AES 145TH CONVENTION PROGRAM

OCTOBER 17–20, 2018

JAVITS CONVENTION CENTER, NY, USA

The Winner of the 145th AES Convention Best Peer-Reviewed Paper Award is:

The Effect of Pinnae Cues on Lead-Signal Localization in Elevated, Lowered, and Diagonal Loudspeaker Configurations—*Wesley Bulla*, Belmont University, Nashville, TN, USA; *Paul Mayo*, University of Maryland, College Park, MD, USA

Convention Paper 10066

To be presented on Thursday, Oct. 18, in Session 7—Perception—Part 2

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

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

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

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

(a) The paper was accepted for presentation at the AES 145th 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 145th AES Convention Student Paper Award is:

Musical Instrument Synthesis and Morphing in Multidimensional Latent Space Using Variational, Convolutional Recurrent Autoencoders—*Emre Çakir, Tuomas Virtanen*, Tampere University of Technology, Tampere, Finland

Convention Paper 10035
To be presented on Wednesday, October 17, in Session 2—Signal Processing—Part 1

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Session P1 **Wednesday, Oct. 17**
9:00 am – 12:00 noon **Room 1E11**

PERCEPTION

Chair: **Elizabeth McMullin**, Samsung Research America, Valencia, CA, USA

9:00 am

P1-1 Improved Psychoacoustic Model for Efficient Perceptual Audio Codecs—*Sascha Disch,[1,2] Steven van de Par,[3] Andreas Niedermeier,[1] Elena Burdiel Pérez,[1] Ane Berasategui Ceberio,[1] Bernd Edler[2]*
[1] Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Erlangen, Germany
[2] Friedrich Alexander University, International Audio Laboratories Erlangen, Erlangen, Germany
[3] University of Oldenburg, Oldenburg, Germany

Since early perceptual audio coders such as mp3, the underlying psychoacoustic model that controls the encoding process has not undergone many dramatic changes. Meanwhile, modern audio coders have been equipped with semi-parametric or parametric coding tools such as audio bandwidth extension. Thereby, the initial psychoacoustic model used in a perceptual coder, just considering added quantization noise, became partly unsuitable. We propose the use of an improved psychoacoustic excitation model based on an existing model proposed by Dau et al. in 1997. This modulation-based model is essentially independent from the input waveform by calculating an internal auditory representation. Using the example of MPEG-H 3D Audio and its semi-parametric Intelligent Gap Filling (IGF) tool, we demonstrate that we can successfully control the IGF parameter selection process to achieve overall improved perceptual quality.
Convention Paper 10029

9:30 am

P1-2 On the Influence of Cultural Differences on the Perception of Audio Coding Artifacts in Music—*Sascha Dick,[1] Jiandong Zhang,[2] Yili Qin,[2] Nadja Schinkel-Bielefeld,[1] Anna Katharina Leschanowsky,[1] Frederik Nagel[1]*
[1] International Audio Laboratories Erlangen, a joint institution of Universität Erlangen-Nürnberg and

Fraunhofer IIS, Erlangen, Germany
[2] Academy of Broadcasting Planning, SAPPRFT, Beijing, China

Modern audio codecs are used all over the world, reaching listeners with many different cultures and languages. This study investigates if and how cultural background influences the perception and preference of different audio coding artifacts, focusing on musical content. A subjective listening test was designed to directly compare different types of audio coding and was performed with Mandarin Chinese and German speaking listeners. Overall comparison showed largely consistent results, affirming the validity of the proposed test method. Differential comparison indicates preferences for certain artifacts in different listener groups, e.g., Chinese listeners tended to grade tonality mismatch higher and pre-echoes worse compared to German listeners, and musicians preferred bandwidth limitation over tonality mismatch when compared to non-musicians.
Convention Paper 10030

10:00 am

P1-3 Perception of Phase Changes in the Context of Musical Audio Source Separation—*Chungeun Kim, Emad M. Grais, Russell Mason, Mark D. Plumbley*, University of Surrey, Guildford, Surrey, UK

This study investigates into the perceptual consequence of phase change in conventional magnitude-based source separation. A listening test was conducted, where the participants compared three different source separation scenarios, each with two phase retrieval cases: phase from the original mix or from the target source. The participants' responses regarding their similarity to the reference showed that (1) the difference between the mix phase and the perfect target phase was perceivable in the majority of cases with some song-dependent exceptions, and (2) use of the mix phase degraded the perceived quality even in the case of perfect magnitude separation. The findings imply that there is room for perceptual improvement by attempting correct phase reconstruction in addition to achieving better magnitude-based separation.
Convention Paper 10031

10:30 am

P1-4 Method for Quantitative Evaluation of Auditory Perception of Nonlinear Distortion: Part II—Metric for Music Signal Tonality and its Impact on Subjective Perception of Distortions—*Mikhail Pakhomov, Victor Rozhnov*, SPb Audio R&D Lab, St. Petersburg, Russia

In the first part of the paper we have noticed that the impact of audible nonlinear distortions on subjective listener preference is strongly dependent on the spectral structure of a test signal. In the second part we propose a method for considering the spectral characteristics of a test signal in the evaluation of the subjective perception of audible nonlinear distortions. To describe the tonal structure of a music signal, a qualitative characteristic, tonality, is taken as a metric, and tonality coefficient is proposed as a measure of this characteristic. Subjective listening tests were performed to estimate how the auditory perception of nonlinear distortions depends on the tonal structure of a signal and the spectral distribution of the noise-to-mask ratio (NMR)
Convention Paper 10032

11:00 am

P1-5 Developing a Method for the Subjective Evaluation of Smartphone Music Playback—*Elisabeth McMullin, Victoria Suha, Yuan Li, Will Saba, Pascal Brunet*, Samsung Research America, Valencia, CA USA

To determine the preferred audio characteristics for media playback over smartphones, a series of controlled double-blind listening experiments were run to evaluate the subjective playback quality of six high-end smartphones. Listeners rated products based on their audio quality preference and left comments categorized by attribute. The devices were tested in different orientations in level-matched and maximum-volume scenarios. Positional variation and biases were accounted for using a motorized turntable and audio playback was controlled remotely with remote-access software. Test results were compared to spatially-averaged measurements made using a multitone stimulus and demonstrate that the smoothness of the frequency response is the most important aspect in smartphone preference. Low frequency extension, decreased levels of nonlinear distortion, and higher maximum playback level did not correlate with higher phone ratings.
Convention Paper 10033

11:30 am

P1-6 Investigation into the Effects of Subjective Test Interface Choice on the Validity of Results—*Nicholas Jillings,[1] Brecht De Man,[1] Ryan Stables,[1] Joshua D. Reiss[2]*
[1] Birmingham City University, Birmingham, UK
[2] Queen Mary University of London, London, UK

Subjective experiments are a cornerstone of modern research with a variety of tasks being undertaken by subjects. In the field of audio, subjective listening tests provide validation for research and aid fair comparison between techniques or devices such as coding performance, speakers, mixes, and source separation systems. Several interfaces have been designed to mitigate biases and to standardize procedures, enabling indirect comparisons. The number of different combinations of interface and test design make it extremely difficult to conduct a truly unbiased listening test. This paper resolves the largest of these variables by identifying the impact the interface itself has on a purely auditory test. This information is used to make recommendations for specific categories of listening tests.
Convention Paper 10034
[Paper presented by Brecht De Man]

Session P2 **Wednesday, Oct. 17**
9:00 am – 11:00 am **Room 1E12**

SIGNAL PROCESSING—PART 1

Chair: **Emre Çakir**, Tampere University of Technology, Tampere, Finalnd

9:00 am

P2-1 Musical Instrument Synthesis and Morphing in Multidimensional Latent Space Using Variational, Convolutional Recurrent Autoencoders—*Emre Çakir, Tuomas Virtanen*, Tampere University of Technology, Tampere, Finland

In this work we propose a deep learning based method—

namely, variational, convolutional recurrent autoencoders (VCRAE)—for musical instrument synthesis. This method utilizes the higher level time-frequency representations extracted by the convolutional and recurrent layers to learn a Gaussian distribution in the training stage, which will be later used to infer unique samples through interpolation of multiple instruments in the usage stage. The reconstruction performance of VCRAE is evaluated by proxy through an instrument classifier and provides significantly better accuracy than two other baseline autoencoder methods. The synthesized samples for the combinations of 15 different instruments are available on the companion website.
Convention Paper 10035

9:30 am

P2-2 Music Enhancement by a Novel CNN Architecture—
Anton Porov,[1] *Eunmi Oh,*[2] *Kihyun Choo,*[2] *Hosang Sung,*[2] *Jonghoon Jeong,*[2] *Konstantin Osipov,*[1] *Holly Francois*[3]
[1] PDMI RAS, St. Petersburg, Russia
[2] Samsung Electronics Co., Ltd., Seoul, Korea
[3] Samsung Electronics R&D Institute UK, Staines-Upon Thames, Surrey, UK

This paper is concerned with music enhancement by removal of coding artifacts and recovery of acoustic characteristics that preserve the sound quality of the original music content. In order to achieve this, we propose a novel convolution neural network (CNN) architecture called FTD (Frequency-Time Dependent) CNN, which utilizes correlation and context information across spectral and temporal dependency for music signals. Experimental results show that both subjective and objective sound quality metrics are significantly improved. This unique way of applying a CNN to exploit global dependency across frequency bins may effectively restore information that is corrupted by coding artifacts in compressed music content.
Convention Paper 10036

10:00 am

P2-3 The New Dynamics Processing Effect in Android Open Source Project—*Ricardo Garcia*, Google, Mountain View, CA, USA

The Android "P" Audio Framework's new Dynamics Processing Effect (DPE) in Android Open Source Project (AOSP), provides developers with controls to fine-tune the audio experience using several stages of equalization, multi-band compressors, and linked limiters. The API allows developers to configure the DPE's multichannel architecture to exercise real-time control over thousands of audio parameters. This talk additionally discusses the design and use of DPE in the recently announced Sound Amplifier accessibility service for Android and outlines other uses for acoustic compensation and hearing applications.
Convention Paper 10037

10:30

P2-4 On the Physiological Validity of the Group Delay Response of All-Pole Vocal Tract Modeling—*Aníbal Ferreira*, University of Porto, Porto, Portugal

Magnitude-oriented approaches dominate the voice analysis front-ends of most current technologies addressing, e.g., speaker identification, speech coding/compression, and voice reconstruction and re-synthesis. A popular technique is all-pole vocal tract modeling. The phase response of all-pole models is known to be non-linear and highly dependent on the magnitude frequency response. In this paper we use a shift-invariant phase-related feature that is estimated from signal harmonics in order to study the impact of all-pole models on the phase structure of voiced sounds. We relate that impact to the phase structure that is found in natural voiced sounds to conclude on the physiological validity of the group delay of all-pole vocal tract modeling. Our findings emphasize that harmonic phase models are idiosyncratic, and this is important in speaker identification and in fostering the quality and naturalness of synthetic and reconstructed speech.
Convention Paper 10038

Acoustics/Psychoacoustics 1 — Wednesday, October 17
9:00 am – 10:30 am — Room 1E07

ACOUSTICAL SIMULATIONS: HOW ALGORITHMS PREDICT AUDIBLE REALITY

Chair: **Dirk Noy**, Walters-Storyk Design Group, Basel, Switzerland

Panelists: *Renato Cipriano*, Walters Storyk Design Group, Belo Horizonte, Brazil
Gabriel Hauser, Walters Storyk Design Group, Basel, Switzerland

Modern acoustical engineering processes are largely based on full virtual (digital) prototyping and engineering, e.g., an acoustical situation is modeled in a specialized software environment to accurately reflect the space in question, whether it is a sports stadium, theater, classroom, boardroom, airport terminal or convention hall. Acoustical causes and effects are then studied and benchmarked within the virtual environment. This 75-minute tutorial will explore some of the methods and algorithms employed within the software tools, and illustrate recent advances such as 3D room modeling, acoustical surface definitions, and loudspeaker representation for electro acoustical prediction. Examples of effective corrective and predictive methods available to study certain acoustical problems will be reviewed, and detailed case studies will be illustrated.

Broadcast/Online Delivery 1 — Wednesday, October 17
9:00 am – 10:30 am — Room 1E17

BROADCASTING IN EMERGENCY SITUATIONS: "IT CAN HAPPEN TO YOU … BECAUSE IT HAPPENED TO US"

Chair: **Howard Price**

Panelists: *Jason Ornellas*, Bonneville Interantional, Sacramento, CA, USA
Alex Roman, Emmis Broadcasting, New York, NY, USA

When disaster strikes your broadcast operation, when audiences and clients alike may be counting on you the most, it's too late to plan or resource your response. From those who lived through Sandy and Harvey, hear the first-person stories of broadcast engineers who experienced catastrophe, and survived to teach you their operational secrets.

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

Networked Audio 1 **Wednesday, October 17**
9:00 am – 10:00 am **Room 1E10**

NETWORK FUNDAMENTALS FOR AUDIO ENGINEERS

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

Everything in our industry is going to IP. It isn't just audio—video, lighting, control and intercom are there, as well. This session forms a great base networking skill for anyone in live sound or media production. If you're comfortable setting up a home network and joining a WiFi hotspot at Starbucks, then this session will launch you comfortably to an intermediate-level user.

By the end of this session, you'll have the concepts to structurally design a medium sized network for an installation, recording studio or live production. You'll also have the basic terminology to interact with an IT department, as needed.

This session will be taught by Patrick Killianey from Yamaha. Patrick produced a YouTube video series called "Network Fundamentals for Professional Audio" which has become part of the curriculum in college programs towards IT and audio production alike.

This live variant of this class will review the key pieces of that video series and chart new territory. We'll refocus on key concepts on Subnet Mask, Gateway, Reserved LAN ranges, IP addresses to avoid, DNS and DHCP. From there, we'll add on common structural topics in audio designs like trunk lines, VLANs and VLAN tagging, Spanning Tree Protocol (STP), Link Aggregation Groups (LAGs), non-blocking backplane architecture, and basics of fiber

Product Development 1 **Wednesday, October 17**
9:00 am – 10:30 am **Room 1E09**

TIME TO MARKET: WHAT CAN YOU DO ABOUT IT?

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

Time to Market may be the most important feature in a product today. Product lifecycles and windows of opportunity are shrinking. Most product companies miss their time to market needs because they don't make major changes to their development strategy and process. In this Product Development session, the presenter will lead a discussion on the nine causes of late to market projects and how to leverage best practices, inside and outside resources, build vs buy, and development technologies to bring your product to market on time!

Sound Reinforcement 1 **Wednesday, October 17**
9:00 am – 10:00 am **Room 1E09**

LIVE SOUND PRODUCTION IN A TV WORLD

Presentes: **Jenny Montgomery**, IATSE
Matt Kraus, IATSE

Jenny Montgomery was the lead A2 on the recent live event broadcast of *Jesus Christ Superstar*. She and other members of the production team will discuss the issues of a one off live event and how the live sound is integrated into the overall production.

Archiving/Restoration 1 **Wednesday, October 17**
9:30 am – 10:30 am **Room 1E13**

ARCHIVING AND BEST PRACTICES FOR MODERN PRODUCTION WORKFLOWS

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

Panelists: *Chuck Ainlay*, METAlliance, Nashville, TN USA
Maureen Droney, The Recording Academy, Los Angeles, CA, USA
Bob Koszela, Iron Mountain
Michael Romanowski, Coast Mastering, Berkeley, CA, USA; The Tape Project

While technology has democratized music creation, the abundance of recording formats and the rate at which technology evolves present a dizzying array of choices to the artist. Additionally, many artists are working outside the music industry pipeline, reaching their audience directly via online portals. In effect, they are their own curators, preserving their artistic output.

The Producers and Engineers Wing of the Recording Academy recently revised their Recommendations for Delivery of Recorded Music Projects and released Recommendations for High-Resolution music production. They have also been a leader in the development of the DDEX recording metadata standard called RIN (Recording Information Notification). This panel will discuss technical best practices and the many challenges artists and producers face when creating and archiving music using modern technology.

Special Event
SE1: AES DIVERSITY AND INCLUSION COMMITTEE TOWN HALL
Wednesday, October 17, 9:30 am – 10:30 am
Room 1E15+16

Moderator: **Leslie Gaston-Bird**, Mix Messiah Productions, Brighton, UK

Panelists: *Ezequiel Morfi*, TITANIO, Buenos Aires, Capital Federal, Argentina
Piper Payne, Neato Mastering, San Francisco Bay Area, CA, USA
Terri Winston, Women's Audio Mission, San Francisco, CA, USA

The AES Diversity and Inclusion Committee strives to ensure diversity in the AES worldwide and the audio industry as a whole by improving accessibility, welcoming diverse genres, embracing emergent audio fields and research, and radiating inclusiveness to all gender and gender identities, races, physical abilities, ages, and nationalities. At this Town Hall, committee chair Leslie Gaston-Bird will give an update on the activities of the committee to date and field questions from audience members.

Student Events/Career Development
EC1: IF I CAN DO IT, YOU CAN DO IT: TONY BONGIOVI'S CAREER AT MOTOWN, THE POWER STATION, AND BEYOND
Wednesday, October 17, 9:30 am – 10:30 am
Room 1E21

Presenter: **Tony Bongiovi**, Bongiovi Acoustics

What do you do when it seems no studio will hire you? How do you go straight from high school to Motown? You don't wait for opportunities to come to you. You don't hope for good luck. You do the work. You knock on doors. You study. You listen. And you go for it! Tony Bongiovi shares the many stories of his success with timeless lessons that inform any career in audio today.

Standards Committee Meeting
SC-04-03 WORKING GROUP ON LOUDSPEAKER MODELING AND MEASUREMENT
Wednesday, October 17, 9:30 am – 11:00 am
Room 1B03

The scope of SC-04-03 includes the specification, modeling, and measurement of the electromechanical and free-field radiation characteristics of loudspeakers, individually and in arrays, intended for propagation of audio signals that are used in professional audio recording, reinforcement, and reproduction applications.

PMC Masters of Audio Program
Wednesday, October 17, 9:30 am – 10:45 am
Room 1E06

UMG/CAPITOL STUDIOS DOLBY ATMOS PLAYBACK

Presenter: **Steve Genewick**

UMG/Capitol Studios Dolby Atmos Playback sessions with Steve Genewick. Featuring music mixed for Dolby Atmos from Elton John, LL Cool J, Chris Walden, INXS, REM, Public Enemy, Bastille, Arturo Sandoval, Snoh Aleegra and many others.

Session P3 **Wednesday, Oct. 17**
10:00 am – 11:30 am **Poster Area**

POSTERS: RECORDING AND PRODUCTION

10:00 am

P3-1 Characterizing the Effect on Linear and Harmonic Distortions of AC Bias and Input Levels when Recording to Analog Tape—*Thomas Mitchell, Christopher Bennett*, University of Miami, Coral Gables, FL, USA

Analog tape recorders introduce both linear distortions and nonlinear distortions to the audio. While the role of the AC bias and input levels on these distortions are well-understood by recording engineers, the impact on specific audio features, for example SNR, fatness, brightness, roughness, and harmonic count is less well described. In this study we examined with high granularity the impact and interactions of several AC bias and input levels on each of these features. We utilized the exponential swept sine acquisition and deconvolution technique to analyze a Scully 280. The results provide a detailed characterization on the tonal character introduced by the recorder. We conclude with level recommendations that could prove important for primary capture, effects processing, and digital emulation.
Convention Paper 10039

10:00 am

P3-2 Microphone Comparison for Snare Drum Recording—*Matthew Cheshire, Jason Hockman, Ryan Stables*, Birmingham City University, Birmingham, UK

We present two experiments to test listener preference for snare microphones within real-world recording scenarios. In the first experiment, listeners evaluated isolated recordings captured with 25 microphones. In the second experiment, listeners performed the same task with the addition of a kick drum and hi-hat as part of a performed drum sequence. Results indicate a prominent contrast between the highest and lowest rated microphones and that condensers were rated higher than other subsets tested. The preference for three microphones significantly changed between the two listening test conditions. A post-test survey revealed that most listeners compared high-frequency characteristics, which were measured using spectral features. A positive correlation was observed between test scores of cardioid microphones and the brightness feature.
Convention Paper 10040

10:00 am

P3-3 Automatic Mixing of Multitrack Material Using Modified Loudness Models—*Steven Fenton*, University of Huddersfield, Huddersfield, West Yorkshire, UK

This work investigates the perceptual accuracy of the ITU-Recommendation BS.1770 loudness algorithm when employed in a basic auto mixing system. Optimal filter parameters previously proposed by the author, which incorporate modifications to both the pre-filter response and the integration window sizes are tested against the standard K-weighted model and filter parameters proposed through other studies. The validation process encompassed two stages, the first being the elicitation of preferred mix parameters used by the mixing system and the subsequent generation of automatic mixes based on these rules utilizing the various filter parameters. A controlled listening test was then employed to evaluate the listener preferences to the completed mixes. It is concluded that the optimized filter parameter set based upon stem type, results in a more perceptually accurate automatic mix being achieved.
Convention Paper 10041

10:00 am

P3-4 Double-MS Decoding with Diffuse Sound Control—*Alexis Favrot*,[1] *Christof Faller*,[1] *Helmut Wittek*[2]
[1] Illusonic GmbH, Uster, Switzerland
[2] SCHOEPS GmbH, Karlsruhe, Germany

The double MS (DMS) setup provides a coincident recording configuration in the horizontal plane for surround sound recording, similar to Ambisonics B-format. DMS uses two cardioids and one dipole microphones, arranged coincidentally. An algorithm for processing DMS recordings is described, which in addition to linear processing provides a diffuse sound gain control and diffuse sound de-correlation. The target stereo or multichannel signals can be made more dry or reverberant with diffuse gain. Diffuse de-correlation improves spaciousness and its importance scales with the number of channels. The consequences of these new controls for the stereophonic image will be depicted.
Convention Paper 10042

10:00 am

P3-5 Real-Time System for the Measurement of Perceived Punch—*Andrew Parker, Steven Fenton, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Previous work has proposed a perceptually motivated model for the objective measurement of punch in a music recording. This paper presents a real-time implementation of the proposed model as a punch metering plug-in. The plug-in presents both momentary and historical punch metrics in the form of a P95 (95th percentile punch measure), Mean punch score, and P95M (95th

percentile punch divided by the mean punch score). The meter's outputs are compared to subjective punch scores derived through a controlled listening test and show a "strong" correlation with Pearson and Spearman coefficients r=0.840 (p<0.001) and rho=0.937 (p<0.001) respectively. The real-time measure of punch could prove useful in allowing objective control and optimization of this feature during mixing, mastering and broadcast.
Convention Paper 10043

10:00 am

P3-6 Algorithm to Determine Downmixing Coefficients from Specific Multichannel Format to Reproduction Format with a Smaller Number of Channels—*Hiroki Kubo, Satoshi Oode, Takehiro Sugimoto, Shu Kitajima, Atsuro Ito, Tomoyasu Komori, Kazuho Ono*, NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

A unified algorithm to derive downmixing coefficients from a specific multichannel format to arbitrary reproduction formats with a smaller number of channels was investigated. The proposed method attaches importance to maintaining both basic audio quality and spatial impression. It involves appropriately changing the positions of channels in source formats, or locating the channels in source formats at equal intervals between the two channels in destination formats when phantom sources are used. Owing to this feature, this method can minimize the use of phantom sources and avoid the deterioration of spatial impression. A subjective evaluation was carried out and the obtained results implied that the proposed algorithm satisfies our requirements.
Convention Paper 10044

10:00 am

P3-7 Subjective Evaluation of Stereo-9.1 Upmixing Algorithms Using Perceptual Band Allocation—*Sungsoo Kim*, New York University, New York, NY, USA

The purpose of this study was to investigate preexisting algorithms for building an upmixing algorithm that converts a stereo signal to 5.1 and 9.1 multichannel audio formats. Using three algorithms (the passive surround decoding method, the Least Mean Squares algorithm, the adaptive panning algorithm), a stereo audio signal was upmixed to 5.1 and 9.1 in Max. The Max patch provides a GUI in which listeners can select one of the upmixing algorithms and control EQ during playback. Perceptual Band Allocation (PBA) is applied for converting the upmixed 5.1 channel audio to 9.1 that contains four height channels (top front left, top front right, top back left, top back right). A subjective listening test was conducted in New York University's MARL Research Lab. LMS algorithm was found to provide more natural and spatial sounds than the other two algorithms. The passive surround decoding method and the adaptive panning algorithm were found to show similar characteristics in terms of low frequency and spaciousness.
Convention Paper 10045

Technical Committee Meeting
Wednesday, October 17, 10:00 am – 11:00 am
Room 1B05
ACOUSTICS AND SOUND REINFORCEMENT

Mix with the Masters Workshop
Wednesday, October 17, 10:00 am – 11:00 am
Booth 458

ANDREW SHEPS

Game Audio & XR 1 — **Wednesday, October 17**
10:15 am – 11:30 am — **Room 1E08**

PRACTICAL RECORDING TECHNIQUES FOR LIVE MUSIC PRODUCTION IN 6DOF VR

Co-moderators: **Calum Armstrong**, University of York, York, UK
Gavin Kearney, University of York, York, UK
David Rivas Méndez, University of York, York, UK

Panelists: *Hashim Riaz*, Abbey Road Studios, London, UK
Mirek Stiles, Abbey Road Studios, London, UK

As virtual and augmented reality technologies move towards systems that can deliver full six degrees of freedom (6DOF), it follows that good strategies must be employed to create effective 6DOF audio capture. In a musical context, this means that if we record an ensemble then we must give the end user the potential to move close and even around audio sources with a high degree of plausibility to match the visuals. This workshop looks at recording strategies that enable 3DOF/3DOF+ and 6DOF for live music performances..

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

Networked Audio 2 — **Wednesday, October 17**
10:15 am – 11:15 am — **Room 1E10**

THE NEW APPLICATION LAYER PROTOCOL FOR GUARANTEED, FUTURE-PROOF AV NETWORKS

Presenters: **Tim Boot**, Digital Product Experience, Meyer Sound
Henning Kaltheuner, d&b audiotechnik GmbH. Backnang, Germany
Jeff Rocha, L-Acoustics

Deterministic networking on AV networks was impossible 10 years ago before the Avnu Alliance was founded. Its members have since solved this by collaborating to develop the original Avnu Pro AV specifications and certification program, which focused on creating a baseline foundation for deterministic networking. Now, the industry is focused on building a truly interoperable application layer on top of it.

Major manufacturers in the Pro AV space under the umbrella of Avnu Alliance have taken the lead to create a new application layer protocol to directly address this focus. This presentation will offer background on the new application layer protocol and discuss how it helps AV manufacturers promise deterministic, reliable, and future-proof delivery of networked media.

Broadcast/Online Delivery 2 — **Wednesday, October 17**
10:30 am – 12:00 noon — **Room 1E07**

PODCAST PRODUCTION ROUNDTABLE

Chair: **Rob Byers**, NPR, Washington, DC

Panelists: *Haley Shaw*, Gimlet Media
Alex Trajano, Audible

Building on the success of last year's session, Rob Byers (American Public Media and Criminal) will host a special roundtable discussion on the art and techniques of podcast production. Specifically focused on productions that tell stories, this session will bring together engineers, sound designers, and producers to talk about their top-of-the-charts work. The conversation will range from specific mixing and sound design techniques to considerations for an audience that is often listening on smart speakers and mobile devices. Join us for a unique view in to this blossoming format—and come with questions!

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

AoIP Pavilion
Wednesday, October 17, 10:30 am – 11:00 am
AoIP Pavilion Theater

INTRODUCTION TO THE AOIP TECHNOLOGY PAVILION

Presenter: **Terry Holton**, Yamaha R&D Centre, London, UK

The Audio-over-IP Technology Pavilion is a significant new initiative created by the AES in partnership with the Alliance for IP Media Solutions (AIMS). The pavilion will promote professional IP media networking as well as providing the latest information about this rapidly developing field through practical demonstrations and an extensive presentation program. This session will provide an introduction to the various aspects of the pavilion including the AIMS demonstration system, exhibitors' Pods, and the AoIP Presentation Theater.

Project Studio Expo
Wednesday, October 17, 10:30 am – 10:45 am
PSE Stage

OPENING SESSION & NEUMANN ANNOUNCEMENT

Neumann will be presented with a "Service to Industry" award to mark the company's 90th anniversary being celebrated throughout the Convention—it's long, standard-setting legacy of engineering prowess in the development of microphones for studio and live performance and for Neumann's advancement of microphone technology. All attendees are invited to attend the presentation, which takes place on the PSE Recording Stage, inside the AES New York 2017 Exhibition Hall in New York City's Jacob Javits Convention Center at 10:30am on the Wednesday, October 17. Presenting the award on behalf of the AES will be Graham Kirk, AES International Sales Director and Al Schmitt, multi-GRAMMY Award winning engineer/mixer. Receiving the award will be Wolfgang Fraissinet, President of Neumann.

Software@AES
Wednesday, October 17, 10:30 am – 11:00 am
Software Pavilion

AUDIOSOURCERE

Immersive/Spatial Audio 1 **Wednesday, October 17**
10:45 am – 12:15 pm **Room 1E17**

SPATIAL AUDIO-VIDEO CREATIONS FOR MUSIC, VIRTUAL REALITY AND 3D PROJECTORS—CASE STUDIES

Moderator: **Tomasz Zernicki**, Zylia sp. z o.o., Poznan, Poland

Panelists: *Florian Grond*
Yao Wang
Edward Wersocki

The goal of the workshop is to present spatial audio-video creations in practice. Professional audio engineers and musicians will talk about their 360, 3D and ambient productions combining the sound and the vision. Among discussed projects there are going to be "Unraveled"—360 spatial experience where the listener finds themselves in the middle of the entire recording. Furthermore, the speakers will tell about the process of making 3D audiovisual footage displayed in the 360° dome as well as spatial recordings of the concert music. The workshop will focus especially on the usage of spherical microphone arrays that enable to record entire 3D sound scene. The separation of individual sound sources in post-production and Ambisonics give creators unlimited possibilities to achieve unique audio effects.

Product Development 2 **Wednesday, October 17**
10:45 am – 12:15 pm **Room 1E09**

POWER AMPLIFICATION FOR HIGH RESOLUTION AUDIO

Presenter: **John Dawson**, Jade Electronics Ltd., Cambridge, UK

Driving modern passive loudspeakers to realistic levels without audible distortion can be surprisingly demanding—more so when the source is high resolution audio. And, although the peak power levels are much lower, achieving full dynamic range with headphones can also present tricky design problems. This tutorial reviews the various types of linear amplifier—particularly classes A, AB, B, D, G and H—and examines some of the trade-offs involved in designing and implementing some recent commercial designs for both speakers and headphones. A number of the author's measurements of real products—not all of them good—will be shown.

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

Recording & Production 1 **Wednesday, October 17**
10:45 am – 12:15 pm **Room 1E21**

MICROPHONES—CAN YOU HEAR THE SPECS? A MASTER CLASS

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

Panelists: *Eddy Bøgh Brixen*, EBB-consult, Smørum, Denmark; DPA Microphones, Allerød, Denmark
David Josephson, Josephson Engineering, Inc., Santa Cruz, CA, USA

There are numerous microphones available to the audio engineer. It's not easy to compare them on a reliable basis, often the choice of the model is made on the basis of experience or perhaps just habits—or just because it looks nice. Nevertheless, there is valuable information in the microphone specifications. This master class held by well-known microphone experts of leading microphone manufacturers demystifies the most important microphone specs and provides the attendees with up-to-date information on how these specs are obtained and can be interpreted. Furthermore, many practical audio demonstrations are given in order to help everyone to understood how the numbers relate to the perceived sound.

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

Student Events/ Career Development
EC2: MAKING LIVE JAZZ AND SOUND LIKE JAZZ!
Wednesday, October 17, 10:45 am – 12:15 pm
Room 1E13

Presenters: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA; off-beat-open-hats recording & sound reinforcement
Nadezhda Rakhmanova, Mariinsky Theatre, Russia

This workshop, ideal for students, new engineers, and more experienced engineers new to jazz and acoustic music situations, will discuss what "the sound" of jazz and acoustic music might be and different approaches to amplifying a selection of ensembles in order to appropriately scale the sound to different sized venues, from small clubs to large theaters and festival stages. Are there desirable characteristics? Should miking techniques be the same as, or different from (more common) rock shows? Should the sound system be the same as for other genres? What situations do engineers working local and regional shows find themselves in? What can help make the day go smoothly for you and the musicians and make the event a musically appropriate sonic experience for the audience?

Special Event
SE00: HOW NEXT GENERATION AUDIO INNOVATIONS WILL RESHAPE THE MOBILE DEVICE EXPERIENCE
Wednesday, October 17, 11:00 am – 12:00 noon
Room 1E15+16

Presenter: **Jyri Huopaniemi**, Head of Product and Technology Strategy at Nokia Technologies

Sound has been a fundamental building block of mobile telephony and mobile devices since the very beginning of the connectivity era. Significant advances have since been made in the quality and variety of audio-based applications. However, there are areas where audio has not yet reached its true potential. In this talk we focus on areas of mobile audio that are currently gaining consumer attention: communication, immersion, and user-generated content.

Today, people are telling their stories with technology, and they are increasingly using mobile devices to capture the full richness of their lives; in pictures, videos, and sounds. We discuss solutions and opportunities which allow consumers to capture and share audio with depth, direction, and detail.

The use of voice has quickly expanded from person-to-person communication in voice and conference calls to voice-based control. The rapid emergence of intelligent assistants places significant new requirements on the quality of sound capture and rendering. In parallel, the basic function of mobile devices, namely voice calls, has undergone significant improvements in quality. We offer a glimpse to some of the challenges and future opportunities in voice-based applications and immersive voice and audio services.

Mix with the Masters Workshop
Wednesday, October 17, 11:00 am – 12:00 noon
Booth 458

PETER KATIS

AoIP Pavilion
Wednesday, October 17, 11:00 am – 11:30 am
AoIP Pavilion Theater

OPTIMIZING NETWORKS FOR MEDIA

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

This session will examine the network technologies used to optimize a network for modern media transport. With this knowledge, attendees will have a much better understanding of how to manage networks with mixed traffic and have the basic knowledge to begin diagnosing networked audio issues. This session will cover TCP vs. UDP, Unicast vs. Multicast and Quality of Service (QoS).

Live Sound Events
Wednesday, October 17, 11:00 am – 11:45 am
Live Production Stage

LS01 - SPORTS BROADCASTING IN DOLBY ATMOS

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

Panelists: *Rob France*, Dolby Laboratories, London, UK
Karl Malone, NBC Universal, Stamford, CT, USA

Dolby Atmos is ready for prime time. It has proven to be the leading format for immersive audio for live sport broadcasts. With Dolby Atmos, fans are brought into the action, hearing not only the commenters but also the crowd noise all around them, immersing them in the action like never before. And while immersive audio won't necessarily be right for every sport, for the ones that it is a good match for, the impact will be incredible. Moderated by Roger Charlesworth and featuring, Karl Malone, Director of Sound Design, NBC Sports and NBC Olympics, this session cover the elements involved in the production and delivery of immersive audio for live sports. Don't miss this opportunity to hear firsthand about NBC Sports' broadcasts of Atmos for Olympics coverage at both the 2016 Rio Summer Games, the 2018 PyeongChang Winter Games and now Notre Dame home football.

Project Studio Expo
Wednesday, October 17, 11:00 am – 11:45 am
PSE Stage

PSE01 - THE ART OF COMPRESSION

Presenter: **Jack Joseph Puig**, Record Executive/Producer/Mixer, Hollywood, CA, USA

Compression is the only real tool we have that can move the time due to the fact it's based on time constants. The Art of Compression is not adding lots of level but understanding that compression is dynamic! Compression is not static. EQ is static. The ability to look at it as a shaper of attack and release as it relates to how the ear and brain perceives how instruments and vocals related to each other "time" / "feel" wise is the magic and expert level of compression use. This does not preclude the importance of raising the "Average/RMS" level! Or even peak limiting. Remember in the end we are making music/art that must have a emotional effect on the listener and of course delivering the artist intent emotionally. I will demonstrate how it is possible to change the feel and why it matters.

Sponsored by Neumann, Shure, SSL

Software@AES
Wednesday, October 17, 11:00 am – 11:30 am
Software Pavilion

BEST SERVICE

Sound Reinforcement 2 **Wednesday, October 17**
11:15 am – 12:15 pm **Room 1E12**

AUDIO PRODUCTION FOR CORPORATE EVENTS

Moderator: **Lee Kalish**, Positive Feedback llc, Kingston, NY, USA

Lee Kalish will host a panel of experts in live sound production in the corporate event world. This will include sound system design to fit the venue, RF coordination for wireless mics and communications, music production and band support.

PMC Masters of Audio Program
Wednesday, October 17, 11:15 am – 12:15 pm
Room 1E06

RECURRENCE

Presenter: **Daniel Shores**, Sono Luminus, Boyce, VA, USA; Shenandoah Conservatory Music Production and Recording Technology, Winchester, VA, USA

Daniel Shores "Recurrence" by the Icelandic Symphony Orchestra in Dolby Atmos 7.1.4.

Applications in Audio 1 **Wednesday, October 17**
11:30 am – 12:00 noon **Room 1E10**

VACUUM TUBE ELECTRONICS FOR SOLID STATE ENGINEERS

Presenters: **Scott Dorsey**, Kludge Audio, Williamsburg, VA, USA
David Hill, Crane Song Inc., Superior, WI, USA; Dave Hill Designs, Superior, WI, USA
David Josephson, Josephson Engineering, Inc., Santa Cruz, CA, USA

Vacuum tube electronics is conceptually different than solid state devices, in part because the size of each stage requires minimizing the number of stages of electronics, in part because the high impedances involved almost always require the use of matching transformers. All of these changes have severe consequences for the designer in how circuits are laid out, how feedback is used, and in fundamental circuit and system design. Younger engineers who grew up with solid state theory would benefit from an understanding of the constraints on vacuum tube systems and the consequent decisions and compromises to be made in their design.

AoIP Pavilion
Wednesday, October 17, 11:30 am – 12:00 noon
AoIP Pavilion Theater

THE AUDIO PARTS OF SMPTE ST 2110 EXPLAINED

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

This session will explain the fundamentals and possible variations of audio transport within ST 2110 and its compatibility with AES67.

Software@AES
Wednesday, October 17, 11:30 am – 12:00 noon
Software Pavilion

SONIBLE

Game Audio & XR 2 **Wednesday, October 17**
11:45 am – 12:15 pm **Room 1E08**

LIVE CODING TUTORIAL: REAL-TIME OPEN SOUND COMMUNICATION FROM UNREAL ENGINE 4 TO MAX 7

Presenter: **Timothy Vallier**, Orion Healthcare Technology, Omaha, NE, USA

Developers often face challenges when attempting to integrate visual based virtual reality environments and auditory based virtual auditory environment signal processing. This live coding tutorial session will demonstrate the ease of bridging two disparate environments vis Open Sound Control (OSC). Alone, these two environments offer a great deal of control and complexity for their respective domains (Max in Audio and UE4 in Visual). Together, they offer complete control and synchrony for virtual reality applications requiring multichannel virtual auditory environments. This tutorial will demonstrate a from-scratch approach to getting the two platforms set up to communicate with one another.

Mix with the Masters Workshop
Wednesday, October 17, 12:00 noon – 1:00 pm
Booth 458

RUSSELL ELEVADO

AoIP Pavilion
Wednesday, October 17, 12:00 noon – 12:30 pm
AoIP Pavilion Theater

AOIP: ANATOMY OF A FULL-STACK IMPLEMENTATION

Presenter: **Ievgen Kostiukevych**, European Broadcasting Union, Le Grand-Saconnex, Genéve, Switzerland

The presentation will explain that there is much more to consider when building an AoIP infrastructure than just the AES67 standard. The challenges of synchronization and clocking, discovery and registration, device and network control will be explained, and some solutions will be offered.

Project Studio Expo
Wednesday, October 17, 12:00 noon – 1:00 pm
PSE Stage

PSE02 - PRODUCING & RECORDING; FROM THE MIC TO THE TRACK

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

A guest producer matches our singer with a microphone, then demonstrates the process of producing a vocal track.

Software@AES
Wednesday, October 17, 12:00 noon – 12:30 pm
Software Pavilion

RME

Special Event
SE2: AWARDS PRESENTATION AND KEYNOTE ADDRESS
Wednesday, October 17, 12:15 pm – 2:15 pm
Room 1E15+16

Opening Remarks: President David Scheirman
Convention Chairs Paul Gallo, Valerie Tyler, Jonathan Wyner
Program: AES Awards Presentation by John Krivit
Keynote Address by Thomas Dolby

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.

Board of Governors Award
Eddie B. Brixen
Edgar Choueiri
Linda Gedemer
Matt Klassen
Andres Mayo
Valeria Paolomin
Alberto Pinto
Daniel Rappaport
Angieska Roginska
Lawrence Schwedler
Jeff Smith
Nadja Wallaszkovits

Fellowship Award
Gustavo Borner
Christopher Freitag
Leslie Ann Jones
Hyunkook Lee
Andres Mayo
Bruce Olson
Xiaojun Qiu
Rafa Sardina
Frank Wells

Distinguished Service Medal Award
David Bialik

Keynote Speaker
Thomas Dolby—musician, technologist and educator—has a 35-year career of technical innovation. Perhaps most widely known for his seminal song and music video "She Blinded Me with Science," Dolby blazed the trail for electronic music creators with his recordings and imaginative videos. He is also known for his work as a producer, as a composer of film scores, as a technical consultant on emerging entertainment platforms and as a filmmaker. Since the Fall of 2014, Dolby has held the post of Homewood Professor of the Arts at Johns Hopkins University in Baltimore, MD. Thomas Dolby's AES Keynote address will focus on next-generation sound technologies, in particular adaptive/non-linear music and audio for games, VR/AR, "hearables" and other new media platforms. The title of his speach is "The Conscious Sound Byte."

A big difference between "real" and "electronic" sounds is that electronic sounds have zero awareness of each other." Sound bytes blindly follow orders, and fire off (usually) as instructed by a human. Yet musicians playing "real" instruments listen, resonate, and respond to the music, the room, and to each other, in a matter of microseconds.

In the hands of master arranger or programmer, this is not a problem. Many of the nuances of real music can be simulated quite effectively as processor speed, bandwidth, and resolution improve. But as entertainment becomes more interactive, with games and augmented reality and "wearable" technologies, it is increasingly vital that electronic sounds and music learn an awareness of the context in which they are playing.

