AES 147TH CONVENTION PROGRAM

OCTOBER 16 – 19, 2019

JAVITS CONVENTION CENTER, NY, USA

The Winner of the 147th AES Convention Best Peer-Reviewed Paper Award is:

Sound Design and Reproduction Techniques for Co-Located Narrative VR Experiences—*Marta Gospodarek, Andrea Genovese, Dennis Dembeck, Corinne Brenner, Agnieszka Roginska, Ken Perlin*, New York University, New York, NY, USA

Convention Paper 10287

To be presented on Saturday, Oct. 19, *in Session 16—Posters: Spatial Audio*

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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 147th 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 147th AES Convention Student Paper Award is:

Effect of a Global Metronome on Ensemble Accuracy in Networked Music Performance—*Robert Hupke, Lucas Beyer, Marcel Nophut, Stephan Preihs, Jürgen Peissig*, Leibniz Universität Hannover, Hannover, Germany

Convention Paper 10218

To be presented on Wednesday, October 16, in Session 1—Applications in Audio

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Paper Session P01 — **Wednesday, Oct. 16**
9:00 am – 11:30 am — **Room 1E10**

APPLICATIONS IN AUDIO

Chair: **Kevin Bastyr**, Harman International, Novi, MI, USA

9:00 am

P01-1 Personal Sound Zones: A Comparison between Frequency and Time Domain Formulations in a Transportation Context—*Lucas Vindrola,*[1,2] *Manuel Melon,*[1] *Jean-Christophe Chamard,*[2] *Bruno Gazengel,*[1] France; *Guy Plantier*[1]
[1] LAUM, Le mans Université, Les Mans, France
[2] PSA Group, Rueil-Malmison, France

This paper compares the formulation of a least-squares pressure matching algorithm in the frequency and time domains for the generation of Personal Sound Zones (PSZ) for a transportation application. Due to variations in the transportation's acoustic environment, the calculation time is added to the usually found metrics in the PSZ bibliography (like Acoustic Contrast, Effort, etc.). Both formulations are implemented to control two zones in three configurations (4, 6, and 8 sources), using monopole simulations and anechoic measurements. In spite of not always presenting perfectly causal filters—pre-ringing in some filters occurs in some cases—the frequency domain formulation allows achieving equal levels of Acoustic Contrast, Effort, and Reproduction error more than 500 times faster than the time domain formulation.
Convention Paper 10216

9:30 am

P01-2 Mitigating the Effect of In-Vehicle Road Noise Cancellation on Music Playback—*Tao Feng, Kevin Bastyr*, Harman International, Novi, MI, USA

A Road Noise Cancellation (RNC) system is an Active Noise Cancellation (ANC) system implemented in a vehicle in order to minimize undesirable road noise inside the passenger cabin. Current RNC systems undesirably affect the frequency response of music playback. The RNC system's error microphones sense all the sound in the passenger cabin, including the music. Hence, RNC systems will cancel this total sensed sound and not only the road induced noise. A new True Audio algorithm can directly remove the music signal from the error microphone signals and

leave only the interior noise portion. In order to correctly estimate the music portion at the error microphones, True Audio implements a novel control topology based on a new multiple channel, real time modeling of the music's secondary path transfer function. To validate the effectiveness of the proposed algorithm, experimental and numerical simulations were performed. The numerical studies use logs of real sensors mounted on a vehicle forming an RNC system with six reference accelerometers, five control speakers and six error microphones. Both the models and measurements show that the True Audio algorithm preserves the frequency response of music when the RNC system is activated.
Convention Paper 10217

10:00 am

P01-3 Effect of a Global Metronome on Ensemble Accuracy in Networked Music Performance—*Robert Hupke, Lucas Beyer, Marcel Nophut, Stephan Preihs, Jürgen Peissig*, Leibniz Universität Hannover, Hannover, Germany

Several rhythmic experiments with pairs drawn from a group of 23 subjects were performed to investigate the effect of a global metronome on the ensemble accuracy in Networked Music Performance (NMP). Artificial delays up to 91 ms were inserted into the audio transmission between the subjects. To investigate the dependencies between delay times, ensemble accuracy and the highly synchronized global metronome, the experiments were evaluated in terms of tempo acceleration, imprecision and subjective judgment of the ensemble play. The results show that the global metronome leads to a stabilization of the tempo acceleration caused by the delay. The imprecision stays constant to a threshold of about 28 ms and 36 ms, depending on the delay compensating strategy the subjects used.
Convention Paper 10218

10:30 am

P01-4 Evaluation of Multichannel Audio in Automobiles versus Mobile Phones—*Fesal Toosy, Muhammad Sarwar Ehsan*, University of Central Punjab, Lahore, Pakistan

Multichannel surround and 3D audio are slowly gaining popularity and eventually commercial content in these formats will become common. Many automobiles still have a stereo sound system with some firmware or software that is capable of rendering multichannel audio into stereo. This paper shows the results of a listening test for multichannel audio conducted in a medium-sized car. The results of this test are compared to the results of a listening test for the same audio excerpts but conducted on a mobile phone with headphones. The results show that on mobile phones, multichannel audio clearly outperforms stereo in terms of perceived audio quality as rated by a user. However in automobiles, multichannel audio only shows marginal improvement in the rated audio quality.
Convention Paper 10219

11:00 am

P01-5 Realizing An Acoustic Vector Network Analyzer—*Marcus MacDonell, Jonathan Scott*, University of Waikato, Hamilton, Waikato, New Zealand

Acoustic absorption, reflection, and transmission is typically measured using an impedance tube. We present the design and initial measurements of a radically different measurement system. The instrument builds on the rich history and deep mathematics developed in pursuit of electromagnetic Vector-corrected Network Analyzers (VNAs). Using acoustic directional couplers and a traditional VNA mainframe we assembled an "Acoustic Vector Network Analyzer" (AVNA). The instrument measures acoustic scattering parameters, the complex reflection and transmission coefficients, of materials, transmission lines, ported structures, ducts, etc. After the fashion of electromagnetic VNAs we have constructed millimeter-wave measurement heads that span the 800 Hz–2200 Hz (420–150 mm) and 10 kHz–22 kHz (35–15 mm) bands, demonstrating scalability. We present initial measurement results.
Convention Paper 10220

Paper Session P02 **Wednesday, Oct. 16**
9:00 am – 12:00 noon **Room 1E11**

AUDIO SIGNAL PROCESSING

Chair: **Scott Hawley**, Belmont University, Nashville, TN, USA

9:00 am

P02-1 Analyzing and Extracting Multichannel Sound Field—*Pei-Lun Hsieh*, Ambidio, Glendale, CA, USA

Current post production workflow requires sound engineers to create multiple multichannel audio delivery formats. Inaccurate translation between formats may lead to more time and cost for extra manual adjustment; whereas in sound reproduction, it causes misinterpretation of the original mix and deviation from the intended story. This paper proposes a method that combines both analyzing an encoded Ambisonics field from the input multichannel signal and analyzing between each pair of adjacent channels. This allows an overall understanding of the multichannel sound field while having the ability to have a fine extraction from each channel pair. The result can be used to translate between multichannel formats and also to provide a more accurate rendering for immersive stereo playback.
Convention Paper 10221

9:30 am

P02-2 Profiling Audio Compressors with Deep Neural Networks —*Scott Hawley,*[1] *Benjamin Colburn,*[2] *Stylianos Ioannis Mimilakis*[3]
[1] Belmont University, Nashville, TN, USA
[2] ARiA Acoustics, Washington, DC, USA
[3] Fraunhofer Institute for Digital Media Technology (IDMT), Ilmenau, Germany

We present a data-driven approach for predicting the behavior of (i.e., profiling) a given parameterized, non-linear time-dependent audio signal processing effect. Our objective is to learn a mapping function that maps the unprocessed audio to the processed, using time-domain samples. We employ a deep auto-encoder model that is conditioned on both time-domain samples and the control parameters of the target audio effect. As a test-case, we focus on the offline profiling of two dynamic range compressors, one software-based and the other analog. Our results show that the primary characteristics of the compressors can be captured, however there is still sufficient audible noise to merit further investigation before such methods are applied to real-world audio processing workflows.
Convention Paper 10222

10:00 am

P02-3 Digital Parametric Filters Beyond Nyquist Frequency—*Juan Sierra*, Stanford University, Stanford, CA, USA

Filter Digitization through the Bilinear Transformation is often considered a very good all-around method to produce equalizer sections. The method is well behaved in terms of stability and ease of implementation; however, the frequency warping produced by the transformation leads to abnormalities near the Nyquist frequency. Moreover, it is impossible to design parametric sections whose analog center frequencies are defined above the Nyquist frequency. These filters, even with center frequencies outside of the hearing range, have effects that extend into the hearing bandwidth with desirable characteristics during mixing and mastering. Surpassing these limitations, while controlling the abnormalities of the warping produced by the Bilinear Transform through an alternative definition of the Bilinear constant is the purpose of this paper. In the process, also a correction factor is discussed for the bandwidth of the parametric section to correct abnormalities affecting the digitization of this parameter.
Convention Paper 10224

10:30 am

P02-4 Using Volterra Series Modeling Techniques to Classify Black-Box Audio Effects—*Ethan Hoerr, Robert C. Maher*, Montana State University, Bozeman, MT, USA

Digital models of various audio devices are useful for simulating audio processing effects, but developing good models of nonlinear systems can be challenging. This paper reports on the in-progress work of determining attributes of black-box audio devices using Volterra series modeling techniques. In general, modeling an audio effect requires determination of whether the system is linear or nonlinear, time-invariant or –variant, and whether it has memory. For nonlinear systems, we must determine the degree of nonlinearity of the system, and the required parameters of a suitable model. We explain our work in making educated guesses about the order of nonlinearity in a memoryless system and then discuss the extension to nonlinear systems with memory.
Convention Paper 10225

11:00 am

P02-5 Modifying Audio Signals for Reproduction with Reduced Room Effect—*Christof Faller*, Illusonic GmbH, Uster, Switzerland

Conventionally, equalizers are applied when reproducing audio signals in rooms to reduce coloration and effect of room resonances. Another approach, filtering audio signals with an inverse of the room impulse response (RIR), can theoretically eliminate the effect of the room in one point. But practical issues arise such as impaired sound at other positions, a need to update when RIRs change, and loudspeaker-challenging signals. A technique is presented, which modifies the time-frequency envelopes (spectrogram) of audio signals, such that the corresponding spectrogram in the room is more similar to the original signal's spectrogram, i.e., room effect is attenuated. The proposed technique has low sensitivity on RIR and listener position changes.
Convention Paper 10226

11:30 am

P02-6 On the Similarity between Feedback/Loopback Amplitude and Frequency Modulation—*Tamara Smyth*, University of California, San Diego, San Diego, CA, USA

This paper extends previous work in loopback frequency modulation (FM) to a similar system in which an oscillator is looped back to modulate its own amplitude, so called feedback amplitude modulation (FBAM). A continuous-time closed-form solution is presented for each, yielding greatly improved numerical properties, reduced dependency on sampling rate, and a more accurate representation of the feedback by eliminating the unit-sample delay required for discrete-time implementation. Producing similar waveforms, it is shown that FBAM for a known input frequency, is actually a scaled and offset version of loopback FM having a different carrier frequency but same sounding frequency. Two distinct representations are used to show mathematical equivalence between systems while validating the closed-form solution for each.
Convention Paper 10223

Game Audio & XR
GA01 - 4-Pi REVERB EFFECTS FOR IN-GAME SOUNDS
Wednesday, October 16, 9:00 am – 10:30 am, Room 1E06

Presenters: **Tomoya Kishi**, CAPCOM Co., Ltd., Japan
Steve Martz, THX Ltd., San Rafael, CA, USA
Masataka Nakahara, ONFUTURE Ltd. ,Tokyo, Japan; SONA Corp., Tokyo, Japan
Kazutaka Someya, beBlue Co., Ltd. ,Tokyo, Japan

In a video game, sound fields of virtual spaces are created in a 4-pi field which is free from channel restrictions. That is, in a video game, there are no picture frames, which cut out a part of sound fields, nor channel borders, which divide sound fields into finite numbers of areas. The workshop introduces how to create 4-pi channel-free reverberations for in-game sounds both from current and future technical points of the views. Demonstrations will be also provided, and reverberations that are generated by proposed methods will be listened to be compared with conventional reverb sounds that are created by a skilled mixing engineer.

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

Networked Audio
NA01 - SOLUTION-BASED APPROACHES FOR NETWORKED AUDIO IN LIVE PRODUCTION
Wednesday, October 16, 9:00 am – 10:30 am, Room 1E13

Moderator: **Dan Ferrisi**, Sound & Communications, Port Washington, NY, USA

Panelists: *Tim Boot*, Meyer Sound Laboratories
Genio Kronauer, L-Acoustics, Marcoussis, France
Morten Lave, Morten Lave, Toronto, ON, Canada
Patrice Lavoie, Solotech
Jamie Pollock, Kore Audio Design LLC

With AV increasingly residing on networks, engineers, end-users, integrators and others responsible for system design and implementation face new challenges. Products and systems are becoming increasingly complex with new functionality; managing networks require IT expertise; and audiences expect even greater experiences – particularly in live-sound applications where size and scale can range substantially. Organizations like the Avnu Alliance's Milan workgroup are providing solutions for networked audio that directly improve both the engineer and audience experience, with audio, video, and data able to coexist in one network via a single cable connection that does not cause interference between the data packets.

This session's panel will feature live sound industry experts and end users discussing the challenges that face AV systems engineers and others today, and how solutions for networked audio like the Milan protocol help address these needs. Technical aspects will be examined, such as scalability; how much planning is involved; do you have to be an IT expert; as well as real-world systems case studies in networked live sound applications with lessons learned shared and technical solutions discussed.

Attendees will leave with an understanding of the importance of solutions-based networked audio, and how to integrate technology like Milan into their systems, how to build it into the design and layout on an AVB network, and the benefits it brings to both the integrator or sound engineer, as well as the audience.

Product Development
PD01 - THE PREDICTABLE HORN: THE WHY, HOW, AND WHEN OF HORN SIMULATION
Wednesday, October 16, 9:00 am – 10:30 am, Room 1E09

Presenters: **Mark Dodd**, Celestion, Woodbridge, Suffolk, UK
Bjørn Kolbrek, Celestion, Ipswich, UK
David Murphy, Krix Loudspeakers, Henley Beach, South Australia, Australia
Joerg Panzer

In the "good old days," horns were designed to have an exponential wave front area expansion, and otherwise a geometry that "looked good." Today numerical simulation of directivity and frequency response has become an important tool in the horn loudspeaker design process. What simulation methods exist, and what are their pros and cons? How does the chosen simulation method influence the work flow? And is simulation really necessary, or can horns with the desired properties be designed using experience, rules of thumb, and cut and try methods? In this workshop we will take a closer look at these questions.

Student Events & Career Development
SC00 - RESUME REVIEW (FOR STUDENTS, RECENT GRADS, AND YOUNG PROFESSIONALS)
Wednesday, October 16, 9:00 am – 5:00 pm, SDA Booth

Moderator: **Alex Kosiorek**, Central Sound at Arizona PBS, Phoenix, AZ, USA

Students, recent graduates and young professionals… Often your resume is an employer's first impression of you. Naturally, you want to make a good one. Employer's often use job search websites to search for candidates. Some use automated software to scan your resume and in some cases, your LinkedIn/social media profiles as well. Questions may arise regarding formatting, length, keywords and phrases so it shows up in searches and lands on the desk of the hiring manager. No matter how refined your resume may be, it is always good to have someone else review your materials. Receive a one-on-one 20-25 minute review of your resume from a hiring manager who is in the audio engineering business. Plus, if time allows, your cover letter and online presence will be reviewed as well.

Sign up at the student (SDA) booth immediately upon arrival. For those who would like to have your resume reviewed on Wednesday, October 17th prior to SDA-1, please email the request to: aesresumereview@outlook.com. You may be requested to upload your resume prior to your appointment for review. Uploaded resumes will only be seen by the moderator and will be deleted at the conclusion of the 147th Pro Audio Convention.

This review will take place during the duration of the convention by appointment only.

AES Standards Meeting
SC-02-12-R TASK GROUP ON STREAMING METADATA
Wednesday, October 16, 9:00 am – 10:30 am, Room 1C03

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.

Game Audio & XR
GA02 - ABBEY ROAD SPATIAL AUDIO FORUM— MUSIC PRODUCTION IN VR AND AR
Wednesday, October 16, 9:15 am – 10:45 am, Room 1E08

Chair: **Gavin Kearney**, University of York, York, UK

Panelists: *Stephen Barton*, Afterlight Inc.
Etienne Corteel, L-Acoustics, Marcoussis, France
Oliver Kadel, 1.618 Digital, London, UK; University of West London, London, UK
Muki Kulhan, Muki International, UK
Hyunkook Lee, University of Huddersfield, Huddersfield, UK
Mirek Stiles, Abbey Road Studios, London, UK

Virtual and augmented reality offers a new platform for the creation, production, and consumption of immersive music experiences. Immersive technologies now have the power to create experiences that transform how we experience music, from transporting the listener to the original recording studio in VR or even bringing the musicians to their own living room in an AR scenario. However, the creation of such audio experiences has many challenges. With different parts of the immersive audio production chain being developed by various third parties, there is a danger of confusing the producer/musician and perhaps scaring off talent before we even get off the ground. Can we do better? What are the barriers and how can they be broken down? What are the strengths and weak-nesses of existing tools? Can we achieve better clarity in the different formats that are available, and should we move towards standardization? In this open panel discussion of the Abbey Road Spatial Audio Forum, we will be looking at workflow challenges for recording, mixing, and distributing music for VR and AR.

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

Historical Event
H01 - INTEGRATING HISTORY INTO THE MODERN AUDIO CURRICULUM
Wednesday, October 16, 9:15 am – 10:15 am, Room 1E12

Chair: **Scott Burgess**, University of Colorado Denver, Denver, CO, USA

Panelists: *Gabe Herman*, The University of Hartford, The Hartt School, Hartford, CT, USA
Susan Schmidt Horning, St. John's University
Jessica Thompson, Jessica Thompson Audio, Berkeley, CA, USA

Audio engineers know the value of an old microphone and understand the uses of classic equipment and techniques. However, many current audio students still need to be connected with the rich history of our craft. This panel of experienced educators will discuss how to incorporate history into the curriculum of audio schools. Several approaches to a stand-alone history class will be dis-cussed, as well as methods of including history in survey courses. Among these are the use, maintenance, and repair of historical equipment; examination of documents relating to audio histo-ry; preservation and restoration of older recordings; and utilization of recording techniques from bygone days. Our ultimate goal is to inspire students to take this history to heart by incorporating it into their present-day careers.

Recording & Production
RP01 - MINIATURE MICROPHONES—
WHY ARE THEY SO ATTRACTIVE?
Wednesday, October 16, 9:15 am – 10:45 am, Room 1E21

Chair: **Eddy Bøgh Brixen**, EBB-consult, Smørum, Denmark; DPA Microphones, Allerød, Denmark

Panelists: *John Born*, Senior Product Manager, Shure, USA
Benjamin Nimkin, independent sound designer, NY, USA
Ryan Rumery, Sounddesigner & Composer, Pen o Three, Brooklyn, NY, USA

Miniature microphones are used all over the place: For stage sound, for film sound, for television, for sound effects, etc. Traditionally these fine instruments are electret condenser types. This is why it has been possible to make them small with data comparable to many of the bigger brothers and sisters. This event has a panel of experts both from the manufacturing side as well as the user side. In this workshop some of the leading manufacturers will present their unique technology and some of the top designers from the live theater/film will demonstrate how the stuff works in practical sound design. Also a look into future technologies will be given that includes binaural applications for VR/AR.

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

Student Events & Career Development
SC01 - EMPOWERING THE NEXT-GENERATION
OF AUDIO INDUSTRY LEADERS
Wednesday, October 16, 9:15 am – 10:15 am, Room 1E07

Moderator: **Jay LeBoeuf**, Executive Director at Real Industry, Lecturer at Stanford University

This panel presentation describes the initiatives of educational nonprofit, Real Industry, and 6 universities to support career discovery and career preparation of a more diverse population. Our specific goal was to design a mentorship and internship program to increase the number of women and underrepresented students discovering and entering the intersection of arts and technology.

Students at 6 universities received a 1-year mentorship and experiential opportunities program. Students in undergraduate programs often do not develop the skills that they need to succeed in the private sector. Without having years of industry experience, students are often unaware of roles, day-to-day responsibilities, skill sets, and inner-workings of companies. Since they do not know how companies work, they do not know how they can fit in, or even what job titles to apply to. We outline our research, work, and steps the audience can take to provide students with 3 critical, measurable keys to career success:

1. Inspiration. With inspiration, Students become aware of the roles, career paths, jobs, or companies in our industry.
2. Access. With access, students have access to mentors and role models, to explore and prepare for careers.
3. Experience. With small project experience (“micro-internships”), students have real-world experience and are able to translate their in-school learnings to professional skills and knowledge.

Our plans are ambitious! Over the next 3 years, this program supports 5,000 students. We describe resources that university educators can use utilize, including Real Industry’s 400+ mentors, an engaged network of companies, and organizational expertise in bridging academia and industry through events and mentorship. Speakers to include the pilot faculty from 3-5 of our audio education partners!

AES Mix with the Masters Workshops
MM01 - JOE CHICCARELLI
Wednesday, October 16, 10:00 am – 11:00 am
Mix with the Masters Workshop Stage

Paper Session P03 Wednesday, Oct. 16
10:30 am – 12:00 noon South Concourse A

POSTERS: TRANSDUCERS

10:30 am

P03-1 Acoustic Beamforming on Transverse Loudspeaker Array Constructed from Micro-Speakers Point Sources for Effectiveness Improvement in High-Frequency Range—*Bartlomiej Chojnacki, Klara Juros, Daniel Kaczor, Tadeusz Kamisinski*, AGH University of Science and Technology, Cracow, Poland

Variable directivity speaker arrays are very popular in many acoustic aspects, such as wearable systems, natural sources simulations or acoustic scanners. Standard systems constructed from traditional drivers, despite great DSP, have limited beamforming possibilities because of the very narrow directivity patterns for loudspeakers in high frequencies. This paper presents a new approach for micro-speakers array design from monopole sources, based on isobaric speaker configuration. New solutions allow to reach high efficiency in broadband frequency range keeping matrix size small. This presentation will contain an explanation of used isobaric speaker principles and comparisons between standard transverse transducers matrix and innovative point-source matrix in two configurations. Achieved results allow to improve beamforming effectiveness in a high frequency range with new driver matrix construction.
Convention Paper 10227

10:30 am

P03-2 Spherical Microphone Array Shape to Improve Beamforming Performance—*Sakurako Yazawa, Hiroaki Itou, Ken’ichi Noguchi, Kazunori Kobayashi, Noboru Harada*, NTT Media Intelligence Laboratories, NTT Corporation, Japan

A 360-degree steerable super-directional beamforming are proposed. We designed a new acoustic baffle for spherical microphone array to achieve both small size and high performance. The shape of baffle is a sphere with parabola-like depressions; therefore, sound-collection performance can be enhanced using reflection and diffraction. We first evaluated its beamforming performance through simulation then fabricated a 3D prototype of an acoustic baffle microphone array with the proposed baffle shape and compared its performance to that of a conventional spherical 3D acoustic baffle. This prototype exhibited better beamforming performance. We built microphone array system that includes the proposed acoustic baffle and a 360-degree camera, our system can pick up match sound to an image in a specific direction in real-time or after recording. We have received high marks from users who experienced the system demo.
Convention Paper 10228
Paper presented by Hiroaki Itou

10:30 am

P03-3 Infinite Waveguide Termination by Series Solution in Finite Element Analysis—*Patrick Macey*, PACSYS Limited, Nottingham, UK

The acoustics of an audio system may comprise of several

components, e.g., a compression driver producing plane waves, a transition connecting to the throat of a horn, and a cylindrical horn which is baffled at the mouth. While finite elements/boundary elements can model the entire system, it is advantageous from the design perspective to consider simplified systems. A compression driver might be used in many situations and should be designed radiating plane waves, without cross modes, into a semi-infinite tube. The pressure field in the tube can be represented by a series that is coupled to the finite element mesh by a DtN approach. The method is generalized to cater for ducts of arbitrary cross section and infinite cylindrical horns.
Convention Paper 10229

10:30 am

P03-4 Evaluating Listener Preference of Flat-Panel Loudspeakers—*Stephen Roessner, Michael Heilemann, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Three flat-panel loudspeakers and two conventional loudspeakers were evaluated in a blind listening test. Two of the flat-panel loudspeakers used in the test were prototypes employing both array-based excitation methods and constrained viscoelastic damping to eliminate modal resonant peaks in the mechanical response of the vibrating surface. The remaining flat-panel speaker was a commercially available unit. A set of 21 listeners reported average preference ratings of 7.00/10 and 6.81/10 for the conventional loudspeakers, 6.48/10 and 5.90/10 for the prototype flat-panel loudspeakers, and 2.24/10 for the commercial flat-panel speaker. The results are consistent with those given by a predictive model for listener preference rating, suggesting that designs aimed at smoothing the mechanical response of the panel lead to improved preference ratings.
Convention Paper 10230

10:30 am

P03-5 Modelling of a Chip Scale Package on the Acoustic Behavior of a MEMS Microphone—*Yafei Nie, Jinqiu Sang, Chengshi Zheng, Xiaodong Li*, Chinese Academy of Sciences, Beijing, China; University of Chinese Academy of Sciences, Beijing, China

Micro-electro-mechanical system (MEMS) microphones have been widely used in the mobile devices in recent decades. The acoustic effects of a chip scale package on a MEMS microphone needs to be validated. Previously a lumped equivalent circuit model was adopted to analyze the acoustic frequency response of the package. However, such a theoretical model cannot predict performance at relatively high frequencies. In this paper a distributed parameter model was proposed to simulate the acoustic behavior of the MEMS microphone package. The model illustrates how the MEMS microphone acoustic transfer function is affected by the size of sound hole, the volumes of the front and back chamber. This model also can illustrate the mechanical response of the MEMS microphone. The proposed model provided a more reliable way towards an optimized MEMS package structure.
Convention Paper 10231

10:30 am

P03-6 Personalized and Self-Adapting Headphone Equalization Using Near Field Response—*Adrian Celestinos, Elisabeth McMullin, Ritesh Banka, Pascal Brunet*, Samsung Research America, Digital Media Solutions, Valencia, CA USA

Variability in the acoustical coupling of headphones to human ears depends on a number of factors. Placement, size of user's head and ears, the headband and ear-pad material are all major contributors to the sound quality delivered by the headphone to the user. By measuring the transfer function from the driver terminals to a miniature microphone set near the driver inside the cavity produced by the headphone and the ear, the degree of acoustical coupling and the fundamental frequency of the cavity volume was acquired. An individualized equalization on these measurements was applied to every user. Listeners rated the personalized EQ significantly higher than a generic target response and slightly higher than the bypassed headphone.
Convention Paper 10232

10:30 am

P03-7 Applying Sound Equalization to Vibrating Sound Transducers Mounted on Rigid Panels—*Stefania Cecchi,[1] Alessandro Terenzi,[1] Francesco Piazza,[1] Ferruccio Bettarelli[2]*
[1] Universitá Politecnica della Marche, Ancona, Italy
[2] Leaff Engineering, Osimo, Italy

In recent years, loudspeaker manufacturers have proposed to the market vibrating sound transducers (also called shakers or exciters) that can be installed on a surface or a panel to be transformed in invisible speakers capable of delivering sound. These systems show different frequency behaviors mainly depending on the type and size of the surface. Therefore, an audio equalization is crucial to enhance the sound reproduction performance achieving flat frequency responses. In this paper a multi-point equalization procedure is applied to several surfaces equipped with vibrating transducers, showing its positive effect from objective and subjective point of view.
Convention Paper 10233

Immersive and Spatial Audio
IS01 - ISSP: IMMERSIVE SOUND SYSTEM PANNING. AN INTERACTIVE SOFTWARE APPLICATION AND TOOLS FOR LIVE PERFORMANCES
Wednesday, October 16, 10:30 am – 12:00 noon, Room 1E06

Presenter: **Ianina Canalis**, National University of Lanús, Buenos Aires, Argentina

Spatial audio has been gaining popularity in the area of commercial live performances, and immersive audio systems are now available for large as well as small concerts. There are a number of challenges in developing and implementing immersive and spatial audio systems; in particular the use of dedicated hardware interfaces. This workshop introduces the Immersive Sound System Panning (ISSP) software application, which allows a free choice in the position of the speakers and sound sources. ISSP has two versions; the first one processes the audio in a DiGiCo mixer and the second one, in a computer.

This software was designed with the specific aim of making it user-friendly and to make an immersive system that is more intuitive to mixing engineers, artists, and the public. The tools have been designed with the express needs of sound engineers and artists in mind, specifically as spatial sound is a new way of expressing music. The idea is that artists and the public can also take part in what happens with the audio as involving them will intensify the experience for everyone.

This workshop showcases the Immersive Sound System Panning (ISSP) application, the main features of the software and the tools that have been developed to spatialize the sound within the space. Within the workshop, the audience will be encouraged to get/use hands-on with the app and demonstrate its features, showing the intuitive nature of the app design and the ease with which sounds can be panned around the space. https://www.youtube.com/watch?v=Aw-YseB23R0

Broadcast & Online Delivery
B01 - PODCAST PRODUCTION STUDIOS
Wednesday, October 16, 10:30 am – 12:00 noon, Room 1E07

Moderators: **Romina Larregina**, WSDG Partner/Project Manager
John Storyk, Walters-Storyk Design Group, Highland, NY, USA

Panelists: *John DeLore*, Chief Engineer, Stitcher (Scripps), New York, NY, USA
Judy Elliott-Brown, Walters-Storyk Design Group
Austin Thompson, Technical Director, Gimlet Media, Brooklyn, NY, USA

2019 has proved itself a watershed year for podcasting. In addition to achieving global recognition as an influential information/entertainment format, podcasts have motivated the construction of major production facilities in New York, Los Angeles, and other major cities around the U.S. While these complex facilities would seem to be focused on traditional production/post-production services, the nature of Podcast production requires a number of purpose-built design considerations unique to the medium. This panel will explore the priorities of Gimlet Media and Stitcher, two of the largest, most successful, most sophisticated and most prominent players in the field. Parallels and contrasts in technology choices, physical design and acoustic requirements will be discussed. Floor plans and a selection of high quality photos will reveal the inner workings of both these showplace facilities.

Product Development
PD02 - DIRECTIVITY OPTIMIZATION OF A PASSIVE LOUDSPEAKER SYSTEM
Wednesday, October 16, 10:30 am – 12:00 noon, Room 1E09

Presenter: **Charles Hughes**, Excelsior Audio, Gastonia, NC, USA; AFMG, Berlin, Germany

Almost all types of loudspeaker systems will benefit from having a consistent directivity response with respect to frequency. It is an important contributor for the loudspeaker system to be able to sound good in different rooms with different acoustical properties. This is often overlooked or relegated in importance in order to give priority to the on-axis frequency response. This seems particularly true when dealing with two-way loudspeaker systems employing a passive crossover. However, this need not be the case. In this session the performance of a product recently evaluated from an overseas ODM / CM will be examined. We will detail how software modeling and simulations can be used to optimize the passive crossover to improve the directivity of the loudspeaker system through the crossover region. A single-channel of front-end DSP can then be used to equalize the overall response of the loudspeaker system. Since the directivity response is more consistent, the equalization can be more effective at improving the sound quality of the loudspeaker system.

Student Events & Career Development
SC02 - EDUCATION FORUM—
EDUCATION OUTSIDE SCHOOL
Wednesday, October 16, 10:30 am – 12:00 noon, Room 1E12

Moderator: **Magdalena Piotrowska**, Gdansk University of Technology, Poland; Hear Candy Mastering, Poland

How do we learn after we finish school? How do we get familiar with newest standards and technologies? During this year Education Forum participants are going to discuss various forms of education including online resources, summer programs, internships, self-education, MOOCs, training programs in work environment and many others. Participants are invited to bring their perspectives into the discussion. Short, ca. 1-2 min contributions from attendees are welcome.

AES Standards Meeting
SC-02-12-N TASK GROUP ON MEDIA NETWORK DIRECTORY ARCHITECTURE
Wednesday, October 16, 10:30 am – 12:00 noon, Room 1C03

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 standard that specifies directory system requirements across the range of professional audio and video applications.

AoIP Pavilion
AIP01 - INTRODUCTION TO THE AUDIO/VIDEO-OVER-IP TECHNOLOGY PAVILION
Wednesday, October 16, 10:30 am – 11:00 am
AoIP Pavilion Theater

Presenter: **Terry Holton**, AIMS Audio Group Chairman – UK, London, UK

The Audio/Video-over-IP Technology Pavilion is an initiative created by the AES in partnership with the Alliance for IP Media Solutions (AIMS). Following the success of last year's pavilion which provided valuable insights and practical information about audio over IP networking, the scope of this year's pavilion has been expanded to also include video and other related professional media networking topics. This presentation will provide an overview of the various aspects of the pavilion as well as an introduction to AES67 and its relationship to the SMPTE ST 2110 standard.

Game Audo & XR
GA03 - INTERACTIVE AR AUDIO USING SPARK
Wednesday, October 16, 10:45 am – 11:45 am, Room 1E13

Moderator: **Dren McDonald**, Facebook

Panelists: *Kristi Boeckmann*
Matt Nichols, Facebook
Grey Robbins, Facebook

In the past year, over a billion people have used SparkAR experiences (https://developers.facebook.com/videos/f8-2019/day-1-keynote/ at 1 hour 24min in). It's a re-markable number. In apps such as Messenger, Messenger Kids, Instagram, the Facebook app, and in devices such as Portal, AR is becoming mainstream. All of those experiences were generated with Spark AR, a free tool that anyone can use to develop AR effects. The Facebook AR Audio team would like to demonstrate some of our AR effects and discuss the challenges we faced in bringing our ideas to fruition. Topics within the AR audio design discussion will include dynamic music, ambience, audio triggers based on face and hand gestures, asset variation/randomization, microphone DSP capabilities, target plane tracking triggers, importing beat maps/beat detection, and javascripting. We can demonstrate the Spark audio workflow using the patch editor and also demonstrate effects that were designed using javascript. Several AR games have been developed in Spark and we'd like to show how those were created, in addition to showing off the interactive AR stories on Portal called Story Time. Spark AR is free and available on both Windows and MacOS. While there is some documentation about audio within Spark AR, we feel that by showing effects that we have built and prototypes that we are developing, we can show what is possible within Spark AR. Hopefully we can inspire others in the interactive audio community to dive into AR au-dio to see what they can create. This short talk from F8 highlights topics that we will go into a deep dive on with this talk (https://developers.facebook.com/videos/2019/ar-music/)

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

Immersive & Spatial Audio
IS02 - MUSIC PRODUCTION IN IMMERSIVE FORMATS: ALTERNATIVE PERSPECTIVES
Wednesday, October 16, 10:45 am – 12:00 noon, Room 1E17

Presenters: **Thomas Aichinger**, scopeaudio, Austria
Zachary Bresler, University of Agder, Kristiansand S, Vest-Agder, Norway
Sally Kellaway, Microsoft
Eve Klein, University of Queensland, Brisbane, Queensland, Australia
Jo Lord, University of West London, London, UK

Evidenced by the large volume of presentations on immersive audio at the previous AES conventions in Dublin and New York and the AES conferences on Immersive and Interactive Audio, Virtual and Augmented Reality, and Spatial Reproduction, 3D audio is of growing interest within our field. As Anastasia Devana of Magic Leap stated in her keynote at the IIA conference, it is the "bleeding edge" of our industry. In spatial audio, there are many competing ideas and technologies from delivery formats to production standards and aesthetics. From the perspective of music creators, the norms of music production and the terms used to describe practice are often clumsy or not helpful when applied in 3D. It is in this context that we propose this workshop on immersive music production. We will discuss several questions from the perspective of creatives in immersive and interactive music content. What are the changing ways that creators use and exploit 3D technologies? How do we describe the way that we make content for such systems? What are the practices, standards, and formats that creators use, and which ones should they use in the future? What are the interesting use-cases for 3D audio that challenge the way we think about music and audio production?

