp will peal back the green curtain, and tell us how he approaches working with multi platinum artists Beck, Radiohead, and McCartney. He will have "raw tracks" from the influential Beck record "Morning Phase" which won the trifecta of Album of the Year, Best Rock Record, AND Best Engineered [Non Classical] in 2015. Darrell will show how he crafts these amazing records, and what it's like to be in the studio with legends Nigel Goodrich, Mac, Beck, and the new legends of Nashville. Producer / Engineer Jonathan Pines, Director of Strategic Operations for Rupert Neve Designs, is very excited to guide this lively discussion with top professionals for the third year in a row!

Special Event
GRAMMY SOUNDTABLE
Friday, September 30, 10:45 am – 12:15 pm
Room 502AB

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

Panelists: *Leslie Ann Jones*
Jeri Palumbo, NFL/NBA/MLB/NASCAR
Paul Sandweiss
Eric Schilling
James Stoffo

Life in the Hotseat—Audio Production for Live Global Telecast Events

GRAMMY, EMMY, OSCAR, Super Bowl, NFL, NBA, MLB, NASCAR, live broadcasts that capture a global audience with myriad moving parts and zero margin for error. Going live with tens of millions of viewers hanging on every note and play, there are no second chances to get it right. Join members of the most experienced broadcast audio teams in the business as they pull back the curtain on the most technically advanced and logistically challenging audio productions on the planet.

Student Event and Career Development
STUDENT DESIGN COMPETITION
Friday, September 30, 10:45 am – 12:45 pm
Room 515A

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: Colin Broad, Charlie DeVane, Doug Fearn, Erik Gaskell, Dave Hill, Mike Mabie, Charlie Slee, Andrew W. Smith.

Student Event and Career Development
MIC IT LIKE YOU MEAN IT!
Friday, September 30, 11:00 am – 12:30 pm
Room 511A
Presenters: **Lenise Bent**, Independent Engineer, Producer, Post-Production, Los Angeles, CA, USA
Ian Corbett, Kansas City Kansas Community College, Kansas City, KS, USA; off-beat-open-hats recording & sound reinforcement
Mark Rubel, The Blackbird Academy, Nashville, TN, USA; Pogo Studio, Nashville, TN, USA

You should not start a recording project without knowing how you want the mix to sound! Come and learn how to approach your recording session from the perspective of the mix, and to capture the sound you really need for the mix. You will not only produce more spacious multi-dimensional mixes, but also create a more efficient workflow that speeds up your mixing process. Topics explored will include how different microphone technologies can beneficially color and shape the sounds you're recording, how different instrument, vocal, and stereo mic techniques affect the sounds captured, and how the recording room's effect on the sound source and the microphones can be beneficially exploited and explored. For students and professionals alike, understanding these techniques will increase the variety of mix styles and sounds you are able to produce, making you a more versatile audio engineer ready to meet the needs of clients and employers today and tomorrow

Live Sound Expo 7 **Friday, September 30**
11:00 am – 11:45 am **LSE Stage**

LIVE MIXING ACADEMY—MONITOR MIXING APPS

Presenters: **Chris Brouelette**
Kevin Kimmell
Joe Mendez

Remote control allows a dual-function console to be operated by two or more users to make adjustments from meaningful locations, in the house or on stage, replacing personal mixers or a second console.

Friday, September 30 **11:00 am** **Room 405**

Technical Committee Meeting on Loudspeakers and Headphones

Friday, September 30 **11:00 am** **Room 508A**

Standards Committee Meeting SC-04-04, Microphones

Project Studio Expo 8 **Friday, September 30**
11:30 am – 12:15 pm **PSE Stage**

FOCUSRITE RED 4PRE AND DANTE BASICS

Presenter: **Matt Pliskin**, Focusrite RedNet, El Segundo, CA, USA

Focusrite's finest mic pres yet!

Live Sound Expo 8 **Friday, September 30**
12:00 noon – 12:45 pm **LSE Stage**

THE GRAND MOTHER: MIKING THE PIANO

Presenters: **Ken Newman**, Newman Audio, Inc., Canyon Country, CA, USA
Jeramiah Slovarp

The grand piano is the Mother of Musical Creation, every singer's lover and the heart of most genres, yet it's often replaced by synthesizers due to its inconvenient size and a propensity to resonate with adjacent sounds.

Friday, September 30 **12:15 pm** **Room 408A**

Technical Committee Meeting on Broadcast and On-Line Delivery

Project Studio Expo 9 **Friday, September 30**
12:30 pm – 1:15 pm **PSE Stage**

HOW TO GET GREAT SOUNDS FROM AMP SIMS

Presenter: **Craig Anderton**, Gibson Brands, Nashville, TN, USA; Harmony Central.com

If you like amp sims that sound harsh and cold, you can skip this workshop. But if you prefer warm, creamy amps that respond to the way you play, this workshop shows how placing certain effects before and after a sim, in addition to specific amp sim techniques, can make playing with amp sims an inspiring experience—and let you create effects that would be difficult, if not impossible, to realize with conventional hardware.

Student Event and Career Development
EDUCATION AND CAREER/JOB FAIR
Friday, September 30, 1:00 pm – 2:30 pm
Room 515A

The combined AES 141st 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 141st 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 9 **Friday, September 30**
1:00 pm – 1:45 pm **LSE Stage**

LIVE MIXING ACADEMY—ROCK AUTOMATION

Presenter: **Robert Scovill**, Avid Technologies, Scottsdale, AZ, USA; Eldon's Boy Productions Inc.

Snapshots automate consoles, with a scene for each song in a set list. Each channel's recall "scope" determines what parameters remain untouched. Including only mutes or just faders creates an analog workflow, while automating each tune's "starting point." Scenes can be examined "blind" and parameters in multiple scenes can be adjusted at once. Effects and plugins update song by song and recall extends to VCA and Group assignments, allowing creative use of routing and flexible VCA mix control.

Session P12 **Friday, September 30**
1:30 pm – 3:00 pm **Room 403A**

APPLICATIONS IN AUDIO—PART 1

Chair: **Bob Schulein**, ImmersAV Technology, Schaumburg, IL, USA

1:30 pm

P12-1 Development of Shotgun Microphone with Extra-Long Leaky Acoustic Tube—*Yo Sasaki,[1] Toshiyuki Nishiguchi,[1] Kazuho Ono,[1] Takeshi Ishii,[2] Yutaka Chiba,[2] Akira Morita[2]*
[1]NHK Science & Technology Research Laboratories, Kinuta, Setagaya-ku, Tokyo, Japan
[2]Sanken Microphone Co. Ltd., Tokyo, Japan

A shotgun microphone having sharper directivity than a conventional microphone has been studied to capture distant sound clearly. The directivity of the shotgun microphone is known to become sharper with a longer leaky acoustic tube. Thus, we developed a prototype microphone that uses a 1-m-long leaky acoustic tube, which is longer than the conventional one. We also conducted a numerical simulation based on an acoustical distribution constant circuit to develop such a shotgun microphone. The measurement results of the prototype microphone were in fairly good agreement with the simulation results, and they showed that the directivity is very narrow.
Convention Paper 9639

2:00 pm

P12-2 Non-Traditional Noise and Distortion Sources in Headphones and Earbuds and Their Impact on System Performance and Ear Fatigue—*Dennis Rauschmayer*, REVx Technologies/REV33, Austin, TX, USA

Non-Traditional dynamic noises and distortions that couple into earbuds and headphones are measured and analyzed. The effective noise and distortion level relative to the signal level for a number of commercially available headphones and earbuds are reported. The impact of this noise and distortion on listener experience is quantified and discussed. Finally, the impact on listener fatigue is quantified. Results presented show that the non-traditional sources studied are significant, audible and that they present a fundamental limitation to the performance of many earbuds and headphones. In many cases, these sources reduce the effective signal to noise ratio (SNR) of the audio system into the 30-40 dB range, well below the SNR level that would result from by a 0.1% total harmonic distortion (THD) system and even below that of a 1% THD system. In addition to limiting system performance, the non-traditional noise and distortion are found to increase ear fatigue experienced by the user when ear fatigue tests are conducted with identical audio levels. Otoacoustic Emission (OAE) results from tested subjects show degradation that is significantly greater when the impairments are present vs. when they are mitigated.
Convention Paper 9640

2:30 pm

P12-3 In Situ Subjective and Objective Acoustic Seal Performance Tests for Insert Earphones—
Bob Schulein,[1] Brian Fligor[2]
[1]ImmersAV Technology, Schaumburg, IL, USA
[2]Lantos Technologies Inc., Wakefield, MA, USA

Insert–type earphones are unique in that when tightly sealed in an ear canal, they have the potential to deliver sound down to 20 Hz and below. In practice however, obtaining an extended low frequency response is challenged by a difficulty for users to achieve an adequate seal between the ear canal and the insert earphone [1], [2], [3]. Achieving such a seal is not intuitive to users new to insert earphones. Measurements made on actual earphones show that a leak as small as .5 mm in diameter and 2.5 mm long can result in a reduction of bass at 50 Hz of approximately 15 dB. Such a loss results when the leak is present, since the actual volume seen by the transducer at low frequencies is considerably greater than for higher frequencies, where the impedance of the leak becomes much higher in value. The result is a very perceptible and often disappointing reduction in bass performance. This paper describes two methods by which a user can confirm the level of acoustic seal obtained for a given combination of transducer, ear tip / ear mold and ear canal by subjective and objective means. The subjective method is based on an experimental observation that even with a poor seal the output of insert earphones at 500 Hz, is quite independent of seal quality. By consequently subjectively comparing the output of a recorded tone at 500 Hz to one at 50 Hz adjusted to be equal in perceived level based on ISO 226:2003 equal-loudness contours, one can confirm a good seal when the levels tend to subjectively match. The objective method involves fitting an insert earphone with a miniature flat pressure microphone into the sound port, and observing the "in canal" frequency response by means of spectral analysis, while the earphone is in use. As the seal quality improves the measured low frequency response approaches that of the coupler response of the earphone as measured with a high quality seal.
Convention Paper 9641

Session P13 **Friday, September 30**
1:30 pm – 3:00 pm **Room 403B**

POSTERS: PERCEPTION AND FORENSIC AUDIO

1:30 pm

P13-1 Determining the Muzzle Blast Duration and Acoustical Energy of Quasi-Anechoic Gunshot Recordings—*Tushar Routh, Rob Maher*, Montana State University, Bozeman, MT, USA

Investigation of gunshot waveforms largely includes analyzing the muzzle blast. Generated by the combustion of

gunpowder immediately after firing, these brief duration directional shock waves travel outward in all directions at the speed of sound. Features of these waveforms are analyzed to identify characteristics of a particular shot, for example, the combination of firearm type, ammunition, and orientation. This paper includes measured muzzle blast durations for several common firearms and calculation of the total acoustical energy during the muzzle blast period.
Convention Paper 9635

1:30 pm

P13-2 Analysis and Localization for ENF Signals in the Tokyo Metropolitan Area—*Akira Nishimura*, Tokyo Univeristy Information Sciences, Chiba-shi, Japan

This paper addresses the first investigation and analysis of electronic network frequency (ENF) signals in the Tokyo metropolitan area, Japan. Electric power signals are recorded directly from a clean power line at seven different sites simultaneously for several weeks. Instantaneous frequency measurements based on time-domain analytic signals are performed on band-pass-filtered electric power signals, therein providing higher temporal resolution compared with the conventional FFT-based method combined with quadratic interpolation for extracting ENFs. Spectro-temporal analysis of the fluctuations of the ENF signals reveals that temporal correlations between the fluctuation energy in the frequency range of 0.4 Hz to 1.0 Hz obtained at different sites are inversely correlated to the geographical distances between the sites. The similarities of the spectro-temporal ENFs obtained from different sites show generally higher correlations with the geographical distances than the similarities of high-pass-filtered ENFs. Location estimation using linear regression between the similarities of spectro-temporal ENFs and the geographical distances of the anchor sites predicts the location of a target site with a mean prediction error of approximately 20 to 30 km.
Convention Paper 9636

1:30 pm

P13-3 Does Environmental Noise Influence Preference of Background-Foreground Audio Balance?—*Tim Walton*,[1,2] *Michael Evans*,[2] *David Kirk*,[1] *Frank Melchior*[2]
[1]Newcastle University, Newcastle-Upon-Tyne, UK
[2]BBC Research and Development, Salford, UK

With an increase in the consumption of mobile media, audio is being consumed in a range of contexts. The literature describes several techniques to improve the experience of mobile listening by utilizing information about the environmental noise of the listening environment, however, none of the previous work utilizes object-based audio. This paper investigates the possibility of using object-based audio to improve the experience of mobile listening by investigating whether environmental noise influences preference of background-foreground audio balance. A listening test was carried out in which listeners were asked to adjust the background-foreground balance to their preference while in the presence of reproduced environmental noise. It was found that environmental noise can have a significant effect on preferred background-foreground balance.
Convention Paper 9637

1:30 pm

P13-4 Evaluation of a Perceptually-Based Model of "Punch" with Music Material—*Steven Fenton, Hyunkook Lee, Jonathan Wakefield*, University of Huddersfield, Huddersfield, UK

This paper evaluates a perceptually motivated objective model for the measurement of "punch" in musical signals. Punch is a perceptual attribute that is often used to characterize music that conveys a sense of dynamic power or weight. A methodology is employed that combines signal separation, onset detection, and low level parameter measurement to produce a perceptually weighted "punch" score. The model is evaluated against subjective scores derived through a forced pairwise comparison listening test using a wide variety of musical stimuli. The model output indicates a high degree of correlation with the subjective scores. Correlation results are also compared to other objective models such as Crest Factor, Inter-Band-Ratio (IBR), Peak-to-Loudness Ratio (PLR), and Loudness Dynamic Range (LDR).
Convention Paper 9638

Session EB2 Friday, September 30
1:30 pm – 3:15 pm Room 409B

LECTURE: PRODUCTION AND BROADCAST

Chair: **Amandine Pras**, CNSMDP, Paris, France

1:30 pm

EB2-1 A Broadcast Film Leader with Audio Channel, Frequency, and Synchronism Test Properties—*Luiz Fernando Kruszielski, Rodrigo Meirelles*, Globo TV Network, Rio de Janeiro, Brazil

Universal film leaders, commonly known as "countdowns," have been an important tool to synch audio and video. In a broadcast production, the material goes through several stages where audio and video are edited and processed, and time is a very precious resource. Also, it is important to minimize possible errors in the production chain. We propose a film leader format that, in a single 10 second clip, would be possible to do a preliminary check on aspects such as surround and stereo channel identification, relative channel level and frequency response, as well as synchronism. The proposed film leader has been tested and integrated in a Brazilian Television Network with very good results.
Engineering Brief 286

1:45 pm

EB2-2 Live vs. Edited Studio Recordings: What Do We Prefer?—*Amandine Pras*, Paris Conservatoire (CNSMDP), Paris, France; Stetson University, DeLand, FL, USA

This pilot study examines a common belief in written classical music that a live recording conveys a more expressive musical performance than a technically flawless studio production. Two tonmeister students of the Paris Conservatoire recorded a six-dance baroque suite and a four-movement romantic sonata in concert and in studio sessions with the same microphone techniques and in the same venue for both conditions. Twenty listeners completed an online survey to rate three versions of the dances and movements, i.e., the concert performance, the firstt studio take, and the edited version. Results show that listeners preferred the edited versions (44%) more often than the firstt studio takes (29%) and the concert performances (27%).
Engineering Brief 287

2:00 pm

EB2-3 Rondo360: Dysonics' Spatial Audio Post-Production Toolkit for 360 Media—*Robert Dalton*, *Jimmy Tobin*, *David Grunzweig,* Dysonics, San Francisco, CA, USA

Rondo360 is Dysonics' toolkit for spatial audio post-production, supporting multiple workflows including multichannel, Ambisonics, and Dysonics' own native 360 Motion-Tracked Binaural (MTB) format. Rondo360 works with all input formats—live or prerecorded—from traditional or sound field microphones, and exports to a wide array of formats depending on desired content distribution. Rondo360 integrates seamlessly with all DAWs by adding a final layer onto the creator's existing workflow, and it comes bundled with a suite of custom mastering tools (Mixer, Compressor, Limiter, and Reverb) that work on multichannel sound field content. With support for RondoMotion, Dysonics' wireless head-tracking device, creators can monitor their 360 mixes in real-time. Rondo360 also provides an intuitive audio/video sync and export functionality along with live broadcasting support
Engineering Brief 288

Engineering Brief 289 was withdrawn

2:15 pm

EB2-5 Mixing Hip-Hop with Distortion—*Paul "Willie Green" Womack*, Willie Green Music, Brooklyn, NY, USA

The grit and grime of Hip-Hop doesn't have to be metaphorical. With the vast array of saturation tools available, distortion is no longer just something to remove from recordings; and the huge and aggressive sounds in Hip-Hop music can benefit specifically. From subtly warming drums and keyboards to mangling vocals and samples, this brief will demonstrate techniques for creatively distorting urban music. Exploring tape emulation, parallel vocal distortion, drum crushing, and more, I will investigate how a bit of dirt can drastically affect a mix.
Engineering Brief 290

2:30 pm

EB2-6 Smart Audio Is the Way Forward for Live Broadcast Production—*Peter Poers*, Junger Audio GmbH, Berlin, Germany

Today's broadcast facilities are facing ever-increasing demands on their resources as they strive to keep up with consumers who expect more content on more devices both where and when they want it. To attract and retain viewers, consistent, stable, and coherent audio is a vital requirement. One aspect that is particularly important to pay attention to is speech intelligibility. This is most critical and difficult in a live broadcast situation. The Smart Audio concept is to utilizing real time processing algorithms that are both intelligent and adaptive. Devices need to be fully interoperable with others in the broadcast environment and need to seamlessly integrate with both playout automation systems and logging and monitoring processes. The Engineering Brief will present some dedicated and proofed algorithms and practical use cases for Smart Audio.
Engineering Brief 291

2:45 pm

EB2-7 Towards Improving Overview and Metering through Visualization and Dynamic Query Filters for User Interfaces Implementing the Stage Metaphor for Music Mixing—*Steven Gelineck*,[1] *Anders Kirk-Uhrenholt*[2]
[1]Aalborg University Copenhagen, Copenhagen, Denmark
[2] Copenhagen University, Copenhagen, Denmark

This paper deals with challenges involved with implementing the stage metaphor control scheme for mixing music. Recent studies suggest that the stage metaphor outperforms the traditional channel-strip metaphor in several different ways. However, the implementation of the stage metaphor poses issues including clutter, lack of overview and monitoring of levels, and EQ. Drawing upon suggestions in recent studies, the paper describes the implementation of a stage metaphor prototype incorporating several features for dealing with these issues, including level and EQ monitoring using brightness, shape, and size. Moreover we explore the potential of using Dynamic Query filtering for localizing channels with certain properties of interest. Finally, an explorative user evaluation compares different variations of the prototype, leading to a discussion of the importance of each feature.
Engineering Brief 292

Workshop 5 — Friday, September 30
1:30 pm – 3:00 pm — Room 404AB

**THE DREADED SIGN-OFF:
WHEN CAN I CALL THIS SONG FINISHED?**

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

Panelists: *George Massenburg*, Schulich School of Music, 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, Eemnes, Netherlands
Darcy Proper, Darcy Proper Mastering, Eemnes, The Netherlands; Wisseloord Studios, Hilversum, The Netherlands
Michael Romanowski, Coast Mastering, Berkely, CA, USA; The Tape Project

There are many challenges with signing-off a song. Digital technologies have allowed final decisions to be pushed further back in the production process. There are clearly benefits to these tools, allowing songs to be restructured during mixing or to correct specific issues that emerge during mastering, for example. Critical decision-making, however, is an integral part of all processes in music production, from songwriting and composition to tracking, mixing and mastering. The sign-off process is a fearful and intrepid one which many artists, A&R and producers find challenging; wrestling with their own definitions of 'perfect' and coming to terms with potential deficiencies in their process or tools. In this workshop, a number of esteemed recording, mixing and mastering engineers will discuss their own experiences with signing-off.