Soon, all the accumulated craft and skills of a century of post-production legacy will have to operate in real time, controlled not by the programmer, but by the users themselves via the choices they make. Is it time for us to reconsider why our sound and music files are so "dumb" and rigid?

AoIP Pavilion
Wednesday, October 17, 12:30 pm – 1:00 pm
AoIP Pavilion Theater

RAVENNA AND ST 2110

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

This session will explain the fundamentals of RAVENNA and how it relates to AES67 & ST 2110, explaining why RAVENNA offers the fastest and most flexible options for AES67 & ST 2110 compliance.

Software@AES
Wednesday, October 17, 12:30 pm – 1:00 pm
Software Pavilion

MELODYNE

Mix with the Masters Workshop
Wednesday, October 17, 1:00 pm – 2:00 pm
Booth 458

ALAN MEYERSON

AoIP Pavilion
Wednesday, October 17, 1:00 pm – 1:30 pm
AoIP Pavilion Theater

TELOS INFINITY: BREAKING THE MATRIX WITH AES67, NEXT GENERATION INTERCOM

Presenter: **Martin Dyster**, The Telos Alliance

Telos Infinity IP Intercom is a complete reimagining of broadcast communications technology developed by the Telos Alliance engineering team that invented AoIP for broadcast in 2003. Infinity replaces outmoded matrix technology with an advanced, distributed fully AES67 compliant network solution that provides superior functionality in a simplified, more elegant form. Being matrix-free allows plug-and-play networked hardware and software devices to be added to the system as part of a planned or ad-hoc change, without ever worrying that you might exceed the number of available ports on a matrix.

Live Sound Events
Wednesday, October 17, 1:00 pm – 1:45 pm
Live Production Stage

LS02 - MIKING THE SPORTS BROADCAST

Presenter: **Ben Escobedo**, Shure Market Development & Michael Mason, President of CP Communications

Pro sports broadcasts attract millions of viewers every weekend, with audio being a critical part of the experience. Join Ben Escobedo of the Shure Market Development team and Michael Mason, President of CP Communications, to walk through some of the microphone types and sound capture techniques required to flawlessly execute a high-profile sports broadcast.

Software@AES
Wednesday, October 17, 1:00 pm – 1:30 pm
Software Pavilion

FABFILTER

AoIP Pavilion
Wednesday, October 17, 1:30 pm – 2:00 pm
AoIP Pavilion Theater

AES70 AT A GLANCE

Presenter: **Ethan Wetzell**, OCA Alliance

In the world of media networking, much attention is given to content transport, but an equally important component is how connected devices can be controlled. This presentation will discuss the AES70 open control standard, how it fits within the media networking landscape, and will serve as an introduction to its structure and capabilities for device control.

Software@AES
Wednesday, October 17, 1:30 pm – 2:00 pm
Software Pavilion

SONARWORKS

Standards Committee Meeting
SC-02-12-L TASK GROUP ON OPEN CONTROL ARCHITECTURE
Wednesday, October 17, 2:00 pm – 4:00 pm
Room 1B03

This task group is responsible for the AES70 network audio system control standard, also known as the Open Control Architecture, or OCA, and for related information documents to help developers of AES70-compliant equipment. AES70 offers a wide range of device control and media stream connection management features for professional media networks of all sizes, and includes robust reliability and security features. AES70 became a standard in 2015; the group has recently completed a revision that is available for public comment.

Mix with the Masters Workshop
Wednesday, October 17, 2:00 pm – 3:00 pm
Booth 458

MICHAEL BRAUER

AoIP Pavilion
Wednesday, October 17, 2:00 pm – 2:30 pm
AoIP Pavilion Theater

AUDIO OVER IP: PRACTICAL REQUIREMENTS FOR REAL-WORLD USABILITY

Presenter: **Brad Price**, Audinate, Portland, OR, USA

As professional audio installations become synonymous with IP networks, the industry has been abuzz with discussions of protocols and other necessary transport foundations. In reality, people work with solutions that build on these underlying concepts and provide a plethora of additional functionality that makes audio-over-IP usable in the real world. This presentation explores how coherent solutions enhance the experience of audio networking, and how features beyond transport are crucial to the widespread adoption of the technology by the channel and end users.

Live Sound Events
Wednesday, October 17, 2:00 pm – 2:45 pm
Live Production Stage

LS03 - AUDIO NETWORKING IN PRODUCTION

Presenter: **TBD**

Project Studio Expo
Wednesday, October 17, 2:00 pm – 2:45 pm
PSE Stage

PSE03 - VOCALS: A MASTER'S VIEW

Presenter: **David Darlington**, Waves, London, UK

The producer/engineer explains the methods of how to get the most from the vocals with some advice and tricks. Dave Darlington: Grammy Awards-winner, Composer, Recording & Mastering Engineer (Avicii, David Guetta, Sting, Oz).

Software@AES
Wednesday, October 17, 2:00 pm – 2:30 pm
Software Pavilion

MAGIX

Product Development 3 **Wednesday, October 17**
2:15 pm – 3:15 pm **Room 1E09**

SYSTEM-LEVEL SOLUTIONS FOR ACTIVE NOISE CANCELING HEADPHONES

Moderator: **Mike Klasco**, Menlo Scientific Ltd.

Presenters: *Mark Donaldson*
Mark Noon

Active noise canceling headphones were first commercialized over 30 years ago yet remain a challenging product to design, manufacture, and quality control. Best practice design methods for high performance passive headphones are not fully adequate when developing those with an active noise canceling feature. Physical construction from materials, earcup wall thickness and ribbing, intra-cavity sealing and mechanical integrity of glue joints, speaker driver parameters and circuit topology all contribute to an active noise cancellation headphone's unit-to-unit performance and stability in mass production. Our talk will focus on the unique construction and performance aspects that require particular attention during the development phase, as well as the need for precision, speed, and automation during end of line testing and calibration.

Session P4 **Wednesday, Oct. 17**
2:30 pm–5:30 pm **Room 1E11**

TRANSDUCERS—PART 1

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

2:30 pm

P4-1 **Numerical Optimization Strategies for Acoustic Elements in Loudspeaker Design**—*Andri Bezzola*, Samsung Research America, Valencia, CA USA

Optimal design of acoustic loudspeaker design elements, such as waveguides and phase plugs, often requires extensive experience by the designer. Numerical simulations and optimization algorithms can aid in reducing the design-test-optimize cycle that is traditionally applied. A general mathematical framework for numerical optimization techniques is introduced and three algorithms of design optimization are reviewed: parameter optimization, shape optimization, and topology optimization. This paper highlights strengths and drawbacks of each method with the use of real-world design of a waveguide and two phase plugs. Where possible, the results are confirmed with prototypes and measurements. The work shows that excellent results can be achieved in just one design iter-

ation with the help of numerical optimization methods.
Convention Paper 10046

3:00 pm

P4-2 An Acoustic Model of the Tapped Horn Loudspeaker— *Marco Berzborn,*[1,2] *Michael Smithers*[3]
[1] Dolby Laboratories, San Francisco, CA, USA
[2] RWTH Aachen University, Aachen, Germany
[3] Dolby Laboratories, McMahons Point, NSW, Australia

A lumped-parameter model of the Tapped Horn loudspeaker–a design where the loudspeaker driver radiates into the throat as well as the mouth of the horn simultaneously–is presented. The model enables the estimation of the far-field sound pressure response from the Thiele/Small parameters of the loudspeaker driver and an additional analytic two-port matrix representation of the Tapped Horn. Simulations, performed using the model for subwoofers, are compared to measurements from an actual loudspeaker.
Convention Paper 10047

3:30 pm

P4-3 A Survey and Analysis of Consumer and Professional Headphones Based on Their Objective and Subjective Performances—*Sean Olive, Omid Khonsaripour, Todd Welti*, Harman International Inc., Northridge, CA, USA

In previous studies [1–3], we presented two statistical models that predict listeners' sound quality preferences of headphones based on deviations in their measured frequency response. In this paper the models are applied to 156 different consumer and professional headphones that include a wide range of brands, prices, and headphone categories (e.g., in-ear, on-ear, and around-ear). The goal was to gain a better understanding of how these factors influence the subjective and objective performances of the headphone. The predicted preference ratings of the headphones were compared to ratings given by five different headphone review organizations to determine their correlation. Headphones designed to the current IEC/ITU/EBU standards produce significantly lower sound quality ratings.
Convention Paper 10048

4:00 pm

P4-4 Minimizing Costs in Audio Devices through Efficient End-of-Line Testing—*Wolfgang Klippel*, Klippel GmbH, Dresden, Germany

Variances in the parts and uncertainties in the assembling process degrade the performance and the reliability of the manufactured devices. Defective units increase the manufacturing cost if detected during end-of-line testing or increase the after-sales cost if an undetected failure occurs in the field. This paper addresses the role of end-of-line testing to reduce both kinds of failures and to maximize the performance/cost ratio as seen by the end-user. The selection of sensitive and fast measurements, which can be performed under manufacturing conditions, is the basis for the PASS/FAIL classification. The paper shows that optimal production limits used in EoL-testing minimize the overall cost by considering a clear product definition, information from the particular design, statistical data from manufacturing process, and traceability of the field rejects. The general concept presented here is illustrated using practical examples from automotive and other applications.
Convention Paper 10049

4:30 pm

P4-5 Horn Driver Based on Annular Diaphragm and the Side-Firing Compression Chamber—*Alexander Voishvillo*, JBL/Harman Professional Solutions, Northridge, CA, USA

This work proposes a new type of compression driver based on an annular flexural diaphragm and a topology that combines part of the diaphragm radiating directly into the horn and the other part loaded by a side-firing compression chamber. This configuration's design is very simple compared to its predecessors. Acoustical theory, numerical simulations, matrix electrical-mechanical-acoustical model, and the results of real transducer measurements substantiate the new design. The new driver provides performance on par with more complex transducers of a similar format.
Convention Paper 10050

5:00 pm

P4-6 On the Efficiency of Flown vs. Ground Stacked Subwoofer Configurations—*Etienne Corteel, Hugo Coste-Dombre, Christophe Combet, Youchim Horyn, François Montignies*, L-Acoustics, Marcoussis, France

Modern live loudspeaker systems consist of broadband sources, often using variable curvature line sources, combined with subwoofers. While it is common practice to fly the broadband sources to improve energy distribution in the audience, most subwoofer configurations remain ground-stacked because of practical constraints and alleged efficiency loss of flown configurations. This article aims at evaluating the efficiency of flown subwoofers for large audiences as compared to their ground-stacked counterparts. We use finite element simulations to determine the influence of several factors: baffling effect, trim height. We show that flown configurations remain efficient at the back of the venue while reducing the SPL excess at the front of the audience.
Convention Paper 10051

Session P5 Wednesday, Oct. 17
2:30 pm – 5:30 pm Room 1E12

SIGNAL PROCESSING—PART 2

Chair: **Remi Audfray**, Magic Leap, San Francisco, CA, USA

2:30 pm

P5-1 A Pseudoinverse Technique for the Pressure-Matching Beamforming Method—*Miller Puckette, Tahereh Afghah, Elliot Patros*, University of California San Diego, San Diego, CA, USA

In this work an extension to the pressure-matching beamforming method (PMM) that's well-suited for transaural sound field control is presented. The method aims to improve performance at *dark* points, locations relative to the array where sound pressure is minimized; without producing noticeable artifacts at *bright* points, locations where acoustic interference is minimized. The method's new performance priorities result from replacing Tikhonov regularization, which is conventionally used in PMM, with a purpose-built regularization strategy for solving

the pseudoinverse of ill-conditioned matrices. Discussions of how this method's formulation affects the filter design process and of performance comparisons between this and conventional PMM filters are included.
Convention Paper 10052

3:00 pm

P5-2 Analog Circuits and Port-Hamiltonian Realizability Issues: A Resolution Method for Simulations via Equivalent Components—*Judy Najnudel,*[1,2] *Thomas Hélie,*[2] *Henri Boutin,*[2] *David Roze,*[2] *Thierry Maniguet,*[1] *Stéphane Vaiedelich*[1]
[1] CNRS-Musé de la Musique, Paris, France
[2] IRCAM, Paris, France

In order to simulate the Ondes Martenot, a classic electronic musical instrument, we aim to model its circuit using Port-Hamiltonian Systems (PHS). PHS have proven to be a powerful formalism to provide models of analog electronic circuits for audio applications, as they guarantee the stability of simulations, even in the case of non-linear systems. However, some systems cannot be converted directly into PHS because their architecture cause what are called realizability conflicts. The Ondes Martenot circuit is one of those systems. In this paper a method is introduced to resolve such conflicts automatically: problematic components are replaced by equivalent components without altering the overall structure nor the content of the modeled physical system.
Convention Paper 10053

3:30 pm

P5-3 Practical Realization of Dual-Shelving Filter Using Proportional Parametric Equalizers—*Rémi Audfray, Jean-Marc Jot, Sam Dicker,* Magic Leap, Inc., San Francisco, CA, USA

Proportional Parametric Equalizers have been proposed as an efficient tool for accurate magnitude response control within defined constraints. In particular, a combination of shelving filters can be used to create a 3-band parametric equalizer or tone control with minimal processing overhead. This paper picks up on this concept, fully develops the filter control equations, and proposes a look up table based implementation of the dual-shelving filter design.
Convention Paper 10054

4:00 pm

P5-4 Measuring Audio when Clocks Differ—*Mark Martin, Jayant Datta, Xinhui Zhou*, Audio Precision, Beaverton, OR, USA

This paper examines what happens when a digital audio signal is measured with respect to a digital reference signal that was created based on a different clock. The resultant change in frequency and time depends on the degree of mismatch between clocks and may introduce a significant amount of distortion into the measurements. But the distortion is different from the types normally considered important in audio systems and may obscure other types of distortion, e.g., harmonic, IMD, and noise, that are more important to accurately assess. A difference in clocking essentially creates a difference in sample rates. Therefore, sample rate conversion methods can be used to mitigate the discrepancy. Although this approach is effective, it cannot be used when the sample rate difference is too small, it can be computationally intensive, and it cannot entirely eliminate the effects of a clocking difference. This paper describes a much simpler and more effective technique that requires minimal computation.
Convention Paper 10055

4:30 pm

P5-5 Reducing Musical Noise in Transform Based Audio Codecs—*Elias Nemer, Jeff Thompson, Ton Kalker*, DTS/XPERI, Calabasas, CA, USA

This paper addresses the problem of musical noise in transform-based audio codecs. This artifact occurs when encoding audio segments with a noise-like spectrum—at low bit rates where signal quantization results in significant zero-valued coefficients. Due to the quantization commonly used, bands containing several non-zero coefficients are quantized to only one or two, giving rise to a musical artifact. This has been identified in other codecs, such as in CELT/OPUS , where a special transform is used prior to quantization to remedy the problem. In this paper we provide a modified approach consisting of a *Hadamard* transform combined with an interleaving scheme. Simulation shows the proposed method has a lower complexity and yields improved perceptual scores as measured by PEAQ.
Convention Paper 10056

5:00 pm

P5-6 Statistical and Analytical Approach to System Alignment—*Juan Sierra,*[1] *Jonathan Kamrava,*[2] *Pablo Espinosa,*[2] *Jon M. Arneson,*[2] *Paul Kohut*[2]
[1] Stanford University, Stanford, CA, USA
[2] Meyer Sound Laboratories, Berkeley, CA, USA

The current project describes the design of a tuning or alignment processor for a system based on multiple satellite speakers and a single subwoofer. It explains the methodology used to solve this problem and the procedure to arrive to a viable solution. On one side, the procedure was based on filter design techniques to optimize phase relationships; however, these phase relations are often unknown due to the possibility of changing the relative positions of the speakers. Accordingly, a statistical analysis was used to determine the most stable set of parameters across different speaker locations and acoustical environments, implying that the same alignment parameters can be implemented in multiple circumstances without significant performance degradation.
Convention Paper 10057

Archiving/Restoration 2 **Wednesday, October 17**
2:30 pm – 3:30 pm **Room 1E13**

HELP! I HAVE A TAPE MACHINE

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

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

Analog tape is increasingly becoming a vintage format. As this medium heads towards its twilight, it's crucial to share knowledge of how to maintain, evaluate, and source machines. This panel discussion will cover topics such as assessing the tape path, recognizing issues with motors, calibration techniques, maintaining electronics and mechanics, evaluation of heads and what to look for when purchasing a machine. The panel will consist of highly esteemed maintenance engineers with centuries of analog tape machine knowledge between them, including John French of JRF Magnetics, Bob Shuster, and Dan Zellman, well-known technicians in the New York area, moderated by Noah Simon from the Recorded Music program at New York University.

Broadcast/Online Delivery 3 **Wednesday, October 17**
2:30 pm – 4:00 pm **Room 1E07**

ADVANCED AUDIO FOR TELEVISION BROADCAST: UNDERSTANDING AUDIO FOR ATSC 3.0

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

Panelistss: *Robert Bleidt*, Fraunhofer USA Digital Media Technologies, San Jose, CA, USA
Tim Carroll, Dolby Laboratories, San Francisco, CA, USA
Kazuho Ono, NHK Science & Technology Research Laboratories, Tokyo, Japan
Skip Pizzi, NAB, Washington DC, USA
Jim Starzynski, ATSC

Immersive and personalized audio workflows are now ubiquitous in the industry and finally breaching into consumer delivery; not just through cable TV and streaming but also over consumer broadcast channels.

Whether to a personal device in binaural through headphones, or to a multichannel home theater playout system, advanced transmission systems are now up and running to deliver the excitement and realism of a three-dimensional playback space with relative ease, carrying metadata to adapt to any listening environment.

ATSC 3.0, Hi-Vision 22.2, and Mpeg-H have shot beyond theory and are now up and running in approaching 100 transmitters in cities and countries around the world.

Again, AES brings you the world's foremost experts on the subject as they bring us up to speed on the very exciting milestones bested over the past year.

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

Historical Events 1 **Wednesday, October 17**
2:30 pm – 4:00 pm **Room 1E08**

LIFE AND DEATH OF THE 30TH STREET STUDIO

Presenter: **Dan Mortensen**, Dansound Inc.; Friends of the 30th Street Studio, Seattle, WA, USA

Expanded technical look at entire history of the legendary Columbia Records recording studio in NYC, from its construction as a Presbyterian church in 1875, through its conversion to and use as a studio starting in 1948, until its sale in 1981 and demolition in 1982.

This talk will trace the history of the building itself and the evolution and changes in microphones, recording media and devices, studio monitoring, mixing consoles, outboard gear, and recording and studio staff. We will see some of the incredible talent that went through the studio, but there will not be time to do justice to the amazing array of people who worked there in all capacities and at one time or another, although the author is separately trying.

Recording & Production 2 **Wednesday, October 17**
2:30 pm – 3:30 pm **Room 1E21**

MASTERING CONTENT FOR LOUDNESS SPECIFICATIONS: A DISCUSSION OF BEST PRACTICES, PITFALLS, AND OTHER PRACTICAL CONSIDERATIONS FOR ENGINEERS AND EDUCATORS

Presenters: **Adam Ayan**, Gateway Mastering Studios, Portland, ME USA
Adam Grover, Sterling Sound
Bob Katz, Digital Domain Mastering, Orlando, FL, USA
Alex Kosiorek, Central Sound at Arizona PBS, Phoenix, AZ, USA
Jamie Tagg, Indiana University, Bloomington, IN, USA; Stagg Sound Services, LLC

Navigating today's loudness standards can be confusing to young engineers and others not frequently working with these requirements. This panel of audio experts and educators will address the justification for these standards, what they mean for us and how to optimize our workflow to accommodate their requirements over a variety of commercial delivery formats. Particular attention will be payed to balancing problematic sound sources with dialogue, the state-of-the-art in metering, formats and delivery methods, and workflow considerations with which young engineers and interns should be familiar.

Sound Reinforcement 3 **Wednesday, October 17**
2:30 pm – 5:30 pm **Room 1E10**

RF SUPER SESSION

Chair: **James Stoffo**, Independent, Key West, FL, USA

Panelists: *Mark Brunner*, Shure Incorporated, Niles, IL, USA
Joe Ciaudelli, Sennheiser, Old Lyme, CT USA
Henry Cohen, CP Communications
Jim Dugan, JetWave Wireless
Jason Glass, Clean Wireless
Jackie Green, Alteros, Stow, OH, USA
Dave Missal, Sennheiser, Old Lyme, CT, USA
Cameron Stuckey, Professional Wireless Systems, New York, NY, USA
Gary Trenda, Prairie Du Sac, WI, USA
Karl Winkler, Lectrosonics, Rio Rancho, NM, USA

Tremendous changes in the RF ecosystem are currently taking place affecting available spectrum, equipment, workflow, and required knowledge base in the deployment of wireless microphones and similar equipment. This RF Super Session will discuss in depth the main topics concerning RF purchasing decisions and deployments; RF spectrum and regulatory changes, new equipment to comply with the updated technical rules, best and advanced RF practices including advanced filtering techniques and RF over fiber. Panelists include leading RF technicians and engineers, manufacturer product specialists and the manufacturers' government relations people.

Student Events/Career Development
EC3: STUDENT RECORDING CRITIQUES
Wednesday, October 17, 2:30 pm – 3:30 pm
Room 1E06

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

Students! Come and get 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 at any stage of their studies can sign up to participate. Sign up at the student (SDA) booth immediately on arrival at the convention and deliver stereo or non-interleaved 5.1 channel mixes as 44.1 Khz/24 bit AIFF or WAVE files, to the SDA booth when you sign up. If you sign up, please make sure you arrive on time at the start of the session, otherwise alternates will be placed on the schedule in your place. 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.

AoIP Pavilion
Wednesday, October 17, 2:30 pm – 3:00 pm
AoIP Pavilion Theater

SAMPLE-ACCURATE SYNCHRONIZATION OF SMPTE ST 2110 AUDIO STREAMS

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

Detailed explanation of the synchronization fundamentals of ST 2110 and how these can be applied to achieve sample-accurate synchronization among audio streams.

Software@AES
Wednesday, October 17, 2:30 pm – 3:00 pm
Software Pavilion

RELAB

Session EB1 **Wednesday, Oct. 17**
3:00 pm – 4:30 pm **Poster Area**

POSTERS—SPATIAL AUDIO

3:00 pm

EB1-1 A Head-Related Transfer Function Database Consolidation Tool for High Variance Machine Learning Algorithms—*Benjamin Tsui, Gavin Kearney*, University of York, York, UK

Binaural based machine learning applications generally require a large number of HRTF (Head-Related Transfer Function) measurements. However, building an HRTF database from measurements of a large number of participants can be a time-consuming and tedious process. An alternative method is to combine the data from different existing databases to create a large training dataset. This is a significant challenge due to the large difference in measurement angles, filter size, normalization schemes, and sample rates inherent in different databases. Consequently, training of some machine learning algorithms can be cumbersome, requiring significant trial and error with different data and settings. To facilitate convenient preparation of datasets, this paper presents a Matlab-based tool that allows researchers to prepare and consolidate various HRTF datasets across different databases in a robust and fast manner. The tool is available online: https://github.com/Benjamin-Tsui/HRTF_preprocessing
Engineering Brief 451

3:00 pm

EB1-2 SoundFields: A Mixed Reality Spatial Audio Game for Children with Autism Spectrum Disorder—*Daniel Johnston, Hauke Egermann, Gavin Kearney*, University of York, York, UK

SoundFields is an interactive mixed reality experience developed for children with Autism Spectrum Disorders (ASD). The project aims to provide a technical intervention framework that has the potential to promote the improvement of joint attention, social interaction, and cognitive development through full-body interaction with virtual spatitalized auditory events. The SoundFields system is based in a 360-degree visual environment in which players can move freely around without the need for head mounted displays. By means of optical motion tracking, 3rd order ambisonic audio is transmitted wirelessly to headphones, reacting to head rotation and their position within the physical space.
Engineering Brief 452

3:00 pm

EB1-3 Studio for Immersive Media Research and Production: Immersive Audio Lab at HAW Hamburg— *Philipp Kessling, Thomas Görne*, Hamburg University of Applied Sciences, Hamburg, Germany

Spatial audio is becoming increasingly important in media production since the availability of adequate distribution channels and budget hardware for production and consumer side playback is increasing likewise. To not only provide a studio for the production of spatial audio content, but also accommodate research on immersive media, a novel facility has been implemented at Hamburg University of Applied Sciences (HAW). The "Immersive Audio Lab" comprises a 33.2 High Density Loudspeaker Array (HDLA) suitable for a diverse set of spatial audio coding formats including HOA, complemented with VR technology and a broadband tracking system.
Engineering Brief 453

3:00 pm

EB1-4 Evaluation of Binaural Renderers in Virtual Reality Environments: Platform and Examples—*Thomas Robotham, Olli Rummukainen, Jürgen Herre, Emanuël A. P. Habets*, International Audio Laboratories Erlangen, Erlangen, Germany

One of the challenges of virtual reality technology is to provide convincing sensory information to users, to give the illusion of presence within the virtual environment. Audio-visual input combined with self-motion is a step beyond traditional cinematic content, whereby the audio renderer must accommodate a limitless number of potential user interactions and movements within an acoustic ?eld. In this e-Brief a framework for an online (real-time) 6 degrees-of-freedom evaluation platform is detailed. The platform allows psychoacoustic research and subjective testing of binaural audio renderers for virtual reality applications and ?nds application in the development of the MPEG-I Audio Standard.
Engineering Brief 454

3:00 pm

EB01-5 Rapid HRTF Measurement in a Loudspeaker Dome— *Noé Philip Chevalier,*[1] *Piotr Majdak,*[2] *Eva Wilk,*[1] *Thomas Görne*[1]

[1] Hamburg University of Applied Sciences, Hamburg, Germany
[2] Austrian Academy of Sciences, Vienna, Austria

Spatial audio implementations with binaural playback benefit from personalized HRTF sets. Thus access to an efficient procedure for capturing individual Head Related Transfer Functions (HRTF) is beneficial for media production as well as for research and development in the ?eld. In the newly established Immersive Audio Lab at Hamburg University of Applied Sciences we implemented a fast HRTF measurement procedure in a 33-channel loudspeaker dome, utilizing the Multiple Exponential Sweep Method (MESM) introduced by Majdak, Balazs, and Laback [1]. One measurement of about 4 minutes results in a set of 289 discrete HRIRs, covering 360° in the horizontal plane and roughly -15°...90° elevation.
Engineering Brief 455

3:00 pm

EB1-6 Survey of Media Device Ownership, Media Service Usage, and Group Media Consumption in UK Households—*Craig Cieciura*,[1] *Russell Mason*,[1] *Philip Coleman*,[1] *Matthew Paradis*[2]
[1] University of Surrey, Guildford, Surrey, UK
[2] BBC Research and Development, Salford, UK

Homes contain a plethora of devices for audio-visual content consumption, which intelligent reproduction systems can exploit to give the best possible experience. To investigate media device ownership in the home, media service-types usage and solitary versus group audio/audio-visual media consumption, a survey of UK households with 1102 respondents was undertaken. The results suggest that there is already significant ownership of wireless and smart loudspeakers, as well as other interconnected devices containing loudspeakers such as smartphones and tablets. Questions on group media consumption suggest that the majority of listeners spend more time consuming media with others than alone, demonstrating an opportunity for systems that can adapt to varying audience requirements within the same environment.
Engineering Brief 456

3:00 pm

EB1-7 A Perceptual Spectral Difference Model for Binaural Signals—*Calum Armstrong, Thomas McKenzie, Damian Murphy, Gavin Kearney*, University of York, York, UK

This paper presents a perception based model for calculating the difference between two binaural signals to more accurately represent the perceptual relevance of spectral differences. A basic spectral difference calculation, the difference between the fast Fourier transforms (FFTs) of two audio signals, is not an accurate metric for human perception as the auditory system differs greatly in sensitivity depending on relative amplitude, frequency, and temporal aspects. The presented model is evaluated through objective measures and comparison to the results of a previously published listening test.
Engineering Brief 457

Technical Committee Meeting
Wednesday, October 17, 3:00 pm – 4:00 pm
Room 1B05

NETWORK AUDIO SYSTEMS

Mix with the Masters Workshop
Wednesday, October 17, 3:00 pm – 4:00 pm
Booth 458

TCHAD BLAKE

AoIP Pavilion
Wednesday, October 17, 3:00 pm – 3:30 pm
AoIP Pavilion Theater

MONITORING AUDIO STREAMS IN THE IP NETWORK-BASED WORKFLOW

Presenter: **Aki Mäkivirta**, Genelec Oy, Iisalmi, Finland

In this presentation,Aki will explain why the entire studio audio signal paths are now being networked, how IP-connectable monitoring loudspeakers are being used across the broadcast industry to directly monitor IP audio streams, and how installed audio applications can also benefit from this technology.

Live Sound Events
Wednesday, October 17, 3:00 pm – 3:45 pm
Live Production Stage

LS04 - PRODUCTION FOR BROADCASTING THE GRAMMYS —THE TALENT BEHIND THE SCENES

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

Project Studio Expo
Wednesday, October 17, 3:00 pm – 3:45 pm
PSE Stage

PSE04 - MICROPHONE MODELING

Presenter: **Donald Spacht, II**, Antelope Audio, New York, NY, USA

Antelope Audio introduces you to the world of microphone modeling technology, featuring our custom microphones and both native and FPGA-powered processing. Antelope's Edge unlocks massive potential to achieve creative stereo/surround recording techniques with both classic and modern tonality. Emulate the sound of the past or craft the sound of the future with Edge!

Sponsored by Antelope Audio

AoIP Pavilion
Wednesday, October 17, 3:00 pm – 3:30 pm
AoIP Pavilion Theater

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Software@AES
Wednesday, October 17, 3:00 pm – 3:30 pm
Software Pavilion

BEST SERVICE

Product Development 4 **Wednesday, October 17**
3:30 pm – 5:45 pm **Room 1E09**

PATENTS AND OTHER KEY LEGAL TOPICS

Presenters: **Linda Cole**
Phillipp Lengeling, RafterMarsh
Thomas Millikan, Perkins Coie LLP, San Diego, CA, USA
Heather Rafter, RafterMarsh
Mark Rifkin
John Strawn, S Systems Inc., Larkspur, CA, USA

This Legal Topics Supersession is two sessions in one covering the legal world of product development.

The first 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.

Our second session will provide an overview of legal considerations associated with product development and launch, including IP protection and licensing, certification and branding, privacy regulations (including GDPR), and sales, both direct and through resellers. The diverse panel of legal experts welcomes an interactive discussion around software and hardware development, data collection and storage, and industry practice and pitfalls.

AoIP Pavilion
Wednesday, October 17, 3:30 pm – 4:00 pm
AoIP Pavilion Theater

ANEMAN: KEEPING AUDIO NETWORKS UNDER CONTROL

Presenter: **Dominique Brulhart**, Merging Technologies, Puidoux, Switzerland

With the raising and ubiquitous adoption of AES67, audio networks are rapidly becoming more open but as a consequence more and more heterogenic. The new challenge is to keep these networks under control and offer tools allowing managing them as easily as proprietary networks.

Software@AES
Wednesday, October 17, 3:30 pm – 4:00 pm
Software Pavilion

FL STUDIO

Recording & Productionl 3 **Wednesday, October 17**
3:45 pm – 4:45 pm **Room 1E21**

METADATA—GETTING FOUND, GETTING PAID

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

Metadata used to be hurried notes on tape boxes. Now it is the way music is found, celebrated, and paid for. It may not be as sexy as high resolution or surround mixing—but it is arguably just as important. For each of us in music, it is part of our present and our future and it needs to be done right.

Paul Jessop has been working on standards in this area for decades and will present an overview of the practical steps that audio sector professionals need to take. He will cover ISRC, ISWC, ISNI, IPI, DDEX, RIN, and indeed other terms found in the left-over alphabet soup bowl.

Student Events/Career Developments
EC4: OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Wednesday, October 17, 3:45 pm – 5:15 pm
Room 1E13

Moderator: **Kyle P. Snyder**, Ohio University, School of Media Arts & Studies, Athens, OH, USA

Presenters: *Justin Chervony*, McGill University, Montreal, Quebec, Canada
Bartlomiej Chojnacki, AGH University of Science and Technology, Cracow, Poland; Mega-Acoustic, Kepno, Poland
Brecht De Man, Birmingham City University, Birmingham, UK
Mitchell Graham, University of Michigan, Ann Arbor, MI, USA
Nyssim Lefford, Luleå University of Technology, Luleå, Sweden
Maryam Safi, Hamburg, Germany
Rob Toulson, University of Westminster, London, UK

The first Student Delegate Assembly (SDA) meeting is the official opening of the Convention's student program which includes the SDA Keynote featuring Rob Toulson and is 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 Saturday, October 20.

Audio Builders Workshop 1 **Wednesday, October 17**
4:00 pm – 6:00 pm **Crystal Palace**

CUSTOM GEAR SHOW

Presenters: **Eddie Ciletti**, Manhattan Sound Technicians, Inc.
Owen Curtin, Audio Builders Workshop,

Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA
Bob Katz, Digital Domain Mastering, Orlando, FL, USA
Joe Vezzetti
Ethan Winer, Real Traps

There is room for improvement and customization in consumer and professional gear from the past or today. We will review some popular projects and modifications. Come see what you can do and meet other builders who will display their projects and answer your questions. Audio Builder Workshop is a workgroup of the Boston AES and is hosting 7 events at the 145th Convention.

Standards Committee Meeting
SC-02-12-N TASK GROUP ON MEDIA NETWORK DIRECTORY ARCHITECTURE
Wednesday, October 17, 4:00 pm – 6:00 pm
Room 1B03

This task group is responsible for AES standards and standards-related activities for media network directories. Its scope includes media network device registration, media network device discovery, network membership administration and security activities, and the data models that support them. The group's current activity is AES project X238, a survey that aims to collect and document directory system requirements across the range of professional audio and video applications. The result of X238 will be a report to inform future directory standards activities.

Technical Committee Meeting
Wednesday, October 17, 4:00 pm – 5:00 pm
Room 1B05

PERCEPTION AND SUBJECTIVE EVALUATION OF AUDIO SIGNALS

Mix with the Masters Workshop
Wednesday, October 17, 4:00 pm – 5:00 pm
Booth 458

AL SCHMITT

AoIP Pavilion
Wednesday, October 17, 4:00 pm – 4:30 pm
AoIP Pavilion Theater

NMOS: THE KEY TO WIDE ADOPTION OF IP INFRASTRUCTURES

Presenter: **Rick Seegull**, Riedel Communications, Burbank, CA, USA

This session will explain the differences between NMOS specifications IS-04, IS-05 and IS-06. It will also provide a behind the scenes look into IS-04 and IS-05.

Live Sound Events
Wednesday, October 17, 4:00 pm – 4:45 pm
Live Production Stage

LS05 –CONSIDERING THE IMPLICATIONS OF ST-2110 ON AUDIO OVER IP

Presenter: **Stephen Brownsill**, Audio Product Manager, TSL Products

While Audio Over IP is not new to Broadcast, questions are now forming around what it means in light of SMPTE's ratification of ST-2110. What are the key differences between proprietary and industry standards? and what will need to be taken in to account if / when considering a COTS infrastructure? In this session, Stephen Brownsill will provide recap on the advantages of IP infrastructures and breakdown what we already know about Audio Over IP, whilst citing real-life applications to anchor this topic in context.

Project Studio Expo
Wednesday, October 17, 4:00 pm – 4:45 pm
PSE Stage

PSE05 - HI-RESOLUTION RECORD PRODUCTION – IT'S NOT ROCKET SCIENCE!

Presenter: **Leslie Ann Jones**, Recording Engineer and Producer, Director of Music Recording and Scoring, Skywalker Sound, San Rafael, CA, USA

Meyer Sound and Leslie Ann Jones (Skywalker Sound) will present "Hi-Resolution Record Production – It's Not Rocket Science!" with special guests including GRAMMY Award winning producer/engineer Chuck Ainlay.

Sponsored by Meyer Sound

Software@AES
Wednesday, October 17, 4:00 pm – 4:30 pm
Software Pavilion

ACCUSONUS

Broadcast/Online Delivery 4 **Wednesday, October 17**
4:15 pm – 5:45 pm **Room 1E07**

OTT LOUDNESS IS CRITICAL TO CONTENT PROVIDERS, DISTRIBUTORS AND AUDIENCES ALIKE

Chair: **Jim Starzynski**, ATSC

Panelists: *Robert Bleidt*, Fraunhofer USA Digital Media Technologies, San Jose, CA, USA
Scott Kramer, Netflix, Los Angeles, CA USA
Scott Norcross, Dolby Labs, San Francisco, CA USA
Sean Richardson, STARZ, Denver, CO, USA

OTT (over the top television) and OVD (online video distribution) are the two preeminent services that are the focus of this important session on loudness of streaming video content. Essential documents like The Advanced Television Systems Committee's (ATSC) A/85 provides the solution to government's concern with over the air (OTA) and cable loudness raised by the CALM Act. AES-71 takes loudness a step further, recommending OTA's successful international practices and more, to address audio of online OTT and OVD content.

This session will give background on how ATSC A85, EBU r128, Japan's ARIB TR-832 and Free TV Australia's OP-59 were the basis for creating OTT / OVD loudness specs and how an international group of audio experts developed these current streaming guidelines.

Content consumed on phones, tablets, and devices with varied fidelity even further motivate the need to optimize loudness, dynamic range, and the listening experience. The Consumer Technology Association has stepped up and supports R4WG8, their group made up of these familiar audio experts determined to complete the OTT loudness initiative by creating a standard for the audio design of these devices. Come and participate with a distinguished expert panel and learn how and why standardizing audio quality for OTT is so vital to content creators, distributors and their audiences. Review the practical business ramifications and technical framework required to implement AES-71, the loudness recommended practice of the AES standards organization, intended to optimize listening and avert any potential OTT loudness concerns. Hear about each expert panelist's unique perspective on the topic.

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

Immersive and Spatial Audio 2 **Wednesday, October 17**
4:15 pm – 5:45 pm **Room 1E08**

DELIVERING INTERACTIVE EXPERIENCES USING HOA THROUGH MPEG-H

Chair: **Patrick Flanagan**, THX Ltd., San Francisco, CA, USA

Panelists: *Stephen Barton*, Afterlight Inc.
Simon Calle, THX Ltd.
Mick Laviers, Respawn Entertainment
Aaron McLeran, Epic Games
Nils Peters, Qualcomm

A panel discussion about HOA content creation and how HOA should transform the way we produce audio for all types of media. Discussion includes DAW's for creating HOA; MPEG-H file compression for delivery of up to 6th order Ambisonics, Broadcasting tools, and possibilities; and how HOA and MPEG-H are gaining traction in the world.

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

Recording & Production 4 **Wednesday, October 17**
4:15 pm – 5:45 pm **Room 1E06**

THE WOW FACTOR

Presenter: **Jim Anderson**, Anderson Audio NY, New York, NY, USA; Clive Davis Institute of Recorded Music, New York University, New York, NY, USA

What is Wow? Who has Wow? Where is Wow? Why is Wow needed? When can I get Wow? How can I get Wow?

Over one hundred years ago, audiences experienced "Wow" listening to a singer and comparing their sound with a recording. Observers at the time found that it was "almost impossible to tell the difference" between what was live sound and what was recorded. Sixty years ago, the transition from monaural sound to stereophonic brought "realism" into listener's homes and today audiences can be immersed in sound. This talk will trace a history of how listeners have been educated and entertained to the latest sonic developments and said to themselves and each other: "Wow!"

AoIP Pavilion
Wednesday, October 17, 4:30 pm – 5:00 pm
AoIP Pavilion Theater

AOIP AND AES67—WHY SHOULD YOU CARE?

Presenter: **Ievgen Kostiukevych**, European Broadcasting Union, Le Grand-Saconnex, Genéve, Switzerland

Are you building a radio studio or designing a live venue? Are you confused by the number of audio solutions on the market? The presentation will explain the difference between audio over IP solutions and legacy audio networking solutions and why you should consider going IP.

Software@AES
Wednesday, October 17, 4:30 pm – 5:00 pm
Software Pavilion

MELODYNE

Special Event
SE3: SMALLS CHANGE—DECONSTRUCTING A RECORD WITH DEREK SMALLS
Wednesday, October 17, 4:45 pm – 5:45 pm
Room 1E015+16

Presenters: **Derek Smalls**
CJ Vanston

Legendary Bass Player Derek Smalls (Derek Smalls - Formerly of the Band Formerly known as Spinal Tap) discusses his latest solo album featuring multiple superstar guest musicians and full orchestra, with Producer/Co-Songwriter/Musician CJ Vanston.

Recording & Production 5 **Wednesday, October 17**
5:00 pm – 6:00 pm **Room 1E21**

PODCAST PRODUCTION

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

Panelists: *Mitra Kaboli*, The Heart, WNYC
Chaquita Paschal, Uncivil/Gimlet, Another Round/Buzzfeed
Haley Shaw, Gimlet

An inside look at the audio production and sound design of podcasts. The panel will feature podcast producers and engineers deconstructing clips and discussing the production process from story inception to sound design to editing and mixing.

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

Mix with the Masters Workshop
Wednesday, October 17, 5:00 pm – 6:00 pm
Booth 458

RAFA SARDINA

AoIP Pavilion
Wednesday, October 17, 5:00 pm – 5:30 pm
AoIP Pavilion Theater

AOIP, AES67, AND SMPTE 2110-30, IMPLEMENTATION IN THE REAL WORLD

Presenter: **Ken Tankel**, Linear Acoustic, Malvern, PA, USA

What are some of the benefits of AES67 and how does AES67 fit into the SMPTE ST 2110 standard? What are the practical requirements of putting an AoIP network in place that can allow equipment from different manufacturers to share audio over IP (AoIP) audio streams? What are the pitfalls and what are the benefits?

Live Sound Events
Wednesday, October 17, 5:00 pm – 5:45 pm
Live Production Stage

LS06 - MUSIC MIXING FOR BROADCAST

Presenter: **Lawrence Manchester**

Lawrence Manchester, Grammy-award winning music engineer and producer, takes a look at the special demands and challenges of mixing live music for broadcast. Lawrence will explain his process for mixing NBC's "The Tonight Show Starring Jimmy Fallon" in 5.1 surround and the crossover between his work in TV, film, and the recording studio.