Zachary Bresler, Ph.D. fellow, University of Agder. Research explores music production in immersive formats and the staging of listeners in immersive compositional design.

Jo Lord, Ph.D. student, University of West London. Research investigates the development, practical application, and aesthetic suitability of 3D mix technique for record production.

Dr. Eve Klein, senior lecturer in music technology and popular music, University of Queensland. Currently researches within the VR/AR space, creating large-scale immersive festival experiences.

Thomas Aichinger, founder, scopeaudio. Studio specialized in sound design and post-production, focusing on spatial audio productions for VR and 360° videos.

Audio Builders Workshop
AB01 - CLASSROOM DIY PROJECTS
Wednesday, October 16, 11:00 am – 12:00 noon, Room 1E08

Moderator: **Chris Kincaid**, Audio Builders Workshop, Louisville, KY, USA

Panelists: *Charlie DeVane*, Mathworks, Natick, MA, USA; University of Massachusetts Lowell, Lowell, MA, USA
Robert-Eric Gaskell, McGill University, Montreal, QC, Canada
Paul Lehrman, Tufts University, Medford, MA, USA

Classroom-ready DIY projects that your students can successfully and affordably build. Lesson plans and slides will be available for download at audiobuildersworkshop.com

Recording & Production
RP02 - EVOLUTION OF ALBUM PRODUCTION FROM START TO FINISH
Wednesday, October 16, 11:00 am – 12:00 noon, Room 1E21

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

Panelists: *Gloria Kaba*, Little Underground
Heba Kadry, Timeless Mastering, Brooklyn, NY, USA
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

AES Mix with the Masters Workshops
MM02 - GAVIN LURSSEN & RUBEN COHEN
Wednesday, October 16, 11:00 am – 12:00 noon
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP02 - DESIGNING WITH DANTE AND AES67/SMPTE 2110
Wednesday, October 16, 11:00 am – 11:30 am
AoIP Pavilion Theater

Presenter: **Patrick Killianey**, Audinate, Buena Park, CA, USA

In September 2019, Audinate added support for SMPTE ST 2110 in Dante devices. In this presentation, Audinate will share its vision on how Dante and AES67/SMPTE 2110 can be used together in a modern studio, including Dante Domain Manager. A tour of Dante's managed and unmanaged integration with open standards will lead to a discussion of the need for network security, Layer 3 traversal, and fault monitoring in real time. All will be laid out with practical diagrams of broadcast studios from location sound, to small studios and large properties for practical application.

Live Production Stage
LS01 - TBD
Wednesday, October 16, 11:00 am – 11:45 am
Live Production Stage

Project Studio Expo Recording Stage
RS01 - SWEETWATER PRESENTS: EVERYTHING YOU NEED TO KNOW ABOUT MICROPHONES
Wednesday, October 16, 11:00 am – 11:45 am
Recording Stage

Presenter: **Lynn Fuston, Sweetwater**, Fort Wayne, IN, USA

Lynn Fuston covers the topic of microphones from the very basics to advanced applications, including a live vocal demo of different mics from Shure and Sennheiser/Neumann.

AoIP Pavilion
AIP03 - AUDIO MONITORING SOLUTIONS FOR AUDIO-OVER-IP
Wednesday, October 16, 11:30 am – 12:00 noon
AoIP Pavilion Theater

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

ST 2110 and AES67 have established audio-over-IP as the next generation standard for audio monitoring. This presentation discusses the benefits of using audio-over-IP over traditional audio monitoring methods, and why the change to audio-over-IP is happening with increasing speed. Practical case examples of using audio-over-IP in professional broadcasting as well as in AV install audio applications are presented.

Special Event
SE01 - OPENING CEREMONIES / AWARDS / KEYNOTE SPEECH
Wednesday, October 16, 12:00 pm – 1:30 pm, Room 1E15+16

Presenters: **Agnieszka Roginska**, New York University, New York, NY, USA
Valerie Tyler, College of San Mateo, San Mateo, CA, USA
Jonathan Wyner, M Works Studios/iZotope/Berklee College of Music, Boston, MA, USA; M Works Mastering
Prince Charles Alexander introducing
Grandmaster Flash, Keynote Speaker

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.

Silver Medal Award
Louis D Fielder
Marina Bosi

Distinguished Service Medal
Garry Margolis

Fellowship Award
Anthony Agnello
Martha de Fancisco
Michael Kelly
John Krivit
Jan Abildgaard Pedersen
Joshua Reiss
Takehiro Sugimoto
Terri Winston

Honorary Membership
Grand Master Flash

Board of Governers Award
Gavin Kearney
Mariana Lopez
Michael Santucci
Jim Starzynski
Jonathan Wyner

Citation
Patricia Parker

Keynote Speaker

The Keynote Speaker for the 147th Convention is Grandmaster Flash. Emerging from the South Bronx in the early 1970s, Grandmaster Flash is inarguably one of Hip Hop's original innovators. In the earliest days of the genre, he manipulated music by placing his fingers on the vinyl, perfected beat looping, and discovered many of the most iconic beats still commonly sampled today. His influence on how music is created has been profound and it's no surprise that The New York Times calls him Hip Hop's first virtuoso. The title of his address is "GRANDMASTER FLASH: EVOLUTION OF THE BEAT.

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

Immersive & Spatial Audio
IS11 - GENELEC PLAY IMMERSIVE
Wednesday, October 16, 12:00 noon – 1:00 pm, Room 1E17

A wide collection of excellent immersive music, nature and film recordings, conveyed via a point source 7.1.4 reproduction system.

AES Mix with the Masters Workshops
MM03 - TBA
Wednesday, October 16, 12:00 noon – 1:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP04 - DEPLOYING SMPTE ST 2110 IN A DISTRIBUTED CAMPUS SYSTEM
Wednesday, October 16, 12:00 noon – 12:30 pm
AoIP Pavilion Theater

Presenters: **Cassidy Lee Phillips**, Imagine Communications, Plano, TX, USA
Tony Pearson, North Carolina State University

When NC State University (NCSU) was looking to upgrade the Pro-AV gear driving its Distance Education and Learning Technology Applications (DELTA) program, they decided to future-proof their infrastructure with a standards-based approach to IP, while continuing to leverage existing SDI gear. Using SMPTE ST 2110, NCSU significantly multiplied the capacity of their fiber infrastructure — where four single-mode fiber strands had supported four HD video channels, now just two fiber strands deliver 32 bidirectional HD channels. This presentation will share the real-world experiences and lessons learned from adopting AES67 and ST 2110 to successfully deploy a first-of-its-kind inter-campus Pro-AV system.

Audio Builders Workshop Booth Talks
ABT01 - FIND YOUR DIY VOICE, HACK YOUR OWN STUFF
Wednesday, October 16, 12:00 noon – 12:30 pm, Booth 266

Presenter: **Michael Swanson**

Project Studio Expo Recording Stage
RS02 - MAKE THE MOST OF YOUR VOCALS WITH JACK JOSEPH PUIG
Wednesday, October 16, 12:00 noon – 12:45 pm
Recording Stage

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

Multi Grammy Award winner Jack Joseph Puig, has had a successful and varied career, having worked with blues legend Eric Clapton and John Mayer; with roots rock revisionists like The Black Crowes, Sheryl Crow and The Counting Crows; with pop superstars like The Goo Goo Dolls, Robbie Williams, Lady Gaga, Florence and the Machine, and The Pussycat Dolls; country artists like Keith Urban, Faith Hill, and Sugarland, indie heroes Chris Isaak, Jellyfish, Dinosaur Jr, Guided By Voices, Beck, and as well as the Black Eyed Peas to Green Day, No Doubt, 311, U2, Weezer, Fiona Apple, Klaxons, Fergie, Mary J Blidge, Panic at the Disco and The Rolling Stones. In the process of building such a catalogue, Puig has won himself a Grammy Award and a strong reputation as a sound engineer.

Software@AES
S01 - MELODYNE
Wednesday, October 16, 12:00 noon – 12:30 pm
Software@AES Pavilion

AoIP Pavilion
AIP05 - INTRODUCTION TO AES70
Wednesday, October 16, 12:30 pm – 1:00 pm
AoIP Pavilion Theater

Presenter: **Ethan Wetzell**, OCA Alliance, New York, NY, USA

AES70, also known as OCA, is an architecture for system control and connection management of media networks and devices. AES70 is capable of working effectively with all kinds of devices from multiple manufacturers to provide fully interoperable multivendor networks. This session will provide an overview of the AES70 standard, including the current and future objectives of the standard.

Software@AES
S02 - FABFILTER
Wednesday, October 16, 12:30 pm – 1:00 pm
Software@AES Pavilion

AES Mix with the Masters Workshops
MM04 - LESLIE BRATHWAITE
Wednesday, October 16, 1:00 pm – 2:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP25 - AUDIO IN ST 2110 FACILITY AND ACROSS WAN
Wednesday, October 16, 1:00 pm – 1:30 pm
AoIP Pavilion Theater

Presenter: **Andy Rayner**, Nevion

The ST2110 and NMOS standards are now maturing and in serious use globally. In an all-IP production environment, there are key requirements for the manipulation of audio within a facility. Further to this there are challenges of how essence-flow synchronization is automatically maintained. When sharing audio between facilities for remote/meshed/distributed production there are further challenges for integrity, timing and control. The presentation will look at all of these facility and wide area challenges and use recent deployments as case studies to demonstrate how these have been overcome.

Electronic Dance Music Stage
EDJ01 - ALLEN & HEATH REELOP, XONE & HERCULES PRESENTS: BRIDGING THE GAP BETWEEN HOME PRODUCTION AND THE STAGE
Wednesday, October 16, 1:00 pm – 2:00 pm
Electronic Dance Music & DJ Stage

Presenter: **Jamie Thompson**

Come learn what you will need to take your home music production to the stage and perform using tools like DJ mixers and midi controllers, as well as software programs like Ableton Live and Traktor Pro. There will be a live demonstration using all of these tools along with a Q&A segment to answer any questions you may have.

Live Production Stage
LS02 - SHURE PRESENTS: WIRELESS WORKFLOW SOFTWARE
Wednesday, October 16, 1:00 pm – 1:45 pm
Live Production Stage

RF can be your sound system's best friend, and its worst enemy. Wireless Workbench 6 is the Shure software solution that keeps the signal on your side, in every environment. In this session, the Shure product team will demonstrate how to remotely monitor and manage every piece of gear connected to your system, calculate and analyze frequencies so you can coordinate the entire show, and track RF data for later review using the Timeline feature. All this from one application. Find out how WWB6 can streamline your workflow so you can operate your wireless with more confidence.

Software@AES
S03 - ACCUSONUS
Wednesday, October 16, 1:00 pm – 1:30 pm
Software@AES Pavilion

Audio Builders Workhop
AB02 - CUSTOM CONSOLES / POWER AND GROUNDING
Wednesday, October 16, 1:15 pm – 2:15 pm, Room 1E09

Chair: **Owen Curtin**, Audio Builders Workshop, Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA

Panelists: *Eddie Ciletti*, Manhattan Sound Technicians, Inc., West Saint Paul, MN, USA
David Thibodeau, Analog Devices, Wilmington, MA, USA

After you win the auction, you will need to power up your new toy. Be careful not to release the magic smoke! Oh no, why is the noise floor louder than your lawnmower? Dave Thibodeau and Eddie Ciletti have restored more vintage gear then most of us have seen, and they will explain the ins and out of powering and grounding the projects on your bench.

Paper Session P04 **Wednesday, Oct. 16**
1:30 pm – 5:00 pm **Room 1E10**

ROOM ACOUSTICS

Chair: **David Griesinger**, David Griesinger Acoustics, Cambridge, MA, USA

1:30 pm

P04-1 Use of Wavelet Transform for the Computation of Modal Decay Times in Rooms—Roberto Magalotti,[1] Daniele Ponteggia[2]
[1] B&C Speakers S.p.A., Bagno a Ripoli (FI), Italy
[2] Audiomatica Srl, Firenze, Italy

The acoustic behavior of small rooms in the modal frequency band can be characterized by the modal decay times MT60. The paper explores a method for computing modal decay times from measurements of Room Impulse Responses (RIR) based on the wavelet transform. Once the resonance frequencies have been selected, the procedure computes a series of wavelet transforms of the Morlet type with decreasing bandwidth, exploiting the property that Morlet wavelets preserve the time history of energy decay. Then decay times can be calculated either by linear regression of the non-noisy portion of the curve or by nonlinear fitting of a model of decay plus noise. Examples of application of the method to real RIR measurements are shown.
Convention Paper 10235

2:00 pm

P04-2 What's Old Is New Again: Using a Physical Scale Model Echo Chamber as a Real-Time Reverberator—Kevin Delcourt,[1,2] Franck Zagala,[2] Alan Blum,[1] Brian F. G. Katz[2]
[1] École Nationale Supérieure Louis Lumière, Saint-Denis, France
[2] Sorbonne Université, Paris, France

This paper presents a method using physical scale models as echo chambers. The proposed framework creates a partitioned convolution engine where the convolution processing is carried out physically on an up-scaled live audio stream in the model. The resulting reverberated sound is captured and down-scaled, providing the result to the user in real-time. The scale factor can be dynamically changed to explore different room sizes and the reduced dimensions of the scale model make it a tangible reverberation tool. Scale factors up to 1:5 have been tested for full bandwidth, with higher factor possible with improved

hardware or in exchange for lowering the upper frequency range, primarily due to driver performance.
Convention Paper 10236
Presented by Brian F. G. Katz

2:30 pm

P04-3 Synthesis of Binaural Room Impulse Responses for Different Listening Positions Considering the Source Directivity—*Ulrike Sloma, Florian Klein, Stephan Werner, Tyson Pappachan Kannookadan*, TU- Ilmenau, Ilmenau, Germany

A popular goal in research on virtual and augmented acoustic realities is the implementation of realistic room acoustics and sound source characteristics. Additionally, listeners want to move around, explore the virtual or augmented environments. One way to realize position-dynamic synthesis is the use of binaural technologies on the basis of real measurements. While this approach allows to successfully reproduce the real acoustic environment, many positions need to be measured. To reduce the time effort new methods are invented to calculate binaural room impulse responses from few positions. The presented work enhances existing synthesis methods by including predefined sound source directivities into calculation of binaural room impulse responses. The results are analyzed in a physical and in a perceptive way.
Convention Paper 10237

3:00 pm

P04-4 Extracting the Fundamental Mode from Sound Pressure Measurements in an Acoustic Tube—*Joerg Panzer*, R&D Team, Salgen, Germany

Acoustic tubes are used to provide a load to loudspeakers or to measure material properties. If the wavelength is comparable to the diameter of the tube cross-modes can be excited. This paper demonstrates a method that allows to extract only the fundamental mode from the measurement of the sound-pressure response. The only requirement is the use of three microphones mounted into the sides of the tube-wall as well as a circular cross-section.
Convention Paper 10238

3:30 pm

P04-5 Accurate Reproduction of Binaural Recordings through Individual Headphone Equalization and Time Domain Crosstalk Cancellation—*David Griesinger*, David Griesinger Acoustic, Cambridge, MA, USA

We have developed software apps that allow a user to non-invasively match headphones to reproduce the identical spectrum at the eardrum as that from a frontal source. The result is correct timbre and forward localization without head tracking. In addition we have developed a non-individual crosstalk cancelling algorithm that creates virtual sound sources just outside a listener's ears. Both systems reproduce binaural recordings with startling realism. The apps enable researchers and students to hear what acoustical features are essential for clarity, proximity, and preference. Listening to any type of music with our apps is beautiful and highly engaging.
Convention Paper 10239

4:00 pm

P04-6 Concert Hall Acoustics' Influence on the Tempo of Musical Performances—*Jan Berg*, Luleå University of Technology, Piteå, Sweden

The acoustics of a concert hall is an integral and significant part of a musical performance as it affects the artistic decisions made by performer. Still, there are few systematic studies on the phenomenon. In this paper the effect of concert hall acoustics, mainly reverberation, on musical tempo for a selection of different genres and ensemble types is analyzed quantitatively. The study utilizes audio recordings made in a concert hall equipped with a movable ceiling enabling a variable volume and thus a variable reverberation time. The results show that there are cases where the tempo follows a change in acoustics as well as cases where it remains more or less unchanged.
Convention Paper 10240

4:30 pm

P04-7 Optimum Measurement Locations for Large-Scale Loudspeaker System Tuning Based on First-Order Reflections Analysis—*Samuel Moulin, Etienne Corteel, François Montignies*, L-Acoustics, Marcoussis, France

This paper investigates how first-order reflections impact the response of sound reinforcement systems over large audiences. On the field, only few acoustical measurements can be performed to drive tuning decisions. The challenge is then to select the right measurement locations so that it provides an accurate representation of the loudspeaker system response. Simulations of each first-order reflection (e.g., floor or side wall reflection) are performed to characterize the average frequency response and its variability over the target audience area. Then, the representativity of measurements performed at a reduced number of locations is investigated. Results indicate that a subset of eight measurement locations spread over the target audience area represents a rational solution to characterize the loudspeaker system response.
Convention Paper 10234

Paper Session P05 — **Wednesday, Oct. 16**
1:30 pm – 5:00 pm — **Room 1E11**

SIGNAL PROCESSING—PART 2

Chair: **Todd Welti**, Harman International Inc., Northridge, CA, USA

1:30 pm

P05-1 Nonlinear Control of Loudspeaker Based on Output Flatness and Trajectory Planning—*Pascal Brunet, Glenn S. Kubota*, Samsung Research America, Valencia, CA, USA

A loudspeaker is inherently nonlinear and produces timbre alterations, roughness, harshness, lack of clarity, and modulation noise. This may impair reproduction quality and speech intelligibility. These issues increase rapidly with high levels and especially high bass levels. Industrial design and marketing constraints demand smaller speaker systems without sacrificing sound output level. This results in higher distortion. To obtain "big bass from little boxes," an anti-distortion system is needed. We present a new approach that is based on the direct control and linearization of the loudspeaker diaphragm displacement that allows the maximization of the bass output and the minimization of the nonlinearities while keeping the diaphragm displacement within the range of safe operation.
Convention Paper 10241

2:00 pm

P05-2 Perceptual Assessment of Distortion in Low-Frequency Loudspeakers—*Louis Fielder*,[1] Michael Smithers[2]
[1] Retired, Millbrae, CA, USA
[2] Dolby Laboratories, Sydney, NSW, Australia

A perceptually-driven distortion metric for loudspeakers is proposed that is based on a critical-band spectral comparison of the distortion and noise to an appropriate masking threshold. The loudspeaker is excited by a sinewave signal composed of windowed 0.3 second bursts. Loudspeaker masking curves for sine waves between 20–500 Hz are derived from previously published ones for headphone distortion evaluation and expanded to curves at 1 decibel increments by linear interpolation and extrapolation. For each burst, the ratios of measured distortion and noise levels to the appropriate masking curve values are determined for each critical band starting at the second harmonic. Once this is done the audibility of all these contributions are combined into various audibility values.
Convention Paper 10242

2:30 pm

P05-3 Rethinking Flat Panel Loudspeakers—An Objective Acoustic Comparison of Different Speaker Categories— *Benjamin Zenker, Sebastian Merchel, M. Ercan Altinsoy*, TU Dresden, Dresden, Germany

The home entertainment market is growing, but connected devices like multi-room and streaming loudspeakers are increasingly replacing traditional audio systems. Compromises in the acoustic quality are made to satisfy additional requirements such as smaller, lighter, and cheaper products. The number of smart speakers sold suggests that the customers accept speakers with lower acoustical quality for their daily use. Concepts like soundbars aim to achieve better spatial reproduction but try to stay visually unobtrusive. Thanks to the low visual profile flat panel loudspeakers give opportunities for invisible integration. This paper presents an objective acoustic comparison of four speaker categories: smart speaker, flat panel, soundbar, and studio monitor. The comparison reveals that recent technological advances could make flat panel loudspeakers an alternative.
Convention Paper 10243

3:00 pm

P05-4 Modelling and Measurement of Nonlinear Intermodal Coupling in Loudspeaker Diaphragm Vibrations— *William Cardenas*, ORA Graphine Audio Inc., Montreal, Quebec, Canada

Accurate prediction of the nonlinear transfer response of loudspeakers in the full band is relevant to optimize the development of audio products. Small size, light, and efficient transducers require low density and thin diaphragms, which may vibrate nonlinearly even at low amplitudes impairing the sound quality. This paper proposes an extension of the existing transducer model comprising breakup modes with geometrical nonlinearities, adding the nonlinear coupling effect between the piston mode and the breakup modes responsible for large intermodulation problems. A novel measurement technique to estimate the breakup frequency modulation induced by the piston mode excursion is presented, the model is validated with measurements of harmonic and intermodulation distortion and other symptoms relevant for assessment of acoustic performance.
Convention Paper 10244

3:30 pm

P05-5 Sound Capture by Microphone Vibration inside Playback Devices—*Rivanaldo De Oliveira*, Qualcomm Technologies, Inc., San Diego, CA, USA

Integration of voice capture into devices that formerly were used only as a sound source, or that now need to include the ability to interface with a variety of cloud-provided services available to users and/or expand voice control capability to otherwise simple devices has become commonplace, and integration of multiple microphones inside a case that houses playback transducers requires careful attention to certain design aspects as will be discussed in this article where a prototype of a "Smart Speaker" will be used as an example.
Convention Paper 10245

4:00 pm

P05-6 Low Deviation and High Sensitivity—Optimized Exciter Positioning for Flat Panel Loudspeakers by Considering Averaged Sound Pressure Equalization—*Benjamin Zenker, TShanavaz Sanjay Abdul Rawoof, Sebastian Merchel, M. Ercan Altinsoy*, TU Dresden, Dresden, Germany

Loudspeaker panels represent a class of loudspeakers, whose electrical, mechanical, and acoustical properties differ completely from conventional loudspeakers. However, the acoustic properties are mostly associated with lower performance. The position of the excitation is one of the crucial parameters to optimize multiple parameters such as the frequency response in terms of linearity and sensitivity. This paper describes an approach to find the best excitation position for an exemplary distributed mode loudspeaker (DML) by considering efficiency and the averaged sound pressure equalization. An evaluation of the measured response in the horizontal plane of 25 excitation positions is presented and an optimization algorithm is used to filter every position to a certain acoustical quality standard.
Convention Paper 10246

4:30 pm

P05-7 A Comparison of Test Methodologies to Personalize Headphone Sound Quality—*Todd Welti, Omid Khonsaripour, Sean Olive, Dan Pye*, Harman International, Northridge, CA, USA

There exist many different methods to gather subjective equalization preference data from listeners. From one method to another there are generally tradeoffs between speed, accuracy, and ease of use. In this study four different types of test were compared to see which tests performed the best in each of these categories. The purpose was to select the best methods for headphone personalization applications for mobile devices. All four tests involve test subjects setting filter gain values for bass and treble shelving filters, and thus selecting their preferred response curves for listening to music through headphones. The results of each test, the time taken to complete them, and the ease of use based on a post-test questionnaire are presented.
Convention Paper 10247

Broadcast & Online Delivery
B02 - INNOVATIONS IN AUDIO PROCESSING
Wednesday, October 16, 1:30 pm – 3:30 pm, Room 1E07

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

Panelists: *Tim Carroll*, Dolby Laboratories, San Francisco, CA, USA
Steve Dove, Wheatstone

Frank Filipetti, The Living Room, New York, NY, USA
Frank Foti, The Telos Alliance, Cleveland, Ohio, USA
Robert Orban, Orban Labs Inc., Pennsauken, NJ, USA

Audio processing helps create the signature sound of a produced audio production. Compression, expansion, ATSC 3.0, equalization, and loudness will be part of the discussion of technology and techniques.

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

Game Audio & XR
GA04 - REAL-TIME MIXING AND MONITORING BEST PRACTICES FOR VIRTUAL, MIXED, AND AUGMENTED REALITY
Wednesday, October 16, 1:30 pm – 2:30 pm, Room 1E13

Moderator: **Scott Selfon**, Audio Experiences Lead, Facebook Reality Labs (Oculus Research)

Presenting sound to a person experiencing a dynamic virtual reality experience is, by definition, a just-in-time activity. How do we take advantage of more than a century of mixing and monitoring practices based on linear content—and more than 20 years of interactive game mixing—to create a coherent, believable, and emotionally satisfying soundscape for these new realities? This talk dis-cusses the current state of the art for mixing and monitoring techniques, from the actual process, to ever-evolving standards, to robustly handling the variety of authored and implemented content, playback environments, and scenarios.

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

Recording & Production
RP03 - SO UNFORGETTABLE—2 ICONIC ALBUMS FROM 2 NON-STUDIO SPACES
Wednesday, October 16, 1:30 pm – 3:00 pm, Room 1E21

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

Presenter: *Kevin Killen, New York, NY, USA*

U2's *The Unforgettable Fire*, and Peter Gabriel's *So* were released just two years apart, and Kevin Killen was part of both. These records represented major points of creative inflection for each artist. Most of the production took place outside of any traditional studio, recording U2 in a castle and Gabriel in a converted cow shed. These special projects placed unique demands on the producers and engineers to deliver. *The Unforgettable Fire* was released in 1984, *So* hit the world in 1986, and the results speak for themselves. The records have become sonic touchstones for many artists and engineers since. Interviewed by Alex Case, Kevin Killen takes us into the sessions so that we might learn from his experiences.

Sound Reinforcement
SR01 - LARGE-SCALE LOUDSPEAKER SYSTEM CALIBRATION, FROM SIMULATION-BASED DESIGN TO ONSITE TUNING
Wednesday, October 16, 1:30 pm – 3:00 pm, Room 1E12

Presenters: **Etienne Corteel**, L-Acoustics, Marcoussis, France
François Montignies, L-Acoustics, Marcoussis, France
Scott Sugden, L-Acoustics, Oxnard, CA, USA

The objective of a calibration is to ensure that the onsite performances of a loudspeaker system meet specific project requirements. It implies the optimization of the solution that was defined at the design stage, accounting for the environment in which the system was implemented. Onsite tuning is at the core of the process. It adopts a check and adjust approach, using measurements and electronic settings to reach a defined level of expectation. The calibration should not be a solution to correct major design issues, whether they come from the manufacturer or from the user. Nowadays, high-end manufacturers provide the user with loudspeaker processing presets that optimize direct-sound and multi-way crossovers. That removes a lot of the onsite optimization process that was performed by the users in the past. However, a meticulous definition of the sound system solution is still critical, especially in the physical deployment of a system. On this aspect, the use of advanced modeling and simulation software minimizes the risk for design errors. Simulation-based design also allows to anticipate most of the electronic settings to further reduce the onsite tuning time. One additional and major benefit is that a 3D simulation gives an overview of the results over the whole audience area, whereas onsite tuning needs to rely on a limited number of measurements locations, leading to high risk of wrong electronic adjustment choices. This tutorial presents a complete optimization process that starts at the design stage, planning the physical deployment parameters, the zoning groups within line source arrays, the optimal main-sub alignment location and the time-alignment of fill systems. Onsite tuning can then use measurements and critical listening to validate the planned solution and to address remaining simulation uncertainties.

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

Special Event
SE02 - MIXING & MASTERING FOR IMMERSIVE AUDIO
Wednesday, October 16, 1:30 pm – 2:30 pm, Room 1E08

Moderator: **Rafa Sardina**

Panelists: *Reuben Cohen*, Lurssen Mastering, Burbank, CA, USA
Gavin Lurssen, Lurssen Mastering, Los Angeles, CA, USA
Michael Romanowski

As the industry moves toward a more immersive environment, hear from these five experts the latest news regarding workflow, standards, and how to get the maximum immersive experience for your tracks.

Student Events & Career Development
SC03 - STUDENT RECORDING CRITIQUES
Wednesday, October 16, 1:30 pm – 2:30 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.

AES Standards Meeting
SC-04-04 WORKING GROUP ON MICROPHONE

MEASUREMENT AND CHARACTERIZATION
Wednesday, October 16, 1:30 pm – 2:30 pm, Room 1C03

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.

Software@AES
S04 - BEST SERVICE
Wednesday, October 16, 1:30 pm – 2:00 pm
Software@AES Pavilion

Immersive & Spatial Audio
IS12 - FLORIAN CAMERER TALK&PLAY
Wednesday, October 16, 2:00 pm – 3:00 pm, Room 1E17

Presenter: **Florian Camerer**, ORF - Austrian TV - Vienna, Austria; EBU - European Broadcasting Union

Having finally arrived where human beings are all the time, immersive audio recording and reproduction of sound is here to stay. Besides the ubiquitous 3D-audio bombardment of action movies, music and sound effects provide potentially more subtle but certainly not less compelling listening experiences. In the latter realm (atmosphere recording), Florian Camerer has gone the extra mile to explore the frontiers of quality for location sound setups in 3D audio. "Nothing is more practical than a good theory" is the foundation of his immersive outdoor rig. The thinking behind its dimensions will be covered, and the audience can judge themselves if the practical result holds up to the theory through the examples that will be played.

AES Mix with the Masters Workshops
MM05 - EDDIE KRAMER
Wednesday, October 16, 2:00 pm – 3:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP06 - BOLERO WIRELESS INTERCOM SYSTEM IN AES67 NETWORKS
Wednesday, October 16, 2:00 pm – 2:30 pm
AoIP Pavilion Theater

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

Riedel Communications will provide an overview of their Bolero wireless intercom system, including the newest AES67 mode that facilitates the distribution of antennas over AES67 networks. Then, after a review of how to assemble and configure a Bolero standalone system, two people will be randomly selected from the audience to compete to see who can configure a system in the shortest amount of time.

Electronic Dance Music Stage
EDJ02 - WAVES PRODUCT WORKSHOP
Wednesday, October 16, 2:00 pm – 2:45 pm
Electronic Dance Music & DJ Stage

Presenter: **Michael Pearson-Adams**, Waves, Knoxville, TN, USA

Live Production Stage
LS03 - MILAN PROTOCOL: DETERMINISTIC AV NETWORKS
Wednesday, October 16, 2:00 pm – 2:45 pm
Live Production Stage

Presenters: **Tim Boot**, Global Brand Strategist, Meyer Sound, Berkeley, CA, USA
Henning Kaltheuner, d&b audiotechnik GmbH, Backnang, Germany
Genio Kronauer, L-Acoustics, Marcoussis, France
Morten Lave, Adamson Systems Engineering

Today, interoperability between devices has become a key element of the AV industry's value proposition, decreasing the value of isolated individual products and focusing more on an integrated system. The network, which provides connectivity for individual components to work together, now becomes the grid that defines system architectures. The future of AV requires more than just connecting individual products and components together – it requires value and functionality that can only come from deep system integration. However, today planning and handling of networks requires deep IT management skills. This is not what audio and systems engineers signed up for. Created by Pro AV market leaders in Avnu Alliance, Milan is a standards-based, user-driven deterministic network protocol for professional media, that assures networked AV devices will work together at new levels of convenience, reliability and functionality. Milan combines the technical benefits of the AVB standard with Pro AV market-defined device requirements at both the network and the application layer for media streams, formats, clocking and redundancy.

Project Studio Expo Recording Stage
RS03 - WAVES PRESENTS: JC LOSADA "MR. SONIC"
Wednesday, October 16, 2:00 pm – 2:45 pm
Recording Stage

Presenter: **JC Losada**, Grammy and Latin Grammy Award winning Engineer & Producer

Juan Cristobal Losada is an accomplished songwriter, engineer, mixer, and producer, with a string of top-selling collaborations and credits including Shakira, Santana, Ricky Martin, Enrique Iglesias, and Plácido Domingo. He has a Latin Grammy award for his work on the Best Traditional Tropical Album To Beny Moré With Love by Jon Secada, and a Grammy Award for Best Tropical Latin Album category for Luis Enrique's Ciclos.

Software@AES
S05 - BITWIG
Wednesday, October 16, 2:00 pm – 2:30 pm
Software@AES Pavilion

Product Development
PD03 - PATENTS AND PRODUCT DEVELOPMENT
Wednesday, October 16, 2:30 pm – 3:45 pm, Room 1E09

Presenters: **Joseph Reid**, Perkins Coie, San Diego, CA, USA
John Strawn, S Systems, Inc., Larkspur, CA, USA
Babak Tehranchi, Perkins Coie, San Diego, CA, USA

Forewarned is forearmed. How often has a company found itself in a lawsuit regarding patents, a lawsuit that could have been avoided? The best defense is to incorporate intellectual property into every stage of the product development process: market opportunity, your competitors, concept, design, implementation, marketing, release, and protecting your inventiveness. We will cover the basics of patents including what a patent is, how you can get a patent for your work, and what happens in litigation if you are attacked or you need to protect your rights.

AES Standards Meeting
SC-04-09 WORKING GROUP ON ASSESSMENT OF ACOUSTIC ANNOYANCE
Wednesday, October 16, 2:30 pm – 4:00 pm, Room 1C03

This effort identifies practices for estimation of annoyance of manmade 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.

AoIP Pavilion
AIP07 - AES67 / ST 2110 / NMOS—AN OVERVIEW ON CURRENT SDO ACTIVITIES
Wednesday, October 16, 2:30 pm – 3:00 pm
AoIP Pavilion Theater

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

Update and report on current standardization activities, including AES67, SMPTE ST2110. Refresh / summary on commonalities and constraints between AES67 and ST2110. Brief overview on NMOS developments and activities.

Audio Builders Workshop Booth Talks
ABT02 - GET TO KNOW YOUR GEAR WITH FREE ANALYSIS SOFTWARE
Wednesday, October 16, 2:30 pm – 3:00 pm, Booth 266

Presenter: **Peterson Goodwyn**, DIY Recording Equipment, Philadephia, PA, USA

Software@AES
S06 - MAGIX
Wednesday, October 16, 2:30 pm – 3:00 pm
Software@AES Pavilion

Game Audio & XR
GA05 - SPATIAL STORYTELLING IN GAMES
Wednesday, October 16, 2:45 pm – 4:15 pm, Room 1E08

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

Panelists: *Cedric Diaz*, Senior Sound Designer, People Can Fly
Jason Kanter, Audio Director, Avalanche Studios, New York, NY, USA
Phillip Kovats, WWS Sound, Sony Interactive Entertainment
Mark Petty, Gearbox Software

In this exciting panel discussion join several industry experts, all with deep experience in authoring narrative content for spatial audio entertainment platforms, in discussing some of the incredible op-portunities and challenges of bringing stories to life using spatial elements. Our goal is to discuss the techniques, thinking, approaches, and execution of how 3D spaces interface with and infiltrate our storytelling practices. As audio directors, sound designers, mixers and storytellers, we will fo-cus on how we are able to leverage spatial audio to bring a greater level of engagement, spectacle, and immersion for audiences inside our story worlds.