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

Broadcast/Streaming Media 6 — Friday, September 30
1:30 pm – 3:00 pm — Room 408A

INTERVIEW WITH BOB ORBAN

Moderator: **Greg Ogonowski**, Orban, San Leandro, CA, USA

Presenter: **Robert Orban**, Orban, San Leandro, CA, USA

Bob Orban is best known in the professional broadcast industry for the Orban Optimod FM audio processor. It introduced patented non-overshooting lowpass filters to the FM MPX audio chain, dramatically increasing loudness and signal to noise ratio without audible side-effects. While Opti-mod could be used as a lethal weapon in the early FM loudness wars, Bob's original goal was cre-ate a processor with much lower distortion that anything then available. Optimod was a powerful tool that didn't always need to run at "11" to achieve its goals. This was a true game-changer for FM Radio broadcasting.

In addition, Bob created innovative recording studio and production tools, such as the Orban Ste-reo Synthesizer, de-essers and reverbs. Bob has also been an active participant in the National Radio Systems Committee.

This interview by longtime friend and colleague Greg Ogonowski will discuss the vast progress in audio processing technology that has shaped and molded delivery of audio for broadcast and streaming throughout the world. Join us for a true trip down tech memory lane, and up to current developments in broadcast audio processing.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Game Audio 6 **Friday, September 30**
1:30 pm – 3:00 pm **Room 406AB**

THE SECRET TO VR AUDIO

Presenter: **Alex Wilmer**, Electronic Arts, San Francisco, CA, USA

In this talk Alex will discuss the relationship between audio and storytelling. He will draw from his experience in film, games, and VR to demonstrate how focusing on narrative and emotion has been the solution to his most difficult creative challenges.

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

Product Development 5 **Friday, September 30**
1:30 pm – 3:00 pm **Room 402AB**

USER EXPERIENCE FOR MOBILE APPS

Presenters: **Marc Krolczyk**, Three Design Group, Rochester, NY, USA
Frank Marino, Jr., Three Design Group, LLC., Rochester, NY, USA

This session will discuss how Design Thinking using the User Centered Design process influences mobile application design for audio products. Mobile applications are becoming a central component of the overall product and brand experience. These applications are becoming personalized, technically advanced, and can define the product experience. Learn common practices in the User Experience discipline that place the user at the center of your product design process.

Recording & Production 7 **Friday, September 30**
1:30 pm – 3:00 pm **Room 502AB**

PLATINUM VOCAL PRODUCTION (SPECIAL EVENT)

Moderator: **Terri Winston**

Panelists: *Jimmy Douglass*
Leslie Ann Jones
Neal Pogue

A round table focusing specifically on recording chains used and why, compression techniques, de-essing, vocal editing, punch in or comps, headphone mix tricks, etc.

Sound for Picture 1 **Friday, September 30**
1:30 pm – 3:00 pm **Room 501ABC**

PRODUCTION SOUND: THE SOUND PROFESSIONALS RESPONSIBLE FOR TELLING THE STORY

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

Moderator: **Jeff Wexler**, JW Sound

Panelists: *Devendra Cleary*, Santa Monica, CA, USA
Peter Kurland

A reprise of the hugely-popular Film & TV Production Sound workshop held in Los Angeles in 2014. Production sound, which is the primary method of capturing the dialog for film and TV, uses very specific and unique methods and equipment for this work, often done in the field. The top professionals will discuss their methods for capturing the so-important words.

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

Sound Reinforcement 6 **Friday, September 30**
1:30 pm – 3:00 pm **Room 408B**

WIRELESS SPECTRUM UPDATE

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

Panelists: *Mark Brunner,* Shure Incorporated, Niles, IL USA
Joe Ciaudelli, Sennheiser, Old Lyme, CT USA
Jackie Green, Atus
James Stoffo, Radio Active Designs, Key West, FL, USA

Recent rulings by the FCC on RF spectrum as applies to wireless microphones will have a profound effect on our industry in the years to come. Added to the loss of the 700 MHz band just a few short years ago, now the 600 MHz has been auctioned as well. Due to this upcoming loss of spectrum and the resulting crowding in the remaining UHF bands, the panel will also discuss the additional changes in regulations for wireless microphone compliance. Also covered will be the potential for new frequency bands to become available for wireless mic use on a shared basis along with some of the other bands currently available including VHF, 902-928 MHz, 941-960 MHz, 1.4 GHz, 2.4 GHz, 3.5 GHz and 6 GHz.

Live Sound Expo 10 **Friday, September 30**
2:00 pm – 2:45 pm **LSE Stage**

SHELF THE LOWS: TIME TO KILL THE SUBS?

Presenter: **Howard Page**, Clair Global, Lititz, PA, USA

Subwoofers in sound reinforcement overpower the main mix to the point of distraction at many large-scale live concert events in recent history. A leading industry expert examines the role and of subwoofers and a means of taming them.

Friday, September 30 **2:00 pm** **Room 405**

Technical Committee Meeting on Network Audio Systems

Project Studio Expo 10 **Friday, September 30**
2:15 pm – 3:00 pm **PSE Stage**

MIXING SECRETS: PRODUCTION TRICKS TO USE WITH ANY DAW

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

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

Student Event and Career Development
STUDENT RECORDING CRITIQUES
Friday, September 30, 2:30 pm – 3:30 pm
Room 511A

Moderators: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA; off-beat-open-hats recording & sound reinforcement
David Greenspan, University of Michigan, Ann Arbor, MI, USA

Students! Come and get pointers, tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system and then receive feedback from a panel of renowned industry professionals. Students should sign up at the student (SDA) booth immediately following the first SDA meeting, and bring mix files on USB memory stick or hard drive to the SDA booth at least two hours prior to the session they sign up for. Files should be AIFF or WAVE, 44.1 Khz, 24 bit. Stereo mixes should be a single interleaved file, up to 5.1 surround mixes should be a set of clearly labeled discrete mono files. Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by PMC and Genelec.

Live Sound Expo 11 **Friday, September 30**
3:00 pm – 3:45 pm **LSE Stage**

SHED & ARENA OPTIMIZATION: SIDES & SUBS

Presenters: **Philip Reynolds**, System Tech, Foo Fighters, Gilbert, AZ, USA
Scott Sugden, L-Acoustics, Oxnard, CA, USA

Today's touring speakers sound amazing, but two problems must be conquered in typical shed and arena venues: the interaction of the mains and the side arrays and the management of low end supplemented with subwoofers driven from an auxiliary send. This session examines solutions for large venue applications.

Friday, September 30 **3:00 pm** **Room 405**

Technical Committee Meeting on Perception and Evaluation of Audio Signals

Friday, September 30 **3:00 pm** **Room 508A**

Standards Committee Meeting SC-04-03, Loudspeakers

Session P14 **Friday, September 30**
3:15 pm – 4:15 pm **Room 403A**

APPLICATIONS IN AUDIO—PART 2

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

3:15 pm

P14-1 The Fender 5F6-A Bassman Circuit: A 21st Century Adaptation—*Bryan Martin*, McGill University, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, QC, Canada

This investigation involves the design of a guitar amplifier conducive to low-volume environments, such as the home studio. Design goals were increased headroom, a relatively flat frequency response (in the context of classic American and British designs), and exceptional tone at low volume. This lead to the adaptation of the Fender 5F6-A circuit. Measurements of the completed unit are provided as well as assessment by guitarists. All design goals were met.
Convention Paper 9649

3:45 pm

P14-2 Design of Efficient Sound Systems for Low Voltage Battery Driven Applications—*Niels Elkjær Iversen*,[1] *Rien Oortgiesen*,[2] *Arnold Knott*,[1] *Michael A. E. Andersen*,[1] *Mikkel Høyerby*[2]
[1]Technical University of Denmark, Kogens Lyngby, Denmark;
[2]Merus Audio, Herlev, Denmark

The efficiency of portable battery driven sound systems is crucial as it relates to both the playback time and cost of the system. This paper presents design considerations when designing such systems. This include loudspeaker and amplifier design. Using a low resistance voice coil realized with rectangular wire one can boost the efficiency of the loudspeaker driver and eliminate the need of an additional power supply. A newly developed switching topology is described that is beneficial to near-idle efficiency (< 2 W), which is crucial for real audio applications in the consumer electronics spaceA small sized sound system was implemented using the discussed design considerations. The amplifier efficiency performance was found to be very high with near-idle efficiency reaching a remarkably 88% at 2W. The average output SPL was estimated to be up to 90 dB in half spheric anechoic conditions. Measured results are compared with current state-of-art and shows a 14% points efficiency improvement.
Convention Paper 9650

Session P15 **Friday, September 30**
3:15 pm – 4:45 pm **Room 409B**

PERCEPTION—PART 1

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

3:15 pm

P15-1 In-Vehicle Audio System Distortion Audibility versus Level and Its Impact on Perceived Sound Quality— *Steve Temme,*[1] *Patrick Dennis*[2]

[1]Listen, Inc., Boston, MA, USA
[2]Nissan Technical Center North America, Farmington Hills, MI, USA

As in-vehicle audio system output level increases, so too does audio distortion. At what level is distortion audible and how is sound quality perceived as level increases? Binaural recordings of musical excerpts played through the in-vehicle audio system at various volume levels were made in the driver's position. These were adjusted to equal loudness and played through a low distortion reference headphone. Listeners ranked both distortion audibility and perceived sound quality. The distortion at each volume level was also measured objectively using a commercial audio test system. The correlation between perceived sound quality and objective distortion measurements are discussed.
Convention Paper 9651

3:45 pm

P15-2 Effect of Presentation Method Modifications on Standardized Listening Tests—*Julian Villegas,*[1] *Tore Stegenborg-Andersen,*[2] *Nick Zacharov,*[2] *Jesper Ramsgaard*[3]

[1]University of Aizu, Aizu Wakamatsu, Fukushima, Japan
[2]DELTA SenseLab, Hørsholm, Denmark
[3]Widex, Lynge, Denmark

This study investigates the impact of relaxing presentation methods on listening tests by comparing results from two identical listening experiments carried out on two countries and comprising two presentation methods: the ITU-T P.800 Absolute Category Rating (ACR) recommendation and a modified version of it where assessors had more control on the reproduction of the samples. Compared with the standard method, test duration was reduced on average 37% in the modified version. No significant effects of the method used on the ratings of codecs were found, but a significant effect of site on ratings and duration were found. We hypothesize that in the latter case, cultural differences and instructions to the assessors could explain these effects.
Convention Paper 9652

4:15 pm

P15-3 Can We Hear The Difference? Testing the Audibility of Artifacts in High Bit Rate MP3 Audio—*Denis Martin,*[1,2] *Richard King,*[1,2] *Wieslaw Woszczyk,*[1,2] *George Massenburg,*[1,2] *Martha DeFrancisco*[1,2]

[1]McGill University, Montreal, QC, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology - Montreal, Quebec, Canada

A new type of listening test for testing very small impairments in audio systems is proposed using audio engineer participants and a mix matching task based approach. A pilot test was conducted in an attempt to reveal perceptual differences between WAV (44.1 k-16 bit) and MP3 (256 kbps) encodings of the same musical material. The participant mixing data was analyzed and trends in the data generally coincide with the hypotheses. Several factors were also found that can influence participant accuracy and speed in completing this type of test: age, experience (production, musical, listening test), and preferred genre for audio work.
Convention Paper 9653

Workshop 6 **Friday, September 30**
3:15 pm – 4:45 pm **Room 404AB**

RECORDING THE MODERN BIG BAND: PART 2

Presenters: **Jim Anderson**
Rich Breen
Freddie Breitberg

Following up on the workshop presented at AES 139, this event will continue exploring the topic of recording large ensemble jazz in modern contexts. This panel of esteemed engineers will explore the intricacies of documenting the modern jazz orchestra and break down their preferences for ensemble configuration, microphone selection and placement, session management, and mixing techniques. The presentation will include session documentation, setup diagrams, photographs, and audio examples from recent releases.

Broadcast/Streaming Media 7 **Friday, September 30**
3:15 pm – 4:45 pm **Room 408A**

AUDIO CONSIDERATIONS FOR OVER THE TOP TELEVISION

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

Panelists: *Roger Charlesworth*, DTV Audio Group, New York, NY, USA
Berhard Grill, Frauhfer IIS, Erlangen, Germany
Sean Richardson, Starz Entertainment, Englewood, Co, USA
Jeff Riedmiller, Dolby Labs, San Francisco, CA, USA
Jim Starzynski, NBC Universal, New York, NY, USA
Adrian Wisbey, BBC Design & Engineering, London, UK

No single area of media distribution is developing as quickly and with as much agility as Over-the-Top (OTT) Television. New media formats (such as 4K video) are typically making their first ap-pearances there, so it could be expected that the immersive and personalized features of Next-Generation Audio (NGA) services for television sound may also debut in the OTT world. Mean-while, current challenges of interoperability and loudness management in OTT TV still require some sorting out. Find out the latest on audio in today's and tomorrow's OTT TV environment, from experts currently working on solutions in this dynamic session, which will also report on de-velopment of AES Guidelines for audio practices in OTT TV and online video services.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Game Audio 7 **Friday, September 30**
3:15 pm – 4:45 pm **Room 406AB**

VR AUDIO RENDERER PANEL— PROCESS AND IMPLEMENTATION

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

Panelists: *Martin Dufour*, Audiokinetic
Patrick Flanagan, THX Ltd.
Michel Henein, Visisonics
Alex Wilmer, Electronic Arts

Delivering a convincing audio experience in VR is not just challenging but seemingly unobtainable at times. There is a complicated chain of variables that must work together in order to create realistic sonic environments (plugins, HRTFs, middleware, headphones, etc.). The goal of the session is to help game makers better understand the spatializer rendering process and be able to more easily work in this format. But more importantly, it will focus on problems with implementation and discuss the key ingredients to getting the most out of your game.

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

Networked Audio 4 **Friday, September 30**
3:15 pm – 4:45 pm **Room 408B**

AES67 DISCOVERY

Chair: **Greg Shay**, the Telos Alliance

Presenters: *Jeff Berryman*, Bosch Communications, Ithaca, NY, USA
Tom de Brouwer, Bosch Communications Systems, Burnsville, MN USA; OCA Alliance
Gints Linis, the Telos Alliance

AES67 promises to unite different audio networking technologies but allows for multiple exclusive methods for discovery of networked devices, potentially hindering adoption. This presentation covers these methods, their pros and cons, and highlights the importance of determining a standard upon which the AV industry can rely.

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

Product Development 6 **Friday, September 30**
3:15 pm – 4:45 pm **Room 402AB**

MODERN DIGITAL PROCESSING OF MICROPHONE SIGNALS

Presenter: **Paul Beckmann**, DSP Concepts, Inc., Santa Clara, 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 pro-cessors for reducing plosives and handling noise ("de-poppers"). We show how these algorithms are designed and tuned in practice

Sound for Picture 2 **Friday, September 30**
3:15 pm – 4:45 pm **Room 501ABC**

MUSIC SCORING FOR FILM & TV: HOW IT WAS, WHERE IT IS, WHERE IT'S GOING

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

Presenters: **Simon Franglen**, Class1 Media, Los Angeles, CA, USA; London
Jim Henrikson
Leslie Ann Jones, Skywalker Sound, San Rafael, CA, USA
Jason LaRocca, La-Rocc-A-Fella, Inc., Los Angeles, CA, USA

Like all areas of film and TV production, the craft of Music Scoring is undergoing significant changes to meet the growing demand for productions. These leading professionals discuss the process of preparing and recording the music.

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

Project Studio Expo 11 **Friday, September 30**
3:15 pm – 4:00 pm **PSE Stage**

THE TIPPING POINT

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

So you like to mix loud, but what if too loud actually makes the result softer in the end? Pre-recorded tracks for the European Song Contest in 2015 were generally extremely hot, but content providers were unaware that pushing beyond a tipping point made too much of a good thing become bad. Not only were music tracks more distorted than need be, most also ended up being quieter, considering the jury and the consumer. Hear the latest about ideal leveling for platforms such as Tidal, Spotify, iTunes, YouTube and TV. The presentation also details EBU, AES, Nordic Dynamic and mobile device standards; before discussing loudness-savvy alignment and calibration of loudspeakers in pro.