Bio: Whether recording a symphony orchestra for Martin Scorsese's OSCAR-winning masterpiece, The Departed, tracking vocals with Beyoncé, engineering the Spider-Man: Turn Off The Dark album for Bono and The Edge, "Slow Jamming The News" with President Obama, mixing The Roots in 5.1 surround sound to a live TV audience of millions, or collaborating on viral videos surpassing ten billion views, Grammy-winner Lawrence Manchester's experience and versatility as a producer, mixer, and engineer keep him very busy. Lawrence served as the Supervising Music Producer for the film adaptation of Jason Robert Brown's musical The Last Five Years starring Anna Kendrick. He also recorded and mixed the cast recordings for The Hunchback of Notre Dame, A Bronx Tale, Summer: The Donna Summer Musical, and co-produced the Grammy-nominated Original Cast Album for the Broadway smash Something Rotten!. Lawrence served as the Post Audio Mixer for Adele Live in New York City and recently Lawrence recorded the film scores for The Girl On The Train and Keeping Up With The Joneses and mixed the scores for I, Tonya, Godless, and Bel Canto. Lawrence is also the music mixer for NBC's The Tonight Show Starring Jimmy Fallon and co-producer of Fallon's Grammy-winning comedy album, Blow Your Pants Off. During his nine years with Jimmy, he's mixed live performances by Ed Sheeran, Meghan Trainor, Prince, Lady Gaga, Beastie Boys, Bruce Springsteen, Paul McCartney, Sting, Metallica, Madonna, Ariana Grande, Camila Cabello, Green Day, Justin Timberlake, Blake Shelton, Public Enemy, Dave Matthews Band, Shakira, U2, Foo Fighters and The Who. Manchester is represented by Joe D'Ambrosio Management Inc.

Project Studio Expo
Wednesday, October 17, 5:00 pm – 5:45 pm
PSE Stage

PSE06 - BLUETOOTH RECORDING:
NOW PROFESSIONAL GRADE

Presenters: **Glenn Lorbecki**, Glenn Sound Inc., Seattle, WA, USA
Anthony Mattana, Hooke Audio Corp., Brooklyn, NY, USA

Learn about the advancements in Bluetooth audio recording and its potential to bring binaural audio main stream.

Sponsored by Iron Mountain

Software@AES
Wednesday, October 17, 5:00 pm – 5:30 pm
Software Pavilion

FABFILTER

AoIP Pavilion
Wednesday, October 17, 5:30 pm – 6:00 pm
AoIP Pavilion Theater

ROUTING AES67 AUDIO

Presenter: **Anthony Kuzub**, Ward-Beck Systems, Toronto, ON, Canada; AES - Vice Chair - Toronto Section

What was once a piece of masking tape and a sharpie marker is now a dynamic database running on a network; there must be a simpler way. Using a common language leveraging the work done in the already accepted RFC standards, a hugely complex audio system can be augmented with metadata.

Software@AES
Wednesday, October 17, 5:30 pm – 6:00 pm
Software Pavilion

MAGIX

Historical Event 2 **Wednesday, October 17**
6:15 pm – 8:30 pm
Dolby Theater
1350 Avenue of the Americas, Dolby 88 Screening Room
(corner 6th Ave. & W 55th St.)
doors open at 5:30 pm

RAY DOLBY AND THE TECHNICAL INNOVATIONS
OF DOLBY LABORATORIES

Presenters: **Ioan Allen**, Dolby Laboratories, Inc., San Francisco, CA, USA
Thomas Kodros, Dolby Laboratories, Inc., New York, NY, USA

This program traces over 50 years of audio developments from Ray Dolby and Dolby Laboratories and uses contemporaneous demonstration material for illustration. The content is primarily non-technical and will conclude with a demonstration of the combination of Dolby Atmos and Dolby Vision—the ultimate cinema experience of 2018.

Special thanks for Meyer Sound for Providing additional sound equipment.

This event will be at the Dolby Theater. Program will start at 6:45 pm and end at 8:30. We must be completely out of the room by 9:00 pm.

Come early (5:30 pm) for finger food hors d'oeuvres and beer/wine reception. This is a ticketed event

Student Events/Career Development
EC5: AES STUDENT PARTY
Wednesday, October 17, 7:00 pm – 9:00 pm
New York University
James L. Dolan Music Recording Studio, 6th Floor
35 W. 4th Street, Suite 1077
New York, NY

Audio Students! The AES Student Party is open to any 145th Convention participant with an ALL ACCESS STUDENT BADGE—a great opportunity to meet fellow students from around the world. Join us for a fun and exciting evening to be held at NYU's James L. Dolan Music Recording Studio, a 7,500 square foot multifunctional teaching, recording, and research space. This is one of the most technologically advanced audio teaching facilities in the United States and a great place for pizza and music. Space is limited so register in advance at the SDA Booth.

Archiving/Restoration 3 **Thursday, October 18**
9:00 am – 10:00 am **Room 1E10**

ARCHIVE THIS! HOW MASTERING ENGINEERS HAVE BECOME DEFAULT GUARDIANS OF AUDIO ASSETS

Presenters: **Michael Graves**
Andreas Meyer, Swan Studios NYC / Meyer Media LLC, New York, NY, USA
Cheryl Pawelski, Omnivore Records

Mastering engineers rarely deliver a physical product for manufacturing anymore. Once the final product is released, we may not hear from the artist or label again ... until years later when the phone rings and we hear, "Remember that album we did back in 200X? Now we want to create high resolution masters, vinyl masters, and a few remixes."

Responsibility for long term preservation of recorded assets has shifted from the record label to the expectation that someone else (often, the mastering engineer!) will take care of it. This panel explores the issues in asset management, the decision to archive or not to archive, the legal ramifications, and whether there ought to be a new, subscription-based business model for long term storage of client data.

Broadcast/Online Delivery 5 **Thursday, October 18**
9:00 am – 10:30 am **Room 1E15+16**

SPECIAL EVENT: TECHNIQUES FOR SUCCESS WITH AOIP TECHNOLOGY

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

Panelists: *Andreas Hildebrand*, ALC NetworX GmbH, Munich, Germany
Gary Kline, Broadcast Engineering and IT Consultant
Greg Shay, The Telos Alliance, Cleveland, OH, USA
Kent Terry, Audio Technology, Dolby Labs
Chris Tobin, Content Creator Solutions, IP Codecs

Audio over IP, a technology first deployed at radio stations some 15 years ago, is mature, stable, flexible, and most certainly viable for decades to come. There are now over ten-thousand radio studios constantly producing roughly a half-million IP-audio streams. Equipment manufacturers and engineers have come to understand what's critical in designing and installing studios built on AoIP infrastructure. The AES67 standard for AoIP interoperability plays a key role in IP-audio adoption.

Now, the television world is moving to an IP-centric infrastructure for both audio and video. Notably, the AES67 AoIP standard is at the center of the new SMPTE 2110-30 TV-audio specification. This panel presentation and discussion reveals techniques that radio engineers have been learning and perfecting for over a decade. Some panelists will also suggest the best practices for television engineers as AoIP is implemented alongside video-over-IP infrastructure.

This session is designed for audio and video engineers and technicians from radio, TV, and post-production facilities. IT engineers will also benefit from by learning key concepts in these real-time applications of IP-audio and IP-video.

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

Education 1 **Thursday, October 18**
9:00 am – 10:00 am **Room 1E13**

OFF-CAMPUS LEARNING BOTH DOMESTIC AND ABROAD

Presenters: **Leslie Gaston-Bird**, Mix Messiah Productions, Brighton, UK
Neal Schmitt, Capital University, Bexley, OH, USA

Music crosses many borders and boundaries, and an immersive educational short-term study abroad experience can be an exciting addition to traditional music curriculum. Music and Music Technology provide a modular and creative platform with which to explore and challenge our worldview. With experience in both domestic trips (Nashville, Chicago, St. Louis, and NYC) and international trips (London, Berlin, Northern Ireland, and Denmark), this tutorial will share tips and best practices for Coordinators and Educators interested in researching, planning and implementing credit and non-credit educational trips for students. Covered topics include; learning outcomes and assessment; researching and outreaching potential hosts; building buy-in with faculty, administration and students; budgeting and flight arrangements; pre, post-trip activities and reflections; and security, insurance, and behavior concerns.

Game Audio & XR 3 **Thursday, October 18**
9:00 am – 10:00 am **Room 1E08**

ANATOMY OF GREAT VOICE OVER: A CASTING & RECORDING PRIMER

Presenter: **Andrea Toyias**, Blizzard Entertainment, Irvine, CA, USA

Game dialogue is one of the final ingredients that breathes life into a video game. The story flows, the gameplay engages, and the characters come to life through memorable VO performances. . . thus, bringing depth and immersion to the overall experiences. Andrea Toyias will bring her 10 years as Head of the VO department at Blizzard Entertainment to give you tools, tips, and insights on how to best create the vocal performances you are after. Topics will include: how to fully flesh out characters specs when casting; what to listen for when auditioning; best ways to prep for a recording session; and how to successfully work with voice actors in order to better create, collaborate and experiment with them so you can bring your vision to life in new and exciting ways. The relationship between game team, director, and talent will be broken down, examined, and explored in its purest form.

Immersive and Spatial Audio 3 **Thursday, October 18**
9:00 am – 10:00 am **Room 1E17**

WORKFLOWS AND TECHNIQUES FOR AMBISONIC RECORDING AND POSTPRODUCTION

Chair: **Ming-Lun Lee**, University of Rochester, Rochester, NY, USA

Panelists: *Olivia Canavan*, University of Rochester, Rochester, NY, USA
Steven Philbert, University of Rochester, Rochester, NY, USA

For a 360/3D VR video with head-tracked binaural audio, we have to use Ambisonics to capture the overall sound field. Although some high-end VR cameras have built-in Ambisonic microphones, most of them can only record first-order Ambisonics in the horizontal plane. The sound qualities are also not good in general. To achieving high-quality 3D audio, a recommended solution is to record audio with a separate Ambisonic microphone and then mix with the video during recording or post-editing. With the experience of recording Ambisonics using the em32 Eigenmike microphone array, Zylia ZM-1 microphone, Core Sound Tetramic, and Sennheiser Ambeo VR microphone for many concerts, this workshop aims to offer optimized workflows and practical techniques for Ambisonic recording, conversion, editing, rendering, playback, and publishing.

Product Development 5 **Thursday, October 18**
9:00 am – 10:15 am **Room 1E09**

MAKING YOUR PRODUCT GREAT THROUGH ADDING PRE-PRODUCT INNOVATION DEVELOPMENT

Presenters: **Scott Leslie**, PD Squared, Irvine, CA, USA
Jacques Spillmann, PD Squared, Los Angeles, CA, USA

Most products today are developed in a define, design, validate, and verify process. Because of schedules, resource availability, and traditional thinking, true innovation cannot fit into most projects. In this session the presenter will introduce the subject of pre-product innovation. This method separates innovation development from product development allowing an innovation specialist, inside or outside the organization, to create and produce concepts and pre-designs that program the direction of the product before the definition of the final product is completed or even started in some cases. Further, a method for incorporating pre-product innovation into the overall development process will be presented.

Recording & Production 6 **Thursday, October 18**
9:00 am – 10:00 am **Room 1E21**

FROM STREAMING TO VINYL (AND EVEN CASSETTE!)— UNDERSTANDING DISTRIBUTION FORMATS

Moderator: **Jonathan Wyner**, M Works Studios/iZotope/ Berklee College of Music, Boston, MA, USA; M Works Mastering

Panelists: *Margaret Luthar*, Chicago Mastering Service, Chicago, IL, USA
Dave Polster, Well Made Music, Cleveland, OH, USA
Mandy Parnell, Black Saloon Studios, London, UK

The way music gets distributed from artist to consumer is varied. In this panel we'll discuss the distribution formats that are used today; vinyl, streaming services, CD, cassette—and what are the limitations, benefits, and myths about each format. Technical specs for streaming, as well as the vinyl production process will be described as well as best practices for providing audio to mastering engineers. The hope is to de-mystify distribution and allow everyone involved in the production process be able to communicate effectively with regards to concerns, suggestions, and achieving the best result.

Technical Committee Meeting
Thursday, October 18, 9:00 pm – 10:00 am
Room 1B05

RECORDING TECHNOLOGY AND PRACTICES

Recording & Production 7 **Thursday, October 18**
9:15 am – 10:15 am **Room 1E07**

CIRCLES OF CONFUSION

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

Panelists: *Bob Ludwig*, Gateway Mastering Studios, Inc., Portland, ME, USA
Sean Olive, Harman International, Northridge, CA, USA
Floyd Toole, Acoustical consultant to Harman, ex. Harman VP Acoustical Engineering, Oak Park, CA, USA

30.000 year old cave paintings are among human beings' most impressive cultural heritage, while it's unknown how excellent composers sounded even 300 years ago. We recently acquired technical skills to record sound, but that asset is now degenerating because of cognitive limitations and the circles of confusion, addressed by this panel.

Without proper anchoring of spectral balance and level, drifting over time is inevitable in self-referenced systems, thereby putting legacy recordings at the risk of sounding dated for no good reason, or causing irreversible distortion to be added to pieces of art. The panel will discuss baseline listening requirements for in-room and headphone spectral balance and level that stand the test of time, putting our interests as a species above commercial trivialities.

Session P6 **Thursday, Oct. 18**
9:30 am – 12:00 noon **Room 1E11**

TRANSDUCERS—PART 2

Chair: **Pascal Brunet**, Samsung Research America, Valencia, CA, USA

9:30 am

P6-1 On the Interdependence of Loudspeaker Motor Nonlinearities—*Finn T. Agerkvist, Franz Heuchel*, Technical University of Denmark, Kgs. Lyngby, Denmark

Two of the main nonlinearities in the electrodynamic loudspeaker are the position dependence of the force factor, *Bl*, and the voice coil inductance, L_e. Since they both are determined by the geometry of the motor structure, they cannot be independent. This paper investigates this dependence both analytically and via FEM simulations. Under certain simplifying assumptions the force factor can be shown to be proportional to the spatial derivative of the inductance. Using FEM simulations the implications of this relation is illustrated for drivers with more realistic geometry and material parameters.
Convention Paper 10058

10:00 am

P6-2 Comparison of Dynamic Driver Current Feedback Methods—*Juha Backman*, Huawei Technologies, Tampere, Finland; Genelec Oy, Iisalmi, Finland

Current feedback is a versatile method of modifying the behavior of a loudspeaker driver, with opportunity for linearization and matching the driver to the enclosure design targets, but depending on the chosen approach a potential risk of increasing the effects of either voice coil impedance variation or driver mechanical parameter nonlinearity. This work compares using a nonlinear simulation model various forms of current feedback, including current drive, finite positive or negative amplifier resistances, and negative resistance combined with a reactance.
Convention Paper 10059

10:30 am

P6-3 Non-Linear Optimization of Sound Field Control at Low Frequencies Produced by Loudspeakers in Rooms—*Adrian Celestinos, Pascal Brunet, Glenn Kubota*, Samsung Electronics, Valencia, CA, USA

The low-frequency response of loudspeakers can be affected severely when placed in typical living rooms. Past investigations have focused on reducing the energy at the room resonances but not reducing seat-to-seat variation. Other works, using multiple loudspeakers, nullify the room modes or eliminate them with acoustic interference but are restricted by loudspeaker position and room geometry. In this work the loudspeaker position is not restricted and both frequency and time-domain methods are explored. Nonlinear optimization has been computed to improve the time response of the speaker in the room. Results have shown significant reduction of seat-to-seat variation. The time-frequency analysis reveals elimination of room resonances; producing a clear tight bass response.
Convention Paper 10060

11:00 am

P6-4 Evaluation of Efficiency and Voltage Sensitivity in Horn Drivers—*Alexander Voishvillo, Brian McLaughlin*, Harman Professional, Northridge, CA, USA

There is a belief in horn driver design that if the resistive component of the input impedance of the diaphragm's acoustical load is equal to the DC component of the voice coil's electrical impedance, a maximum efficiency of 50% can be reached. This work shows that in reality, the compression driver's real efficiency differs from the classical theory prediction. The discrepancy is explained by the fact that the driver's output impedance and the acoustical loading impedance are different functions, not mere resistances. Additionally, the input electrical power and the output acoustical power are essentially integral functions of frequency and can be expressed by single numbers rather than frequency-dependent functions. Examples illustrating dependence of efficiency and sensitivity on various parameters are given.
Convention Paper 10061

11:30 am

P6-5 Characterization of Nonlinear Port Parameters in Loudspeaker Modeling—*Doug Button,*[1] *Russ Lambert,*[2] *Pascal Brunet,*[3] *James Bunning*[1]
[1] Harman International, Northridge, CA, USA
[2] Harman International, Richardson, TX, USA
[3] Samsung Research America, Valencia, CA USA

The outputs from ports used in common vented box loudspeakers are not linear with input level. With the goal of developing accurate modeling of vented boxes, a new method for estimation of nonlinear port parameters is shown. Acoustic mass and acoustic resistance parameters change with pressure in the enclosure and velocity in the port. Along with all nonlinear speaker parameters required for modeling, a practical way to characterize the changing acoustic mass and acoustic resistance is presented and tested with measurements. Using the new method, nonlinear port parameters are extracted for multiple box and port types.
Convention Paper 10062
[Paper will be presented by Pascal Brunet]

Session P7 **Thursday, Oct. 18**
9:30 am – 12:00 noon **Room 1E12**

PERCEPTION—PART 2

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

9:30 am

P7-1 Reproducing Low Frequency Spaciousness and Envelopment in Listening Rooms—*David Griesinger*, David Griesinger Acoustics, Cambridge, MA, USA

Envelopment—the perception of being surrounded by a beautiful acoustic space—is one of the joys of concert halls and great recordings. Recording engineers strive to capture reverberation independently in each channel, and reverberation from digital equipment is similarly non-coherent. But in playback rooms with a single low frequency driver or a single subwoofer reverberation is often flat, frontal, and without motion. In this paper we will show that full range loudspeakers or at least two independent subwoofers can bring low frequency envelopment back to a playback room. In some rooms minimizing room modes with high pressure at the listening position while maximizing lateral modes with minima near the listeners can help. If necessary, we put two independent subwoofers at the sides of the listeners.
Convention Paper 10063

10:00 am

P7-2 Evaluation of Implementations of the EBU R128 Loudness Measurement—*Brecht De Man*, Birmingham City University, Birmingham, UK

The EBU R128 / ITU-R BS.1770 specifications are widely followed in various areas of the audio industry and academic research. Different implementations exist in different programming languages with subtly differing design parameters. In this work we assess these implementations on their performance in terms of accuracy and functionality. As filter equations are notably absent from the standard, we reverse engineer the prescribed filter coefficients and offer more universal filter specifications. We also provide a simple implementation of the integrated loudness measure in JavaScript, MATLAB, and Python.
Convention Paper 10064

10:30 am

P7-3 Hyper-Compression in Music Production: Testing the "Louder Is Better" Paradigm—*Robert Taylor*, University of Newcastle, Callaghan, NSW, Australia

Within the scope of the literature surrounding the Loudness War, the "louder is better" paradigm plays a cornerstone role in the motivation and also presumed justification for the continued use of hyper-compression. At the core of this assumption is the non-linearity of frequency response of the human auditory system first identified by Fletcher and Munson [11]. Previous research into listener preferences concerning hyper-compression has attempted to rationalize this production practice with audience expectations. The stimuli used in these studies have invariably been loudness normalized to remove loudness bias in audition so that only the perceptual cues of dynamic range compression (DRC) are under examination. The results of these studies have proven less than conclusive and varied. The research study presented herein examines the extent of influence the "louder is better" paradigm has

on listener preferences via a direct comparison between listener preference tasks that present music that is loudness normalized and music that retains the level differentiation which is a by-product of the hyper-compression process. It was found that a level differential of 10 dB had a significant influence of listener preferences as opposed to the arguably weak perceptual cues of DRC.
Convention Paper 10065

11:00 am

P7-4 The Effect of Pinnae Cues on Lead-Signal Localization in Elevated, Lowered, and Diagonal Loudspeaker Configurations—*Wesley A. Bulla,*[1] *Paul Mayo*[2]
[1] Belmont University, Nashville, TN, USA
[2] University of Maryland, College Park, MD, USA

In a follow-up to AES-143 [convention paper] #9832, this experiment employed a novel method that altered subjects' pinna and examined the effects of modifying salient spectral cues on time-based vertical-oriented precedence in raised, lowered, and diagonal sagittal and medial planes. As suggested in the prior study, outcomes confirm perceptual interference from acoustic patterns generated via lead-lag signal interaction. Results provide clear physical and psychophysical evidence that reliable elevation cues may be rendered ineffective by stimuli such as those used in typical precedence-based experiments. Outcomes here demonstrate the salient and powerful influence of spectral information during lead-lag precedence-oriented tasks and suggest that prior research, in particular that concerned with so-called "vertical" precedence, may have been erroneously influenced by simple—yet profound—acoustic comb-filtering.
Convention Paper 10066

11:30 am

P7-5 Perceptual Audio Evaluation of Media Device Orchestration Using the Multi-Stimulus Ideal Profile Method—
Alex Wilson,[1] *Trevor Cox,*[1] *Nick Zacharov,*[2] *Chris Pike*[3]
[1] University of Salford, Salford, UK
[2] Force Technology, SenseLab, Hørsholm, Denmark
[3] BBC Research & Development, Salford, UK

The evaluation of object-based audio reproduction methods in a real-world context remains a challenge as it is difficult to separate the effects of the reproduction system from the effects of the audio mix rendered for that system. This is often compounded by the absence of explicitly-defined reference or anchor stimuli. This paper presents a perceptual evaluation of five spatial audio reproductions including Media Device Orchestration (MDO) in which a stereo mix is augmented by four additional loudspeakers. The systems are evaluated using the Multi-Stimulus Ideal Profile Method (MS-IPM), in which an assessor's preferred value of each perceptual attribute is recorded in addition to their ratings of the systems. Principal Component Analysis was used to gain insight into the perceptual dimensions of the stimulus set and to investigate if the real systems overlap with the ideal profile. The results indicate that perceptual envelopment is the dominating perceptual factor for this set of reproduction systems, and the ideal system is one of high-envelopment.
Convention Paper 10067

Student Event/Career Development
EC6: SAUL WALKER STUDENT DESIGN EXHIBITION
Thursday, October 18, 9:30 am – 11:00 am
Poster Area

Moderators: **Justin Chervony**, McGill University, Montreal, Quebec, Canada
Bartlomiej Chojnacki, AGH University of Science and Technology, Cracow, Poland; Mega-Acoustic, Kepno, Poland
Mitchell Graham, University of Michigan, Ann Arbor, MI, USA
Maryam Safi, Hamburg, Germany

All accepted entries to the AES Saul Walker 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.

PMC Masters of Audio Program
Thursday, October 18, 9:30 am – 10:30 am
Room 1E06

UMG/CAPITOL STUDIOS DOLBY ATMOS PLAYBACK

Presenter: **Steve Genewick**

UMG/Capitol Studios Dolby Atmos Playback sessions with Steve Genewick. Featuring music mixed for Dolby Atmos from Elton John, LL Cool J, Chris Walden, INXS, REM, Public Enemy, Bastille, Arturo Sandoval, Snoh Aleegra and many others.

Session P8 — Thursday, Oct. 18
10:00 am – 11:30 am — Poster Area

POSTERS: ACOUSTICS AND SIGNAL PROCESSING

10:00 am

P8-1 Estimation of MVDR Beamforming Weights Based on Deep Neural Network—*Moon Ju Jo,*[1] *Geon Woo Lee,*[1] *Jung Min Moon,*[1] *Choongsang Cho,*[2] *Hong Kook Kim*[1]
[1] Gwangju Institute of Science and Technology (GIST), Gwangju, Korea
[2] Artificial Intelligence Research Center, Korea Electronics Technology Institute (KETI), Sungnam, Korea

In this paper we propose a deep learning-based MVDR beamforming weight estimation method. The MVDR beamforming weight can be estimated based on deep learning using GCC-PHAT without the information on the source location, while the MVDR beamforming weight requires information on the source location. As a result of an experiment with REVERB challenge data, the root mean square error between the estimated weight and the MVDR weight was found to be 0.32.
Convention Paper 10068

10:00 am

P8-2 A Statistical Metric for Stability in Instrumental Vibrato—*Sarah R. Smith, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

When instrumentalists perform with vibrato, they add a quasi-periodic frequency modulation to the note. Although this modulation is rarely purely sinusoidal, many methods for vibrato parameterization focus exclusively on the rate and depth of the frequency modulation, with less attention given to measuring how a performer's vibrato changes over the course of a note. In this paper we interpret the vibrato trajectories as instantiations of a random process that can be characterized by an associated autocorrelation function and power spectral density. From these distributions, a coherence time can be estimated that describes the stability of the vibrato within a note. This metric can be used to characterize individual performers as well as for resynthesizing vibratos of different styles.
Convention Paper 10069

10:00 am

P8-3 Machine Learning Applied to Aspirated and Non-Aspirated Allophone Classification–An Approach Based on Audio "Fingerprinting"—*Magdalena Piotrowska,*[1] *Grazina Korvel,*[2] *Adam Kurowski,*[1] *Bozena Kostek*,[1] *Andrzej Czyzewski*[1]
[1] Gdansk University of Technology, Gdansk, Poland
[2] Vilnius University, Vilnius, Lithuania

The purpose of this study is to involve both Convolutional Neural Networks and a typical learning algorithm in the allophone classification process. A list of words including aspirated and non-aspirated allophones pronounced by native and non-native English speakers is recorded and then edited and analyzed. Allophones extracted from English speakers' recordings are presented in the form of two-dimensional spectrogram images and used as input to train the Convolutional Neural Networks. Various settings of the spectral representation are analyzed to determine adequate option for the allophone classification. Then, testing is performed on the basis of non-native speakers' utterances. The same approach is repeated employing learning algorithm but based on feature vectors. The archived classification results are promising as high accuracy is observed.
Convention Paper 10070

10:00 am

P8-4 Combining the Signals of Sound Sources Separated from Different Nodes after a Pairing Process Using an STFT-Based GCC-PHAT—*César Clares-Crespo, Joaquín García-Gómez, Alfredo Fernández-Toloba, Roberto Gil-Pita, Manuel Rosa-Zurera, Manuel Utrilla-Manso*, University of Alcalá, Alcalá de Henares, Madrid, Spain

This paper presents a Blind Source Separation (BSS) algorithm proposed to work into a multi-node network. It is intended to be used indoors as an improvement to overcome some limitations of classical BSS algorithms. The goal is to improve the quality of the speech sources combining the signals of the same source separated from the different nodes located in the room. The algorithm matches the signals belonging to the same source through cross-correlations. It provides some stability to the quality of the separated sources since it takes advantage of the fact that the sources are likely to be correctly separated in some nodes.
Convention Paper 10071

10:00 am

P8-5 Reverberation Modeled as a Random Ensemble of Images—*Stephen McGovern*, Wire Grind Audio, Sunnyvale, CA, USA

Modeling box-shaped rooms with the image method requires many mathematical operations. The resulting reverberation effect is qualitatively flawed due to sweeping echoes and flutter. Efficiency is improved with the Fast Image Method, and sweeping echoes can be suppressed by using randomization. With both approaches, however, there is still a remaining problem with flutter. To address all of these issues, the Fast Image Method is modified to have both randomized image locations and increased symmetry. Additional optimizations are proposed and applied to the algorithm. The resulting audio effect has improved quality, and the computation time is dramatically reduced (by a factor usually exceeding 200) when compared to its ancestral Allen and Berkley algorithm. Some relevant perceptual considerations are also discussed.
Convention Paper 10072

10:00 am

P8-6 On the Accuracy of Audience Implementations in Acoustic Computer Modelling—*Ross Hammond,*[1] *Adam J. Hill*,[1] *Peter Mapp*[2]
[1] University of Derby, Derby, Derbyshire, UK
[2] Peter Mapp Associates, Colchester, Essex, UK

Performance venue acoustics differ significantly due to audience size, largely from the change in absorption and reflection pathways. Creating acoustic models that accurately mimic these changes is problematic, showing significant variance between audience implementation methods and modelling techniques. Changes in total absorption per person due to audience size and density makes absorption coefficients selection difficult. In this research, FDTD simulations confirm that for densely packed audiences, diffraction leads to a linear correlation between capacity and total absorption at low frequencies, while at high frequencies there is less increase in total absorption per person. The significance of diffraction renders ray-tracing inaccurate for individually modelled audience members and has further implications regarding accuracy of standard audience modelling procedures.
Convention Paper 10073

10:00 am

P8-7 High Resolution Horizontal Arrays for Sound Stages and Room Acoustics: Concepts and Benefits—*Cornelius Bradter*,[1,2] *Kichel Ju*[1]
[1] Shinan Information & Communication Co., Ltd., Gyeonggi-do, Korea
[2] State University of New York Korea, Incheon, Korea

Wavefield-synthesis systems were the first to apply large horizontal arrays for sound reproduction and live sound. While there has been some success for experimental music or sound art, results for music and live sound are limited. The reasons for this are array sound quality, insufficient sound pressure level, spatial aliasing, and interference and inflexible array control algorithm. Nevertheless, the application of high resolution horizontal arrays with appropriate array control systems produces exceptional sound quality, extensive flexibility to adjust to acoustic conditions, and remarkable control over sound distribution. Advanced systems can overcome many shortcomings in traditional sound systems. This paper describes the con-

cepts and technology of high performance HRH-arrays and their utilization for church sound through examples of application in Korean churches.
Convention Paper 10074

Standards Committee Meeting
SC-02-01 WORKING GROUP ON DIGITAL AUDIO MEASUREMENT TECHNIQUES
Thursday, October 18, 10:00 am – 11:00 am
Room 1B03

The scope of SC-02-01 includes measurement methods for equipment used for the recording, reproduction, and transmission of digital audio signals for professional recording and broadcasting. It includes effects of perceptually based coding algorithms on audio signals for professional recording and broadcasting. It inclu des psychophysical and electrical analysis under all operational and listening conditions. It includes ranking of codecs and test methods to determine presence of coders or their proper operation.

Technical Committee Meeting
Thursday, October 18, 10:00 pm – 11:00 am
Room 1B05

MICROPHONES AND APPLICATIONS

Audio for Cinema 1 **Thursday, October 18**
10:15 am – 11:15 am **Room 1E10**

BEST PRACTICES IN RE-RECORDING MIXING: FOCUS ON EPISODIC

Presenter: **Tom Fleischman**, Soundtrack Film & Television, New York, NY, USA

This session on re-recording will reprise some of Tom Fleischman's master class from AES143 but with an added focus on the particular challenges associated with episodic television and series. Tom will discuss how he approaches a project from the beginning through to the end, on how the mix can enhance storytelling, the importance of clarity of dialogue, and how music and sound effects engage the audience. Also covered will be the distinctions between cinema and episodic productions, dealing with continuity between episodes, changing creative hierarchies, and the blinding speed at which mixes are made for series.

Mix with the Masters Workshop
Thursday, October 18, 10:00 am – 11:00 am
Booth 458

JOE CHICCARELLI

Game Audio & XR 4 **Thursday, October 18**
10:15 am – 11:15 am **Room 1E08**

GAMES V. CINEMA: GRUDGE MATCH

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

Panelists: *Lydia Andrews*, Ubisoft Montreal, Quebec City, Montreal, Canada
Jason Kanter, Avalanche Studios
Harold Kilianski, Fanshaew College, Ontario, Canada
John Whynot, Berklee College of Music, Boston, MA, USA

Game audio and audio for cinema. Two worlds that create epic soundscapes. How much are they similar or different? Do they share the same tools, design plan and challenges, or are they completely individual? Join this session to hear from four leaders in the industry, two from cinema and two from games, as they discuss audio design for their fields. Learn how sound design, dialog and FX strategies differs between the two realms, and how they sometimes even work together.

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

Immersive and Spatial Audio 4 **Thursday, October 18**
10:15 am – 11:15 am **Room 1E17**

STAAG IMPLEMENTATION EXPANDED: A PRACTICAL DEMO OF STAAG MIC TECHNIQUE AND ITS USE IN STEREO, SURROUND, AND HEIGHT-CHANNEL CAPTURE FOR RECORDING AND BROADCAST

Presenters: **David Angell**, Central Sound at Arizona PBS, Phoenix, AZ, USA
Jamie Tagg, Indiana University, Bloomington, IN, USA; Stagg Sound Services, LLC

While working on location, audio engineers are often challenged by insufficient monitoring, mak-ing choices which lead to timbral, wet/dry balance, and stereo image problems. This tutorial examines the use of STAAG (Stereo Technique for Augmented Ambience Gradient) and its established advantages for addressing stereo image, acoustic realism, and flexibility in the mix. While original-ly optimized for immersive headphone listening, this technique has proven advantageous when up-scaling to stereo, surround, and even height-channel loudspeaker systems and stereo/surround broadcast streams. Also, this setup is advantageous when working on location and in live perfor-mance scenarios due to its compact arrangement and compatibility with a wide variety of micro-phones. This tutorial will feature physical configurations and playback of audio examples in a vari-ety of formats.

Recording & Production 8 **Thursday, October 18**
10:15 am – 11:15 am **Room 1E21**

SPACE, PLACE, AND BASS: PROVIDING MODERN METAL MUSIC WITH AN APPROPRIATE BALANCE BETWEEN HEAVINESS AND CLARITY

Presenters: **Steven Fenton**, University of Huddersfield, Huddersfield, West Yorkshire, UK
Mark Mynett, University of Huddersfield, Huddersfield, UK

Distinct challenges are posed when providing the various sounds/performances of a modern metal mix with appropriate space, place, and bass. This is especially the case when down-tuning is combined with fast performance subdivisions and ensemble rhythmic synchronization.

This workshop covers intermediate-to-advanced "space, place and bass" mix principles that afford this production style an appropriate balance between heaviness and clarity, including: frequency bracketed kick and bass approaches; anti-masking mix theory (with a focus on different "designs"); dynamic EQ and multi-band compression use; series and parallel dynamics approaches; and time-based processing principles.

Mark Mynett, who lectures in Music Technology and Production at Huddersfield University, is a record producer and author of *Metal Music Manual*, the world's first book on producing/engineering/mixing and mastering contemporary heavy music.

Sound Reinforcement 4 **Thursday, October 18**
10:15 am – 11:45 am **Room 1E13**

DESIGNING FOR BROADWAY: THE BAND'S VISIT

Moderator: **Kai Harada**

Among the many Tony Awards won by "The Band's Visit" was the Tony for Best Sound for a Musical. Kai Harada and his team will discuss the process of bringing this production and others to the stage.

Product Development 6 **Thursday, October 18**
10:30 am – 11:45 am **Room 1E09**

LOUDSPEAKER LIFE TESTING AND POWER AMPLIFIER REQUIREMENTS

Chair: **Steven Hutt**, Equity Sound Investments, Bloomington, IN, USA

Panelists: **John Dawson**, Jade Electronics Ltd., Cambridge, UK
Richard Little, Goertek USA, Fremont, CA, USA

Loudspeaker life testing is a critical aspect of characterizing the capacity of loudspeaker (drivers/systems) to accept input voltage. The AES2 (1984/2012) standard defines an accelerated two hour test to define a safe input level from which a device under test (DUT) can recover without permanent alteration. AES2-2012 adopted use of pink noise with a 12 dB Crest Factor that has potential to demand extreme peak capability from power amplifiers. This discussion will address the challenge of updating the life test section of AES2 with incite from industry experts. Discussion may include pre/post measurements, test duration, Crest Factor, source signal, duty cycle, mechanical vs. thermal stress, maximum useful acoustic output, power amplifier requirements, and the challenge of assigning a frequency dependent power rating to loudspeakers.

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

AoIP Pavilion
Thursday, October 18, 10:30 am – 11:00 am
AoIP Pavilion Theater

INTRODUCTION TO THE AOIP TECHNOLOGY PAVILION

Presenter: **Terry Holton**, Yamaha R&D Centre, London, UK

The Audio-over-IP Technology Pavilion is a significant new initiative created by the AES in partnership with the Alliance for IP Media Solutions (AIMS). The pavilion will promote professional IP media networking as well as providing the latest information about this rapidly developing field through practical demonstrations and an extensive presentation program. This session will provide an introduction to the various aspects of the pavilion including the AIMS demonstration system, exhibitors' Pods, and the AoIP Presentation Theater.

Software@AES
Thursday, October 18, 10:30 am – 11:00 am
Software Pavilion

AUDIOSOURCERE

Broadcast/Online Delivery 6 **Thursday, October 18**
10:45 am – 12:15 pm **Room 1E07**

AUDIO OF MARCH MADNESS

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

Panelists: *Bob Baker*, Sr. Media Transport Architect, Broadcast Infrastructure & Technology, BEST Engineering
Erinn Thorp, Manager, Broadcast Infrastructure & Technology, BEST Studio Engineering

The NCAA Men's Basketball Tournament has often been described as one of the greatest sports tournaments, fue to the unpredictability of college sports wrapped in the tradition of generations of fans. Bringing this unique experience to fans across a variety of delivery platforms is a complex technical challenge. Starting with games originating from eight venues ranging from on campus sports facilities to professional sports arenas, maintaining a consistent audio experience across all of the games and platforms pushes the limits of technology.

This panel follows the entire audio signal path from originating the mix at the game venues, capturing and packaging all of the game feeds into an assortment of consumer products, and delivering that experience across a wide assortment of devices.

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

Special Event
SE4: PLATINUM LATIN PRODUCERS & ENGINEERS
Thursday, October 18, 10:45 am – 12:15 pm
Room 1E15+16

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

Presenters: *Rafael Arcaute*
Gustavo Borner, igloo music, Burbank, CA, USA
Eduardo Cabra, (Visitante)
Andres Levin, Music Has No Enemies, New York / Havana; Habanico, New York/Havana
Eduardo Pereyra, Buenos Aires, Argentina
Rafa Sardina

The Platinum Latin Panel brings every year some of the finest audio professionals. Multi-platinum record producers and GRAMMY winners, the presenters will tell you all about producing great music for the Latin market.

Standards Committee Meeting
SC-02-12-R TASK GROUP ON STREAMING METADATA
Thursday, October 18, 11:00 am – 1:00 pm
Room 1B03

This group will define a standardized method for transporting metadata associated with audio in an AES67 stream in a separate parallel stream. It shall define synchronization between the audio metadata transport and the associated AES67 transport. The transmission method shall be low latency and have a level of network performance equivalent to AES67. Within the scope is formatting of the streaming audio metadata for transport. Suggested is an open standards based framework that supports both static and dynamic, time synchronous metadata that is optimized for live workflow applications. The standard shall consider all use cases for metadata associated with AES67, support existing AES audio metadata standards, and be extensible for future metadata requirements. The standard will consider binding between the audio metadata transport and the associated AES67 transport.

Technical Committee Meeting
Thursday, October 18, 11:00 pm – 12:00 noon
Room 1B05

ARCHIVING RESTORATION AND DIGITAL LIBRARIES

Mix with the Masters Workshop
Thursday, October 18, 11:00 am – 12:00 noon
Booth 458

JACK JOSEPH PUIG

AoIP Pavilion
Thursday, October 18, 11:00 am – 11:30 am
AoIP Pavilion Theater

AES67-2018 PICS: A BASIS FOR INTEROPERABILITY ASSESSMENT

Presenter: **Gints Linis**, Telos Alliance, Riga, Latvia

AES67-2018 has a new Annex G - Protocol implementation conformance criteria. The presentation will provide background information on what PICS is about as well as a brief history and the current status of the AES67 conformance criteria work, and explain how it can help equipment manufacturers and system integrators. Structure of the provided PICS proforma will be discussed and hints for completing it will be provided. A closer look at AES67 clauses will reveal examples of potential ambiguities and demonstrate how the provided conformance criteria help to resolve them.

Live Sound Events
Thursday, October 18, 11:00 am – 11:45 am
Live Production Stage

LS07 - WIRELESS ISSUES FOR LIVE THEATER: BROADWAY AND BEYOND

Presenter: **TBA**

PMC Masters of Audio Program
Thursday, October 18, 11:00 am – 12:15 pm
Room 1E06

SHOWMAN

Presenter: **Greg Wells**

Making of *The Greatest Showman* in Dolby Atmos 7.1.4 by Greg Wells.

Project Studio Expo
Thursday, October 18, 11:00 am – 11:45 am
PSE Stage

PSE07 - ART OF MONITORING

Presenter: **Richard Chycki**

Renowned mix engineer/producer Richard Chycki talks about his approach to creating a mix: revealing what you should listen for, what mistakes you should avoid, and why good studio monitoring is essential for helping your mixes translate beautifully to other systems. Not to be missed!

Sponsored by Genelec

Software@AES
Thursday, October 18, 11:00 am – 11:30 am
Software Pavilion

MELODYNE

Audio for Cinema 2 **Thursday, October 18**
11:30 am – 12:30 pm **Room 1E10**

COLLABORATION AT A DISTANCE: REAL-TIME REMOTE RECORDING TOOLS FOR SCORING AND POST AUDIO

Presenter: **Robert Marshall**, Source Elements

Market forces and the advent of new technology and the internet have greatly altered the world of film scoring, along with other related enterprises in audio post. Excellent tools exist for collaborating remotely on recording sessions. Workflows have built up and developed over time to allow those constrained by budget, or desiring the artistic services of someone who is geographically remote, to conduct full sessions with talkback and good monitor audio. This workshop is an exploration of one of the major tools, along with examples of how remote recording is being used in scoring and other disciplines.

Game Audio & XR 5 **Thursday, October 18**
11:30 am – 12:30 pm **Room 1E17**

A SYSTEMIC APPROACH TO INTERACTIVE DIALOGUES ON ASSASSINS CREED ODYSSEY—FROM SPEECH TO SFX TO MUSIC

Presenters: **Lydia Andrew**, Ubisoft, Quebec City, Canada
Greig Newby, Ubisoft, Quebec, Canada

From the beginning of Assassin's Creed Odyssey, we recognized that this rich, continually unfolding open world game demanded more than the traditional manual, minute-by-minute approach to audio design, integration and mixing. The dual protagonists, the interactive dialogues, and the massive scale meant we needed to build systems that were both responsive to the complexity of our game world and to the individuality of our players' choices.