Networked Audio
NA02 - AN INTRODUCTORY TUTORIAL TO AES70
Wednesday, October 16, 2:45 pm – 4:15 pm, Room 1E13

Presenters: **Jeff Berryman**, OCA Alliance
Simon Jones, Focusrite
Ethan Wetzell, OCA Alliance

AES70, also known as OCA, is an architecture for system control and connection management of media networks and devices. AES70 is capable of working effectively with all kinds of devices from multiple manufacturers to provide fully interoperable multivendor networks. In this session the presenters will provide an overview of the AES70 standard. Aimed at developers, users, and technical decision makers, members of the OCA Alliance will cover topics such as the object model, developer resources, and the current and future objectives of the standard.

Special Event
SE03 - THE LOUDNESS WAR IS OVER (IF YOU WANT IT)
Wednesday, October 16, 2:45 pm – 4:15 pm, Room 1E15+16

Moderator: **George Massenburg**, McGill University, Montreal, Quebec, Canada

Panelists: *Serban Ghenea*
Gimel "Guru" Keaton
Bob Ludwig, Gateway Mastering Studios, Inc., Portland, ME, USA
Thomas Lund, Genelec Oy, Iisalmi, Finland
Ann Mincieli, Jungle City Studios, New York, NY, USA

Now that streaming dominates the music listening landscape, it's time to revisit what loudness really is and how to manage it. Companies including Apple, YouTube, and Spotify each have their own measurement standards and loudness targets, while today's production paradigm often lacks a traditional infrastructure of project managers and gatekeepers with technical expertise. Artists and record companies—as they always have—want their songs to sound at least as loud as the ones playing before and after them. The stakes are high. So, what to do?

The truth is, we, the creators, are responsible for understanding all of the issues in the loudness discussion. No one else is going to do it. Join us for this lively and informative conversation with some of the best minds in the business who will shed light on both the current unhappy state of loudness and what creators can do to make it better.

Paper Session P06 **Wednesday, Oct. 16**
3:00 pm – 4:30 pm **South Concourse A**

POSTERS: AUDIO SIGNAL PROCESSING

3:00 pm

P06-1 Modal Representations for Audio Deep Learning—*Travis Skare, Jonathan S. Abel, Julius O. Smith, III*, Stanford University, Stanford, CA, USA

Deep learning models for both discriminative and generative tasks have a choice of domain representation. For audio, candidates are often raw waveform data, spectral data, transformed spectral data, or perceptual features. For deep learning tasks related to modal synthesizers or processors, we propose new, modal representations for data. We experiment with representations such as an N-hot binary vector of frequencies, or learning a set of modal filterbank coefficients directly. We use these representations discriminatively–classifying cymbal model based on samples–as well as generatively. An intentionally naive application of a basic modal representation to a CVAE designed for MNIST digit images quickly yielded results, which we found surprising given less prior success when using traditional representations like a spectrogram image. We discuss applications for Generative Adversarial Networks, towards creating a modal reverberator generator.
Convention Paper 10248

3:00 pm

P06-2 Distortion Modeling of Nonlinear Systems Using Ramped-Sines and Lookup Table—*Paul Mayo,*[1] *Wesley Bulla*[2]

[1] University of Maryland, College Park, MD, USA
[2] Belmont University, Nashville, TN, USA

Nonlinear systems identification is used to synthesize black-box models of nonlinear audio effects and as such is a widespread topic of interest within the audio industry. As a variety of implementation algorithms provide a myriad of approaches, questions arise whether there are major functional differences between methods and implementations. This paper presents a novel method for the black-box measurement of distortion characteristic curves and an analysis of the popular "lookup table" implementation of nonlinear effects. Pros and cons of the techniques are examined from a signal processing perspective and the basic limitations and efficiencies of the approaches are discussed.
Convention Paper 10249

3:00 pm

P06-3 An Open Audio Processing Platform Using SoC FPGAs and Model-Based Development—*Trevor Vannoy,*[1,2] *Tyler Davis,*[2] *Connor Dack,*[2] *Dustin Sobrero,*[2] *Ross Snider*[1,2]
[1] Montana State University, Bozeman, MT, USA
[2] Flat Earth Inc., Bozeman, MT, USA

The development cycle for high performance audio applications using System-on-Chip (SoC) Field Programmable Gate Arrays (FPGAs) is long and complex. To address these challenges, an open source audio processing platform based on SoC FPGAs is presented. Due to their inherently parallel nature, SoC FPGAs are ideal for low latency, high performance signal processing. However, these devices require a complex development process. To reduce this difficulty, we deploy a model-based hardware/software co-design methodology that increases productivity and accessibility for non-experts. A modular multi-effects processor was developed and demonstrated on our hardware platform. This demonstration shows how a design can be constructed and provides a framework for developing more complex audio designs that can be used on our platform.
Convention Paper 10250

3:00 pm

P06-4 Objective Measurement of Stereophonic Audio Quality in the Directional Loudness Domain—*Pablo Delgado,*[1] Jürgen Herre[1,2]
[1] International Audio Laboratories Erlangen, Erlangen, Germany
[2] Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Automated audio quality prediction is still considered a challenge for stereo or multichannel signals carrying spatial information. A system that accurately and reliably predicts quality scores obtained by time-consuming listening tests can be of great advantage in saving resources, for instance, in the evaluation of parametric spatial audio codecs. Most of the solutions so far work with individual comparisons of distortions of interchannel cues across time and frequency, known to correlate to distortions in the evoked spatial image of the subject listener. We propose a scene analysis method that considers signal loudness distributed across estimations of perceived source directions on the horizontal plane. The calculation of distortion features in the directional loudness domain (as opposed to the time-frequency domain) seems to provide equal or better correlation with subjectively perceived quality degradation than previous methods, as con?rmed by experiments with an extensive database of parametric audio codec listening tests. We investigate the effect of a number of design alternatives (based on psychoacoustic principles) on the overall prediction performance of the associated quality measurement system.
Convention Paper 10251

3:00 pm

P06-5 Detection of the Effect of Window Duration in an Audio Source Separation Paradigm—*Ryan Miller, Wesley Bulla, Eric Tarr,* Belmont University, Nashville, TN, USA

Non-negative matrix factorization (NMF) is a commonly used method for audio source separation in applications such as polyphonic music separation and noise removal. Previous research evaluated the use of additional algorithmic components and systems in efforts to improve the effectiveness of NMF. This study examined how the short-time Fourier transform (STFT) window duration used in the algorithm might affect detectable differences in separation performance. An ABX listening test compared speech extracted from two types of noise-contaminated mixtures at different window durations to determine if listeners could discriminate between them. It was found that the window duration had a significant impact on subject performance in both white- and conversation-noise cases with lower scores for the latter condition.
Convention Paper 10252

3:00 pm

P06-6 Use of DNN-Based Beamforming Applied to Different Microphone Array Configurations—*Tae Woo Kim, Nam Kyun Kim, Geon Woo Lee, Inyoung Park, Hong Kook Kim,* Gwangju Institute of Science and Tech (GIST), Gwangju, Korea

Minimum variance distortionless response (MVDR) beamforming is one of the most popular multichannel signal processing techniques for dereverberation and/or noise reduction. However, the MVDR beamformer has the limitation that it must be designed to be dependent on the receiver array geometry. This paper demonstrates an experimental setup and results by designing a deep learning-based MVDR beamformer and applying it to different microphone array configurations. Consequently, it is shown that the deep learning-based MVDR beamformer provides more robust performance under mismatched microphone array configurations than the conventional statistical MVDR one.
Convention Paper 10253

3:00 pm

P06-7 Deep Neural Network Based Guided Speech Bandwidth Extension—*Konstantin Schmidt,*[1] Bernd Edler[1,2]
[1] Friedrich-Alexander-University (FAU), Erlangen, Germany
[2] Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Up to today telephone speech is still limited to the range of 200 to 3400 Hz since the predominant codecs in public switched telephone networks are AMR-NB, G.711, and G.722 [1, 2, 3]. Blind bandwidth extension (blind BWE, BBWE) can improve the perceived quality as well as the intelligibility of coded speech without changing the transmission network or the speech codec. The BBWE used in this work is based on deep neural networks (DNNs) and has already shown good performance [4]. Although this BBWE enhances the speech without producing too many artifacts it sometimes fails to enhance prominent fricatives that can result in muffled speech. In order to better synthesize prominent fricatives the BBWE is extended by

sending a single bit of side information—here referred to as guided BWE. This bit may be transmitted, e.g., by watermarking so that no changes to the transmission network or the speech codec have to be done. Different DNN con?gurations (including convolutional (Conv.) layers as well as long short-term memory layers (LSTM)) making use of this bit have been evaluated. The BBWE has a low computational complexity and an algorithmic delay of 12 ms only and can be applied in state-of-the-art speech and audio codecs.
Convention Paper 10254

3:00 pm

P06-8 Analysis of the Sound Emitted by Honey Bees in a Beehive—*Stefania Cecchi, Alessandro Terenzi, Simone Orcioni, Francesco Piazza*, Universitá Politecnica della Marche, Ancona (AN), Italy

The increasing in honey bee mortality of the last years has brought great attention on the possibility of intensive bee hive monitoring in order to better understand the problems that are seriously affecting the honey bee health. It is well known that sound emitted inside a beehive is one of the key parameters for a non-invasive monitoring capable of determining some aspects of their condition. The proposed work aims at analyzing the bees' sound introducing features extraction useful for sound classification techniques and to determine dangerous situations. Taking into consideration a real scenario, several experiments have been performed focusing on particular events, such as swarming, to highlight the potentiality of the proposed approach.
Convention Paper 10255

3:00 pm

P06-9 Improvement of DNN-Based Speech Enhancement with Non-Normalized Features by Using an Automatic Gain Control—*Linjuan Cheng, Chengshi Zheng, Renhua Peng, Xiaodong Li*, Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, Beijing, China; University of Chinese Academy of Sciences, Beijing, China

Speech enhancement performance may degrade when the peak level of the noisy speech is significantly different from the training datasets in Deep Neural Networks (DNN)-based speech enhancement algorithms, especially when the non-normalized features are used in practical applications, such as log-power spectra. To overcome this shortcoming, we introduce an automatic gain control (AGC) method as a preprocessing technique. By doing so, we can train the model with the same peak level of all the speech utterances. To further improve the proposed DNN-based algorithm, the feature compensation method is combined with the AGC method. Experimental results indicate that the proposed algorithm can maintain consistent performance when the peak of the noisy speech changes in a large range.
Convention Paper 10256

Sound Reinforcement
SR02 - CANCELED
Wednesday, October 16, 3:00 pm – 4:30 pm, Room 1E12

Student Events & Career Development
SC04 - MIX IT! IMPROVING YOUR MIXES AND YOUR MIXING WORKFLOW
Wednesday, October 16, 3:00 pm – 4:00 pm, Room 1E17

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

Panelists: *Cosette Collier*, Middle Tennessee State University, Murfreesboro, TN, USA
Yuri Lysolvanov, Flashpoint Chicago, a campus of Columbia College Hollywood, Chicago, IL, USA

This workshop, ideal for students, new engineers, and more experienced engineers looking to speed up their workflow and productivity in studio mixing, live sound and event production, will discuss topics including:
• Starting a mix, and the key elements of a mix.
• Identifying problems or challenges, and deciding on the best solutions to potential issues.
• Best practice workflows or procedures that can be followed in order to make the process of mixing an easier, faster, and more predictable experience.
• Common mistakes to avoid.
• Gear!
• Getting the job done and finishing the mix.

AoIP Pavilion
AIP08 - INNOVATION THROUGH OPEN TECHNOLOGIES
Wednesday, October 16, 3:00 pm – 3:30 pm
AoIP Pavilion Theater

Presenter: **Nestor Amaya**, Ross Video / COVELOZ Technologies, Ottawa, ON, Canada

Where would today's Cloud, smartphones and embedded IoT engines be without Linux? And could we have gotten here with a proprietary OS such as Windows? Similarly, this presentation argues that Pro Audio/Video applications need open technologies to serve the unique needs of their users. Our markets need a fully open stack, beyond just AES67/ST 2110 transport, to reap the benefits of our investment in A/V networking technologies. Learn how NMOS, EmBER+ and similar open technologies enable you to innovate and serve your users better when compared to proprietary control systems.

AES Mix with the Masters Workshops
MM06 - SYLVIA MASSY
Wednesday, October 16, 3:00 pm – 4:00 pm
Mix with the Masters Workshop Stage

Audio Builders Workshop Booth Talks
ABT03 - DIY DSP PLATFORM—THE SHARC AUDO MODULE
Wednesday, October 16, 3:00 pm – 3:30 pm
Booth 266 (Exhibit Floor)

Presenter: **Denis Labrecque**, DeLab Consulting, Half Moon Bay, CA, USA

Live Production Stage
LS04 - DANTE FOR LIVE PRODUCTION, BROADCASTING AND RECORDING
Wednesday, October 16, 3:00 pm – 3:45 pm
Live Production Stage

Presenter: **Mike Picotte**, Producer Engineer Sweetwater Sound, Fort Wayne, IN, USA

Expert Engineer and Installer Mike Picotte will show you Dante Basics and how to avoid common mistakes. He will take a deeper dive into using Dante Control Manager, Virtual Dante Sound Card, Clocking via Dante, Switches, Peripherals that can up your game. Mike will show how to scale and systems from a Club to an Arena, and from multi building Campus to Houses of Worship. You will leave more confident and ready to move into the future of Broadcast, Live, and Recorded digital audio with Dante!

PMC Masters of Audio
PM01 - DOLBY AND PMC PRESENT: "A DUMMY'S GUIDE TO MIXING FOR DOLBY ATMOS MUSIC
Wednesday, October 16, 3:00 pm – 4:00 pm, Room 1E067

What is different compared to mixing 5.1/7.1 and what do you need to be able to mix in Dolby Atmos in your studio. With ample time for Q&A.

Project Studio Expo Recording Stage
RS04 - AMBEO: 3D AUDIO TECHNOLOGY BY SENNHEISER
Wednesday, October 16, 3:00 pm – 3:45 pm
Recording Stage

Presenter: **Greg Simon**, Sennheiser

Greg Simon from Sennheiser will discuss Ambeo technology, products and applications, including the work flow of Ambisonic recordings.

Software@AES
S07 - KILOHEARTS
Wednesday, October 16, 3:00 pm – 3:30 pm
Software@AES Pavilion

Recording & Production
RP04 - SPATIAL AUDIO MICROPHONES
Wednesday, October 16, 3:15 pm – 4:45 pm, Room 1E21

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

Presenters: *Svein Berge*, HARPEX, Berlin, Germany
Gary Elko, mh acoustics, Summit, NJ USA
Len Moskowitz, Core Sound LLC, Teaneck, NJ, USA
Tomasz Zernicki, Zylia sp. z o.o., Poznan, Poland

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 (one-point) arrays.

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

Audio Builders Workshop Booth Talks
ABT04 - OPEN SOURCE MODULAR HARDWARE FOR ANALOG AND DSP AUDIO SYSTEM BUILDING
Wednesday, October 16, 3:30 pm – 4:00 pm, Booth 266

Presenter: **Brewster LaMacchia**, Clockworks Signal Processing LLC, Andover, MA, USA

AoIP Pavilion
AIP09 - REINVENTING INTERCOM WITH SMPTE ST 2110-30
Wednesday, October 16, 3:30 pm – 4:00 pm
AoIP Pavilion Theater

Presenter: **Martin Dyster**, The Telos Alliance, Cleveland, OH USA

This presentation looks at the parallels between the emergence of audio over IP standards and the development of a product in the Intercom market sector that has taken full advantage of IP technology.

Software@AES
S08 - SONIBLE
Wednesday, October 16, 3:30 pm – 4:00 pm
Software@AES Pavilion

Broadcast & Online Delivery
B03 - ADVANCED AUDIO FOR SPORTS
Wednesday, October 16, 3:45 pm – 5:15 pm, Room 1E07

Moderator: **Peter Wharton**, Happy Robotz; SMPTE

Panelists: *Mike Babbitt*, Dolby Laboratories
Dennis Baxter, Dennis Baxter Sound, Gainesville, GA, USA
Karl Malone, NBC Universal, Stamford, CT, USA
Patrick Warrington, Lawo AG

Immersive audio is being used today in live sports broadcasts to create a more compelling experience for the viewer. Creating this immersive experience presents a number of creative and technical challenges and questions. What is the perspective of the listener – is it one as an observer, a participant, or a mix of both? How does one leverage height in a mix when there are no obvious sound cues from above? Is a multi-bus mix console sufficient for mixing immersive audio? What new skills must an A1 master with immersive audio? How do you mix for a the wide range of listening environments including immersive, 5.1, stereo, soundbars and headphones?

Consumers are also looking for a more personalized experience, one where they can choose their perspective and modify the mix accordingly. Can this be accomplished while maintaining control over the quality of the listener's experience?

These presentations and panel discussion will look at how immersive audio is being implemented today and the techniques and best practices for production that have been developed from micing and mixing through monitoring and distribution. Case studies from a number of live sports events will be used to illustrate the concepts including Olympics, NHRA and FIFA World Cup .

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

Product Development
PD04 - MICROPHONE ELECTROACOUSTICS
Wednesday, October 16, 4:00 pm – 5:30 pm, Room 1E09

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

This presentation introduces the basic concepts of microphones, principles of operation, and common applications. Equivalent circuits for condenser, dynamic moving coil, ribbon, and piezoelectric microphone technologies are described. Well-known, contemporary versions – some replicas of classic designs – are shown as examples of each type. Pressure and pressure gradient microphones are described. Various configurations of 1st order directional microphones are described and their associated directionality metrics are detailed. The response of shotgun and parabolic microphones are given. An overview of the principles of higher order directional microphones and their performance is given. Examples of special purpose devices such as boundary layer (PZM) microphones and manikins for binaural recording or electroacoustic measurements are shown. Line arrays and adaptive systems are also discussed. Preamplifiers, powering, and balanced vs. unbalanced operation of microphones are described. Measurements, standards, and specifications are explained, including: Frequency response, sensitivity, polar response, equivalent input noise, 2-tone distortion, and calibration. The effects of wind noise and stands and adaptors are also given. References for additional detailed information are also provided.

AES Standards Meeting
SC-02-01 WORKING GROUP ON DIGITAL AUDIO

MEASUREMENT TECHNIQUES
Wednesday, October 16, 4:00 pm – 5:00 pm, Room 1C03

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 includes 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.

AES Mix with the Masters Workshops
MM07 - BOB POWER
Wednesday, October 16, 4:00 pm – 5:00 pm
Mix with the Masters Workshop Stage

Project Studio Expo Recording Stage
RS05 - SOUNDGIRLS PRESENTS: WHAT IT TAKES TO HAVE A SUCCESSFUL CAREER IN AUDIO
Wednesday, October 16, 4:00 pm – 4:45 pm
Recording Stage

Moderator: **Leslie Gaston-Bird**, Mix Messiah Productions, Brighton, UK; Audio Engineering Society, London, UK

Panelists: *Karrie Keyes*, Executive Director SoundGirls.org
Piper Payne, Piper Payne, Mastering Engineer, Infrasonic Sound, San Francisco Bay Area, CA, USA
Michelle Sabolchick Pettinato, Sound Engineer- Elvis Costello, Scranton, PA USA; MixingMusicLive.com
Jessica Thompson, Jessica Thompson Audio, Berkeley, CA, USA
April Tucker
Catherine Vericolli, Fivethirteen Recording, Tempe, AZ, USA; Useful Industries, Nashville, TN, USA

Immersive & Spatial Audio
IS03 - REPRODUCTION AND EVALUATION OF SPATIAL AUDIO THROUGH SPEAKERS
Wednesday, October 16, 4:15 pm – 5:45 pm, Room 1E06

Presenters: **Juan Simon Calle Benitez**, THX Ltd., San Francisco, CA, USA
Patrick Flangan, THX Ltd.
Gavin Kearney, University of York, York, UK
Nils Peters, Qualcomm, Advanced Tech R&D, San Diego, CA, USA
Marcos Simon, AudioScenic, Southampton UK

A lot has been discussed about the reproduction of spatial audio in headphones as it is a controlled environment to generate the signals that trick our brain to believe there are sources outside of our head. Speakers are another way to reproduce spatial audio, but less development and evaluation has been done as it is harder to reproduce binaural audio without coloration to the signal. In this panel we will discuss the benefits, the challenges, and the ways we can evaluate spatial audio reproduction in speakers. We will discuss topics like crosstalk cancellation, wavefield synthesis and multichannel arrays applied to real life applications like virtual surround, virtual reality and augmented reality and how to test subjectively and objectively the different characteristics of 3D audio.

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

AoIP Pavilion
AIP10 - AES67 / ST 2110 AUDIO TRANSPORT & ROUTING (NMOS IS-08)
Wednesday, October 16, 4:00 pm – 4:30 pm
AoIP Pavilion Theater

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

This session explains the details of networked audio transport (as defined in AES67 and ST 2110) and how connections among devices with dedicated channel mapping utilizing NMOS IS-08 can be established.

Live Production Stage
LS05 - AUDIO CODEC TECHNOLOGY FOR RELIABLE TRANSPORT OVER UNRELIABLE NETWORKS— IP AUDIO IN BROADCAST APPLICATIONS
Wednesday, October 16, 4:00 pm – 4:45 pm
Live Production Stage

IP Audio Codec technology for reliable transport over unreliable networks is a class designed to educate attendees about the world of IP Audio Codecs for Broadcast Applications. This class will give attendees the foundations of IP Audio, IP Audio Transport over a LAN / WAN, as well as the keys to reliable audio transport for Broadcast-Critical applications.

Key topics will include:

• Relationship between network speed, audio compression, and latency

• Choosing the right audio compression algorithm for your given application

• Error correction technology available to insure secure and reliable audio transport

• Key technologies available for broadcasting Dante, MADI, AES 67, AES/EBU using your existing internet connection

• Cost-effective solutions available for Remote Broadcasting, STL, SSL, DVB Audio, and Web Radio

This presentation will look into the advantages each technology has to offer, what they are, where and how they are used, and answer any questions the participants have.

Software@AES
S09 - ACON DIGITAL
Wednesday, October 16, 4:00 pm – 4:30 pm
Software@AES Pavilion

Student Events & Career Development
SC05 - STUDENT DELEGATE ASSEMBLY, PART 1
Wednesday, October 16, 4:15 pm – 5:30 pm, Room 1E13

The first Student Delegate Assembly (SDA) meeting is the official opening of the Convention's student program which includes the SDA Keynote 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 19.

Game Audio & XR
GA06 - SIMULATING REAL WORLD ACOUSTIC PHENOMENA: FROM GRAPHICS TO AUDIO
Wednesday, October 16, 4:30 pm – 5:30 pm, Room 1E08

Presenter: **Christophe Tornieri**, Audiokinetic

Simulating real world acoustic phenomena in virtual environments drastically enhances immersion. However, computing high order reflections and convincing diffraction towards more dynamic and realistic audio scenes is a challenging task to achieve in real-time. In this talk we propose an ap-proach derived from modern 3D graphic rendering techniques used in CGI. We will first introduce the general concepts behind ray tracing and stochastic methods, then present the adaptation of these techniques to spatial audio, focusing on how to compute reflections and diffraction while maintain-ing time and spatial coherence.

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

Recording & Production
RP05 - 3D MICROPHONE TECHNIQUE SHOOTOUT—9.1 DEMO AND DISCUSSION
Wednesday, October 16, 4:30 pm – 6:00 pm, Room 1E06

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

3D audio is rapidly becoming an industry standard for film, music, virtual reality, etc. Various types of 3D microphone arrays have been proposed over the recent years, and it is important to understand the perceptual differences among different techniques. This workshop presents a large-scale 3D mic array shootout recording session that has been conducted in St. Paul's concert hall in Huddersfield, UK. An open-access database for research and education has been created from this session. A total of 104 channels of audio were recorded simultaneously for 7 different main array configurations for 9.1 (a.k.a. 5.1.4) reproduction (OCT, 2L-Cube, PCMA v1&2, Decca Tree, Hamasaki-Cube v1&2), a 32-channel spherical microphone array, a First-Order Ambisonics microphone and a dummy head. The microphones used for the main arrays were from the same manufacturer to minimize spectral differences. Various sound sources including string quartet, piano trio, a cappella singers, organ, clarinet, piano, etc., were recorded. This workshop will explain the basic psychoacoustic principles of the arrays used, with accompanying 9.1 demos of the recordings and pictures. The pros and cons of each technique depending on the type of sound source will be discussed in depth.

Special Event
SE04 - THE MAKING OF SHERYL CROW'S "THREADS"
Wednesday, October 16, 4:30 pm – 5:30 pm, Room 1E15+16

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

Panelists: *Dave O'Donnell*, Mixer/Engineer
TBA

An analysis of the making of the new Sheryl Crow record *Threads* featuring a star-studded cast of talented artists.

AoIP Pavilion
AIP11 - TECHNICAL: SYNCHRONIZATION & ALIGNMENT (ST 2110 / AES67)
Wednesday, October 16, 4:30 pm – 5:00 pm
AoIP Pavilion Theater

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

A deep dive into the secrets behind timing & synchronization of AES67 & ST2110 streams. The meaning of PTP, media clocks, RTP, synchronization parameters (SDP) and the magic of stream alignment will be unveiled in this compact presentation.

Audio Builders Workshop Booth Talks
ABT05 - HEALING POWER OF DIY GEAR
Wednesday, October 16, 4:30 pm – 5:00 pm, Booth 266

Presenter: **Buddy Lee Dobberteen**

Software@AES
S10 - FL STUDIO
Wednesday, October 16, 4:30 pm – 5:00 pm
Software@AES Pavilion

Acoustics & Psychoacoustics
AP01 - NEW DEVELOPMENTS IN ACOUSTIC SIMULATION: REFLECTIONS ON YEAR ONE IN AN ACOUSTIC LAB (PREDICTING AUDIO ACCURACY WITH ALGORITHMS & VIRTUAL REALITY)
Wednesday, October 16, 4:45 pm – 5:45 pm, Room 1E12

Presenters: **Renato Cipriano**, WSDG Director of Design
Peter D'Antonio, WSDG, Walters Storyk Design Group
Dirk Noy, WSDG, Basel, Switzerland

In 2018 prototype Acoustic Lab listening rooms were constructed in WSDG offices in Basel, Switzerland, Berlin, Germany, and Highland, NY. Over the past eighteen months projected acoustic properties of a series of diverse environments ranging from a sports stadium to theaters, classrooms, a railway terminal, and a sophisticated high end listening room were among those simulated in these Labs. This tutorial will review the evolution of methodology for developing accurate audio and 3D VR room modeling simulations of pre-construction acoustic properties. Programs include audio and visual simulations illustrating various acoustic options resulting from the engagement of a variety of acoustic treatments and speaker placement configurations. New developments in software, client response and completed projects will be discussed.

Historical Event
H02 - BUILDING SUCCESS THROUGH YOUR TEAM—VIEWS FROM THE FIRST WOMAN BROADCAST ENGINEER
Wednesday, October 16, 5:00 pm – 6:00 pm, Room 1E21

Presenter: **Pamela Gibson**

Pamela Gibson was the first female broadcast engineer. She will discuss how she built success through her team, including her experience being the first woman in broadcast engineering and how she got the job. Also, she will discuss her success at team building and attracting the best crew, who have moved on to become stars in their respective careers, and their unique ability to exceed their goals and propel her to exceed her limits. She will also touch on the subject of changing professions mid career, since it is never too late to start over and build another career.

AES Mix with the Masters Workshops
MM08 - CHRIS LORD-ALGE
Wednesday, October 16, 5:00 pm – 6:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP12 - ST 2110 ENABLED CENTRALIZED PRODUCTION
Wednesday, October 16, 5:00 pm – 5:30 pm
AoIP Pavilion Theater

Presenter: **Lucas Zwicker**, Lawo AG

While traditional OB trucks still play a major role in broadcast, more and more companies rely on centralized workflows and remote production. How can ST 2110 help to enable these workflows? And why does it help to streamline REMI procedures? Let us find out about the building blocks of this approach based on a real life example.

Live Production Stage
LS06 - LAWO PRESENTS: MIXING MUSIC FOR TV BROADCAST
Wednesday, October 16, 5:00 pm – 5:45 pm
Live Production Stage

Presenter: **Josiah Gluck**

Josiah Gluck, Emmy-award winning sound engineer, takes a look at the special demands and challenges of mixing live music for broadcast. Josiah will explain the process of creating some of the most iconic and recognizable TV live music shows including Saturday Night Live – and how to turn inspiration into success.

Josiah Gluck just started his 28th season as Co-Music Engineer for "Saturday Night Live." He received an Emmy for his music mixing work on the SNL 40th Anniversary Show. He was previously nominated for an Emmy for music mixing on the 2nd live "30Rock" show broadcast from Studio 8H. Also recipient of 3 Grammy nominations for engineering, Josiah has been the producer and/or engineer on over 200 CDs for artists such as Karrin Allyson, Kevin Eubanks, Dave Grusin, Diane Schuur, Nnenna Freelon, Curtis Stigers, Patti Austin, B.B. King, Billy Cobham, Dave Stryker. He's also engineered numerous CDs for USAF Bands all over the country. Additional television work includes "Night Music," "The Rosie O'Donnell Show," "Late Night with Conan O'Brien," "Last Call with Carson Daly," and "The Tonight Show" and "Christmas In Rockefeller Center."

Project Studio Expo Recording Stage
RS06 - KIMBRA - TECHNOLOGY FACILITATES CREATIVITY
Wednesday, October 16, 5:00 pm – 5:45 pm
Recording Stage

Presenters: **Kimbra**
Chris Tabron

Acclaimed singer/songwriter Kimbra talks with producer Chris Tabron about the creative process in recording.

Product Development
PD05 - VENDOR EVENT 1: AUDIO PRECISION
Wednesday, October 16, 5:30 pm – 6:00 pm, Demo Room 1E01

Broadcast & Online Delivery
B04 - AN INTIMATE EVENING WITH TESLA AND TWAIN
Wednesday, October 16, 7:00 pm and 9:00 pm
The Radio Waves Building (49 West 27th Street, NYC)
Penthouse

Presented by the HEAR Now Festival in tandem with SueMedia Radio Waves Studios at 7:00 pm and 9:00 pm. Tickets are required for each performance; free with your badge. Limited to 35 people per performance

The unique relationship between the two will be examined in this evening that will feature Twain artist Robert Alvey (also a recently retired EPA Scientist, www.gcnews.com/articles/rob-alvey-how-i-became-mark-twain/) in conversation with the great Nicola Tesla, played P.J. Ochlan.

According to the science blog It's "Okay to be Smart," Tesla had a bout of severe illness in the 1870s before his emigration to the U.S. His condition was serious enough that his doctors thought he might not survive. Since there was relatively little else he could do during that period of time, he read all the books he could from the local library. Among those books were several volumes of Twain's earlier works, which Tesla described as "so captivating as to make me utterly forget my hopeless state." He went on to say that those books may have been the reason for his recovery. It would be 25 years before the men met, but meet they did in NYC — and when Tesla told Twain about his illness and the role Twain's writing played in his recovery, Twain was moved to tears. The Irish Times reports that the writer and the inventor became friends in the 1890s. Tesla was living in New York, and even though Twain and his family lived in Europe at the time, Twain was a frequent traveler to New York. Twain became interested in Tesla and his work after hearing about an electric motor the scientist had invented while working for Westinghouse.

This performance piece will focus on a number of Tesla's inventions including the Tesla motor which interested Twain. According to their writings, Twain recognized that Tesla's motor was a better than a model his partner, James Paige, had come up with, which Twain had been thinking about investing in. Tesla advised against Paige's motor, but Twain still invested in another of Paige's machines – a mechanical typesetter – and took significant losses.

"An Intimate Evening with Tesla and Twain" will be a wonderful opportunity to meet two of the men who helped to build the foundation of the communication industry with their inventions and writings.

Special Event
SE00 - DIVERSITY & INCLUSION COCKTAIL PARTY
Wednesday, October 16, 5:30 pm – 7:00 pm
Houndstooth Pub, 520 8th Avenue @ 37th Street

The AES Diversity & Inclusion Committee has been set up to acknowledge, celebrate and encourage diversity within the audio engineering community. This informal meet-and-greet is a fantastic opportunity to schmooze with our committee and to meet one another.