Session P16 **Friday, September 30**
3:30 pm – 5:00 pm **Room 403B**

POSTERS: SIGNAL PROCESSING

3:30 pm

P16-1 Analysis of Binaural Features for Supervised Localization in Reverberant Environments—*Jiance Ding*,[1,2] *Jie Wang*,[3] *Chengshi Zheng*,[1,4] *Renhua Peng*,[1] *Xiaodong Li*[1,4]
[1]Chinese Academy of Science, Beijing, China
[2]University of Chinese Academy of Sciences, Beijing, China;
[3]Guangzhou University, Guangzhou, China
[4]Chinese Academy of Sciences, Shanghai, China

Recent research on supervised binaural sound source localization methods shows that the performance is promising even in reverberant environments when the training and testing environments can match perfectly. However, these supervised methods may still suffer somewhat performance degradation when the intensity of the reverberation increases markedly. This paper studies the impact of reverberation on binaural features theoretically. This study reveals that reverberation is a major factor in reducing the accuracy of supervised binaural localization. Accordingly, we use a binaural dereverberation algorithm to reduce the effect of reverberation and thus to improve the performance of the existing supervised binaural localization methods. Experimental results demonstrate that dereverberation can improve the localization accuracy of these supervised binaural localization methods in reverberant environments.
Convention Paper 9642

3:30 pm

P16-2 Acoustic Echo Cancellation for Asynchronous Systems Based on Resampling Adaptive Filter Coefficients—*Yang Cui*,[1,2,4] *Jie Wang*,[3] *Chengshi Zheng*,[1,4] *Xiaodong Li*[1,4]
[1]Chinese Academy of Science, Beijing, China
[2]University of Chinese Academy of Sciences, Beijing, China

[3]Guangzhou University, Guangzhou, China
[4]Chinese Academy of Sciences, Shanghai, China

In asynchronous systems, most of traditional acoustic echo cancellation (AEC) algorithms couldn't track echo path correctly because of the asynchronization of D/A and A/D converters, which can reduce the performance dramatically. Based on multirate digital signal processing theory, this paper proposes to solve this problem by resampling adaptive filter coefficients (RAFC), where the adaptive filter coefficients are updated by normalized least mean square (NLMS) algorithm with a variable step control method. The simulation results indicate that the proposed can estimate the clock offset quite accurately. Objective test results also show that the proposed RAFCN-LMS is much better than the previous adaptive sampling rate correction algorithm in terms of the convergence rate and clock offset tracking performance.
Convention Paper 9643

3:30 pm

P16-3 Single-Channel Speech Enhancement Based on Reassigned Spectrogram—*Jie Wang*,[1] *Chengcheng Yang*,[1] *Chunliang Zhang*,[1] *Renhua Peng*[2]
[1]Guangzhou University, Guangzhou, China
[2] Chinese Academy of Sciences, Beijing, China

Most of the traditional a priori SNR estimators, such as the decision-directed approach and its improved versions, only consider the correlation of adjacent frames. Whereas, it is well-known that voiced speech is a typical harmonic signal that results in strong correlation of harmonics. We can expect that the a priori SNR estimator can be improved if the correlation of adjacent frames and harmonics can be used simultaneously. With this motivation, we propose to use the reassigned spectrogram (RS) to control the forgetting factor of the decision-directed approach. Experimental results indicate that the proposed RS-based SNR estimator is much better than the traditional decision-directed approach.
Convention Paper 9644

3:30 pm

P16-4 The a Priori SNR Estimator Based on Cepstral Processing—*Jie Wang*,[1] *Guangquan Yang*,[1] *JingJing Liu*,[1] *Renhua Peng*[2]
[1]Guangzhou University, Guangzhou, China
[2] Chinese Academy of Sciences, Beijing, China

For single-channel speech enhancement systems, the a priori SNR is a key parameter for Wiener-type algorithms. The a priori SNR estimators can reduce the noise efficiently when the noise power spectral density (NPSD) can be estimated accurately. However, when the NPSD is overestimated/underestimated, the a priori SNR may lead to the speech distortion and the residual noise. To solve this problem, this paper proposes to estimate the a priori SNR based on cepstral processing, which not only can suppress harmonic speech components in the noisy speech segments, but also can reduce strong noise components in noise-only segments. Simulation results show that the proposed algorithm has better performance than the traditional DD and Plapous's two-step algorithms.
Convention Paper 9645

3:30 pm

P16-5 Quantitative Analysis of Masking in Multitrack Mixes Using Loudness Loss—*Gordon Wichern, Hannah Robertson, Aaron Wishnick*, iZotope, Cambridge, MA, USA

The reduction of auditory masking is a crucial objective when mixing multitrack audio and is typically achieved through manipulation of gain, equalization, and/or panning for each stem in a mix. However, some amount of masking is unavoidable, acceptable, or even desirable in certain situations. Current automatic mixing approaches often focus on the reduction of masking in general, rather than focusing on particularly problematic masking. As a first step in focusing the attention of automatic masking reduction algorithms on problematic rather than known and accepted masking, we use psychoacoustic masking models to analyze multitrack mixes produced by experienced audio engineers. We measure masking in terms of loudness loss and present problematic masking as outliers (values above the 95th percentile) in instrument and frequency-dependent distributions.
Convention Paper 9646

3:30 pm

P16-6 Log Complex Color for Visual Pattern Recognition of Total Sound—*Stephen Wedekind, P. Fraundorf*, University of Missouri - St. Louis, St. Louis, MO, USA

While traditional audio visualization methods depict amplitude intensities vs. time, such as in a time-frequency spectrogram, and while some may use complex phase information to augment the amplitude representation, such as in a reassigned spectrogram, the phase data are not generally represented in their own right. By plotting amplitude intensity as brightness/saturation and phase-cycles as hue-variations, our complex spectrogram method displays both amplitude and phase information simultaneously, making such images canonical visual representations of the source wave. As a result, the original sound may be reconstructed (down to the original phases) from an image, simply by reversing our process. This allows humans to apply our highly-developed visual pattern recognition skills to complete audio data in new way.
Convention Paper 9647

3:30 pm

P16-7 Material for Automatic Phonetic Transcription of Speech Recorded in Various Conditions—*Bozena Kostek, Magdalena Plewa, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

Automatic speech recognition (ASR) is under constant development, especially in cases when speech is casually produced or it is acquired in various environment conditions, or in the presence of background noise. Phonetic transcription is an important step in the process of full speech recognition and is discussed in the presented work as the main focus in this process. ASR is widely implemented in mobile devices technology, but the need is also encountered in applications such as automatic recognition of speech in movies for non-native speakers, for impaired users, and as a support for multimedia systems. This work contains an attempt to analyze speech recorded in various conditions. First, audio and video recordings of specially constructed list of words in English were prepared in order to perform dedicated audio and video analyses in the future stages of the research aiming at audio-visual speech recognition systems (AVSR) development. A dataset of audio-video recordings was prepared and examples of analyses are described in the paper.
Convention Paper 9648

Special Event
THE GREAT BRITISH RECORDING STUDIOS
Friday, September 30, 3:30 pm – 5:00 pm
Room 502AB

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

Panelists: *Geoff Emerick*
Dave Harries
Adam Moseley

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.

Student Event and Career Development
STUDENT RECORDING CRITIQUES
Friday, September 30, 4:00 pm – 5:00 pm
Room 515B

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

Students! Come and get pointers, tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system and then receive feedback from a panel of renowned industry professionals. Students should sign up at the student (SDA) booth immediately following the first SDA meeting, and bring mix files on USB memory stick or hard drive to the SDA booth at least two hours prior to the session they sign up for. Files should be AIFF or WAVE, 44.1 Khz, 24 bit. Stereo mixes should be a single interleaved file, up to 5.1 surround mixes should be a set of clearly labeled discrete mono files. Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by PMC and Genelec.

Live Sound Expo 12 — **Friday, September 30**
4:00 pm – 4:45 pm — **LSE Stage**

ENTERTAINMENT RF: MOVING TARGET

Presenters: **Mark Brunner**, Shure Incorporated, Niles, IL USA
Joe Ciaudelli, Sennheiser, Old Lyme, CT USA
James Stoffo, Radio Active Designs, Key West, FL, USA
Karl Winkler, Lectrosonics, Rio Rancho, NM, USA

It's deja vu all over again. UHF TV spectrum shared by entertainment wireless as secondary users is in the process of being reduced for the second time in as many decades through voluntary auctions that will reduce UHF spectrum by over 100 MHz, rendering equipment operating in the 600 MHz bands (and below) obsolete in a few years. Our experts examine what to expect and when to expect it.

Project Studio Expo 12 — **Friday, September 30**
4:15 pm – 5:00 pm — **PSE Stage**

BEYOND EQ & COMPRESSION

Presenter: **Sam Inglis**, Sound on Sound

As more and more music is recorded outside of professional studios, and track counts grow bigger and bigger, the challenges facing the mix engineer grow and grow—and so does the range of processors on hand to solve them. Equalization and compression have always been the standard tone-shaping tools, but modern DAW software offers much more. Today's plug-in folders are bursting with transient shapers, dynamic equalizers, frequency-dependent dynamics, harmonic enhancement, spectral editors and even faders that write their own automation. The possibilities are endless.

These new types of plug-in can save time, solve difficult problems and help take your mixes to the next level—but only if you know how to deploy them. When, why and how should you go beyond conventional EQ and dynamics? Taking universal mix issues and real-world situations as examples, Sound On Sound Features Editor Sam Inglis explains how to bring these cutting-edge software processors to bear in your studio.

Session P17 — **Friday, September 30**
5:00 pm – 6:30 pm — **Room 409B**

APPLICATIONS IN AUDIO—PART 3

Chair: **Josh Reiss**, Queen Mary University of London, London, UK

5:00 pm

P17-1 Measuring Frequency and Amplitude Modulation Effects in Cross-Modulation Distortion from Audio Amplifiers—*Ronald Quan*, Ron Quan Designs, Cupertino, CA, USA

In the SMPTE IM (Intermodulation) distortion test using 60 Hz and 7000 Hz signals, it is normally assumed that the IM distortion products form amplitude modulation (AM) sidebands, which are commonly measured with an envelope AM detector. However, with other IM test signals the output of the AM detector is minimized or close to zero, while instead producing FM (Frequency Modulation) distortion. This paper investigates testing for phase and frequency modulation from cross-modulation and intermodulation distortions in audio amplifiers. The cross modulation test signal includes a 3 kHz tone and a high frequency amplitude modulation signal. Alternatively the test signal may include the 3 kHz signal and two high frequency tones. Amplifiers with feedback are tested.
Convention Paper 9654

5:30 pm

P17-2 Finite Element Simulation of Ring Radiators with Acoustic Filter—*Lakshmikanth Tipparaju, Allan Devantier, Andri Bezzola*, Samsung Research America, Valencia, CA USA

Ring radiator loudspeakers consist of a phase plug in front of the diaphragm and are typically used to create omnidirectional sound. Potential resonances between the diaphragm and the phase plug create a design challenge and put additional requirements on the equalizer to obtain a flat amplitude response. We present a finite element model to predict and mitigate the undesirable peaks and dips in the amplitude response of ring radiator loudspeakers. Simulations show that a properly designed acoustic filter can minimize resonant behavior between diaphragm and phase plug. A maximum peak attenuation of 12 dB was obtained using this method. We observe good correlation between simulations and experimental results.
Convention Paper 9655

6:00 pm

P17-3 Android Based Mobile Application for Home Audio—Visual Localization to Benefit Alzheimer Patients—"Remember It"—*Raul Rincón Flórez, Christopher Vottela Pérez, Esteban Polania Gutierrez, Luis Felipe Rios Zamudio*, Universidad de San Buenaventura, Bogota, Cundinamarca, Colombia

This article targets the development of a Mobile application to benefit people affected by Alzheimer's Disease implementing visual resources like RGB Leds, using Arduino as a medium between Android and these. On the other hand, the implementation of the program "App inventor," the Arduino algorithm programming and the circuit mapping.
Convention Paper 9656
[This paper was not presented]

Tutorial 5 **Friday, September 30**
5:00 pm – 6:30 pm **Room 404AB**

RECORDING STUDIO TECHNOLOGIES FOR LIVE EDM AND CONCERT PRODUCTION

Presenters: **Richard Larsen**
Scott Eric Olivier, Casa Distortion, Inc., North Hollywood, CA, USA

This Master Class will review the control and monitoring requirements of EDM performers and concert tours using extensive electronic instruments and backing tracks, then walk through several case studies using recording studio technologies to build custom, real-time control and audio networks for use on stage. Application technologies include using SMPTE time code, MADI audio networking, and multiple control surfaces to manage content, control code, and audio signals from Ableton Live, Pro Tools, and Virtual Instruments. Case studies will include custom solutions for high profile concert tours and development of a solution for the EDM group UNA Music.

Networked Audio 5 **Friday, September 30**
5:00 pm – 6:15 pm **Room 408A**

UNDERSTANDING AUDIO CAPABILITIES AND BANDWIDTH IN MIXED USE NETWORKS

Presenter: **Brad Price**, Audinate

The IT convergence is upon us, but many in AV are unfamiliar with the details of audio networking from the perspective of an IT manager. This presentation seeks to dispel some commonly held misconceptions about audio bandwidth requirements and the capabilities of modern switched networks when using real-time audio protocols.

1. What network engineers really care about, and how this helps AV installers.
2. How big is that? Multichannel audio transport as compared with common network capacities.
3. Unicast and Multicast: where understanding matters in network design and use.

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

Product Development 7 **Friday, September 30**
5:00 pm – 6:30 pm **Room 402AB**

DEVELOPING NOVEL AUDIO PROCESSING ALGORITHMS—MOVING QUICKLY FROM IDEAS TO REAL-TIME PROTOTYPES

Presenter: **Gabriele Bunkheila**, MathWorks, Cambridge, UK

High-level programming languages are frequently used by DSP and Audio Engineers in audio product design. These languages allow engineers to rapidly create and understand new audio pro-cessing ideas which later are targeted for implementation in various audio products. In this work-flow, real-time tuning is an important component.

In this tutorial, we will use our industry knowledge to summarize the best programming practices adopted by audio companies to reuse research code directly for real-time prototypes. We will show a number of examples, tips, and tricks to minimize latency, maximize efficiency, run in real-time on a PC, and generate native VST plugins for testing and prototyping without writing any C or C++ code.

Sound for Picture 3 **Friday, September 30**
5:00 pm – 6:30 pm **Room 508B**

DIALOG INTELLIGIBILITY: THE CHALLENGE OF RECORDING THE WORDS SO THE AUDIENCE CAN UNDERSTAND THEM

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

Presenters: **Peter Mapp**, Peter Mapp Associates, Colchester, Essex, UK
Simon Tuff
Jeff Wexler, JW Sound

The intelligibility of the dialog is of great importance to the audience. There have been growing reports of both film and TV productions where the audience has been unhappy with what they're hearing in the cinema, and at home. This workshop looks at the issues from the microphone through the editing and mixing process, and then into the listening environment with the leading professionals in this area.

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

Historical Event
H1 - HISTORY OF STUDIO ACOUSTIC DESIGN
Friday, September 30, 5:00 pm – 6:30 pm
Room 406AB

Presenter: **George Augspurger**, Perception Inc., Los Angeles, CA, USA

One of the leaders in U.S. studio design over six decades will give his personal perspective on the evolution of various design approaches. George Augspurger is an audio and acoustical engineer who has been active in recording studio work since the 1960s. He spent more than a decade with JBL starting in 1958, then started his own consulting firm in 1970 and continues to be actively engaged in architectural acoustics and studio design projects. He has designed hundreds of studios, mastering rooms, and screening rooms as well as custom monitor loudspeaker systems. He is a fellow of both the Audio Engineering Society and the Acoustical Society of America. In this presentation he will outline the development of modern studio design practices, with examples of important projects and events over a 65-year timeline. The presentation will be followed by questions from the audience.

Student Event and Career Development
RECORDING COMPETITION—PART 1
Friday, September 30, 5:00 pm – 7:00 pm
Room 501ABC

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.

5:00: Category 1—Traditional Acoustic Recording
6:00: Category 2– Traditional Studio Recording

Friday, September 30 5:00 pm Room 405

Technical Committee Meeting on Signal Processing

Student Event and Career Development
TELEFUNKEN ELEKTROAKUSTIK /
AES SDA MEET UP & PARTY DOWN
Friday, September 30, 8:00 pm — 11:00 pm
Echo Park

"Audio Students! Join us for a fun and exciting evening at our second AES Student Party, sponsored by Telefunken and hosted by Bedrock LA in the Echo Park neighborhood of Los Angeles. The event will feature live In-Studio performances by Ivory Deville & Matt Szlachetka, Door prizes and give-a-ways, light food and refreshments, and a Pinball Arcade!***

Co-sponsors TELEFUNKEN Elektroakustik, Moog, Ableton, DirectOut, Latch Lake Music, Direct Sound, and more will also be on-hand demonstrating their products and offering insight on techniques and designs.

Additional details will be announced at SDA1, will be available at the SDA booth, and can be found online at http://bit.ly/141TFparty"

Session P18 Saturday, October 1
9:00 am – 10:30 am Room 403A

PERCEPTION—PART 2

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

9:00 am

P18-1 Hyper-Compression, Environmental Noise and Preferences for the Ear Bud Listening Experience—
Robert W. Taylor,[1] *Luis Miranda*[2]
[1]University of Newcastle, Callaghan, NSW, Australia
[2]University of Sydney, Sydney, NSW, Australia

The notion that compressed music performs more effectively in automobiles as a consequence of the background noise present has been widely accepted and particularly relevant to classical music with a very large dynamic range. The environmental noise can act as a masking agent that can interrupt the listening experience when sections of the music fall below the noise level. Similarly, it is assumed that the hyper-compression of contemporary popular music fulfills a similar function when using ear bud headphones in noisy environments. This study examines this assumption and can find no evidence to support the practice. It is suggested that contemporary music most likely does not have a sufficiently large enough dynamic range regardless to support its use in this instance.
Convention Paper 9657

9:30 am

P18-2 Validation of a Virtual In-Ear Headphone Listening Test Method—*Todd Welti, Sean Olive, Omid Khonsaripour*, Harman International, Northridge, CA, USA

Controlled, comparative double blind listening tests on different in-ear (IE) headphones are logistically impractical. One solution is to present listeners virtualized versions of the headphones through a high quality IE replicator headphone equalized to match their measured frequency responses. To test the accuracy of method, ten trained listeners evaluated the overall quality of both actual and virtualized versions of twelve different IE headphones binaurally recorded and reproduced through replicator headphone. The results show evidence that the virtualized headphones produce sound quality ratings that are similar to those produced by the actual headphones.
Convention Paper 9658

10:00 am

P18-3 The Physics of Auditory Proximity and its Effects on Intelligibility and Recall—*David Griesinger*, David Griesinger Acoustics, Cambridge, MA, USA

Cutthroat evolution has given us seemingly magical abilities to hear speech in complex environments. We can tell instantly, independent of timbre or loudness, if a sound is close to us, and in a crowded room we can switch attention at will between at least three different simultaneous conversations. And we involuntarily switch attention if our name is spoken. These feats are only possible if, without conscious attention, each voice has been separated into an independent neural stream. We believe the separation process relies on the phase relationships between the harmonics above 1000 Hz that encode speech information, and the neurology of the inner ear that has evolved to detect them. When phase is undisturbed, once in each fundamental period harmonic phases align to create massive peaks in the sound pressure at the fundamental frequency. Pitch-sensitive filters can detect and separate these peaks from each other and from noise with amazing acuity. But reflections and sound systems randomize phases, with serious effects on attention, source separation, and intelligibility. This talk will detail the many ways ears and speech have co-evolved, and recent work on the importance of phase in acoustics and sound design.
Convention Paper 9659

Tutorial 6 Saturday, October 1
9:00 am – 10:30 am Room 404AB

FUNDAMENTAL ACOUSTICS

Presenter: **Anthony Grimani**, PMI Engineering, Novato, CA, USA

Fundamental acoustics are an important building block to any audio engineering application. This tutorial will review fundamental wave theory, reflections, absorption, diffraction, echoes, reverb, and standing waves. We will also review implications for listening locations, speaker placement, intelligibility, and the audible symptoms of various acoustic anomalies: comb filters, standing waves, sound field integration and accuracy. The tutorial will include case studies of principles in action: tuning a studio (absorption/diffusion/bass traps/EQ), speaker placement for live.