The presentation will cover this systemic approach, showing how we created and used tools and pipelines to support our player's freedom of choice. We will talk about the complexity of constructing, recording, and integrating the voice into the interactive dialogue system, focusing on the new tools and pipelines we developed. We will show how music is used in the interactive dialogues to support character, emotion, and player choice. We will talk about how we aimed to maintain the consistency of the player experience with Foley, sfx, and ambiences by seamlessly moving in and out of the interactive dialogues. Finally, we will discuss how we brought all these elements together through systems that were the friend not the enemy of creativity.

The attendees will walk away with an understanding of the potential challenges of implementing a branching interactive dialog system in an open world game and some insights on how to transform their traditional linear pipelines.

Immersive and Spatial Audio 5 **Thursday, October 18**
11:30 am – 12:30 pm **Room 1E08**

MEASURING HEAD RELATED TRANSFER FUNCTIONS: PRACTICALITIES, PROCESSING AND APPLICATIONS

Presenters: **Calum Armstrong**, University of York, York, UK
Gavin Kearney, University of York, York, UK

At the heart of good audio spatial audio reproduction over headphones is the measurement of high quality binaural filters, commonly known as head related transfer functions (HRTFs). This workshop explores the practical challenges involved in the measurement of datasets of HRTFs and the subsequent signal processing required for high quality binaural rendering. We discuss measurement techniques, microphone choices, subject considerations and equalization strategies. We also explore anechoic binaural measurements vs. binaural room impulse responses as well as spatial sampling considerations for applications such as Ambisonic rendering for virtual and augmented reality. Finally we look at what makes a good quality HRTF set—is it the resultant timbre, the sense of externalization or other factors that make people prefer one HRTF dataset over another?

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

Recording & Production 9 **Thursday, October 18**
11:30 am – 12:30 pm **Room 1E21**

CREATIVE WAYS TO USE AN ANALOG TAPE MACHINE WITH A DAW

Presenter: **Chris Mara**, Mara Machines, Nashville, TN, USA

Fast and easy ways to use an analog tape machine with a DAW in new and creative ways, without the use of synchronization. Demonstration and application in a modern music production environment.

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

Student Event/Career Development
EC7: SPARS MENTORING
Thursday, October 18, 11:30 am – 1:30 pm
Crystal Palace

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

This event is especially suited for students, recent graduates, young professionals, and those interested in career advice. Hosted by SPARS in cooperation with the AES Education Committee, 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 include: *David Amlen; Chris Mara; Leslie Mona-Mathus; Jamie Baker; Karrie Keyes; Leslie Ann Jones; Piper Payne; Scott Adamson; Barry Rudolph; Dave Hampton; Fred Guarino; Lenise Bent; Geovana Gaitan; Narima Wong; Eve Anna Manley*

AoIP Pavilion
Thursday, October 18, 11:30 am – 12:00 noon
AoIP Pavilion Theater

RAV2SAP: A VALUE-ADDED AES67 MICRO SERVICE

Presenter: **Anthony Kuzub**, Ward-Beck Systems, Toronto, ON, Canada; AES Vice Chair, Toronto Section

AES67 Devices are not required to implement discovery services. Ravenna to SAP is an application that manages, monitors and creates announcements to networked devices. With this SDP translation and creation tool, RAV2SAP can provide announcements of known transmitters that don't have announcement and discovery mechanisms.

Software@AES
Thursday, October 18, 11:30 am – 12:00 noon
Software Pavilion

FABFILTER

Technical Committee Meeting
Thursday, October 18, 12:00 noon – 1:00 pm
Room 1B05

HIGH RESOLUTION AUDIO

Mix with the Masters Workshop
Thursday, October 18, 12:00 noon – 1:00 pm
Booth 458

CHRIS LORD-ALGE

AoIP Pavilion
Thursday, October 18, 12:00 noon – 12:30 pm
AoIP Pavilion Theater

FROM ANALOG, TO DIGITAL, TO AUDIO-OVER-IP— A MANUFACTURER'S PERSPECTIVE

Presenter: **Gordon Kapes**, Studio Technologies, Inc., Skokie, IL, USA

Project Studio Expo
Thursday, October 18, 12:00 noon – 12:45 pm
PSE Stage

PSE08 - GEAR CLUB PODCAST WITH TONY VISCONTI (DAVID BOWIE, T.REX)

Presenter: **Tony Visconti, (David Bowie, T.Rex)i**

Described as one of the most important players in the history of rock, Tony Visonti is a multiple Grammy-winning record producer, arranger, mixer and musician. Since the late 1960s, he has worked with an array of performers including David Bowie, T. Rex, The Moody Blues, Angelique Kidjo, and Kristeen Young.

Sponsored by Eventide

Software@AES
Thursday, October 18, 12:00 noon – 12:30 pm
Software Pavilion

SONARWORKS

Game Audio & XR 6 **Thursday, October 18**
12:30 pm – 1:30 pm **Room 1E06**

SHADOW OF THE TOMB RAIDER: A CASE STUDY DOLBY ATMOS VIDEO GAME MIX

Presenter: **Rob Bridgett**, Eidos Montreal, Montreal, Canada

Shadow of the Tomb Raider was developed at Eidos Montreal over a three year period and had its final mix at Pinewood Studios in the UK over a two week period. The game was mixed entirely in Dolby Atmos for Home Theatre and was one of the first console games to author specific height-based, 3D-sound specifically for this exciting new surround format.

Audio Director, Rob Bridgett, will cover all aspects of bringing this mix to fruition, from planning to execution, in this fascinating post-mortem. Highlights include: • Mix philosophy overview for a blockbuster AAA action title. • Unexpected side-effects of height-based surround. • Critical tools and techniques for surround and overhead-based mixing. • Implementing loudness guidelines. • Differences and benefits of Atmos and object-based surround sound systems for games, over and above those of movies. • Middleware and live-tuning workflow examples and descriptions. • Mix team composition and roles.

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

Special Event
SE5: LUNCHTIME KEYNOTE: THE DISTRIBUTION OF INDEPENDENCE: FROM THE GREEN LINE TO THE BLOCKCHAIN
Thursday, October 18, 12:30 pm – 1:30 pm
Room 1E15+16

Moderator: **Rich Jensen**, formerly Sub Pop Records

The origins of the Sub Pop Record label lead back to an Independent music policy, called "The Green Line," put in place at a tiny community radio station serving Olympia, Washington, in the mid-1970s. Fifteen years later Sub Pop achieved international fame for promoting Seattle-based bands such as Nirvana, Soundgarden, and Mudhoney during the Seattle grunge movement. The label has a history of irreverence, innovation, and a cultural sensitivity that has made it a role model for other so-called Indie labels and artists around the world. In his keynote address, former "Sub-President'"and early Sub Pop recording artist Rich Jensen explores opportunities the new digital technologies such as Blockchain, Smart Contracts, Music Metadata, and the Creative Passport to offer to unleash artists and their communities today into a world that seems hungry for their guidance and inspiration. If Sub Pop had those technologies at its disposal in the 1980s, how would the world be different today?

AoIP Pavilion
Thursday, October 18, 12:30 pm – 1:00 pm
AoIP Pavilion Theater

AES67 AND SMPTE ST 2110—HOW ARE THEY THE SAME, HOW ARE THEY DIFFERENT?

Presenter: **Rick Seegull**, Riedel Communications, Burbank, CA, USA

This session will provide a comparison between the broad specification of AES67 and SMPTE 2110-30 requirements.

Software@AES
Thursday, October 18, 12:30 pm – 1:00 pm
Software Pavilion

SONIBLE

Product Development 7 **Thursday, October 18**
1:00 pm – 2:30 pm **Room 1E09**

B CHAIN SOUND SYSTEM PRODUCT SPECIFICATION: HOW DO WE ADDRESS THE NEEDS AND SUCCESS OF OUR CUSTOMERS?

Presenters: **Klas Dalbjörn**, Powersoft S.P.A., Scandicci, Italy
Claudio Lastrucci, Powersoft S.p.a., Scandicci (FI), Italy
Bennett Prescott, B&C Speakers
, QSC

The missing part of existing and proposed loudspeaker system specifications is the lack of attention to customer requirements. The authors will lead a discussion about a future AES standard that is based on the work that Consultants, Integrators, and End Users do to successfully design, install, and operate a sound system focused on the B Chain.

The specification must include the entire B chain including processing, amplification, and acoustic components. It must have representation from the customer community as well as manufacturers. The goal is offer to the customer a method to specify the components and how they match the sound system requirements. No more marketing driven specifications and disconnected specifications of different components.

Special Event
SE6: DTVAG FORUM
Thursday, October 18, 1:00 pm – 4:00 pm
Room 1E13

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

Panelists: *Tim Carroll*, Senior Director Technology, Sound Group, Office of the CTO, Dolby Laboratories
Stacey Foster, Coordinating Producer, Saturday Night Live
Jackie Green, President and Chief Technology Officer, Alteros
Scott Kramer, Manager Sound Technology, Netflix
Sean Richardson, Executive Director and Principal Audio Engineer, Starz Entertainment
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

Television in Transition: Expanding Possibilities for Audio

The entire television consumption and distribution ecosystem is being transformed at breakneck speed. Ubiquitous and cheap wireless and broadband networking; smart TVs and mobile devices; and massively-scalable cloud computing are building a completely new entertainment distribution system practically overnight, upending the traditional broadcast model, and changing viewing habits forever. This transition from "hardwired" to "virtualized" distribution is expanding the possibilities for television audio innovation, further raising the bar on ultimate quality of premium viewing experiences, while presenting creative challenges in translating these experiences to an ever-widening range of devices.

"The rule book for television distribution is being completely re-written. The migration away from traditional broadcasting to IP delivery will continue to accelerate the uptake of advanced encoding solutions and sophisticated audio services. This transition creates new challenges in providing quality and consistency across an ever-widening range of device and environments. Please join

the DTVAG for a discussion of these and other important television audio issues."—Roger Charlesworth, Executive Director, DTV Audio Group

Forum topics will include:

Chasing Quality

The advent of affordable consumer 4K and HDR on TVs and other devices is transforming the home viewing experience. Combined with the story-telling power of premium episodic content, upscale home viewing is supplanting cinema as the ultimate Hollywood entertainment consumption experience. Audio has been front and center in this transition as more and more premium content becomes available in Dolby Atmos. Is this trend sustainable? How are broadcasters and others responding to the demand for premium audio content?

Surround Virtualization

An important aspect of next-generation audio systems is the ability to virtualize surround presentations over a range of devices and environments. Consumers are already being offered increasingly sophisticated immersive-audio-capable soundbars and TV sets; what are the prospects for enhanced surround virtualization on headphones, earbuds and mobile devices?

Infrastructure and Workflow for Next-Generation Audio

Next-generation audio services greatly increase network operations payloads with additional immersive channels, alternate languages, accessibility features, and all their attendant metadata. Can linear routing systems keep up with these demands or is SMPTE ST 2110 IP media infrastructure arriving just in time to save the day? If so, where are we on the attendant standards and operating protocols to make next-gen audio work in the IP domain? We will debate the issue and take an expert look at the standards crafting work going on behind the scenes.

The Wireless Spectrum Crunch Marches On

Carriers are quickly rolling out services in their newly-acquired 600 MHz blocs making life difficult for wireless mics and other low-power users in many parts of the country. As the rollout in open blocks continues, stations are beginning to exit their existing allocations and head for new frequencies in an already over-crowded 500 MHz TV band. What is the long-term feasibility of operating in what remains of the UHF TV band, and are there practical alternatives on the horizon?

Technical Committee Meeting
Thursday, October 18, 1:00 pm – 2:00 pm
Room 1B05

AUTOMOTIVE AUDIO

AoIP Pavilion
Thursday, October 18, 1:00 pm – 1:30 pm
AoIP Pavilion Theater

AES70 AT A GLANCE

Presenter: **Ethan Wetzell**, OCA Alliance

In the world of media networking, much attention is given to content transport, but an equally important component is how connected devices can be controlled. This presentation will discuss the AES70 open control standard, how it fits within the media networking landscape, and will serve as an introduction to its structure and capabilities for device control.

Software@AES
Thursday, October 18, 1:00 pm – 1:30 pm
Software Pavilion

RELAB

Session P9 **Thursday, Oct. 18**
1:30 pm – 3:30 pm **Room 1E11**

SPATIAL AUDIO—PART 1

Chair: **Jean Marc Jot**, Magic Leap, San Francisco, CA, USA

1:30 pm

P9-1 Impression of Spatially Distributed Reverberation in Multichannel Audio Reproduction—*Sarvesh Agrawal, Jonas Braasch*, Rensselear Polytechnic Institute, Troy, NY, USA

Auditory immersion and spatial impression in multichannel audio reproduction can be altered by changing the number of loudspeakers and independent reverberation channels. The spatial impression can change drastically as one moves away from the sweet-spot. Since multichannel audio reproduction is not limited to one position, it is critical to investigate Listener Envelopment (LEV) and immersion at off-axis positions. This work discusses the impression of spatially distributed decorrelated reverberation at on- and off-axis positions. Laboratory environment is used to reproduce a diffused sound field in the horizontal plane through 128 independent audio channels and loudspeakers. Results from psychoacoustical experiments show that there are perceptible differences even at higher channel counts. However, spatial impression does not change significantly beyond 16 channels of decorrelated reverberation and equally spaced loudspeakers at on- and off-axis positions.
Convention Paper 10076

2:00 pm

P9-2 From Spatial Recording to Immersive Reproduction—Design & Implementation of a 3DOF Audio-Visual VR System—*Maximillian Kentgens, Stefan Kühl, Christiane Antweiler, Peter Jax*, RWTH Aachen University, Aachen, Germany

The complex mutual interaction between human visual perception and hearing demands combined examinations of 360° video and spatial audio systems for Virtual Reality (VR) applications. Therefore, we present a joint audio-visual end-to-end chain from spatial recording to immersive reproduction with full rotational three degrees of freedom (3DOF). The audio-subsystem is based on Higher Order Ambisonics (HOA) obtained by Spherical Microphone Array (SMA) recordings, while the video is captured with a 360° camera rig. A spherical multi-loudspeaker setup for audio in conjunction with a VR head-mounted video display is used to reproduce a scene as close as possible to the original scene with regard to the perceptual modalities of the user. A database of immersive content as a basis for future research in spatial signal processing was set up by recording several rehearsals and concerts of the Aachen Symphony Orchestra. The data was used for a qualitative assessment of the eligibility of the proposed end-to-end system. A discussion shows the potential and limitations of the approach. Therein, we highlight the importance of coherent audio and video to achieve a high degree of immersion with VR recordings.
Convention Paper 10077

2:30 pm

P9-3 Required Bit Rate of MPEG-4 AAC for 22.2 Multichannel Sound Contribution and Distribution—

Shu Kitajima,[1] *Takehiro Sugimoto,*[1] *Satoshi Oode,*[1] *Tomoyasu Komori,*[1] *Joji Urano*[2]
[1] NHK Science & Technology Research Laboratories, Tokyo, Japan
[2] Japan Television Network Corporation, Tokyo, Japan

22.2 multichannel sound (22.2 ch sound) is currently broadcast using MPEG-4 Advanced Audio Coding (AAC) in 8K Super Hi-Vision broadcasting in Japan. The use of MPEG-4 AAC for contribution and distribution transmissions is also planned. Contribution and distribution transmissions require sufficient audio quality to withstand repeated coding and decoding processes. In this study the bit rate of MPEG-4 AAC for a 22.2 ch sound signal satisfying tandem transmission quality was investigated by the subjective evaluation specified in Recommendation ITU-R BS.1116-3. The basic audio quality of 72 stimuli made from a combination of 6 bit rates, 3 different numbers of tandems, and 4 contents were evaluated by 28 listeners. The required bit rates of 22.2 sound material transmission for 3, 5, and 7 tandems were concluded to be 96, 144, and 160 kbit/s per channel, respectively.
Convention Paper 10078

3:00 pm

P9-4 Effect of Binaural Difference in Loudspeaker Directivity on Spatial Audio Processing—
Daekyoung Noh,[1] *Oveal Walker*[2]
[1] Xperi Corp., Santa Ana, CA, USA
[2] XPERI/DTS, Calabasas, CA, USA

Head Related Transfer Functions (HRTFs) are typically measured with loudspeakers facing the listener. Therefore, it is assumed that loudspeaker directivity to the left and the right ear is equal. However, in practice the directivity to both ears may not be equal. For instance, differences can be caused by changes to a listener's location or uncommon loudspeaker driver orientations. This paper discusses the effect of binaural difference in directivity of loudspeaker on spatial audio processing and proposes an efficient solution that improves the spatial effect by compensating the directivity difference. Subjective evaluation is conducted to measure the performance of the proposed solution.
Convention Paper 10079

Mix with the Masters Workshop
Thursday, October 18, 1:00 pm – 2:00 pm
Booth 458

TCHAD BLAKE

Live Sound Events
Thursday, October 18, 1:00 pm – 1:45 pm
Live Production Stage

LS08 - CONCEALED MIKING WITH DPA MICROPHONE

Presenter: **José Frías**, Production Sound Mixer (DPA)

Learn concealed miking techniques for film and television using DPA Microphones discreet line of lavalier microphones. This live demonstration will go through different mic choices, wardrobe considerations, mounting accessories, and proper etiquette when dealing with talent or subjects.

Session P10 **Thursday, Oct. 18**
1:30 pm – 3:30 pm **Room 1E12**

RECORDING AND PRODUCTION

Chair: **Doug Bielmeier**, Purdue University, Indianapolis, IN, USA

1:30 pm

P10-1 The Impact of Compressor Ballistics on the Perceived Style of Music—*Gary Bromham, David Moffat, Mathieu Barthet, György Fazekas,* Queen Mary University London, London, UK

Dynamic range compressors (DRC) are one of the most commonly used audio effect in music production. The timing settings are particularly important for controlling the manner in which they will shape an audio signal. We present a subjective user study of DRC, where a series of different compressor attack and release setting are varied and applied to a set of 30 sec audio tracks. Participants are then asked to rate which ballistic settings are most appropriate for the style of music in their judgment and asked to select one of a series of tag words to describe the style or setting of the song. Results show that the attack parameter influences perceived style more than the release parameter. From the study this is seen more evidently in the case of Jazz and Rock styles than in EDM or Hip-Hop. The area of intelligent music production systems might benefit from this study in the future as it may help to inform appropriateness for certain DRC settings in varying styles.
Convention Paper 10080

2:00 pm

P10-2 Active Multichannel Audio Downmix—
Aleksandr Karapetyan, Felix Fleischmann, Jan Plogsties, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Mixing down signals of a multichannel configuration into a format with fewer channels is widely used in many areas of audio coding, production, and recording. Commonly used downmix methods are based on fixed downmix coefficients or mixing equations. Experience has shown that these passive methods cause quality losses in terms of speech intelligibility, vocal/instrumental clarity, and timbre-changes. In this paper a novel method is introduced that addresses these problems. The method aims to preserve the energy of the input signals during the downmix. In doing so, magnitude and phase are retrieved from two different approaches and combined afterwards. A listening test was conducted. The results prove that the introduced method has a significant positive effect on the aforementioned quality aspects.
Convention Paper 10081

2:30 pm

P10-3 Microphone Array Geometry for Two Dimensional Broadband Sound Field Recording—*Wei-Hsiang Liao, Yuki Mitsufuji, Keiichi Osako, Kazunobu Ohkuri,* Sony Corporation, Tokyo, Japan

Sound field recording with arrays made of omnidirectional microphones suffers from an ill-conditioned problem due to the zero and small values of the spherical Bessel function. This article proposes a geometric design of a microphone array for broadband two dimensional (2D)

sound field recording and reproduction. The design is parametric, with a layout having a discrete rotationally symmetric geometry composed of several geometrically similar subarrays. The actual parameters of the proposed layout can be determined for various acoustic situations to give optimized results. This design has the advantage that it simultaneously satisfies many important requirements of microphone arrays such as error robustness, operating bandwidth, and microphone unit efficiency.
Convention Paper 10082

3:00 pm

P10-4 Risk of Sound-Induced Hearing Disorders for Audio Post Production Engineers: A Preliminary Study—
Laura Sinnott, Barbara Weinstein, City University of New York, New York, NY, USA

In this preliminary study, sound dosimetry measurements were conducted at film studios to assess whether audio post-production engineers are at risk for sound-induced hearing loss. Additionally, we measured 23 engineers' hearing thresholds and assessed their self-perception of hearing disorders via a new questionnaire. Our results show that most participants had at least one audiometric notch, which is an early indicator of noise-induced hearing loss, and most reported experiencing hearing disorders such as tinnitus. Dosimetry suggested that sound levels pose a low risk of permanent hearing loss according to NIOSH criteria, but these criteria are not protective for disorders such as tinnitus, cochlear synaptopathy or even early threshold shifts. We recommend routine hearing evaluations and the use of hearing protection to maintain healthy hearing.
Convention Paper 10083

Broadcast/Online Delivery 7 — Thursday, October 18
1:30 pm – 3:00 pm — Room 1E07

STANDARDIZING STREAMING AUDIO DESCRIPTIVE METADATA

Chair: **Samuel Sousa**, Triton Digital, Montreal, QC, Canada

Panelists: *Dean Mitchell*, StreamGuys
Mike Smith, MainStreaming, Inc., San Francisco, CA, USA

It is easy to say that metadata is available of files and for on demand delivery most CMS systems will have some mechanism of storing metadata along with the file. But having it available during live broadcasts is something that seems to be very neglected. The main reason is often because of technical reasons where the automation system is just limited and making this information available. What can we do about that and which standard should we be pushing forwards to help alleviate this.

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

Standards Committee Meeting
SC-04-04 WORKING GROUP ON MICROPHONE MEASUREMENT AND CHARACTERIZATION
Thursday, October 18, 1:30 pm – 3:00 pm
Room 1B03

The scope of SC-04-04 includes the specification, measurement, and description of the pressure and pressure gradient transduction characteristics in amplitude, time, phase, and spatial domains of microphones intended for the reception of audio signals that are used in professional audio recording, reinforcement, and reproduction applications, individually and in arrays, with and without accessory response-modifying devices, and the interface, environmental, and compatibility characteristics of such microphones.

AoIP Pavilion
Thursday, October 18, 1:30 pm – 2:00 pm
AoIP Pavilion Theater

THE AUDIO PARTS OF SMPTE ST 2110 EXPLAINED

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

This session will explain the fundamentals and possible variations of audio transport within ST 2110 and its compatibility with AES67.

Software@AES
Thursday, October 18, 1:30 pm – 2:00 pm
Software Pavilion

BEST SERVICE

Historical Events 3 — Thursday, October 18
1:45 pm – 2:45 pm — Room 1E08

STANLEY WATKINS, A BELL LABS SOUND PIONEER

Presenter: **Doug Slocum**, Synthetic Sound Labs, Toms River, NJ, USA

Stanley Watkins was a British audio engineer who worked for Western Electric and Bell Labs for the duration of his career. A documentary is in progress about his work with the Warner Brothers launching talking pictures. In 1939 he dazzled audiences at the World's Fair presenting the early talking machine known as the VODER. This presentation will include a demonstration of one of the few remaining VODERs.

Networked Audio 3 — Thursday, October 18
1:45 pm – 2:45 pm — Room 1E10

OPTIMIZING NETWORKS FOR MEDIA

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

This session will examine the network technologies used to optimize a network for modern media transport like Dante. With this knowledge, attendees will have a much better understanding of how to manage networks with mixed traffic and have the basic knowledge to begin diagnosing networked audio issues.

This session will cover TCP vs UDP, Unicast vs Multicast and Quality of Service (QoS). A discussion of clocking will also ensue, contrasting traditional clock distribution on (baseband) BNC vs the networked Precision Time Protocol (PTP). System examples will be given where these technologies have been used.

Recording & Production 10 — Thursday, October 18
1:45 pm – 2:45 pm — Room 1E21

EVOLUTION OF ALBUM PRODUCTION FROM START TO FINISH

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

Panelists: *Heba Kadry*, Chief Mastering Engineer, Timeless Mastering (Bjork, Beach House, Neon Indian)
Erin Tonkon, Producer,Engineer, Mixer (David Bowie, Esperanza Spalding, The Damned)
Simone Torres, Vocal Producer and Multi Platinum engineer (Sia, Dua Lipa, Usher)

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, topline writing, and mastering.

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

Student Events/Career Development
EC8: STUDENT RECORDING CRITIQUES
Thursday, October 18, 1:45 pm – 2:45 pm
Room 1E06

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

Students! Come and get 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 at any stage of their studies can sign up to participate. Sign up at the student (SDA) booth immediately on arrival at the convention and deliver stereo or non-interleaved 5.1 channel mixes as 44.1 Khz/24 bit AIFF or WAVE files, to the SDA booth when you sign up. If you sign up, please make sure you arrive on time at the start of the session, otherwise alternates will be placed on the schedule in your place. 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.

Technical Committee Meeting
Thursday, October 18, 2:00 pm – 3:00 pm
Room 1B05

AUDIO FOR TELECOMMUNICATIONS

Mix with the Masters Workshop
Thursday, October 18, 2:00 pm – 3:00 pm
Booth 458

GREG WELLS

AoIP Pavilion
Thursday, October 18, 2:00 pm – 2:30 pm
AoIP Pavilion Theater

AUDIO OVER IP: PRACTICAL REQUIREMENTS FOR REAL-WORLD USABILITY

Presenter: **Brad Price**, Audinate, Portland, OR, USA

As professional audio installations become synonymous with IP networks, the industry has been abuzz with discussions of protocols and other necessary transport foundations. In reality, people work with solutions that build on these underlying concepts and provide a plethora of additional functionality that makes audio-over-IP usable in the real world. This presentation explores how coherent solutions enhance the experience of audio networking, and how features beyond transport are crucial to the widespread adoption of the technology by the channel and end users.

Live Sound Events
Thursday, October 18, 2:00 pm – 2:45 pm
Live Production Stage

LS09 - AUDIO NETWORKING FOR THEATER & SYSTEM REQUIREMENTS

Presenter: **TBA**

Learn concealed miking techniques for film and television using DPA Microphones discreet line of lavalier microphones. This live demonstration will go through different mic choices, wardrobe considerations, mounting accessories, and proper etiquette when dealing with talent or subjects.

Project Studio Expo
Thursday, October 18, 2:00 pm – 2:45 pm
PSE Stage

PSE09 - MIXING A HIT RECORD

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

Sponsored by Waves

Software@AES
Thursday, October 18, 2:00 pm – 2:30 pm
Software Pavilion

FL STUDIO

AoIP Pavilion
Thursday, October 18, 2:30 pm – 3:00 pm
AoIP Pavilion Theater

SAMPLE-ACCURATE SYNCHRONIZATION OF SMPTE ST 2110 AUDIO STREAMS

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

Detailed explanation of the synchronization fundamentals of ST 2110 and how these can be applied to achieve sample-accurate synchronization among audio streams.

Software@AES
Thursday, October 18, 2:30 pm – 3:00 pm
Software Pavilion

ACCUSONUS

Session P11 **Thursday, Oct. 18**
2:45 pm – 4:15 pm **Poster Area**

POSTERS: APPLICATIONS IN AUDIO

2:45 pm

P11-1 Optimal Exciter Array Placement for Flat-Panel Loudspeakers Based on a Single-Mode, Parallel-Drive Layout—*David Anderson, Michael Heilemann, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Flat-Panel Loudspeakers are most effective at reproducing audio non-directionally when operating at frequencies with many overlapping modes. Frequency regions with low modal overlap produce directional acoustic radiation, long decay times, as well as sharp peaks and notches in pressure. Exciter arrays re-enable use of these frequency regions by restricting structural excitation to a single mode until the frequency region of high modal overlap. An optimization method is described here for determining the placement of exciters such that they are all driven by a single amplifier yet only excite a single structural mode. Experimental results are reported for an acrylic panel with 4, 8, and 11 exciters that demonstrate successful operation of the exciter arrangements.
Convention Paper 10084

2:45 pm

P11-2 Solar Powered Autonomous Node for Wireless Acoustic Sensor Networks Based on ARM Cortex M4—*Alfredo Fernández-Toloba, Héctor A. Sánchez-Hevia, Rubén Espino-Sanjosé, César Clares-Crespo, Joaquín García-Gómez, Roberto Gil-Pita*, University of Alcalá, Alcalá de Henares, Madrid, Spain

The project aims to show the hardware and software of a solar powered autonomous node for wireless acoustic sensor networks based on ARM Cortex M4. The device consists of the following components: a microcontroller, four microphones for audio processing, a radio frequency communication module, a microSD to store and read data, four buzzers to emit a sound, a GPS, and a temperature sensor. Furthermore, the device can be powered by a battery or a solar panel. The device is characterized by low consumption and a small size.
Convention Paper 10085

2:45 pm

P11-3 Vibrational Contrast Control for Local Sound Source Rendering on Flat Panel Loudspeakers—*Ziqing Li*,[1,2] *Pingzhan Luo*,[1,2] *Chengshi Zheng*,[1,2] *Xiaodong Li*[1,2]
[1] Institute of Acoustics, Chinese Academy of Sciences, Beijing, China
[2] University of Chinese Academy of Sciencs, Beijing, China

A vibrational contrast control method is proposed for two-dimensional audio display on a thin flat panel, which is based on maximizing the contrast of the average kinetic energy of the transverse motion between the radiated zone and the total zone. With the measured mobility matrix from the actuators to the measurement points, the optimal filter for each actuator is obtained based on the solutions of the maximization problem. The proposed method does not need to estimate both the natural frequencies and the modal shapes and therefore is easy to be implemented. Experimental results on an aluminum plate show that the proposed method can achieve high contrast level over a wide frequency range.
Convention Paper 10086

2:45 pm

P11-4 Quantifying Listener Preference of Flat-Panel Loudspeakers—*Michael Heilemann*,[1] *David Anderson*,[2] *Stephen Roessner*,[1] *Mark F. Bocko*[1]
[1] University of Rochester, Rochester, NY, USA
[2] University of Pittsburgh, Pittsburgh, PA, USA

We present a perceptual evaluation of flat-panel loudspeakers derived from anechoic amplitude response measurements. Seventy measurements were used to formulate frequency response curves for each loudspeaker characterizing the effects of listener position and in-room reflections. A model developed by Olive [1] was applied to the response curves to predict a preference rating for each loudspeaker. A commercial flat-panel speaker and a flat-panel speaker with a modal crossover network enabled/disabled were measured along with a conventional speaker. The modal crossover speaker scored over 10 points higher than the other flat-panel speakers and displayed a smooth low-frequency response. For flat-panel loudspeakers to produce preference ratings comparable with conventional speakers, structural improvements must be made to reduce the narrow band deviation at high-frequencies.
Convention Paper 10087

2:45 pm

P11-5 Virtual Venues—An All-Pass Based Time-Variant Artificial Reverberation System for Automotive Applications—*Friedrich von Türckheim, Adrian von dem Knesebeck, Tobias Münch*, Harman Becker Automotive Systems GmbH, Munich, Germany

This paper presents an artificial reverberation system for automotive applications. The core reverberation algorithm is based on a time-variant all-pass filter and delay network. It supports an arbitrary number of de-correlated ambience channels and enables the creation of individual direction-dependent early reflection patterns for each output channel. Incorporating de-reverberation technology and microphones, the system allows for 3D direct-ambience upmixing of stereo content simulating the acoustics of existing concert halls as well as for actively modifying and improving the acoustics of car interiors. Ambisonics measurements of real rooms and concert halls serve as starting point for designing the virtual rooms. Adaptive effect stabilization guarantees a consistent spatial impression in presence of masking driving noise.
Convention Paper 10088

2:45 pm

P11-6 Audio Portraiture—The Sound of Identity, an Indigenous Artistic Enquiry—*Maree Sheehan*, Auckland University of Technology, Auckland, New Zealand

To date the potential of 3D immersive and binaural sound technologies have not been applied to audio portraiture nor considered as a means of approaching and expressing indigenous identity. This paper looks at an artistic, practice-led study that utilizes the technology of 3D immersive and binaural sound technologies to create audio portraits and depictions of indigenous Maori women (wahine) from New Zealand/Aotearoa. This enquiry is part of my Ph.D. doctoral research that seeks to artistically interpret the identity and multiple-dimensionality of these women through sound. By multiple-dimensionality, I refer to historical, physical, cognitive, social, emotional, political, and spiritual dimensions of being.
Convention Paper 10089

2:45 pm

2:45 pm

P11-7 Precision Maximization in Anger Detection in Interactive Voice Response Systems—*Inma Mohíno-Herranz, Cosme Llerena-Aguilar, Sr., Joaquín García-Gómez, Manuel Utrilla-Manso, Manuel Rosa-Zurera*, University of Alcalá, Alcalá de Henares, Madrid, Spain

Detection is usually carried out following the Neyman-Pearson criterion to maximize the probability of detection (true positives rate), maintaining the probability of false alarm (false positives rate) below a given threshold. When the classes are unbalanced, the performance cannot be measured just in terms of true positives and false positives rates, and new metrics must be introduced, such as Precision. "Anger detection" in Interactive Voice Response (IVR) systems is one application where precision is important. In this paper a cost function for features selection to maximize precision in anger detection applications is presented. The method has been proved with a real database obtained by recording calls managed by an IVR system, demonstrating its suitability.
Convention Paper 10090

2:45 pm

P11-8 Automatic Guitar Tablature Transcription from Audio Using Inharmonicity Regression and Bayesian Classification—*Jonathan Michelson,*[1] *Richard Stern,*[2] *Thomas Sullivan*[2]
[1] New Sensor Corp. / Electro-Harmonix, Brooklyn, NY, USA
[2] Carnegie Mellon University, Pittsburgh, PA, USA

We propose two new methods to classify guitar strings for automated tablature transcription using only monophonic audio. The first method estimates the linear regression of log-inharmonicities of guitar strings with respect to their pitches and assigns unseen notes to the strings whose means and variances maximize the probability of their measured inharmonicities. The second method, developed as a baseline, characterizes the inharmonicity distribution of each fretboard position as a normal probability density, and then similarly assigns unseen notes to the fretboard positions that maximize the likelihood of their observed inharmonicities. Results from the standard Real World Corpus of guitar recordings show that exploiting regressions generally improves accuracy compared to our baseline, while both achieve adequate performance in guitar-independent test scenarios.
Convention Paper 10091

2:45 pm

P11-9 Harmonic Drum Design Based on Multi-Objective Shape Optimization—*Adam Szwajcowski, Adam Pilch*, AGH University of Science and Technology, Krakow, Poland

A vast majority of drums used in music have round membranes. Sound produced by such an instrument is non-harmonic, so one cannot perceive its pitch clearly. The paper aims to present possibilities of fusing additive synthesis and multi-objective optimization in order to find the relatively simple shape for which a membrane could produce harmonic sound. The proposed approach is based on Multi-Objective Particle Swarm Optimization and uses an original shape parametrization method based on Fourier series. Harmonicity of a drum was assessed based on additive synthesis using solutions of two-dimensional Helmholtz equation on irregular domains by means of the finite difference method.
Convention Paper 10092

2:45 pm

P11-10 Troubleshooting Resource Reservation in Audio Video Bridging Networks—*Christoph Kuhr, Alexander Carôt*, Anhalt University of Applied Sciences, Köthen, Germany

The research project fast-music investigates the requirements of an infrastructure for 60 musicians of a conducted orchestra to do rehearsals via the public internet. Since a single server would not be able to handle the required amount of interleaved audio and video streams, process and distribute them again in a reasonable amount of time, a scalable cloud concept is more promising. The design of the realtime audio and video signal processing cloud, at the heart of a distributed live music session in the public internet, is operating on an Audio Video Bridging network segment. Such a realtime processing cloud requires a proper resource management for network resources. In this paper we present the concept for the processing cloud, evaluate on the resource management, and discuss a troubleshooting strategy for the stream reservation in Audio Video Bridging networks.
Convention Paper 10093

2:45 pm

P11-11 The Sound Diffusion Simulation Software Basing on Finite-Difference Time-Domain Method—*Kamil Piotrowski, Adam Pilch*, AGH University of Science and Technology, Krakow, Poland

The aim of the project was to create an application that allows users to simulate acoustic wave propagation according to given input parameters. The program was based on MATLAB environment and most parts of it were designed using k-Wave toolbox, the package operating on a finite-difference time-domain method calculations (FDTD). The application enables to create a heterogeneous medium and measure sound pressure distribution in a simulated scenario. Separate program module contains time and frequency analysis of obtained waveforms and gives the user a possibility to visualize the results. What is more, the software also computes directional diffusion coefficient *d* in accordance with ISO 17497-2:2012 of defined sound diffusers and makes one independent from complex measurements in an anechoic chamber.
Convention Paper 10075

Product Development 8 — **Thursday, October 18**
2:45 pm – 4:15 pm — **Room 1E09**

BENEFITTING FROM NEW LOUDSPEAKER STANDARDS

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

This tutorial focuses on the development of new IEC standards, addressing conventional and modern measurement techniques applicable to all kinds of transducers, active and passive loudspeakers and other sound reproduction systems. The first proposed standard (IEC 60268-21) describes important acoustical measurements for evaluating the generated sound field and signal distortion. The second standard (IEC 60268-22) is dedicated to the measurement of electrical and mechanical state variables (e.g. displacement), the identification of lumped and distributed parameters (e.g. T/S) and long-term testing to assess power handling, thermal capabilities, product reliability and climate impact. The third standard (IEC 63034) addresses the particularities of micro-speakers used in mobile and other personal audio devices. The tutorial gives a deeper insight into the background, theory and practical know-how behind those standards and shows the relevance for transducer and system design.

Special Event
SE7: AL SCHMITT—ON THE RECORD
Thursday, October 18, 2:45 pm – 3:45 pm
Room 1E15+16)

Presenters: **Maureen Droney**, The Recording Academy, Los Angeles, CA, USA
Al Schmitt, Los Angeles, CA, USA

Al Schmitt is a multiple GRAMMY Award winning engineer and producer; a music legend whose career spans more than six decades. In fact, Al has received the most GRAMMYs ever awarded to an engineer. He's worked with such iconic artists as Paul McCartney, Ray Charles, Toto, Diana Krall, Steely Dan, Bob Dylan, Barbra Streisand, Neil Young, Quincy Jones, Henry Mancini, Tony Bennett, Linda Ronstadt, Natalie Cole, and so many more. In this wide-ranging discussion with Maureen Droney, Managing Director of the Recording Academy Producers & Engineers Wing, Schmitt will share stories from his life in music along with practical tips and sage advice about what it takes to become one of the most in-demand talents in the business. This discussion also commemorates the release of Al's autobiography: *Al Schmitt on the Record: The Magic Behind the Music* (Hal Leonard).

Audio for Cinema 3 **Thursday, October 18**
3:00 pm – 4:30 pm **Room 1E17**

MODERN SCORING WORKFLOWS: MIXING MULTI-STEM SCORE IN THE BOX—A MASTER CLASS FEATURING ALAN MEYERSON

Presenter: **Alan Meyerson**

Legendary film score mixer Alan Meyerson (200+ major film score credits) will elucidate his approach to mixing multi-stemmed scores in the box, using AVID Pro Tools.

Audio for Cinema 4 **Thursday, October 18**
3:00 pm – 4:30 pm **Room 1E06**

***THE 5TH ELEMENT*—HOW A SCI-FI CLASSIC SOUNDS WITH A NEW 3D AUDIO MIX**

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

The 5th Element—it's certainly a milestone in Sci-Fi film history. Recently it was completely overworked doing a completely new film scan in 4k and mixing the whole audio elements again in Dolby Atmos and Headphone Surround 3D. This version was released in Germany as UHD Blu-ray and offers a fantastic new adventure of this great production from Luc Besson. The session offers listening examples and inside information of the production.

Broadcast/Online Delivery 8 **Thursday, October 18**
3:00 pm – 4:30 pm **Room 1E07**

LOUDNESS FOR STREAMING

Moderator: **David Bialik**, Entercom.com, New York, NY, USA

Panelists: *Robert Bleidt*, Fraunhoffer USA
Frank Foti, Telos Systems/Omnia Audio, New York, NY, USA
John Kean, Kean Consulting LLC, Washington, DC, USA
Thomas Lund, Genelec Oy, Iislami, Finland
Mike Smith, MainStreaming, Inc., San Francisco, CA, USA
Samuel Sousa, Triton Digital, Montreal, QC, Canada

Audio fidelity and loudness control are becoming more important as streaming audio grows. Audio being injected from multiple sources, smart speakers becoming a preferred listening device, and being able to listen anywhere to anything make audio quality paramount.

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

Immersive and Spatial Audio 6 **Thursday, October 18**
3:00 pm – 5:00 pm **Room 1E08**

SPATIAL AUDIO MICROPHONES

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

Panelists: *Gary Elko*, mhacoustics
Brian Glasscock, Sennheiser AMBEO
Hyunkook Lee, University of Huddersfield
Len Moskowitz, Core-Sound
Tomasz Zernicki, Zylia

Multichannel loudspeaker setups as well as Virtual Reality applications enable Spatial Sound to be reproduced with large resolution. However, on the recording side it is more complicated to gather a large spatial resolution. Various concepts exist in theory and practice for microphone arrays. In this workshop the different concepts are presented by corresponding experts and differences, applications as well as pros and cons are discussed. The different array solutions include coincident and spaced Ambisonics arrays as well as Stereophonic multi-microphone arrays.

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

Networked Audio 4 **Thursday, October 18**
3:00 pm – 4:00 pm **Room 1E10**

AV FIBER OPTIC CONNECTORS, SELECTION, AND TERMINATIONS

Presenter: **Ronald Ajemian**, Owl Fiber Optics, Flushing, NY, USA; Institute of Audio Research, New York, NY, USA

Now that AV is rapidly increasing in its deployment of fiber optic technology, a need to better understand how fiber optic connectors can affect the signals in a AV system. This workshop is designed to introduce and educate the design engineer, technician, user or student how to select and terminate a fiber optic connector. Examples of popular fiber optic connector types will be discussed with a live or video demonstration on how to terminate a popular fiber optic connector.
Discussion:

• What is a fiber optic connector (FOC)?
• Why use a FOC?
• Which FOC is best for my application?
• How easy is it to put/terminate a FOC together?