Paper Session P07 **Thursday, Oct. 17**
9:00 am – 12:00 noon **Room 1E10**

PERCEPTION

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

9:00 am

P07-1 A Binaural Model to Estimate Room Impulse Responses from Running Signals and Recordings—*Jonas Braasch, David Dahlbom, Nate Keil*, Rensselaer Polytechnic Institute, Troy, NY, USA

A binaural model is described that can use a multichannel signal to robustly localize a sound source in the presence of multiple reflections. The model also estimates a room impulse response from a running multichannel signal, e.g., from a recording, and determines the spatial locations and delays of early reflections, without any prior or additional knowledge of the source. A dual-layer cross-correlation/autocorrelation algorithm is used to determine the interaural time difference (ITD) of the direct sound source component and to estimate a binaural activity pattern. The model is able to accurately localize broadband signals in the presence of real room reflections.
Convention Paper 10257

9:30 am

P07-2 Describing the Audible Effects of Nonlinear Loudspeaker Distortion—*Elisabeth McMullin, Pascal Brunet, Zhongran Wang*, Samsung Research America, Valencia, CA, USA

In order to evaluate how and when listeners hear distortion in a nonlinear loudspeaker model, a three-part study was designed. A variety of audio files were processed

through both a linear and a nonlinear loudspeaker model and the input signals were calibrated to produce a prescribed level of distortion in the nonlinear model. Listeners completed subjective experiments in which they heard both versions of the clips, selected the audible attributes they believed changed, and described the differences in their own words. In later tests, listeners marked in time they heard changes in the most commonly used descriptors. A full analysis of listener comments and time-based relationships is explored with theoretical explanations of the results obtained.
Convention Paper 10258

10:00 am

P07-3 Spatial Auditory Masking for Three-Dimensional Audio Coding—*Masayuki Nishiguchi, Kodai Kato, Kanji Watanabe, Koji Abe, Shouichi Takane*, Akita Prefectural University, Yurihonjo, Akita, Japan

Spatial auditory masking effects have been examined for developing highly efficient audio coding algorithms for signals in three-dimensional (3D) sound fields. Generally, the masking threshold level is lowered according to the increase of the directional difference between masker and maskee signals. However, we found that when a maskee signal is located at the symmetrical position of the masker signal with respect to the frontal plane of a listener, the masking threshold level is not lowered, which counters the expectations. A mathematical model is proposed to estimate the masking threshold caused by multiple masker signals in the 3D sound field. Using the model, the perceptual entropy of a tune from a two channel stereo CD was reduced by approximately 5.5%.
Convention Paper 10259

10:30 am

P07-4 Investigation of Masking Thresholds for Spatially Distributed Sound Sources—*Sascha Dick,*[1] *Rami Sweidan,*[1,2] *Jürgen Herre*[1]
[1] International Audio Laboratories Erlangen, a joint institution of Universität Erlangen-Nürnberg and Fraunhofer IIS, Erlangen, Germany
[2] University of Stuttgart, Stuttgart, Germany

For perceptual audio coding of immersive content, the investigation of masking effects between spatially distributed sound sources is of interest. We conducted subjective listening experiments to determine the masking thresholds for "tone-masking-noise" conditions when masker (1 kHz sine tone) and probe (1 kHz narrow band noise) are spatially distributed using an immersive 22.2 loudspeaker setup. Our results show masking thresholds in the range of –35 dB to –26 dB probe-to-masker-ratio. As expected, least masking was found between left/right opposed sources with up to 5 dB lower than for coincident sources. Other noteworthy observations included an increase of masking for certain elevations and cases of selective masking decrease due to interaural phase difference phenomena.
Convention Paper 10260

11:00 am

P07-5 An Attempt to Elicit Horizontal and Vertical Auditory Precedence Percepts without Pinnae Cues—*Wesley Bulla,*[1] *Paul Mayo*[2]
[1] Belmont University, Nashville, TN, USA
[2] University of Maryland, College Park, MD, USA

This investigation was a continuation of AES-143 paper #9832 and AES-145 paper #10066 where reliable auditory precedence in the elevated, ear-level, and lowered horizontal planes was examined. This experiment altered and eliminated the spectral influences that govern the detection of elevation and presented two different horizontal and vertical inter-channel time delays during a precedence-suppression task. A robust precedence effect was elicited via ear-level horizontal plane loudspeakers. In contrast, leading signal identification was minimal in the vertical condition and no systematic influence of the leading elevated and lowered median plane loudspeakers was witnessed suggesting that precedence was not active in the vertical condition. Observed influences that might have been generated by the lead-lag signal in the vertical plane was not consistent with any known precedence paradigms.
Convention Paper 10261

11:30 am

P07-6 Perceptual Weighting to Improve Coding of Harmonic Signals—*Elias Nemer, Zoran Fejzo, Jeff Thompson*, XPERI/DTS, Calabasas, CA, USA

This paper describes a new approach to improving the coding of harmonic signals in transform-based audio codecs employing pulse vector quantization. The problem occurs when coding at low rate signals with varying levels of harmonics. As a result of vector quantization (VQ), some lower level harmonics may be missed or fluctuating and cause perceptual artifacts. The proposed solution consists of applying perceptual weighting to the computed synthesis error in the search loop of the VQ. The objective being to de-emphasize the error in the high tonal peaks where signal energy partially masks the quantization noise. Simulation results over mixed musical content showed a noticeable improvement in perceptual scores, particularly for highly harmonic signals.
Convention Paper 10262

Paper Session P08 **Thursday, Oct. 17**
9:00 am – 11:30 am **Room 1E11**

RECORDING, PRODUCTION, AND LIVE SOUND

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

9:00 am

P08-1 Microphone Comparison: Spectral Feature Mapping for Snare Drum Recording—*Matthew Cheshire, Ryan Stables, Jason Hockman*, Birmingham City University, Birmingham, UK

Microphones are known to exhibit sonic differences and microphone selection is integral in achieving desired tonal qualities of recordings. In this paper an initial multi-stimuli listening test is used to categorize microphones based on user preference when recording snare drums. A spectral modification technique is then applied to recordings made with a microphone from the least preferred category, such that they take on the frequency characteristics of recordings from the most preferred category. To assess the success of the audio transformation, a second experiment is undertaken with expert listeners to gauge pre- and post-transformation preferences. Results indicate spectral transformation dramatically improves listener preference for recordings from the least preferred category, placing them on par with those of the most preferred.
Convention Paper 10263

9:30 am

P08-2 An Automated Approach to the Application of Reverberation—*Dave Moffat, Mark Sandler*, Queen Mary University of London, London, UK

The field of intelligent music production has been growing over recent years. There have been several different approaches toautomated reverberation. In this paper we automate the parameters of an algorithmic reverb based on analysis of the input signals. Literature is used to produce a set of rules for the application of reverberation, and these rules are then represented directly as direct audio feature. This audio feature representation is then used to control the reverberation parameters from the audio signal in real time.
Convention Paper 10264

10:00 am

P08-3 Subjective Graphical Representation of Microphone Arrays for Vertical Imaging and Three-Dimensional Capture of Acoustic Instruments, Part II—*Bryan Martin, Denis Martin, Richard King, Wieslaw Woszczyk*, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, QC, Canada;

This investigation employs a simple graphical method in an effort to represent the perceived spatial attributes of three microphone arrays designed to create vertical and three-dimensional audio images. Three separate arrays were investigated in this study: Coincident, M/S-XYZ, and Non-coincident/Five-point capture. Instruments of the orchestral string, woodwind, and brass sections were recorded. Test subjects were asked to represent the spatial attributes of the perceived audio image on a horizontal/vertical grid and a graduated depth grid, via a pencil drawing. Results show that the arrays exhibit a greater extent in every dimension—vertical, horizontal, and depth—compared to the monophonic image. The statistical trends show that the spatial characteristics of each array are consistent across each dimension. In the context of immersive/3D mixing and post production, a case can be made that the arrays will contribute to a more efficient and improved workflow due to the fact that they are easily optimized during mixing or post-production.
Convention Paper 10265

10:30 am

P08-4 Filling The Space: The Impact of Convolution Reverberation Time on Note Duration and Velocity in Duet Performance—*James Weaver,*[1] *Mathieu Barthet,*[1] *Elaine Chew*[2]
[1] Queen Mary Universit London, London, UK
[2] CNRS-UMR9912/STMS (IRCAM), Paris, France

The impact of reverberation on musical expressivity is an area of growing interest as technology to simulate, and create, acoustic environments improves. Being able to characterize the impact of acoustic environments on musical performance is a problem of interest to acousticians, designers of virtual environments, and algorithmic composers. We analyze the impact of convolution reverberation time on note duration and note velocity, which serve as markers of musical expressivity. To improve note clarity in situations of long reverberation times, we posit musicians performing in a duo would lengthen the separation between notes (note duration) and increase loudness (note velocity) contrast. The data for this study comprises of MIDI messages extracted from performances by 2 co-located pianists playing the same piece of music 100 times across 5 different reverberation conditions. To our knowledge, this is the largest data set to date looking at piano duo performance in a range of reverberation conditions. In contrast to prior work the analysis considers both the entire performance as well as an excerpt at the opening part of the piece featuring a key structural element of the score. This analysis ?nds convolution reverberation time is found to be moderately positively correlated with mean note duration ($r = 0.34$ and $p =< 0.001$), but no significant correlation was found between convolution reverberation time and mean note velocity ($r = -0.19$ and $p = 0.058$).
Convention Paper 10266

11:00 am

P08-5 The Effects of Spectators on the Speech Intelligibility Performance of Sound Systems in Stadia and Other Large Venues—*Peter Mapp,*[1] *Ross Hammond*[2]
[1] Peter Mapp Associates - Colchester, Essex, UK
[2] University of Derby, Derby, Derbyshire

Stadiums and similar venues in the UK and throughout most of Europe are subject to strict safety standards and regulations, including the performance of their Public Address systems. The usual requirement is for the PA system to achieve a potential speech intelligibility performance of 0.50 STI, though some authorities and organizations require a higher value than this. However, a problem exists with measuring the performance of the system, as this can only be carried out in the empty stadium. The paper shows that with occupancy, the acoustic conditions change significantly, as the spectators introduce significant sound absorption and also increase the background noise level. The effect this can have on the intelligibility performance of the sound system is examined and discussed. The relationship between the unoccupied starting conditions and audience absorption and distribution are also investigated.
Convention Paper 10267

Paper Session P09 — Thursday, Oct. 17
9:00 am – 10:30 am — South Concourse A

POSTERS: APPLICATIONS IN AUDIO

9:00 am

P9-1 Analyzing Loudness Aspects of 4.2 Million Musical Albums in Search of an Optimal Loudness Target for Music Streaming—*Eelco Grimm*, HKU University of the Arts, Utrecht, Netherlands

In cooperation with music streaming service Tidal, 4.2 million albums were analyzed for loudness aspects such as loudest and softest track loudness. Evidence of development of the loudness war was found and a suggestion for music streaming services to use album normalization at –14 LUFS for mobile platforms and –18 LUFS or lower for stationary platforms was derived from the data set and a limited subject study. Tidal has implemented the recommendation and reports positive results.
Convention Paper 10268

9:00 am

P9-2 Audio Data Augmentation for Road Objects Classification—*Ohad Barak, Nizar Sallem*, Mentor Graphics, Mountain View, CA, USA

Following the resurgence of machine learning within

the context of autonomous driving, the need for acquiring and labeling data expanded by folds. Despite the large amount of available visual data (images, point clouds, . . .), researchers apply augmentation techniques to extend the training dataset, which improves the classification accuracy. When trying to exploit audio data for autonomous driving, two challenges immediately surfaced: first, the lack of available data and second, the absence of augmentation techniques. In this paper we introduce a series of augmentation techniques suitable for audio data. We apply several procedures, inspired by data augmentation for image classification, that transform and distort the original data to produce similar effects on sound. We show the increase in overall accuracy of our neural network for sound classification by comparing it to the non-augmented version.
Convention Paper 10269

9:00 am

P09-3 Is Binaural Spatialization the Future of Hip-Hop?— *Kierian Turner,*[1] *Amandine Pras*[1,2]
[1] University of Lethbridge, Lethbridge, AB, Canada
[2] School for Advanced Studies in the Social Sciences (EHESS), Paris, France

Modern hip-hop is typically associated with samples and MIDI and not so much with creative source spatialization since the energy-driving elements are usually located in the center of a stereo image. To evaluate the impact of certain element placements behind, above, or underneath the listener on the listening experience, we experimented beyond standard mixing practices by spatializing beats and vocals of two hip-hop tracks in different ways. Then, 16 hip-hop musicians, producers, and enthusiasts, and three audio engineers compared a stereo and a binaural version of these two tracks in a perceptual experiment. Results showed that hip-hop listeners expect a few elements, including the vocals, to be mixed conventionally in order to create a cohesive mix and to minimize distractions.
Convention Paper 10270

9:00 am

P09-4 Alignment and Timeline Construction for Incomplete Analogue Audience Recordings of Historical Live Music Concerts—*Thomas Wilmering, Florian Thalmann, Mark Sandler*, Queen Mary University of London, London, UK

Analogue recordings pose specific problems during automatic alignment, such as distortion due to physical degradation, or differences in tape speed during recording, copying, and digitization. Oftentimes, recordings are incomplete, exhibiting gaps with different lengths. In this paper we propose a method to align multiple digitized analogue recordings of same concerts of varying quality and song segmentations. The process includes the automatic construction of a reference concert timeline. We evaluate alignment methods on a synthetic dataset and apply our algorithm to real-world data.
Convention Paper 10271

9:00 am

P09-5 Noise Robustness Automatic Speech Recognition with Convolutional Neural Network and Time Delay Neural Network—*Jie Wang,*[1] *Dunze Wang,*[1] *Yunda Chen,*[1] *Xun Lu,*[2] *Chengshi Zheng*[3]
[1] Guangzhou University, Guangzhou, China
[2] Power Grid Planning Center, Guandgong Power Grid Company, Guangdong, China
[3] Institute of Acoustics, Chinese Academy of Sciences, Beijing, China

To improve the performance of automatic speech recognition in noisy environments, the convolutional neural network (CNN) combined with time-delay neural network (TDNN) is introduced, which is referred as CNN-TDNN. The CNN-TDNN model is further optimized by factoring the parameter matrix in the time-delay neural network hidden layers and adding a time-restricted self-attention layer after the CNN-TDNN hidden layers. Experimental results show that the optimized CNN-TDNN model has better performance than DNN, CNN, TDNN, and CNN-TDNN. The average recognition word error rate (WER) can be reduced by 11.76% when comparing with the baselines.
Convention Paper 10272

Acoustics & Psychoacoustics
AP02 - OPTICAL/FIBER OPTIC MICROPHONE: WHEN AND WHY SHOULD I CONSIDER USING AN OPTICAL MICROPHONE?
Thursday, October 17, 9:00 am – 10:00 am, Room 1E08

Presenters: **Ronald Ajemian**, Owl Fiber Optics, Flushing, NY, USA
Yuvi Kahana, Optoacoustics, Israel
Moshav Mazor, Israel

A brief history of the Optical/Fiber Optic Microphone will be discussed with descriptions and uses in audio related fields for picking up sounds in pro and non-pro audio applications. Dr. Yuvi Kahana obtained his Ph.D. in numerical acoustics from ISVR, Southhampton, UK and is CEO of Optoacoustics. Dr. Kahana has over 20 years eperience as he leads technology development and patent registration of optical microphone technology, while creating new innovative products in new markets. He will present the current and related applications of this amazing breakthrough for recording sound compared to conventional electroacoustic transducer types of microphones. A Questions and Answers period will be taken after the discussion.

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

Broadcast & Online Delivery
B05 - PERFORMANCE SPACES FOR BROADCAST
Thursday, October 17, 9:00 am – 10:30 am, Room 1E07

Moderator: **Paul Mclane**, Radio World

Panelists: *Sam Berkow*, SIA Acoustics, Los Angeles, CA, USA
John Caracciola, JVC Broadcasting Co., Inc.
Gary Kline, Kline Consulting Group LLC, Atlanta, GA, USA
Jason Ornellas, Bonneville International, Sacramento, CA, USA

Forward thinking ideas about designing, running, and maintaining performances spaces. Video streaming, recording live sound, revenue opportunity, and numerous other details that go into a multi-purpose performance space. This session will help provide you tech ideas, design tips, and guidelines to navigate the concept to completion of building you performance space in your facility.

Electronic Instrument Design & Applications
EI01 - DRUM MACHINES, GROOVE-BOXES, AND TEMPO-BASED ELECTRONIC INSTRUMENTS
Thursday, October 17, 9:00 am – 10:00 am, Room 1E17

Presenters: **Michael Bierylo**, Berklee College of Music, Brookline, MA, USA
Peter Brown
Dan Gill
Dave Rossum

Early drum machines such and the Roland TR-808 and the Linn drum ushered in a new era in mu-sic production forming the bedrock of hit songs from the 1980s onward. These legendary instru-ments have helped define musical genres like Hip Hop and Dance music, and the influence of these designs can't be underestimated. This session will examine the design philosophies behind these instruments and how they continue to be a vital part of the electronic instrument industry.

Game Audio & XR
GA07 - MPEG-H 3D AUDIO GOES VR
Thursday, October 17, 9:00 am – 10:30 am, Room 1E21

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

Panelists: *Adrian Murtaza*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Nils Peters, Qualcomm, Advanced Tech R&D, San Diego, CA, USA

The MPEG-H 3D Audio is a recent MPEG standard that was designed to represent and render 3D audio experiences while supporting all known production paradigms (channel-based, object-based, and Higher Order Ambisonics based audio) and reproduction setups (loudspeaker, head-phone/binaural). As the audio production world moves forward to embrace Virtual and Augmented Reality (VR/AR), MPEG-H found considerable adoption and re-use in recently finalized VR standards, such as MPEG-I OMAF(Omnidirectional Media Format), VR Industry Forum (VR-IF) Guidelines as well as 3GPP "VRStream" (Virtual Reality profiles for streaming applications) where it was selected as the audio standard for VR content delivered over 5G networks.

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

Product Development
PD06 - CAN DSP FIX A BAD SPEAKER?
Thursday, October 17, 9:00 am – 10:30 am, Room 1E09

Presenters: **Jonathan Gerbet**, Klippel GmbH, Dresden, Germany

Adaptive nonlinear loudspeaker control algorithms implemented in low-cost DSPs allows speakers to be driven very close to their physical limits, producing the maximum SPL and increasing the reliability. While distortion caused by deterministic motor and suspension nonlinearities can be canceled by digital signal processing, residual distortion commonly known as "Rub&Buzz" becomes more audible and has to be fixed by the transducer manufacturer or system integrator as it can never be compensated by electrical means. This tutorial focuses on particularities, design goals, and requirements of optimal passive systems for adaptive nonlinear control. Measurement methods that combine perceptual and physical metrics are introduced to provide a comprehensive evaluation of the transducer, enclosure and amplifier intended for adaptive nonlinear control.

Sound Reinforcement
SR03 - RF SUPER SESSION
Thursday, October 17, 9:00 am – 12:00 noon, Room 1E13

Spectrum Update Panel

Moderator : **James Stoffo**

Panelists: *Mark Brunner*, Shure
Joe Ciaudelli, Sennheiser
Karl Winkler, Lectrosonics

Manufacturers

Moderator : **James Stoffo**
[TBD]

Advanced Practices Panel

Moderator: **Jason Glass**, Clean Wireless Audio

Panelists: *Jim Dugan*, JetWave Wireless
Cameron Stuckey, Firehouse Productions
James Stoffo
Gary Trenda, Professional Wireless Systems

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 three part full [morning/afternoon] super session will discuss in depth RF spectrum and regulatory changes; a manufacturers update; and best and advanced RF practices including 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
SC06 - AES MATLAB PLUGIN STUDENT COMPETITION
Thursday, October 17, 9:00 am – 10:30 am, Room 1E12

MathWorks is supporting the 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.

The following submissions were chosen by the judges to be presented in front of the audience. These projects are considered to win cash and software prizes. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2). Learn more about this competition at aes.org/students/awards.

- Edward Ly, "Inner Space," University of Aizu
- Domenico Andrea Giliberti, Festim Iseini, Nicola Ignazio Pelagalli, Alessandro Terenzi, "BEStEx - Bass Enhancer Stereo Expander," Universita Politecnica delle Marche
- Christian Steinmetz, "flowEQ," Universitat Pompeu Fabra
- Sean Newell, "Shift Drive," Belmont University
- John Kolar, "Dynamizer: Amplitude Modulator & Audio Effects Plugin," West Virginia University
- Michael Nuzzo, "Spectrum Pixelator," University of Massachusetts-Lowell

Student Events & Career Development
SC07 - RESUME REVIEW (FOR STUDENTS, RECENT GRADS, AND YOUNG PROFESSIONALS)
Thursday, October 17, 9:00 am – 5:00 pm, SDA Booth

Moderator: **Alex Kosiorek**, Central Sound at Arizona PBS, Phoenix, AZ, USA

Students, recent graduates and young professionals… Often your resume is an employer's first impression of you. Naturally, you want to make a good one. Employer's often use job search websites to search for candidates. Some use automated software to scan your resume and in some cases, your LinkedIn/social media profiles as well. Questions may arise regarding formatting, length, keywords and phrases so it shows up in searches and lands on the desk of the hiring manager. No matter how refined your resume may be, it is always good to have someone else review your materials. Receive a one-on-one 20-25 minute review of your resume from a hiring manager who is in the audio engineering business. Plus, if time allows, your cover letter and online presence will be reviewed as well.

Sign up at the student (SDA) booth immediately upon arrival. For those who would like to have your resume reviewed on Wednesday, October 17th prior to SDA-1, please email the request to: aesresumereview@outlook.com. You may be requested to upload your resume prior to your appointment for review. Uploaded resumes will only be seen by the moderator and will be deleted at the conclusion of the 147th Pro Audio Convention.

This review will take place during the duration of the convention by appointment only.

Technical Committee Meeting
MICROPHONES AND APPLICATIONS
Thursday, October 17, 9:00 am – 10:00 am, Room 1C02

AES Standards Meeting
SC-02-08 WORKING GROUP ON AUDIO-FILE TRANSFER AND EXCHANGE
Thursday, October 17, 9:00 am – 10:00 am, Room 1C03

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.

Hip Hop & R&B
HH01 - CHOPPED AND LOOPED—INSIDE THE ART OF SAMPLING FOR HIP-HOP (AN AES SPECIAL EVENT)
Thursday, October 17, 9:30 am – 11:00 am, Room 1E15+16

Moderator: **Paul "Willie Green" Womack**, Willie Green Music, Brooklyn, NY, USA

Panelists: *Just Blaze*, Jay-Z, Kanye West
Breakbeat Lou, Ultimate Breaks and Beats
Hank Shocklee, Shocklee Entertainment, New York, NY, USA
Ebonie Smith, Atlantic Records/Hamilton Cast Album

Celebrating the art of the audio collage, the panelists will discuss the production technique that launched a genre. Exploring the process of digging for samples, techniques and tools for composi-tion, and concepts for processing and mixing, this panel will cover all aspects of sample driven production.

Technical Committee Meeting
AUDIO FOR GAMES
Thursday, October 17, 10:00 am – 11:00 am, Room 1C02

AES Standards Meeting
SC-04-03-A TASK GROUP ON MEASURING LOUDSPEAKER MAXIMUM LINEAR PEAK SPL USING NOISE
Thursday, October 17, 10:00 am – 11:00 am, Room 1C03

The scope of SC-04-03-A is to develop a method for measuring the maximum linear peak SPL of a loudspeaker driver or system. It uses a mathematically derived test signal that effectively emulates the dynamic characteristics of music as a function of frequency as well as its spectral content.

AES Mix with the Masters Workshops
MM09 - PETER KATIS
Thursday, October 17, 10:00 am – 11:00 am
Mix with the Masters Workshop Stage

Electronic Instrument Design & Applications
EI02 - NOT YOUR EVERYDAY FILTERS
Thursday, October 17, 10:15 am – 11:45 am, Room 1E17

Presenter: **Jayant Datta**, Audio Precision, Beaverton, OR, USA

The purpose of this tutorial is to introduce and expose audio engineers to additional filters that have interesting properties but are not as commonly known to our community.

In this tutorial

- We take a deeper look at allpass filters – where phase (instead of frequency shaping) is used as a manipulating tool
- We take a closer look at real-time zero-phase IIR filtering techniques
- Finally, we look at multi-rate sampling systems, where in addition to polyphase FIR filters we also look at polyphase IIR approaches

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

Game Audio & XR
GA08 - INTRODUCTION TO MACHINE LEARNING FOR GAME AUDIO
Thursday, October 17, 10:15 am – 11:45 am, Room 1E08

Presenters: **Krystle Becknauld**
John Byrd, Gigantic Software, Santa Ana, CA, USA

In the past four years, every tech industry has put machine learning systems into production. Virtu-al assistants, cybersecurity systems, and self-driving cars are all built on machine learning. However, none of these novel algorithms are being used in the interactive entertainment industries.

This talk summarizes the most important new research in this exciting new field and describes novel applications of machine learning, specifically for interactive audio. You'll see demonstrations and code for: automatic feature extraction, classification, silence removal, tempo extraction, automatic speaker detection, emotion detection, convolutional and recurrent neural networks, variational auto-encoders, and WaveNet. Additionally, this talk will show you the state of the art in interactive audio, and it will also tell you what will be possible, until the year 2030.

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

Recording & Production
RP06 - IMMERSIVE MUSIC LISTENING SESSION: CRITICAL LISTENING AND RECORDING TECHNIQUES
Thursday, October 17, 10:30 am – 12:00 noon, Room 1E106

Presenters: **David Bowles**, Swineshead Productions LLC, Berkeley, CA, USA
Paul Geluso, New York University, New York, NY, USA

In this workshop, new immersive music recordings will be presented followed by a brief technical discussion by their creators. Paul Geluso and David Bowles will host the session presenting their recent work and invite other recording engineers and music producers working in immersive formats to present recent works as well. Tracks will be played in their entirety to preserve their artistic impact and create an environment for critical listening. Following playback of each work will be a brief presentation and Q and A session. New immersive recording techniques designed specifically to optimize Dolby ATMOS compatibility will be presented by Geluso and Bowles as well.

AoIP Pavilion
AIP13 - INTRODUCTION TO THE AUDIO/VIDEO-OVER-IP TECHNOLOGY PAVILION
Thursday, October 17, 10:30 am – 11:00 am
AoIP Pavilion Theater

Presenter: **Terry Holton**, AIMS Audio Group Chairman – UK, London, UK

The Audio/Video-over-IP Technology Pavilion is an initiative created by the AES in partnership with the Alliance for IP Media Solutions (AIMS). Following the success of last year's pavilion which provided valuable insights and practical information about audio over IP networking, the scope of this year's pavilion has been expanded to also include video and other related professional media networking topics. This presentation will provide an overview of the various aspects of the pavilion as well as an introduction to AES67 and its relationship to the SMPTE ST 2110 standard.

Broadcast & Online Delivery
B06 - LIVE BROADCASTING WITH OBJECT BASED AUDIO
Thursday, October 17, 10:45 am – 12:00 noon, Room 1E07

Presenters: **Frédéric Changenet**, Radio France, Paris, France
Adrian Murtaza, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Matthieu Parmentier, francetélévisions, Paris, France

Object Based Audio for immersive and interactive contents is now taking off in Europe thanks to close collaborations between EBU members and NGA industrials. With the participation of such panelists, this workshop will highlight the lessons learnt from 3 major use-cases:

• A live sport event: the 2019 Roland Garros tennis tournament, where object-based audio is used to create simultaneous audio versions at the same time

• A live music event: the 2019 Eurovision Song Contest, where NGA offers multiple languages and musical mix versions on the receiver side

• A live musical tour of electronic music, where 20 dynamic objects from the DJ set fly over the crowd thanks to a 360 sound reinforcement system, together with a binaural broadcasting on radio

• An Audio programs collection containing dramas and documentaries produced with object-based technologies, allowing versatile renderings from Stereo FM Broadcasting to binaural and multichannel streaming, WFS equipped venues and movie theater broadcast

Product Development
PD07 - INTRODUCING A2B—A LOW-LATENCY DIGITAL AUDIO BUS STANDARD
Thursday, October 17, 10:45 am – 12:15 pm, Room 1E09

Presenters: **Joseph Beauchemin**, Analog Devices
Denis Labrecque, DeLab Consulting, Half Moon Bay, CA, USA

A2B is a de-facto digital audio communication standard developed to input and deliver deterministic, very low latency (< 50 microseconds) audio across multiple nodes. Multiple microphone arrays and speakers can be connected to the same daisy-chained unshielded twisted-pair (UTP) wires carrying both digital microphone and I2S/TDM audio signals (up to 32 channels), as well as control information and bus power. This presentation will explain the capabilities of A2B and how it works, illustrating how the low-latency A2B bus enables new applications like simplified multichannel pickup cabling and other multichannel digital I/O for music instruments/effects, as well as mixing consoles, robotics applications, and more. The presentation will also demonstrate how easy it is to use the A2B bus to bidirectionally transmit synchronous audio data (I2S/TDM-to-I2S/TDM) and control/status information (I2C-to-I2C) across multiple bus nodes in an A2B system.

Recording & Production
RP07 - PODCAST PRODUCTION
Thursday, October 17, 10:45 am – 11:45 am, Room 1E12

Moderator: **Elena Botkin-Levy**

Panelists: *Chiquita Paschal*, Uncivil, Another Round
Haley Shaw, Gimlet Media, Brooklyn, NY, USA
Jenna Weiss-Berman, Pineapple Media

What makes a good podcast? What makes it worth listening to? Producers from top podcasts like Uncivil, The Daily, The Habitat and podcasting companies like Gimlet Media and Pineapple Media break-down the production process and technical elements of successful podcasting.

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

Student Events & Career Development
SC08 - SAUL WALKER STUDENT DESIGN COMPETITION
Thursday, October 18, 10:45 am – 12:15 pm, South Concourse A

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 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.

Technical Committee Meeting
FIBER OPTICS FOR AUDIO
Thursday, October 17, 11:00 am – 12 noon, Room 1C02

AES Standards Meeting
SC-04-03 WORKING GROUP ON LOUDSPEAKER MODELING AND MEASUREMENT
Thursday, October 17, 11:00 am – 12:30 pm, Room 1C03

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.

AES Mix with the Masters Workshops
MM10 - MICHAEL BRAUER
Thursday, October 17, 11:00 am – 12:00 noon
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP14 - HOW TO CONFIGURE YOUR RAVENNA SETUP WITH GLOBCON—NEW, "GLOBAL CONTROL" SOFTWARE PLATFORM
Thursday, October 17, 11:00 am – 11:30 am
AoIP Pavilion Theater

Presenter: **Luca Giaroli**, DirectOut

Globcon is a new, free, global control software platform for the management of professional audio equipment. It has been conceived and designed from the end-user's point of view to allow consistent and coherent control of multiple pieces of equipment from different brands as if they were just one. The goal of globcon is to offer to professional users an ever-increasingly powerful tool to manage environments of any size in a modern, ergonomic, flexible, efficient and cost-effective way. In September 2019 the official version of globcon has been released, following the completion of a successful beta testing program that has proved the stability and capabilities of the software. RAVENNA partner DirectOut is the first adopter of globcon, many other RAVENNA partners are currently working on adding globcon support.

Audio Builders Workshop Booth Talks
ABT06 - FIND YOUR DIY VOICE, HACK YOUR OWN STUFF
Thursday, October 17, 11:00 am – 11:30 am
Booth 266 (Exhibit Floor)

Presenter: **Michael Swanson**

Project Studio Expo Recording Stage
RS07 - SONICSCOOP PODCAST LIVE: MONITORS, ACOUSTICS, AND CORRECTION: THE THREE KEYS TO UPGRADING YOUR CONTROL ROOM
Thursday, October 17, 11:00 am – 11:45 am
Recording Stage

Presenter: **Justin Colletti**

Good monitoring is arguably the most important asset in any professional studio, and bad monitoring is a liability that is a major challenge to overcome. Whatever budget and room size you have to work with, getting the most out of your listening environment hinges on three things: 1. Selecting the right speakers for your room and your needs, 2. Having a smart and cost-effective plan for acoustic treatment and 3. Using room and speaker correction to keep coloration to an absolute minimum. In this live recording of the SonicScoop Podcast, Justin Colletti invites three special guests to talk about these three most essential elements in upgrading your studio's monitoring situation so that you can finally trust what you hear and make great-sounding work with a minimum of guesswork.

Recording & Production
RP08 - CHANGING ROLES IN THE AUDIO INDUSTRY
Thursday, October 17, 11:15 am - 12:15 pm, Room 1E21

Moderator: **Paul "Willie Green" Womack**, Willie Green Music, Brooklyn, NY, USA

Panelists: *Ariel Borujow*, Westward Music Group Inc.
Ken "Duro" Ifill, Jay-Z & Alicia Keys/Erykah Badu/ Will Smith
Mike Kuz, Drake, J. Cole, Offset, Teyana Taylor

As the music industry and technology change at a rapid rate, the previously clearly-defined roles in the studio have blurred the lines as well. Modern music professionals often find themselves wearing the hats of producer, engineer, mixer, and more. This panel will explore the shifting responsibilities of the modern music industry.

Electronic Dance Music Stage
EDJ09 - AVID
Thursday, October 17, 11:00 am – 12:00 noon
Electronic Dance Music & DJ Stage

Live Production Stage
LS07 - MEYER SOUND PRESENTS: M-NOISE 101
Thursday, October 17, 11:00 am – 11:45 am
Live Production Stage

Special Event
SE05 - SHOW ME THE MONEY: FUNDING YOUR AUDIO DREAM
Thursday, October 17, 11:15 am – 12:15 pm, Room 1E15+16

Moderator: **Heather D. Rafter**, RafterMarch US

Panelists: *Phil Dudderidge*, Executive Chariman, Focusrite PLC
Mark Ethier, CEO iZotope, Cambridge, MA, USA
*Ethan Jack*s, Founder, MediaBridge Capital
Piper Payne, Neato Mastering, San Francisco Bay Area, CA, USA
Wisam Reid, Harvard Medical School

This panel of industry insiders will share their tips on funding your audio passion, whether you're a student, start up, or an established company wishing to expand. We'll take you through every scenario: from scholarships and grants, to crowd funding via Kickstarter and other campaigns, and onto raising money through friends and family rounds and more. We'll demystify venture capital, debt financing, investment banking, and private equity, and we'll also explore growth through merger and acquisition or IPO. Whether you're a student, solo audio developer, new or well-established company, this program will guide you through the financing path that best meets your needs.

AoIP Pavilion
AIP15 - JT-NM TESTED PROGRAM—TEST PLANS AND RESULTS
Thursday, October 17, 11:30 am – 12:00 noon
AoIP Pavilion Theater

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

The JT-NM Tested Program was repeated in August 2019 with the addition of AMWA NMOS/JT-NM TR-1001-1 testing. New revisions of the test plans were produced. What does all this mean to the end customers? The editor and coordinator of the program will explain the reasoning behind, the technical details, what was changed in the new revisions, how it all was executed and everything else you wanted to know about the JT-NM Tested Program, but were too afraid to ask!

Imersive & Spatial Audio
IS13 - MORTEN LINDBERG TALK&PLAY
Thursday, October 17, 12:00 pm – 1:00 pm, Room 1E17

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

Morten: "We should create the sonic experience that emotionally moves the listener to a better place. Immersive Audio is a completely new conception of the musical experience." Listen to Morten and some of his most legendary recordings.

Student Events & Career Development
SC09 - STUDENT RECORDING CRITIQUES
Thursday, October 17, 12:00 noon – 1:00 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
ACOUSTICS AND SOUND REINFORCEMENT
Thursday, October 17, 12:00 noon – 1:00 pm, Room 1C02

Technical Committee Meeting
BROADCAST AND ONLINE DELIVERY: ADVANCED GUIDELINES FOR OVER THE TOP TV & STREAMING
Thursday, October 17, 12:00 noon – 1:00 pm, Room 1E07

AES Mix with the Masters Workshops
MM11 - CHRIS LORD-ALGE
Thursday, October 17, 12:00 noon – 1:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP16 - NMOS—A GENERAL OVERVIEW OF THE CURRENT STATE
Thursday, October 17, 12:00 noon – 12:30 pm
AoIP Pavilion Theater

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

Overview of the current state of NMOS (Automated Discovery and Connection Management of ST 2110 Media Devices), including a look into some of the security aspects, and a short discussion on how Riedel is applying NMOS functionality into its product line.

Audio Builders Workshop Booth Talks
ABT07 - HEALING POWER OF DIY GEAR
Thursday, October 17, 12:00 noon – 12:30 pm, Booth 266

Presenter: **Buddy Lee Dobberteen**

Electronic Dance Music Stage
EDJ03 - WAVES PRODUCT WORKSHOP
Thursday, October 17, 12:00 noon – 12:45 pm
Electronic Dance Music & DJ Stage

Presenter: **Michael Pearson-Adams**, Waves, Knoxville, TN, USA

Live Production Stage
LS08 - DPA PRESENTS: THE HIDDEN MIC FOR BROADWAY AND THEATER PRODUCTIONS
Thursday, October 17, 12:00 noon – 12:45 pm
Live Production Stage

Software@AES
S11 - BITWIG
Thursday, October 17, 12:00 noon – 12:30 pm
Software@AES Pavilion

Product Development
PD08 - VENDOR EVENT 2: ANALOG DEVICES
Thursday, October 17, 12:15 pm – 1:30 pm, Demo Room 2D03

Analog Devices' A2B digital audio bus is a new way to distribute bi-directional multi-channel audio. Previously only available to tier 1 automotive customers, A2B is now being released for the broad market for pro audio applications such as conference rooms, studio monitoring, rack-to-rack and board-to-board digital communication, and more. This session will feature a live, interactive demonstration of the A2B topology discussed in PD07 and will focus on a real-time configuration of multiple microphones and speakers utilizing the SigmaStudio development/deployment tools.