Broadcast/Streaming Media 8 Saturday, October 1
9:00 am – 10:30 am Room 408A

DESIGNING, BUILDING, AND MAINTAINING A RADIO STATION PER-FORMANCE SPACE

Presenters: **Lynn Duke**, CBS Radio
Gary Kline, Kline Consulting
Steve Shultis, New York Public Radio
Andrew Stern, Cumulus Media, San Francisco, CA, USA
Tracy Teagarden, CBS Radio, Las Vegas, NV, USA

Many radio stations are building performance spaces to further engage listeners with more original content. These spaces are not only for traditional radio but are also for multi-media production to be presented on the internet. Not all spaces are originally built for this, some are re-purposed space. We will explore choosing and equipping a performance space in an existing facility with limited resources.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Game Audio 8 **Saturday, October 1**
9:00 am – 10:30 am **Room 406AB**

GAME AUDIO EDUCATION – GET SMART! HOW AND WHERE TO GET THE TRAINING YOU NEED

Moderator: **Scott Looney**, Game Audio Institute, San Francisco, CA, USA; Nickelodeon Digital

Panelists: *Sable Cantus*, Golden West College, Buena Park, CA, USA
Matt Donner, Pyramind, San Francisco, CA, USA
Steve Horowitz, Game Audio Institute, San Francisco, CA, USA; Nickelodeon Digital
Chanel Summers, Syndicate 17 LLC, Bellevue, WA, 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.

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

Networked Audio 6 **Saturday, October 1**
9:00 am – 10:30 am **Room 409B**

THE INTERNET OF MEDIA THINGS FOR INSTALLED AV, RECORDING, AND LIVE EVENTS

Presenters: **Brent Butterworth**, Industry Expert & Consultant
Greg Schlechter, Intel, Hillsboro, OR, USA

Data and network connections are growing at rapid rates. Just as the data is growing, the expectations for audio and video continue to grow and evolve, demanding more from networks, including connecting more devices. With AV increasingly residing on the network, it becomes part of the larger IT ecosystem. The next natural step of the communications evolution is for Pro AV to become part of the IoT. With more media on the network, the need for network infrastructure that can support that becomes essential. This presentation will answer questions about network infrastructure and considerations for how to scale from small to large, as well as coexistence of multiple AV and control systems . The session will also look at how interoperability and open standards play into the evolving network.

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

AES SUPER SESSION: HOW TO DEVELOPE A KILLER AUDIO PRODUCT IN ONE DAY!

This will be a very special "Super Session" in the Product Development (PD) Track. On Saturday, October 1, The PD Track will dedicate the day to a workshop where product development professionals learn about the latest technologies and best practices in bringing new products to market across the entire development process. The PD Track is targeted to Product Design Engineers, Product Managers, Product Marketing, and Engineering Managers.

This day-long session will be presented by a team of product development experts, each one discussing best practices and technologies in their specific disciplines of Product Management, User Experience, Industrial Design, Acoustic Design, Natural Voice Processing, Validation and Testing, and Sourcing and Supply.

Unlike other AES sessions, this is a day-long session where the presenters will be working as a team with our audience throughout the day in presenting, discussing, demo'ing and more. Please reserve the day for Super Saturday at the PD Track.

To bring this session to life, the team will develop an actual product (codenamed Speak2Me) in front of and with the help of the audience. Each of the disciplines will be tied to the killer product being designed. The proposed product will be one that competes with Sonos, Alexa and other high volume consumer AoT (Audio of Things) products. We will also have a demo room to get a hands on look at the Speak2Me and interact with the development team.

By applying the best practices of Product Development to an actual product will connect the audience into how these topics apply in the real world. It's like taking the lecture and lab at the same time. The attendees will in one day get a learning experience in all facets of product development and may be headed back to their labs to create their 'killer' product!

Our Development Team and their expertise is:

• Scott Leslie, PD Squared – Product Management
• Frank Marino and Marc Krolczyk, Three Design – User Experience
• Myk Lum, LDA – Industrial Design
• Mark Trainer, Minimum Phase – Acoustic Design
• Paul Beckmann, DSP Concepts – DSP Product Engineering
• Dan Carter, VoiceBox – Natural Voice Input
• Jonathan Novick, Audio Precision – Design Validation and Production Verification
• Mike Klasco, Menlo Scientific – Sourcing and Supply Chain

Product Development SS1 **Saturday, October 1**
9:00 am – 11:00 am **Room 402AB**

PRODUCT MANAGEMENT, INDUSTRIAL DESIGN AND USER EXPERIENCE

Presenters: **Scott Leslie**, Ashly Audio, Webster, NY, USA; PD Squared, Irvine, CA USA
Myk Lum, Lum Design Associates, Irvine, CA, USA

Frank Marino, Jr., Three Design Group, LLC., Rochester, NY, USA

In the leadoff session of Super Saturday, we will explore the initial phases of product development. These three areas are done most effectively when they are done in parallel to speed time to market and ensure development team collaboration.

Product Management has become one of the most important function in developing and bringing products to market today. In the past product managers mainly focused on product features and pricing. Today the Product Manager must take a 360 degree view in include a broader variety of expertise and content. Scott Leslie of PD Squared will present the elements of such a complete view of product management:

1. Business case
2. Sales forecasting by geography and product segment
3. Market assessment
4. The constantly changing competitive landscape
5. Sales channels
6. Detailed pro forma product costing
7. Building the development budget
8. Creating a Project Schedule
9. Designing the development team
10. Selecting partners to win rather than selecting suppliers on cost

User Experience Design (UX) covers all aspects of consumer touch points of a product. These touch points include everything from initial product awareness all the way through product disposal. Marc Krolczyk and Frank Marino of Three Design Group will present what it takes to make User Experience the difference between a winner and a loser in the market. They will present the following topics:

1. Historical killer consumer audio user experience examples
2. User centered design process overview
3. Highlights of the design process including: Insights, Ideation, and Design
4. Writing user scenarios
5. Creating concept videos
6. Defining user tasks
7. Designing information architecture and user interaction ideation
8. User testing
9. Commercialization pragmatics

Industrial Design is all important function of determining the "look and feel" of a product. From shape and size to weight to materials to placement of features, industrial design is making sure that the product appeals to the buyer from the pre-sales phase to when it has lived out its useful life. Our expert in Industrial Design, Myk Lom of Lum Design Associates (LDA) will engage with our audience in what it takes to develop products that connect with consumers today and cause them to hit the "buy" button as soon as they see the product!

Sound for Picture 4 — **Saturday, October 1**
9:00 am – 10:30 am — **Room 501ABC**

WORLD CLASS FILM & TV SOUND DESIGN

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

Presenters: **Lon Bender**, Soundelux
Brett Hinton
Karen Baker Landers

The Sound Design of a film or TV production is an important and specific task to ensure the director's vision is delivered to the audience. We hear from leading sound design professionals in this workshop.

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

Sound Reinforcement 7 — **Saturday, October 1**
9:00 am – 10:30 am — **Room 408B**

NEW PARADIGMS FOR MIXING

Moderator: **Steven Gelineck**, Aalborg University Copenhagen, Copenhagen, Denmark

Panelists: *Steven Fenton*, University of Huddersfield, Huddersfield, UK
Robert Frye, Harman International
Joshua D. Reiss, Queen Mary University of London, London, UK

While digital mixing environments today incorporate many new exiting features, the main form of control is still based on channel strips adopted from analog mixers. But is this really the most intuitive, effective and creative way to work with audio production? Since everything today is digital and the control interface is now completely separated from the audio processing, we can design our control interfaces in any way we want. In this seminar future thinkers from academia, from R & D, as well as practitioners will discuss what the future mixing environment might look like. The discussion will look at alternative paradigms for controlling audio in various contexts asking questions about the importance of tradition, intuition, layout, touch, tangibility, speed, overview, feedback, visuals, etc.

Special Event
BREAKING THE AUDIO CEILING
Saturday, October 1, 9:00 am – 10:30 am
Room 502AB

Moderators: **Eileen Sweeney**, Global Data Management Group at Iron Mountain
Kai Scheer, Student

Panelists: *Deston Bennett*, Audio Relations
Leslie Lewis, Producer, GRAMMY Nominees Album Series and President, Leslie Lewis Consulting
Less Lincoln, Google, Alphabet
EveAnna Manley, Manley Labs

This panel addresses the invisible barriers and challenges that keep an individual from rising beyond a certain level in their career—the so-called "glass ceilings" encountered by women, minorities and industry outsiders of all types. Panel topics include how to get started in the industry, determining if you are on the right path, how to advance your career, equal pay for men and women, and more. Breaking The Audio Ceiling panelists include a wide range of men and women with unique perspectives, from seasoned executives to those just starting out in the industry.

Saturday, October 1 — **10:00 am** — **Room 405**

Technical Committee Meeting on Audio Forensics

Project Studio Expo 13 — **Saturday, October 1**
10:30 am – 11:15 am — **PSE Stage**

WHEN AUTHENTICITY MATTERS: CHOOSING BETWEEN PLUG-INS, DSP, AND FPGA EFFECTS

Presenter: **David Hytinen**, Antelope

Antelope's proprietary FPGA engine is capable of real-time performance before the signal even reaches the DAW component. By combining the industry-leading performance of our clocking and conversion technology with the real-time effects capabilities of our FPGA engine, we are now giving Antelope Audio users all the firepower they need to create professional mixes in the box. Conventional Buffered DSP plug-in FX require thousands of lines of code to operate, while consuming significant computer resources or additional off-line processing muscle. However, they do not necessarily improve the latency or sound quality. Such plug-ins might often add several milliseconds of processing delay and introduce artifacts to the sound, causing various types of distortion and potentially smearing the audio's stereo image. Antelope's FX maintain audio realism and feel just like real hardware gear—because the FPGA is in fact replicating actual hardware circuits. When you open a new effect, you aren't introducing additional lines of code and adding buffering, but you are adjusting the parameters of actual circuitry in real time, which are constantly operating on the Antelope interface's hardware engine.

Session P19 Saturday, October 1
10:45 am – 12:15 pm Room 409B

SIGNAL PROCESSING—PART 1

Chair: **Jean-Marc Jot**, DTS, Inc., Los Gatos, CA, USA

10:45 am

P19-1 Efficient Multichannel Audio Transform Coding with Low Delay and Complexity—*Florian Schuh,[1] Sascha Dick,[1] Richard Füg,[1] Christian R. Helmrich,[2] Nikolaus Rettelbach,[1] Tobias Schwegler[1]*
[1] Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories, Erlangen, Germany

For multichannel input such as 5.1-surround material, contemporary transform-based perce ptual audio codecs provide good coding quality even at very low bit-rates. These codecs rely on discrete joint-channel coding and parametric spatial coding schemes. The latter require dedicated complex-valued filter-banks around the core-codec, which increase both the algorithmic complexity and latency. This paper demonstrates that the discrete joint-channel as well as known semi-parametric spatial coding principles can also be realized directly within the real-valued modified discrete cosine transform (MDCT) domain of the core-coder, thereby eliminating the need for auxiliary filter-banks. The resulting fully flexible signal-adaptive coding scheme, when integrated into the MPEG-H 3D Audio codec, offers the same quality as the state of the art even at bit-rates as low as 80 kbit/s for 5.1-surround.
Convention Paper 9660

11:15 am

P19-2 Intelligent Gap Filling in Perceptual Transform Coding of Audio—*Sascha Disch,[1,2] Andreas Niedermeier,[1] Christian R. Helmrich,[2] Christian Neukam,[1] Konstantin Schmidt,[2] Ralf Geiger,[1] Jérémie Lecomte,[1] Florin Ghido,[1] Frederik Nagel,[1,2] Bernd Edler[2]*
[1] Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories, Erlangen, Germany

Intelligent Gap Filling (IGF) denotes a semi-parametric coding technique within modern codecs like MPEG-H-3D-Audio or the 3gpp-EVS-codec. IGF can be applied to fill spectral holes introduced by the quantization process in the encoder due to low-bitrate constraints. Typically, if the limited bit budget does not allow for transparent coding, spectral holes emerge in the high-frequency (HF) region of the signal first and increasingly affect the entire upper spectral range for lowest bitrates. At the decoder side, such spectral holes are substituted via IGF using synthetic HF content generated out of low-frequency (LF) content, and post-processing controlled by additional parametric side information. This paper provides an overview of the principles and functionalities of IGF and presents listening test data assessing the perceptual quality of IGF coded audio material.
Convention Paper 9661

11:45 am

P19-3 Sonic Quick Rresponse Codes (SQRC) for Embedding Inaudible Metadata in Sound Files—*Mark Sheppard,[1] Rob Toulson,[2] Mariana Lopez[1]*
[1]Anglia Ruskin University, Cambridge, Cambridgeshire, UK
[2]University of Westminster, London, UK

With the advent of high definition recording and playback systems, a proportion of the ultrasonic frequency spectrum can potentially be used as a container for unperceivable data and used to trigger events or to hold metadata in the form of text, ISRC (International Standard Recording Code) or a website URL. The Sonic Quick Response Code (SQRC) algorithm is proposed as a method for embedding inaudible acoustic metadata within a 96 kHz audio file in the 30–35 kHz bandwidth range. Thus any receiver that has sufficient bandwidth and decode software installed can immediately find metadata on the audio being played. SQRC data was mixed at random periods into 96 kHz music audio files and listening subjects were asked to identify if they perceived the introduction of the high frequency content. Results show that none of the subjects in this pilot study could perceive the 30–35 kHz material. As a result, it is shown that it is possible to conduct high-resolution audio testing without significant or perceptible artifacts caused by intermodulation distortion.
Convention Paper 9662

Session EB3 Saturday, October 1
10:45 am – 11:30 am Room 403A

LECTURE: SPATIAL AUDIO, SIGNAL PROCESSING & TRANSDUCERS

Chair: **Dylan Menzies**, University of Southampton, Southampton, UK

10:45 am

EB3-1 A Perceptual Approach to Object-Based Room Correction—*Dylan Menzies, Filippo Maria Fazi,* University of Southampton, Southampton, UK

Object-based audio offers some advantages over conventional channel-based reproduction. Objects can be adapted based on conditions at each reproduction site, in order to improve the overall quality or according to listener preferences. In particular, if the direct and reverberant parts of objects are separately available, more freedom is available to compensate for the effects of the reproduction room. An overview is provided here of a practical approach to such room correction that can modify the object stream in real-time based on captured acoustic properties of the room.
Engineering Brief 295

11:00 am

EB3-2 The Physical Limit of Microspeakers—*Kang Hou*,[1] *Shawn Shao*[2]
[1]GoerTek Electronics, Santa Clara, CA, USA
[2] GoerTek Audio Technologies, China

Audio playback in portable devices might be the most challenging and least satisfactory in acoustic fields. The rising of new audio hardware and software bring some silver lights and push the components to it limits. The physical limit and some practical design guidelines of micro-speakers are discussed in this paper.
Engineering Brief 296

11:15 am

EB3-3 A General Study of Space Dependent Frequency Response—*Mario Di Cola, Paolo Martignon*, Audio Labs Systems, Parma (PR), Italy

Vertical line arrays are nowadays the standard solution for large-scale sound reinforcement due to several well-known advantages. At the same time they also carry some critical issues along, for example like distance dependent frequency response, which is one of the most important. This phenomenon is governed by parameters like single box height, HF vertical dispersion, as well as array length and curvature. In order to minimize frequency response variability several solutions can be involved, ranging from mechanical optimization to multichannel DSP processing (all relying on a prediction software). The authors felt the necessity to investigate distance dependent frequency response with a simple and quite general (Matlab) model. The model is based on parametric Curved Sub-Arrays elements that we call CSA, instead of single element measurements. This was found to be very helpful to understand, with simplicity and generality, the very nature of the involved phenomena and what are the real necessary directivity requirements for each element.
Engineering Brief 297

Workshop 7 Saturday, Otober 1
10:45 am – 12:15 pm Room 404AB

GENIUS! LIVE: THAT LIGHTBULB MOMENT

Moderator: **David Robinson**

Panelists: *Joe Bull*, JoeCo Limited, Cambridge, UK
Dave Gunness, Fulcrum Acoustic, Sutton, MA, USA
Pat Quilter

What defines the art of "genius"? PSNEurope offers one possible definition. It's the moment when the clouds part, when the route to success is clear; the "lightbulb moment," when everything you knew before has changed, and nothing will be the same again. . . The Genius Live! session will celebrate these unique moments of clarity with a number of leading names in the industry, each of whom you might just want to call a "genius." What were they doing when the lightbulb moment arrived? And what happened next?

The event will be based on the tried-and-tested "PSNPresents" events in the UK, with host Dave Robinson (editor of *PSNEurope,* aka *Pro Sound News Europe*) ensuring no one gets bogged down in equations but everyone leaves entertained and informed!

Broadcast/Streaming Media 9 Saturday, October 1
10:45 am – 12:15 pm Room 408A

IMPLEMENTING IP WIRING FOR AUDIO APPLICATIONS

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

Panelists: *Kurt Denke*, Blue Jeans Cable, Seattle, WA, USA
Dan Mortensen, Dansound Inc., Seattle, WA, USA
Tony Peterle, Worldcast Systems, Miami, FL, USA
Greg Shay, Telos
Ron Tellas, Technology and Applications Manager, LAN, Belden, USA
Paul Vanderlaan, Nexans

Everything is going IP. And the "Internet of Things" means that, even in our world of audio, IP will soon be everywhere. What does this mean about the wire and cable used in installations? There are now thousands of products that could be used. And, while fiber optic cable and wireless trans-mission might give you options, the majority of these installations will probably be done on "tra-di-tional" twisted pairs, such as those in Category 5, 5e, 6, 6a, 7 and —soon to arrive—Category 8. You think you have a hard time figuring out which mic cable to use? These IP/Ethernet cables will be even more of a challenge. How do you approach this? How do you decide? Our panel will at-tempt to give you clear guidelines to choose cables for IP applications.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Networked Audio 7 Saturday, October 1
10:45 am – 12:15 pm Room 408B

OPTIMIZING AUDIO NETWORKS

Presenter: **Patrick Killianey**, Yamaha Professional Audio, Buena Park, CA, USA

This event covers key methods audio networks use to achieve low latency and high (optimized) bandwidth. Key technologies will include TCP vs UDP, Unicast vs Broadcast vs Multicast, QoS, IGMP Snooping, and PTP. This will be presented in practical, relatable ways for audio engineers with live demonstrations and examples from real installations. Concludes with some guidance on choosing network switches.

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

Recording & Production 8 Saturday, October 1
10:45 am – 12:15 pm Room 502AB

PLATINUM ENGINEERING (SPECIAL EVENT)

Moderator: **Michael Romanowski**, Coast Mastering, Berkely, CA, USA; The Tape Project

Panelists: *Chuck Ainlay*, METAlliance, Nashville, TN, USA
Lynne Earls
Bob Ohlsson
Andrew Scheps
Ryan Ulyate, Producer / Engineer, Topanga, CA, USA

A platinum panel of renowned engineers will discuss the key elements of the creative & technical process that they apply as they work with a wide range of artists in all popular music genres. We will be playing examples and taking questions from the audience.