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

Recording & Production 11 **Thursday, October 18**
3:00 pm –3:30 pm
Room 1E21

MASTERING IN THE BOX—WHY YOU DON'T NEED ANYTHING ELSE

Presenters: **Ezequiel Morfi**, TITANIO, Buenos Aires, Capital Federal, Argentina
Jonathan Wyner, iZotope, Boston, MA, USA

There has been already much controversy on the topic "analog vs. digital" when it comes to regular use inside the modern studio for mixing and mastering. Not willing to join in that discussion, here is a proposal for a processing-chain entirely conceived inside the digital domain that will deliver on any mastering application needed without the operator "having to do with" or miss anything from the analog world—an entire ITB procedure for common mastering practices for any music genre. Technical explanation of the "workarounds" as done with plug-ins for the common digital shortcomings will be fully explained along with a simple but effective mastering processing chain that is designed to be a template for most studio applications regardless of type of program material.

Student Events/Career Development
EC9: STUDENT RECORDING COMPETITION—PART 1
Thursday, October 18, 3:00 pm – 5:00 pm
Room 1E06

Moderators: **Justin Chervony**, McGill University, Montreal, Quebec, Canada
Bartlomiej Chojnacki, AGH University of Science and Technology, Cracow, Poland; Mega-Acoustic, Kepno, Poland
Mitchell Graham, University of Michigan, Ann Arbor, MI, USA
Maryam Safi, Hamburg, Germany

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 Saturday. The competition is a great chance to hear the work of your fellow students at other educational institutions. A complete list of judges can be found on the SDA website.

3:00 pm Category 1—Traditional Acoustic Recording

4:00 pm Category 2—Traditional Studio Recording

Standards Committee Meeting
SC-04-09 WORKING GROUP ON ASSESSMENT OF ACOUSTIC ANNOYANCE
Thursday, October 18, 3:00 pm – 4:30 pm
Room 1B03

This effort identifies practices for estimation of annoyance of man-made sounds in the presence of background sounds, for automotive/aircraft, consumer, professional and land use planning applications. It includes intentionally generated sounds such as music and sporting events, and unintentional sound such as transportation system noise. It is to include auditory masking, not only integrated sound pressure. It considers relative disturbance but does not set thresholds for acceptability. It does not consider health impacts of sound.

Technical Committee Meeting
Thursday, October 18, 3:00 pm – 4:00 pm
Room 1B05

AUDIO FOR GAMES

Mix with the Masters Workshop
Thursday, October 18, 3:00 pm – 4:00 pm
Booth 458

TONY MASERATI

AoIP Pavilion
Thursday, October 18, 3:00 pm – 3:30 pm
AoIP Pavilion Theater

AOIP: ANATOMY OF A FULL-STACK IMPLEMENTATION

Presenter: **Ievgen Kostiukevych**, European Broadcasting Union, Le Grand-Saconnex, Genéve, Switzerland

The presentation will explain that there is much more to consider when building an AoIP infrastructure than just the AES67 standard. The challenges of synchronization and clocking, discovery and registration, device and network control will be explained, and some solutions will be offered.

Live Sound Events
Thursday, October 18, 3:00 pm – 3:45 pm
Live Production Stage

LS10 - THE PROBLEM WITH MONO – MULTICHANNEL MIXING TECHNIQUES FOR THEATER

Presenter: **Jesse Stevens**, L-Acoustics

This panel will explore common reinforcement techniques for the theatre, weighing the advantages and disadvantages of each approach, with a focus on how to get the most out of mono sources.

Project Studio Expo
Thursday, October 18, 3:00 pm – 3:45 pm
PSE Stage

PSE10 - PROGRAMMING AUDIO SPECIFIC DSPS USING A GUI BASED PROGRAMMING TOOL TO OPTIMIZE CODE DEVELOPMENT

Presenter: **David Thibodeau**, Analog Devices, Wilmington, MA, USA

Analog Devices Inc., will be demonstrating the SigmaStudio GUI based programming tool used to program the SigmaDSP line of digital signal processors. These processors are optimized for processing audio and include some micro-controller like features. The SigmaDSP line of products produced by Analog Devices INC., are a range of products starting with small cost-efficient DSPs with integrated analog and digital converters to large powerful DSPs that can interface to a wide variety of converters and other audio processing systems like Bluetooth and audio networks. The processors include integrated general purpose input and outputs (GPIO) to simplify interfacing to control switches and potentiometers for the adjustments of DSP parameters in real time. The higher-end processors are also capable of many micro-controller-like functions such as booting up external codecs (ADC and DAC ICs) and polling external ICs for errors. Most of the SigmaDSP products are capable of self-booting using an external EEPROM to enable the design of standalone systems that boot themselves and operate without a system controller. Of course, system controllers can be utilized for advanced system solutions. Our demonstration will consist of developing an application using SigmaStudio starting from a blank file and coming up with a functioning solution and programming it into an EEPROM while you watch. Other more advanced concepts will be detailed using prepared projects that showcase capabilities and ease of programming.

Sponsored by Analog Devices

Software@AES
Thursday, October 18, 3:00 pm – 3:30 pm
Software Pavilion

MELODYNE

AoIP Pavilion
Thursday, October 18, 3:30 pm – 4:00 pm
AoIP Pavilion Theater

MONITORING AUDIO STREAMS IN THE IP NETWORK-BASED WORKFLOW

Presenter: **Aki Mäkivirta**, Genelec Oy, Iisalmi, Finland

In this presentation, Aki will explain why the entire studio audio signal paths are now being networked, how IP-connectable monitoring loudspeakers are being used across the broadcast industry to directly monitor IP audio streams, and how installed audio applications can also benefit from this technology.

Software@AES
Thursday, October 18, 3:30 pm – 4:00 pm
Software Pavilion

FABFILTER

Historical Event 4 **Thursday, October 18**
3:45 pm – 4:15 pm **Room 1E11**

FOUND! LAB NOTES OF SHURE'S BEN BAUER

Presenter: **Michael Pettersen**, Shure Inc., Niles, IL, USA

Benjamin B. Bauer (1913–1979) held over 100 patents for acoustical/audio technology, with his first patent, at age 25, being arguably the most significant: invention of the Uniphase principle integral to the Shure Unidyne model 55 microphone. Introduced in 1939 and still manufactured today, the Shure Unidyne was the first unidirectional microphone using a single dynamic element. Today, the Uniphase principle is employed in the vast majority of directional microphones.

In September 2016, Bauer's engineering lab notebooks dating from 1936 to 1944 were located; they had not been seen for over 50 years. The presentation provides a peek into these Bauer notebooks as he discovers and refines the Uniphase principle, as well as numerous other electro-acoustical concepts—some decades ahead of their time.

Special Event
SE8: MASTERED BY BOB LUDWIG: AN EXPLORATION OF HIS CAREER AND TECHNIQUES
Thursday, October 18, 3:45 pm – 4:45 pm
Room 1E21

Presenters: **Jett Galindo**, Bakery Mastering, Los Angeles, CA, USA
Bob Ludwig, Gateway Mastering Studios, Inc., Portland, ME, USA

Jett Galindo of Bakery Mastering hosts a conversation with legendary mastering engineer Bob Ludwig of the famed Gateway Mastering Studios. Ludwig's career began during an era when mastering technology looked very different than it does today. The interview explores Ludwig's perspective on all things mastering including how he has navigated those changes, his observations about what has changed, exploration of technology old and new, and insights into how he approaches his work. We'll hear a sampling of some of his work and learn, what does "Mastered by Bob Ludwig" mean?

Education 2 **Thursday, October 18**
4:00 pm – 5:30 pm **Room 1E12**

WHAT'S DRIVING THE INTERNATIONAL EDU/PRO AUDIO EDUCATION BOOM?

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

Panelists: *Rob Jaczko*, Berklee Collegeo of Music, Boston, MA, USA
Paul Lehrman, Tufts University, Medford, MA, USA
Dana Roun, Full Sail University, Orlando, FL USA
Mary Simoni, Rensselaer Polytechnic Institute, Troy, NY, USA
Cyrille Taillandier, Drexel University's, Westphal College, Philadelphia, PA, USA

Many global universities are making significant investments in faculty, real estate, and cutting-edge technology in support of professional audio production/teaching facilities. 2018 looms as a watershed year for new entries: Drexel University, Rensselaer Polytechnic Institute, Concordia College, ICESI U (Cali, Colombia), are just some of the institutions opening teaching complexes for audio in 2019. This panel of leading educators will explore the escalating commitment to pro audio education. Topics include: • What are the issues in designing for technical and creative training? • How do we "future-proof" our new facilities? • Where do today's (and tomorrow's) students end up working? • Engineering vs. liberal arts / Dedicated schools vs. traditional colleges: How do we incorporate intensive audio training into a more comprehensive curriculum—or should we?

Mix with the Masters Workshop
Thursday, October 18, 4:00 pm – 5:00 pm
Booth 458

BOB POWER & MANNY MARROQUIN

AoIP Pavilion
Thursday, October 18, 4:00 pm – 4:30 pm
AoIP Pavilion Theater

ANEMAN: KEEPING AUDIO NETWORKS UNDER CONTROL

Presenter: **Dominique Brulhart**, Merging Technologies, Puidoux, Switzerland

With the raising and ubiquitous adoption of AES67, audio networks are rapidly becoming more open but as a consequence more and more heterogenic. The new challenge is to keep these networks under control and offer tools allowing managing them as easily as proprietary networks.

Live Sound Events
Thursday, October 18, 4:00 pm – 4:45 pm
Live Production Stage

LS11 - MIXING A MUSICAL

Presenters: **Matt Larson**, DiGiCo- Group One Limited National Sales Manager, Farmingdale, NY
Scott Sanders, Veteran Broadway Mix Engineer

What does it take to mix a musical in a small community theater to Broadway/West End and what technologies can help you with the best results? Scott Sanders & Matt Larson will look at the scope of a small to large show and discuss current tools that will help you in the real-world with a limited or a proper budget as you weave through the Sound Designers visions and what the Director wants.

Project Studio Expo
Thursday, October 18, 4:00 pm – 4:45 pm
PSE Stage

PSE11 - THE NETWORKED STUDIO, A DREAM OR REALITY?

Presenter: **Jan Lykke**, NTP Technology, Gentofte

The presentation will look at the feasibility of a networked studio. Topics such as latency, synchronization and handling multiple sample rates will be covered. This presentation will include some interesting case stories.

Sponsored by NTP/DAD

Software@AES
Thursday, October 18, 4:00 pm – 4:30 pm
Software Pavilion

MAGIX

Networked Audio 5 **Thursday, October 18**
4:15 pm – 5:45 pm **Room 1E10**

AES67/ST2110 TECHNICAL: SYNCHRONIZATION AND REDUNDANCY

Presenters: **Claudio Becker-Foss**, DirectOut GmbH, Mittweida, Germany
Andreas Hildebrand, ALC NetworX GmbH, Munich, Germany

With a shared interest in media operability, the AES and SMPTE have both been developing standards for networked media. The audio part of SMPTE's new standard for "Transport of Professional Media over Managed IP Networks", ST 2110, builds on AES67, the "High-performance streaming audio-over-IP interoperability" standard. This session features several noted authors in the field to discuss these standards from a technical perspective, namely explaining commonalities and constraints among both standards, the principles of sample-accurate synchronization among individual essence streams and how to implement increased operating safety utilizing stream redundancy as defined in SMPTE ST2022-7.

Sound Reinforcement 5 **Thursday, October 18**
4:15 pm – 5:15 pm **Room 1E13**

UNDERSTANDING LINE SOURCE BEHAVIOR FOR BETTER OPTIMIZATION

Presenters: **Etienne Corteel**, L-Acoustics, Marcoussis, France
François Montignies, L-Acoustics, Marcoussis, France

The key to a good loudspeaker system design is the balance between coverage, SPL, and frequency response performances. It deals with various challenges such as directivity control, auditory health preservation, and sonic homogeneity.

In solutions based on a variable curvature line source, the parameters linked to its physical deployment are often overlooked. The temptation to rely on electronic processing to fix resulting problems may then arise. However, it always compromises other performances to some extent, whether system headroom or wavefront integrity.

Using Fresnel analysis, this tutorial points at important aspects of line source behavior and identify the effect of determinant parameters, such as inter-element angles. It shows how an optimized physical deployment allows for rational electronic adjustments, which just become the icing on the cake.

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

Acoustics/Psychoacoustics 2 **Thursday, October 18**
4:30 pm – 6:00 pm **Room 1E11**

TELEPHONOMETRY: THE PRACTICAL ACOUSTICS OF HANDSETS, HEADSETS, AND MOBILE DEVICES

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

This tutorial introduces the basic concepts of Telephonometry with respect to electroacoustic measurements on analogue and digital telephones. Objective methods are emphasized, however, the relationship to subjective measures is also discussed. The historical concept of Loudness Rating and standardized methods for its calculation are reviewed. Standard objective measurements of send, receive, and sidetone are explained. The selection and use of appropriate instrumentation, including ear and mouth simulators, is also described. Techniques for the evaluation of handsets, headsets, speakerphones, and other handsfree devices are presented. Applications of these measurements to analogue, digital, cellular, and VOIP devices are explained. Issues with extended bandwidth devices are discussed and various methods specified in the ITU-T, IEEE, TIA, ETSI, and 3GPP standards are explained.

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

Broadcast/Online Delivery 9 **Thursday, October 18**
4:30 pm – 6:00 pm **Room 1E07**

SPECIAL EVENT: MODERN TRANSMISSION FACILITIES

Chair: **Scott Fybush**, Northeast Radio Watch

Presenters: *Karl Lahm*, Broadcast Transmission Services, LLC, Traverse City. MI, USA
Jim Leifer, American Tower, Boston, MA, USA
John Lyons, Durst Broadcasting LLC, New York, NY, USA
Shane O'Donoghue, Empire State Building, New York, NY, USA

The broadcast transmission facility of the 21st century is a different beast from its 20th century predecessor. Bulky tube transmitters have largely given way to sleek solid-state rigs, and radio broadcasters are beginning to adopt the same liquid-cooled technology that has transformed TV broadcasting. In New York and other cities, older master FM antennas are being replaced with new combiners and antennas for the first time in decades. Ownership consolidation has brought former competitors together to share engineers and transmitter rooms. In this session the site managers and station engineers who oversee many of the biggest broadcast facilities will discuss the challenges they face in keeping their plants at the cutting edge of technical innovation.

Co-organized by the Society of Broadcast Engineers.

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

Product Development 9 **Thursday, October 18**
4:30 pm – 6:00 pm **Room 1E09**

IN-SITU ACOUSTIC MEASUREMENTS OF AUTOMOTIVE AUDIO SYSTEMS

Presenter: **Alfred Svobodnik**, MVOID Group, Karlsruhe, Germany

The small room acoustics of vehicles is a highly challenging topic. Both in the modal as well as in the statistical region there are acoustical deteriorations, unique to automotive spaces, that lead to significant losses in sound quality. This workshop aims to discuss the major challenges and requirements of in-situ measurements of automotive loudspeakers and audio systems and possible solutions to gain an acoustical footprint by means of reliable and representative measurements. Industry experts from OEMs and Tier 1 suppliers will contribute to this topic.

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

Standards Committee Meeting
SC-05-05 WORKING GROUP ON GROUNDING AND EMC PRACTICES
Thursday, October 18, 4:30 pm – 6:00 pm
Room 1B03

The scope of SC-05-05 includes all practices affecting usage and performance of analog audio hardware, with respect to the susceptibility of the signals it carries to effects such as noise and cross-talk due to the manner of its connection and construction, and the effects of its signals on other hardware and systems in its vicinity for professional audio recording, reproduction, and reinforcement. It shall not set standards for personal safety with regard to such connections and construction, but shall keep safety considerations in mind in its recommendations.

AoIP Pavilion
Thursday, October 18, 4:30 pm – 5:00 pm
AoIP Pavilion Theater

NMOS: THE KEY TO WIDE ADOPTION OF IP INFRASTRUCTURES

Presenter: **Rick Seegull**, Riedel Communications, Burbank, CA, USA

This session will explain the differences between NMOS specifications IS-04, IS-05 and IS-06. It will also provide a behind the scenes look into IS-04 and IS-05.

Software@AES
Thursday, October 18, 4:30 pm – 5:00 pm
Software Pavilion

SONARWORKS

Special Event
SE9: MUSIC MAVENS: MANAGING LEGACY STUDIOS IN CHALLENGING TIMES
Thursday, October 18, 4:45 pm – 5:45 pm
Room 1E15+16

Moderator: **Ellis Sorkin**, Studio Referral Service Inc.

Panelists: *Paula Salvatore*
Candace Stewart

The recording process has changed. What you were once able only do in a commercial recording facility, to some extent, is now possible to do in your home or bedroom. But throughout the world there are still new commercial facilities being built every day and there are numerous legacy studios that are consistently busy and booked. This panel discusses how these grand palaces of production continue to thrive in a new recording environment and how these studios have adapted to changing workflow and new technologies. Hear first-hand from industry leaders on what they are doing from a strategic standpoint and how they are keeping these facilities going for artists and the music industry in general. Topics covered also include: Why live acoustic spaces matter and how they enhance any production; how 5-star service still exists and how these top professionals select, manage and train their staff to deliver the highest service and treatment that their clients have come to expect.

Immersive and Spatial Audio 7 **Thursday, October 18**
5:00 pm – 6:00 pm **Room 1E17**

VIRTUAL REALITY AUDIO: B-FORMAT PROCESSING

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

B-Format has had a revival in recent years and has established itself as the audio format of choice for VR videos and content. Experts in signal processing and production tools are presenting and discussing latest innovations in B-Format processing. This includes processing on the recording and rendering side and B-Format post-production.

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

Recording & Production 12 **Thursday, October 18**
5:00 pm – 6:00 pm **Room 1E21**

BEAT BREAKDOWN

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

Panelists: *Zukye Ardella*
Crystal Caines, Producer, Rapper (A$AP Ferg, MIA)
Ms. Madli, Composer, Producer, Beatmaker
Emmanuel "Manny" Mozart

PProducers and beat makers talk about the tools and workflow they use to create tracks for everyone from A$AP Ferg to MIA to Crystal Caines and performing in international producer showcases and and beatmaking competitions like "Beast of the Beats."

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

Mix with the Masters Workshop
Thursday, October 18, 5:00 pm – 6:00 pm
Booth 458

ANDREW SCHEPS & TCHAD BLAKE

AoIP Pavilion
Thursday, October 18, 5:00 pm – 5:30 pm
AoIP Pavilion Theater

TELOS INFINITY: BREAKING THE MATRIX WITH AES67, NEXT GENERATION INTERCOM

Presenter: **Martin Dyster**, The Telos Alliance

Telos Infinity IP Intercom is a complete reimagining of broadcast communications technology developed by the Telos Alliance e ngineering team that invented AoIP for broadcast in 2003. Infinity replaces outmoded matrix technology with an advanced, distributed fully AES67 compliant network solution that provides superior functionality in a simplified, more elegant form. Being matrix-free allows plug-and-play networked hardware and software devices to be added to the system as part of a planned or ad-hoc change, without ever worrying that you might exceed the number of available ports on a matrix.

Live Sound Events
Thursday, October 18, 5:00 pm – 5:45 pm
Live Production Stage

LS12 - LOUDSPEAKER SYSTEM CASE STUDIES FOR MUSICAL AND THEATRICAL APPLICATIONS

Presenter: **TBD**

Project Studio Expo
Thursday, October 18, 5:00 pm – 5:45 pm
PSE Stage

PSE12 - THE PRODUCERS & ENGINEERS WING PRESENTS: GETTING PAID AND CREDITED - LESSONS IN SELF PRESERVATION

Moderator: **Jeff Balding**, NARAS P&E Wing

Panelists: *Danny Kortchmar*, Legendary GRAMMY nominated guitarist, songwriter and producer (Jackson Browne, Don Henley, James Taylor)
Will Lee, GRAMMY Award-winning bassist and Musicians Hall of Fame inductee
Gebre Waddell, Soundways, Memphis, TN, USA

There was a time in the music industry when album credits were second nature and easily found with vinyl records and CD packaging. In recent years, with downloading and streaming, there has been a steep decline in credit delivery and access. As a musician, singer, songwriter, producer or engineer, your credits most likely influence your ability to get your next gig, and it's often also how you are identified for royalties. Music streaming services are beginning to come on board to show credits, and products have emerged to facilitate the process. Join us to hear how crediting is back in the forefront and how you can take advantage of it.

Software@AES
Thursday, October 18, 5:00 pm – 5:30 pm
Software Pavilion

RELAB

Game Audio & XR 7 **Thursday, October 18**
5:15 pm – 5:45 pm **Room 1E08**

DESIGNING GAME AUDIO PLUGINS

Presenter: **Kris Daniel**, McDSP

With the growth of game audio middleware, plugin developers face new engineering challenges to produce plugins that can be deployed to multiple hosting situations. This session will explore how designing for multiple platforms upfront can save time during porting, by analyzing the fundamental pieces of a plugin and learning how those map to multiple game audio environments. Other upfront design considerations such as choice of language, repository organization, and processing strategies will be discussed, as well as pitfalls encountered from the trenches of development.

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

AoIP Pavilion
Thursday, October 18, 5:30 pm – 6:00 pm
AoIP Pavilion Theater

WHAT'S NEW IN AES70-2018

Presenter: **Jeff Berryman**, OCA Alliance

A revision of the AES70 media network control standard is currently in its public review phase, and will become official soon. It is a compatible upgrade to AES70-2015, the current standard. This talk will summarize the features of the new version.

Software@AES
Thursday, October 18, 5:30 pm – 6:00 pm
Software Pavilion

FL STUDIO

Immersive and Spatial Audio 8 **Thursday, October 18**
6:00 pm – 7:00 pm
New York University
Frederick Loewe Theater
35 West 4th St.

OZARK HENRY ON THE HOLODECK: MAPS TO THE STARS

Presenters: **Tom Beyer**, New York University, New York, NY, USA
Paul Geluso, New York University, New York, NY, USA
Agnieszka Roginska, New York University, New York, NY, USA

Distributed concert featuring live performances by international gold and platinum Sony Music recording artist Ozark Henry. The concert will include latest immersive sound technologies and internationally distributed musicians, motion captured-driven avatars interacting with live dancers. This is an ongoing exploration into the creative application of the NYU Holodeck including Immersive sound technologies such as Ambisonics, live MPEG-H broadcast, multi-channel immersive sound and visual system. Remote locations include Trondheim, Norway, and Buenos Aires, Argentina. This concert is in collaboration with THX and Qualcomm and will demonstrate the usage of an off-the-shelf Ateme real-time encoder to stream MPEG-H audio and render THX Spatial audio over loudspeakers in a 5.1.4 configuration.

This event is free but you may want to pick up a ticket at the Registration Area.

Historical Event 5 **Thursday, October 18**
6:45 pm – 8:30 pm
Dolby Theater
1350 Avenue of the Americas, Dolby 88 Screening Room
(corner 6th Ave. & W 55th St.)
doors open at 6:15 pm

BOTH SIDES NOW:
JONI MITCHELL AT THE ISLE OF WIGHT 1970

Presenter: **Eliot Kissileff**

Producer Eliot Kissileff discusses the restoration ordeal of creating the movie and soundtrack, filmed by Murray Lerner and recorded by Teo Macero and Stanley Tonkel.

Featuring concert footage as well as contemporaneous interviews at the festival with attendees as well as a 2003 interview Ms. Mitchell, the film will be released for the first time at the end of September. This will be the first New York showing.

This event will be at the Dolby Theater. Doors will be open at 6:15 pm, program will start at 6:45 pm and end at 8:30. We must be completely out of the room by 9:00 pm.

No food or drink other than water allowed in theater. This is a ticketed event

Special Event
SE10: ORGAN CONCERT
Thursday, October 18, 8:00 pm – 9:00 pm
Church of the Ascension
36–38 Fifth Avenue and West 10th Street

Organist Graham Blyth's concerts are a highlight of every AES convention at which he performs. This year's recital will be held at the Church of the Ascension.

The most outstanding recent development in Ascension's musical life has been the building, installation, and inauguration of the Manton Memorial Organ. Designed specifically for Ascension and built by one of the world's finest artisans, Pascal Quoirin of St. Didier, France, the new organ made its debut in late 2010 following an extensive renovation of the church interior and a long, careful installation process. With two consoles, one electric-action and one mechanical-action ("tracker"), 95 stops, and 111 ranks, the organ is tailor-made to perform the eclectic repertory favored at Ascension. It is the first French-made organ ever installed in New York City and was made possible by a generous grant from The Manton Foundation.

Session P12 **Friday, Oct. 19**
9:00 am – 12:00 noon **Room 1E11**

TRANSDUCERS—PART 3

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

9:00 am

P12-1 A Stepped Acoustic Transmission Line Model of Interference Tubes for Microphones—*Francesco Bigoni,*[1] *Finn T. Agerkvist,*[2] *Eddy Bøgh Brixen*[3,4]
[1] Aalborg University, Copenhagen, Denmark
[2] Technical University of Denmark, Kgs. Lyngby, Denmark
[3] EBB-consult, Smørum, Denmark
[4] DPA Microphones, Allerød, Denmark

This paper presents an extension of the standing-wave model of interference tubes for microphones by Ono et al. The original model accounts for three acoustic parameters: tube length, tube radius, and constant acoustic conductance per unit length. Our extension allows a varying conductance per unit length along the side wall. The assumptions behind the extended model and its ability to predict the frequency response of interference tubes are validated through simulations and by fitting the model parameters to frequency response measurements of a tube with varying conductance per unit length, using two different mountings. Results suggest that a tube with varying conductance per unit length is most effective at attenuating the off-axis sound if the conductance per unit length is decreased towards the tail end of the tube.
Convention Paper 10094

9:30 am

P12-2 Improving Audio Performance of Microphones Using a Novel Approach to Generating 48 Volt Phantom Powering—*Joost Kist,*[1] *Dan Foley*[2]
[1] Triton Audio & PREMA Semiconductor GmbH, Alkmaar, Netherlands
[2] Audio Precision, Beaverton, OR, USA

The introduction of the 48-volt phantom powering circuit in 1966 led to IEC 61938:1996. A key aspect of this powering circuit are the 6.81 kΩ precision resistors that are in parallel to the emitter-follower of the microphone preamplifier. These resistors act as a load on the emitter-follower that causes added distortion. A new approach is presented whereby, in series of these 6K8 resistors, an electronic circuit is placed that acts as a high input-impedance current source, which does not load the emitter-follower. By making this change, THD is decreased by 10 dB while also slightly improving the gain. Measurement results are presented comparing audio performance of a conventional 48-volt phantom power circuit and this new circuit along with circuit details.
Convention Paper 10095

10:00 am

P12-3 Challenges and Best Practices for Microphone End-of-

Line Testing—*Gregor Schmidle,*[1] *Mark Beach,*[2] *Brian MacMillan*[3]
[1] NTi Audio AG, Schaan, Liechtenstein
[2] Beach Dynamics, Cincinnati, OH, USA
[3] NTi Audio Inc., Tigard, OR, USA

Due to the increasing use of microphones in many applications such as automotive or artificial intelligence, the demand for fast and reliable microphone test processes is growing. This paper covers various aspects of the design of an end-of-line microphone test system. A prevailing challenge is to properly control the sound source, as loudspeakers have a tendency to vary their performance due to many influences. The acoustic environment for the test must provide reproducible conditions and is ideally anechoic. Noise from outside must be damped across the measurement bandwidth, so that it doesn't affect the results. Different testing requirements for various types of microphones are shown. Different methods for defining limit criteria are discussed.
Convention Paper 10096

10:30 am

P12-4 Shotgun Microphone with High Directivity by Extra-Long Acoustic Tube and Digital Noise Reduction— *Yo Sasaki, Kazuho Ono*, NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

A prototype of a shotgun microphone having higher directivity than a conventional microphone has been developed to capture target sounds clearly. The shotgun microphone has a structure in which an acoustic tube is attached on a directional microphone capsule. The directivity pattern is formed by adjusting an acoustic resistor attached to orifices along the length of the tube. The prototype we developed has a 1-m long acoustic tube designed on the basis of a numerical calculation. It also includes additional microphone capsules and a digital signal processing circuit that reduce undesired acoustical signals arriving from directions other than the front. Measurements show that the developed shotgun microphone prototype achieves even higher directivity than conventional shotgun microphones.
Convention Paper 10097

11:00 am

P12-5 High Power Density for Class-D Audio Power Amplifiers Equipped with eGaNFETs—*Andreas Stybe Petersen, Niels Elkjær Iversen, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lyngby, Denmark

This paper presents how to optimize the power density of Class-D audio power amplifiers. The main task is to ensure that the ratio between the ripple current and the continuous output current is larger than one. When this is satisfied soft switching conditions are facilitated. Optimizing the amplifier power stage for soft switching while playing audio result in a more evenly distribution of the power dissipation between switching devices and filter inductors. Measured results on 150 Wrms test amplifiers equipped with eGaNFETs shows that the power density can reach 14.3 W/cm3, with THD+N levels as low as 0.03%. Moreover safe operating temperatures below 100°C when playing music with peaking powers of 200 W is achieved. Compared to state-of-the art, the power density of the amplifier module is improved with a factor 2–3.
Convention Paper 10098

11:30 am

P12-6 Estimation of the Headphone "Openness" Based on Measurements of Pressure Division Ratio, Headphone Selection Criterion, and Acoustic Impedance—*Roman Schlieper, Song Li, Stephan Preihs, Jürgen Peissig*, Leibniz Universität Hannover, Hannover, Germany

The study presented here investigates and compares three different methods regarding their suitability for determining the relative openness of circumaural and supraaural headphone types, namely: (1) the Pressure Division Ratio (PDR), (2) the Headphone Selection Criterion (HPC), and the Acoustic Impedance Curve (AIC). Measurements were conducted by using a custom built acoustic impedance measuring tube and an artificial dummy head (KEMAR 45BC-12). The results show that the openness of headphones can be determined best by their low-frequency acoustic impedance curves. Estimations using PDR and HPC show large measurement variations especially in the low frequency range where the perceptual occlusion effect dominates. We introduce the Occlusion Index (OI) that characterizes well the acoustical "openness" and possibly can be used as a reliable indicator for the perceived headphone occlusion.
Convention Paper 10099

Session EB2 Friday, Oct. 19
9:00 am – 10:00 am Room 1E10

LIVE SOUND, RECORDING, AND PRODUCTION

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

9:00 am

EB2-1 Loudness Metering in Sound Reinforcement: Utilities and Practical Considerations—*Cristian Eduardo Becerra Benítez*, Universidad Tecnologica de Chile INACAP, Santiago de Chile, Chile

Loudness meters were created to standardize levels in broadcasting (radio and TV), to minimize differences between program level and commercial level. This standard is already used in several countries but will it be useful for live sound? Will it allow for better mixing results in sound reinforcement situations? The following Engineering Brief aims to answer these questions and provide some considerations for loudness meters use in sound reinforcement.
Engineering Brief 458

9:15 am

EB2-2 Streamlined 3D Sound Design: The Capture and Composition of a Sound Field—*Wieslaw Woszczyk,*[1] *Paul Geluso*[2]
[1] McGill University, Montreal, QC, Canada
[2] New York University, New York, NY, USA

A pragmatic approach to 3D sound design is described that employs a minimum number of sound fields captured with tetrahedral microphones. The captured sound fields, each extended in a horizontal and vertical dimension, are combined to provide the essential segments of the entire 360° sound design. Supplemental single-capsule microphones are used as needed for balancing of spaciousness and clarity. A compatible scaling of sound design from 3D to 2D can be easily accomplished without distortion of timbre or space.
Engineering Brief 459

9:30 am

EB2-3 Interactive Recorded Music: Past, Present, and Future—*Justin Paterson,*[1] *Rob Toulson*[2]

[1] London College of Music, University of West London, London, UK
[2] University of Westminster, London, UK

This Engineering Brief charts the story of user-interactivity with recorded music. Audio technologies and creative compositional techniques are discussed with particular regard to scenarios where creativity has driven the demand for technological advance, and vice-versa, where technical advance has enabled new creative-practice approaches. This is contextualized through discussion of relevant implementation in legacy systems, mobile applications, video games, artificial intelligence, and extended realities. In identifying seminal applications of music interactivity from the past and linking them to present capabilities and practices, future trajectories for interactive recorded-music are extrapolated.
Engineering Brief 460

9:45 am

EB2-4 Producing Audio Drama Content for an Array of Orchestrated Personal Devices—*Jon Francombe,*[1] *James Woodcock,*[2] *Richard J. Hughes,*[2] *Kristian Hentschel,*[1] *Eloise Whitmore,*[3] *Tony Churnside*[3]

[1] BBC Research and Development, Salford, UK
[2] University of Salford, Salford, UK
[3] Naked Productions, Manchester, UK

Personal devices with loudspeakers can be orchestrated to increase immersion from low channel count reproduction systems. A trial production was conducted to investigate the content creation work?ow and delivery mechanism for orchestrated devices. The content (a 13-minute science-?ction drama entitled "The Vostok-K Incident") included: a stereo bed; elements only replayed from auxiliary devices; and elements that could either be in the stereo bed or replayed from auxiliary devices. A bespoke production environment was established, including plugins for authoring the metadata needed to utilize the rendering ruleset. Ambiguity in the reproduction system, coupled with ?exible and complex metadata authoring requirements, made the production challenging and time-consuming. Future work will focus on re?ning the production process and developing delivery tools.
Engineering Brief 461

Broadcast/Online Delivery 10 **Friday, October 19**
9:00 am – 10:30 am **Room 1E07**

LISTENER FATIGUE AND AUDITORY STRESS

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

Panelists: *Bob Ludwig*, Gateway Mastering Studios, Inc., Portland, ME, USA
Susan Rogers, Berklee College of Music, Boston, MA, USA

Listening with acuity for long hours is discussed from recording, mixing and mastering perspectives. We also review recent physiological and psychological studies on listener fatigue, along with research into efferent components of perception. Hearing might be more accurately understood as primarily a reach-out phenomena, and therefore prone to exhaustion of pathways in both directions.

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

Historical Event 6 **Friday, October 19**
9:00 am – 10:30 am **Room 1E17**

THE COMMERCIALIZATION OF STEREOPHONY, 1955–1960

Presenter: **Thomas Fine**, Tom Fine Audio Services, Brewster, NY, USA

In 1955, stereo reel to reel tapes and players came to market, representing the beginning of the record business's commercialization of stereophony. By late 1958, 60 years ago, most of the major companies were releasing stereo LPs.

Continuing on his previous presentation, "The Roots of Stereophony," Fine will detail the technologies, key people, and software of the early commercialization of stereophonic sound recordings. The presentation will include rare audio examples, including excerpts from the first commercially available stereo LP, plus excerpts from the author's collection of stereo demonstration records and tapes.

Also under discussion will be the marketing techniques used to convince people to ditch their one-speaker listening system and embrace 2-channel stereophony.

Immersive and Spatial Audio 9 **Friday, October 19**
9:00 am – 10:00 am **Room 1E08**

SPATIAL REPRODUCTION ON MOBILE DEVICES

Presenter: **Yesenia Lacouture Parodi**, HUAWEI Technologies Duesseldorf GmbH, Munich, Germany

Several techniques to convey almost realistic 3D audio scenes through loudspeakers have already exist for decades, though most of them rely on large amount of loudspeakers and of course good sound quality. However, with mobile devices such smart-phones and tablets we have usually access to maximum 2 channels, the location of the speakers is not always optimal and the quality of micro-speakers is far from being what we would call reasonable. In this tutorial we will discuss how it is possible to overcome some of the limitations we encounter when reproducing spatial audio with mobile devices, what kind of applications can benefit from the use of these technologies and what challenges remain for us to solve.

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

Networked Audio 6 **Friday, October 19**
9:00 am – 10:30 am **Room 1E13**

INTEROPERABILITY STANDARDS FOR IP MEDIA NETWORKING

Chair: **Terry Holton**, Yamaha R&D Centre, London, UK

Panelists: *Mike Cronk*, AIMS Alliance/Grass Valley Group
Kevin Gross, AVA Networks, Boulder, CO, USA
Andreas Hildebrand, ALC NetworX GmbH, Munich, Germany

In the past few years, two major standards have been published with very significant implications for media networking interoperability: AES67 and SMPTE ST 2110. This session will review the background and objectives behind each of these standards. The relationship between these two standards (the commonalities and constraints) will be explained. Recent developments and the future roadmap for both of these important standards will also be explored.

The session will address a range of topics in relation to the AES67 and ST 2110 standards, particularly emphasizing the symbiotic relationship between these standards. The session will also clarify the differences in motivation and objectives behind of the creation of these two standards, as well as looking at recent updates and possible future developments for these standards.

Product Development 10 **Friday, October 19**
9:00 am – 10:15 am **Room 1E09**

DESIGNING HARDWARE AND SOFTWARE COMPONENTS IN MODERN LOUDSPEAKER SYSTEMS

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

Modern loudspeakers use digital signal processing to generate more output with less energy and optimized hardware while coping with signal distortion, instabilities, overload, aging, climate, and other external influences. The control software provides self-learning capabilities and valuable diagnostic information that simplifies the selection of hardware components, integration in an active system, and assessing the performance of the product in the target application. The tutorial explains essential technical characteristics, the physical background, and practical simulation tools to exploit the new opportunities in optimized amplifier and transducer design.

Sound Reinforcement 6 **Friday, October 19**
9:00 am – 10:30 am **Room 1E21**

A COOKBOOK APPROACH TO SOUND SYSTEM OPTIMIZATION WITH BOB MCCARTHY

Presenter: **Bob McCarthy**, Meyer Sound

Delve into the details and methods of sound system optimization using speaker aiming, splay, equalization, phase and amplitude matching in a real-time workshop environment.

Standards Committee Meeting
SC-02-02 WORKING GROUP ON DIGITAL INPUT/OUTPUT INTERFACING
Friday, October 19, 9:00 am – 10:30 am
Room 1B03

The scope of SC-02-02 includes synchronization and the specification of configurations and operating limits for digital interfaces carrying audio, labeling, and control data for professional recording and broadcasting.

Technical Committee Meeting
Friday, October 19, 9:00 am – 10:00 am
Room 1B05

SEMANTIC AUDIO ANALYSIS

Immersive and Spatial Audio 10 **Friday, October 19**
9:30 am – 12:30 pm **Room 1E12**

FACEBOOK 360 TRAINING

Presenters: **Andres A. Mayo**, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina
Abesh Thakur, Facebook, California, USA

Crash course on the end-to-end workflow for spatial audio design and asset preparation of 360 and 180 immersive videos using the Facebook 360 Spatial Workstation tools. This workshop will go through:

a) What is spatial audio, and why is it important for Immersive videos;

b) Formats and standards that are accepted on popular delivery platforms such as Facebook, Oculus or YouTube;

c) Using plugins in a DAW to author ambisonic mixes that respond to real-time headtracking during runtime;

d) Basic setup for live streaming 360 videos with ambisonic audio;

e) Overview of popular 360 cameras and ambisonic microphones that can be used for linear spatial audio design.

Space is limited to 35 people; this is a ticketed event, priority will be given to AES Members ($50) Non-members ($100).

PMC Masters of Audio Program
Friday, October 19, 9:30 am – 10:45 am
Room 1E06

SHOWMAN

Presenter: **John O'Mahony**, Electric Lady Studios

In Conversation with mixer John O'Mahony. From his studio at New York's iconic Electric Lady Studios, John O'Mahony has mixed for everyone from Coldplay to Sara Bareilles, Vance Joy to Twenty One Pilots. He will play back some of his work, discuss moving from Ireland to New York, working with Andy Wallace, and his mix approach.

Student Events/Career Development
EC10: EDUCATION AND CAREER/JOB FAIR
Friday, October 19, 10:00 am – 12:00 noon
Crystal Palace

The combined AES 145thd 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 145tht 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.

Technical Committee Meeting
Friday, October 19, 10:00 am – 11:00 am
Room 1B05

HEARING AND HEARING LOSS PREVENTION

Mix with the Masters Workshop
Friday, October 19, 10:00 am – 11:00 am
Booth 458

JIMMY DOUGLASS

Immersive and Spatial Audio 11 **Friday, October 19**
10:15 am – 11:15 am **Room 1E08**

THE AUDIO EDGE: AMBIENT COMPUTING, ARTIFICIAL INTELLIGENCE AND MACHINE LEARNING WITH SOUND

Presenters: **Sally Kellaway**, Microsoft, Seattle, WA, USA
George Valavanis, Microsoft, Seattle, WA, USA

Audio professionals understand that sound is a powerful signal for capturing and conveying information about the world. From sound designers to composers, we use the communicative capacity of sound to tell stories. Advancements in Artificial Intelligence and Machine Learning are introducing new ways to process sound as data to better understand our environment and expand our awareness.

Presenting findings from Microsoft's Mixed Reality at Work development team, George Valavanis and Sally Kellaway discuss audio's role in our cloud-connected future. From ML data capture workflows to the Microsoft Azure and Dynamics 365 tools used to develop data insights, we'll uncover how audio will expand the way we interact with our world, define a new class of hardware technologies, and become the data stream of the future.