AoIP Pavilion
AIP17 - NMOS—A GENERAL OVERVIEW OF THE CURRENT STATE
Thursday, October 17, 12:30 pm – 1:00 pm
AoIP Pavilion Theater

Presenter: **Jeff Berryman**, OCA Alliance

NMOS Convergence is a multi-organization collaboration for compatibly evolving the current NMOS specification set into a suite of formal public networking standards suitable for long-term strategic use. Initial version release is planned for April 2021 timeframe. The project is based on the pioneering set of NMOS specifications created by the AMWA NMOS team.

Audio Builders Workshop Booth Talks
ABT08 - TROUBLESHOOTING BY SOLDERING AND BUILDING
Thursday, October 17, 12:30 pm – 1:00 pm, Booth 266

Presenter: **Joyce Lieberman**

Software@AES
S12 - SONIBLE
Thursday, October 17, 12:30 pm – 1:00 pm
Software@AES Pavilion

Acoustics & Psychoacoustics
AP03 - SHAME ON US: PHASE IS NOT POLARITY!
Thursday, October 17, 1:00 pm – 1:45 pm, Room 1E12

Presenter: **Cesar Lamschtein**, Kaps Audio Production Services, Montevideo, Uruguay; Mixymaster, Montevideo, Uruguay

Being disappointed with the industry regarding how messy the concepts of PHASE and POLARITY get to students, general amateur audio enthusiasts and even seasoned professionals. I think that AES is to the only one that can and SHOULD do something to encourage the stopping of the practice of freely exchanging these terms.

E.g, • major console brands, either analog and digital in either plain word or the ø symbol; • Daw software (nuendo, reaper, etc.); • outboard equipment (including such classics as 1073 neve eq); • videos (Allan Parsons MTCS DVD); • books.

During this presentation I will: • Acknowledge the situation: I´ll show the results of a survey done on the 2 past NY conventions, where 100% of the polarity switches in the exhibition floor was reviewed and photographed, showing inconsistency in the naming of the switch even in different products from the same manufacturer; • Teach terms: I´ll state actual definitions and clarify the concepts of phase and polarity with audio examples of each; • Encourage the discussion: I want to gather people together to nourish this venture, discussing possible abbreviation and symbol (ICON) to represent polarity in order to gather information that may lead to an AES TD; • I´ll invite fellow audio men and women to speak properly about those important concepts (and THEN be able to think properly about this quite basic concepts, albeit misunderstood).

Broadcast & Online Delivery
B07 - ANATOMY OF A STREAM
Thursday, October 17, 1:00 pm – 2:30 pm, Room 1E07

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

Panelists: *Kirk Harnack*, Telos Alliance
John Kean, Kean Consultants LLC, Washington DC, USA; Cavell Mertz & Associates, Manassas VA, USA
Robert Orban, Orban Labs Inc., Pennsauken, NJ, USA
Mike Smith, MainStreaming, Inc., San Francisco, CA, USA

An audio and video stream can contain many components as different protocols allow for different things. So what's inside of those signals when can they be used over the Internet? Let's talk about the streaming formats available on the internet, when they are used and when they are privileged over other formats, the pros and cons of each, and how to prepare for future approaches. Additionally, if we have the time, let`s delve into 5G and ATSC3.0 and see if audio has not been forgotten.

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

Technical Committee Meeting
AUDIO FOR CINEMA
Thursday, October 17, 1:00 pm – 2:00 pm, Room 1C02

AES Mix with the Masters Workshops
MM12 - TOM LORD-ALGE
Thursday, October 17, 1:00 pm – 2:00 pm
Mix with the Masters Workshop Stage

Project Studio Expo Recording Stage
RS08 - API PRESENTS THE CASE FOR ANALOG
Thursday, October 17, 1:00 pm – 1:45 pm
Recording Stage

Presenters: **Joe Chiccarelli**, Producer, mixer, engineer, Boston, MA, USA
Daniel Schlett, Strange Weather Studio, Brooklyn, NY, USA
Dave Trumfio, Gold-Diggers Sound, Los Angeles, CA, USA

Join API for a spirited panel discussion featuring Grammy Award-winning Engineer/Producers Daniel Schlett (Strange Weather studio, Brooklyn, NY), Joe Chiccarelli, and Dave Trumfio (Gold-Diggers Sound, Los Angeles, CA).

Software@AES
S13 - MAGIX
Thursday, October 17, 1:00 pm – 1:30 pm
Software@AES Pavilion

Paper Session P10 **Thursday, Oct. 17**
1:15 pm – 4:15 pm **Room 1E10**

SPATIAL AUDIO, PART 1

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

1:15 pm

P10-1 Use of the Magnitude Estimation Technique in Reference-Free Assessments of Spatial Audio Technology— *Alex Brandmeyer, Dan Darcy, Lie Lu, Richard Graff, Nathan Swedlow, Poppy Crum*, Dolby Laboratories, San Francisco, CA, USA

Magnitude estimation is a technique developed in psychophysics research in which participants numerically estimate the relative strengths of a sequence of stimuli along a relevant dimension. Traditionally, the method has been used to measure basic perceptual phenomena in different sensory modalities (e.g., "brightness," "loudness"). We present two examples of using magnitude estimation in the domain of audio rendering for different categories of consumer electronics devices. Importantly, magnitude estimation doesn't require a reference stimulus and can be used to assess general ("audio quality") and domain-specific (e.g., "spaciousness") attributes. Additionally, we show how this data can be used together with objective measurements of the tested systems in a model that can predict performance of systems not included in the original assessment.
Convention Paper 10273

1:45 pm

P10-2 Subjective Assessment of the Versatility of Three-Dimensional Near-Field Microphone Arrays for Vertical and Three-Dimensional Imaging—*Bryan Martin*,[1,2] *Jack Kelly*,[1,2] *Brett Leonard*[3,4]
[1] McGill University, Montreal, QC, Canada
[2] Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, QC, Canada
[3] University of Indianapolis, Indianapolis, IN, USA
[4] BLPaudio, Indianapolis, IN, USA

This investigation examines the operational size-range of audio images recorded with advanced close-capture microphone arrays for three-dimensional imaging. It employs a 3D panning tool to manipulate audio images. The 3D microphone arrays used in this study were: Coincident-XYZ, M/S-XYZ, and Non-coincident-XYZ/five-point. Instruments of the orchestral string, woodwind, and brass sections were recorded. The objective of the test was to determine the point of three-dimensional expansion onset, preferred imaging, and image breakdown point. Subjects were presented with a continuous dial to manipulate the three-dimensional spread of the arrays, allowing them to expand or contract the microphone signals from 0° to 90° azimuth/elevation. The results showed that the M/S-XYZ array is the perceptually "biggest" of the capture systems under test and displayed the fasted sense of expansion onset. The coincident and non-coincident arrays are much less agreed upon by subjects in terms of preference in particular, and also in expansion onset.
Convention Paper 10274

2:15 pm

P10-3 Defining Immersion: Literature Review and Implications for Research on Immersive Audiovisual Experiences—*Sarvesh Agrawal*,[1,2] *Adèle Simon*,[1] *Søren Bech*,[1,3] *Klaus Bærentsen*,[4] *Søren Forchhammer*[2]
[1] Bang & Olufsen a/s, Struer, Denmark
[2] Technical University of Denmark, Lyngby, Denmark
[3] Aalborg University, Aalborg, Denmark
[4] Aarhus University, Aarhus, Denmark

The use of the term "immersion" to describe a multitude of varying experiences in the absence of a definitional consensus has obfuscated and diluted the term. This paper presents a non-exhaustive review of previous work on immersion on the basis of which a definition of immersion is proposed: a state of deep mental involvement in which the subject may experience disassociation from the awareness of the physical world due to a shift in their attentional state. This definition is used to contrast and differentiate interchangeably used terms such as presence and envelopment from immersion. Additionally, an overview of prevailing measurement techniques, implications for research on immersive audiovisual experiences, and avenues for future work are discussed briefly.
Convention Paper 10275

2:45 pm

P10-4 Evaluation on the Perceptual Influence of Floor Level Loudspeakers for Immersive Audio Reproduction— *Yannik Grewe, Andreas Walther, Julian Klapp*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Listening tests were conducted to evaluate the perceptual influence of adding a lower layer of loudspeakers to a setup that is commonly used for immersive audio reproduction. Three setups using horizontally arranged loudspeakers (1M, 2M, 5M), one with added height loudspeakers (5M+4H), and one with additional ?oor level loudspeakers (5M+4H+3L) were compared. Basic Audio Quality was evaluated in a sweet-spot test with explicit reference, and two preference tests (sweet-spot and off sweet-spot) were performed to evaluate the Overall Audio Quality. The stimuli, e.g., ambient recordings and sound design material, made dedicated use of the lower loudspeaker layer. The results show that reproduction comprising a lower loudspeaker layer is preferred compared to reproduction using the other loudspeaker setups included in the test.
Convention Paper 10276

3:15 pm

P10-5 Investigating Room-Induced Influences on Immersive Experience Part II: Effects Associated with Listener Groups and Musical Excerpts—*Sungyoung Kim,*[1] *Shuichi Sakamoto*[2]
[1] Rochester Institute of Technology, Rochester, NY, USA
[2] Tohoku University, Sendai, Japan

The authors previously compared four distinct multichannel playback rooms and showed that perceived spatial attributes of program material (width, depth, and envelopment) were similar across all four rooms when reproduced through a 22-channel loudspeaker array. The present study further investigated perceived auditory immersion from two additional variables: listener group and musical style. We found a three-way interaction of variables, MUSIC x (playback) ROOM x GROUP for 22-channel reproduced music. The interaction between musical material and playback room acoustics differentiates perceived auditory immersion across listener groups. However, in the 2-channel reproductions, the room and music interaction is prominent enough to flatten inter-group differences. The 22-channel reproduced sound fields may have shaped idiosyncratic cognitive bases for each listener group.
Convention Paper 10277

3:45 pm

P10-6 Comparison Study of Listeners' Perception of 5.1 and Dolby Atmos—*Tomas Oramus, Petr Neubauer*, Academy of Performing Arts in Prague, Prague, Czech Republic

Surround sound reproduction has been a common technology in almost every theater room for several decades. In 2012 Dolby Laboratories, Inc. announced a new spatial 3D audio format – Dolby Atmos [1] that (due to its object-based rendering) pushes the possibilities of spatial reproduction and supposedly listeners' experience forward. This paper examines listeners' perception of this format in comparison with today's unwritten standard for cinema reproduction – 5.1. Two sample groups were chosen for the experiment - experienced listeners (sound designers and sound design students) and inexperienced listeners; the objective was to examine how these two groups perceive selected formats and whether there is any difference between these two groups. We aimed at five aspects – Spatial Immersion (Envelopment), Localization, Dynamics, Audio Quality, and Format Preference. The results show mostly an insignificant difference between these two groups while both of them slightly leaned towards Dolby Atmos over 5.1.
Convention Paper 10278

Paper Session P11 **Thursday, Oct. 17**
1:15 pm – 2:45 pm **Room 1E11**

SEMANTIC AUDIO

Chair: **Robert C. Maher**, Montana State University, Bozeman, MT, USA

1:15 pm

P11-1 Impact of Statistical Parameters of Late Reverberation on the Instantaneous Frequencies of Reverberant Audio—*Sarah R. Smith, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

This paper addresses the impact of late reverberation on the instantaneous frequency tracks of reverberant audio. While existing models of early reflections and low frequency room modes enable prediction of instantaneous frequency tracks of a filtered signal, the effects of late reverberation are best modeled statistically. After reviewing the parameterization of late reverberation, the effects of frequency dependent decay time and direct to reverberant ratio on instantaneous frequency are investigated using synthetic impulse responses derived from velvet noise. These effects are quantified using the autocorrelation function of the reverberant instantaneous frequency tracks. Finally, the instantaneous frequency deviations that occur when an anechoic sound is filtered with a recorded impulse response are compared to those resulting from synthesized late reverberation.
Convention Paper 10279

1:45 pm

P11-2 Precise Temporal Localization of Sudden Onsets in Audio Signals Using the Wavelet Approach—*Yuxuan Wan, Yijia Chen, Keegan Yi Hang Sim, Lijia Wu, Xianzheng Geng, Kevin Chau*, Hong Kong University of Science and Technology, Clean Water Bay, Hong Kong

Presently reported is a wavelet-based method for the temporal localization of sudden onsets in audio signals with sub-millisecond precision. The method only requires $O(n)$ operations, which is highly efficient. The entire audio signal can be processed as a whole without the need to be broken down into individual windowed overlapping blocks. It can also be processed in a streaming mode compatible with real-time processing. In comparison with time-domain and frequency-domain methods, the wavelet-based method proposed here offers several distinct advantages in sudden onset detection, temporal localization accuracy, and computational cost, which may therefore find broad applications in audio signal processing and music information retrieval.
Convention Paper 10280

2:15 pm

P11-3 Forensic Comparison of Simultaneous Recordings of Gunshots at a Crime Scene—*Robert C. Maher, Ethan Hoerr*, Montana State University, Bozeman, MT, USA

Audio forensic evidence is of increasing importance in law enforcement investigations because of the growing use in the United States of personal audio/video recorders carried by officers on duty, by bystanders, and by surveillance systems of businesses and residences. These recording systems capture speech, background environmental sounds, and in some cases, gunshots and other firearm sounds. When there are multiple audio recording devices near the

scene of a gunfire incident, the similarities and differences of the various recordings can either help or hamper the audio forensic examiner's efforts to describe the sequence of events. This paper considers several examples and provides recommendations for audio forensic examiners in the interpretation of this gunshot acoustic evidence.
Convention Paper 10281

Game Audio & XR
GA09 - AUDIO PRODUCTIVITY IN MIXED REALITY
Thursday, October 17, 1:15 pm – 2:15 pm, Room 1E13

Presenters: **Sally Kellaway**, Microsoft, Seattle, WA, USA
Joe Kelly, Microsoft, Seattle, WA, USA

Hearing is a human sense that subconsciously enables humans to detect and categorize stimulus to make decisions and take actions. Virtual, Augmented and Mixed Reality has allowed audio design-ers and developers in general to explore what framework of audio is needed to have mostly enter-tainment focused immersive experiences. This has enabled users to have an immense capacity for understanding and empathy with a myriad of experiences. In enterprise scenarios there are a vast array of industries that deploy immersive technologies to expedite workflows, however, enterprise technology has historically focused on productivity at the expense of experience and has never be-fore needed to factor in the complexity of the environment of the experience to this extent. The au-dio team developing Microsoft Dynamics 365 Mixed Reality applications has spent 2+ years refin-ing a framework for audio productivity in MR, and will discuss a number of approaches to design-ing and implementing audio features for enterprise and consumer applications where productivity of the user is the goal.

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

Historical Event
H03 - RUDY VAN GELDER:
A LEGACY IN AUDIO ENGINEERING
Thursday, October 17, 1:15 pm – 2:45 pm, Room 1E08

Presenters: **Richard Capeless**, Deep Groove Mono, New York, NY, USA
Tom Fine
Ashley Kahn
Don Sickler, Second Floor Music
Maureen Sickler, Second Floor Music

Rudy Van Gelder, or "RVG" as most jazz fans know him, was responsible for laying to tape hundreds of classic jazz recordings spanning a period of over five decades, which were released on a variety of labels including Blue Note, Prestige, Savoy, Impulse, Verve, and CTI. His rise from obscurity to engineering fame centered on a unique approach to his work, about which he was famously secretive. Beginning in a makeshift living room recording studio in his parents' Hackensack, New Jersey, home, Van Gelder would eventually build a one-of-a-kind cathedral-like studio in nearby Englewood Cliffs that is still in use today.

Always on the cutting edge of technology, Van Gelder used a plethora of state-of-the-art tools to create "the Van Gelder Sound," a distinct and easily recognizable sonic fingerprint that helped define the sound of jazz on record during the classic analog era of the 1950s and 1960s. Van Gelder's legacy is that of an individual who fully realized his dream to work in a creative capacity, and his story serves as inspiration for anyone striving to make their lifelong dreams a reality.

Sharing that story will be Michael Cuscuna, producer and founder of Mosaic Records, and Don Sickler, publisher and co-founder of Second Floor Music, who along with his wife Maureen, currently runs the Englewood Cliffs studio.

Product Development
PD09 - DOES AUTOMOTIVE AUDIO NEED A SYSTEMS APPROACH?
Thursday, October 17, 1:30 pm – 3:00 pm, Room 1E09

Moderator: **Roger Shively**, JJR Acoustics

Panelists: *John Busenitz*, Bose Corporation, Framingham, MA, USA
Pietro Massini, ASK
Greg Sikora, Harman International, Munich, Germany
Steve Temme, Listen Inc., Boston, MA,USA

Some component specifications do not translate to good system performance. A good example is resonance frequency. This often does not correspond well to performance characteristics in the automotive environment. Some potential improvements would be low-frequency SPL, or a parameter combination such as Fs/Qts or EBP (Fs/Qes). The system performance goals should drive the transducer component specification. Hence, this workshop will host leading industry experts in automotive audio and test/measurement solutions to discuss pros and cons of component vs. system specifications.

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

Recording & Production
RP09 - THE DOPPLER GANG—A PANEL OF PROS DISCUSS THE PROS AND CONS OF PITCH CHANGE
Thursday, October 17, 1:30 pm – 2:30 pm, Room 1E21

Moderators: **Anthony Agnello**, Eventide Inc., Little Ferry, NJ, USA
Richard Factor, Eventide Inc., Little Ferry, NJ, USA

Panelists: *Alex Case*, University of Massachusetts Lowell, Lowell, MA, USA
Bob Clearmountain, Mix This!, Pacific Palisades, CA, USA
George Massenburg, McGill University, Montreal, Quebec, Canada
Susan Rogers, Berklee College of Music, Boston, MA, USA
Tony Visconti, David Bowie, T.Rex

Pitch change processing is the basis for many tracking and mixing effects. Analog tape invited tape speed manipulation, and the effect became digitally available in the mid-70s. A broad set of pitch effects has evolved ever since, led in part by these panelists. Learn about their approaches and listen to the results, from obvious to subliminal, from logical to just plain wacky.

Special Event
SE06 - LUNCHTIME KEYNOTE: STEVE JORDAN
Thursday, October 17, 1:30 pm – 2:30 pm, Room 1E15+16

Presenter: **Steve Jordan**

Student Events & Career Development
SC10 - MATLAB PLUGIN AES STUDENT COMPETITION—OPEN DEMOS
Thursday, October 17, 1:30 pm – 2:30 pm, South Concourse A

The shortlisted finalists participating in the 2019 edition of the MATLAB Plugin AES Student Competition will provide live interactive demonstrations and answer questions on their MATLAB-based VST plugins.

To learn more about the competition visit aes.org/students/awards/mpsc/. This session follows up and complements the presentation-style showcase event taking place earlier on the day.

The following submissions were chosen by the judges to be presented in front of the audience. These projects are considered to win cash and software prizes. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2). Learn more about this competition at aes.org/students/awards.

- Edward Ly, "Inner Space," University of Aizu
- Domenico Andrea Giliberti, Festim Iseini, Nicola Ignazio Pelagalli, Alessandro Terenzi, "BEStEx - Bass Enhancer Stereo Expander," Universita Politecnica delle Marche
- Christian Steinmetz, "flowEQ," Universitat Pompeu Fabra
- Sean Newell, "Shift Drive," Belmont University
- John Kolar, "Dynamizer: Amplitude Modulator & Audio Effects Plugin," West Virginia University
- Michael Nuzzo, "Spectrum Pixelator," University of Massachusetts-Lowell

Audio Builders Workshop Booth Talks
ABT09 - LEARN TO SOLDER
Thursday, October 17, 1:30 pm – 2:00 pm, Booth 266

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

PMC Masters of Audio
PM02 - PMC MASTERS OF AUDIO: PMC, CAPITOL STUDIOS PRESENT : "MUSIC IN DOLBY ATMOS
Thursday, October 17, 1:30 pm – 2:30 pm, Room 1E06

Presenters: **Maurice Patist**, President of PMC USA
Nick Rives, Capitol Studios

Two years ago Capitol Studios partnered with PMC and Dolby to build the first "Dolby Atmos Music" studio in the world famous Capitol Studios Tower. Now hundreds of mixed tracks later, Amazon Music just announced the Atmos Streaming service allowing people to finally hear the results of this project. Nick Rives, engineer for Capitol Studios who mixed a multitude of these tracks, and Maurice Patist, President of PMC USA, will take you on a journey through Dolby Atmos Music, into what they believe is a new chapter in music history.

Software@AES
S14 - MELODYNE
Thursday, October 17, 1:30 pm – 2:00 pm
Software@AES Pavilion

Electronic Dance Music
EDM01 - MIXING EDM MASTERCLASS
Thursday, October 17, 2:00 pm – 3:00 pm, Room 1E12

Presenter: **Ariel Borujow**, Westward Music Group Inc.

Ariel Borujow is a multi platinum/Grammy winning mix engineer who has been making records for over 2 decades. Currently, Ariel serves on the New York committee for the Producer & Engineer Wing of the Recording Academy. He has spoken on countless panels and conducted many workshops both in the United States and Europe.

Ariel will be breaking down a full mix in a 45 minute presentation followed by a 15 minute Q&A. He will discuss all aspects of his approach from his mindset and communication with the client to the techniques that go into creating the feel while keeping the clients vision of the record. Today everyone has access to the same tools. By the time the mix engineer receives files, there is a clear direction. However, how those tools are used dictates the end result. The question becomes, how can a mixer use those tools to enhance the established direction and feel while making their client comfortable with the process?

Technical Committee Meeting
ARCHIVING, RESTORATION, AND DIGITAL LIBRARIES
Thursday, October 17, 2:00 pm – 3:00 pm, Room 1C02

AES Mix with the Masters Workshops
MM13 - TCHAD BLAKE
Thursday, October 17, 2:00 pm – 3:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP18 - BOLERO WIRELESS INTERCOM SYSTEM IN AES67 NETWORKS
Thursday, October 17, 2:00 pm – 2:30 pm
AoIP Pavilion Theater

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

Riedel Communications will provide an overview of their Bolero wireless intercom system, including the newest AES67 mode that facilitates the distribution of antennas over AES67 networks. Then, after a review of how to assemble and configure a Bolero standalone system, two people will be randomly selected from the audience to compete to see who can configure a system in the shortest amount of time.

Audio Builders Workshop Booth Talks
ABT10 - BASIC SIGMASTUDIO AND THE ANALOG DEVICES ADAU1701 SIGMADSP
Thursday, October 17, 2:00 pm – 2:30 pm, Booth 266

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

Electronic Dance Music Stage
EDJ04 - ALLEN & HEATH REELOP, XONE & HERCULES PRESENTS: BRIDGING THE GAP BETWEEN HOME PRODUCTION AND THE STAGE
Thursday, October 17, 2:00 pm – 3:00 pm
Electronic Dance Music & DJ Stage

Presenter: **Jamie Thompson**

Come learn what you will need to take your home music production to the stage and perform using tools like DJ mixers and midi controllers, as well as software programs like Ableton Live and Traktor Pro. There will be a live demonstration using all of these tools along with a Q&A segment to answer any questions you may have.

Live Production Stage
LS09 - THEATRICAL SOUND DESIGNERS AND COMPOSERS ASSOCIATION PANEL
Thursday, October 17, 2:00 pm – 2:45 pm
Live Production Stag

Presenters: **Ien DeNio**
Sam Kusnetz
Beth Lake
Emma Wilk

A presentation from AES and TSDCA on different working situations for theatre sound designers, from Broadway to off-Broadway to regional to storefront/low budget. The panel consists of Ien DeNio, Sam Kusnetz, Beth Lake, and Emma Wilk, all of whom are working designers and associates in and around New York City. The panelists are also members of TSDCA - the Theatrical Sound Designers and Composers Association - a national organization working to further the work and concerns of Sound Designers and Composers within the theatrical community.

Project Studio Expo Recording Stage
RS09 - GENELEC PRESENTS: NAVIGATING THE CIRCLES OF CONFUSION. MONITORING IN THE DIGITAL AGE
Thursday, October 17, 2:00 pm – 2:45 pm
Recording Stage

Presenters: **Will Eggleston**, Technical Marketing USA Genelec
Aki Mäkivirta, Genelec Oy, Iisalmi, Finland

Join Genelec R&D Director, Aki Makivirta and Will Eggleston, Technical Marketing USA as they discuss the obstacles facing modern production and monitoring environments, from stereo to immersive.

Software@AES
S15 - FABFILTER
Thursday, October 17, 2:00 pm – 2:30 pm
Software@AES Pavilion

Electronic Instrument Design & Application
EI03 - MODULAR SYNTHESIZER DESIGNS— FROM THE PAST TO FUTURE DIRECTIONS
Thursday, October 17, 2:15 pm – 3:45 pm, Room 1E17

Moderator: **Michael Bierylo**, Berklee College of Music, Brookline, MA, USA

Panelists: *Dan Green*
Andrew Ikenberry
Andrew Morelli
Dave Rossum

Interest in modular synthesizers has come full circle from the earliest instruments available in the 1960s to obscurity as newer technologies dominated the field, and on to the current renaissance of interest with the embrace of Eurorack as a standard format. Systems run the gamut from classic recreations to bold new deigns with some of the most innovative engineering coming from small, home grown developers. This session will feature the perspectives of industry leaders discussing the roots and future development of modular synthesis.

Game Audio & XR
GA10 - LESSONS LEARNED IN GAME AUDIO
Thursday, October 17, 2:15 pm – 3:15 pm, Room 1E13

Presenter: **Alex Wilmer**, Wilmer Sound, San Francisco, CA, USA
Sae Wilmer, Wilmer Sound, San Francisco, CA, USA

Alex Wilmer will present his guiding principles, learned during his career in Film, Console, Mo-bile, VR, and AR audio. He will share anecdotes to illustrate why these principles are crucial for success in the future of game audio. He will discuss the successes and the failures in the hopes that people can learn how to succeed and not make the same mistakes that he did!

Broadcast & Online Delivery
B08 - ADVANCED AUDIO FOR ATSC 3.0 BROADCAST
Thursday, October 17, 2:30 pm – 4:00 pm, Room 1E07

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

Panelists: *Robert Bleidt*, Fraunhofer USA Digital Media Technologies, San Jose, CA, USA
Tim Carroll, Dolby Laboratories, San Francisco, CA, USA
Jim Starzynski, NBCUniversal, New York, NY, USA; ATSC Group, Washington D.C.

2019 has become a watershed year for the new ATSC 3.0 terrestrial broadcast television standard, as multiple stations across the country have been granted full-time operation licenses from the FCC, and stations in 40 markets have committed to full-time ATSC 3.0 operation by the end of 2020.

This next-generation TV transmission system offers ultra HD video (higher visual resolution, increased video dynamic range and higher frame rates) along with superior over-the-air reception, including mobile and automotive capability.

Moreover, the new standard incorporates enhanced audio with 3D immersive capabilities (either channel-, object- or scene-based), along with personalized sound preferences, including dialog, dynamic range and loudness level adjustments.

The Republic of Korea has been broadcasting in ATSC 3.0 for nearly three years (prior to the 2018 Winter Olympics in Seoul), and most recently has presented live FIFA World Cup coverage in immersive audio.

Several of the world's foremost broadcast audio experts convene in this lively session to present the latest advancements and plans for ATSC 3.0, including the perspectives of broadcasters, SDOs and equipment manufacturers.

AoIP Pavilion
AIP19 - DESIGNING WITH DANTE AND AES67/SMPTE ST 2110
Thursday, October 17, 2:30 pm – 3:00 pm
AoIP Pavilion Theater

Presenter: **Patrick Killianey**, Audinate, Buena Park, CA, USA

In September 2019, Audinate added support for SMPTE ST 2110 in Dante devices. In this presentation, Audinate will share its vision on how Dante and AES67/SMPTE ST 2110 can be used together in a modern studio, including Dante Domain Manager. A tour of Dante's managed and unmanaged integration with open standards will lead to a discussion of the need for network security, Layer 3 traversal, and fault monitoring in real time. All will be laid out with practical diagrams of broadcast studios from location sound, to small studios and large properties for practical application.

Software@AES
S16 - ACON DIGITAL
Thursday, October 17, 2:30 pm – 3:00 pm
Software@AES Pavilion

Historical Event
H04 - SPIKE JONES: PREPOSTEROUS PRECISION
Thursday, October 17, 2:45 pm – 4:15 pm, Room 1E08

Chair: **Mike Wisland**, Utah Valley University, Orem, UT, USA

Panelists: *Arlen Card*, Utah Valley University, Orem, UT, USA
Leslie Ann Jones, Recording Engineer and Producer, Director of Music Recording and Scoring, Skywalker Sound, San Rafael, CA, USA
Emily Taggart

Professor Mike Wisland will present Spike's life including his childhood, early professional jobs, leading to the formation of his massively successful parody band, Spike Jones and his City Slickers. There will also be a video presentation by Emily Taggart on the bandboy of Spike's and Leslie Ann Jones Spike's daughter, who will also be present for a Q&A After the presentation.

Recording & Production
RP10 - TUNING THE VOCAL—WATCH, LISTEN, AND LEARN
Thursday, October 17, 2:45 pm – 3:45 pm
Room 1E21

Presenters: **Alex Case**, University of Massachusetts Lowell, Lowell, MA, USA
Ian Kagey, Berklee NYC, New York, NY, USA

Ian Kagey is a New York City-based engineer and producer constantly called upon to tune vocal tracks. Clients count on him to tune vocals while preserving emotion, timbre, and the artist's personality. Budgets demand that such care and precision happen quickly. In this live demonstration, you'll observe his approaches and tap into his thoughts as he takes a raw vocal up to professional tuning standards. In a parallel interview, by Alex U. Case, practical considerations are reviewed, detailing workflow and keyboard short cuts. Aesthetic issues are discussed, finding the balance between pitch perfection and musical expression.

Paper Session P12 — Thursday, Oct. 17
3:00 pm – 4:30 pm — South Concourse A

POSTERS: ROOM ACOUSTICS

3:00 pm

P12-1 Transparent Office Screens Based on Microperforated Foil—*Krzysztof Brawata,*[1] *Katarzyna Baruch,*[1] *Tadeusz Kamisinski,*[2] *Bartlomiej Chojnacki*[2]
[1] Gorycki&Sznyterman Sp. z o.o., Cracow, Poland
[2] AGH University of Science and Technology, Cracow, Poland

In recent years, providing comfortable working conditions in open office spaces has become a growing challenge. The ever-increasing demand for office work implies the emergence of ever new spaces and the need to use available space, which generates the need for proper interior design. There are many acoustic solutions available on the market that support the acoustic comfort in office spaces by ensuring appropriate levels of privacy and low levels of acoustic background. One of such solutions are desktop screens, which divide employees' space. These solutions are based mainly on sound absorbing materials, i.e., mineral wool, felt, as well as sound insulating ones, such as glass or MDF. The article presents methods of using microperforated foils for building acoustic screens. The influence of dimensions and parameters of microperforated foil were examined. The method of its assembly as well as the use of layered systems made of microperforated foil and sound insulating material were also considered in this paper.
Convention Paper 10282

3:00 pm

P12-2 A Novel Spatial Impulse Response Capture Technique for Realistic Artificial Reverberation in the 22.2 Multichannel Audio Format—*Jack Kelly, Richard King, Wieslaw Woszczyk*, McGill University, Montreal, QC, Canada, The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

As immersive media content and technology begin to enter the marketplace, the need for truly immersive spatial reverberation tools takes on a renewed significance. A novel spatial impulse response capture technique optimized for the 22.2 multichannel audio format is presented. The proposed technique seeks to offer a path for engineers who are interested in creating three-dimensional spatial reverberation through convolution. Its design is informed by three-dimensional microphone techniques for the channel-based capture of acoustic music. A technical description of the measurement system used is given. The processes by which the spatial impulse responses are captured and rendered, including deconvolution and loudness normalization, are described. Three venues that have been measured using the proposed technique are presented. Preliminary listening sessions suggest that the array is capable of delivering a convincing three-dimensional reproduction of several acoustic spaces with a high degree of fidelity. Future research into the perception of realism in spatial reverberation for immersive music production is discussed.
Convention Paper 10283

3:00 pm

P12-3 Impulse Response Simulation of a Small Room and in situ Measurements Validation—*Daniel Núñez-Solano, Virginia Puyana-Romero, Cristian Ordóñez-Andrade, Luis Bravo-Moncayo, Christiam Garzón-Pico*, Universidad de Las Américas, Quito, Ecuador

The study of reverberation time in room acoustics presents certain drawbacks when dealing with small spaces. In order to reduce the inaccuracies due to the lack of space for placing measurement devices, finite element methods become a good alternative to support measurement results or to predict the reverberation time on the bases of calculating impulse responses. This paper presents a comparison of the reverberation time obtained by means of in situ and simulated impulse responses. The impulse response is simulated using time-domain finite elements methods. The used room for measurements and simulations is a control room of Universidad de Las Americas. Results show a measured mean absolute error of 0.04 s compared to the computed reverberation time.
Convention Paper 10284

3:00 pm

P12-4 Calculation of Directivity Patterns from Spherical Microphone Array Recordings—*Carlotta Anemüller, Jürgen Herre*, International Audio Laboratories Erlangen, Erlangen, Germany

Taking into account the direction-dependent radiation of natural sound sources (such as musical instruments) can help to enhance auralization processing and thus improves the plausibility of simulated acoustical environments as, e.g., found in virtual reality (VR) systems. In order to quantify this direction-dependent behavior, usually so-called directivity patterns are used. This paper investigates two different methods that can be used to calculate directivity patterns from spherical microphone array recordings. A comparison between both calculation methods is performed based on the resulting directivity patterns. Furthermore, the directivity patterns of several musical instruments are analyzed and important measures are extracted. For all calculations, the publicly available anechoic microphone array measurements database recorded at the Technical University Berlin (TU Berlin) was used.
Convention Paper 10285

Sound Reinforcement
SR04 - IP BASED LIVE PRODUCTION INTERCOM SYSTEMS
Thursday, October 17, 3:00 pm – 5:00 pm, Room 1E11

Moderator: **Mac Kerr**

Panelists: **Joe Foley**, Capital City Sound
Brian Maddox
Brian Ready, CP Communications

IP based intercom systems have been around for a while but are now quickly replacing proprietary backbone schemes for both large matrix and small partyline systems in live production environments and studio installations. This session will look at best practices and considerations to design, deploy, and configure an IP based system as well as look at the many product offerings from a range of manufacturers.

Special Event
SE07 - TRIPLE THREAT: THE ART, PRODUCTION & TECHNOLOGY OF MAKING MUSIC
Thursday, October 17, 3:00 pm – 4:00 pm, Room 1E15+16

Moderator: **Paul Verna**

Panelists: *Steve Jordan*
Danny Kortchmar, Legendary GRAMMY nominated guitarist, songwriter and producer (Jackson Browne, Don Henley, James Taylor)

With a background of dozens of platinum record credits as a musician, artist, songwriter, producer and engineer, Danny Kortchmar sits down to discuss all aspects of making music both creatively and technically. From going to top commercial studios of the world, to building his own home studio to performing on stage with some of the top artists in the world, Kortchmar shares his secrets and experiences, digital and analog, then and now. Kortchmar credits include but not limited to: Billy Joel, Carole King, James Taylor, Jackson Browne, Toto, Foo Fighters, Don Henley, Neil Young

Danny Kortchmar's resume reads like a Who's Who of the music industry. A renown guitarist, producer, songwriter and session musician, Kortchmar has played, produced and written for James Taylor, Don Henley, Carole King, Linda Ronstadt and Jackson Browne, and many many more. As a songwriter, Kortchmar has either written alone or collaborated with numerous artists and has penned such indelible tracks like Don Henley's "Dirty Laundry," "All She Wants to Do Is Dance" and "New York Minute," as well as Jackson Browne's "Somebody's Baby" and "Shaky Town," to name a few. In the 1970's and 1980's, Kortchmar was a member of The Section, best known for both their studio and live work in support of some of the top selling singer/songwriters and solo singers in the history of music. Together, The Section helped define the sound of a generation. Recently, Kortchmar put members of The Section back together again and they now perform regularly live around the world as The Immediate Family.