Sound for Picture 5 Saturday, October 1
10:45 am – 12:15 pm Room 501ABC

WORLD CLASS SOUND MIXERS DISCUSS THEIR CRAFT

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

Presenters: **Lon Bender**, Soundelux
Bob Bronow, Max Post, Burbank, CA; Audio Cocktail
Sherry Klein

Once the dialog, music, and sound effects have been prepared, the dubbing mixers for film and television have the final impact on these productions. The complexity of the task has increased dramatically over the years and the skills of these professionals is quite often make-or-break for the project. We bring the leading professionals in this craft to discuss the challenges they face in their work.

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

Historical Event
H2 - HISTORIC TRANSDUCER TECHNOLOGIES AND THEIR CONNECTION TO VOCAL PERFORMANCE TECHNIQUES
Saturday, October 1, 10:45 am – 12:15 pm
Room 406AB

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 initial limitations and slow-but-steady advancement of the audio technologies available at the time. First there were no microphones and recording was an unplugged acoustic and mechanical experience. Carbon, ribbon, condenser, and moving coil designs followed, each having an audible impact on the pop vocal. With the extraordinary capabilities of gear available today, we face fewer constraints. In fact, a contemporary challenge might be that, freed of technical restrictions, we have too many possibilities —too broad a 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.

Student Event and Career Development
SPARS SPEED COUNSELING WITH EXPERTS —MENTORING ANSWERS FOR YOUR CAREER
Saturday, October 1, 11:00 am – 1:00 pm
Room 515A

Moderator: **Drew Waters**, Valley Village, CA, USA

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.

Student Event and Career Development
THIS IS A MIX! THIS IS A MASTER!
Saturday, October 1, 11:00 am – 12:30 pm
Room 511A

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

Panelists *Adam Ayan*, Gateway Mastering, Portland, ME, USA
Mandy Parnell, Black Saloon Studios, London, UK
Jonathan Wyner, M Works Studios/iZotope/Berklee College of Music, Boston, MA, USA; M Works Mastering

Whether you are a student, home studio or project studio user, or someone entering the professional industry, most of the music mixes you hear and try to emulate have been professionally mastered. Too many novices try to recreate a "mastered" sound in their mix. This is undesirable and limits what the mastering engineer can do. Join our panel of mastering engineers as they present some "off-the-console" mixes, discuss what they did to the mix, play the resulting master, and discuss some other common issues they see in some of the material sent to them to master.

Live Sound Expo 13 **Saturday, October 1**
11:00 am – 11:45 am **LSE Stage**

WORSHIP PRODUCTION: DIVISION OF LABOR

Presenter: **Mark Frink**, Program Coordinator/Stage Manager, Jacksonville, Florida, USA; Independent Engineer and Tech Writer, IATSE 115

The spheres of skills in a Worship Tech Team establishes the Audio Division of Labor and determines the scope of each member's role in audio production. This ability-based method of appropriate roles for audio tech teammates builds skills and confidence. Covers mute groups, VCAs, and apps.

Saturday, October 1 **11:00 am** **Room 405**

Technical Committee Meeting on Audio for Games

Product Development SS2 **Saturday, October 1**
11:15 am – 1:00 pm **Room 402AB**

ACOUSTIC DESIGN AND DSP ENGINEERING

Presenters: **Paul Beckmann**, DSP Concepts, Inc., Santa Clara, CA USA
Mark Trainer, Minimum Phase LLC, Northridge, CA, USA

In the second session of Super Saturday, our team will explore the depths of acoustic design and DSP, two disciplines when put together, can be used to create a product that it more than the sum of its parts.

Acoustic Design when it comes to successful products is always a compromise. Size, bass and loud are three competing goals of any audio product. Today there are rich tools, methods and technologies to take on acoustic challenges and bring amazing acoustic performance to the consumer. Mark Trainer of Minimum Phase will explore what is possible with transducer selection, acoustic modeling, DSP manipulation and 3D printing and show how he applied it to design the Speak2Me product that you will see for the first time at the AES Super Saturday event.

DSP Engineering today has evolved rapidly. The amount of processing power from even the lowest cost silicon devices has changed the game from a hardware to a software based engineering effort. The high level software tools of today and now compete and exceed the capability of even the best DSP coders. This changes the approach and even the method of developing audio products as DSP simulations can be tuned and iterated in very short times allowing for significant advances in optimization of both cost and performance that was not possible until recently. Paul Beckmann of DSP Concepts will discuss the details of how to approach product development from the beginning with DSP as the game changer for any audio product. During the session Paul will design and demo the DSP chain developed for the Speak2Me product. He'll solicit help from the audience in making tradeoffs in the loudspeaker and microphone processing.

Project Studio Expo 14 **Saturday, October 1**
11:30 am – 12:15 pm **PSE Stage**

HOW TO MAKE YOUR RECORDED VOCALS AT LEAST TWICE AS GOOD!

Presenter: **Craig Anderton**, Gibson Brands, Nashville, TN, USA; Harmony Central.com

Vocals are what links your soul to your listener, so you don't want anything to get in the way—and you don't want to crush your soul with processing, either. Find out how to make your recorded vocals sound at least twice as good through a variety of innovative techniques that polish your vocal, enhance clarity, and reduce the need for processors, with an emphasis on retaining the human qualities that make for a compelling vocal performance.

Live Sound Expo 14 **Saturday, October 1**
12:00 noon – 12:45 am **LSE Stage**

SAFE SOUND & IEM FUNDAMENTALS

Presenters: **Mark Frink**, Program Coordinator/Stage Manager, Jacksonville, Florida, USA; Independent Engineer and Tech Writer, IATSE 115
Michael Santucci, Sensaphonics, Chicago, IL USA

In-ear monitors provide many benefits, but there's no guarantee they can save performers' hearing. This discussion examines the physiology of hearing and fundamentals of in-ear monitoring that can help with hearing conservation.

Saturday, October 1 **12:00 noon** **Room 405**

Technical Committee Meeting on Spatial Audio

Project Studio Expo 15 **Saturday, October 1**
12:30 pm – 1:15 pm **PSE Stage**

CLOUD COLLABORATION IN PRO TOOLS

Presenter: **Greg Stryke Chin**

With the cloud now a practical vehicle for storage, sharing and working interactively, this session looks at how cloud computing tools enable a new era of collaboration. Greg Stryke Chin explains, with Avid's Pro Tools as a vehicle for his demonstrations.

Live Sound Expo 15 **Saturday, October 1**
1:00 pm – 1:45 pm **LSE Stage**

EASY LIVE RECORDING FOR VIRTUAL SOUNDCHECK

Presenters: **Jon Graves**, QSC, Costa Mesa, CA, USA
Leland Green, Yamaha Commercial Audio, Long Beach, CA, USA
Robert Scovill, Avid Technologies, Scottsdale, AZ, USA; Eldon's Boy Productions Inc.

Multi-track recording live shows has never been easier or more affordable Modern live consoles easily record individual channels of live shows with a single cable and then play them back through the console the next day.

Saturday, October 1 **1:00 pm** **Room 405**

Technical Committee Meeting on Microphones and Applications

Saturday, October 1 **1:00 pm** **Room 508A**

Standards Committee Meeting SC-02-12, Audio Networks

Session P20 **Saturday, October 1**
1:30 pm – 3:00 pm **Room 403A**

PERCEPTION—PART 3

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

1:30 pm

P20-1 Listener Perceptual Threshold for Image Shift Caused by Channel Delays in Stereo Audio—*Elisabeth McMullin*, Samsung Research America, Valencia, CA USA

To determine a threshold for listener perception of image shift caused by imperfectly synchronized stereo signals, a series of experiments using method of adjustment and ABX procedures was run over headphones and loudspeakers. Listeners adjusted an endless knob to vary delays between the stereo channels of music programs in search of a centered stereo image. The results demonstrated that 9 out of 10 listeners could reliably detect delays between loudspeaker or headphone channels at levels of 0.06 ms or lower. Furthermore, when centering a stereo image 95% of all listener adjustments were under 0.16 ms for headphones and 0.22 ms for loudspeakers. Many variables that may have affected the experiments are explored, including hearing balance, program material, and listening environments.
Convention Paper 9663

2:00 pm

P20-2 Phantom Image Elevation Explained—*Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

A subjective experiment was conducted to identify frequency bands that produce the effect of phantom image elevation. Subjects judged the perceived image regions of phantom center images for a broadband pink noise burst and its octave bands with seven different loudspeaker base angles. The 500 Hz and 8 kHz bands were found to be the most effective bands for the perception of above image with the base angle of 180°. The role of acoustic crosstalks for the elevation effect was also examined using binaural stimuli created for the 180° angle. It was found that the elevation effect was significantly reduced when the crosstalks were removed or their delay times were manipulated to 0 ms. Furthermore, the low frequency component of the crosstalk was found to produce greater elevation and externalization effects than the high frequency component.
Convention Paper 9664

2:30 pm

P20-3 Validation of a Perceptual Distraction Model in a Complex Personal Sound Zone System—*Jussi Rämö,[1,2] Steven Marsh,[3] Søren Bech,[1,2] Russell Mason,[3] Søren Holdt Jensen[1]*

[1]Aalborg University, Aalborg, Denmark
[2]Bang & Olufsen a/s, Struer, Denmark
[3]University of Surrey, Guildford, Surrey, UK

This paper evaluates a previously proposed perceptual model predicting user's perceived distraction caused by interfering audio programs. The distraction model was originally trained using a simple sound reproduction system for music-on-music interference situations and it has not been formally tested using more complex sound systems. A listening experiment was conducted to evaluate the performance of the model, using music target and speech interferer reproduced by a complex personal sound-zone system. The model was found to successfully predict the perceived distraction of a more complex sound reproducing system with different target-interferer pairs than it was originally trained for. Thus, the model can be used as a tool for personal sound-zone evaluation and optimization tasks.
Convention Paper 9665

Session P21 Saturday, October 1
1:30 pm – 3:00 pm Room 409B

SIGNAL PROCESSING—PART 2

Chair: **Leslie Gaston Bird**, University of Colorado Denver, Denver, CO, USA

1:30 pm

P21-1 Automatic Design of Feedback Delay Network Reverb Parameters for Impulse Response Matching
—*Jay Coggin, Will Pirkle*, University of Miami, Coral Gables, FL, USA

Traditional reverberation algorithms generally fall into two approaches: physical methods, which involve either convolving with room impulse responses (IRs) or modeling a physical space, and perceptual methods, which allow the use of practically any reverberation modules in various combinations to achieve a perceptually realistic reverberation sound. Perceptual reverberator algorithms are typically "hand tuned" where many of their parameters are found empirically. In this paper we present an automatic method of matching Feedback Delay Network parameters to real room impulse responses so that we may produce computationally efficient reverberation algorithms that perceptually match linear convolution with the target room IRs. Features are extracted from the target room IR and used to guide a Genetic Algorithm search to find the reverberator parameters.
Convention Paper 9666

2:00 pm

P21-2 Perceptually Alias-Free Waveform Generation Using the Bandlimited Step Method and Genetic Algorithm—
Francisco Valencia,[1] *Samarth Behura*,[2] *Will Pirkle*[2]
[1]codigoriginal, Medellin, Colombia
[2]University of Miami, Coral Gables, FL, USA

Quasi-Bandlimited waveforms may be synthesized by smoothing the discontinuities of trivial waveforms using the Bandlimited Step Method (BLEP) that produces excellent results with low computational overhead [1]. The correction scheme first starts with a sinc function—the impulse response of a low-pass filter—and uses it to generate offset values that are applied to the points around the discontinuity. Windowing the sinc function prior to use is found to reduce aliasing at the expense of the harmonic envelope, whose shape is no longer ideal. In this paper we explore two methods for generating the initial sinc function in an effort to achieve a perceptually alias-free waveform. Both approaches involve using the Genetic Algorithm as a search method.
Convention Paper 9667

2:30 pm

P21-3 Preserving Reverberation in a Sinusoidally Modeled Pitch Shifter—*Sarah R. Smith, Mark F. Bocko*,
University of Rochester, Rochester, NY, USA

Many pitch shifting algorithms suffer when the signal contains reverberation. In general, it is possible to preserve the spectral envelope of the original sound, however, an appropriate phase response can only be estimated for minimum phase systems such as vocal formants. This paper presents a pitch shifting algorithm that preserves the reverberant qualities of the original signal by modifying the instantaneous amplitude and frequency trajectories of a sinusoidal model. For each overtone, the sinusoidal trajectories are decomposed into correlated and uncorrelated components and a deviation spectrum is calculated. To synthesize the modified sound, the uncorrelated components are adjusted to preserve the deviation spectrum. The resulting trajectories and sounds are then compared with those of a standard pitch shifter.
Convention Paper 9668

Session EB4 Saturday, Oct. 1
1:30 pm – 3:00 pm Room 403B

POSTERS: EDUCATION NETWORK AUDIO, & SIGNAL PROCESSING

1:30 pm

EB4-1 SAE Parametric Equalizer Training: Development of a Technical Ear Training Program Using Max—
Mark Bassett,[1,2] *William L. Martens*[2]
[1]SAE Institute, Byron Bay, NWS, Australia
[2]University of Sydney, Sydney, NSW, Australia

Spectral-based technical ear training (TET) programs generally require the user to identity, by means of matching or absolute identification, one or more parameters of an equalizer applied to a stimulus signal. Numerous TET programs have been developed to date, targeted at either consumers, employees (for in-house training), or students (delivered within educational institutions). Corey's 2010 suite of programs featured the first commercially available TET programs developed using Max software, deployed as stand-alone applications on CD-ROM. This paper details the development of a new TET program developed in Max, successfully deployed in the Apple App Store. "SAE Parametric Equalizer Training" is a TET application designed to teach students to identify the center frequency of a parametric equalizer applied to any imported audio files.
Engineering Brief 298

1:30 pm

EB4-2 Implementation and Demonstration of Applause and Hand-Clapping Feedback System for Live Viewing—
Kazuhiko Kawahara,[1] Akiho Fujimori,[1] *Yutaka Kamamoto*,[2] *Akira Omoto*,[1,3] *Takehiro Moriya*[2]
[1]Kyushu University, Fukuoka, Japan

[2]NTT Communication Science Laboratories, Kanagawa, Japan
[3]Onfuture Ltd., Tokyo, Japan

Recent progress of network capacity enables real-time distribution of high-quality content of multimedia contents. This paper reports on our attempt to transmit the applause and hand-clapping in music concerts. We built a system that has an efficient implementation scheme for low-delay coding of applause and hand-clapping sounds. The system relayed applause and hand-clapping by viewers back to the performance site to provide these sounds in a synthesized and simulated manner. With this system, we conducted an experimental concert using a network distributed site. We observed some interactions between the performers and the receiver site audience. Responses to our questionnaire distributed to the audience and performers also confirmed that applause and hand-clapping feedback were effective for improving the sense of unity established in live viewings.
Engineering Brief 299

1:30 pm

EB4-3 Preliminary Experimental Study on Deep Neural Network-Based Dereverberation—*Ji Hyun Park,[1] Kwang Myung Jeon,[1] Ji Sang Yoo,[2] Hong Kook Kim[1]*
[1]Gwangju Institute of Science and Technology (GIST), Gwangju, Korea
[2]Kwangwoon University, Seoul, Korea

This paper deals with the issues associated with the dereverberation of speech or audio signals using deep neural networks (DNNs). They include feature extraction for DNNs from both clean and reverberant signals and DNN construction for generating dereverberant signals. To evaluate the performance of the proposed dereverberation method, artificially processed reverberant speech signals are obtained and a feed-forward DNN is constructed. It is shown that log spectral distortion (LSD) after applying DNN-based dereverberation is reduced by around 1.9 dB, compared with that of reverberant speech signals.
Engineering Brief 300

1:30 pm

EB4-4 JSAP: A Plugin Standard for the Web Audio API with Intelligent Functionality—*Nicholas Jillings,[1] Yonghao Wang,[1] Joshua Reiss,[2] Ryan Stables [1]*
[1] Birmingham City University, Birmingham, UK
[2] Queen mary University of London, London, UK

In digital audio, software plugins are commonly used to implement audio effects and synthesizers, and integrate them with existing software packages. While these plugins have a number of clearly defined formats, a common standard has not been developed for the web, utilizing the Web Audio API. In this paper we present a standard framework that defines the plugin structure and host integration of a plugin. The project facilitates a novel method of cross-adaptive processing where features are transmitted between plugin instances instead of audio routing, saving on multiple calculations of features. The format also enables communication and processing of semantic data with a host server for the collection and utilization of the data to facilitate intelligent music production decisions.
Engineering Brief 301

Tutorial 7 — Saturday, October 1
1:30 pm – 2:15 pm — Room 404AB

SPATIAL MUSIC EXPERIENCES WITH COMMON HEADPHONES

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

Over 85% of people listen to their music with headphones. So listening to simply stereo on headphones shouldn't be the end but the beginning of involving music experiences with the flexibility of headphone virtualizations (binaural). The tutorial will show how to do this with common DAWs and tools and provide a lot of listening examples to inspire music producers.

Game Audio 9 — Saturday, October 1
1:30 pm – 3:00 pm — Room 406AB

CAREERS IN GAME AUDIO—UNDERSTANDING THE BUSINESS OF GAMES

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

Panelists: *Brennan Anderson*, Senior Producer, Disney Interactive/Pyramind Studios
Pete Johnson, Composer and Sound Designer, Workinman Interactive, Rochester NY
Sally-Anne Kellaway, Zero Latency VR, Melbourne, Australia
Richard Warp, Intonic

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!

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

Recording & Mastering 9 — Saturday, October 1
1:30 pm – 3:00 pm — Room 502AB

PLATINUM MASTERING (SPECIAL EVENT)

Moderator: **Tom Kenny**, Mix Magazinet

Panelists: *Adam Ayan*, Gateway Mastering Studios, Portland, ME USA
Emily Lazar, The Lodge, New York, NY, USA
Gavin Lurssen, Lurssen Mastering
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA
Michael Romanowski, Coast Mastering, Berkely, CA, USA; The Tape Project

A platinum panel of renowned mastering engineers will discuss the creative and technical process behind selected recent work. We will focus on those details that are important to various consumer release formats. We will be playing examples and taking questions from the audience.

Sound Reinforcement 8 — Saturday, October 1
1:30 pm – 3:00 pm — Room 408B

SOUND FOR LIVE VR EVENTS

Moderator: **Dennis Kornegay**, JBA Consulting

Panelists: *Ryan Sheridan*, NextVR, SVP Imaging and Production Technologies
TBA

Producing live events directly to the various VR platforms requires special techniques and workflows. This session will discuss the following:

1) 3d Live Sound Acquisition—positional significant, location specific microphones, such as the basketball rim, cup on the 18th hole, etc., and the associated challenges, as well as binaurals, b-format, etc.;

2) Creating the Bed—the VR audio must not jar the viewer out of the moment, so creating a realistic but familiar feeling 3d bed to layer on the special tracks is critical to success;

3) Monitoring and Mixing for Live VR—how does one both monitor for all possible viewing angles and also keep up with the live action and primary viewing angle at the same time—special challenges require some special solutions

4) QC checking the final product—The best laid plans can sometimes get complicated when the various VR platforms attempt to render out the streams provided. Care must be taken to insure success for all viewing platforms.