Session EB3 **Friday, Oct. 19**
10:30 am – 12:00 noon **Poster Area**

POSTERS—RECORDING AND PRODUCTION

10:30 am

EB3-1 Microphone Positions in Acoustic Field Reconstruction: Robustness Analysis and Optimization—*Yuchen Shen,*[1] *Ziyun Liu,*[1] *Yong Shen,*[1] Ning Xiang[2]
[1] Nanjing University, Nanjing, China
[2] Rensselaer Polytechnic Institute, Troy, NY, USA

This paper introduces a low-cost, simple structure, and high-convenience 3D microphone positioning system for scanning acoustic field of sound sources. For the inherent errors caused by motors, mechanical fluctuations, and string deformations this work describes an anti-error method based on experimentally measured data to eliminate the corresponding effect. The method can determine a weighting strategy for every measured data and could be extended to any scanning system.
Engineering Brief 462

10:30 am

EB3-2 Creating Object-Based Stimuli to Explore Media Device Orchestration Reproduction Techniques—*Craig Cieciura,*[1] *Russell Mason,*[1] *Philip Coleman,*[1] *Matthew Paradis*[2]
[1] University of Surrey, Guildford, Surrey, UK
[2] BBC Research and Development,Salford, UK

Media Device Orchestration (MDO) makes use of interconnected devices to augment a reproduction system, and could be used to deliver more immersive audio experiences to domestic audiences. To investigate optimal rendering on an MDO-based system, stimuli were created via: (1) object-based audio (OBA) mixes undertaken in a reference listening room; and (2) up to 13 rendered versions of these employing a range of installed and ad-hoc loudspeakers with varying cost, quality, and position. The program items include audio-visual material (short film trailer and big band performance) and audio-only material (radio panel show, pop track, football match, and orchestral performance). The object-based program items and alternate MDO con?gurations are made available for testing and demonstrating OBA systems.
Engineering Brief 463

10:30 am

EB3-3 Practical Recording Techniques for Music Production with Six-Degrees of Freedom Virtual Reality—*David Rivas Méndez,*[1] *Calum Armstrong,*[1] *Jessica Stubbs,*[1] *Mirek Stiles,*[2] *Gavin Kearney*[1]
[1] University of York, York, UK
[2] Abbey Road Studios, London, UK

This paper presents practical spatial audio recording techniques for capturing live music performances for reproduction in a six-degrees of freedom (6DOF) virtual reality (VR) framework. The end-goal is to give the listener the ability to move close to or even around musical sources with a high degree of plausibility to match the visuals. The recording workflow facilitates three major rendering schemes–object-based using spot microphones and diffuse field capture microphone arrays, Ambisonics with multiple-placed sound-field microphones, and hybrid approaches that utilize the prior two methods. The work is presented as a case-study where a jazz ensemble is recorded at Studio 3 of Abbey Road Studios London using the proposed techniques.
Engineering Brief 464

10:30 am

EB3-4 A DAW-Based Interactive Tool for Perceptual Spatial Audio Evaluation—*Tomasz Rudzki, Damian Murphy, Gavin Kearney*, University of York, York, UK

A software tool for subjective audio evaluation is presented. The tool helps to overcome the limits of the existing listening test tools by allowing DAW-based multichannel playback with required signal processing and enabling the use of novel test participant interfaces: mobile app, physical controller, and VR interface. Test preparation is done by importing audio samples into the spatial audio standard DAW and setting up the required signal processing plugins. The listening test tool triggers the playback of the desired audio samples inside the DAW, according to the participant's choice. The tool described in this paper can be used for various perceptual audio tests, including evaluation of spatial audio codecs, virtual acoustics, and binaural rendering engines.
Engineering Brief 465

10:30 am

EB3-5 Stationary Music from Users' Viewpoint in VR Applications—*Sungsoo Kim, Sripathi Sridhar*, New York University, New York, NY, USA

The ultimate goal in virtual reality (VR) is to achieve complete immersion in terms of audio and video, where background music is typically included to keep users absorbed in a game or 360-video content. This paper explores a multichannel loudspeaker configuration to anchor the background music to the user's viewpoint in VR. To that

end, an evenly-spaced octagonal loudspeaker configuration is implemented in order to anchor the background music using head tracking data. The real-time panning is achieved through Vector-Base Amplitude Panning (VBAP). This paper also describes a demo interface built using the Oculus Rift in Unity and Max/MSP, as proof of concept.
Engineering Brief 466

10:30 am

EB3-6 Implementation of 4-pi Reverberation Effects in Immersive Sound Contents—Balance of Object-Based Tracks and Channel-Based Tracks—
Akiho Matsuo,[1] *Ritsuko Tsuchikura,*[1] *Masumi Takino,*[2] *Masataka Nakahara*[1]
[1] SONA corporation, Tokyo Japan
[2] be Blue Co., Ltd., Tokyo, Japan

The paper describes effective use of 4-pi (all directional) acoustical information of reverberation for post-production works. Measurement and analysis technique of sound intensities, VSV(Virtual Source Visualizer), is used for capturing 4-pi reverberations, and the obtained reverberations, VSVerb, are mapped on audio tracks of a DAW. In order to obtain precise rendering of spatial characteristics of the VSVerb, it is ideal to assign one object track to one reflection component. However, Dolby Atmos has the number of reflections restricted to 118. According to the hearing impressions by the authors show this restriction is impractical. This paper proposes a practical method to balance objects' and beds' tracks with less auditory deterioration.
Engineering Brief 467

10:30 am

EB3-7 Development of a 4-pi Sampling Reverberator, VSVerb—Source Reduction—*Masataka Nakahara,*[1]
Akira Omoto,[1,2] *Yasuhiko Nagatomo*[3]
[1] ONFUTURE Ltd. ,Tokyo, Japan
[2] Kyushu University, Fukuoka, Japan
[3] Evixar Inc., Tokyo, Japan

The authors developed a 4-pi sampling reverberator, named "VSVerb," which restores a 4-pi reverberant field by using information of dominant reflections that are captured in a target space. The timings and amplitudes of reflections are obtained from the analyses results of the sound intensities that are measured at the site in orthogonal three directions. The generated reverberation provides high S/N performance and enables to adjust various acoustic parameters with no additional measurements. These advantages provide the VSVerb with high affinity with post-production works. In order to enhance its affinity with object-based production schemes, this manuscript proposes a practical method to reduce a number of reflections from generated reverberations. The method, called "Source Reduction," thins out reflections with less auditory deterioration.
Engineering Brief 468

10:30 am

EB3-8 In-Ear Headphone System with Piezoelectric MEMs Driver—*Andreas Männchen,*[1] *Fabian Stoppel,*[2] *Daniel Beer,*[1] *Florian Niekiel,*[2] *Bernhard Wagner*[2]
[1] Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[2] Fraunhofer Institute for Silicon Technology ISIT, Itzehoe, Germany

This article presents a prototype in-ear headphone system based on a previously disclosed piezoelectric MEMS driver technology (piezoMEMS). The centerpiece of the earphone is a 4 mm x 4 mm piezoMEMS chip loudspeaker that on its own achieves broadband sound pressure levels of up to 110 dB in an IEC 60318-4 ear simulator. A specifically designed enclosure allows for easy installation of the piezoMEMS driver and takes first steps in optimizing the acoustic performance. Furthermore, the system comprises a specially tailored amplifier as well as a dedicated signal processing concept. The article describes the ideas behind the system, discusses the particular challenges in designing the piezoMEMS earphone, shows measurement results, and, finally, discusses the vast opportunities for future research.
Engineering Brief 469

Product Development 11 **Friday, October 19**
10:30 am – 12:00 noon **Room 1E09**

VOICE INPUT: WHAT IS THE MAGIC AND WHAT'S NEXT?

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

Voice user interface is being added to a myriad of products. The trend started with smart speakers but is now expanding to just about any product with a user interface: appliances, automobiles, and TVs, just to name a few. This event is a primer for product developers wanting to incorporate this feature into their products. We cover the underlying algorithms: beamforming, echo cancellation, noise reduction, and direction of arrival estimation and show how they work in concert to achieve far-field reception. The presentation combines basic theory with actual real world demonstrations in order to reinforce concepts.

Standards Committee Meeting
SC-04-08 WORKING GROUP ON MEASUREMENT AND EQUALIZATION OF SOUND SYSTEMS IN ROOMS
Friday, October 19, 10:30 am – 12:00 noon
Room 1B03

The scope of SC-04-08 includes the description, specification, measurement, and calibration of electroacoustic sound systems in rooms and the characteristics of sound presented to an audience.

AoIP Pavilion
Friday, October 19, 10:30 am – 11:00 am
AoIP Pavilion Theater

AES67-101: THE BASICS OF AES67

Presenter: **Anthony Kuzub**, Ward-Beck Systems, Toronto, ON, Canada; AES - Vice Chair, Toronto Section

An exploration of the AES67 standards document. Experience an overview of the basics of synchronization, transport, audio encoding, packet timing, buffering mechanisms, and sessions description.

Software@AES
Friday, October 19, 10:30 am – 11:00 am
Software Pavilion

SONIBLE

Broadcast/Online Delivery 11 **Friday, October 19**
10:45 am – 12:15 pm **Room 1E07**

THE FUTURE OF AUDIO DELIVERY TO THE CONSUMER—HOME & MOBILE

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

Panelists: *Jeffrey Riedmiller*, Dolby Laboratories, San Francisco, CA USA
Brian Savoie, National Association of Broadcasters, Washington, DC, USA
Samara Winterfeld, Xperi, Inc.

Audio technology has never stood still, newer and better ways to deliver audio are constantly being developed. While some legacy audio delivery systems, like AM and FM radio, survive and continue to evolve, others, like cassette tapes and even compact discs, go by the wayside. This session will include presentations focusing on where audio delivery is headed in the internet age and what the proliferation of mobile broadband in particular is doing to increase competition and opportunities for audio service providers. Also to be discussed is the advent of object-based audio formats and the new opportunities available to broadcasters from smart speakers and hybrid (over-the-air plus mobile broadband) radio platforms.

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

Networked Audio 7 Friday, October 19
10:45 am – 12:15 pm Room 1E13

HOW TO GET AES67 INTO YOUR SYSTEMS OR PRODUCTS

Presenters: **Claude Cellier**, Merging Technologies
Mike Dosch, Lawo
Andreas Hildebrand, ALC NetworX GmbH, Munich, Germany
Bill Rounopoulos, Ross Video
Greg Shay, Telos Aliance
Arie van den Broek, Archwave Technologies

This workshop introduces several options to implement AES67 networking capabilities into existing or newly designed products. The session starts with a quick recap on the technical ingredients of AES67 and points out the principal options on implementing AES67 into new or existing products. After providing an overview on commercially available building blocks (modules, software libraries, and reference designs), the workshop commences in a discussion on the value of providing AES67 compatibility from the perspective of providers of existing AoIP networking solutions. The workshop is targeted towards product manufacturers seeking ways to implement AES67 into their products but should also provide valuable insight to those with general technical interest in AES67.

Recording & Production 14 Friday, October 19
10:45 am – 12:15 pm Room 1E21

CLASSICAL MUSIC PRODUCTION: THEN AND NOW

Moderator: **Theresa Leonard**

Presenters: *Steve Epstein*
David Frost
Leslie Ann Jones, Skywalker Sound, San Rafael, CA, USA
John Kilgore, John Kilgore Sound & Recording, New York, NY, USA
Richard King, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
Judith Sherman

This workshop will look at the changing role of the classical music producer over that past 30 years, the producer's relationship with the engineer, and how changes in technology and recording practice have and will continue to evolve for classical—and more broadly, acoustic—music producer of the future.

Recording & Production 15 Friday, October 19
10:45 am – 12:15 pm Room 1E15+16

SPECIAL EVENT: KNOW IT BEFORE YOU TRACK IT—GUITAR LITERACY FOR RECORDING ENGINEERS

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

Join educator and author Alex U. Case for this technical tour of the musical acoustics, psychoacoustics, and electroacoustics of the guitar. Deeper knowledge of the instrument–acoustic and electric–enables the recordist to communicate and collaborate more effectively with the artist. Mastery of the many elements of guitar tone empowers the engineer to get better sounds more quickly. Develop your own creative recording and mixing strategies for this essential instrument built on these technical foundations.

Technical Committee Meeting
Friday, October 19, 11:00 am – 12:00 noon
Room 1B05

CODING OF AUDIO SIGNALS

Mix with the Masters Workshop
Friday, October 19, 11:00 am – 12:00 noon
Booth 458

GAVIN LURSSEN & REUBEN COHEN

AoIP Pavilion
Friday, October 19, 11:00 am – 11:30 am
AoIP Pavilion Theater

AUDIO OVER IP: PRACTICAL REQUIREMENTS FOR REAL-WORLD USABILITY

Presenter: **Brad Price**, Audinate, Portland, OR, USA

As professional audio installations become synonymous with IP networks, the industry has been abuzz with discussions of protocols and other necessary transport foundations. In reality, people work with solutions that build on these underlying concepts and provide a plethora of additional functionality that makes audio-over-IP usable in the real world. This presentation explores how coherent solutions enhance the experience of audio networking, and how features beyond transport are crucial to the widespread adoption of the technology by the channel end users.

Live Sound Events
Friday, October 19, 11:00 am – 11:45 am
Live Production Stage

LS13 - THE 7 MOST COMMON WIRELESS MIC MISTAKES (AND WHAT YOU CAN DO ABOUT THEM)

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

Anyone who has set up or used a wireless mic system, large or small, has faced some of the same problems. This panel of industry experts will explore the most common problems users bring upon themselves, and provide best practice advice for how to improve your results next time around. The basics of wireless mic technology and how to apply it in the real world will be covered along the way.

Project Studio Expo
Friday, October 19, 11:00 am – 11:45 am
PSE Stage

PSE13 - LIVE STREAMING ON FACEBOOK WITH A FOCUSRITE SCARLETT USB INTERFACE

Presenter: **TBA**

Focusrite will be presenting a live, in person tutorial, in which you will learn to add high quality audio to your livestream using an interface from the world's best-selling USB audio interface range, Scarlett. Focusrite staff, along with local New York artist Chelsea Takami, will walk you through the steps to connect a Scarlett interface to your Facebook stream allowing you to present the best sound quality to go along with your video. Chelsea will perform a couple of her original songs which will be livestreamed to the Focusrite Facebook page during the presentation and a short Q&A will follow the livestream.

Sponsored by Focusrite

Software@AES
Friday, October 19, 11:00 am – 11:30 am
Software Pavilion

RELAB

Archiving/Restoration 4 **Friday, October 19**
11:15 am – 11:45 am **Room 1E10**

RECENT ADVANCES IN NOISE REDUCTION: FROM MULTIBAND GATES TO MACHINE LEARNING

Presenter: **Alexey Lukin**, iZotope, Inc., Cambridge, MA, USA

Since 1940s single-band and multiband gates dominated the field of static noise reduction in audio signals. FFT processing enabled by digital computers has dramatically increased the number of bands but hasn't gone far from the multiband gate concept. Recent advances in deep neural nets are able to break free from the multiband gate approach by learning to predict signal features. This talk by iZotope's Principal DSP Engineer Alexey Lukin will describe company's evolution of noise reduction algorithms and provide some insights on what the future may bring for the audio engineering community.

PMC Masters of Audio Program
Friday, October 19, 11:15 am – 12:15 pm
Room 1E06

UMG/CAPITOL STUDIOS DOLBY ATMOS PLAYBACK

Presenter: **Steve Genewicks**

UMG/Capitol Studios Dolby Atmos Playback sessions with Steve Genewick featuring music mixed for Dolby Atmos from Elton John, LL Cool J, Chris Walden, INXS, REM, Public Enemy, Bastille, Arturo Sandoval, Snoh Aleegra and many others.

Game Audio & XR 8 **Friday, October 19**
11:30 am – 12:30 pm **Room 1E08**

WRITING A NEW AUDIO ENGINE FOR UE4: INNOVATION UNDER PRESSURE

Presenter: **Aaron McLeran**, Epic Games

In this talk I will describe the technical design challenges and opportunities inherent in writing a new next-gen-capable audio engine for Unreal Engine 4 (UE4), a widely licensed game engine. I will present the prior state of the audio engine and analyze technical constraints and issues which guided API design choices and feature prioritization. I will then describe challenges we faced while testing for quality, stability and correctness and the process of launching a new audio engine within a regular release schedule of UE4 engine updates while not breaking licensees and backward compatibility. Finally, I will discuss the launch of the audio engine on Fortnite on 6 platforms without slowing down audio content production or interrupting an ambitious 2-week release cadence.

This session is presented in association with the AES Technical Committees on Audio for Games, Spatial Audio, and Audio for Cinema

AoIP Pavilion
Friday, October 19, 11:30 am – 12:00 noon
AoIP Pavilion Theater

AES67 AND SMPTE ST 2110—HOW ARE THEY THE SAME, HOW ARE THEY DIFFERENT?

Presenter: **Rick Seegull**, Riedel Communications, Burbank, CA, USA

Software@AES
Friday, October 19, 11:30 am – 12:00 noon
Software Pavilion

ACCUSONUS

Archiving/Restoration 5 **Friday, October 19**
11:45 am – 12:15 pm **Room 1E10**

AN OVERVIEW OF OPTIMIZING SIGNAL-TO-NOISE RATIO PRIMARILY IN ANALOG AUDIO TAPE RECORDING

Presenter: **Richard Hess**, Richard L Hess Audio Tape Restoration, Aurora, ON, Canada

Optimizing signal-to-noise ratio in audio tape recordings has been part of the design process almost from the beginning and continued throughout the life of the medium. The first step was equalization to better optimize the placement of a typical program spectrum within the linear operating area of the tape. Basic equalization was used throughout the life of the medium, while specialty equalization curves were used for shorter periods of time. Next, a two-track-per-audio-channel approach, recording at two different levels and automatically switching between the two recordings on playback was introduced. A form of this process, without the automatic switching on playback, is still with us today in the "backup recording" mode in some digital field recorders. The two-track approach was quickly replaced by systems that compressed the audio signal in a predetermined way and then expanded the signal upon playback in a complementary manner. These compander systems provided a wider dynamic range without sacrificing half the available tracks. Later, equalization was used with some early digital recordings. Archive and music reissue engineers face the difficult challenge of maintaining aging equipment (some 50 years old) in order to properly replay the companded recordings. A software solution to this challenge is described.

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

Standards Committee Meeting
SC-02-08 WORKING GROUP ON AUDIO-FILE TRANSFER AND EXCHANGE
Friday, October 19, 12:00 noon – 1:00 pm
Room 1B03

The scope of SC-02-08 includes the specification, user implementation, and adoption of technologies for the exchange of audio data files and editing information among systems, by either transfer over a network or by exchange of physical media, for professional recording and broadcasting.

Technical Committee Meeting
Friday, October 19, 12:00 noon – 1:00 pm
Room 1B05

AUDIO FOR CINEMA

Mix with the Masters Workshop
Friday, October 19, 12:00 noon – 1:00 pm
Booth 458

TOM LORD-ALGE

AoIP Pavilion
Friday, October 19, 12:00 noon – 12:30 pm
AoIP Pavilion Theater

THE TWO CONTROL LAYERS OF LARGE MEDIA SYSTEMS

Presenter: **Jeff Berryman**, OCA Alliance

The growth of IP networking technology is giving rise to larger IP projects, in which media networks may interconnect large facilities—campuses, studio complexes, and cities. For such projects, it is important to provide a full range of features, extending from overall asset and workflow management down to detailed control of device operating parameters. This talk offers a two-layer design concept to help cover this range and examines how current standards fit into the picture.

Live Sound Events
Friday, October 19, 12:00 noon – 12:45 pm
Live Production Stage

LS14 - SPECTRUM UPDATE - "WE'RE IN IT NOW!"

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

The 600 MHz spectrum auction concluded last year and many changes to the available UHF frequencies are well underway. Join a panel of experts covering these changes, new FCC regulations, and the affects these changes are having on all UHF wireless microphone, intercom, IEM and IFB users in the core-TV bands.

Project Studio Expo
Friday, October 19, 12:00 noon – 12:45 pm
PSE Stage

PSE14 - GEAR CLUB PODCAST WITH BOB CLEARMOUNTAIN

Presenter: **Bob Clearmountain**

Grammy-winning producer, engineer and mixer, Bob Clearmountain is behind some of the best-selling albums in music history. He has worked with many prominent names in music including Bruce Springsteen, The Rolling Stones, Bryan Adams, Robbie Williams, Toto, Bon Jovi, Altered State, and Simple Minds.

Sponsored by Eventide

Software@AES
Friday, October 19, 12:00 noon – 12:30 pm
Software Pavilion

MAGIX

Recording & Production 16 **Friday, October 19**
12:30 pm – 1:30 pm **Room 1E17**

SMALL ROOM STUDIOS . . . ARE THEY HERE TO STAY?

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

Panelists *Louis Benedetti*, Thompson/Nomad Studios, New York, NY, USA
Judy Elliott-Brown, Rocket Science, Surprise, NY USA
Pastor Joe Cortese, Studio On The Hill, Bronx, NY, USA
Matt MacPhail, Undisclosed Location Studio, Arlington, VA, USA

Big Studios are back! Big Studios are going extinct!! Small Room Studios are the wave of the future! As with idiosyncratic gear selection, studio size has and will always be a matter of personal inclination (and budget). 2017 and now 2018 have seen a steady trickle of new studio introductions around the world. From a spectacular high end "Destination Studio" in Belo Horizonte, Brazil, to an innovative satellite studio in midtown Manhattan, studios keep popping up, burning out, and rising from the ashes in different incarnations. This panel will feature new and established studio owners of all stripes and situations. They will explore this constantly shifting but apparently indomitable field of audio endeavor and illustrate the multiplicity of styles available.

Special Event
SE11: LUNCHTIME KEYNOTE: PRINCE CHARLES ALEXANDER
Friday, October 19, 12:30 pm – 1:30 pm
Room 1E15+16

Presenters: **Prince Charles Alexander**, Berklee College of Music
Paul "Willie Green" Womack, Willie Green Music, Brooklyn, NY, USA

Hip-Hop: The Early Adopter of Emerging Technologies

Hip-Hop is well known for pushing boundaries musically but also has a long history of embracing new technology. From drum machines to samplers to computer based recording and editing, Hip-Hop artists have a legacy of embracing cutting edge technology to drive forward not only their music but the industry at large. This keynote discussion with recording and mixing engineer Prince Charles Alexander (The Notorious B.I.G., P. Diddy, Destiny's Child, Aretha Franklin) explores Hip-Hop's innovating role in the fast paced world of music technology. Moderated by Paul "Willie Green" Womack.

AoIP Pavilion
Friday, October 19, 12:30 pm – 1:00 pm
AoIP Pavilion Theater

AOIP, AES67, AND SMPTE 2110-30, IMPLEMENTATION IN THE REAL WORLD

Presenter: **Ken Tankel**, Linear Acoustic, Malvern, PA, USA

What are some of the benefits of AES67 and how does AES67 fit into the SMPTE ST 2110 standard? What are the practical requirements of putting an AoIP network in place that can allow equipment from different manufacturers to share audio over IP (AoIP) audio streams? What are the pitfalls and what are the benefits?

PMC Masters of Audio Program
Friday, October 19, 12:30 pm – 1:30 pm
Room 1E06

JC LOSADA: A "MUSICAL LATIN LUNCH"K

Presenter: **Juan Cristobal Losada**, New York University, New York, NY, USA

Spend your lunch hour with Venezuelan-born Juan Cristóbal Losada is a GRAMMY and Latin GRAMMY Award winning engineer, producer, and songwriter. Losada has worked with an impressive roster of world-renowned artists, including Carlos Santana, Ricky Martin, Enrique Iglesias, Plácido Domingo, José Feliciano, Shakira, José José, and Chayanne among many others.

Software@AES
Friday, October 19, 12:30 pm – 1:00 pm
Software Pavilion

FABFILTER

Audio Builders Workshop 2 **Friday, October 19**
1:00 pm – 3:00 pm **Crystal Palace**

AUDIO BUILDERS WORKSHOP

Presenters: **Jason Bitner**, Traffic Entertainment Group, Somerville, MA, USA
Dereck Blackburn, Quiethouse Recording, Bedford, MA, USA; Audio Builders Workshop, Boston, MA, USA
Owen Curtin, Audio Builders Workshop, Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA
Brewster LaMacchia, Clockworks Signal Processing LLC, Andover, MA, USA

This special exhibit will walk attendees thru the process of designing and testing a circuit. Mouser Electronics has provided the parts so attendees can keep the project they build Audio Builder Workshop is a workgroup of the Boston AES and is hosting 7 events at the 145th Convention.

Technical Committee Meeting
Friday, October 19, 1:00 pm – 2:00 pm
Room 1B05

LOUDSPEAKERS AND HEADPHONES

Mix with the Masters Workshop
Friday, October 19, 1:00 pm – 2:00 pm
Booth 458

CHRIS LORD-ALGE

AoIP Pavilion
Friday, October 19, 1:00 pm – 1:30 pm
AoIP Pavilion Theater

AES67 PICS, CERTIFICATION, SELF-CERTIFICATION AND PLUGFESTS

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

Live Sound Events
Friday, October 19, 1:00 pm – 1:45 pm
Live Production Stage

LS15 - MIXING FOH & MONITORS – COHABITATION AND COOPERATION

Presenter: **TBA**

Project Studio Expo
Friday, October 19, 1:00 pm – 1:45 pm
PSE Stage

PSE15 - THE SPECIAL SAUCE FOR MIXING A HIT RECORD

Presenters: **Fab Dupont**, Flux Studios, New York, NY, USA
Kevin Killen

Producer Fab Dupont (Mark Ronson, Jennifer Lopez) talks with Kevin Killen (Peter Gabriel, U2, David Bowie) as they walk through one of today's hottest tracks. Hear how the pros approach crafting a hit with the same tools available to you and what that "special sauce" is too.

Software@AES
Friday, October 19, 1:00 pm – 1:30 pm
Software Pavilion

MELODYNE

Session P13 **Friday, Oct. 19**
1:15 pm – 4:15 pm **Room 1E11**

ACOUSTICS AND LIVE SOUND

Chair: **Finn Agerkvist**, Technical University of Denmark, Kgs. Lyngby, Denmark

1:15 pm

P13-1 A Quiet Zone System, Optimized For Large Outdoor Events, Based on Multichannel FxLMS ANC—*Daniel Plewe, Finn T. Agerkvist, Jonas Brunskog*, Technical University of Denmark, Kgs. Lyngby, Denmark

As part of the bigger EU project MONICA (Horizon2020) a local quiet zone system is being developed. This system provides a zone of quiet close to loud outdoor concerts in order to support communications or minimize the noise exposure of staff. Because the noise sources are the loudspeakers of the venue's PA system an ideal reference signal can be obtained from the sound engineer's mixing console, which can be used to apply methods of feedforward active noise control. This paper presents a real time application of the multichannel filtered reference least mean square algorithm (MCFxLMS), shows how it has been designed, implemented, and tested under laboratory conditions.
Convention Paper 10104

1:45 pm

P13-2 Advancements in Propagation Delay Estimation of Acoustic Signals in an Audience Service for Live Events—*Marcel Nophut, Robert Hupke, Stephan Preihs, Jürgen Peissig*, Leibniz Universität Hannover, Hannover, Germany

In the course of the project PMSE-xG an audience service for live events—the Assistive Live Listening Service—was developed. The service uses supplementary augmented audio content presented through transparent headphones to enhance the traditional audio playback of a PA loudspeaker system. Augmented audio content and ambient sound are temporally aligned by the service. This paper proposes signal processing techniques to improve existing methods for estimating the propagation delay of acoustic signals in strongly reverberant environments. These refined methods aim to reduce the computational costs and allow the service to keep track of moving listeners. The proposed methods are evaluated based on realistic recordings of music and speech samples.
Convention Paper 10105

2:15 pm

P13-3 Perceptual Evaluation of an Augmented Audience Service under Realistic Live Conditions—*Robert Hupke, Marcel Nophut, Stephan Preihs, Jürgen Peissig*, Leibniz Universität Hannover, Hannover, Germany

With the usage of future 4G+/5G technologies in wireless equipment for "Programme Making and Special Events" (PMSE) new innovative live services and applications are possible. In this paper our novel "Assistive Live Listening Service" is presented. The service provides individualized additional augmented audio content to every single listener at a concert or voice-based live event by using an augmented reality audio headset that is able to provide both environmental sounds and supplemental audio content. A listening experiment was performed under realistic live conditions to investigate if the service enhances speech intelligibility without a loss of perceptual live experience. The results show that a perceptual enhancement is possible. Further steps to improve the service are discussed.
Convention Paper 10106

2:45 pm

P13-4 Sound Field Control for Reduction of Noise from Outdoor Concerts—*Franz Heuchel, Diego Caviedes Nozal, Finn T. Agerkvist*, Technical University of Denmark, Kgs. Lyngby, Denmark

We investigate sound field control based on the concept of sound zones for the mitigation of low frequency noise from outdoor concerts to the surrounding area by adding secondary loudspeakers to the existing primary sound system. The filters for the secondary loudspeakers are the result of an optimization problem that minimizes the total sound pressure level of both primary and secondary loudspeakers in a sensitive area and the impact of the secondary loudspeakers on the audience area of the concert. We report results from three different experiments with increasing complexity and scale. The sound field control system was reducing the sound pressure level in the dark zone on average by 10 dB below 1 kHz in a small scale experiment in anechoic conditions, by up to 14 dB in a controlled large scale open-air experiment and by up to 6 dB at a pilot test at a music festival.
Convention Paper 10107

3:15 pm

P13-5 Analysis of Piano Duo Tempo Changes in Varying Convolution Reverberation Conditions—*James Weaver, Mathieu Barthet, Elaine Chew*, Queen Mary University London, London, UK

Acoustic design of performance spaces often places the performer relatively low in the hierarchy of needs in comparison to the quality of sound for an audience, and while there are a number of studies relating to solo performers' and symphony orchestras preferred acoustic environments, there is a paucity of literature on objective measurements of the impact of acoustic spaces on smaller ensembles. This study aims to build a methodology for analysis of changes in ensemble musical expression caused by different acoustic environments and extends previous research in the area of acoustics and musical performance.
Convention Paper 10108

3:45 pm

P13-6 Performance and Installation Criteria for Assistive Listening Systems—*Peter Mapp*, Peter Mapp Associates, Colchester, Essex, UK

Approximately 12–15% of the population has a noticeable hearing loss and would benefit from a hearing aid or some form of assistive listening system. However, it is not generally appreciated that hearing aids have a limited operating distance and so are often used in conjunction with an assistive listening system. In recent years, as disability legislation has strengthened, the number of Assistive Listening System installations has dramatically increased, yet there is little guidance or standards available concerning the electroacoustic performance that these systems should meet. The paper reports novel research findings into the acoustic effectiveness of both hearing aids and assistive listening systems, reviews current and upcoming technologies, and sets out a number of potential performance guidelines and criteria.
Convention Paper 10109

Product Development 12 **Friday, October 19**
1:15 pm – 2:45 pm **Room 1E09**

DSP BASED LOUDSPEAKER DESIGN: FASTER AND BETTER AGAIN!

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

The majority of loudspeaker products use DSP based processing. This is because most content is delivered in digital form, and digital processing can improve the sound quality over traditional analog approaches. 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.

Networked Audio 8 **Friday, October 19**
1:30 pm – 2:30 pm **Room 1E13**

AES67 PRACTICAL

Chair: **Claudio Becker-Foss**, DirectOut GmbH, Mittweida, Germany

Panelists: *Andreas Hildebrand*, ALC NetworX GmbH, Munich, Germany
Ievgen Kostiukevych, EBU

Since the publication of AES67 and its recent adoption by the ST 2110 media over IP standard, we see a growing number of installations and studio built around this standard. This workshop and Q&A session will focus on practical aspects of using AES67 and ST 2110-30 in the field: what are the benefits of using AES67? What are the difficulties? What should I do / not do, when configuring my network? Panelists experienced in real-world applications will provide valuable insights and share opinions on the importance of AES67 and its role as part of ST 2110 for the wider broadcast market.

Recording & Production 17 **Friday, October 19**
1:30 pm – 3:00 pm **Room 1E21**

MUSIC MIXING, PART V

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

Panelists: *Pablo Arraya*, Audio Piranha LLC, Brooklyn, NY, USA
Frank Filipetti
Leslie Ann Jones, Skywalker Sound, San Rafael, 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

A panel of award-winning expert practitioners from varying backgrounds and genres within the industry will spark interesting discussion and debate. Topics will include the process of mixing, methodologies that have yielded successful results in a constantly changing industry. Focus will include the different ways to approach a mix, how to improve a mix, how to interpret comments from clients. Balancing, processing, and listening will be addressed. Lots of time for Q&A so the audience can engage the panel members. This series has become very popular with young engineers, educators and seasoned professionals.

A continuation of the successful "Mixing Music" workshop series at AES 139 (New York), AES 140 (Paris), and AES 142 (Berlin) and AES 143 (NY).

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

Standards Committee Meeting
SC-04-01 WORKING GROUP ON ACOUSTICS AND SOUND SOURCE MODELING
Friday, October 19, 1:30 pm – 2:30 pm
Room 1B03

The scope of SC-04-01 includes the specification, modeling, and measurement of electroacoustic transduction sources in spaces used in professional audio recording, reinforcement, and reproduction applications, including the airborne propagation of the signals.

AoIP Pavilion
Friday, October 19, 1:30 pm – 2:00 pm
AoIP Pavilion Theater

HOW AES67 & RAVENNA ENABLES INNOVATION

Presenter: **Bill Rounopoulos**, Ross Video

Software@AES
Friday, October 19, 1:30 pm – 2:00 pm
Software Pavilion

SONARWORKS

Game Audio & XR 9 **Friday, October 19**
1:45 pm – 2:45 pm **Room 1E17**

JUST CAUSE 4: GUNS AND MUSIC AND MIX . . . OH MY!

Presenters: **Ronny Mraz**, Avalanche Studios, New York, NY, USA
Dominic Vega, Avalanche Studios, New York, NY USA

The Just Cause franchise has always presented players one of the largest free-roaming envi-ronments in the open world action genre of games and Just Cause 4 is no exception. During this presentation we'll cover the music, weapons, and mix of JC4 with an in depth look at the ap-proach taken for design and implementation of these systems.

Sound Reinforcement 7 **Friday, October 19**
1:45 pm – 5:00 pm **Room 1E10**

INTERCOM SYSTEMS SUPER SESSION

Chair: **Pete Erskine**, Best Audio, Mt. Vernon NY, USA

Presenters: *Jim Andersen*, RTS
David Beckwith, Riedel
Charles Downs, Unity Intercom
Craig Fredrickson, ClearCom
Scott Gillman, RTS
Donald Kuser, Radio Active Designs
Michael Marston, Unity Intercom
Mark Rehfuss, Pliant Technologies
Gary Rosen, Pliant Technologies
Rom Rosenblum, ClearCom
Rick Seegull, Riedel
Geoff Shearing, Radio Active Designs
Ken Smalley, RTS

An in depth look and comparison of the major wireless intercom systems available today including their associated control platforms and user interfaces. Working systems will be presented by their respective manufacturer product specialists. Currently confirmed manufacturers include Clear-Com, Pliant Technologies, Radio Active Designs, Riedel, RTS, and Unity Intercom.

Student Events/Career Development
EC11: STUDENT RECORDING CRITIQUES
Friday, October 19, 1:45 pm – 2:45 pm
Room 1E06 (PMC Demo Room)

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

Students! Come and get 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 at any stage of their studies can sign up to participate. Sign up at the student (SDA) booth immediately on arrival at the convention and deliver stereo or non-interleaved 5.1 channel mixes as 44.1 Khz/24 bit AIFF or WAVE files, to the SDA booth when you sign up. If you sign up, please make sure you arrive on time at the start of the session, otherwise alternates will be placed on the schedule in your place. 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.

Student Events/Career Development
EC12: AES MATLAB PLUGIN STUDENT COMPETITION
Friday, October 19, 1:45 pm – 3:15 pm
Room 1E08

Moderators: **Justin Chervony**, McGill University, Montreal, Quebec, Canada
Bartlomiej Chojnacki, AGH University of Science and Technology, Cracow, Poland; Mega-Acoustic, Kepno, Poland
Mitchell Graham, University of Michigan, Ann Arbor, MI, USA
Maryam Safi, Hamburg, Germany

MathWorks is supporting the first AES MATLAB Plugin Student Competition and Showcase which invites students to design a new kind of audio production VST plugin using MATLAB Software. The competition provides students with the opportunity to challenge both their signal processing skills and creativity, and to share their results with the audio engineering community.

Join us to watch shortlisted teams present their work in person, for a chance to win cash and software prizes. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2).

Immersive and Spatial Audio 12 **Friday, October 19**
2:00 pm – 5:00 pm **Room 1E12**

FACEBOOK 360 TRAINING

Presenters: **Andres A. Mayo**, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina
Abesh Thakur, Facebook, California, USA

Crash course on the end-to-end workflow for spatial audio design and asset preparation of 360 and 180 immersive videos using the Facebook 360 Spatial Workstation tools. This workshop will go through:

a) What is spatial audio, and why is it important for Immersive videos;

b) Formats and standards that are accepted on popular delivery platforms such as Facebook, Oculus or YouTube;

c) Using plugins in a DAW to author ambisonic mixes that respond to real-time headtracking during runtime;

d) Basic setup for live streaming 360 videos with ambisonic audio;

e) Overview of popular 360 cameras and ambisonic microphones that can be used for linear spatial audio design.

Space is limited to 35 people; this is a ticketed event, priority will be given to AES Members ($50) Non-members ($100).

Technical Committee Meeting
Friday, October 19, 2:00 pm – 3:00 pm
Room 1B05

AUDIO FORENSICS

Mix with the Masters Workshop
Friday, October 19, 2:00 pm – 3:00 pm
Booth 458

LESLIE BRATHWAITE

AoIP Pavilion
Friday, October 19, 2:00 pm – 2:30 pm
AoIP Pavilion Theater

FROM ANALOG, TO DIGITAL, TO AUDIO-OVER-IP—
A MANUFACTURER'S PERSPECTIVE

Presenter: **Gordon Kapes**, Studio Technologies, Inc., Skokie, IL, USA

Live Sound Events
Friday, October 19, 2:00 pm – 2:45 pm
Live Production Stage

LS16 - PRESENTING L-ISA: REINVENTING THE LIVE SOUND EXPERIENCE

Presenter: **Scott Sugden**, L-Acoustics, Oxnard, CA, USA

An introduction to L-ISA Hyperreal Sound and Immersive Hyperreal Sound via an exploration of a rapidly growing number of over 600 real world deployments. L-ISA is the most comprehensive and robust technology for real-time mixing live productions with the confidence that the loudspeaker configuration, the multidimensional mixing tools, and the processing in the L ISA ecosystem will combine to exceed production expectations.

Project Studio Expo
Friday, October 19, 2:00 pm – 2:45 pm
PSE Stage

PSE16 - HOW TO LEARN RECORDING THE WRONG WAY

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

Based on a series of his Tape Op "End Rants" (issues 122 and 127), editor and studio owner/producer Larry Crane will explain how the current state of learning about recording equipment and techniques is dangerous, and can lead to poor recordings, dead ends, and the homogenization of sounds. He will discuss how to improperly utilize resources in the age of the internet, and how to avoid attitudes that make for lousy engineering and production decisions.

Sponsored by Meyer Sound

Software@AES
Friday, October 19, 2:00 pm – 2:30 pm
Software Pavilion

BEST SERVICE

AoIP Pavilion
Friday, October 19, 2:30 pm – 3:00 pm
AoIP Pavilion Theater

OPTIMIZING NETWORKS FOR MEDIA

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

This session will examine the network technologies used to optimize a network for modern media transport. With this knowledge, attendees will have a much better understanding of how to manage networks with mixed traffic and have the basic knowledge to begin diagnosing networked audio issues. This session will cover TCP vs. UDP, Unicast vs. Multicast and Quality of Service (QoS).

Software@AES
Friday, October 19, 2:30 pm – 3:00 pm
Software Pavilion

FL STUDIO

Session P14 **Friday, Oct. 19**
2:45 pm – 4:15 pm **Poster Area**

POSTERS: PERCEPTION

2:45 pm

P14-1 Perception of Stereo Noise Bursts with Controlled Interchannel Coherence—*Steven Crawford, Michael Heilemann, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Lateralization in stereo-rendered acoustic fields with controlled interchannel cross-correlation properties was explored in subjective listening tests. Participants indicated the perceived lateral locations of a series of 2 ms stereo white noise bursts with specified interchannel cross-correlation properties. Additionally, participants were asked to indicate the spatial location and apparent source width of a series of 2 sec white noise bursts composed of one-thousand 2 ms bursts with specified interchannel cross-correlation. The distribution of peak locations in the signals' cross-correlation corresponds to the perceived spatial extent of the auditory image. This illustrates the role of the averaging time in the short-time windowed cross-correlation model of binaural hearing and how the coherence properties of audio signals determine source image properties in spatial audio rendering.
Convention Paper 10110
[Paper presented by Mark Bocko]

2:45 pm

P14-2 Analysis of the performance of Evolved Frequency Log-Energy Coefficients in Hearing Aids for Different Cost Constraints and Scenarios—
Joaquín García-Gómez, Inma Mohíno-Herranz, César Clares-Crespo, Alfredo Fernández-Toloba, Roberto Gil-Pita, University of Alcalá, Alcalá de Henares, Madrid, Spain

Hearing loss is a common problem in old people. Nowadays hearing aids compensate these losses and make their life better, but they present some important issues (reduced battery life, requirement of real-time processing). Because of that, the algorithms implemented in these devices must work at low clock rates. Voice Activity Detection (VAD) is one of the main algorithms used in hearing aids since it is useful for reducing the environmental noise and enhancing the speech intelligibility In this paper a VAD algorithm will be tested using QUT-NOISE-TIMIT Corpus, with different computational cost constraints and at different locations.
Convention Paper 10111

2:45 pm

P14-3 Evaluation of Additional Virtual Sound Sources in a 9.1 Loudspeaker Configuration—*Sungsoo Kim, Sripathi Sridhar*, New York University, New York, NY, USA

This study aims to evaluate the addition of virtual sound sources to a 9.1 loudspeaker configuration in terms of spatial attributes such as envelopment and sound image width. It is the second part of a previous study where different upmixing algorithms to convert stereo to a 9.1 mix were examined. Four virtual sound sources (VSS) are added to a 9.1 configuration to simulate virtual loudspeakers in the height layer with the help of Vector-Based Amplitude Panning (VBAP). A subjective test is conducted to determine whether listeners perceive an improvement due to the addition of VSS channels in the height layer.
Convention Paper 10112

2:45 pm

P14-4 Noticeable Rate of Continuous Change of Intensity for Naturalistic Music Listening in Attentive and Inattentive Audiences—*Yuval Adler*, Stanford University, Stanford, CA, USA

An investigation was done into the threshold of noticeability for a continuous rate of change in intensity and how listener attention affects this threshold rate. Results suggest listener attention has a strong effect on the threshold. Much previous work has been done to try and find the intensity discrimination threshold of human hearing involving comparison of consecutive stimuli differing in intensity but not with a constant change over a long time for a continuous stimulus. Exposure to high intensity sounds over time can damage hearing, so the driving goal behind this investigation is to inform development of a mechanism that will lower the intensity of sound listeners are subjected to when consuming music while having minimal effect on perceived loudness.
Convention Paper 10113

2:45 pm

P14-5 Environment Replication with Binaural Recording: Three-Dimensional (3-D) Quadrant and Elevation Localization Accuracy—*Joseph Erichsen*, *Wesley Bulla*, Belmont University, Nashville, TN, USA

The purpose of this experiment was to examine the spatial accuracy of an environmental image created via binaural recording. Listeners were asked to localize 10 sources each positioned in one of four horizontal quadrants in three vertical planes. A binaural-recording was created in both anechoic and reverberant environments and subjective tests were conducted. The experiment yielded data for a comparative study of the effectiveness of the binaural recording in recreating the perceived source locations in "3-D" space around a specified listening position. ANOVA for overall accuracy and

target hit/miss binomial measures in the free-field and with binaural recordings via headphone reproduction revealed areas of concern for future investigation as well as measures of relative accuracy for the experimental environments.
Convention Paper 10114

2:45 pm

P14-6 Localization of Elevated Virtual Sources Using Four HRTF Datasets—*Patrick Flanagan, Juan Simon Calle Benitez*, THX Ltd., San Francisco, CA, USA

At the core of spatial audio renderers are the HRTF filters that are used to virtually place the sounds in space. There are different ways to calculate these filters, from acoustical measurements to digital calculations using images. In this paper we evaluate the localization of elevated sources using four different HRTF datasets. The datasets used are SADIE (York University), Kemar (MIT), CIPIC (UC Davis), and finally, a personalized dataset that uses an image-capturing technique in which features are extracted from the pinnae. Twenty subjects were asked to determine the location of randomly placed sounds by selecting the azimuth and the elevation from where they felt the sound was coming from. It was found that elevation accuracy is better for HRTFs that are located near elevation = 0°. There was a tendency to under-aim and over-aim towards the area between 0° and 20° in elevation. A high impact of elevation in azimuth location was observed in sounds placed above 60°.
Convention Paper 10115

Special Event
SE12: STUDIO STORIES—REMASTERING JACKSON BROWNE'S "RUNNING ON EMPTY"
Friday, October 19, 2:45 pm – 3:45 pm
Room 1E15+16

Presenters: **Reuben Cohen**
Danny Kortchmar
Gavin Lurssen
Ron McMaster

This panel focuses on the recently re-mastered release of Jackson Browne's iconic LP "Running On Empty" on vinyl, the relevance of working on historical recordings, and revamping the aural experience using and vintage and modern mastering techniques to satisfy the current consumer audio expectations for these recordings. The album was recorded and mixed by the late Greg Ladanyi on tour, live and on stage, or in locations such as backstage, on tour busses, and hotel rooms.