Student Events & Career Development
SC11 - STUDENT RECORDING COMPETITION—PART 1
Thursday, October 17, 3:00 pm – 6:00 pm, Room 1E06

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.

Technical Committee Meeting
AUDIO FORENSICS
Thursday, October 17, 3:00 pm – 4:00 pm, Room 1C02

AES Standards Meeting
SC-05-02 WORKING GROUP ON AUDIO CONNECTORS
Thursday, October 17, 3:00 pm – 4:00 pm, Room 1C03

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.

AES Mix with the Masters Workshops
MM14 - JACK JOSEPH PUIG
Thursday, October 17, 3:00 pm – 4:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP20 - DESIGNING WITH DANTE AND AES67/SMPTE ST 2110
Thursday, October 17, 3:00 pm – 3:30 pm
AoIP Pavilion Theater

Presenters: **Cassidy Lee Phillips**, Imagine Communications, Plano, TX, USA
Tony Pearson, North Carolina State University

When NC State University (NCSU) was looking to upgrade the Pro-AV gear driving its Distance Education and Learning Technology Applications (DELTA) program, they decided to future-proof their infrastructure with a standards-based approach to IP, while continuing to leverage existing SDI gear. Using SMPTE ST 2110, NCSU significantly multiplied the capacity of their fiber infrastructure — where four single-mode fiber strands had supported four HD video channels, now just two fiber strands deliver 32 bidirectional HD channels. This presentation will share the real-world experiences and lessons learned from adopting AES67 and ST 2110 to successfully deploy a first-of-its-kind inter-campus Pro-AV system.

Audio Builders Workshop Booth Talks
ABT11 - REDUCE, REUSE, RECYCLE: AN APPROACH TO REPURPOSING BROKEN GEAR IN DIY BUILDS
Thursday, October 17, 3:00 pm – 3:30 pm, Booth 266

Presenter: **Jason Bitner**, Traffic Entertainment Group, Somerville, MA, USA

Live Production Stage
LS10 - SHURE PRESENTS: DAY IN THE LIFE OF AN RF COORDINATOR
Thursday, October 17, 3:00 pm – 3:45 pm
Live Production Stag

Project Studio Expo Recording Stage
RS10 - THE TECHNOLOGY BEHIND GRAMMY-WINNING RECORDS—MICHAEL BRAUER AND IGOR LEVIN ON THE ART AND SCIENCE OF PRO AUDIO
Thursday, October 17, 3:00 pm – 3:45 pm
Recording Stage

Presenters: **Michael Brauer**, Michael Brauer, New York, NY, USA
Igor Levin, Antelope Audio, New York, NY, USA

Two industry greats, Grammy Award-winning producer Michael Brauer and Antelope Audio founder Igor Levin, take the stage to share the secrets of their mutual success. Michael will expose his unique approach to the art of making hit records, revealing the role of legendary Antelope Audio equipment like the Atomic Reference Clock in his achievements. Igor will reflect on innovation and ingenuity as the driving forces behind his challenging products which showcase original developments like 6-transistor preamps, custom controllers, DSP + FPGA FX hardware and Acoustically Focused Clocking. In a dialog where art and science intertwine, expect to earn formidable insight and know-how from two consummate professionals whose work inevitably moves the music industry forward.

Software@AES
S17 - FL STUDIO
Thursday, October 17, 3:00 pm – 3:30 pm
Software@AES Pavilion

Electronic Dance Music
EDM02 - THE ART & ORIGINS OF SAMPLING: FROM VINYL TO DAW; FROM HIP-HOP TO DANCE MUSIC
Thursday, October 17, 3:15 pm – 4:15 pm, Room 1E12

Presenters: **Thavius Beck**
Nate Mars, Electronic Music Collective

Sampling is an art form that has evolved dramatically over the past decade with the changing land-scape in music technology and in the many emerging and long-established genres that incorporate sampling as the foundation for a song. In this workshop, presented by Electronic Music Collective instructors, you will be guided through the journey of sampling where techniques will be demon-strated using traditional vinyl record sampling all the way to using a sample as a key element of a wavetable synthesizer within a DAW.

*About Electronic Music Collective—Electronic Music Collective began with the idea that community is where we find connection, and that connections are what drive our artistic goals. We have created a music education community that will give you the support to develop fully as an artist, producer and performer, and to inspire, guide and develop your own sound. Located in a brand-new, state-of-the-art facility in historic downtown Manhattan, our curriculum is designed to teach you a new language of music that blurs the lines between musician, DJ, and producer, empowering you to be your best creative self.

Product Development
PD10 - DIAGNOSTICS FOR PRODUCTION VEHICLE AUDIO SYSTEMS
Thursday, October 17, 3:15 pm – 4:30 pm, Room 1E09

Presenter: **Jonathan Gerbet**, Klippel GmbH, Dresden, Germany
Steven Hutt, Equity Sound Investments, Bloomington, IN, USA
Steve Temme, Listen Inc., Boston, MA, USA

Audio systems in production vehicles are known to exhibit vehicle to vehicle performance variance [1]. The root causes of variance can include loudspeaker driver manufacturing tolerance, mounting issues such as missing or misaligned gaskets, or wrong loudspeaker drivers mounted in the system. A diagnostics method to compare actual production vehicle audio systems is defined along with a method for correction and calibration of production vehicle audio systems. The diagnostics procedure may be implemented at production end-of-line, at vehicle distribution center or at a dealer service center in the field after delivery to a customer.

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

Game Audio & XR
GA11 - CANCELED
Thursday, October 17, 3:30 pm – 4:30 pm, Room 1E13

AoIP Pavilion
AIP21 - TECHNICAL: SYNCHRONIZATION & ALIGNMENT (ST 2110 / AES67)
Thursday, October 17, 3:30 pm – 4:00 pm
AoIP Pavilion Theater

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

A deep dive into the secrets behind timing & synchronization of AES67 & ST 2110 streams. The meaning of PTP, media clocks, RTP, synchronization parameters (SDP) and the magic of stream alignment will be unveiled in this compact presentation.

Software@AES
S18 - ACCUSONUS
Thursday, October 17, 3:30 pm – 4:00 pm
Software@AES Pavilion

Broadcast & Online Delivery
B09 - THE TECHNICAL HISTORY OF WNYC
Thursday, October 17, 4:00 pm – 5:30 pm, Room 1E07

Presenters: **Andy Lanset**, Director of Archives, New York Public Radio
Steve Shultis, Chief Technology Officer, New York Public Radio
Jim Stagnitto, Director of Engineering, New York Public Radio

This year WNYC radio marked its 95th year on the air. The broadcaster's remarkable engineering story will be the focus of this presentation beginning with an auspicious launch by the Mayor of New York employing the exact same model Westinghouse transmitter as used by KDKA, Pittsburgh. Nearly a century later, WNYC is the flagship station for the nation's public radio networks as well as leading producer and distributor of audio content to listeners on multiple platforms.

This talk will cover the veteran broadcaster's pioneering history on both AM and FM, the evolution of its respective signal patterns and reach as well as WNYC's early work with shortwave, binaural broadcasting, early implementation of streaming and podcasting, and HD radio.

The stations engineering challenges post-9/11 and hurricane Sandy will also be discussed.

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

Electronic Instrument Design & Applications
EI04 - EMBEDDED COMPUTING AND INSTRUMENT DESIGN
Thursday, October 17, 4:00 pm – 5:30 pm, Room 1E17

Moderator: **Paul Lehrman**, Tufts University, Medford, MA, USA

Panelists: *Richard Graham*
Andrew Ikenberry
Denis Labrecque
Teresa Marrin Nakra
Dave Rossum

Embedded computing systems based on common platforms such as Arduino, Raspberry Pi, Bea-gleBone, and Bela offer powerful, flexible platforms for prototyping new electronic instrument de-signs. This session will offer perspectives from both research and independent builder communi-ties on innovative use cases and future directions.

Recording & production
RP11 - PITCH SHIFT FXPERTISE—RECORD AND MIX STRATEGIES FOR THE BROAD RANGE PITCH EFFECT POSSIBILITIES
Thursday, October 17, 4:00 pm – 5:30 pm, Room 1E21

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

Pitch shift as an effect has been a part of sound recording from the very beginning of audio and continues to evolve briskly today. Recording at one speed but playing back at another has been the pitch shift modus operandi across all analog formats—cylinders, disks, and tape. Digital audio continued these time-domain techniques, exploiting sample rate differences between record and playback. The digital luxury of frequency-domain pitch shift, offering new effects possibilities, became an important part of the pop engineer's tool kit in the late 90s.

From this are born a seemingly endless set of pitch-based outcomes. But what are we to make of the aesthetic possibilities? How do we organize its creative potential? Alex U. Case auditions a curated set iconic examples of pitch shifting effects in pop music and analyses them through multiple lenses—music, signal processing and psychoacoustics—to define the full range of the effect, giving it structure, and empowering you to develop your own strategies for the use of Pitch FX in your next project.

Technical Committee Meeting
SIGNAL PROCESSING
Thursday, October 17, 4:00 pm – 5:00 pm, Room 1C02

AoIP Pavilion
AIP22 - AUDIO META DATA TRANSPORT
Thursday, October 17, 4:00 pm – 4:30 pm
AoIP Pavilion Theater

Presenter: **Kent Terry**, Dolby Laboratories Inc., San Francisco, CA, USA

Audio metadata, particularly dynamic, time varying audio metadata, is now a requirement for many audio applications. As AoIP systems use IP connections many possibilities for metadata transport exist, however standards in many areas are lacking which can lead to multiple solutions and market fragmentation. This session will discuss current and emerging IP based audio metadata transport standards that support existing AoIP standards and allow interoperability between AoIP systems and devices. Standards relevant to the ST 2110 suite and AES70 will be covered.

AES Standards Meeting
SC-04-08 WORKING GROUP ON MEASUREMENT AND EQUALIZATION OF SOUND SYSTEMS IN ROOMS
Thursday, October 17, 4:00 pm – 5:30 pm, Room 1C03

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.

AES Mix with the Masters Workshops
MM15 - LESLIE BRATHWAITE
Thursday, October 17, 4:00 pm – 5:00 pm
Mix with the Masters Workshop Stage

Electronic Dance Music Stage
EDJ05 - IZOTOPE PRESENTS: MASTERING EDM
Thursday, October 17, 4:00 pm – 4:45 pm
Electronic Dance Music & DJ Stage

Presenter: **Alex Psaroudakis**, Alex Psaroudakis Mastering, Brooklyn, NY, USA

• The dilema between detrimental loudness vs efficient masters on loud PA playback for clubs and festivals
• Gain staging to maintain punch and integrity for electronic music mastering
• Choosing limiters for electronic music
• The importance of understanding the aesthetics of all of the different electronic music and which process will serve each best

Live Production Stage
LS11 - OBJECT BASED AUDIO FOR MUSICALS—CASE STUDY: BROADWAY BOUNTY HUNTER
Thursday, October 17, 4:00 pm – 4:45 pm
Live Production Stag

Presenters: **Cody Spencer**
Jesse Stevens, L-Acoustics

Sound Designer Cody Spencer joins the panel for a case study of his work on the recent Joe Iconis musical "Broadway Bounty Hunter." Spencer chose an object-based approach to the sound design, which allowed for increased clarity, control, and new creative possibilities.

Project Studio Expo Recording Stage
RS11 - WHEN LOUD IS NOT LOUD: WHAT YOU NEED TO KNOW ABOUT LOUDNESS MEASUREMENT TODAY
Thursday, October 17, 4:00 pm – 4:45 pm
Recording Stage

Presenter: **Alex Kosiorek**, Central Sound at Arizona PBS, Phoenix, AZ, USA; Arizona State University, Phoenix, AZ, USA

With streaming dominating the music listening landscape, it is time to revisit both what loudness actually is and how to manage it. Companies such as Apple, YouTube, Spotify and others each have their own measurement standards and loudness targets. Whether it is music or spoken word (such as podcasts), care is needed to preserve the artistic intent of the content's creators. It is critical that producers, recording, mixing and mastering engineers understand what truly is at stake, and how to read, measure and manage the loudness of audio files. Join representatives from four highly regarded audio tech companies who will inform and enlighten about the proper use of today's loudness meters and measurement tools.

Software@AES
S19 - KILOHEARTS
Thursday, October 17, 4:00 pm – 4:30 pm
Software@AES Pavilion

Archiving & Restoration
AR01 - LONG TERM PRESERVATION OF AUDIO ASSETS (AN AES SPECIAL EVENT)
Thursday, October 17, 4:30 pm – 5:30 pm, Room 1E15+16

Moderator: **Jessica Thompson**, Jessica Thompson Audio, Berkeley, CA, USA

Panelists: *Jeff Balding*, Recording Academy Producers & Engineers Wing
Rob Friedrich, Library of Congress
Jamie Howarth, Plangent Process
Pat Kraus, UMG
Greg Parkin, Iron Mountain
Cheryl Pawelski, Omnivore Recording
Toby Seay, Drexel University; IASAsA

Throughout the history of the recorded music industry, masters have burned, been lost in floods, been mislabeled and misfiled, neglected, forgotten, even systematically destroyed to salvage the raw materials. This panel is an opportunity to learn from the past and move the conversation forward, addressing current challenges with long term preservation of audio assets. Beyond rehashing well-established best practices, panelists will discuss barriers to preservation including technical hurdles, cost, long term storage, deteriorating media, maintaining legacy playback equipment, legalities, and the very simple fact that we cannot and will not save everything.

Acoustics & Psychoacoustics
AP04 - CIRCLES OF CONFUSION
Thursday, October 17, 4:30 pm – 5:30 pm, Room 1E13

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

Panelists: **Sean Olive**, Harman International, Northridge, CA, USA
Susan Rogers, Berklee College of Music, Boston, MA, USA

Circles of confusion in pro audio are replacing technical limitations with cognitive limitations. Without proper anchoring of spectral balance and level, drifting over time is foreseeable in self-referenced systems, thereby putting legacy recordings at the risk of sounding dated for no good reason.

The panel will discuss monitoring requirements that stand the test of time, recent studies on active sensing, between listener variation and "slow listening"; and a possible revision of ITU-R BS.1116. The topics are addressed from a more practical perspective in Friday's AP06 session.

Audio Builders Workshop
AB03 - PROJECT TO PRODUCT
Thursday, October 17, 4:30 pm – 5:30 pm, Room 1E09

Chair: **Owen Curtin**, Audio Builders Workshop, Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA

Panelists: *Robert-Eric Gaskell*, McGill University, Montreal, QC, Canada
Peterson Goodwyn, DIY Recording Equipment, Philadephia, PA, USA
Denis Labrecque, DeLab Consulting, Half Moon Bay, CA, USA
Brewster LaMacchia, Clockworks Signal Processing LLC, Andover, MA, USA

Techniques and Resources for Turning Your DIY Hobby into a Scalable Business.

Electronic Dance Music
EDM03 - REMIXING—BREAKING THE ILLUSION
Thursday, October 17, 4:30 pm – 5:30 pm, Room 1E12

Presenter: **Rick Snoman**, Dance Music Productio, Manchester, UK

Remixing now forms the cornerstone of artist and music marketing. Both labels are artists solicit remixers to extend the lifespan of their music and expose it to a different demographic. But there is an illusion that remixing is easy because the remixer is supplied with many of the musical elements from the original but the reality is very different.

In this workshop, we walk through the art of a remix, from the initial decisions of what to keep, how to develop on the original idea, and how to imprint your own identity onto it.

Immersive & Spatial Audio
IS04 - 3D AUDIO PHILOSOPHIES & TECHNIQUES FOR COMMERCIAL MUSIC
Thursday, October 17, 4:30 pm – 5:30 pm, Room 1E08

Presenter: **Bt Gibbs**, Skyline Entertainment and Publishing, Morgan Hill, CA, USA; Tool Shed Studios, Morgan Hill, CA, USA

3D Audio (360 Spatial) for immersive content has made massive strides forward in just the first five months of 2019. However, 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 (headphone) delivery options for spatial audio as a topic of discussion for many major hi-res audio delivery platforms, commercial music delivery options are coming very soon. The ability for commercial artists to deliver studio quality audio (if not MQA) to consumers with an "in-the-studio" experience soon will be delivered in ambisonic formats.

This presentation will demonstrate studio sessions delivered in 360 video and stereo mixes translated to static (non-HRTF) 360 audio, which was originally captured for standard stereo delivery through traditional streaming and download sites. All of this audio is prepared to be delivered in a simultaneous (and rapid) turn around from pre-production to final masters delivered on both 360 and stereo platforms. To do so, requires planning in even in the earliest of (pre-production) stages prior to actual recording.

Special Event
SE08 - DTVAG FORUM: AUDIO FOR A NEW TELEVISION LANDSCAPE
Thursday, October 17, 4:30 pm – 5:30 pm, Room 1E10

Presenters: **Roger Charlesworth**, DTV Audio Group, New York, NY, USA
Tim Carroll, Dolby Laboratories, San Francisco, CA, USA
Scott Kramer, Netflix, Los Angeles, CA, USA
Sean Richardson, Starz Entertainment, Denver, CO, USA
Tom Sahara, Turner Sports, Atlanta, GA, USA
Jim Starzynski, NBCUniversal, New York, NY, USA; ATSC Group, Washington D.C.

We appear to have now entered the post-television, television era. In a few short years, the entire nature of television distribution and consumption has changed so significantly as to be unrecognizable. Ubiquitous and inexpensive wireless and broadband networking; smart TVs and mobile devices; and massively-scalable cloud computing have created a completely new entertainment distribution system, upending the traditional broadcast model, and changing viewing habits forever. The transition from "hardwired" to "virtualized" distribution has expanded 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 advent of affordable consumer 4K and HDR on smart TVs and other devices has radically transformed the at-home viewing experience. Combined with the story-telling power of premium episodic content and streaming movies, upscale home viewing has supplanted 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 immersive surround.

The dramatic resurgence of surround sound, and emerging interest in next-generation enhanced-surround, is built on the ability to virtualize surround presentations over a growing range of devices and environments including increasingly sophisticated immersive-audio-capable soundbars and TV sets, alongside enhanced surround virtualizing headphones, earbuds and mobile devices.

Please join us for a discussion of how the post-television era is re-inventing television sound.

The DTV Audio Group Forum at AES is produced in association with the Sports Video Group and is sponsored by: Brainstorm, Calrec, Dale Pro Audio, Dolby, Lawo, Sanken, Shure

"The rule book for television distribution is being completely re-written. The migration away from traditional broadcasting to IP delivery continues 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 devices and environments. Please join the DTVAG for a discussion of these and other important television audio issues."

—Roger Charlesworth, Executive Director, DTV Audio Group

AoIP Pavilion
AIP23 - INNOVATION THROUGH OPEN TECHNOLOGIES
Thursday, October 17, 4:30 pm – 5:00 pm
AoIP Pavilion Theater

Presenter: **Nestor Amaya**, Ross Video / COVELOZ Technologies, Ottawa, ON, Canada

Where would today's Cloud, smartphones and embedded IoT engines be without Linux? And could we have gotten here with a proprietary OS such as Windows? Similarly, this presentation argues that Pro Audio/Video applications need open technologies to serve the unique needs of their users. Our markets need a fully open stack, beyond just AES67/ST 2110 transport, to reap the benefits of our investment in

A/V networking technologies. Learn how NMOS, EmBER+ and similar open technologies enable you to innovate and serve your users better when compared to proprietary control systems.

Software@AES
S20 - BEST SERVICE
Thursday, October 17, 4:30 pm – 5:00 pm
Software@AES Pavilion

AES Mix with the Masters Workshops
MM16 - ANDY WALLACE
Thursday, October 17, 5:00 pm – 6:00 pm
Mix with the Masters Workshop Stage

Live Production Stage
LS12 - DIGICO PRESENTS: BROADWAY TO BLACK BOX THEATRE MIXING
Thursday, October 17, 5:00 pm – 5:45 pm
Live Production Stag

Moderator: **Matt Larson**, DiGiCo- Group One Limited National Sales Manager, Farmingdale, NY

Panelists: *Lew Mead*, AutographA2D Managing Director, NY, USA
Dan Page

Benefits of DiGiCo Theater Software and Workflow

Brief overview and backstory that illustrates how the pioneering theatre software was conceived and developed by Autograph and DiGiCo. We will demonstrate the T-Software and how it's implemented in musical theatre productions. This session will cover an in-depth introduction to all theatre-specific features and how they can be used in the demanding world of theatre sound design and discuss controlling other external systems via the DiGiCo platform via protocols as OSC, GPIO, MIDI and the discuss what the future may bring in Sound Design. Attendees will leave understanding the additional tools that the T-Software can provide for fast & efficient programing of a simple to complex show.

Project Studio Expo Recording Stage
RS12 - BEYOND BASICS IN VOCAL RECORDING & PRODUCTION: ENGINEERING WITH FOCUS ON THE CREATIVE PROCESS
Thursday, October 17, 5:00 pm – 5:45 pm
Recording Stage

Presenters: **Neal Cappellino**, Multiple Grammy-winning Engineer
Mike Picotte, Producer Engineer Sweetwater Sound, Fort Wayne, IN, USA
Jonathan Pines, Rupert Neve Designs, Wimberley, TX, USA; sE Electronics

Master the basics in order to facilitate the Creative Process that's happening on the other side of the glass. Grammy award winning engineer Neal Cappellino and Producer-Engineers Mike Picotte and Jonathan Pines will detail five aspects of vocal recording & production they see as foundational skills Engineers must be fluid with in order to be effective creative collaborators.

iTeam = Interpersonal, Technical, Environmental, Administrative, and Musical; skills that combine to foster a supportive and successful collaboration.

Sponsored by: Sweetwater Sound, sE Electronics, and Rupert Neve Designs

Historical Event
H05 - LIMITED ENGAGEMENT SCREENING, TOM DOWD AND THE LANGUAGE OF MUSIC
Thursday, October 18, 5:30 pm – 7:00 pm
Dolby Theater, 1350 Ave. of the Americas

Filmmaker Mark Moormann premiered this independently produced feature-length documentary at the 2003 Sundance Film Festival, with its international premiere at the 2003 Toronto International Film Festival. It has screened at film festivals, theaters, and television screens around the world to widespread critical acclaim.

A long-time engineer and producer for Atlantic Records, Tom Dowd was responsible for some of the most important R&B, rock, and jazz records ever made. In his own words, Tom relates how he went from working on the Manhattan Project, while still high school age, to recording some of the greatest music ever made over the last half of the 20th Century.

In the course of the film we see interviews with Ahmet Ertegun, Jerry Wexler, Al Schmitt, Eric Clapton, Aretha Franklin, Ray Charles, Les Paul, Phil Ramone, Joe Bonamassa, and many more of Tom's musical collaborators.

After the film screening, we will be honored to have Tom's daughter Dana Dowd share memories and answer questions.

Seating is limited and by advance ticket only.

NOTE: No food or drink (including water) is allowed in the Dolby Theater, so come hydrated and fed.

Doors will open shortly after 5, the program will start promptly at 5:30. Movie at 5:35 pm, Q&A will follow at 7:05; event ends at 8:00 sharp.

Presented by the AES Historical Track in conjunction with Language of Music Films LLC and Dana Dowd

Special Event
SE09 - HEYSER LECTURE
Thursday, October 17, 6:30 pm – 8:00 pm, Room 1E15+16

Lecturer: **Louis Fielder**, Retired, Millbrae, 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 **Louis Fielder**.

Psychoacoustics Applied to Dynamic-Range and Nonlinear-Distortion Assessment

The psychoacoustics of noise detection, measurements of noise in the digital-audio recording storage reproduction chain, and measurements of peak-acoustic pressures in music performances are combined to determine the requirements for noise-free reproduction of music. It is found that the required ratio between the maximum reproduction levels and the perceived audibility of noise can be as much as 124 decibels. When more practical circumstances are considered, this requirement is shown to drop to more feasible values. Next, the concept of auditory masking is introduced to allow for the assessment of nonlinear distortions in digital-audio conversion systems operating at low signal levels and then several examples of digital-audio conversion systems are examined. Finally, an expanded use of masking and a model to calculate a total nonlinear-distortion audibility number are used to determine the audibility of nonlinear distortions in headphones when driven by sine-wave signals at or below 500 Hz. Examples of headphone distortion assessment are examined and extension of this measurement technique to low-frequency loudspeaker evaluation is also discussed.

Broadcast & Online Delivery
B10 - TOUR OF STITCHER STUDIOS
Thursday, October 17, 7:00 pm – 8:00 pm
Stitcher Studios, 5 Bryant Park (6th Ave. & 40th St.)

Podcasts currently reach over 73 million affluent, highly educated and exceedingly mobile listeners in the U.S. alone. Stitcher is among the earliest, the most creative, and most successful proponents of this vibrant medium. Providing a 360-degree suite of production, distribution, and monetization services to artists, entertainers, and thought leaders Stitcher is the parent company of leading comedy podcasting network, Earwolf and of Midroll, the company's advertising and sales arm. To provide their programing creators and producers with the optimal production environment, Stitcher's management team engaged WSDG Walters-Storyk Design Group to create production facilities in both its NYC and LA offices.

The recently completed NYC studio complex occupies a 2000 square foot sector of the firm's 20,000 square foot space in a midtown Manhattan office building overlooking Bryant Park. The suite of three studios, two edit rooms, and two additional ISO booths can accommodate anything from a one-on-one interview to an 8-person roundtable and live music recording sessions in Studio A. Stitcher Chief Engineer John DeLore reports that "Superb sound quality was a major design priority, and reaching that goal began with the room design. WSDG understood that vital concern and specified complete room-within-room, floating floor isolation to preclude any leakage."

WSDG Partner/Project Manager, Romina Larregina believes Stitcher's studios epitomize the concept of uncompromising attention to detail. "Decoupled floors, wall and ceiling structures were implemented to insure the level of isolation required between the various studios positioned adjacent to each other," she says. "A clean, bright, straightforward floor plan optimizes every inch of allotted floor space with the most appropriate acoustic treatments to further enhance the overall high quality sound environment. Studio A was designed to set frequent musical guests completely at ease for their performance sessions. Stitcher Studios represent a new standard for podcast production."

Limited to 20 people; tickets available at Registration

Paper Session P13 — **Friday, Oct. 18**
9:00 am – 11:00 pm — **Room 1E10**

SPATIAL AUDIO, PART 2

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

9:00 am

P13-1 Simplified Source Directivity Rendering in Acoustic Virtual Reality Using the Directivity Sample Combination—*Georg Götz, Ville Pulkki*, Aalto University, Espoo, Finland

This contribution proposes a simplified rendering of source directivity patterns for the simulation and auralization of auditory scenes consisting of multiple listeners or sources. It is based on applying directivity filters of arbitrary directivity patterns at multiple, supposedly important directions, and approximating the filter outputs of intermediate directions by interpolation. This reduces the amount of required filtering operations considerably and thus increases the computational efficiency of the auralization. As a proof of concept, the simplification is evaluated from a technical as well as from a perceptual point of view for one specific use case. The promising results suggest further studies of the proposed simplification in the future to assess its applicability to more complex scenarios.
Convention Paper 10286

9:30 am

P13-2 Classification of HRTFs Using Perceptually Meaningful Frequency Arrays—*Nolan Eley*, New York University, New York, NY, USA

Head-related transfer functions (HRTFs) are essential in binaural audio. Because HRTFs are highly individualized and difficult to acquire, much research has been devoted towards improving HRTF performance for the general population. Such research requires a valid and robust method for classifying and comparing HRTFs. This study used a k-nearest neighbor (KNN) classifier to evaluate the ability of several different frequency arrays to characterize HRTFs. The perceptual impact of these frequency arrays was evaluated through a subjective test. Mel-frequency arrays showed the best results in the KNN classification tests while the subjective test results were inconclusive.
Convention Paper 10288

10:00 am

P13-3 An HRTF Based Approach towards Binaural Sound Source Localization—*Kaushik Sunder*,[1] *Yuxiang Wang*[2]
[1] Embody VR, Mountain View, CA, USA
[2] Rochester Institute of Technology, Rochester, NY, USA

With the evolution of smart headphones, hearables, and hearing aids there is a need for technologies to improve situational awareness. The device needs to constantly monitor the real world events and cue the listener to stay aware of the outside world. In this paper we develop a technique to identify the exact location of the dominant sound source using the unique spectral and temporal features listener's head-related transfer functions (HRTFs). Unlike most state-of-the-art beamforming technologies, this method localizes the sound source using just two microphones thereby reducing the cost and complexity of this technology. An experimental framework is setup at the EmbodyVR anechoic chamber, and hearing aid recordings are carried out for several different trajectories, SNRs, and turn-rates. Results indicate that the source localization algorithms perform well for dynamic moving sources for different SNR levels.
Convention Paper 10289
Paper presented by Yuxiang Wang

10:30 am

P13-4 Physical Controllers vs. Hand-and-Gesture Tracking: Control Scheme Evaluation for VR Audio Mixing—*Justin Bennington, Doyuen Ko*, Belmont University, Nashville, TN, USA

This paper investigates potential differences in performance for both physical and hand-and-gesture control within a Virtual Reality (VR) audio mixing environment. The test was designed to draw upon prior evaluations of control schemes for audio mixing while presenting sound sources to the user for both controller schemes within VR. A VR audio mixing interface was developed in order to facilitate a subjective evaluation of two control schemes. Response data was analyzed with t- and ANOVA tests. Physical controllers were generally rated higher than the hand-and-gesture controls in terms of perceived accuracy, efficiency, and satisfaction. No significant difference in task completion time for either control scheme was found. The test participants largely preferred the physical controllers over the hand-and-gesture control scheme. There were no significant differences in the ability to make adjustments in general when comparing groups of more experienced and less experienced audio engineers.
Convention Paper 10290

Engineering Briefs Session EB01 **Friday, Oct. 18**
9:00 am – 10:00 am **Room 1E11**

RECORDING AND PRODUCTION

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

9:00 am

EB01-1 Recording and Mixing of Classical Music Using Non-Adjacent Spherical Microphone Arrays and Audio Source Separation Algorithms—*Eduardo Patricio, Mateusz Skrok, Tomasz Zernicki*, Zylia sp. z o.o., Poznan, Poland

The authors present a novel approach to recording classical music, making use of non-adjacent 3rd order Ambisonics microphone arrays. The flexible combination of source separated signals with varied degrees of beamforming focus enable independent levels control, while maintaining the spatial coherence and reverberation qualities of the recorded spaces. The non-coincidental arriving locations of multiple arrays allow for post-production manipulations without disrupting the inherent classical music logic that values the overall sound as opposed to individual single sound sources. In addition, this method employs portable and lightweight equipment to record decorrelated signals, which can be mixed in surround formats with enhanced sense of depth.
Engineering Brief 525

9:15 am

EB01-2 Exploring Preference for Multitrack Mixes Using Statistical Analysis of MIR and Textual Features—*Joseph Colonel, Joshua D. Reiss*, Queen Mary University of London, London, UK

We investigate listener preference in multitrack music production using the Mix Evaluation Dataset, comprised of 184 mixes across 19 songs. Features are extracted from verses and choruses of stereo mixdowns. Each observation is associated with an average listener preference rating and standard deviation of preference ratings. Principal component analysis is performed to analyze how mixes vary within the feature space. We demonstrate that virtually no correlation is found between the embedded features and either average preference or standard deviation of preference. We instead propose using principal component projections as a semantic embedding space by associating each observation with listener comments from the Mix Evaluation Dataset. Initial results disagree with simple descriptions such as “width” or “loudness” for principal component axes.
Engineering Brief 526

9:30 am

EB01-3 Machine Learning Multitrack Gain Mixing of Drums—*Dave Moffat, Mark Sandler*, Queen Mary University of London, London, UK

There is a body of work in the field of intelligent music production covering a range of specific audio effects. However, there is a distinct lack of any purely machine learning approaches to automatic mixing. This could be due to a lack of suitable data. This paper presents an approach to used human produced audio mixes, along with their source multitrack, to produce the set of mix parameters. The focus will be entirely on the gain mixing of audio drum tracks. Using existing reverse engineering of music production gain parameters, a target mix gain parameter is identified, and these results are fed into a number of machine learning algorithms, along with audio feature vectors of each audio track. This allow for a machine learning prediction approach to audio gain mixing. A random forest approach is taken to perform a multiple output prediction. The prediction results of the random forest approach are then compared to a number of other published automatic gain mixing approaches. The results demonstrate that the random forest gain mixing approach performs similarly to that of a human engineer and outperforms the existing gain mixing approaches.
Engineering Brief 527

9:45 am

EB01-4 Why Microphone Arrays Are Not Better than Single-Diaphragm Microphones with Regard to Their Single Channel Output Quality—*Helmut Wittek, Hannes Dieterle*, SCHOEPS Mikrofone GmbH, Karlsruhe, Germany

A comparison of the directional characteristics of single-diaphragm vs multi-microphone arrays is performed on the basis of frequency response and polar diagram measurements. The simple underlying question was: Is a conventional first-order pressure-gradient microphone better than an M/S array or an Ambisonics microphone regarding the quality of their individual outputs? The study reveals significant differences and a clear superiority of single-diaphragm microphones regarding the smoothness of on- and off-axis curves which is believed to highly correlate with timbral fidelity. Array microphones, on the other hand, can potentially create variable patterns and a higher order directivity.
[Presentation only; not in E-Library

10:00 am

EB01-5 Predicting Objective Difficulty in Peak Identification Task of Technical Ear Training—*Atsushi Marui, Toru Kamekawa*, Tokyo University of the Arts, Adachi, Tokyo, Japan

Technical ear training is a method to improve the ability to focus on a specific sound attribute and to communicate using the vocabularies and units shared in Audio Engineering. In designing the successful course in a sound engineers’ educational institution, it is essential to have a gradual increase in the task difficulty. In this e-Brief, the authors investigated creating a predictive model of objective difficulty for a given music excerpt when it is used in a peak identification task of technical ear training. The models consisting of six or seven acoustic features, including statistics on attack transients and power spectrum, showed overall better results.
Engineering Brief 565

Engineering Briefs Session EB02 **Friday, Oct. 18**
9:00 am – 10:30 am **South Concourse A**

POSTERS: APPLICATIONS IN AUDIO

9:00 am

EB02-1 Withdrawn

9:00 am

EB02-2 A Latency Measurement Method for Networked Music Performances—*Robert Hupke,[1] Sripathi Sridhar,[2] Andrea Genovese,[2] Marcel Nophut,[1] Stephan Preihs,[1] Tom Beyer,[2] Agnieszka Roginska,[2] Jürgen Peissig[1]*
[1] Leibniz Universität Hannover, Hannover, Germany