Special Event
DTVAG AES FORUM
Saturday, October 1, 1:30 pm – 6:30 pm
Room 408A

Moderator **Roger Charlesworth**, DTV Audio Group, New York, NY, USA

The Changing Face of Television Audio:Objects, Immersivity, and Personalization

The explosion in streamed-content delivery to fixed and mobile devices has accelerated the adoption of advanced audio services for television. These offer immersive sound, enhanced personalization, and improved bandwidth efficiency. Cinema-quality immersive soundtracks are now starting to show up on popular streaming platforms. At the same time, the VR craze is driving interest and innovation in personalization and virtualized surround sound on mobile devices.

How is Hollywood coping with streamlining object workflows for episodic production? What tools are being developed to manage the demands of live production? While next-generation audio systems are becoming more widely deployed, a great deal of content is still being pushed through outdated format-and-dynamic-range-limited encoding workflows. How do we manage the loudness and consistency issues this creates? Can cinema-like television experiences coexist with cat videos and tornado alerts?

"The impact of streaming is upending the entire television business and audio is benefiting. The migration from traditional broadcasting to an IP stream-based model is accelerating the uptake of advanced encoding solutions with sophisticated audio services. This is good news, but expect turbulence along the way."—Roger Charlesworth, Executive Director, DTV Audio Group

Discussion topics will include:

The Impact of VR on Immersivity and Personalization in Television: VR is the ultimate personalized immersive experience. How will technologies and trends driven by VR re-calibrate our thinking about television sound?

Object Audio Deliverables and Emerging Tools for Interchange and Content Management: As content creators ramp up the production of premium immersive audio content for online delivery, are scalable workflows in place to absorb and manage this content on the distributor side? We will explore the progress on universal interchange standards and practical mezzanine deliverables, and discuss future requirements for versioning and editing for re-exploitation.

The Challenges of Loudness Management in Multi-Platform Streamed-Content Delivery: As the center of gravity for television viewing shifts to an online experience, are the best practices for loudness, dynamic range and format management in danger of being lost along the way? Can a line still be drawn between fixed and mobile or desktop streaming? How do content preparation and audio encoding processes need to catch up? Our panelists will discuss the challenges of bring the same commitment to consistency and quality from network linear television to the online experience.

Atmos Mixing for Episodic Television: This fall, post-production of premium episodic content in advanced surround is in full swing. With dubbing stages all over Hollywood filling up with Atmos productions, how are mix teams adapting cinema audio production values to television budgets and timelines.

Wireless Spectrum Update: With the first forward-auction round completed, how is the future UHF spectrum picture shaping up? Initial clearing targets would make UHF microphone operation virtually impossible in many areas. Wherever we end up, things are going to get a lot more crowded. We will indulge in some speculation on the final band plan and examine the limited relocation options.

Presenters and Panelists Include:

- **Tom Brewer**, Re-Recording Mixer
- **Florian Camerer**, ORF, Vienna
- **Tim Carroll**, Senior Director, Office of the CTO, Dolby Laboratories
- **Roger Charlesworth**, Executive Director, DTV Audio Group
- **Pete Elia**, Re-Recording Mixer
- **Michael Englehaupt**, Vice President and Chief Technology Officer, Graham Media Group
- **Stacey Foster**, President Production Services, Broadway Video, Coordinating Producer, Saturday Night Live, Technical Consultant, Tonight Show with Jimmy Fallon
- **Richard Friedel**, Executive Vice President and General Manager, Fox Networks Engineering and Operations
- **Tim Gedemer**, Re-Recording Mixer
- **Jackie Green**, Vice President R&D and Engineering, Audio-Technica
- **Steve Harvey**, West Coast Editor, *Pro Sound News*
- **Tom Marks**, Re-Recording Mixer
- **Scott Norcross**, Manager Sound Platform Group, Office of the CTO, Dolby Laboratories
- **Tom Ozanich**, Re-Recording Mixer
- **Skip Pizzi**, Senior Director, Media Technology, National Association of Broadcasters
- **Sean Richardson**, Executive Director and Principal Audio Engineer, Starz Entertainment
- **Jeffery Riedmiller**, Senior Director, Sound Group, Office of the CTO, Dolby Laboratories
- **Kevin Roache**, Re-Recording Mixer
- **Tom Sahara**, Vice President Operations and Engineering, Turner Sports, Chairman Sports Video Group
- **Steve Silva**, Consultant, Technology Strategy, Fox Networks Engineering and Operations
- **Jim Starzynski**, Director and Principal Audio Engineer, NBC Universal, Chairman DTV Audio Group
- **Joel Susal**, Director, Virtual and Augmented Reality at Dolby Laboratories
- **Nicolas Tsingos**, Director, E-media technology at Dolby Labs
- **Will Wolcott**, Senior Audio Developer, Netflix

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

Special Event
RESTORATION AUDIO: MUSIC TODAY AND TOMORROW
Saturday, October 1, 1:30 pm – 3:00 pm
Room 501ABC

Moderator: **Bob Koszela**, Iron Mountain Entertainment Services, Kennerdell, PA, USA

Panelists: *Nathaniel Kunkel*
David McEowen
Joe Travers
Brett Zinn

This panel addresses the challenges to restore degraded or damaged media assets including those affected by binder hydrolysis (sticky shed), tape binding adhesion (NOT sticky shed), mold, water damage, bent flanges, de-spooled pancake, salt residue, glue seep, splice repair, lubricant loss, static discharge and acetate spoking. The work-shop will discuss asset degradation encountered by an esteemed panel of archival engineers and the remediation processes developed and used to restore playback. Advancements in audio technology and tremendous changes in how the entertainment industry creates and monetizes content have challenged the engineers to accommodate requests to migrate, mix, master, store and distribute that content securely. Moderated by Bob Koszela of Iron Mountain Entertainment Services Digital Studios (who preserves over 28 million assets for its customers), the panel will show examples of various types of degradation to a wide range of audio formats.

Product Development SS3 **Saturday, October 1**
2:00 pm – 4:00 pm **Room 402AB**

SPEAK2ME DEMO AND NATURAL VOICE INPUT

Presenters: **Paul Beckmann**, DSP Concepts, Inc., Santa Clara, CA USA
Dan Carter, VoiceBox
Mark Trainer, Minimum Phase LLC, Northridge, CA, USA

Following our lunch break, the PDSS Team will host a Demo to show off the Speak2Me product to PDSS attendees in a "mini" trade show floor co-located in the PDSS room. You will see and hear how it performs from a variety of consumer uses. The development team will be available to answer questions from PDSS attendees.

Voice Input using Natural Language Understanding is probably the hottest feature set in consumer audio products today from loudspeakers to remote controls to doorbells to assistive living devices and beyond. With the growth in IoT devices predicted in the billions, a key input to many of these devices is listening and processing information from humans. Barry Roitblat of VoiceBox will lead a discussion of the primary considerations for creating a great user experience using natural language understanding.

Natural Language Understanding is about more than just voice recognition. Once the system understands the words spoken, it must still discern the meaning and then take the appropriate action. To do this, requires Voice AI. Voice AI combines a number of features and techniques, including:

• Context—historical, conversations, environmental, or personal information that lend to the understanding of the current request.

• Semantic Understanding—parsing, knowledge, and reasoning to determine the meaning of a request from the language structure, and infer appropriate queries and actions to fulfill the request

• Machine Learning—adapt to different dialects and pronunciations, understand new words and aliases, derive new relationships, and learn new phrases or other language forms from previous interactions

• Dialog—use natural language responses and multi-modal interaction (speech, touch, gesture, etc.) to request additional information needed to fulfill a request.

Student Event and Career Development
STUDENT RECORDING CRITIQUES
Saturday, October 1, 2:00 pm – 3:30 pm
Room 511A

Moderators: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA; off-beat-open-hats recording & sound reinforcement
David Greenspan, University of Michigan, Ann Arbor, MI, USA

Students! Come and get pointers, tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system and then receive feedback from a panel of renowned industry professionals. Students should sign up at the student (SDA) booth immediately following the first SDA meeting, and bring mix files on USB memory stick or hard drive to the SDA booth at least two hours prior to the session they sign up for. Files should be AIFF or WAVE, 44.1 Khz, 24 bit. Stereo mixes should be a single interleaved file, up to 5.1 surround mixes should be a set of clearly labeled discrete mono files. Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by PMC and Genelec.

Live Sound Expo 16 **Saturday, October 1**
2:00 pm – 2:45 pm **LSE Stage**

CHOOSING THE RIGHT VOCAL MIC

Presenter:s **Pete Keppler**, Self, New York, NY, USA
Ken Newman, Newman Audio, Inc., Canyon Country, CA, USA
Howard Page, Clair Global, Lititz, PA, USA

Selecting the best microphone for a vocalist involves gauging the singer, the environment and the material, matching a model to a stage, a voice and a musical genre, as well as tips for working with vocalists.

Saturday, October 1 **2:00 pm** **Room 405**

Technical Committee Meeting on Hearing and Hearing Loss Prevention

Project Studio Expo 16 **Saturday, October 1**
2:15 pm – 3:30 pm **PSE Stage**

THE SPECIAL SAUCE FOR MIXING A HIT RECORD

Presenters: **Fab Dupont**, Flux Studios, New York, NY, USA; Flux Studios
Ryan Hewitt, Nice Rack Audio Services, Venice, CA, USA

Producer Fab Dupont (Mark Ronson, Jennifer Lopez) walks through one of today's hottest tracks with esteemed mix engineer Ryan Hewitt (Red Hot Chili Peppers, The Lumineers). Hear how the pros bring that "special sauce" to crafting a hit.

Live Sound Expo 17 **Saturday, October 1**
3:00 pm – 3:45 pm **LSE Stage**

ON THE ROAD WITH MURPHY: WHO'S LAUGHING NOW

Presenter:s **Mark Frink**, Program Coordinator/Stage Manager, Jacksonville, Florida, USA; Independent Engineer and Tech Writer - IATSE 115
Ken Newman, Newman Audio, Inc., Canyon Country, CA, USA

Howard Page, Clair Global, Lititz, PA, USA

Stories of Humor and Irony from Touring with Mr. Murphy: How Tony Bennett's monitors got flown and other true stories of "gettin' 'er done" on the road when the show must go on.

Saturday, October 1 3:00 pm Room 405

Technical Committee Meeting on Automotive Audio

Session P22 Saturday, October 1
3:15 pm – 4:45 pm Room 409B

SIGNAL PROCESSING—PART 3

Chair: **Christoph Musailik**, Sennheiser Audio Labs, Waldshut-Tiengen, Germany

3:15 pm

P22-1 Loudspeaker Crossover Network Optimizer for Multiple Amplitude Response Objectives—*William Decanio, Ritesh Banka*, Samsung Research America, Valencia, CA USA

Even though the correlation between multi-spatial amplitude response metrics and listener preferences has been established, most commercial loudspeaker crossover optimization tools operate on only a single axis of loudspeaker system's amplitude response. This paper describes development of a software based crossover network optimizer that is capable of simultaneously optimizing the on-axis, as well as off-axis acoustic response of a loudspeaker system. Choice of off-axis acoustic metrics as well as a general description of software and implementation of numerical optimization algorithms will be briefly discussed. Several design examples are presented, and measured versus predicted results will be shown.
Convention Paper 9669

3:45 pm

P22-2 The Relationship between the Bandlimited Step Method (BLEP), Gibbs Phenomenon, and Lanczos Sigma Correction—*Akhil Singh, Will Pirkle*, University of Miami, Coral Gables, FL, USA

In virtual analog synthesis of traditional waveforms, several approaches have been developed for smoothing the discontinuities in trivial waveforms to reduce or eliminate aliasing while attempting to preserve both the time and frequency domain responses of the original analog waveforms. The Bandlimited Step Method (BLEP) has been found to produce excellent results with low computational overhead. The correction scheme first starts with a sinc function—the impulse response of a low-pass filter—and uses it to generate offset values that are applied to the points around the discontinuity. This paper discusses the relationships that exist between the BLEP method, Gibbs Phenomenon, and the Lanczos Sigma correction method.
Convention Paper 9670

4:15 pm

P22-3 Modeling and Adaptive Filtering for Systems with Output Nonlinearity—*Erfan Soltanmohammadi*, Marvell Semiconductor, Inc., Santa Clara, CA, USA

Many practical systems are nonlinear in nature, and the Volterra series, also known as nonlinear convolution, is widely used to model these systems. For nonlinear systems with infinite memory, such a modeling approach is usually not feasible because of multiple infinite summations. In practice, the full Volterra series representation of such a system is either approximated by just a few terms, or is otherwise simplified. In an audio system, a useful approximation is to model all memoryless and dynamical nonlinear effects as a combined nonlinearity at its output. In this paper we propose a new Volterra-based structure that accommodates nonlinear systems with output nonlinearity and infinite memory. We then propose an adaptation approach to estimate the Volterra kernels based on the Least Mean Squares (LMS) approach.
Convention Paper 9671

Session EB5 Saturday, October 1
3:15 pm – 4:15 pm Room 403A

LECTURE: ACOUSTICS, PSYCHOACOUSTICS, & EDUCATION

Chair: **Matthew Boerum**, McGill University, Montreal, Quebec, Canada

3:15 pm

EB5-1 Digital Waveguide Network Reverberation in Non-Convex Rectilinear Spaces—*Aidan Meacham,*[1] *Lauri Savioja,*[2] *Sara R. Martin,*[3] *Julius O. Smith, III*[1]
[1]Stanford University, Stanford, CA, USA
[2]Aalto University, Aalto, Finland
[3]Norwegian University of Science and Technology, Trondheim, Norway

We present a method to simulate the late reverberation of a non-convex rectilinear space using digital waveguide networks (DWNs). In many delay-line-based reverberators, diffraction effects and even occlusion are often neglected due to the need for hand-tuned, non-physical mechanisms that complicate the extreme computational economy typical of such systems. We contend that a target space can be decomposed into rectangular solids following a succinct set of geometric rules, each of which correspond to a simple DWN reverberator. By defining the interactions between these systems, an approximation of diffraction and occlusion can be achieved while maintaining structural simplicity. This approach provides a promising engine for real-time synthesis of late reverberation with an arbitrary number of sources and receivers and dynamic geometry.
Engineering Brief 303

3:30 pm

EB5-2 Max as an Interactive, Multi-Modal Learning and Teaching Tool for Audio Engineering—*Mark Bassett*, SAE Institute, Byron Bay, NSW, Australia; University of Sydney, Sydney, NSW, Australia

A solid understanding of fundamental audio engineering concepts, specifically signal flow, is paramount to the successful operation of audio technologies. Failure to fully understand the signal path of a system can lead to students learning audio technology "functions by rote, making them inherently non-transferrable" and may also lead to the development of inaccurate conceptual models. This paper discusses the use of Max software to teach fundamental audio engineering concepts to first-year Bachelor of Audio students. Although not designed as a learning and teaching tool, Max is perfectly suited for this purpose as it is interactive, adaptive and facilitates multiple modes

of learning and interaction.
Engineering Brief 304

3:45 pm

EB5-3 A Real-Time Simulation Environment for Use in Psychoacoustic Studies of Aircraft Community Noise—*Kenneth Faller, II*,[1] *Stephen Rizzi*,[2] *Aric Aumann*[3]
[1]California State University, Fullerton, CA, USA
[2]NASA Langley Research Center, Hampton, VA, USA
[3]Analytical Science Applications International Corporation, Hampton, VA, USA

The Exterior Effects Room (EER) is a psychoacoustic test facility located at the NASA Langley Research Center, with a real-time simulation environment that includes a three-dimensional sound-reproduction system. The main purpose of the EER is to support research investigating human response to aircraft community noise. To compensate for the spectral coloration of the installation and room effects, the system required real-time application of equalization filters. The efforts taken to design, implement, and analyze the equalization filters for use in the real-time sound-reproduction system is described. Acoustic performance of the system was assessed for its crossover performance and stationary and dynamic conditions.
Engineering Brief 305

4:00 pm

EB5-4 Blind VST: A Perceptual Testing Tool for Professional Mixing Evaluation—*Matthew Boerum*, McGill University, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, QC, Canada

The Blind VST tool is presented to aid perceptual testing of real time applied digital audio signal processing in professional mixing situations. The software was designed in MaxMSP and runs as a standalone application or MaxMSP external. It hosts any Virtual Studio Technology (VST) plug-in for blind, singular control of any and all accessible VST parameters. Visual biasing from the identification and response of the VST's graphical user interface (GUI) is removed. All user data is recorded in real time. The author proposes this tool as a freely distributed academic resource. It is best integrated as an add-on module for training and research applications when determining audible signal quality, comparison and the analysis of audio descriptors.
Engineering Brief 306

Tutorial 8 — **Saturday, October 1**
3:15 pm – 4:45 pm — **Room 406AB**

INTRODUCTION TO CBT LOUDSPEAKER ARRAYS

Presenter: **D.B. (Don) Keele, Jr.**, DBK Associates and Labs, Bloomington, IN, USA

CBT is a term originated by the U.S. military in a series of declassified Naval Research Lab underwater-sound ASA papers published in the late 1970s and early 80s. Mr. Keele applied the technology to loudspeaker arrays in a series of nine AES papers between 2000 and 2015 and three more which will be presented at this convention.

CBT arrays provide broadband constant directivity/beamwidth behavior with a 3D sound field control that is exceptionally uniform and well behaved with frequency at all distances, and offer directional performance and sound-field coverage control that is outstanding. CBT array possibilities extend over the full loudspeaker product range from professional, commercial, consumer, home theater, computer, and multimedia. Don will discuss the background and history of CBT arrays including implementation of typical arrays. CBT arrays can be implemented very simply in a passive circular-arc loudspeaker configuration that does not require any sophisticated DSP signal processing except for simple level changes. Straight-line CBT arrays can also be implemented, but these require complex multi-amp configurations with individual DSP delay blocks driving each speaker or complex passive RLC delay networks.

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

Workshop 8 — **Saturday, October 1**
3:15 pm – 4:45 pm — **Room 408B**

LEARNING ABOUT AUDIO: APPS, ONLINE AND OTHER HI-TECH

Chair: **Jonathan Wyner**, M Works Studios/iZotope/ Berklee College of Music, Boston, MA, USA

Panelists: *Jason Corey*, University of Michigan, Ann Arbor, MI, USA
David Franz
Jeremy Miller
Bradford Swanson, iZotope, Inc., Cambridge, MA; University of Massachusetts - Lowell

This workshop will offer a survey of various tools and modes available to help people learn about all facets of audio and music production. Panelists will offer a description of applications, online courses, MOOC's, mobile apps, and others.