Archiving/Restoration 6 — Friday, October 19
3:00 pm – 4:00 pm — Room 1E13

PRESERVING CLASSIC MULTI-TRACKS (LIKE EARTH WIND & FIRE'S "SEPTEMBER") FOR USE IN THE CLASSROOM

Presenters: **John Krivit**, Professional Audio Design, Hanover, MA, USA; Emerson College, Boston, MA, 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
Eric Schwartz, National Library Preservation Board, Mitchell Silberberg & Knupp LLP
Toby Seay, Drexel University, Philadelphia, PA, USA

In the wake of the 2018 AES Archiving Conference, a demand was created to make available classic multi-track recordings for educational purposes. Addressed were well-founded concerns that archivists cannot keep up with quantities of decomposing magnetic tape being stored in less than ideal conditions. If tape owners could find opportunity to remonetize original multi-track tapes, they would have incentive to hire archivists, and worries about conservation would be allayed. In proposing an electronic library of classic multitrack master recordings distributed by the AES to educational institutions, the panel will present the pedagogical value of their use, discuss existing arrangements for archiving collections at colleges and unwind the legal and ethical concerns of record companies and copyright holders. Attendees can expect to hear multi-tracks of Earth Wind & Fire's "September" presented by George Massenburg and David Bowie's Young Americans as presented by Toby Seay.

Broadcast/Online Delivery 12 — Friday, October 19
3:00 pm – 5:00 pm — Room 1E07

BUILDING A MODERN BROADCAST FACILITY IN A 152-YEAR OLD HOUSE—HOW WKNY RADIO IS TURNING A 152-YEAR-OLD OCTAGONAL HOUSE INTO A CUTTIN EDGE COMMUNITY SERVICE BROADCAST STUDIO

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

Panelists: *Jimmy Buff*, WKNY Radio, Kingston, NY, USA
John Storyk, Walters-Storyk Design Group, Highland, NY, USA
Kristen Thorne, WKNY Radio

In December 2017 Radio Station WKNY (Kingston, NY) announced plans to move from its long-time home to a brand new, cutting-edge broadcast center. On the air since December 1939, WKNY's new home is singularly unique. Hexagonal in shape, the 152-year-old structure (circa 1866) has served a variety of purposes, from Tavern to (most recently) a Hair Salon. Jimmy Buff, Executive Director of the Radio Kingston Corp., which purchased the station in 2017, reports the building's unique six-sided profile is particularly cool, as it "faces all parts of the city." A Hudson Valley broadcast icon, Buff is exec director and an on-air personality at WKNY Radio, WSDG (Highland, NY) is a global broadcast facility, recording studio, teaching/production complex and performance venue architectural/acoustical design firm, retained to create the new station. Though tied in to the local power grid, WKNY is designed to function as an "all solar power" facility. The new complex will include a separate building, which will serve as a Community Center/video production studio for podcasts. WKNY Technical Director Kale Kaposhilin will cover all aspects of the station design process, and address a variety of issues including the challenges of constructing within a building with six sides, systems integration, and funding.

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

Game Audio & XR 10 — Friday, October 19
3:00 pm – 4:00 pm — Room 1E17

WWISE SPATIAL AUDIO: A PRACTICAL APPROACH TO VIRTUAL ACOUSTICS

Presenter: **Nathan Harris**, Audiokinetic, Montreal, QC, Canada

Wwise Spatial Audio is becoming increasingly advanced and now allows for real-time modeling of acoustic phenomena including reflection, diffraction, and sound propagation by informing the sound engine about 3D geometry in the game or simulation.

In this workshop, Nathan Harris, a software developer on the Audiokinetic research and development team, will give a overview of the technology behind Wise Spatial Audio. He will demonstrate how reflection, diffraction, and sound propagation is simulated and how the Wwise authoring tool can be used to monitor and to enable creative intervention when desired. Using the Wwise Audio Lab, a "sandbox" for experimentation, Nathan will walk through a live listening demonstration.

Product Development 13 **Friday, October 19**
3:00 pm – 4:15 pm **Room 1E09**

PRACTICAL DEEP LEARNING INTRODUCTION FOR AUDIO PROCESSING ENGINEERS

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

Are you an audio engineer working on product development or DSP algorithms and willing to integrate AI capabilities within your projects? In this session we will walk through a simple Deep Learning example for speech classification. We will use MATLAB code and a speech command dataset made available by Google. We will cover creating and accessing labeled data, using time-frequency transformations, extracting features, designing and training deep neural network architectures, and testing prototypes on real-time audio. We will also discuss working with other popular Deep Learning tools, including exploiting available pre-trained networks.

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

Student Events/Career Development
EC13: RECORDING COMPETITION—PART 2
Friday, October 20, 3:00 pm – 6:00 pm
Room 1E06

Moderators: **Justin Chervony**, McGill University, Montreal, Quebec, Canada
Bartlomiej Chojnacki, AGH University of Science and Technology, Cracow, Poland; Mega-Acoustic, Kepno, Poland
Mitchell Graham, University of Michigan, Ann Arbor, MI, USA
Maryam Safi, Hamburg, Germany

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 Saturday. The competition is a great chance to hear the work of your fellow students at other educational institutions. A complete list of judges can be found on the SDA website.

3:00 pm Category 3—Modern Studio Recording & Electronic Music

4:00 pm Category 4—Sound for Visual Media

5:00 pm Category 5—Remix

Technical Committee Meeting
Friday, October 19, 3:00 pm – 4:00 pm
Room 1B05

FIBER OPTICS FOR AUDIO

Mix with the Masters Workshop
Friday, October 19, 3:00 pm – 4:00 pm
Booth 458

MARCELLA ARAICA

AoIP Pavilion
Friday, October 19, 3:00 pm – 3:30 pm
AoIP Pavilion Theater

MONITORING AUDIO STREAMS IN THE IP NETWORK-BASED WORKFLOW

Presenter: **Aki Mäkivirta**, Genelec Oy, Iisalmi, Finland

In this presentation, Aki will explain why the entire studio audio signal paths are now being networked, how IP-connectable monitoring loudspeakers are being used across the broadcast industry to directly monitor IP audio streams, and how installed audio applications can also benefit from this technology.

Live Sound Events
Friday, October 19, 3:00 pm – 3:45 pm
Live Production Stage

LS17 - CASE STUDY OF LARGE-SCALE FESTIVAL SOUND SYSTEMS: ROSKILDE FESTIVAL 2018

Presenter: **Bob McCarthy**, Meyer Sound, New York, NY, USA

Roskilde Festival is one of Europe's largest and longest-running music festivals, featuring 8 live music stages running in scale from a few hundred to 110,000 people. In this presentation, Bob McCarthy, Meyer Sound's Director of System Optimization and one of the chief audio engineers for the festival, will discuss the sound system design objectives and tuning processes, as well as the challenges unique to outdoor festivals, tents, and diverse musical acts playing simultaneously.

Project Studio Expo
Friday, October 19, 3:00 pm – 3:45 pm
PSE Stage

PSE17 - AUDIO POST-PRODUCTION: OUT OF TIME AND OVER BUDGET

Presenters: **David Gross**, Producer/Engineer/Composer, Seattle, WA, USA
Glenn Lorbecki, Glenn Sound Inc., Seattle, WA, USA

From TV commercials to movie and game soundtracks, audio post-production services are in high demand. Corporate marketing departments are producing more videos than ever before. This presentation takes a look at how to deliver high quality audio for video productions, regardless of timeline or budget.

Sponsored by Glenn Sound

Software@AES
Friday, October 19, 3:00 pm – 3:30 pm
Software Pavilion

AUDIOSOURCERE

Recording & Production 18 **Friday, October 19**
3:15 pm – 4:15 pm **Room 1E21**

CREATIVE PROCESS OF AN ALTERNATIVE JAZZ ALBUM: "THE CONSTANT" BY JIM BLACK TRIO (INTAKT RECORDS, 2016)

Presenters: **Amandine Pras**, University of Lethbridge, Lethbridge, Alberta, Canada
Alan Silverman, mastering engineer

Critic Kevin Whitehead (NPR) said that the sound on *The Constant* has a distinctive edge that "pops out of the speakers." Recorded, edited and mixed within three days at Water Music Studio by Amandine Pras and mastered by Alan Silverman, *The Constant* received the 2016 Best Album Award by critic David Cristol (*Jazz Magazine*) with a special mention to production.

This workshop will highlight the design of the recording process, which was based on insights into how to best support creative musical improvisation in the studio, drawn from a two-year study within the New York City alternative jazz scene. Just as the music was improvised, so was the mix. Analog mixing without automation was performed on a Neve 8088 which was, in itself, a musical improvisation mirroring drummer Jim Black's, bassist Thomas Morgan's and pianist Elias Stemeseder's music-making process in the studio. Presenters will also detail the combination of layered stereo ambient pickups to capture the natural acoustics of the studio in time-alignment with the close mics that are typical of an East-Coast jazz production. An outline of the mastering approach and Q&A will conclude the workshop.

Additional attention will be given to the challenges faced by women producer/engineers in working with an all-male group of musicians.

Game Audio & XR 11 **Friday, October 19**
3:30 pm – 4:30 pm **Room 1E08**

MICROTALKS: (LISTENING) INTO THE FUTURE

Moderator: **Sally Kellaway**, Microsoft, Seattle, WA, USA

Panelists: *Jean-Pascal Beaudoin*, HEADSPACE STUDIO
Sadah Espii Proctor, Espii Studios
Linda Gedemer, Source Sound VR, Woodland Hills, CA USA
Margaret Schedel, Stony Brook University, Stony Brook, NY, USA
George Valavanis, Microsoft, Seattle, WA, USA

The Audio industries have been in a perpetual state of technological revolution since their inception, making for a volatile, interesting and fast paced environment that have left many and much trailing in its dust. With the current exploration of Games, Virtual, Augmented and Mixed Reality, Artificial Intelligence continuing full-steam-ahead, how do we fit in this new world, and how important can we make audio? Our 5 speakers have 8 minutes to explore what exists in the future for Audio (or the future of listening).

The microtalk panel format is a panel of 5 speakers, each speaking for exactly 8 minutes, with 24 slides auto-advancing every 20 seconds. This session is designed to explore the space around important audio industry topics, speakers aim to provoke and challenge standards of topic, thought and presentation.

Software@AES
Friday, October 19, 3:30 pm – 4:00 pm
Software Pavilion

SONIBLE

Archiving/Restoration 7 **Friday, October 19**
4:00 pm – 5:00 pm **Room 1E15 + 16**

SPECIAL EVENT: PRESERVING THE ARCHIVES OF MAJOR RECORDING ARTISTS

Moderator: **Jessica Thompson**, Jessica Thompson Audio, Berkeley, Ca, USA

Presenters: *Niko Bolas*
Brad Mindich, Inveniem
Susan Rogers, Berklee College of Music, Boston, MA, USA
Steve Rosenthal

Aretha Franklin, David Bowie, Lou Reed, Merle Haggard, Tom Petty—these past few years, we've lost too many major recording artists to list. They leave behind their legacy as artists and performers as well as vaults filled with recordings, demos, sketches, unfinished projects, documents, and more. This panel will address the challenges of dealing with the archives of major recording artists.

Susan Rogers (Berklee, Prince, Barenaked Ladies), Brad Mindich (CEO, Inveniem), engineer/producer Niko Bolas (Neil Young, LeAnn Rimes, The Mavericks), and engineer/producer/archivist Steve Rosenthal (MARS, Lou Reed, Blondie) will share their memories of working in the studio with major artists and discuss the formidable issues around archiving and preservation. They will speak to the technical challenges of identifying and digitizing decades of analog and digital formats, the legal obstacles, and the question of when and how to curate new releases from archival materials.

Session EB4 **Friday, Oct. 19**
4:30 pm – 5:45 pm **Room 1E11**

APPLICATIONS IN AUDIO

Chair: **Ivan Bourmeyster**, ARKAMYS, Paris, France

4:30 pm

EB4-1 Withdrawn [*Engineering Brief 470]*

4:45 pm

EB4-2 Description of the Single Note Database SNDB—*Esther Fee Feichtner, Bernd Edler*, International Audio Laboratories Erlangen, Erlangen, Germany

Big and well-tagged databases are needed for many tasks in music information retrieval. Therefore we created the single note database (SNDB) containing over 30,000 single notes of 11 orchestra instruments by extracting and combining material of well-established databases. The ground truth was manually checked, corrected, and augmented. The result is a large and easy to handle database providing a reliable ground truth with a high variety for each class. Because of well-known original databases, new scientific results can be easily compared to earlier approaches. Here we show the benefit of the SNDB in a simple example. Moreover, we depict how the SNDB was created in the first place and how it can be conveniently reproduced from the original databases.
Engineering Brief 471

5:00 pm

EB4-3 Withdrawn [*Engineering Brief 472]*

5:15 pm

EB4-4 Why Can You Hear a Difference between Pouring Hot and Cold Water? An Investigation of Temperature Dependence in Psychoacoustics—*He Peng,*[1] *Joshua D. Reiss*[2]

[1] Tianjin University, Zhengzhou, China
[2] Queen Mary University of London, London, UK

Studies have shown that listeners can distinguish between hot and cold water being poured based solely on sonic properties, yet the cause of this is unknown. This acoustic perception of temperature is an interesting aspect of multisensory perception and integration. In this paper a series of experiments were performed to investigate the characteristics of auditory information when water is poured at different temperatures into various containers. Based on the results, it attempts to find physical and psychoacoustic explanations for the phenomenon.
Engineering Brief 473

5:30 pm

EB4-5 Introduction to Acoustic Meshes in Audio Applications— *Jason McIntosh*, SAATI, SPA, Appiano Gentile (C0), Italy

Acoustic applications utilize thin porous materials for a variety of reasons, the primary one being to provide an acoustic resistance. This resistance dampens acoustic modes, allowing the designer to control their behavior. However, to achieve consistent resistive damping requires a level of precision manufacturing not found in common materials. Some understanding of the weaving process is helpful in guiding a designer's choice of materials.
Engineering Brief 474

Product Development 14 **Friday, October 19**
4:30 pm – 5:45 pm **Room 1E09**

AUDIO SOURCE SEPARATION—RECENT ADVANCEMENT, APPLICATIONS, AND EVALUATION

Chair: **Chungeun Kim**, University of Surrey, Guildford, Surrey, UK

Panelists: *Jon Francombe*, BBC Research and Development, Salford, UK
Nima Mesgarani, Columbia University, New York, NY, USA
Bryan Pardo, Northwestern University, Evanston, IL, USA

Audio source separation is one of the signal processing techniques inspired by the humans' corresponding cognitive ability in auditory scene analysis. It has a wide range of applications including speech enhancement, sound event detection, and repurposing. Although initially it used to be only possible to separate the sources in very specific capturing configurations, only to a suboptimal level of quality, advanced signal processing techniques, particularly deep learning-related approaches, have both widened the applicability and enhanced the performance of source separation. In this workshop the recent advancement of source separation techniques in various use-cases will be discussed, along with the challenges that the research community is currently facing. Also, research activities on the quality aspects specific to source separation towards effective performance evaluation will be introduced.

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

Recording & Production 13 **Friday, October 19**
4:30 pm – 5:45 pm **Room 1E21**

SPECIAL EVENT: ROAD STORIES: DAVID HEWITT AND DB BROWN

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

Panelists: *Daryl Bornstein*
Jack Douglas
David Hewitt
Jay Messina

David Hewitt and dB Brown crisscrossed the United States, Canada, and beyond in a series of remote recording trucks, initially for the Record Plant. They recorded such historic artists and events as Live Aid, Farm Aid, Watkins Glen and other festivals, Simon and Garfunkel in Central Park, U2, Prince's Purple Rain, The Rolling Stones, Bruce Springsteen, Frank Sinatra, Earth, Wind and Fire, The Grateful Dead, and countless others. Join us for an hour and fifteen minutes of storytelling, technical investigation, possibly the playing of some live multitrack recordings, and wisdom from a pair who have been there. Moderated by Mark Rubel.

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

Standards Committee Meeting
SC-02-12 WORKING GROUP ON AUDIO APPLICATIONS OF NETWORKS
Friday, October 19, 4:30 pm – 6:00 pm
Room 1B03

The scope of SC-02-12 includes the use of various network types for audio and audio-related applications in professional recording and broadcasting.

Game Audio & XR 12 **Friday, October 19**
4:45 pm – 6:00 pm **Room 1E08**

OMNI-DIRECTIONAL: SOUND CAREER PATHS IN VR/AR

Chair: **Chris Burke**, DamianL, Inc., Brooklyn, NY, USA

Panelists: *Jeanine Cowen*, Berklee College of Music, Boston, MA, USA
Aaron McLeran, Epic Games
Andrew Sheron, Freelance Composer/Engineer
Michael Sweet, Berklee College of Music, Boston, MA, USA

After fits and starts in the film, gaming, home cinema, and cellular industries, VR and AR are no longer in search of a reason for being. Great immersive visuals demand great immersive audio and the speed at which standards are being hammered out would make Alan Blumlein's head spin. While a final interoperability spec is still a ways off, the object-oriented nature of ambisonics means that producers can create now for the system s of the future. This is already having a huge effect on the industry with new career paths emerging in sound design, coding, systems design, and more. Jump in the bitstream and learn everything you need to know with our panel of producers and tool developers, and get ready for your new career in sound for im-mersive media!

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

Recording & Production 19 **Friday, October 19**
5:15 pm – 5:45 pm **Room 1E07**

THE DECCA TREE: AN IN-DEPTH HISTORICAL EXAMINATION OF ITS ORIGINS AND DEVELOPMENT

Presenter: **Michael Gray**, classical-discography.org, Alexandria, VA

The Decca Tree is widely used by audio engineers throughout the world. Though the current configuration of the "Tree" is well-understood today, its history is not. Via archival drawings, diagrams, and photographs, this tutorial will chronicle the birth and development of the "Tree." It will show how different balance engineers changed the Tree's microphones and their configurations as they sought to find the sound they wanted. Finally, it will place the Tree in the evolution of what became known as "The Decca Sound."

Special Event
SE13: METALLIANCE—THE MISSION
Friday, October 19, 5:15 pm – 6:15 pm
Room 1E10

Moderator: **Jim Pace**

Panelists: *Chuck Ainlay*, Nashville, Tennessee, USA
Ed Cherney, Edward Cherney Company, Venice, CA, USA
Frank Filipetti
George Massenburg, Schulich School of Music, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
*Elliot Scheine*r, Producer, USA
*Al Schmit*t, Los Angeles, CA, USA

The mission of the METAlliance (a group of globally recognized and award winning engineers) is to promote standards of quality in the art and science of recording music. The panel will discuss their ongoing efforts and provide examples of their efforts and contributions to current and future professionals who aspire to carry forth the craftsmanship music deserves.

Historical Event 7 **Friday, October 19**
6:30 pm – 8:30 pm
Dolby Theater
1350 Avenue of the Americas, Dolby 88 Screening Room
(corner 6th Ave. & W 55th St.)
doors open at 6:15 pm

***COMPANY*: A COMMENTARY**

Presenter: **Dan Mortensen**, Dansound Inc.; Friends of the 30th Street Studio, Seattle, WA, USA

This will be a showing of the D. A. Pennebaker documentary of the making of the Original Cast Recording of the 1970 Stephen Sondheim musical. It was almost entirely shot in Columbia Records' 30th Street Studio and is the best visual tour of the studio that exists. The recording process of the show is shown in detail, as are the trials, tribulations, and joys of all involved as they proceed and/or stumble through every part of that process.

We will listen to a commentary by 12-time Grammy winner Thomas Z. Shepard, original producer of the recording and featured in the movie, and Dan Mortensen, 30th Street Studio researcher. Other guests TBA.

This event will be at the Dolby Theater. Doors will be open at 6:15 pm, program will start at 6:45 pm and end at 8:30. We must be completely out of the room by 9:00 pm.

No food or drink other than water allowed in theater. This is a ticketed event

Special Event
SE14: TH RICHARD C. HEYSER MEMORIAL LECTURE
Friday, October 19, 6:30 pm – 8:00 pm
Room 1E15+16

Lecturer: **John Meyer**, Meyer Sound Labs, Berkeley, 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 Richard C. Heyser distinguished lecturer for the 145th AES Convention is **John Meyer**.

John Meyer's career as an audio innovator was launched in the late 1960s, starting out as designer of custom amplification systems for guitarist Steve Miller then continuing as creator of the legendary Glyph very large horn loudspeakers and as in-house loudspeaker designer at San Francisco's McCune Sound Service. In the 1970s his audio research at the Institute for Advanced Musical Studies in Switzerland led to his first patent, for a low-distortion horn. In 1979 he co-founded (with his wife, Helen) Meyer Sound Laboratories, a company synonymous with innovations in loudspeaker design and acoustical measurement. Meyer led development of the SIM System for audio measurement, which was awarded the prestigious R&D 100 Award in 1992. Meyer Sound also pioneered development of self-powered sound reinforcement loudspeakers with introduction of the MSL-4 in 1994, launching a trend that has since revolutionized the industry. Throughout his career, he has focused on development of linear loudspeaker systems, with his R&D team most recently achieving phase linearity from 20 kHz down to 27 Hz with the Bluehorn System studio monitor. John Meyer is a Fellow of the AES and recipient of the Silver Medal. The title of his lecture is "Taking the Room Out of the Loudspeaker: New Tools for Transparent Reproduction."

With few exceptions, loudspeakers are not used in a free field environment but rather in an enclosed acoustical space. This is inherently problematic as an acoustical space behaves in a manner similar to that of loudspeakers, making it difficult to separate the problematic characteristics of each using common measurement tools or subjective listening tests. John Meyer's lecture will review the history of loudspeaker measurement tools as used both in the free field and in acoustical space, including Richard Heyser's pioneering TDM methods and Meyer Sound's own SIM (Source Independent Measurement) systems. A key focus will be on a new multi-component studio monitor that exhibits absolutely flat amplitude and phase response from 27 Hz to 20 kHz. Because this system effectively "takes the room out of the loudspeaker" it opens up possibilities for correlating new objective testing techniques with subjective listening observations. The lecture also will discuss a new test signal known as M-Noise, which effectively mimics the dynamics of music and avoids inherent weaknesses in the use of pink noise with third-octave analyzers when testing loudspeaker systems used for music. The retrospective will touch on other benchmarks in the quest for linear sound amplification, including the early Glyph large-horn systems, the Grateful Dead's "Wall of Sound" and the HD-1 high resolution studio monitor, a trusted near-field reference that remains in Meyer Sound's product line 29 years after its introduction.

Broadcast/Online Delivery 13 **Friday, October 19**
7:00 pm – 9:00 pm
The Green Space
44 Charlton St.

80TH ANNIVERSARY OF THE MERCURY THEATER'S "WAR OF THE WORLDS"

Moderator: **Sue Zizza**, SueMedia Productions, Carle Place, NY, USA

Cast: *Frank Beacham*
Sammy Jones
David Shinn, National Audio Theatre Festivals, New York, NY
Joel Spector, Audio Consyltant, New York
Herb Squire, Herb Squire, Martinsville, NJ, USA
Seth Winner, Seth B. Winner Sound Studios, Inc., Merrick, NY, USA

The Orson Welles radio broadcast, T*he War of the Worlds*, was monumental in the history of radio broadcasting. Performed as a Halloween episode of *The Mercury Theater of the Air* broadcast on the CBS network, Sunday, October 30, 1938, it scared the nation and became a classic demonstration of the extraordinary power of the radio medium.

An adaptation of H. G. Wells' novel, "The War of the Worlds" (1898), the afternoon rehearsal for the show was thought to be poor by writer, Howard Koch, Welles, and the cast. Then, the young Welles, only 23-years-old, went into action, using his dramatic genius to create a last minute radio masterpiece.

Martians may not have really landed in Grover's Mill, New Jersey in 1938, but the broadcast did send the young Orson Welles to Hollywood, where he next made *Citizen Kane*, one of the greatest films ever produced.

This 80th anniversary of *The War of the Worlds* will feature Frank Beacham, who worked with Welles, to discuss the dramatic tricks he used to turn the broadcast into a compelling and believable drama, plus recordings of the behind-the-scenes story in the CBS Studio that frantic night.

SueMedia's Sue Zizza and David Shinn along with the HEAR Now Festival and Voicescapes Audio Theater will produce and perform live recreations and interpretations from *The War of the Worlds* with actors.

Seth Winner and Sammy Jones will play excerpts from the newly remastered recording of the original broadcast, while Herb and Laurie Squire will discuss the reactions of the audience and public to the broadcast.

New York Public Radio is hosting this event for the Audio Engineering Society Convention. Tickets will be required for access. Space is limited.

Session P15 Saturday, Oct. 20
9:00 am – 10:30 am Room 1E12

AUDIO EDUCATION

Chair: **Elsa Lankford**, Towson University, Towson, MD, USA

9:00 am

P15-1 Development and Evaluation of an Audio Signal Processing Educational Tool to Support Somatosensory Singing Control—*Evangelos Angelakis, George Kosteletos, Areti Andreopoulou, Anastasia Georgaki*, National and Kapodistrian University of Athens, Athens, Greece

This paper discusses a newly designed educational tool called "Match Your Own Voice!" which aims at complementing modern vocal pedagogy. The software addresses the perceptual challenges faced by aspiring singers and the need for more objective unsupervised practice methods and quantified reference points to guide students through their training course. The tool is a real-time visual feedback application that uses lessons with a professional vocal coach as a reference for the students' unassisted practice guidance. It is designed for use on portable computers. The conducted longitudinal study evaluating the tool's effect on the students' practice accuracy showed promising results.
Convention Paper 10116

9:30 am

P15-2 Future Educational Goals for Audio Recording and Production (ARP) Programs: A Decade of Supporting Research—*Doug Bielmeier*, Purdue School of Engineering and Technology, IUPUI, Indianapolis, IN, USA

This paper presentation reviews research collected over the last decade to forecast best practices and goals for future educators and students. A collective mandate of Audio Recording and Production (ARP) educational programs is to provide students with foundational skills/theory to begin a career in the recording and entertainment industry. This paper draws connections between a series of surveys designed to elicit many perspectives of people involved in the education and application of ARP programs, educators in ARP programs, and students enrolled in ARP programs. The review will provide insight and actionable items for continued innovations in ARP educational institutions and the maintenance of a healthy relationship between Industry and ARP programs for employees, employers, educators, and students.
Convention Paper 10117

10:00 am

P15-3 Case Study: An Interdisciplinary Audio Curriculum—*Elsa Lankford, Adam Schwartz*, Towson University, Towson, MD, USA

Audio is interdisciplinary in nature, connecting the disciplines of music, radio, broadcasting, film, science, liberal and fine arts. Audio students benefit from a wide, multi-disciplined curricular approach. Towson University's Radio/Audio concentration in the Electronic Media and Film department has evolved to cover a range of topics and disciplines to better prepare students for a constantly changing professional environment.
Convention Paper 10118

Session EB5 Saturday, Oct. 20
9:00 am – 10:00 am Room 1E11

SPATIAL AUDIO

Chair: **Gavin Kearney**, University of York, York, UK

9:00 am

EB5-1 Creative Approach to Audio in Corporate Brand Experiences—*Alexander Mayo, Nathan Blum, Leonard Roussel*, Arup, New York, NY, USA

Corporate clients have become focused on turning their digital platforms into human experiences. As designers, we must create unique solutions that speak to the corporation's brand identity. For the Lobby at Salesforce New York, the Salesforce "Trailblazer" brand is brought to life through a sonic landscape. A spatial audio system and custom composition is paired with immersive lighting and LED displays to create a multimedia entry into Salesforce. This 3D sound system installation: • Reinforces the Salesforce brand and is inspired by their events; • Employs multiple loudspeakers, network audio transport, and integrated control; • Makes use of audio samples arranged in a composition which utilizes an object-based spatialization engine; • Integrates seamlessly with architectural features and building systems.
Engineering Brief 475

9:15 am

EB5-2 Subspace-Based HRTF Synthesis from Sparse Data: A Joint PCA and ML-Based Approach—*Sunil G. Bharitkar,*[1] *Timothy Mauer,*[2] *Teresa Wells,*[1] *David Berfanger*[2]
[1] HP Labs., Inc., San Francisco, CA, USA
[2] HP, Inc., Vancouver, WA, USA

Head-related transfer functions (HRTF) are used for creating the perception of a virtual sound source at an arbitrary azimuth-elevation. Publicly available databases use a subset of these directions due to physical constraints (viz., loudspeakers for generating the stimuli not being point-sources) and the time required to acquire and deconvolve responses for a large number of spatial directions. In this paper we present a subspace-based technique for reconstructing HRTFs at arbitrary directions for the IRCAM-Listen HRTF database, which comprises a set of HRTFs sampled every 15 deg along the azimuth direction. The presented technique includes first augmenting the sparse IRCAM dataset using the concept of auditory localization blur, then deriving a set of P=6 principal components, using PCA for the original and augmented HRTFs, and then training a neural network (ANN) with these directional principal components. The reconstruction of HRTF corresponding to an arbitrary direction is achieved by post-multiplying the ANN output, comprising the estimated six principal components, with a frequency weighting matrix. The advantage of using a subspace approach, involving only 6 principal components, is to obtain a low complexity HRTF synthesis ANN-based model as compared to training an ANN model to output an HRTF over all frequencies. Objective results demonstrate a reasonable interpolation with the presented approach.
Engineering Brief 476

9:30 am

EB5-3 Audio Application Programming Interface for Mixed Reality—*Rémi Audfray, Jean-Marc Jot, Sam Dicker,* Magic Leap, Inc., San Francisco, CA, USA

In mixed reality (MR) applications, digital audio objects are rendered via an acoustically transparent playback system to blend with the physical surroundings of the listener. This requires a binaural simulation process that perceptually matches the reverberation properties of the local environment, so that virtual sounds are not distinguishable from real sounds emitted around the listener. In this paper we propose an acoustic scene programming model that allows pre-authoring the behaviors and trajectories of a set of sound sources in a MR audio experience, while deferring to rendering time the specification of the reverberation properties of the enclosing room.
Engineering Brief 477

9:45 am

EB5-4 Accessible Object-Based Audio Using Hierarchical Narrative Importance Metadata—*Lauren Ward,*[1] *Ben Shirley,*[1] *Jon Francombe*[2]
[1] University of Salford, Salford, UK
[2] BBC Research and Development, Salford, UK

Object-based audio has great capacity for production and delivery of adaptive and personalizable content. This can be used to improve the accessibility of complex content for listeners with hearing impairments. An adaptive object-based audio system was used to make mix changes enabling listeners to balance narrative comprehension against immersion using a single dial. Performance was evaluated by focus groups of 12 hearing impaired participants who gave primarily positive feedback. An experienced sound designer also evaluated the function of the control and process for authoring the necessary metadata establishing that the control facilitated a clearer narrative while maintaining mix quality. In future work the algorithm, production tools, and interface will be refined based on the feedback received.
Engineering Brief 478

Audio Builders Workshop 3 — **Saturday, October 20**
9:00 am – 10:00 am — **Room 1E07**

THE STATE OF THE ART OF DO IT YOURSELF AUDIO

Presenter: **Owen Curtin**, Audio Builders Workshop, Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA

The DIY gear market has exploded and some quality pieces can now be had inexpensively if you're willing to wield a soldering iron. Audio Builder's Workshop founder Owen Curtin provides an overview of today' DIY marketplace. Audio Builder Workshop is a workgroup of the Boston AES and is hosting 7 events at the 145th Convention.

Immersive and Spatial Audio 13 — **Saturday, October 20**
9:00 am – 10:00 am — **Room 1E10**

CANCELED

Historical Event 8 — **Saturday, October 20**
9:00 am – 10:30 am — **Room 1E15+16**

SPECIAL EVENT: LES PAUL: HOW THE RECORDING WIZARD AND MUSIC ICON CHANGED THE INDUSTRY

Presenters: **Sue Baker**, Les Paul Foundation
Michael Braunstein, Les Paul Foundation
Gene Paul, G&J Audio, Union City, NJ USA

Les Paul spent his life chasing sound, a sound different from anyone else's. In his quest he developed the techniques that are used every day in the recording industry. Through video, participants will hear Les Paul explaining what motivated him to create his recording evolution.

Sue Baker, Program Director for the Les Paul Foundation, will give an overview of Les Paul's inventions and innovations including the evolution of the electric guitar and Les' recording techniques.

Grammy-winning engineer Gene Paul, Les' son, will explain the technology of his father's inventions and describe how the inventions affect today's recording technology.

Michael Braunstein, Les Paul's most recent manager and Executive Director of the Les Paul Foundation, will explain how Les' foundation is carrying on his work.

Participants will hear examples of Les Paul's recording innovations.

Les won the Honorary Member Award from the AES in 1958 for his extraordinary contributions to the art and science of our industry.

Product Development 15 — **Saturday, October 20**
9:00 am – 10:15 am — **Room 1E09**

NEXT GENERATION MATERIALS FOR AUDIO PRODUCTS

Presenter: **Michael Klasco**, Menlo Scientific Ltd., Richmond, CA, USA

The advances in materials science in the last few years has been remarkable. Product demand in all categories are driving this rapid advancement. In this session we will cover the latest in materials now being used for and being planned for the next generation audio products.

New bio-plastics are stiffer and better damped than commodity plastics result in less enclosure buzz and smoother response. Noise canceling and echo cancelers especially benefit from these materials. Nano-materials enhance water repellent surface treatments to wind screens. Ongoing development of graphene for microphones, headphones, and microspeakers promises significant performance advances. Elastic (helical conductor) audio cables provide elongation to 40%! You will see examples of the materials available so a product developer can ride the next wave of ultra-materials.

Standards Committee Meeting
SC-07-01 WORKING GROUP ON AUDIO METADATA
Saturday, October 20, 9:00 am – 10:30 am
Room 1B03

The scope of SC-07-01 includes formal descriptions and metadata sets for audio and audio-related elements useful to support audio operations based on the use of computers and computer networks.

Technical Committee Meeting
Saturday, October 20, 9:00 am – 10:00 am
Room 1B05

SIGNAL PROCESING

Recording & Production 20 **Saturday, October 20**
9:15 am – 10:00 am **Room 1E13**

GET YOUR CREDITS: THE BREAKTHROUGH IN MUSIC CREDITS, LINER NOTES AND ARTWORK

Presenter: **Gebre Waddell**, Soundways

Music credits have been a painful and costly problem for over 20 years. A series of recent breakthroughs led to a revolution in music credit standards and technology. This presentation will provide the background and skills to secure official credits for yourself and your clients. Attend this presentation and advance to the forefront of modern music crediting.

Broadcast/Online Delivery 14 **Saturday, October 20**
9:30 am – 12:30 pm **Room 2D16**

SBE CERTIFICATION EXAMS

The Society of Broadcast Engineers Certification Program, established in 1975, is a service contributing to the professional development of the broadcast engineer and advancement of the field of broadcast engineering. The SBE Program of Certification is administered by the SBE National Certification Committee, which continually develops exam questions based on changes in the industry and technology. Levels of SBE certification vary based on an indivudual's experience. Membership in the SBE is not required to hold SBE certification. Advance registration is required to take an SBE exam at the convention.

Game Audio & XR 13 **Saturday, October 20**
9:30 am – 10:30 am **Room 1E08**

GET INSIDE THE MUSIC: CONNECTING ARTISTS AND AUDIENCES THROUGH INTERACTIVE MUSIC APPS

Presenters: **Justin Paterson**, London College of Music, University of West London, London, UK
Rob Toulson, University of Westminster, London, UK

Commercial artists are nowadays empowered to invite listeners deeper inside their music through interactive music apps incorporating gameplay, stem mixers, alternative content, and production/remix features. Bjork, Peter Gabriel, and Massive Attack, for example, have all launched interactive music albums in recent years. The tutorial presenters have also been at the forefront of this movement, collaborating with record labels and music artists to evaluate the commercial and creative opportunities of interactive music.

This tutorial will present a history of interactive music and will demonstrate future audience listening-experiences. Relevant mobile programming interfaces will be discussed alongside the music production workflow for developing interactive music applications. This insight is intended to inspire developers and students to consider their own ideas for engaging listeners more actively with recorded music.

This session is presented in association with the AES Technical Committee on Recording Tech-nology and Practices

Recording & Production 21 **Saturday, October 20**
9:45 am – 11:15 am **Room 1E21**

ERA-SHIFTING RECORDING WITH DAVID KALMUSKY

Presenters: **David Kalmusky**, Addiction Sound Studios, Nashville, TN, USA
Mark Rubel, The Blackbird Academy, Nashville, TN, USA; Pogo Studio, Nashville, TN, USA

Acclaimed Nashville-based producer/engineer/artist David Kalmusky (Journey, Keith Urban, John Oates, Vince Gill, Shawn Mendes, etc.) was asked to record a version of "Amazing Grace" for the documentary film Met While Incarcerated. Kalmusky set himself a fascinating challenge: parallel recording of the same music using different eras of recording technology and musical approaches, which can be cross-faded and blended together. The song, band, and performances "time travel" between 1947 record cutting, 1962 open room recording, 1976 vari-speeded tape, 1984 synthesizers, and current technology (Pro Tools, Elastic Audio, etc.) In a far-ranging conversation with Mark Rubel, David Kalmusky will explain the various instruments, microphones, techniques and media used, and demonstrate and play the musical results of this unique process.

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

Audio Builders Workshop 4 **Saturday, October 20**
10:00 am – 11:00 am **Room 1E07**

EDUCATING AUDIO BUILDERS

Presenters: **Doug Bielmeier**, Purdue School of Engineering and Technology, IUPUI, Indianapolis, IN, USA
Owen Curtin, Audio Builders Workshop, Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA
Tom Erbe, University of California San Diego, San Diego, CA, USA
Justin Kurtz, The Hartt School, University of Hartford, West Hartford, CT, USA
Paul Lehrman, Tufts University, Medford, MA, USA

Students are learning to maintain vintage equipment and design the products of the future. Educators from an array of institutions will offer a review of their programs. Students and educators will be inspired by the wide range of possibilities. Audio Builder Workshop is a workgroup of the Boston AES and is hosting 7 events at the 145th Convention.