[2] New York University, New York, NY, USA

The New York University and the Leibniz University Hannover are working on future immersive Networked Music Performances. One of the biggest challenges of audio data transmission over IP-based networks is latency, which can affect the interplay of the participants. In this contribution, two metronomes, utilizing the Global Positioning System to generate a globally synchronized click signal, were used as a tool to determine delay times in the data transmission between both universities with high precision. The aim of this first study is to validate the proposed method by obtaining insights into transmission latency as well as latency fluctuations and asymmetries. This work also serves as baseline for future studies and helps to establish an effective connection between the two institutions.
Engineering Brief 529

9:00 am

EB02-3 An Investigation into the Effectiveness of Room Adaptation Systems: Listening Test Results—*Pei Yu, Ziyun Liu, Shufeng Zhang, Yong Shen, Nanjing University, Nanjing, Jiangsu Province, China*

Loudspeaker-room interactions are well known for affecting the perceived sound quality of low frequencies. To solve this problem, different room adaptation systems for adapting a loudspeaker to its acoustic environment have been developed. In this study two listening tests were performed to assess the effectiveness of four different room adaptation systems under different circumstances. The factors investigated include the listening room, loudspeaker, listening position, and listener. The results indicate that listeners' preference for different adaptation systems is affected by the specific acoustic environment. It was found that the adaptation system based on acoustic power measurement proved to be more preferred, also with stable performance.
Engineering Brief 530

9:00 am

EB02-4 Evaluating Four Variants of Sine Sweep Techniques for Their Resilience to Noise in Room Acoustic Measurements—*Eric Segerstrom,[1] Ming-Lun Lee,[2] Steve Philbert[2]*
[1] Rensselaer Polytechnic Institute, Troy, NY, USA
[2] University of Rochester, Rochester, NY, USA

The sine sweep is one of the most effective methods for measuring room impulse responses; however, ambient room noise or unpredictable impulsive noises can negatively affect the quality of the measurement. This study evaluates four different variants of sine sweeps techniques for their resilience to noise when used as an excitation signal in room impulse response measurements: linear, exponential, noise whitened, and minimum noise. The result shows that in a pseudo-anechoic environment, exponential and linear sine sweeps are most resilient to impulsive noise among the four sweeps, while none of the evaluated sine sweeps are resilient to impulsive noise in an acoustically untreated room. Additionally, it is shown that minimum noise sine sweeps are most resilient to ambient noise.
Engineering Brief 531

9:00 am

EB02-5 Perceptually Affecting Electrical Properties of Headphone Cable—Factor Hunting Approach— *Akihiko Yoneya*, Nagoya Institute of Technology, Nagoya, Aichi-pref., Japan

An approach to find the cause of the perceptual sound quality change by headphone cable has been proposed. This is a method of verifying the validity of the selected candidate by selecting candidate factors from the measurement results, simulating them by digital signal processing, and evaluating the simulated sounds by audition. In the headphone cable, it was found that the factor is that the inductance changes due to the flowing current. It has become clear from the experimental results that changes in transfer characteristics are very sensitively affecting the perceptual sound quality.
Engineering Brief 532

9:00 am

EB02-6 An Investigation into the Location and Number of Microphone Measurements Necessary for Efficient Active Control of Low-Frequency Sound Fields in Listening Rooms—*Tom Bell,[1,2] Filippo Maria Fazi[2]*
[1] Bowers & Wilkins, Southwater, West Sussex, UK
[2] University of Southampton, Southampton, Hampshire, UK

The purpose of this investigation is to understand the minimum number of control microphone measurements needed and their optimal placement to achieve effective active control of the low-frequency sound field over a listening area in a rectangular room. An analytical method was used to model the transfer functions the loudspeakers and a 3-dimensional array of 75 virtual microphones. A least-squares approach was used to create one filter per sound source from a varying number and arrangement of these measurements, with the goal to minimize the error between the reproduced sound field and the target. The investigation shows once enough measurements are taken there is a clear diminishing return in the effectiveness of the filters versus the number of measurements needed.
Presentation only; not in E-Library

9:00 am

EB02-7 Measuring Speech Intelligibility Using Head-Oriented Binaural Room Impulse Responses—*Allison Lam,[1] Ming-Lun Lee,[2] Steve Philbert[2]*
[1] Tufts University, Medford, MA, USA
[2] University of Rochester, Rochester, NY, USA

Speech intelligibility/speech clarity is important in any setting in which information is verbally communicated. More specifically, a high level of speech intelligibility is crucial in classrooms to allow teachers to effectively communicate with their students. Given the importance of speech intelligibility in learning environments, several studies have analyzed how accurately the standard method of measuring clarity predicts the level of speech intelligibility in a room. In the context of speech measurements, C50 has been widely used to measure clarity. Instead of using a standard omnidirectional microphone to record room impulse responses for clarity measurements, this study examines the effectiveness of room impulse responses measured with a binaural dummy head. The data collected for this experiment show that C50 measurements differ between the left and right channels by varying amounts based on the dummy head's position in the room and head orientation. To further investigate the effectiveness of binaural C50 measurements in comparison to the effectiveness of omnidirectional C50 measurements, this research explores the results of psychoacoustic testing to determine which recording method more consistently predicts human speech intelligibility. These results, combined with qualitative observations, predict how precisely acousticians are able to measure C50.
Engineering Brief 533

9:00 am

EB02-8 Compensation Filters for Excess Exciter Excursion on Flat-Panel Loudspeakers—*David Anderson*, University of Pittsburgh, Pittsburgh, PA, USA

Inertial exciters are used to actuate a surface into bending vibration, producing sound, but often have a high-Q resonance that can cause the exciter magnet to displace enough to contact the bending panel. The magnet contacting the panel can cause distortion and possibly even damage to the exciter or panel while having a minimal contribution to acoustic output. A method is outlined for deriving a digital biquad filter to cancel out the excessive displacement of the magnet based on measurements of the exciter's resonant frequency and Q-factor. Measurements of exciter and panel displacement demonstrate that an applied filter reduces magnet excursion by 20 dB at the resonant frequency.
Engineering Brief 534

Broadcast & Online Delivery
B11 - SOUND KNOWLEDGE: LEARN HOW SMPTE ST 2110 HELPS AUDIO STAND OUT
Friday, October 18, 9:00 am – 10:30 am, Room 1E07

Moderator: **Jeff Cohen**, SMPTE NY Section Chair

Panelists: *Andreas Hildebrand*, ALC NetworX GmbH, Munich, Germany
John Mailhot, Imagine Communications, Bridgewater, NJ, USA

Join SMPTE ST 2110 experts John Mailhot and Andreas Hildebrand as they delve into the latest on the video, audio and metadata over IP standard that is changing the broadcast industry. Discover the why and how behind SMPTE ST 2110 and what it means for audio production. Get up to speed on the latest developments within the standard, ST 2110-30: PCM Digital Audio, which is based on AES-67, and ST 2110-31: AES3 Transparent Transport, which originates from RAVENNA's AM824 format. Take a deep dive into audio specific elements including benefits, commonalities, constraints and differences. Learn how SMPTE ST 2110 provides a wealth of flexibility for handling audio signals and practical methods of utilizing the technology in real live use cases. Understand how this standard compares to other approaches and formats in the field. This 90 minute session will bring you up to speed on SMPTE ST 2110 and what it can do for your next audio deployment.

Co-organized by the Audio Engineering Society and the Society of Motion Picture and Television Engineers

Broadcast & Online Delivery
B12 - SBE CERTIFICATION EXAMS— CANCELLED

Game Audio & XR
GA12 - NOT PLAYING GAMES WITH YOUR BUSINESS
Friday, October 18, 9:00 am – 10:30 am, Room 1E13

Chair: **Alex Wilmer**, Wilmer Sound, San Francisco, CA, USA

Panelists: *Jason Kanter*, Audio Director, Avalanche Studios
Adam Levensen, Sr. Dir. of Business Development & Marketing, Waves Audio
Michael Sinterniklaas, President, NYAV Post

The business of game audio is rapidly changing. This panel will include some of the most qualified professionals in the industry to share their views on the opportunities available to artists, produc-ers, directors, and entrepreneurs within game audio. Whether you're a veteran or just starting out, this panel will cover all angles of the business of game audio to help you achieve your goals.

Immersive & Spatial Audio
IS05 - BUILDING LISTENING TESTS IN VR
Friday, October 18, 9:00 am – 10:00 am, Room 1E08

Presenters: **Gavin Kearney**, University of York, York, UK
Tomasz Rudzki, University of York, York, UK
Benjamin Tsui, University of York, York, UK

In this workshop we will demonstrate how to prepare and conduct various listening tests in VR easily with the tools we created. Our open source tool-box consists of Unity objects, VST plugins, and MATLAB based data analysis app. It provides an end-to-end workflow from creating the test to visualizing the results. Researchers can create their own listening tests which can be run on different VR headsets, e.g., Oculus Rift, HTC Vive. We will dive into some of the actual use-cases to show the practicality and robustness of using audio-visual VR presentation for perceptual tests. We would like to encourage researchers to use the toolbox, express feedback, and contribute to the project development.

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

Product Development
PD11 - PRODUCT MANAGEMENT MODELING
Friday, October 18, 9:00 am – 10:30 am, Room 1E09

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

Product Management has advanced to become arguably the most important part of Product Development. Being able to research, develop, and present the product definition before design begins leads to more successful products that come to market in less time. Modeling the different dimensions of a proposed product brings a much higher level of understanding and confidence in making key decisions.

In this workshop you will learn how to model in detail the following dimensions:

1) Pricing
2) Market Share
3) Financial Pro Formas
4) Forecasting integrating existing, overlapping and complimentary products
5) Product performance and specifications
6) Product costing and analysis
7) Project planning, costing, and resourcing
8) High Level Summary of all underlying models

This workshop is targeted to current and future product managers, engineering managers, development engineers, manufacturing engineers, and supply chain people. All of these roles can participate and add value to the product management function by participate in the modeling process.

Recording & Production
RP12 - RECORDING AND REALIZING IMMERSIVE CLASSICAL MUSIC FOR, AND WITH, DOLBY ATMOS
Friday, October 18, 9:00 am – 10:15 am, Room 1E06

Moderator: **John Loose**, Dolby Laboratories, Inc., San Francisco, CA, USA

Panelists: *David Bowles*, Swineshead Productions LLC, Berkeley, CA, USA
Morten Lindberg, 2L (Lindberg Lyd AS), Oslo, Norway
Jack Vad, San Francisco Symphony, San Francisco, CA, USA

Producing for Classical releases in immersive formats like Dolby Atmos has unique considerations unique to the genre. Dolby's John

Loose will moderate this lively panel of Grammy nominated and winning Classical engineers and producers will discuss translation from microphones to immersive playback environments including binaural Dolby Atmos playback.

Sound Reinforcement
SR05 - AC POWER, GROUNDING, AND SHIELDING SUPER SESSION
Friday, October 18, 9:00 am – 12:00 noon, Room 1E12

Presenters: **Bruce Olson**, Olson Sound Design, LLC, Minneapolis, MN, USA; AFMG Services North America, LLC, Minneapolis, MN, USA
Bill Whitlock, Whitlock Consulting, Ventura, CA, USA

Significant misinformation still persists about what is needed for AC power, and proper grounding, for events — much of it potentially life-threatening advice. Two of the leading authorities on the subject will discuss in detail how to provide AC power properly and safely and without causing noise problems. The session will cover theory and practice of AC power for small to large sound systems, from a couple boxes on sticks up to multiple stages in ballrooms, road houses, and event centers; large scale installed systems, including multiple transformers and company switches, service types, generator sets, 1ph, 3ph, 240/120 208/120.

Student Events & Career Development
SC12 - RESUME REVIEW (FOR STUDENTS, RECENT GRADS, AND YOUNG PROFESSIONALS)
Friday, October 18, 9:00 am – 5:00 pm, SDA Booth

Moderator: **Alex Kosiorek**, Central Sound at Arizona PBS, Phoenix, AZ, USA

Students, recent graduates and young professionals… Often your resume is an employer's first impression of you. Naturally, you want to make a good one. Employer's often use job search websites to search for candidates. Some use automated software to scan your resume and in some cases, your LinkedIn/social media profiles as well. Questions may arise regarding formatting, length, keywords and phrases so it shows up in searches and lands on the desk of the hiring manager. No matter how refined your resume may be, it is always good to have someone else review your materials. Receive a one-on-one 20-25 minute review of your resume from a hiring manager who is in the audio engineering business. Plus, if time allows, your cover letter and online presence will be reviewed as well.

Sign up at the student (SDA) booth immediately upon arrival. For those who would like to have your resume reviewed on Wednesday, October 17th prior to SDA-1, please email the request to: aesresumereview@outlook.com. You may be requested to upload your resume prior to your appointment for review. Uploaded resumes will only be seen by the moderator and will be deleted at the conclusion of the 147th Pro Audio Convention.

This review will take place during the duration of the convention by appointment only.

Technical Committee Meeting
AUDIO FOR TELECOMMUNICATIONS
Friday, October 18, 9:00 am – 10:00 am, Room 1C02

AES Standards Meetings
SC-02-12-L TASK GROUP ON OPEN CONTROL ARCHITECTURE
Friday, October 18, 9:00 am – 10:30 am, Room 1C03

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.

Historical Event
H06 - AFRICAN AMERICANS IN AUDIO (AN AES SPECIAL EVENT)
Friday, October 18, 9:30 am – 11:00 am, Room 1E15+16

Moderator: **Leslie Gaston-Bird**, Mix Messiah Productions, Brighton, UK; Institute of Contemporary Music Performance, London, UK

Panelists: *Prince Charles Alexander*, Berklee College of Music, Boston, MA, USA
Abhita Austin, Audio Engineer-Producer and Founder of The Creator's Suite
James Henry, recording engineer/producer and audio educator
Ebonie Smith, Atlantic Records/Hamilton Cast Album
Paul "Willie Green" Womack, Willie Green Music, Brooklyn, NY, USA
Bobby Wright, Hampton University

African Americans have contributed to the popular music recording industry in a number of ways, and although their achievements are visible, their representation at technical conferences and on the exhibit floor is less so. Join our panel of renowned engineers, performers, and educators for a discussion on how African Americans have been blazing trails behind the scenes in the recording industry and how we can best engage and welcome them to access the social and scholarly networks that have benefited us. Topics include a technological pedagogy for hip-hop education and dispelling the stereotype that African-American engineers are only able to work in "Black Music" genres.

Chaired by AES Governor-at-Large Leslie Gaston-Bird, the first African American to sit on the AES Board of Governors, panelists include: James Henry, three time Grammy nominated recording engineer/producer and audio educator; Paul Willie Green Womack, Producer/Engineer and Chair of the Hip-Hop and R&B track of the AES Convention Committee; Prince Charles Alexander, Grammy winning Music Producer/Engineer/Recording Artist and Professor of Music Production and Engineering for Berklee College of Music and Berklee Online; Bobby Wright, Assistant Professor (music, audio engineering) at Hampton University; Abhita Austin, Audio Engineer-Producer and Founder of The Creator's Suite; and Ebonie Smith, Grammy winning engineer, producer, and singer / songwriter.

Student Event & Career Development
SC13 - RECENT GRADUATE PANEL
Friday, October 18, 9:30 am – 11:00 am, Room 1E21

Moderator: **Justin Chervony**, University of Miami, Frost School of Music, Miami, FL, USA

Panelists: *TBA*

Panel for students to explore work, job hunting, and creative experiences of recent graduates in the industry. Students will get to hear about experiences that directly impact them from folks who are closer in age and more closely have dealt with the ever-changing and variable opportunities in the world of audio. Panelists will be of diverse backgrounds and differing career paths, all prior and/or current AES members, and some will have been competition finalists. This will hopefully inspire students about the prospect of going out after school and creating opportunities and a unique career for themselves.

Immersive & Spatial Audio
IS14 - MICK SAWAGUCHI IMMERSIVE PREVIEW
Friday, October 18, 10:00 am – 11:00 am, Room 1E17

Presenter: **Mick Sawaguchi,** Mick Sound Lab, Tokyo, Japan

Mick Sawaguchi recently made unique 7.1.4 music recordings in Finland, based on his vast immersive experience and new microphone techniques. This session is an apetiser for the full suite, to be presented at InterBEE in November.

Technical Committee Meeting
HEARING AND HEARING LOSS PREVENTION
Friday, October 18, 10:00 am – 11:00 am, Room 1C02

AES Mix with the Masters Workshops
MM17 - RAFA SARDINIA
Friday, October 18, 10:00 am – 11:00 am
Mix with the Masters Workshop Stage

Archiving & Restoration
AR02 - CACHE RULES EVERYTHING AROUND ME: ARCHIVING AND PRESERVING HIP-HOP IN A DIGITAL AGE
Friday, October 18, 10:15 am – 11:15 am, Room 1E11

Chair: **Sommer McCoy**, The Mixtape Museum, New York, NY, USA

Panelists: *Rocky Bucano*, Universal Hip-Hop Museum
Manny Faces, The Center for Hip-Hop Advocacy
Syreeta Gates
DJ Dirty Harry
DJ Rich Nice

The early recordings of hip-hop's beginnings are in danger of deterioration. The genesis of hip hop began roughly 46 years ago and was largely captured on the celebrated medium of the time, the compact cassette. Cassettes captured hip-hop's early sounds at live performances and park jams. Decades later, many of these one-of-a-kind recordings live in unstable environments exposed to elements that can erase their existence.

In an effort to rescue, preserve, and restore these original recordings, initiatives like The Mixtape Museum are organizing for solutions. In addition, we are witnessing an unprecedented effort to archive hip-hop in museums, cultural heritage institutions, and libraries, and an expanded presence in the burgeoning field of hip-hop scholarship.

This panel examines:

• The cultural, artistic, and historical significance of hip-hop's early cassette-based recordings (mixtapes, bootleg recordings, "demo" versions, etc.) and the importance of preserving them.

• The technical and engineering challenges unique to hip-hop. Where and how do current technologies play a role?

• Who can implement the technologies needed to preserve, archive, and distribute hip-hop? How can DJs, artists, collectors, archivists, librarians, producers, engineers, and technologists collaborate on these efforts

• The current unprecedented effort to archive hip-hop in private collections, museums, cultural heritage institutions, and libraries. What are the pros and cons of academic institutions vs. community-driving preservation?

Recording & Production
RP13 - PLATINUM MASTERING: MASTERING IMMERSIVE AUDIO
Friday, October 18, 10:15 am – 11:45 am, Room 1E08

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

Panelists: *Stefan Bock*, msm-studios GmbH, Munich, Germany
Gavin Lurssen, Lurssen Mastering, Los Angeles, CA, USA
Andres A. Mayo, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina
Darcy Proper, Darcy Proper Mastering at Valhalla Studios, Auburn, NY, USA
Mark Wilder

The Long Running Platinum Mastering Panel will focus on Mastering in an Immersive environment. Mastering experts from around the world discuss the challenges and expectations of mastering for release and distribution. The focus will be on the many aspects of the emerging popularity of immersive music and the challenges mastering engineers face with delivery, formats, work flow and many other details related to providing the best combination of the Art and Technical aspects for the artist and the consumer. There are many different formats, with different requirements, vying for prominence and mastering engineers need to be prepared for each.

Acoustics & Psychoacoustics
AP05 - HOW TO ORGANIZE UNBIASED PA/SR SHOOT-OUTS
Friday, October 18, 10:30 am – 11:30 am, Room 1E013

Presenters: **Eddy Bøgh Brixen**, EBB-consult, Smørum, Denmark;
DPA Microphones, Allerød, Denmark
Gil Eva Craig, Western Audio Engineering Wellington
Malle Kaas, Women in Live Music
Bob McCarthy, Meyer Sound, New York, NY, USA
Scott Sugden, L-Acoustics, Oxnard, CA, USA

It is very common to arrange for comparisons of PA systems. However, often these comparisons are organized in a way leaving procedures less transparent and results rather unclear. Standards for the assessment of loudspeakers do exist. The assessors basically must be trained for the purpose, and the set-up should support double-blind testing. However, in the test of big systems, the listening panel is not necessarily trained and the practical problems of rigging huge arrays to some degree may weaken the procedures and the results. This workshop seeks to provide good advice from experts in the field. Further, it reports the experience from double blind testing that solved most of the problems in connection with a session testing five line arrays.

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

Broadcast & Online Delivery
B13 - METADATA— WHAT WORKS, WHAT DOES NOT, AND WHY?
Friday, October 18, 10:30 am – 12:00 noon, Room 1E07

Moderator: **David Bialik**

Panelists: *Kirk Harnack*, Telos Alliance
John Kean, Kean Consultants LLC, Washington DC, USA; Cavell Mertz & Associates, Manassas VA, USA
Robert Orban, Orban Labs Inc., Pennsauken, NJ, USA
Samuel Sousa, Triton Digital, Montreal, QC, Canada
Kent Terry, Dolby Laboratories Inc., San Francisco, CA, USA

Metadata seems as simple as describing content, but in reality you have to understand what the dif-ferent distribution systems support, then you have to consider that you might be doing On Air, Online Live, Podcasts, and add some revenue options such as Server-Side Ad Insertion.

Metadata and other trigger information will be crucial and understanding the differences between what is stored in files, how these files are played out, and how data transported through the differ-ent systems. Then you also have to think about the experience that created for the end-user.

Let's try exploring what is possible vs what is effective. To be truly universal is actually a chal-lenge. Let's talk about this.

Game Audio & XR
GA13 - BORDERLANDS 3—THE WILD WEST OF ATMOS
Friday, October 18, 10:30 am – 12:00 noon, Room 1E06

Presenters: **Brian Fieser**, Gearbox Software
Julian Kwasneski, Bay Area Sound
Mark Petty, Gearbox Software
William Storkson, Bay Area Sound

This event will cover: Emitter / object based in game design vs rendered 7.1.4 assets; How do these two approaches differ when it comes to spatialization from the player perspective; Understanding the end user environment and mixing for Atmos virtualization; Atmos for headphones; Linear post / cinematic design for Atmos; Mix perspectives / how aggressive should we be with height in-for-mation— Atmos is the wild west; Using 7.1.4 discreet as a tool to better understand inconsisten-cies in game / run time spatial information.

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

AES Standards Meetings
SC-07-01 WORKING GROUP ON AUDIO METADATA
Friday, October 18, 10:30 am – 11:30 am, Room 1C03

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.

Product Development
PD12 - ADVANCED MATERIALS FOR AUDIO PRODUCTS
Friday, October 18, 10:45 am – 12:15 pm, Room 1E09

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 plastics that are stiffer and better damped than commodity plastics result in less enclosure buzz and smoother response. Active noise canceling and acoustic echo cancelers especially benefit from these materials. Nano materials enhance water repellent surface treatments to CVD coatings for enhanced structural integrity. Thermoplastic carbon fiber composites deliver high strength for speaker diaphragms to horns and enclosures with fast forming speed compared to thermosets. Ongoing development of graphene for microphones, headphones and micro-speakers promises significant performance advances. Elastic (helical conductor) audio cables provide elongation to 40% while delivering non-tangling and non-microphonic (no body noise).

You will see examples of the materials available so you as a product developer can start seeing how your products ride the next wave of ultra-materials.

At the talk tickets to special demos at booth #457 Friday from 12:30 to 1:30 pm for hands-on (and ears-on) for the technologies discussed during the product track talk.

Engineering Briefs Session EB03 **Friday, Oct. 18**
11:00 am – 12:30 pm **South Concourse A**

POSTERS—SPATIAL AUDIO

11:00 am

EB03-1 Comparing Externalization Between the Neumann KU100 Versus Low Cost DIY Binaural Dummy Head—*Kelley DiPasquale*, SUNY Potsdam, Potsdam, NY, USA

Music is usually recorded using traditional microphone techniques. With technology continually advancing, binaural recording has become more popular, that is, a recording where two microphones are used to create a three-dimensional stereo image. Commercially available binaural heads are prohibitively expensive and not practical for use in typical educational environments or for casual use in a home studio. This experiment consisted of gathering recorded stimuli with a homemade binaural head and the Neumann KU 100. The recordings were played back for 34 subjects instructed to rate the level of externalization for each example. The study investigates whether a homemade binaural head made for under $500 can externalize sound as well as a commercially available binaural head the Neumann KU 100.
Engineering Brief 535

11:00 am

EB03-2 SALTE Pt. 1: A Virtual Reality Tool for Streamlined and Standardized Spatial Audio Listening Tests—*Daniel Johnston, Benjamin Tsui, Gavin Kearney*, University of York, York, UK

This paper presents SALTE (Spatial Audio Listening Test Environment), an open-source framework for creating spatial audio perceptual testing within virtual reality (VR). The framework incorporates standard test paradigms such as MUSHRA, 3GPP TS 26.259 and audio localization. The simplified drag-and-drop user interface facilitates rapid and robust construction of customized VR experimental environments within Unity3D without any prior knowledge of the game engine or the C# coding language. All audio is rendered by the dedicated SALTE audio renderer which is controlled by dynamic participant data sent via Open Sound Control (OSC). Finally, the software is capable of exporting all experimental conditions such as visuals, participant interaction mechanisms, and test parameters allowing for streamlined and standardized comparable data within and in-between organizations.
Engineering Brief 536

11:00 am

EB03-3 SALTE Pt. 2: On the Design of the SALTE Audio Rendering Engine for Spatial Audio Listening Tests in VR—*Tomasz Rudzki, Chris Earnshaw, Damian Murphy, Gavin Kearney*, University of York, York, UK

The dedicated audio rendering engine for conducting listening experiments using the SALTE (Spatial Audio Listening Test Environment) open-source virtual reality framework is presented. The renderer can be used for controlled playback of Ambisonic scenes (up to 7th order) over headphones and loudspeakers. Binaural-based Ambisonic rendering facilitates the use of custom HRIRs contained within separate WAV ?les or SOFA ?les as well as head tracking. All parameters of the audio rendering software can be controlled in real-time by the SALTE graphical user interface. This allows for perceptual evaluation of Ambisonic scenes and different decoding schemes using custom HRTFs.
Engineering Brief 537

11:00 am

EB03-4 Mixed Reality Collaborative Music—*Andrea Genovese,* *Marta Gospodarek, Agnieszka Roginska*, New York University, New York, NY, USA

This work illustrates a virtual collaborative experience between a real-time musician and virtual game characters based on pre-recorded performers. A dancer and percussionists have been recorded with microphones and a motion capture system so that their data can be converted into game avatars able to be reproduced within VR/AR scenes. The live musician was also converted into a virtual character and rendered in VR, and the whole scene was observable by an audience wearing HMDs. The acoustic character between the live and pre-recorded audio was matched in order to blend the music into a cohesive mixed reality scene and address the viewer's expectations set by the real-world elements.
Presentation only; not in E-Library

11:00 am

EB03-5 Withdrawn

11:00 am

EB03-6 Field Report: Immersive Recording of a Wind Ensemble Using Height Channels and Delay Compensation for a Realistic Playback Experience—
Hyunjoung Yang, Alexander Dobson, Richard King, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and T echnology, Montreal, Quebec, Canada

Practical examples for orchestra recording in stereo or surround were relatively easy to obtain. Whereas, it was found that recording practice in immersive audio is relatively limited. This paper is intended to share the experience of the immersive recording process for a wind orchestra recording at McGill University. There were concerns that needed to be considered before planning the concert recording, problems encountered during the planning, and lastly the solutions to these issues. In conclusion, the discussions about the final result and the approach will be described.
Engineering Brief 538

Immersive & Spatial Audio
IS15 - HYUNKOOK LEE TALK&PLAY
Friday, October 18, 11:00 am – 12:00 noon, Room 1E17

This session will demonstrate various 7.1.4 and 4.0.4 immersive 3D recordings made using PCMA-3D and ESMA-3D microphone techniques. The demos will include an orchestral concert recorded at Victoria Hall, Geneva, choral performances recorded at Merton College Chapel, Oxford and York Minster, an organ performance at Huddersfield Town Hall and some soundscapes of New York City. The psychoacoustic principles of the microphone arrays used will also be explained.

Recording & Production
RP14 - FOR THE RECORD: ENGINEERING PRINCE
Friday, October 18, 11:00 am – 12:00 noon, Room 1E21

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

Panelists: *Lisa Chamblee*
Sylvia Massy
Peggy McCreary
Susan Rogers, Berklee College of Music, Boston, MA, USA

Women's Audio Mission (WAM) is proud to present a special panel discussion with a group of top engineers who worked closely with the legendary artist Prince. Come hear Sylvia Massy, Susan Rogers, Peggy McCreary, and Lisa Chamblee recount their experiences of working with the prolific star on albums like Purple Rain, Diamonds and Pearls, Fury, 3121, and D.M.S.R., as well as other tales of Paisley Park. Panel will be moderated by the Grammy-winning engineer, Leslie Ann Jones.

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

Technical Committee Meeting
FIBER OPTICS FOR AUDIO
Friday, October 18, 11:00 am – 12:00 noon, Room 1C02

AES Mix with the Masters Workshops
MM18 - JIMMY DOUGLASS
Friday, October 18, 11:00 am – 12:00 noon
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP24 - REINVENTING INTERCOM WITH SMPTE ST 2110-30
Friday, October 18, 11:00 am – 11:30 am
AoIP Pavilion Theater

Presenter: **Martin Dyster**, The Telos Alliance, Cleveland, OH USA

This presentation looks at the parallels between the emergence of audio over IP standards and the development of a product in the Intercom market sector that has taken full advantage of IP technology.

Live Production Stage
LS13 - RF SPECTRUM UPDATE
Friday, October 18, 11:00 am – 11:45 am
Live Production Stage

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

Panelists: *Mark Albergo*
Ben Escobedo, Shure Market Development
Michael Mason, President of CP Communications
David Missall

We're in it now! The results of the 600 MHz spectrum auction are upon us, as the new owners of the 616-698 MHz range are turning on their services. 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 Recording Stage
RS13 - STUDIO DMI PRESENTS LUCA PRETOLESI SUPER SESSION
Friday, October 18, 11:00 am – 11:45 am
Recording Stage

Presenter: **Luca Pretolesi**, Studio DMI, Las Vegas, NV, USA

Immerse yourself in this interactive presentation designed to give attendees a beyond-the-box look at some of Pretolesi's most impactful mixing and mastering tips. From session recalls, live Q&A, and real-time track reviews Pretolesi will present the same cutting-edge production techniques and hit-making insights that have helped to amplify the biggest artists in the world including; David Guetta, Jason Derulo, J Balvin, Diplo, Lil Jon, DJ Snake, Major Lazer, Steve Aoki, Above & Beyond, Nicki Minaj, Daddy Yankee, and Prince Royce.

Archiving & Restoration
AR03 - METADATA FROM CREATION TO CONSUMPTION TO PRESERVATION
Friday, October 18, 11:15 am – 12:15 pm, Room 1E11

Chair: **Brad McCoy**, Library of Congress, Culpeper, VA, USA

Panelists: *Tony Brooke*
Maureen Droney, The Recording Academy, Los Angeles, CA, USA
Paul Jessop

This panel approaches metadata from the from the varied perspectives of audio archivists, music creators, labels, distributors, music services, and standards bodies in order to foster a collaborative conversation around how to capture, store, and distribute information about recorded sound. Panelists will compare best practices in the library world to how labels handle the same information and discuss how metadata travels through the pipeline (or doesn't) to streaming services and end users.

Why is collecting and delivering rich metadata critical for your projects and career? Watch this short video: https://youtu.be/h5TDZDtNqhM

Historical Event
H07 - COMPRESSION DRIVER DNA: THE ORIGIN AND SEEDS OF PROGRESS
Friday, October 18, 11:15 am – 12:15 pm, Room 1E10

Presenters: **Thomas Dunker**
Bjørn Kolbrek, Celestion, Ipswich, UK

The successful commercialization of sound films in 1926 by Western Electric and Vitaphone marked the start of large scale high quality sound reproduction. Central in the success of the Vitaphone was the Western Electric 555 compression driver, the grandmother of modern compression drivers, developed by the Bell Labs. We will take a closer look at the technology inside this remarkable driver, and through never-before-published reports, including unimplemented proposed improvements.

Further Bell Labs development into high-power wide-range drivers resulted in the second generation of compression drivers, in the form of the Western Electric 594A. This driver marks the start of modern compression driver design and is worthy of some detailed discussion.

Special Event
SE10 - HOW WE MAKE MUSIC—CROSSING THE DECADES FROM ANALOG TO DIGITAL
Friday, October 18, 11:15 am – 12:15 pm, Room 1E15+16

Moderator: **Chris Lord-Alge**, Mix LA, Los Angeles, CA, USA

Panelists: *Niko Bolas*, The Surf Shack Studio, Ventura, CA USA; Germano Studio, NYC, NY USA
Danny Kortchmar, Legendary GRAMMY nominated guitarist, songwriter and producer (Jackson Browne, Don Henley, James Taylor)
Tom Lord-Alge, SPANK Studios, South Beach, FL, USA
Dave Way

Hear from top recording engineers and producers techniques they use from analog to digital and how they connect the dots of the past, present, and future of creative technology—how we transaction into Pro Tools. This panel will feature Chris Lord-Alge, Tom Lord-Alge, and others to be announced.

Student Events & Career Development
SC14 - THE ART OF LISTENING
Friday, October 18, 11:30 am – 12:30 pm, Room 1E13

Presenter: **Rick Snoman**, Dance Music Production, Manchester, UK

The real secret behind any great engineer or producer is their ability to hear the finer details of the music. For this session Rick will introduce you to the basic principles you need to learn in order to develop your listening skills at a faster rate. Throughout the 45 minute talk, you'll not only learn what to listen for but also how to listen and identify the details of your music so you can progress at making music faster, more professional and with more confidence.

This is a High School event limited for HS students.

Student Events & Career Development
SC15 - EDUCATION AND CAREER FAIR
Friday, October 18, 11:30 am – 1:30 pm, South Concourse B

The combined AES 147th 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 147th 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.

AES Standards Meetings
SC-02-02 WORKING GROUP ON DIGITAL INPUT/OUTPUT INTERFACING
Friday, October 18, 11:30 am – 12:30 pm, Room 1C03

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.

AoIP Pavilion
AIP26 - JT-NM TESTED PROGRAM—TEST PLANS AND RESULTS
Friday, October 18, 11:30 am – 12:00 noon
AoIP Pavilion Theater

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

The JT-NM Tested Program was repeated in August 2019 with the addition of AMWA NMOS/JT-NM TR-1001-1 testing. New revisions of the test plans were produced. What does all this mean to the end customers? The editor and coordinator of the program will explain the reasoning behind, the technical details, what was changed in the new revisions, how it all was executed and everything else you wanted to know about the JT-NM Tested Program, but were too afraid to ask!