Recording & Production 10 — **Saturday, October 1**
3:15 pm – 4:45 pm — **Room 501ABC**

RAW TRACKS: SOUND OF EWF/SEPTEMBER

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

Panelists: *George Massenburg*, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Kenny Moran

Legendary engineer/producer / inventor / designer / programmer / professor George Massenburg will discuss the recording of iconic Earth Wind and Fire songs in detail, in this special 90-minute interview. Educator/engineer/producer Mark Rubel and engineer/producer/programmer Kenny Moran will engage Professor Massenburg in a wide-ranging discussion, including the stories of recording such songs as "September" and "Fantasy," an analysis of the multi-track recordings, soloing individual elements, related anecdotes, and more. This event will not be recorded, so you must be present to learn!

Special Event
HI RES AUDIO AND SOCCER MOMS—HOW ARE THEY RELATED AND HOW WILL PEOPLE BE GETTING THEIR MUSIC IN THE FUTURE?
Saturday, October 1, 3:30 pm – 5:00 pm
Room 502AB

Moderator: **Jack Joseph Puig**

Both consumer electronics manufacturers and music professionals both have clear objectives and they are not necessarily in harmony and perhaps in conflict. Does Hi Res matter and can the majority of the people hear the difference? What is the disconnect between the record companies, the consumer manufacturers, and the consumer? What are the new music delivery technologies in the future? How will that draw in fans? Where is music delivery going?

Saturday, October 1 3:30 pm Room 508A

AESSC Plenary Meeting

Project Studio Expo 17 Saturday, October 1
3:45 pm – 4:30 pm PSE Stage

QUANTUM ACOUSTICS, LIFE-LIKE IMAGING, AND THE WALL-TO-WALL SWEET SPOT: ANY SPACE, ANY SIZE, ANYWHERE

Presenter: **Hanson Hsu**, Delta H Design Inc. | DHDI, Marina Del Rey, CA, USA

Acoustical design is very much like music composition: both are inherently mathematical and artistic. Until the mid 2000s, studios were trapezoidal amalgamations of architectural and acoustic geometries brimming with sonic standards of the day: saw-toothed/sloping walls, multi-planar speaker soffits, bass traps, suspended clouds, Helmholtz resonators, quadratic diffusors, etcetera, etcetera. Yet all too often imaging, intelligibility, and clarity fell prey to economy and aesthetics. Artists, producers, and engineers alike were reduced to sharing a one foot square sweet spot in even the most prestigious studio facilities. Fortunately, as change and technological evolution go hand in hand, advancements in quantum physics, materials sciences, and computational analysis have given rise to an infinite variety of design possibilities in acoustics.

Student Event and Career Development
STUDENT RECORDING CRITIQUES
Saturday, October 1, 4:00 pm – 5:00 pm
Room 515B

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

Students! Come and get pointers, tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system and then receive feedback from a panel of renowned industry professionals. Students should sign up at the student (SDA) booth immediately following the first SDA meeting, and bring mix files on USB memory stick or hard drive to the SDA booth at least two hours prior to the session they sign up for. Files should be AIFF or WAVE, 44.1 Khz, 24 bit. Stereo mixes should be a single interleaved file, up to 5.1 surround mixes should be a set of clearly labeled discrete mono files. Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by PMC and Genelec.

Product Development SS4 Saturday, October 1
4:15 pm – 6:00 pm Room 402AB

VALIDATION AND TESTING, SOURCING, AND SUPPLY

Presenters: **Mike Klasco**, Menlo Scientific, Richmond, CA USA
Jonathan Novick, Audio Precision, Beaverton, CA, USA

In our last session, the team will discuss how the product moves from the design phase to the production phase. Most product companies have challenges in getting through this phase quickly and without undue stress. Making sure the design is right, it is properly documented and having the right partners is essential to success in getting the product into the customers' hands.

Design Validation is the phase of development where the design is fully evaluated to make sure that the design can transition to be manufactured in high quantities while adhering to product specifications and costing constraints. Production Verification is where the appropriate systems and controls are put in place to ensure that the production facility, equipment and work force are ready to produce the product to meet customer demand for availability price, quality. Jonathan Novick of Audio Precision will discuss these key topics and how a product can successfully move from design phase to the production phase. He will look at how these are applied to the Speak2Me product and what challenges a product company faces in making sure that a high level of confidence about the product's ability to perform as expected when it is in the customer's hands.

Sourcing and Supply Chain has evolved tremendously over the past 15 years. Audio products have morphed from separate components to integrated solutions of analog and digital electronics, wireless, speakers and enclosures and the factory's teams have evolved to support these products.

There is a lot more to getting a product into production than sending out a bid and picking the lowest cost producer. Timing of vendor selection is often early on in the process to get the sign-on of the vendor team and to insure development that fits both the brand's and factory fabrication processes. Product price points and quality, projected quantities, ability and to ramp up production, are just some of the selection criteria. Much of the heavy lifting in product development as well as design for manufacturability are often the responsibility of the vendor with oversight by the Brand.

Mike Klasco with Menlo Scientific has been involved with product development and international factory selection for over 30 years. Mike will discuss specifically how he addresses bringing the Speak2Me to market and the roles that the Sourcing and Supply Chain people must perform to ensure product and market success today.

Tutorial 9 Saturday, October 1
5:00 pm – 6:15 pm Room 406AB

THE PHYSICS OF AUDITORY PROXIMITY AND ITS EFFECTS ON INTELLIGIBILITY AND RECALL

Presenter: **David Griesinger**, David Griesinger Acoustics, Cambridge, MA, USA

This tutorial will detail and demonstrate the many ways ears and speech have evolved to utilize the phase relationships of vocal harmonics to separate sonic information from complex and noisy environments. Early reflections randomize these phases, and in most rooms at some distance the ability to detect these phases is lost. Speech becomes difficult to localize, intelligibility decreases, and information is difficult to recall. We call this the Limit of Localization Distance, or LLD. We believe the number of seats within the LLD is one of the most important determinants of acoustic quality. The tutorial participants will be able to hear for themselves how the LLD can be easily determined by simply walking around with eyes closed during a lecture, a rehearsal or a performance.

Broadcast/Streaming Media 10 Saturday, October 1
5:00 pm – 6:30 pm Room 404AB

CONSIDERATIONS FOR PODCAST AUDIO

Moderator: **Rob Byers**, American Public Media, St. Paul, MN, USA

Panelists: *Brendan Baker*, Love + Radio
Kate Bilinski, Mix Engineer/Music Editor, Serial Season 2
Dylan Keefe, Radiolab, WNYC
Michael Raphael, The New Yorker Radio Hour and Rabbit Ears
Kyle Wesloh, American Public Media/ Minnesota Public Radio

Podcasts are an exciting opportunity for those of us in the audio world. Listenership increases yearly , companies dedicated solely to podcast production are springing up left and right, and a new audience is falling in love with listening. As a result, the traditional broadcast roles of audio engi-neer, sound designer, and producer are morphing and blending, so we must find ways to position ourselves as a valuable resource. Experimentation is rampant and there is more possibility than ever for creating engaging sound—but our audience is listening in challenging environments! None-theless we continue to push our art forward. This panel of experts will deal with these topics as well as discuss the craft of sound design and the challenges of mixing for the podcast audience.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Networked Audio 8 **Saturday, October 1**
5:00 pm – 6:30 pm **Room408B**

AN OVERVIEW OF AES67

Presenters: **Terry Holton**, Yamaha

Interoperability among high-performance, low-latency media networks from various different manufacturers was once a dream but is now becoming reality with the adoption and growing implementation of the AES67 standard. Terry Holton (Yamaha) will discuss how it all works and what lies ahead.

This session is presented in association with the AES Technical Commit-*tee on Network Audio Systems*

Historical Event
H3 - CONTROL SYSTEMS AND ELECTROACOUSTICAL CONSIDERATIONS FOR LARGE-SCALE LOUDSPEAKER ARRAYS: PAST, PRESENT & FUTURE
Saturday, October 1, 5:00 pm – 6:30 pm
Room 403A

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

Four decades of system development will be highlighted, in timelines for both control and electroacoustical domains. In historical review, this presentation highlights the evolutionary process from control-only networks to networked digital audio, and migration paths from powered speaker arrays to line array elements to beam-steerable systems. Various speaker array topologies in use over time will be reviewed as a series of developments having taken place since the 6th International Conference (Sound Reinforcement, Nashville, 1988) and the 13th International Conference (Computer-controlled sound systems, Dallas, 1994), with content at these landmark AES events having foreshadowed today's high-powered loudspeaker arrays that incorporate beam-steering technology. Emerging trends will be examined and potential future developments contemplated. Of potential interest to sound reinforcement technicians, system operators, installed-system designers, rental service providers, and product development engineers.

Student Event and Career Development
RECORDING COMPETITION—PART 2
Saturday, October 1, 5:00 pm – 7:00 pm
Room 501ABC

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.

5:00 Category 3—Modern Studio Recording & Electronic Music
6:00: Category 4—Sound for Visual Media

Session P23 **Sunday, October 2**
9:00 am – 10:30 am **Room 409B**

SIGNAL PROCESSING—PART 4

Chair: **Dave Berners**

9:00 am

P23-1 Physically Derived Synthesis Model of an Aeolian Tone—*Rod Selfridge, Joshua D. Reiss, Eldad J. Avital, Xiaolong Tang*, Queen Mary University of London, London, UK

An Aeolian tone is the whistle-like sound that is generated when air moves past a cylinder or similar object; it is one of the primary aeroacoustic sound sources. A synthesis model of an Aeolian tone has been developed based on empirical formula derived from fundamental fluid dynamics equations. It avoids time consuming computations and allows real-time operation and interaction. Evaluation of the synthesis model shows frequencies produced are close to those measured in a wind tunnel or simulated through traditional offline computations.
Convention Paper 9679

9:30 am

P23-2 Binaural Auditory Steering Strategy: A Cupped Ear Study for Hearing Aid Design—*Changxue Ma,*[1] *Andrew B. Dittberner,*[1] *Rob de Vries*[2]
[1]GN Resound, Glenview, IL, USA
[2] GN Resound, Eindhoven, The Netherlands

The binaural auditory steering strategy for hearing aids focuses to achieve a better hearing experience in terms of both "better ear" listening and situational awareness. We have taken into account the binaural auditory spatial processing strategy to optimize the acoustic beamforming filters. We investigate in this paper how human beings achieve better listening with cupped ears based on the head-related transfer function (HRTF) of the subjects with both open ears and cupped ears. We define the metrics better ear index and situational awareness index. We show that cupped ears can significantly improve the better ear index and the open ears has better situational awareness. We can automatically choose one hearing aid with directivity similar to a cupped ear and another hearing aid similar to an open ear to achieve both better intelligibility and situational awareness in certain acoustic environments.
Convention Paper 9680

10:00 am

P23-3 On the Effect of Artificial Distortions on Objective Performance Measures for Dialog Enhancement— *Matteo Torcoli, Christian Uhle*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

The objective evaluation of dialog enhancement systems using computational methods is desired to complement the subjective evaluation using listening tests. It remains a challenge because for this application neither were performance measures specifically designed, nor were existing measures systematically analyzed. This work investigates eight objective performance measurement tools originally developed for audio and speech coding, speech enhancement, or source separation. To this end, a set of basic distortions is presented and used to simulate degradations that are common in dialog enhancement. The effect of the artificial distortions on the performance measures is quantified by means of a so-called response score that is proposed here.
Convention Paper 9681

Session P24 **Sunday, October 2**
9:00 am – 10:30 am **Room 403B**

POSTERS: SPATIAL AUDIO AND RECORDING & PRODUCTION

9:00 am

P24-1 The Duplex Panner: A Cross-Platform Stereo Widening Tool—*Samuel Nacach*, New York University, Saadiyat Island, Abu Dhabi, UAE; Element Audio Group

Binaural processing, Ambiophonics, and the Haas effect are some of the most popular methods of achieving 3D audio, virtual space, and wide mixes. However, for commercial music applications Binaural audio is limited by its HRTF and headphone dependency; Ambiophonics by its cross-talk cancellation and a listening sweet-spot; and the Haas effect by its hard pans. To solve this, this paper examines the recently developed Duplex Panner technique to better understand its cross-platform capabilities for playback in both headphone and loudspeaker systems. A subjective experiment comparing processed and unprocessed versions of the same material, along with an in depth analysis of the effects on phase and its psychoacoustics implications will help define the limitations of the Duplex Panner algorithm.
Convention Paper 9672

9:00 am

P24-2 Women in Audio: Contributions and Challenges in Music Technology and Production—*Marlene Mathew*,[1] *Jennifer Grossman*,[1] *Areti Andreopoulou*[2]
[1]New York University, New York, NY USA
[2]LIMSI-CNRS, Université Paris-Saclay, Orsay, France

Even though there is a persistent gender gap, the impact of women in audio continues to grow. The achievements of pioneering women in audio go back to the mid-twentieth century. Their accomplishments in the entertainment and academic sectors have helped pave the way for a record number of women in the next generation of women in audio. This paper presents recent contributions as well as discusses the representation of women in audio, the gender gap and challenges women face in this field. Various options, policies, and initiatives are also proposed with the goal towards gender parity. The authors hope to provide a valuable contribution to the research on women in audio, and in particular women's representation in audio engineering, production, and electronic music.
Convention Paper 9673
[Paper presented by Jennifer Grossman]

9:00 am

P24-3 An Evaluation of Two Microphone Techniques for Bleed Reduction Using Independent Component Analysis— *Mark Rau, Wieslaw Woszczyk*, McGill University, Montreal, QC, Canada

Independent component analysis is tested as an approach to reduce the effects of instrument bleed in a multi-track audio recording. Two microphone techniques are examined for a case with two sound sources. The first technique uses each microphone facing a separate source, while the second uses both microphones facing the same source. The separation of the microphones as well as polarity is altered to observe the effects. Both techniques are tested using a simulation as well as with a physical experiment. The first technique is shown to be effective less than 50% of the time with minimal bleed reduction. The second technique is shown to be effective between 68–96% of the time and can have an average bleed reduction of up to 2 dB with instances up to 4.6 dB.
Convention Paper 9674

9:00 am

P24-4 Music Production for Dolby Atmos and Auro 3D— *Robert Jay Ellis-Geiger*, City University of Hong Kong, Hong Kong, SAR China

This paper explores 3D cinematic spatial audio techniques adapted and developed for Dolby Atmos and Auro 3D by the author for the production of music for a feature film. The main objective was to develop a way of recording and mixing music that could translate to both Dolby Atmos and Auro-3D cinema reproduction systems.
Convention Paper 9675

9:00 am

P24-5 Smartphone-Based 360° Video Streaming/Viewing System including Acoustic Immersion—*Kenta Niwa*,[1] *Daisuke Ochi*,[1] *Akio Kameda*,[1] *Yutaka Kamamoto*,[2] Takehiro Moriya[2]
[1]NTT Media Inelligence Laboratories, Tokyo, Japan
[2]NTT Communication Science Labs, Atsugi-shi, Kanagawa-ken, Japan

In virtual reality (VR), 360º video services provided through head mounted displays (HMDs) or smartphones are widely used. Despite the fact that the user's viewpoint seamlessly changes, sounds through the headphones are fixed even when images change in correspondence with user head motion in many 360º video services. We have been studying acoustic immersion technology that is achieved by, for example, generating binaural sounds corresponding to the user head motion. Basically, our method is composed of angular region-wise source enhancement using array observation signals, multichannel audio encoding based on MPEG-4 Audio Lossless Coding (ALS), and binaural synthesizing of enhanced signals using head related transfer functions (HRTFs). In this paper we constructed a smartphone-based real-time system for streaming/viewing 360º video including acoustic immersion and evaluated it through subjective tests.
Convention Paper 9676

9:00 am

P24-6 Analysis on the Timbre of Horizontal Ambisonics with Different Decoding Methods—*Yang Liu*,[1,2] *Bosun Xie*[1]
[1]South China University of Technology, Guangzhou, China
[2]Guoguang Electric Company Limited, Guangzhou, China

Based on different physical and psychoacoustic considerations, there are various Ambisonics decoding methods. The perceived performances of reproduction depend on the order and decoding method of Ambisonics. The present paper analyzes and compares the timbre coloration in horizontal Ambisonics with basic, maximize energy location vector (max-rE) and in-phase decoding methods, respectively. The binaural loudness level spectra (BLLS) for Ambisonics reproduction are calculated by using Moore's revised binaural loudness model and then used as the criterion to evaluate the timbre coloration. Results indicate that, overall, the basic decoding method is superior to the other two methods in terms of the deviation of BLLS. This conclusion is valid for Ambisonics with various orders as well as in the central and off-central listening position.
Convention Paper 9677

9:00 am

P24-7 Distance Factor for Frontal Sound Localization with Side Loudspeakers—*Yuzuru Nagayama, Akira Saji, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

In our laboratory, we conducted auditory experiments to examine elevation localization by changing frequency spectra without using HRTFs. As a result, we achieved elevation localization at (azimuth, elevation) = (0, 0), (0, 30), (0, 60) by two loudspeakers arranged just beside the listener. However, it was only investigated with fixed distance between the ear and the loudspeaker to be 70 cm. When distance was not 70 cm, the elevation perception was unidentified. Therefore, the arrangement of loudspeakers was changed in this research to improve the result. As a result, the rate of in-the-head localization was increased, and the distance of the perceived sound image became shorter and shorter.
Convention Paper 9678

Tutorial 10 **Sunday, October 2**
9:00 am – 10:30 am **Room 404AB**

THE ART OF THE SAMPLE LIBRARY

Presenters: **Douglas Morton**, Q Up Arts
Jim Norman, Guitar Center

Developing sample libraries for virtual and hardware instruments requires a complex balance of recording knowledge, the real-world behavior of the instrument sampled, detailed editing, and final programming of the software of hardware product. This tutorial will review the general process of planning, recording, editing, and programming a sample library using real world case studies on pianos and drums.

Workshop 9 **Sunday, October 2**
9:00 am – 10:30 am **Room 408B**

CANCELED

Product Development 8 **Sunday, October 2**
9:00 am – 10:30 am **Room 402AB**

UI/UX DESIGN THINKING AND BEST PRACTICES FOR AUDIO PLUGINS

Presenters: **Connor Reviere**
Gebre Waddell, Soundways, Memphis, TN, USA

Design thinking encompasses specific methods and approaches used by modern designers during UI/UX development. Our talk will cover this subject through the lends of audio plug-in development. We will also cover best practices and common interface design elements through data analysis of the top 100 commercial plugins. We will cover topics including toolbars, flat design, parameter outlining, cross-modality and 1080 p/4 k compatibility. Included will be a brief video of users discussing their experience with interfaces. We will conclude with honoring our users—the producers and engineers who craft the recordings we love.

Recording & Production 11 **Sunday, October 2**
9:00 am – 10:30 am **Room 403A**

RAW TRACKS: SOUND OF MOTOWN/WHAT'S GOIN ON?

Presenters: **Bob Ohlsson**
Mark Rubel, The Blackbird Academy, Nashville, TN, USA; Pogo Studio, Nashville, TN, USA

Audio engineering legend Bob Ohlsson will discuss in detail the recording of the iconic Marvin Gaye song "What's Goin' On?" in a special 90-minute interview. The wide-ranging conversation will include the stories of recording this and other classic Motown recordings, an analysis of the multi-tracks, soloing individual elements, related anecdotes, and more.