Technical Committee Meeting
Saturday, October 20, 10:00 am – 10:00 am
Room 1B05

SPATIAL AUDIO

Session EB6 Saturday, Oct. 20
10:15 am – 11:15 am Room 1E11

TRANSDUCERS

Chair: **Masataka Nakahara**, ONFUTURE Ltd., Tokyo, Japan

10:15 am

EB6-1 A Dante Powered Modular Microphone Array System— *Mirco Pezzoli,*[1] *Luca Comanducci,*[1] *Joe Waltz,*[2] *Anthony Agnello,*[2] *Luca Bondi,*[1] *Antonio Canclini,*[1] Augusto Sarti[1]
[1] Politecnico di Milano, Milan, Italy
[2] Eventide Inc., Little Ferry, NJ, USA

Eventide Inc. and Politecnico di Milano are collaborating to present a versatile high-performance Network Based Modular Microphone Array System (MMAS). One advantage of the system is the ability to quickly build various linear and planar array geometries with up to 64 microphone elements (sensors) connected to a single workstation, with the option to expand up to 512 sensors by synchronizing multiple sub-systems. The system modules (*eStick*) consist of a 48cm linear array made of 16 MEMS microphones with an integrated Audinate Dante Power-over-Ethernet (PoE) interface. The system is intended to support a wide range of industrial and research applications, ranging from advanced sound scene analysis and manipulation to source separation, extraction, and tracking.
Engineering Brief 479

10:30 am

EB6-2 Low-Complexity Non-Linear Loudspeaker Protection— *Daniil Sinev*, ARKAMYS, Paris, France

These days high-end loudspeakers are not seen with as much reverence as before, and the market is becoming dominated by their cheaper, more efficient counterparts. At the same time the demand for higher sound pressure levels is growing, which creates audible distortion problems. There are various DSP solutions for this, from tried-and-true fixed high pass filter, which, while getting rid of distortion, also cuts out a lot of bass frequencies, to complex new solutions based on physical models and adaptive signal processing. This paper proposes an approach that improves on the former while staying much less demanding than the latter in terms of computational power.
Engineering Brief 480
[eBrief presented by Ivan Boumeyster]

10:45 am

EB6-3 Modeling Point-Source Loudspeaker Arrays in AFMG's EASE Focus 3, and GLLViewer Programs—*D. B. (Don) Keele, Jr.,*[1] *Hugh Sarvis*[2]
[1] DBK Associates and Labs, Bloomington, IN, USA
[2] Presonus Audio Electronics-Worx Audio Technologies, Baton Rouge, LA, USA

To date there has been no easy straightforward way to model simple array configurations of point-sources such as CBT arrays for demonstration and learning purposes in AFMG's simulation and support programs EASE, Focus 3, and GLLViewer. This paper describes a pair of free AFMG-supplied GLLs for creating point-source cluster and array-type configurations. Both GLLs can be used in AFMG's EASE or the GLLViewer but only the array-type GLL can be used in EASE Focus 3. Although free to the end user, AFMG's powerful 3D direct-sound sound reinforcement line-array simulation program EASE Focus 3 is funded when manufacturers provide licensed GLL (Generic Loudspeaker Library) data for their loudspeakers so that customers can model their own branded loudspeakers in Focus 3.
Engineering Brief 481
[This eBrief was not presented but is available in the E-Library]

11:00 am

EB6-4 Influence of Horn's Surface Temperature on its Directivity Control—*Dave "Rat" Levine,*[1] *Paolo Calza,*[2] *Mario Di Cola,*[2] *Paolo Martignon,*[2] *Letizia Chisari*[2]
[1] Rat Sound Systems, Camarillo, CA, USA
[2] Contralto Audio srl, Parma, Italy

Horn directivity control and dispersion angles, especially at high frequency, have always been achieved by carefully design and optimized the horn's surface. It can be experimentally demonstrated that the control properties of constant directivity, high-frequency horns may be influenced by the horn's surface temperature which can be severely affected, for instance, by being exposed to direct sunlight. This e-brief presentation shows various experimental results that demonstrates the phenomenon and shows how much this problem can influence the directivity control at high frequency range.
Engineering Brief 482

Game Audio & XR 14 Saturday, October 20
10:15 am – 11:15 am Room 1E10

THE STANFORD VIRTUAL HEART

Chair: **Daniel Deboy**, Delta Soundworks, Germany

Panelist: *Ana Monte*, DELTA Soundworks, Germany

Pediatric cardiologists at Lucile Packard Children's Hospital Stanford are using immersive virtual reality technology to explain complex congenital heart defects, which are some of the most difficult medical conditions to teach and understand. The Stanford Virtual Heart experience helps families understand their child's heart conditions. For medical trainees, it provides an immersive and engaging new way to learn about the most common and complex congenital heart anomalies. The panelists will give an insight about the challenges for the sound design with a scientific approach and how it was integrated in Unity.

Session P16 Saturday, Oct. 20
10:30 am – 12:00 noon Poster Area

POSTERS: SPATIAL AUDIO

10:30 am

P16-1 Spatial Audio Coding with Backward-Adaptive

Singular Value Decomposition—*Sina Zamani, Kenneth Rose*, University of California Santa Barbara, Santa Barbara, CA, USA

The MPEG-H 3D Audio standard applies singular value decomposition (SVD) to higher-order ambisonics data, and divides the outcome into prominent and ambient sound components, which are then separately encoded. We recently showed that significant compression gains are achievable by moving the SVD to the frequency domain, and ensuring smooth transition between frames. Frequency domain SVD also enables SVD adaptation to frequency, but the increase in side information, to specify additional basis vectors, compromises the gains. This paper overcomes this shortcoming by introducing backward adaptive estimation of SVD basis vectors, at no cost in side information, thereby approaching the full potential of frequency domain SVD. Objective and subjective tests show considerable gains that validate the effectiveness of the proposed approach.
Convention Paper 10119

10:30 am

P16-2 Virtual Source Reproduction Using Two Rigid Circular Loudspeaker Arrays—*Yi Ren, Yoichi Haneda*, University of Electro-Communications, Chofu-shi, Tokyo, Japan

In this paper a virtual sound source reproduction method is proposed using two circular loudspeaker arrays with rigid baffles. This study aims to reproduce virtual sources in front of, or outside the loudspeaker arrays, with each array considered as an infinite-length rigid cylinder with loudspeakers attached to its surface. Transfer functions that consider the reflection between the two arrays are introduced, and the appropriate reflection times to be used in the transfer function are discussed. Using the pressure-matching method and circular harmonic expansion, several methods are proposed and compared via computer simulation
Convention Paper 10120

10:30 am

P16-3 Design and Implementation of a Binaural Reproduction Controller Applying Output Tracking Control—*Atsuro Ito,[1] Kentaro Matsui,[1] Kazuho Ono,[1] Hisao Hattori,[2] Takeaki Suenaga,[2] Kenichi Iwauchi,[2] Shuichi Adachi[3]*
[1] NHK Science & Technology Research Laboratories, Tokyo, Japan
[2] Sharp Corporation, Japan
[3] Keio University, Yokohama-shi, Kanagawa, Japan

We have been studying a design method for a controller for binaural reproduction with loudspeakers. The gain of the controller amplifies errors due to external disturbances and system perturbations, and this leads to deterioration of the sound quality. Therefore, the gain should be suppressed to as low as possible. For this purpose, we formulate the design of the controller as a minimization problem of the gain, in which the H8 norm of the controller is adopted as a measure of the gain. In this article we also introduce a binaural reproduction system as an implementation example. This system virtually reproduces multichannel audio such as 22.2 multichannel audio using line array loudspeakers.
Convention Paper 10121

10:30 am

P16-4 Horizontal Binaural Signal Generation at Semi-Arbitrary Positions Using a Linear Microphone Array—*Asuka Yamazato, Yoichi Haneda*, University of Electro-Communications, Chofu-shi, Tokyo, Japan

Binaural technology using a dummy-head is a powerful technique to provide realistic sound reproduction through headphones. To obtain the binaural signals as if a listener moves around in the sound field, we need to move the dummy head. To overcome this problem, it is a promising approach to convert signals observed by a microphone array into binaural signals at arbitrary positions. In this paper we aim to reproduce horizontal binaural signals at semi-arbitrary listener's positions using linear microphone array signals based on the inverse wave propagation method with spatial over sampling and simulated head-related transfer function (HRTF) directivity pattern. We perform the computer simulation and listening experiments in a reverberant room. A listening test is performed for two cases to verify the performance of the sound localization (case-I) and distance perception (case-II). We confirm that the binaural signals obtained by the proposed method are almost expressed by the HRTF directivity pattern. We find that the angle errors of sound localization is ranged from 1.2° to 4.2° from the results of case-I. According to the results of case-II, the subjects can perceive a distance change of the virtual sound image when the auditory stimulus is white noise
Convention Paper 10122

10:30 am

P16-5 Near-Field Compensated Higher-Order Ambisonics Using a Virtual Source Panning Method—*Tong Wei,[1,2] Jinqiu Sang,[1,2] Chengshi Zheng,[1,2] Xiaodong Li[1,2]*
[1] Institute of Acoustics, Chinese Academy of Sciences, Beijing, China
[2] University of Chinese Academy of Sciences, Beijing, China

The commonly adopted higher order ambisonics (HOA) mainly concentrates on far-field sources and neglects the rendering of near-field sources. Some studies have introduced near-field compensated HOA (NFC-HOA) to preserve the original spherical wave front curvature with lots of loudspeakers. It is worthy to combine the advantages of a physical reproduction approach with a hearing-related model approach to avoid using lots of loudspeakers in regular arrangement. In this paper an all-around virtual source panning method was proposed to improve driving functions of NFC-HOA with panning functions. In this way, a near-field sound source encoded in HOA can be rendered to arbitrary arrangement of only a few loudspeakers. Both the simulation and experimental results show the validity of the proposed method.
Convention Paper 10123
[Paper presented by Chengshi Zheng]

10:30 am

P16-6 Subjective Evaluation of Virtual Room Auralization System Based on the Ambisonics Matching Projection Decoding Method—*Zhongshu Ge,[1] Yue Qiao,[1] Shusen Wang,[2] Xihong Wu,[1] Tianshu Qu[1]*
[1] Peking University, Beijing, China
[2] AES (Beijing) Science & Technology Co., Ltd., Beijing, China

Based on the higher order Ambisonics theory, a loudspeaker-based room auralization system was implemented in this paper in combination with a room acoustics computer model. In the decoding part of the Ambisonics technique, the generally used mode-matching decoding method requires a uniformly arranged loudspeaker array, which sometimes cannot be satisfied. A recently proposed method, the matching projection decoding method, which can

solve this problem, was introduced in the room auralization system to realize reproduction of room re-verberation with non-uniform loudspeaker arrays. Moreover, the performance of the matching projection method was evaluated objectively through room impulse response reconstruction analysis. Besides, the room auralization system is validated through subjective experiments.
Convention Paper 10124

10:30 am

P16-7 A Study of the Effect of Head Rotation on Transaural Reproduction—*Marcos Simón, Eric Hamdan, Dylan Menzies, Filippo Maria Fazi*, University of Southampton, Southampton, Hampshire, UK

The reproduction of binaural audio through loudspeakers, also commonly referred to as Transaural audio, allows for the rendering of immersive virtual acoustic images when the original binaural signal is accurately delivered to the listener's ears. Such accurate reproduction is generally achieved by using a network of cross-talk-cancellation filters designed for a given listener's position and orientation. This work studies the effect of small rotational movements of the listener's head on the perceived location of a virtual sound source when the binaural signal is reproduced using an array of loudspeakers. The results of numerical simulations presented in this paper describe how the perceived virtual source position is affected by the variation of the head orientation.
Convention Paper 10125

10:30 am

P16-8 A Parametric Spatial Audio Coding Method Based on Convolutional Neural Networks—*Qingbo Huang, Xihong Wu, Tianshu Qu*, Peking University, Beijing, China

The channel based 3D audio can be compressed to a downmix signal with side information. In this paper the inter-channel transfer functions (ITF) are estimated through training over fitting convolutional neural networks (CNN) on a specific frame. Perfectly reconstructing the original channel and keeping the spatial cues the same is set as the target of the estimation. By taking this approach, more accurate spatial cues are maintained. The subjective evaluation experiments were carried out on stereo signals to evaluate the proposed method.
Convention Paper 10126

Product Development 16 **Saturday, October 20**
10:30 am – 11:45 am **Room 1E09**

ADVANCED SOFT MAGNETIC MATERIALS FOR NEXT GEN AUDIO PRODUCTS

Presenter: **Md Mehedi**, Carpenter Technology Corporation, Philadelphia, PA, USA

The audio industry is experiencing fast growth in high-end devices such as wireless earphones and headphones, in-ear monitors, hearing aids, smart home assistants, and more. However, are we hitting the wall in achieving higher performance devices? The session will cover advanced magnetic materials for more efficient and more power dense electronics which can enable higher acoustic output and overall miniaturization of audio devices.

Standards Committee Meeting
SC-05-02 WORKING GROUP ON AUDIO CONNECTORS
Saturday, October 20, 10:30 am – 11:30 am
Room 1B03

The scope of SC-05-02 includes the usage, description, and contact designation for connectors for audio and ancillary functions used in professional audio recording, reproduction, and reinforcement; and the wiring among such connectors and the circuits to which they connect.

Applications in Audio 2 **Saturday, October 20**
10:45 am – 12:15 pm **Room 1E12**

THE RAVAGES OF TIME

Presenter: **Bruce Black**, MediaRooms Technology LLC, Ventura, CA, USA

It's easy to think that once we've tuned a room, all it's acoustic issues have been dealt with, and it will provide an honest and correct listening environment. Unfortunately, this is not necessarily the case.

This tutorial shows how Real Time Analyzers leave out critical information that can lead to a classic Dr. Richard Heyser situation, where the RTA measurements look good but the room sounds terrible. Unless we are aware of what an RTA DOES NOT tell us, as well as what it DOES, it's easy to get confused when the room sound does not correlate to what the RTA display says it should be. This can leave us confused and unable to make the proper corrections to our rooms.

This presentation includes actual case histories where causes and remedies are explored in detail.

Recording & Production 22 **Saturday, October 20**
10:45 am – 1:00 pm **Room 1E06**

PLANNING FOR ON-LOCATION AUDIO RECORDING AND PRODUCTION (INCLUDING SURROUND)

Presenters: **Alex Kosiorek**, Central Sound at Arizona PBS, Phoenix, AZ, USA
Steve Remote, Aurasonic
Corey Schreppel, Minnesota Public Radio|American Public Media
George Wellington, New York Public Radio, New York, NY, USA
Eric Xu, Central Sound at Arizona PBS, Phoenix, AZ, USA

Artists and engineers are recording more productions on-location or in locations other than the studio. Whether it's audio for spoken word, chorus, small chamber ensembles to large symphony orchestra, or complex jazz/pop/rock shows involving splits from FOH, pre-production is a critical aspect of any remote recording. Moreover, with new forms of immersive delivery on the horizon, surround production is now part of this equation. Some of the challenges for on-location include maintaining a consistent aesthetic across productions, varying venue acoustics, discretion of microphone placement, monitoring, and redundant backup systems. In this workshop, today's working professionals will give relatable and practical methods of tackling production for mobile/on-location events and discuss how it differs from studio recording. Such topics include venue scoping, bids and quotes, stage plots, communication with venue or ensemble production managers, talent coordination, and other logistics. Some audio examples (including those in surround) will be included.

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

Special Event
SE15: PLATINUM MASTERING—PAST, PRESENT, FUTURE: CHANGES IN AUDIO MASTERING TECHNOLOGY/AESTHETICS

Saturday, October 20, 10:45 am – 12:15 pm
Room 1E15+16

Presenters: **Bob Ludwig**, Gateway Mastering Studios, Inc., Portland, ME, USA
Andres A. Mayo, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina
Ron McMaster, Capitol Studios Mastering, Hollywood, CA, USA

The world of mastering changed dramatically in the past 10+ years due to two facts:

—Music production became a global business, with modern productions involving far more virtual collaborations than physical meetings

—Streaming changed the rules again and led artists to produce singles, much more than entire albums.

So now, mastering just one song for a producer on the other side of the world that you will never meet in person became regular business. How do mastering engineers face these completely new challenges? How do they stand up in the myriad of mastering studios worldwide offering similar services in online platforms? What are the new parameters to consider in order to be competitive?

Student Events/Career Development
EC14: STUDENT DELEGATE ASSEMBLY MEETING—PART 2
Saturday, October 20, 10:45 am – 12:15 pm
Room 1E13

Moderator: **Kyle P. Snyder**, Ohio University, School of Media Arts & Studies, Athens, OH, USA

Presenters: *Justin Chervony*, McGill University, Montreal, Quebec, Canada
Bartlomiej Chojnacki, AGH University of Science and Technology, Cracow, Poland; Mega-Acoustic, Kepno, Poland
Brecht De Man, Birmingham City University, Birmingham, UK
Mitchell Graham, University of Michigan, Ann Arbor, MI, USA
Nyssim Lefford, Luleå University of Technology, Luleå, Sweden
Maryam Safi, Hamburg, Germany

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 so don't miss this important opportunity to represent your section! Judges' comments and awards will be presented for the Recording Competitions and Design Competitions. Plans for future student activities at local, regional, and international levels will be summarized.

Audio Builders Workshop 5 **Saturday, October 20**
11:00 am – 12:15 pm **Room 1E07**

CODE IT YOURSELF: DIY-DSP

Presenters: **Owen Curtin**, Audio Builders Workshop, Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA
Charlie DeVane, MathWorks, Natick, MA, USA; UMass Lowell, Lowell, MA, USA
Tom Erbe, University of California San Diego, San Diego, CA, USA
Brewster LaMacchia, Clockworks Signal Processing LLC, Andover, MA, USA
Jeff Snyder, Princeton University, Princeton, NJ, USA

Programming custom plugins or embedding your code into hardware is now accessible to those with almost no coding experience. Customizing your audio gear is no longer limited to the analog domain; by using DSP (Digital Signal Processing) functions devices can be created that aren't possible any other way. We'll look at a number of topics, including some short demonstrations of some current hardware and corresponding software tools. Audio Builder Workshop is a workgroup of the Boston AES and is hosting 7 events at the 145th Convention.

Archiving/Restoration 8 **Saturday, October 20**
11:30 am – 12:00 noon **Room 1E10**

AUSTIN CITY LIMITS ARCHIVE: APPLYING SOFTWARE DEVELOPMENT METHODOLOGIES TO AUDIOVISUAL ARCHIVING

Presenter: **Amanda Moore**, KLRU Austin Public Television, Austin, TX, USA

Software development and audiovisual archiving are surprisingly similar in many ways. Specifications, file naming, hierarchies, relational materials, metadata, data management, discoverability, storage, use cases, strict standards, best practices, and 3rd party requirements are applicable to both fields and can be applied quite similarly in a lot of cases. Correspondingly, project management between the two fields isn't much different. Using the Austin City Limits Archive as a case study, this discussion will cover how these two fields intersect, what methods can be shared and how software project management tools, such as Agile, can be applied successfully in tracking and managing an audiovisual archive project to completion.

Immersive and Spatial Audio 14 **Saturday, October 20**
11:30 am – 12:30 pm **Room 1E10**

3D AUDIO PHILOSOPHIES & TECHNIQUES FOR COMMERCIAL MUSIC

Presenter: **Bt Gibbs**, Skyline Entertainment and Publishing, Morgan Hill, CA, USA; Tool Shed Studios, Morgan Hill, CA, USA

As 3D Audio (360 Spatial) grows, the majority of content remains in the animated VR world. Commercial audio (in all genres) continues to be delivered across streaming and download platforms in L+R stereo audio. With the binaural delivery options for spatial audio rapidly improving, commercial audio options are being underserved. The ability for commercial artists to deliver studio quality audio (if not MQA) to consumers with an "in-the-studio" experience is at hand. This presentation will demonstrate studio sessions delivered in 360 video and 360 audio, which was simultaneously captured for standard stereo delivery through traditional streaming and download sites. All of this being delivered in a simultaneous, and rapid, turn around from pre-production to final masters delivered on both 360 and stereo platforms.

Standards Committee Meeting
AESSC PLENARY
Saturday, October 20, 12:00 noon – 2:00 pm
Room 1B03

Summaries of all the individual working group meetings are presented.

Special Event
SE16: LUNCHTIME KEYNOTE: PLEASE MAKE MY JOB EASIER
Saturday, October 20, 12:30 pm – 1:30 pm
Room 1E15+16

Presenter: **Greg Wells**, Producer, mixer, songwriter, USA

What makes a device an intuitive tool instead of a burdensome product you have to deal with to try and get the results you are looking for, for your creative project? Some tools inspire artists, engineers or producers to do or try new things, others can make your life difficult.

Multiple GRAMMY-nominated producer, musician, and mixing engineer Greg Wells (Adele, OneRepublic, Keith Urban, Katy Perry, Twenty One Pilots) addresses this topic from both a creative and technical standpoint in this lunchtime keynote address

Product Development 17 **Saturday, October 20**
1:00 pm – 2:15 pm **Room 1E09**

THE CURRENT STATE AND FUTURE OF HIGH RESOLUTION AUDIO VIA BLUETOOTH

Presenter: **Jay Yoo**, Radsone Inc., Seongnam-si, Gyeonggi-do, Korea

This session is about the latest trends in high resolution Bluetooth audio and key technology elements in the fast growing Bluetooth audio market. It also covers practical design guidelines and there will be an audio demo during the session.

Sound quality was the top ranking purchase factor when choosing wireless headphones or speakers according to recent market reports. Bluetooth audio market is increasing every year, the high quality sector is growing even faster. Since 2016, 24 bit audio compression technologies for Bluetooth have been introduced which include Qualcomm aptX HD and Sony LDAC. To make hi-fi Bluetooth products, developers should consider other features like the external audio DACs, analog volume control, balanced output, algorithmic enhancements in addition to adopting 24bit Bluetooth audio codecs. Well designed high resolution Bluetooth products already show better than CD quality and in some use cases better than wired analog from smartphones. Recently there have been technological improvements to make it happen.

The Audio industry is transitioning from 16 bit to 24 bit high resolution in many audio products and services—sources, distributions, players, earphones / headphones. Bluetooth is ready for primetime in world of Hi Res, come find out why!

Immersive and Spatial Audio 15 **Saturday, October 20**
1:15 pm – 3:15 pm **Room 1E06**

3D AUDIO ACOUSTIC RECORDING CAPTURE, DISSEMINATION, AND PERCEPTION

Presenters: **David Bowles**, Swineshead Productions LLC, Berkeley, CA, USA
Paul Geluso, New York University, New York, NY, USA
Hyunkook Lee, University of Huddersfield, Huddersfield, UK
Agnieszka Roginska, New York University, New York, NY, USA

A panel discussion on recording techniques for capturing and disseminating 3D acoustic music recordings, with an emphasis on psychoacoustic and perceptual challenges. The 90-minute panel will be immediately followed by a 90-minute playback session. Q&A will be during the panel, but after the playback [in order to allow audience to experience playback without interruptions]

Session P17 **Saturday, Oct. 20**
1:30 pm – 3:30 pm **Room 1E11**

SEMANTIC AUDIO

Chair: **Rachel Bittner**, New York University, New York, NY, USA

1:30 pm

P17-1 Audio Forensic Gunshot Analysis and Multilateration— *Robert C. Maher, Ethan Hoerr*, Montana State University, Bozeman, MT, USA

This paper considers the opportunities and challenges of acoustic multilateration in gunshot forensics cases. Audio forensic investigations involving gunshot sounds may consist of multiple simultaneous but unsynchronized recordings obtained in the vicinity of the shooting incident. The multiple recordings may provide information useful to the forensic investigation, such as the location and orientation of the firearm, and if multiple guns were present, addressing the common question "who shot first?" Sound source localization from multiple recordings typically employs time difference of arrival (TDOA) estimation and related principles known as multilateration. In theory, multilateration can provide a good estimate of the sound source location, but in practice acoustic echoes, refraction, diffraction, reverberation, noise, and spatial/temporal uncertainty can be confounding.
Convention Paper 10100

2:00 pm

P17-2 Speech Classification for Acoustic Source Localization and Tracking Applications Using Convolutional Neural Networks—*Jonathan D. Ziegler*,[1,2] *Andreas Koch*,[1] *Andreas Schilling*[2]
[1] Stuttgart Media University, Stuttgart, Germany
[2] Eberhard Karls University Tübingen, Tübingen, Germany

Acoustic Source Localization and Speaker Tracking are continuously gaining importance in fields such as human computer interaction, hands-free operation of smart home devices, and telecommunication. A set-up using a Steered Response Power approach in combination with high-end professional microphone capsules is described and the initial processing stages for detection angle stabilization are outlined. The resulting localization and tracking can be improved in terms of reactivity and angular stability by introducing a Convolutional Neural Network for signal/noise discrimination tuned to speech detection. Training data augmentation and network architecture are discussed; classification accuracy and the resulting performance boost of the entire system are analyzed.
Convention Paper 10101

2:30 pm

P17-3 Supervised Source Localization Using Spot Microphones—*Miguel Ibáñez Calvo, Maria Luis Valero, Emanuël A. P. Habets*, International Audio Laboratories Erlangen, Erlangen, Germany

Spatial microphones are used to acquire sound scenes, while spot microphones are commonly used to acquire individual sound sources with high quality. These recordings are essential when producing spatial audio upmixes. However, to automatically create upmixes, or to assist audio engineers in creating these, information about the position of the sources in the scene is also required. We propose a supervised sound source localization method to estimate the direction-of-arrival (DOA) of several simultaneously active sound sources in reverberant and noisy environments that utilizes several spot microphones and a single spatial microphone. The proposed method employs system identification techniques to estimate the relative impulse responses between each spot microphone and the spatial mi-

crophone from which the DOAs can be extracted.
Convention Paper 10102

3:00 pm

P17-4 Multichannel Fusion and Audio-Based Features for Acoustic Event Classification—*Daniel Krause, Konrad Kowalczyk*, AGH University of Science and Technology, Kraków, Poland

Acoustic event classification is of interest for various audio applications. The aim of this paper is to investigate the usage of a number of speech and audio based features in the task of acoustic event classification. Several features that originate from audio signal analysis are compared with features typically used in speech processing such as mel-frequency cepstral coefficients (MFCCs). In addition, the approaches to fuse the information obtained from multichannel recordings of an acoustic event are investigated. Experiments are performed using a Gaussian mixture model (GMM) classifier and audio signals recorded using several scattered microphones.
Convention Paper 10103

Session P18 Saturday, Oct. 20
1:30 pm – 4:00 pm Room 1E12

SPATIAL AUDIO—PART 2 (EVALUATION)

Chair: **Jonas Braasch**, Rensselear Polytechnic Institute, Troy, NY, USA

1:30 pm

P18-1 Prediction of Binaural Lateralization Percepts from the Coherence Properties of the Acoustic Wavefield—*Mark F. Bocko, Steven Crawford, Michael Heilemann*, University of Rochester, Rochester, NY, USA

A framework is presented that employs the space-time coherence properties of acoustic wavefields to compute features corresponding to listener percepts in binaural localization. The model employs a short-time windowed cross-correlator to compute a sequence of interaural time differences (ITDs) from the binaurally-sampled acoustic wavefield. The centroid of the distribution of this sequence of measurements indicates the location of the virtual acoustic source and the width determines the perceived spatial extent of the source. The framework provides a quantitative method to objectively assess the performance of various free-space and headphone-based spatial audio rendering schemes and thus may serve as a useful tool for the analysis and design of spatial audio experiences in VR/AR and other spatial audio systems.
Convention Paper 10127

2:00 pm

P18-2 Influence of Visual Content on the Perceived Audio Quality in Virtual Reality—*Olli Rummukainen,[1] Jing Wang,[2] Zhitong Li,[2] Thomas Robotham,[1] Zhaoyu Yan,[2] Zhuoran Li,[2] Xiang Xie,[2] Frederik Nagel,[1] Emanuël A. P. Habets[1]*
[1] International Audio Laboratories Erlangen, Erlangen, Germany
[2] Beijing Institute of Technology, Beijing, China

To evoke a place illusion, virtual reality builds upon the integration of coherent sensory information from multiple modalities. This integrative view of perception could be contradicted when quality evaluation of virtual reality is divided into multiple uni-modal tests. We show the type and cross-modal consistency of visual content to affect overall audio quality in a six-degrees-of-freedom virtual environment with expert and naïve participants. The effect is observed both in their movement patterns and direct quality scores given to three real-time binaural audio rendering technologies. Our experiments show that the visual content has a statistically significant effect on the perceived audio quality.
Convention Paper 10128

2:30 pm

P18-3 HRTF Individualization: A Survey—*Corentin Guezenoc,[1,2] Renaud Seguier[1]*
[1] Centrale-Supélec, Rennes, France
[2] 3D Sound Labs, Rennes, France

The individuality of head-related transfer functions (HRTFs) is a key issue for binaural synthesis. While, over the years, a lot of work has been accomplished to propose end-user-friendly solutions to HRTF personalization, it remains a challenge. In this article we establish a state-of-the-art of that work. We classify the various proposed methods, review their respective advantages and disadvantages, and, above all, methodically check if and how the perceptual validity of the resulting HRTFs was assessed.
Convention Paper 10129

3:00 pm

P18-4 Spatial Auditory-Visual Integration: The Case of Binaural Sound on a Smartphone—*Julian Moreira,[1,2,3] Laetitia Gros,[1] Rozenn Nicol,[1] Isabelle Viaud-Delmon[3]*
[1] Orange Labs, Lannion, France
[2] CNAM, Paris, France
[3] IRCAM, Paris, France

Binaural rendering is a technology for spatialized sound that can be advantageously coupled with the visual of a mobile phone. By rendering the auditory scene out of the screen, all around the user, it is a potentially powerful tool of immersion. However, this audio-visual association may lead to specific perception artifacts. One of them is the ventriloquist effect, i.e., the perception of a sound and an image as they come from the same location, while they are actually at different places. We investigate the conditions of this effect to occur using an experimental method called Point of Subjective Spatial Alignment (PSSA). Given the position of a visual stimulus, we determine the integration window, i.e., the range of locations in the horizontal plane where auditory stimulus is perceived as matching the visual stimulus location. Several parameters are varied: semantic type of the stimuli (neutral or meaningful) and sound elevation (same elevation as the visual or above subject's head). Results reveal the existence of an integration window in all cases. But, surprisingly, the sound is attracted by the visual as located in the virtual scene, rather than its real location on screen. We interpret it as a mark of immersion. Besides, we observe that integration window is not altered by elevation, provided that stimuli are semantically meaningful.
Convention Paper 10130

3:30 pm

P18-5 Online vs. Offline Multiple Stimulus Audio Quality Evaluation for Virtual Reality—*Thomas Robotham, Olli Rummukainen, Jürgen Herre, Emanuël A. P. Habets*, International Audio Laboratories Erlangen, Erlangen, Germany

Virtual reality technology incorporating six degrees -of-freedom introduces new challenges for the evaluation of audio quality. Here, a real-time "online" evaluation platform is proposed, allowing multiple stimulus comparison of binaural renderers within the virtual environment, to perceptually evaluate audio quality. To evaluate the sensitivity of the platform, tests were conducted using the online platform with audiovisual content, and two traditional platforms with pre-rendered "off-line" audiovisual content. Conditions employed had known relative levels of impairments. A comparison of the results across platforms indicates that only the proposed online platform produced results representative of the known impaired audio conditions. Off-line platforms were found to be not sufficient in detecting the tested impairments for audio as part of a multi-modal virtual reality environment.
Convention Paper 10131

Acoustics/Psychoacoustics 3 — Saturday, October 20
1:30 pm – 3:00 pm — Room 1E13

RECRUITING AND TRAINING PARTICIPANTS FOR LISTENING TESTS

Presenters: **Jan Berg**, Luleå University of Technology, Piteå, Sweden
Jon Francombe, BBC Research and Development, Salford, UK
Todd Welti, Harman International Inc., Northridge, CA, USA
Christer Volk, Force SenseLab, Denmark

Listening tests can produce accurate and reliable results when conducted and analyzed correctly. The participants in such tests are an important contributing factor to the quality of results that are obtained. However, recruiting, training, and maintaining a committed panel of expert listeners is challenging, and reporting often lacks detail.

In this workshop, industrial and academic experts will share best practices from their extensive experience of running tests. How can listeners be recruited and trained? How should listeners be kept motivated, and how should performance be assessed? What details should be presented when reporting results? What are the different requirements for industrial and academic research? What about ethical issues and data security?

Recording & Production 23 — Saturday, October 20
1:30 p m – 2:30 pm — Room 1E21

CREATING SOUNDS FROM SCRATCH— CREATIVE SOUND DESIGN FOR MUSIC PRODUCTION

Presenters: **Scott B. Metcalfe**, Peabody Conservatory, Johns Hopkins, Severna Park, MD, USA; Baltimore, MD, USA
Andrea Pejrolo, Berklee College of Music, Boston, MA, USA

In this hands on presentation you will learn how to create original sounds and patches in order to add a fresh touch to your productions. Based on practical examples and contemporary pro duction techniques Andrea Pejrolo and Scott Metcalfe will guide you through the theory and the practical aspects of original sound design for the modern producer and composer.

Through practical examples you will learn how to: • Choose some of the best software synthesizers for music production; • Pick the best tools for creating specific types of patches; • Create completely original sounds based on different types of synthesis such as sampling, FM, additive, granular, and more.

Audio Builders Workshop 6 — Saturday, October 20
1:45 pm – 2:45 pm — Room 1E07

DESIGN AND BUILD YOUR OWN GEAR

Presenters: **Bob Katz**, Digital Domain Mastering, Orlando, FL, USA
Joe Malone, JLM Audio, South Brisbane, Queensland, Australia

You can make your own custom gear that works and look like vintage or modern gear. Discover what it takes to design a circuit, put it in a box, and power it up. Bob Katz will talk about his DIY project, the "tube blender." Audio Builder Workshop is a workgroup of the Boston AES and its hosting 7 events at the 145th AES Convention.

Game Audio & XR 15 — Saturday, October 20
1:45 pm – 2:45 pm — Room 1E10

MIXING IN VR

Presenters: **Daniel Deboy**, DELTA Soundworks, Germany
Christian Sander, Dear Reality GmbH, Germany

Mixing audio for Virtual Reality (VR) on 2D Displays can be a frustrating job. We present a new workflow that enables the engineer to mix object based audio directly in VR without leaving the HMD. Starting with an overview of Spatial Audio workflows from recording, editing, mix-ing, platforms and playback, we'll be demoing mixing in VR live on stage.

Sound Reinforcement 8 — Saturday, October 20
1:45 pm – 3:00 pm — Room 1E08

THE AUDIO VENDOR AND BAND MIXER RELATIONSHIP

Moderator: **Keith Clark**, Live Sound International

Panelists: *Troy Clair*, Clair Global
Dave Natale, FOH mixer

From the moment both parties are confirmed for a music tour, the relationship between the audio vendor and the band's FOH mixer (and by extension, the monitor mixer) becomes one of the most important factors for the band's touring sound. Keith Clark, editor of *Live Sound International* and ProSoundWeb.com, will discuss the details and nuances of this dynamic with two luminaries of the touring music industry: Troy Clair, CEO of Clair Global and Dave Natale, long-time FOH mixer for the Rolling Stones as well as Jeff Beck, Fleetwood Mac, Tina Turner, Van Halen, and many others.

Session EB7 — Saturday, Oct. 20
2:30 pm – 4:00 pm — Poster Area

POSTERS—APPLICATIONS IN AUDIO

2:30 pm

EB7-1 Noise Reduction for Randomized Speech and Audio Coding in WASNs—*Johannes Fischer*,[1] *Tom Bäckström*[2]
[1] International Audio Laboratories Erlangen, Erlangen, Germany
[2] Aalto University, Espoo, Finland

We are surrounded by a multitude of connected devices with microphones, the signal of which should be combined for best sound quality. Thus, we recently proposed a distributed speech and audio codec that decorrelates quantization

noise applying randomization. In this paper this method is extended attenuating quantization noise using Wiener Filtering at the decoder. We demonstrate that this approach can be used to jointly attenuate quantization noise and background noise present at the microphones. By using orthogonal randomization matrices, computational complexity can be minimized by separating the Wiener Filter from the inverse randomization. Our evaluation shows that Wiener Filtering in combination with a randomized distributed codec is an efficient method to attenuate background and quantization noise at the decoder.
Engineering Brief 483

2:30 pm

EB7-2 Adaptive Ballistics Control of Dynamic Range Compression for Percussive Tracks—*Dave Moffat, Mark Sandler*, Queen Mary University of London, London, UK

Dynamic range compression (DRC) is a very commonly used audio effect. One use of DRC is to emphasize transients in an audio signal. The aim of this paper is to present an approach for automatically setting dynamic range compression timing parameters, adaptively, allowing parameters to adapt to the incoming audio signal, with the aim of emphasizing transients within percussive audio tracks. An implementation approach is presented.
Engineering Brief 484

2:30 pm

EB7-3 A Device for Measuring Auditory Brainstem Responses to Audio—*Piotr Odya, Andrzej Czyzewski, Andrzej Sroczynski, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

Standard ABR devices use clicks and tone bursts to assess subjects' hearing in an objective way. A new device was developed that extends the functionality of a standard ABR audiometer by collecting and analyzing auditory brainstem responses (ABR). The developed accessory allows for the use of complex sounds (e.g., speech or music excerpts) as stimuli. Therefore, it is possible to find out how efficiently different types of sounds are processed in the hearing system including brain. The paper contains technical details related to the design of the device, including its hardware and software parts. The test results that have been carried out to verify the operation of the device are also described.
Engineering Brief 485

2:30 pm

EB7-4 Measuring and Evaluating Excess Noise in Resistors—*Brewster LaMacchia,[1] Bradford Swanson[2]*
[1] Clockworks Signal Processing LLC, Andover, MA, USA
[2] Tufts University, Medford, MA, USA

All resistors generate white (Johnson-Nysquist) noise based on their value and temperature; they can also generate several other types of (excess) noise. The amount or characteristics of a resistor's excess noise could be one factor contributing to variations in perceived sound quality. This research explores methods for measuring the Johnson-Nysquist and excess noise of different resistors with the hopes of quantifying the performance of the components under test. A methodology is proposed for evaluating the audibility of both Johnson-Nysquist and excess noise that requires no special measurement equipment, only a sound system with suitable computer and freely available software.
Engineering Brief 486

2:30 pm

EB7-5 A New Compact 3D Reproduction System: The Tetra-Speaker—*Parichat Songmuang*, New York University, New York, NY, USA

As 3D audio becomes a prominent field of research in audio technology, researchers continue to develop reproduction methods that will best translate 3D recordings. Reproduction techniques have ranged from unique surround systems to 3D speakers such as the dodecahedron. However, these methods may be complex with its disadvantages. These systems tend to be configured for ambient or surround recordings. In this recent project, a new 3D speaker system was created not only for those surround recordings but also for individual sources. The speaker takes influence from the tetra-microphone structure. Testing for the radiation pattern has yet to be conducted. This project is an ongoing research and explores its use with different single-source recordings as well as creative "moving" audio.
Engineering Brief 487

2:30 pm

EB7-6 Music Streaming Platforms—Quality and Technical Comparison—*Pawel Malecki, Dorota Czopek, Katarzyna Sochaczewska*, AGH University of Science and Technology, Krakow, Poland

Music streaming platforms dominate the contemporary music industry. Through such platforms for a fixed monthly fee, or even for free, listeners get access to a huge online music database. In recent years many platforms have been created and they currently compete with each other. They offer different subscription prices, sound quality, and service ranges. This paper presents a comparison of the technical parameters and an auditory assessment of the differences between selected streaming platforms. A listening test containing diversified sound material was carried out using the ABX method (3IFC – 3 interval stimulus with forced choice and hidden reference). Over 50 subjects participated in the listening test. After basic statistical analysis, the results were presented in graphical form.
Engineering Brief 488

2:30 pm

EB7-7 Development of Ambisonic Microphone Design Tools—Part 1—*Charles J. Middlicott, Bruce J. Wiggins*, University of Derby, Derby, UK

In recent years an increase in the capture and production of ambisonic material has occurred as a result of companies such as YouTube and Facebook utilizing ambisonics for spatial audio playback. There is now a greater need for affordable higher order microphone arrays. This work details the development of a set of tools that can be used to simulate and evaluate such microphone arrays, The "Ambisonic Array Design Tool" for simulation and "Ambisonic Array Evaluation Tool" for evaluation. The microphone capsules' position and directivity can be changed, with the effects on the synthesized spherical harmonics frequency and polar responses observed within the GUI. These scripts written in MatLab have been packaged within a GUI and will be available online.
Engineering Brief 489

2:30 pm

EB7-8 Introducing a Dataset of Guitar Amplifier Sounds for Nonlinear Emulation Benchmarking—*Thomas Schmitz, Jean-Jacques Embrechts*, University of Liege, Liege, Belgium

Recent progresses made in the nonlinear system identification field have improved the ability to emulate nonlinear audio systems such as tube guitar amplifiers. As a straightforward comparison of different models cannot always be made, we propose a new reference dataset enabling to highlight the strengths and weaknesses of different nonlinear modeling methods. Our dataset gathers five different styles of guitar sounds passing through different guitar amplifiers with 10 steps of their gain parameter (i.e., the distortion level of the amplifier). Moreover, a Matlab function is also provided to obtain the desired input/output sounds in a matrix form.
Engineering Brief 490

Product Development 18 Saturday, October 20
2:30 pm – 3:45 pm Room 1E09

QUALITY BY DESIGN—SHIPPING ON TIME WITHOUT SACRIFICING QUALITY

Moderator: **Lisa Ferrante-Walsh**, Director of Engineering, iZotope, Cambridge, MA, USA

Panelists: *Karen Stackpole*, QA Engineer, Cinema Technology, Dolby Labs
Emily Wigley, Acoustics Engineer, Shure Incorporated
Samara Winterfeld, VP Product Management, Content, and solutions Xperi corporation

The product release date is coming up quickly—is this speeding train actually accelerating, or does it just seem as if it is? Why, again, are we trading off fixing those last critical bugs in order to meet the date? Our Product Management team wants to consider delivering incremental software value under a new subscription licensing model—how can we get there from here? Why is our manufacturing line demanding additional end of line tests for the new transducer assembly?

Are these common patterns and challenges on your product development team? This workshop will explore some of the latest thinking about building quality into the development process from the beginning—in support of newer and accelerated trends and product release models in the audio industry.

The panel for this 90-minute workshop will provide a variety of product perspectives—software, hardware, integrated software and hardware, technology licensing, cloud platforms. The panel consists of Quality and Product professionals and technologists, experts in their fields, who will share their best practices around building quality products in this evolving environment. This will be an interactive session where we encourage discussion and interchange of ideas.

Special Event
SE17: CRACKING THE CREATIVE PROCESS
Saturday, October 20, 2:45 pm – 3:45 pm
Room 1E15+16

Presenter: **Michael Beinhorn**
Frank Filipeti, The Living Room, New York, NY, USA

An exponential evolution of innovative recording technologies together with seemingly unlimited access to recorded music has generated an amazing era of music creation and consumption unlike any in history. However, with all this emphasis on technology and so much less on individual expression, we feel that something essential is getting lost in the process. Join Frank Filipetti and Michael Beinhorn for a frank, no-holds-barred conversation about the creativity crisis … and how to crack it.

Audio Builders Workshop 7 Saturday, October 20
3:00 pm – 4:00 pm Room 1E07

BUILD YOUR OWN RECORDING EQUIPMENT

Presenters: **Peterson Goodwyn**, DIY Recording Equipment, Philadephia, PA, USA
Matthew McGlynn, Microphone Parts

Learn the steps involved in building your first piece of gear. Discover what tools you need, how much you can expect to spend, what skills you need, and how to choose a good first project. Audio Builder Workshop is a workgroup of the Boston AES and is hosting 7 events at the 145th AES Convention.

Recording & Production 24 Saturday, October 20
3:00 pm – 4:00 pm Room 1E21

FROM STUDIO TO STAGE—THE IMPACT OF TECHNOLOGY ON LIVE PRODUCTION

Presenters: **Erin Barra**, Berklee College of Music, Boston, MA, USA; MAMMA BARRA
DJ Nebraska, Jessie Davis Music
DJ Raydar Ellis, Berklee College of Music
Paul “Willie Green” Womack, Willie Green Music, Brooklyn, NY, USA

Advancements in technology and software have transformed the live performances of DJs, looping artists, and producers. DAWs and samplers do not live solely in the domain of the studio, rather they have become the instruments of live performers across the globe. This round-table discussion will explore the changing workflows and evolving capabilities of technology-based artists.