Sound Reinforcement 9 **Sunday, October 2**
9:00 am – 10:30 am **Room 406AB**

TOO MUCH LOW END?

Presenter: **Howard Page**, Clair Global, Lititz, PA, USA

Is too much low end ruining the listening experience for the audience? Over the years, Howard Page has prescribed low-end diets for those struggling to unveil clarity in their concert audio systems. This session will reveal Howard's approach to removing the variables to deliver a consistent listening experience that places the creative focus where it belongs—at the mixing console.

Student Event and Career Development
STUDENT RECORDING CRITIQUES
Sunday, October 2, 9:30 am – 10:30 am
Room 515B

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

Students! Come and get pointers, tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system and then receive feedback from a panel of renowned industry professionals. Students should sign up at the student (SDA) booth immediately following the first SDA meeting, and bring mix files on USB memory stick or hard drive to the SDA booth at least two hours prior to the session they sign up for. Files should be AIFF or WAVE, 44.1 Khz, 24 bit. Stereo mixes should be a single interleaved file, up to 5.1 surround mixes should be a set of clearly labeled discrete mono files. Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by PMC and Genelec.

Sunday, October 2 10:00 am Room 405

Technical Committee Meeting on Recording Technology and Practices

Session P25 Sunday, October 2
10:45 am – 12:15 pm Room 409B

SIGNAL PROCESSING—PART 5

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

10:45 am

P25-1 An Efficient Algorithm for Clipping Detection and Declipping Audio—*Christopher Laguna, Alexander Lerch*, Georgia Institute of Technology, Atlanta, GA, USA

We present an algorithm for end to end declipping, which includes clipping detection and the replacement of clipped samples. To detect regions of clipping, we analyze the signal's amplitude histogram and the shape of the signal in the time-domain. The sample replacement algorithm uses a two-pass approach: short regions of clipping are replaced in the time-domain and long regions of clipping are replaced in the frequency-domain. The algorithm is robust against different types of clipping and is efficient compared to existing approaches. The algorithm has been implemented in an open source JavaScript client-side web application. Clipping detection is shown to give an f-measure of 0.92 and is robust to the clipping level.
Convention Paper 9682

11:15 am

P25-2 A Two-Pass Algorithm for Automatic Loudness Correction—*Alexey Lukin, Russell McClellan, Aaron Wishnick*, iZotope, Cambridge, MA, USA

Loudness standards for broadcast audio, such as BS.1770, establish target values for the integrated loudness, true peak level, and short-term loudness of a record. Compliance with these three targets can be challenging when the dynamic range of a record is high, so software for automatic loudness correction is important for speeding up the workflow of post-production engineers. This work reviews existing software implementations of automatic loudness correction and proposes a new algorithm that provides efficient simultaneous correction of all three targets.
Convention Paper 9683

11:45 am

P25-3 A Low Computational Complexity Beamforming Scheme Concatenated with Noise Cancellation—*Jin Xie, Sungyub Daniel Yoo, Kapil Jain*, Marvell Technology Group Ltd., Santa Clara, CA, USA

In this paper we present a microphone beamforming algorithm. This algorithm has been implemented in Marvell's proprietary digital signal processor embedded in Marvell's audio codec chip. This beamforming algorithm features (1) easy to implement; (2) sound source localization (SSL) and sound source tracking, (3) single in single out frequency domain noise cancellation. Lab tests show that the performance is better than the reference existing codec.
Convention Paper 9684

Tutorial 11 Sunday, October 2
10:45 am – 12:15 pm Room 404AB

PRACTICAL SCIENCE AND BEST PRACTICES

Presenter: **Greg Riggs**, Guitar Center, Westlake Village, CA, USA

Practical science is a basic review of common audio and physics principles that serve as foundational building blocks for critical listening, system design, and system operation in both live sound and recording environments. Best practices describe how the science fits into audio workflows at all levels.

Workshop 10 Sunday, October 2
10:45 am – 12:15 pm Room 408B

CRITICAL LISTENING: EAR TRAINING IN AUDIO EDUCATION

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

Panelists: *Mark Bassett*, SAE Institute, Byron Bay, NSW, Australia; University of Sydney, Sydney, NSW, Australia
Jason Corey, University of Michigan, Ann Arbor, MI, USA
Kazuhiko Kawahara, Kyushu University, Fukuoka, Japan
Sean Olive, Harman International, Northridge, CA, USA

Considering the interests and growth of ear training in the audio communities, it is timely and important to have a chance to share and discuss the opinions from the experts about necessary features and methods that assist trainees in acquiring the critical listening ability with efficiency, both for personal and group training.

The current workshop aims to let workshop attendees experience and compare the characteristic functions of various ear-training programs through hands-on demonstrations by the panelists. While the workshop locally aims to provide the attendees with chance to experience theoretical and empirical matters of ear training programs around the world, it also globally aims to consider the importance of "listening" in the current video-oriented society.

This session is presented in association with the AES Technical Committee on Perception and Subjective Evaluation of Audio Signals

Networked Audio 9 Sunday, October 2
10:45 am – 12:15 pm Room 403A

AES67 INTEROPERABILITY TESTING —THE PLUGFEST REPORT

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

Panelists: *Fredrik Bergholtz*, Swedish Radio
Terry Holden, Yamaha
Jamie Laundon, BBC
Bruce Olson, AES
Nicholas Sturmel, Digigram

Attendees, audio engineers, and integrators will understand the role of different network technologies, how to communicate with IT with regards to the network, as well as understand the enhancements that are now available to time sensitive networks today.

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

Product Development 9 — **Sunday, October 2**
10:45 am – 12:15 pm — **Room 402AB**

NEW STANDARD FOR ELECTRICAL AND MECHANICAL TRANSDUCER MEASUREMENTS

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

This tutorial reports on the progress made in the development of a new IEC standard dedicated to electrical and mechanical measurements (part B) complementing the acoustical measurements (part A) presented at previous AES conventions. Both standards are applicable to all kinds of transducers, active and passive loudspeakers, and other sound reproduction systems. 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. The electrical and mechanical data are required for transducer and system design based on numerical simulations (FEA, BEA) and digital signal processing protecting the transducer and correcting the transfer behavior actively.

Sound Reinforcement 10 — **Sunday, October 2**
10:45 am – 12:15 pm — **Room 406AB**

LIVE SOUND FOR HOPSCOTCH OPERA

Presenter: **Edward A. Carlson**, Freelance, Los Angeles, CA, USA

In November 2015 Los Angeles was struck by Hopscotch, a large-scale, site-specific mobile opera performance produced by experimental opera company The Industry. A one-of-a-kind experience for audience and crew alike, Hopscotch consisted of 24 simultaneously performed scenes which took place inside 18 limousines and in various locations across downtown Los Angeles. The project's lead A/V Technician, Edward Carlson, talks about the many audio challenges faced while building a show of this scale and complexity. Between fighting frequencies in downtown, lugging antennas to the roof of an apartment building, and live streaming all of it to the audience's headphones in a central hub, it's no understatement that Hopscotch was a tremendous feat.

Student Event and Career Development
STUDENT DELEGATE ASSEMBLY MEETING—PART 2
Sunday, October 2, 10:45 am – 12:15 pm
Room 501ABC

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.

Broadcast/Streaming Media 11 — **Sunday, October 2**
11:30 am – 1:00 pm — **Room 502AB**

GREASE LIVE—THE MIXER'S PERSPECTIVE (SPECIAL EVENT)

Moderator: **J. Mark King**, Production & Music Mixer – Primetime Emmy Awards, Dancing With The Stars, Grammy Nominations Concert, ACMs, CMAs, Santa Clarita, Ca, USA

Panelists: *Biff Dawes*, music mixer
Eric Johnston, booth sfx/underscore mixer
Pablo Munguia, Maraka Music, Beverly Hills, CA, USA

Follow the process from concept to "fade to black" as Mark King shares the methods and tech-niques of mixing television's most challenging live broadcast of *Grease Live*.

This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Special Event
SAMMY NESTICO, "THE SOLDIER SESSIONS" WITH THE US ARMY JAZZ AMBASSADORS—IN SURROUND
Sunday, October 2, 1:00 pm – 2:30 pm
Room 501ABC

Presenter: **Jim Anderson**, New York University, New York, NY, USA

Noted composer and arranger, Sammy Nestico, has had a long career in the television and film industry. He has arranged and conducted projects for artists such as Bing Crosby, Sarah Vaughan, Toni Tennille, Frank Sinatra, Phil Collins, Barbra Streisand, and Count Basie, among others. As orchestrator, he has worked on nearly seventy television programs, including "Mission: Impossible," "Mannix," "Charlie's Angels," and "The Mod Squad." With the US Army Jazz Ambassador's release, *The Soldier Sessions*, Mr. Nestico has gone back to his roots as big band arranger and composer and created twelve new charts for the recordings. Producer/Engineer, Jim Anderson, will present 5.1 mixes performed by America's Big Band.

Session P26 — **Sunday, October 2**
1:30 pm – 3:00 pm — **Room 409B**

SIGNAL PROCESSING—PART 6

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

1:30 pm

P26-1 Discrete-Time Implementation of Arbitrary Delay-Free Feedback Networks—*Dave Berners*,[1,2] *Jonathan S. Abel*[2]
[1]Universal Audio
[2]CCRMA, Stanford University, Stanford, CA, USA

The delay-free feedback loop can be directly implemented in discrete time by separately discretizing the forward and backward transfer functions and simultaneously solving the resulting linear system for the nodes connecting the filters within the loop. The ability to form the solutions rests upon the fact that, at sample n, the output of a discrete-time linear system is a linear function of the input to the system at sample n. This technique allows for relatively simple calculation of coefficients for certain time-varying feedback systems, and allows for inclusion of memoryless nonlinearities inside feedback loops. We show that the technique can be generalized to discretize an arbitrary network of LTI systems arranged in multiple-loop feedback networks. Two examples are presented: one time-varying system and one nonlinear system.
Convention Paper 9685

2:00 pm

P26-2 The Time-Varying Bilinear Transform—*Jonathan S. Abel,*[1] *Dave Berners*[1,2]
[1]CCRMA, Stanford University, Stanford, CA, USA

[2]Universal Audio

The discretization of continuous-time systems is considered, and an extension of the bilinear transform to the case of time-varying systems is introduced. Termed the "time-varying bilinear transform," the transform generates a sequence of digital filter coefficients in response to continuous-time system changes that keeps the digital filter state compatible with the changing digital filter coefficients. Accordingly, transients in the digital filter output that don't appear in the continuous system output are avoided. For an Nth-order continuous-time system, a step change in the system produces a sequence of N intermediate sets of digital filter coefficients, bracketed by what would be generated by the bilinear transform applied to the initial and final systems. Sequences are tabulated for direct and transpose canonical forms and first-order and second-order systems, and examples of first-order and second-order analog filters with time-varying components are presented.
Convention Paper 9686

2:30 pm

P26-3 Active Equalization for Loudspeaker Protection—
Christopher Painter,[1] *Kapil Jain*[2]
[1]Marvell Semiconductor, Inc., Longmont, CO, USA
[2] Marvell Technology Group Ltd., Santa Clara, CA, USA

We present a time-varying linear equalization algorithm whose purpose is to protect a loudspeaker from damage under high drive conditions. It is suitable for implementation on a low-cost digital signal processor, often integrated on the same die as a high-performance audio codec. A typical application is in a portable wireless (e.g., Bluetooth) loudspeaker. For a given driver and enclosure design, the algorithm allows the power output of the loudspeaker to be maximized while introducing only minimal coloration or distortion. During the loudspeaker design phase, the parameters of the algorithm can be easily tuned by the designer, further optimizing the overall design for power output, robustness and low distortion.
Convention Paper 9687

Tutorial 12 **Sunday, October 2**
1:30 pm – 3:00 pm **Room 406AB**

AUDIO CABLE DEMYSTIFIED

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

Overview of common cables used for analog and digital audio transmission. Basic construction, applications, and considerations. Objective comparison of marketing vs science.

Tutorial 13 **Sunday, October 2**
1:30 pm – 3:00 pm **Room 404AB**

AUDIO FORENSICS: OVERVIEW 2

Presenter: **Daniel Shores**, Sono Luminus, Boyce, VA, USA; Shenandoah Conservatory Music Production and Recording Technology, Winchester, VA, USA

Daniel Shores has recorded numerous critically acclaimed solo piano records for several labels. The styles and music have covered everything from Bach, amplified and looped piano, an ensemble with other instruments, and genre from Jazz to chamber orchestra. Each session has its own unique character and voice, highlighting the repertoire and the individual musicians' playing style.

This tutorial will demonstrate different techniques and approaches to accentuating the music and capturing the sound. He will discuss how both music and player can dictate how a few inches can truly make the music jump off the page. The session includes pictures, video, and sound examples both from the Sono Luminus Studios and captured in the new Steinway Hall in New York using their new Spirio piano.

Networked Audio 10 **Sunday, October 2**
1:30 pm – 3:00 pm **Room 403A**

AES70 AS A COMPANION CONTROL PROTOCOL TO AES67 AUDIO TRANSPORT

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

Panelists: *TBA*

A full media networking solution has three parts: (1) an audio transport scheme, to move audio signals around the network; (2) a system control scheme, to provide device control and connection management services; and (3) a directory/discovery scheme—a network "telephone book" to keep track of device names and addresses. For (1), the AES has developed AES67; for (2), the AES has developed AES70; and for (3), the AES is now working on requirements for next-generation standards. This report summarizes the relationships of these three elements and the current status of (2) and (3).

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

Product Development 10 **Sunday, October 2**
1:30 pm – 3:00 pm **Room 402AB**

IMPROVED LOUDSPEAKER PERFORMANCE USING DC-COUPLED AMPLIFIERS

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

DC coupling in audio amplifiers has been considered more dangerous than advantageous because a DC signal generated somewhere in the audio chain might damage the transducer silently. However, the transducer would benefit from a small electrical DC signal that keeps the voice coil at the desired working point in the gap and copes with the DC displacement generated dynamically by the transducer nonlinearities and a shift of the coil's rest position due to production variances, aging, gravity, and load changes. An adaptive control system based on a nonlinear model will synthesize the DC signal automatically by minimizing the voice coil offset that can be detected from voltage and current monitoring. This technique allows to compensate for nonlinear distortion, to protect the transducer reliably against mechanical overload, and to operate the voice coil at the Bl maximum generating highest efficiency and maximal output. The workshop explains the control theory, discusses the practical implementation and illustrates the benefits and power requirements on practical demos.

Sound for Picture 6 **Sunday, October 2**
1:30 pm – 3:00 pm **Room 408B**

IMMERSIVE SOUND DESIGN WITH PARTICLE SYSTEMS

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

Presenter: **Nuno Fonseca**, ESTG/Polytechnic Institute of Leiria, Leiria, Portugal; Sound Particles, Leiria, Portugal

Panelistss: *Benjamin Cook*
Jason Jennings
Mark Mangini

Immersive sound formats are growing in use for film & television. Design of realistic and engaging immersive sound fields is challenging the this fast-paced production environment. These Hollywood-based sound designers will discuss their experiences crafting these immersive sound fields and their use of specialty tools like Sound Particles. Hear the sound design experiences of award-winning professionals Mark Mangini (Mad Max: Fury Road), Jason Jennings (Teenage Mutant Ninja Turtles: Out of the Shadows), and Benjamin Cook (Black Sails).

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

Special Event
MIDI MAKES MUSIC, MIDI MAKES MONEY, AND MIDI MAKES CAREERS!
Sunday, October 2, 1:30 pm – 3:00 pm
Room 502AB

Presenters: **Bryan Lanser**, The MIDI Association, La Mirada, CA, USA; The MIDI Manufacturers Association, La Mirada, CA, USA
Kyle P. Snyder, Ohio University, School of Media Arts & Studies, Athens, OH, USA

MIDI was designed for much more than enhancing and controlling music. Knowing how to use MIDI will help you further your career. Learn how it is used in the studio and live productions and concerts.

Historical Event
H4 - HORNS AND WAVEGUIDES: EVOLUTION OF TECHNOLOGY
Sunday, October 2, 1:30 pm – 3:00 pm
Room 408A

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.

Tutorial 14 **Sunday, October 2**
3:15 pm – 4:45 pm **Room 404AB**

CROSSING OVER INTO IMMERSIVE AUDIO

Presenter: **Daniel Shores**, Sono Luminus, Boyce, VA, USA; Shenandoah Conservatory Music Production and Recording Technology, Winchester, VA, USA

In 2015, Sono Luminus began its experimentation and implementation of 9.1 Auro-3D recording. To date, Sono Luminus has created numerous 9.1 recordings, five of which are now commercially available with more on the way. Immersive audio quite literally takes the listening experience of the home consumer to the next level. It allows Sono Luminus the opportunity to deliver an even more in-depth, intriguing, and unique listening experience.

Recording both on location, and in the 100 year old converted church in the Virginia countryside that is now the home of Sono Luminus Studios, Sono Luminus focuses on techniques for capturing native immersive audio, rather than mixing for the format. In the end though, it is all about the serving the music, and we have taken the opportunity to work with the musicians and composers to develop recordings that bring out the all of the musical nuances in a way not possible before.

Examples in this tutorial will demonstrate various styles of music including choral, early music, Celtic, percussion, electronics, experimental music just to name a few demonstrating vastly different instrumentation and sonic textures.

Tutorial 15 **Sunday, October 2**
3:15 pm – 4:45 pm **Room 406AB**

DIFFUSION: DEFINED AND DEMYSTIFIED

Presenter: **Alejandro Bidondo**, Universidad Nacional de Tres de Febrero - UNTREF, Buenos Aires, Argentina

In order to expand the description of diffusion surfaces towards their effects in the sound field, several technical concepts and new definitions will be presented in this Diffusion Tutorial: Impulse responses: measurement and analysis, “texture” of an impulse response; statistical & temporal acoustic behavior of a room: the Crossover time; room´s early and late sound fields, measuring the Crossover time. What is a diffuser? … It is all about Autocorrelation. How many diffusers should be installed in a...? (theater, studio, concert hall, etc.) Describing the acoustic properties of a surface: absorption, diffusion and scattering coefficients. What is a diffuse sound field? Defining and quantifying the sound field diffusivity: SFDC - A systematic experiment. Calculation method of SFDC. Examples and applications.

Networked Audio 11 **Sunday, October 2**
3:15 pm – 4:45 pm **Room 402AB**

WHO OWNS THE AUDIO NETWORK, IT OR AV?

Presenter: **Patrick Killianey**, Yamaha Professional Audio, Buena Park, CA, USA

Audio networks blur the line between the audio and IT trades. Regardless of who specifies, purchases, and maintains the network infrastructure, both professions are often (and should be) involved in the process. In this workshop we will present some thoughts to help smooth this interaction. We will address differences between professional dispositions and best practices. We will also offer concepts for robust, scalable design that capitalizes on existing IT infrastructure, while creating a clear delineation of responsibility between Audio and IT.

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