Software@AES
S21 - BEST SERVICE
Friday, October 18, 11:30 am – 12:00 noon
Software@AES Pavilion

Student Events & Career Development
SC16 - STUDENT RECORDING CRITIQUES
Friday, October 18, 12:00 noon – 1:00 pm, Room 1E06

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

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
HIGH RESOLUTION AUDIO
Friday, October 18, 12:00 noon – 1:00 pm, Room 1C02

Technical Committee Meeting
BROADCAST AND ONLINE DELIVERY
Friday, October 18, 12:00 noon – 1:00 pm, Room 1E07

AES Mix with the Masters Workshops
MM19 - YOUNG GURU
Friday, October 18, 12:00 noon – 1:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP26 - JT-NM TESTED PROGRAM—TEST PLANS AND RESULTS
Friday, October 18, 12:00 noon – 12:30 pm
AoIP Pavilion Theater

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

When RAVENNA was introduced to the industry back in 2010, several – mostly proprietary – AoIP solutions were existing, but no standard was yet visible. Although RAVENNA is an open technology approach, with its modular architecture fully based on existing and well-accepted standards, people seemed to be confused by the wide choice of competing and non-interoperable solutions. The arrival of AES67 in 2013 made people cheer and pay homage to an AoIP standard they believed would finally render any other solutions obsolete… 4 years later, SMPTE published its suite of ST 2110 documents, with ST 2110-30 (linear PCM audio transport) fully referencing AES67 as its basis. A closer look at these standards reveals that RAVENNA – published some 7 years before the arrival of ST 2110 – was build on exactly the same protocols and functions as AES67 and ST 2110. So, does this mean, RAVENNA is now obsolete, being forced to coma by the AES67/ST 2110 nerve pinch? Andreas Hildebrand, RAVENNA Evangelist at ALC NetworX, Germany, explains why this Vulcan nerve pinch does not have any effect for RAVENNA at all…!

Audio Builders Workshop Booth Talks
ABT12 - GET TO KNOW YOUR GEAR WITH FREE ANALYSIS SOFTWARE
Friday, October 18, 12:00 noon – 12:30 pm, Booth 266

Presenter: **Peterson Goodwyn**, DIY Recording Equipment, Philadephia, PA, USA

Live Production Stage
LS14 - THE 7 MOST COMMON WIRELESS MIC MISTAKES (AND WHAT YOU CAN DO ABOUT THEM)
Friday, October 18, 12:00 noon – 12:45 pm
Live Production Stage

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

Panelists: *Christopher Evans*, The Benedum Center, Pittsburgh, PA, USA
Jason Glass, Clean Wireless

Anyone who has set up or used a wireless mic system, large or small, has faced many of the same problems. This panel of industry experts will explore the most common issues 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 Recording Stage
RS14 - FOCUSRITE PRESENTS: PODCASTING: THE FASTEST GROWING MEDIUM IN AUDIO
Friday, October 18, 12:00 noon – 12:45 pm
Recording Stage

Presenter: **Dan Hughley**, Focusrite, Los Angeles, CA, USA

Join Dan Hughley of Focusrite for this brand-neutral presentation aimed at answering questions that are frequently asked by new podcasters. After learning what a podcast is and the history of the medium, you will learn what you need to get started, the importance of high quality audio to your audience, how to keep your show going when fatigue sets in, and finish up with some quick tips on how to cost effectively promote your show to gain listeners and subscribers. This promises to be a fun and educational session that you surely should not miss.

Software@AES
S22 - FL STUDIO
Friday, October 18, 12:00 noon – 12:30 pm
Software@AES Pavilion

Product Development
PD13 - VENDOR EVENT 3: MENLO SCIENTIFIC
Friday, October 18, 12:15 pm – 1:30 pm, Demo Room 457

AoIP Pavilion
AIP28 - JT-NM TESTED PROGRAM—TEST PLANS AND RESULTS
Friday, October 18, 12:30 pm – 1:00 pm
AoIP Pavilion Theater

Presenter: **Ethan Wetzell**, OCA Alliance, New York, NY, USA

AES70, also known as OCA, is an architecture for system control and connection management of media networks and devices. AES70 is capable of working effectively with all kinds of devices from multiple manufacturers to provide fully interoperable multivendor networks. This session will provide an overview of the AES70 standard, including the current and future objectives of the standard.

Audio Builders Workshop Booth Talks
ABT13 - EURORACK AND PLUGIN DESIGN WITH DELTA SOUND LABS
Friday, October 18, 12:30 pm – 1:00 pm, Booth 266

Presenter: **Richard Graham**, Delta Sound Labs, Cleveland, OH, USA

Software@AES
S23 - ACCUSONUS
Friday, October 18, 12:30 pm – 1:00 pm
Software@AES Pavilion

Acoustics & Psychoacoustics
AP06 - CIRCLES OF EXCELLENCE
Friday, October 18, 1:15 pm – 2:15 pm, Room 1E012

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

Panelists: *Florian Camerer*, ORF, Austrian TV - Vienna, Austria; EBU, European Broadcasting Union
Bob Katz, Digital Domain Mastering, Orlando, FL, 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

30.000 year old cave paintings are among human beings' most impressive cultural heritage, while we are unable to experience how music by great composers sounded just 300 years ago.

The experts on the panel will try to separate the facts from the myths surrounding recording, monitoring and reproduction, and explain which of the elements they have found most essential in their work to make it stand the test of time. The panel is a continuation of the more theoretical AP04 discussion on Thursday.

Recording & Production
RP15 - STUDIO DESIGN FOR 2070 —50 YEARS FROM NOW—DARE WE LOOK?
Friday, October 18, 1:00 pm – 2:00 pm, Room 1E08

Presenters: **Steve Berkowitz**
Jimmy Douglass
Ann Mincieli, Jungle City Studios
David Rosenthal
John Storyk, Walters-Storyk Design Group, Highland, NY, USA

The future is predictably "unpredictable," but this panel of studio designers and users will give it their best shot. What kinds of facilities (and tools) will musicians, engineers, and producers be working with 50 years hence? Will recording artists always need studios? Are CD's & Vinyl totally DOA? Or, will a totally new Gold Standard format emerge? We may not be around to check on our accuracy in 2070, but these production/creative seers including award-winning engineer/producer Eddie Kramer will opine on formats instruments and technological advances they'd like to have. Will recording dates be booked in weightless, anechoic studios circling the globe in outer space stations? Will we finally hear someone wailing away on a Stratocaster from some other galaxy? Tune in tomorrow.

Technical Committee Meeting
CODING OF AUDIO SIGNALS
Friday, October 18, 1:00 pm – 2:00 pm, Room 1C02

AES Mix with the Masters Workshops
MM20 - TCHAD BLAKE
Friday, October 18, 1:00 pm – 2:00 pm
Mix with the Masters Workshop Stage

Audio Builders Workshop Booth Talks
ABT14 - REDUCE, REUSE, RECYCLE: AN APPROACH TO REPURPOSING BROKEN GEAR IN DIY BUILDS
Friday, October 18, 1:00 pm – 1:30 pm, Booth 266

Presenter: **Jason Bitner**, Traffic Entertainment Group, Somerville, MA, USA

Electronic Dance Music Stage
EDJ06 - WAVES PRODUCT WORKSHOP
Friday, October 18, 1:00 pm – 1:45 pm
Electronic Dance Music & DJ Stage

Presenter: **Michael Pearson-Adams**, Waves, Knoxville, TN, USA

Live Production Stage
LS15 - MEYER SOUND PRESENTS: LIVE TOURING SYSTEM
Friday, October 18, 1:00 pm – 1:45 pm
Live Production Stage

Project Studio Expo Recording Stage
RS15 - SYLVIA MASSY—THE SECRET INGREDIENTS
Friday, October 18, 1:00 pm – 1:45 pm
Recording Stage

Presenter: **Dan Hughley**, Focusrite, Los Angeles, CA, USA

But what are the secret ingredients for a good session? Is it a certain piece of gear, a certain song, a certain location? Maybe it's all those things, and so much more! In this lecture, Sylvia Massy shares stories about some of her favorite recording adventures, She also reveals her secrets for making each session memorable.

Sponsored by iZotope

Software@AES
S24 - ACON DIGITAL
Friday, October 18, 1:00 pm – 1:30 pm
Software@AES Pavilion

Broadast & Online Delivery
B14 - TELLING STORIES WITH SOUND: WHAT CAN AUDIO STORYTELLING LEARN FROM SOUND FOR PICTURE?
Friday, October 18, 1:15 pm – 2:30 pm, Room 1E07

Moderator: **Rob Byers**, Minnesota Public Radio | American Public Media, Minneapolis, MN, USA

Panelists: *Lon Bender*, The Formosa Group, Los Angeles, CA, USA
Jocelyn Gonzales
Alexa Zimmerman

Sound: there's no better way to tell stories.

Rob Byers (American Public Media and Criminal) leads a conversation about what the craft of audio-only storytelling can learn from sound for picture. We'll explore what it takes to tell truly engaging stories in sound and apply those concepts to podcasts and radio.

Panelists include Lon Bender (supervising sound editor, Formosa Group), Alexa Zimmerman (dialog editor, *Mary Poppins Returns, Roma*), and Jocelyn Gonzales (executive producer of Studio 360 and a sound for picture educator).

Sound Reinforcement
SR06 - PANEL DISCUSSION ON SOUND SYSTEM OPTIMIZATION
Friday, October 18, 1:15 pm – 3:15 pm, Room 1E21

Presenters: **Jamie Anderson**, Rational Acoustics
Bob McCarthy, Meyer Sound, New York, NY, USA
Merlijn Van Veen, Meyer Sound Labs

An open discussion about sound system optimization with a panel of veterans in this field. Topics will include the latest techniques and improvements in analysis technology. Panelists will discuss equalization, phase alignment, time alignment, speaker aiming, beam steering, subwoofer array control and field examples.

Game Audio & XR
GA14 - JUST CAUSE 4 OST: CREATIVE COLLABORATION
Friday, October 18, 1:30 pm – 2:30 pm, Room 1E17

Presenter: **Zach Abramson**, YouTooCanWoo

Just Cause 4's massive soundtrack incorporates many different styles, sounds, and strategies craft-ed by composer Zach Abramson and a team of friends at their small studio in Brooklyn, NY. A collaborative approach was developed between Abramson, his team in Brooklyn and the audio de-partment at Avalanche Studios across the East River in Manhattan, which led to stronger ideas and results that would have been difficult to achieve if working alone.

This presentation will provide an in-depth analysis of ways composers and audio teams can work together to better interpret creative briefs, design pillars, and game mechanics informing a sound-track's aesthetic as well as its technical design. From there, the discussion will continue into the various approaches used to achieve these results, ranging from broad topics like creative decision-making down to specific production techniques. These approaches directly relate to how compos-ers can rely on a team to help mitigate the struggles of working in difficult environments like New York City, where workspaces and time often come at a premium.

Attendees will learn different composition techniques and how these various approaches relate to big concepts in video game scores, as well as real-world tips for how to collaborate effectively in a fast-paced creative environment. This talk is intended for composers, game audio professionals, music supervisors and anyone who is interested in learning more about video game music creation.

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

Product Development
PD14 - DEEP LEARNING AND AI FOR AUDIO APPLICATIONS—ENGINEERING BEST PRACTICES FOR DATA
Friday, October 18, 1:30 pm – 3:00 pm, Room 1E09

Presenter: **Gabriele Bunkheila**, MathWorks, Madrid, Spain

Audio, speech, and acoustics are increasingly recognized as the second largest application area for deep learning after computer vision. Deep learning and AI are defining a new era in product development as they need vast amounts of task-specific labeled data to be successfully optimized for real-world applications. As deep learning is increasingly used alongside more traditional signal processing methods, the focus of audio DSP engineering is gradually expanding from algorithms to data.

In this session, we discuss the importance of signal processing and audio data engineering for the development of deep learning systems. Using practical examples based on MATLAB, we review best practices for audio data workflows in AI applications, including for signal labeling, data ingestion, data augmentation, feature extraction, and signal transformation.

Recording & Production
RP16 - HYBRID AUDIO: MIXING VIRTUAL AND ACOUSTIC INSTRUMENTS FOR MEDIA PRESENTATIONS
Friday, October 18, 1:30 pm – 2:30 pm, Room 1E11

Presenters: **Brandie Lane**, West Point Band, West Point, NY, USA; New York University, New York, NY, USA
Noah Taylor, Staff Sergeant, United States Military Academy Band, West Point, NY, USA

An increasing amount of recordings are starting to use a mixture of virtual (MIDI) and acoustic instruments. This is especially prevalent in media and film scores. Over the past 4 years, The West Point Band has implemented recording and mixing techniques that marry virtual instruments with live acoustic elements to create soundtracks for media presentations seen on television and web platforms. Clients have included NBC, CBS, Army West Point Sports, The United States Military Academy Public Affairs Office, and the Department of Defense. This workshop will break down the recording and mixing techniques used to create these soundtracks.

Special Event
SE11 - LUNCHTIME KEYNOTE: 1500 OR NOTHIN'
Friday, October 18, 1:30 pm – 2:30 pm, Room 1E15+16

Presenters: **Larrance Dopson**
IZ Avila, Avila Brothers

Inspiring and Educating the Next Generation of Producers, Engineers, Creators

The world of audio education is changing. For the modern audio student, the "traditional" curriculum is not enough to compete in today's music industry. Larrance Dopson (GRAMMY Award-winning producer/instrumentalist and CEO of the 1500 or Nothin' production/songwriting collective), along with GRAMMY Award-winning producer and songwriter IZ Avila (known collectively as The Avila Brothers), discuss what it takes to make it in today's fast-paced music production world, and how these new needs have led some music educators and mentors to evolve their approaches to prepare their students for what they'll encounter out in the real world, while inspiring and motivating these students to create their own opportunities.

AES Standards Meetings
SC-05-05 WORKING GROUP ON GROUNDING AND EMC PRACTICES
Friday, October 18, 1:30 pm – 2:30 pm, Room 1C03

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.

PMC Masters of Audio
PM03 - PMC MASTERS OF AUDIO: PMC, CAPITOL STUDIOS PRESENT : "MUSIC IN DOLBY ATMOS
Friday, October 18, 1:30 pm – 2:30 pm, Room 1E06

Presenters: **Maurice Patist**, President of PMC USA
Nick Rives, Capitol Studios

Two years ago Capitol Studios partnered with PMC and Dolby to build the first "Dolby Atmos Music" studio in the world famous Capitol Studios Tower. Now hundreds of mixed tracks later, Amazon Music just announced the Atmos Streaming service allowing people to finally hear the results of this project. Nick Rives, engineer for Capitol Studios who mixed a multitude of these tracks, and Maurice Patist, President of PMC USA, will take you on a journey through Dolby Atmos Music, into what they believe is a new chapter in music history.

Software@AES
S25 - MAGIX
Friday, October 18, 1:30 pm – 2:00 pm
Software@AES Pavilion

Paper Session P14 **Friday, Oct. 18**
1:45 pm – 4:15 pm **Room 1E10**

SPATIAL AUDIO, PART 3

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

1:45 pm

P14-1 Measurement of Oral-Binaural Room Impulse Response by Singing Scales—*Munhum Park*, King Mongkut's Institute of Technology Ladkrabang, Bangkok, Thailand

Oral-binaural room impulse responses (OBRIRs) are the transfer functions from mouth to ears measured in a room. Modulated by many factors, OBRIRs contain information for the study of stage acoustics from the performer's perspective and can be used for auralization. Measuring OBRIRs on a human is, however, a cumbersome and time-consuming process. In the current study some issues of the OBRIR measurement on humans were addressed in a series of measurements. With in-ear and mouth microphones volunteers sang scales, and a simple post-processing scheme was used to re?ne the transfer functions. The results suggest that OBRIRs may be measured consistently by using the proposed protocol, where only 4~8 diatonic scales need to be sung depending on the target signal-to-noise ratio.
Convention Paper 10291

2:15 pm

P14-2 Effects of Capsule Coincidence in FOA Using MEMS: Objective Experiment—*Gabriel Zalles*, University of California, San Diego, La Jolla, CA, USA

This paper describes an experiment attempting to determine the effects of capsule coincidence in First Order Ambisonic (FOA) capture. While the spatial audio technique of ambisonics has been widely researched, it continues to grow in interest with the proliferation of AR and VR devices and services. Specifically, this paper attempts to determine whether the increased capsule coincidence afforded by Micro-Electronic Mechanical Systems (MEMS) capsules can help increase the impression of realism in spatial audio recordings via objective and subjective analysis. This is the first of a two-part paper.
Convention Paper 10292

2:45 pm

P14-3 Spatial B-Format Equalization—*Alexis Favrot, Christof Faller*, Illusonic GmbH, Uster, Switzerland

Audio corresponding to the moving picture of a virtual reality (VR) camera can be recorded using a VR microphone. The resulting A or B-format channels are decoded with respect to the look-direction for generating binaural or multichannel audio following the visual scene. Existing post-production tools are limited to only linear matrixing and filtering of the recorded channels when only the signal of a VR microphone is available. A time-frequency adaptive method is presented: providing native B-format manipulations, such as equalization, which can be applied to sound arriving from a specific direction with a high spatial resolution, yielding a backwards compatible modified B-format signal. Both linear and adaptive approaches are compared to the ideal case of truly equalized sources.
Convention Paper 10293

3:15 pm

P14-4 Exploratory Research into the Suitability of Various 3D Input Devices for an Immersive Mixing Task—*Diego I Quiroz Orozco, Denis Martin*, McGill University, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technoogy (CIRMMT), Montreal, QC, Canada

This study evaluates the suitability of one 2D (mouse and fader) and three 3D (Leap Motion, Space Mouse, Novint Falcon) input devices for an immersive mixing task. A test, in which subjects were asked to pan a monophonic sound object (probe) to the location of a pink noise burst (target), was conducted in a custom 3D loudspeaker array. The objectives were to determine how quickly the subjects were able to perform the task using each input device, which of the four was most appropriate for the task, and which was most preferred overall. Results show significant differences in response time between 2D and 3D input devices. Furthermore, it was found that localization blur had a significant influence over the subject's response time, as well as "corner" locations.
Convention Paper 10294

3:45 pm

P14-5 The 3DCC Microphone Technique: A Native B-format Approach to Recording Musical Performance—*Kathleen "Ying-Ying" Zhang, Paul Geluso*, New York University, New York, NY, USA

In this paper we propose a "native" B-format recording technique that uses dual-capsule microphone technology. The three dual coincident capsule (3DCC) microphone array is a compact sound?eld capturing system. 3DCC's advantage is that it requires minimal matrix processing during post-production to create either a B-format signal or a multi-pattern, discrete six-channel output with high stereo compatibility. Given its versatility, the system is also capable of producing a number of different primary and secondary signals that are either natively available or derived in post-production. A case study of the system's matrixing technique has resulted in robust immersive imaging in a multichannel listening environment, leading to the possibility of future development of the system as a single six-channel soundfield microphone.
Convention Paper 10295

Audio Builders Workshop
AB04 - REVIVING CLASSIC AND ESOTERIC TECH
Friday, October 18, 1:45 pm – 2:45 pm, Room 1E13

Moderator: **Chris Kincaid**, Audio Builders Workshop, Louisville, KY, USA

Panelists: *Mike Buffington*
Stephen Masucci

Mike Buffington and Stephen Masucci are working to keep historical instruments relevant and on the airwaves. They will discuss the techniques, tools, and methods used to bring back instruments made by G. Jenny, C. Musser, L. Theremin, and others, and create new instruments inspired by classics!

AES Meetings
HISTORICAL COMMITTEE MEETING
Friday, October 18, 2:00 pm – 3:30 pm, Room 1C04

AES Mix with the Masters Workshops
MM21 - DAVID KAHNE
Friday, October 18, 2:00 pm – 3:00 pm
Mix with the Masters Workshop Stage

Immersive & Spatial Audio
IS06 - CAPTURING REALITY WITH THE USE OF SPATIAL SOUND AND HIGH ORDER AMBISONICS—ETHNOGRAPHIC AND SIX DEGREES OF FREEDOM (6DOF) CASE STUDIES
Friday, October 18, 2:15 pm – 3:15 pm, Room 1E08

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

Panelists: *Florian Grond*, McGill University, Montreal, Canada
Eduardo Patricio, Zylia Sp. z o.o., Poznan, Poland
Zack Settel, University of Montreal, Montreal, Quebec, Canada

This workshop will present spatial sound works from a practical perspective. Professional audio engineers and musicians will discuss their 360°, 3D, and ambient productions combining sound, image, and written text. The speakers will address the use of spatial audio and ambisonics for creating immersive representations of reality, including six-degrees-of-freedom live recorded sound. The need for new thinking and specific tools will be discussed and demonstrations of examples created via experimental 6DoF end-to-end workflows will be presented. The workshop will focus especially on the usage of spherical microphone arrays that enables the recording of entire 3D sound scenes as well as six degrees of freedom experiences (6DoF VR). Additionally, the workshop will address Ambisonics and the separation of individual sound sources in post-production, which give creators access to unique sonic possibilities.

Technical Committee Meeting
LOUDSPEAKERS AND HEADPHONES
Friday, October 18, 2:00 pm – 3:00 pm, Room 1C02

AoIP Pavilion
AIP29 - AES67 / ST 2110 / NMOS—AN OVERVIEW ON CURRENT SDO ACTIVITIES
Friday, October 18, 2:00 pm – 2:30 pm
AoIP Pavilion Theater

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

Update and report on current standardization activities, including AES67, SMPTE ST 2110. Refresh / summary on commonalities and constraints between AES67 and ST 2110. Brief overview on NMOS developments and activities.

Electronic Dance Music Stage
EDJ07 - ALLEN & HEATH REELOP, XONE & HERCULES PRESENTS: BRIDGING THE GAP BETWEEN HOME PRODUCTION AND THE STAGE
Friday, October 18, 2:00 pm – 3:00 pm
Electronic Dance Music & DJ Stage

Presenter: **Jamie Thompson**

Come learn what you will need to take your home music production to the stage and perform using tools like DJ mixers and midi controllers, as well as software programs like Ableton Live and Traktor Pro. There will be a live demonstration using all of these tools along with a Q&A segment to answer any questions you may have.

Live Production Stage
LS16 - YAMAHA PRESENTS: MIXING FURRY MONSTERS
Friday, October 18, 2:00 pm – 2:45 pm
Live Production Stage

Presenter: **Chris Prinzivalli**, Sesame Street

Multi Emmy Award winning Chris Prinzivalli, Production Mixer of Sesame Street, speaks about audio techniques for everyone's favorite Street.

Project Studio Expo Recording Stage
RS16 - PRODUCE LIKE A PRO WITH WARREN HUART
Friday, October 18, 2:00 pm – 2:45 pm
Recording Stage

Presenter: **Warren Huart**, Produce Like a Pro

Software@AES
S26 - SONIBLE
Friday, October 18, 2:00 pm – 2:30 pm
Software@AES Pavilion

Broadcast & Online Delivery
B15 - FACILITY DESIGNED FOR IP
Friday, October 18, 2:30 pm – 4:00 pm, Room 1E07

Moderator: **Andy Butler**, PBS

Panelists: *Emeric Feldmar*, WGBH, Boston, MA, USA
Kent Terry, Dolby Laboratories Inc., San Francisco, CA, USA

I'm Designing my New "BLANK" with all IP Connectivity . . . that makes it easy, right?

Oh my Friend, the more you learn the less certain you become . . .

OK – I'll make it easy—I'm going to use all network gear—Really? Which one of the several companies they've absorbed this year are you planning to utilize . . . do they still manufac-ture/market that product line or do you plan on hoping for "inventory clearance" pricing so you can pre-buy spares, after all tubes made a comeback . . . right? Besides nobody ever got fired for buy-ing IBM (seen ANY IBM branded stuff around lately?). OK so I'm gonna be safe – EVERY-THING will be fully Standards Compliant. Interesting Concept—which one you thinking about using? Dante, yep that's a walled off inferno. AES isn't done—or maybe more like barely started; oh and SMPTE—48 flavors and 19 variations and growing every second.

AES and SBE bravely go where sanity ends and lore and legend rule. Are you tough enough to take the ride?

Co-organized by the Audio Engineering Society and the Society of Broadcast Engineers

Hip Hop & R&B
HH02 - THE SOUL OF AN R&B MIXS
Friday, October 18, 2:30 pm – 3:30 pm, Room 1E12

Moderator: **Paul "Willie Green" Womack**, Willie Green Music, Brooklyn, NY, USA

Panelist: *Prince Charles Alexander*, Berklee College of Music, Boston, MA, USA

While Hip-Hop and R&B are usually connected in conversation, mixing for the two genres can require very different skills and mindsets. Here we will discuss R&B specific mixing techniques.

Student Events & Career Development
SC17 - SPARS MENTORING
Friday, October 18, 2:30 pm – 4:00 pm, South Concourse A

Moderator: **Drew Waters**, VEVA Sound

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.

AoIP Pavilion
AIP30 - AUDIO MONITORING SOLUTIONS FOR AUDIO-OVER-IP
Friday, October 18, 2:30 pm – 3:00 pm
AoIP Pavilion Theater

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

ST 2110 and AES67 have established audio-over-IP as the next generation standard for audio monitoring. This presentation discusses the benefits of using audio-over-IP over traditional audio monitoring methods, and why the change to audio-over-IP is happening with increasing speed. Practical case examples of using audio-over-IP in professional broadcasting as well as in AV install audio applications are presented.

Software@AES
S27 - MELODYNE
Friday, October 18, 2:30 pm – 3:00 pm
Software@AES Pavilion

Archiving & Restoration
AR04 - RESTORING HANK WILLIAMS
Friday, October 18, 2:45 pm – 3:45 pm, Room 1E11

Presenters: **Michael Graves**, Osiris Studio, Los Angeles, CA, USA
Cheryl Pawelski, Omnivore Records
Jett Williams
Kelly Zumwalt

To the delight of critics and fans alike, 2013 saw the discovery and release of previously unknown Hank Williams recordings. The "Garden Spot Programs, 1950" album was comprised of four 15-minute radio shows and shed new light on Williams' recording career. It also demonstrated just how good a set of old transcription discs can sound when properly transferred, restored, and mastered. The release ended up winning the Best Historical GRAMMY in 2014. Now, the same team involved with "The Garden Spot Programs, 1950" are revisiting Williams' "Health & Happiness" shows and the "Mother's Best" recordings. Meet the team behind these historic projects; Hank Williams estate representatives Jett Williams and Kelly Zumwalt, Producer Cheryl Pawelski, and mastering engineer/audio restoration specialist Michael Graves.

Recording & Production
RP17 - PLATINUM LATIN ENGINEERS & PRODUCERS (AN AES SPECIAL EVENT)
Friday, October 18, 2:45 pm – 4:15 pm, Room 1E15+16

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

Panelists: *Carli Beguerie*, Studio Instrument Rentals/Mastering Boutique, New York, NY, USA
Mauricio Gargel, Mauricio Gargel Audio Mastering, Murfreesboro, TN, USA
Andres Millan, Diffusion Magazine; Boutique Pro Audio, Bogotá, Cundinamarca, Colombia
Martin Muscatello, 360 Music Lab
Rafa Sardina, Fishbone Productions, Inc., Los Angeles, CA, USA
Camilo Silva F., Camilo Silva F. Mastering, Chia, Cundinamarca, Colombia

This Panel gathers every year a selected group of Latin producers and engineers that will present their multi-Grammy Award-winning work and explain in detail how they deal with the ever growing Latin recording industry.

Acoutics & Psychoacoustics
AP07 - DIGITAL FILTERS, FILTER BANKS AND THEIR DESIGN FOR AUDIO APPLICATIONS —WITH PYTHON EXAMPLES
Friday, October 18, 3:00 pm – 4:00 pm, Room 1E13

Presenter: **Gerald Schuller**, Ilmenau University of Technology, IImenau, Germany; Fraunhofer Institute for Digital Media technology (IDMT), Ilmenau, Germany

This tutorial will teach how to design "Finite Impulse Response" and "Infinite Impulse Response" filters for audio applications, in theory and practice, and will give examples in the popular Open Source programming language Python. Then it will go on to show how to design and use filter banks. Examples will be the "Modified Discrete Cosine Transform" (MDCT) filter bank, the (Integer-to-Integer) "IntMDCT," and Low Delay filter banks, which are widely used in MPEG audio coding standards. Further it will show digital filters as predictors for predictive coding, with applications in MPEG Lossless coding standards. Finally it will show how to implement filter banks as convolutional neural networks, which makes them "trainable" and eases the use of GPU's. This has applications for instance in audio source separation..

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

Immersive & Spatial Audio
IS16 - FLORIAN CAMERER TALK&PLAY
Friday, October 18, 3:00 pm – 4:00 pm, Room 1E17

Presenter: **Florian Camerer**, ORF - Austrian TV - Vienna, Austria; EBU - European Broadcasting Union

Having finally arrived where human beings are all the time, immersive audio recording and reproduction of sound is here to stay. Besides the ubiquitous 3D-audio bombardment of action movies, music and sound effects provide potentially more subtle but certainly not less compelling listening experiences. In the latter realm (atmosphere recording), Florian Camerer has gone the extra mile to explore the frontiers of quality for location sound setups in 3D audio. "Nothing is more practical than a good theory" is the foundation of his immersive outdoor rig. The thinking behind its dimensions will be covered, and the audience can judge themselves if the practical result holds up to the theory through the examples that will be played.

Student Events & Career Development
SC18 - STUDENT RECORDING COMPETITION—PART 2
Friday, October 18, 3:00 pm – 6:00 pm, Room 1E06

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.

Technical Committee Meeting
AUTOMOTIVE AUDIO
Friday, October 18, 3:00 pm – 4:00 pm, Room 1C02

AES Mix with the Masters Workshops
MM22 - MICHAEL BRAUER
Friday, October 18, 3:00 pm – 4:00 pm
Mix with the Masters Workshop Stage

AoIP Pavilion
AIP31 - NETWORK AUTOMATION WITH GOOGLE SHEETS?
Friday, October 18, 3:00 pm – 3:30 pm
AoIP Pavilion Theater

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

There are multiple ways to automate your network infrastructure, but what if you need a foolproof and quick solution to let non-network engineers automate certain functions of your network? We will tell you what we've done at the JT-NM Tested events to overcome that!

Audio Builders Workshop Booth Talks
ABT15 - ADVANCED SIGMASTUDIO USE AND THE ANALOG DEVICES ADAU1452 FAMILY OF SIGMADSP
Friday, October 18, 3:00 pm – 3:30 pm, Booth 266

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

Electronic Dance Music Stage
EDJ08 - STUDIO DMI PRESENTS LUCA PRETOLESI MASTERCLASS ON IZOTOPE OZONE 9
Friday, October 18, 3:00 pm – 3:45 pm
Electronic Dance Music & DJ Stage

Presenter: **Luca Pretolesi**, Studio DMI, Las Vegas, NV, USA

Live Production Stage
LS17 - LARGE SCALE FESTIVAL SOUND SYSTEMS
Friday, October 18, 3:00 pm – 3:45 pm
Live Production Stage

Project Studio Expo Recording Stage
RS17 - LISTENING WITH AMAZON MUSIC HD: HIGH QUALITY STREAMING FOR THE MASSES
Friday, October 18, 3:00 pm – 3:45 pm
Recording Stage

Presenters: **John Farrey**, Amazon Music, Seattle, WA, USA
Jack Rutledge

Streaming music in high definition gives listeners the ability to hear songs the way artists originally recorded them, with all of the emotion, detail, and instrumentation of the original recordings. With the recent launch of Amazon Music HD, music fans no longer need to sacrifice the sound quality that is often compressed for the convenience of streaming music. John Farrey and Jack Rutledge will use this opportunity to talk about the ideation and reception of Amazon Music HD: Amazon Music's new lossless streaming offering, bringing customers the highest quality streaming audio available for the mass market.

Software@AES
S28 - FABFILTER
Friday, October 18, 3:00 pm – 3:30 pm
Software@AES Pavilion

Product Development
PD15 - HARDWARE DEVELOPMENT USING JIRA
Friday, October 18, 3:15 pm – 4:45 pm, Room 1E09

Presenter: **Remi Vaucher**, QSC, Costa Mesa, CA, USA

In line with the "Virtual Product Development" theme this presentation will show how to define "evolving" templates of product development based on AGILE methodology and their adaptation into a real development.

* Problems met during traditional product development and a way to solve them -> Management of all Products REQUIREMENTS, TASKS & QUALIFICATIONS using JIRA/Confluence from Atlassian

* Organizing the execution of the hardware development (in our case a powered speaker, including Acoustic, DSP, Amplifier & mechanical components) using an AGILE "like" process (derived from the SCRUM process used for Software).

* Configuring JIRA/CONFLUENCE to meet our goals

* Real Experiment: this new approach has been prototyped successfully during 10 months, improvements and take away will be discussed.

Sound Reinforcement
SR07 - PSYCHOACOUSTICS FOR SOUND ENGINEERS
Friday, October 18, 3:15 pm – 5:15 pm, Room 1E21

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

The tutorial will discuss and illustrate a number of psychoacoustic phenomena and effects that we use every day when designing and setting up sound systems. Understanding how we hear and discriminate sounds can lead to improved system design, alignment, and optimization. Topics that will be discussed include how we integrate and perceive sounds arriving from different directions and at different times and levels, perception of frequency and frequency balance, the audible effects of latency in systems, IEMs and sound / video synchronization, the Lombard speech and level effect, binaural listening versus monaural measurement.

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

Engineering Briefs Session EB04 **Friday, Oct. 18**
3:30 pm – 5:00 pm **South Concourse A**

POSTERS: RECORDING AND PRODUCTION

3:30 pm

EB04-1 A Comparative Pilot Study and Analysis of Audio Mixing Using Logic Pro X and GarageBand for IOS—*Jiayue Cecilia Wu,[1] Orchisama Das,[2] Vincent DiPasquale[1]*
[1] University of Colorado Denver, Denver, CO, USA
[2] Center for Computer Research in Music and Acoustics (CRMA), Stanford University, Stanford, CA, USA

In this pilot study we compare two mixes of a song done with GarageBand on iOS and Logic Pro X in a professional studio environment. The audio tracks are recorded and mastered in the same controlled environment by the same engineer. A blind listening survey was given to 10 laypersons and 10 professional studio engineers who have at least 10 years of related experience. 80% lay persons and 60% professional studio engineers reported a higher preference for the Logic Pro X version. To further compare these two productions, we look at (1) short-term perceptual loudness to quantify dynamic range and (2) power spectral densities in different frequency bands to quantify EQ. The analysis provides evidence to back the survey results. The purpose of this study is to examine how, in a real-life scenario, a professional studio engineer can produce the best results using the best plugins, effects, and tools available in GarageBand on iOS and Logic Pro X environment, and how these two results are comparatively perceived by both the general audience and professional audio experts.
Engineering Brief 539

3:30 pm

EB04-2 The ANU School of Music Post-Production Suites: Design, Technology, Research, and Pedagogy— *Samantha Bennett, Matt Barnes*, Australian National University, Canberra, Australia

This engineering brief considers the design, construction, technological capacity, research, and pedagogical remit of two post-production suites built at the ANU School of Music. These suites were constructed simultaneously to the recording studio refurbishment, as detailed in AES e-Brief #397 (2017). This new e-Brief first considers the intention and purpose behind the splitting of a single, large control room into two separate, versatile post-production spaces. Secondly, the e-Brief focuses on design and construction, with consideration given to acoustic treatment, functionality, ergonomic workflow, and aesthetics. The e-Brief also focuses technological capacity and the benefits of built-in limitations. Finally, the post-production suites are considered in the broader context of both the research and pedagogical activities of the School.
Engineering Brief 540

3:30 pm

EB04-3 A Case Study of Cultural Influences on Mixing Preference—Targeting Japanese Acoustic Major Students—*Toshiki Tajima, Kazuhiko Kawahara*, Kyushu University, Fukuoka, Japan

There is no clear rule in the process of mixing in popular music production, so even with the same music materials, different mix engineers may arrive at a completely different mix. In order to solve this highly multidimensional problem, some listening experiments of mixing preference have been conducted in Europe and North America in previous studies. In this study additional experiments targeting Japanese major students in the field of acoustics were conducted in an acoustically treated listening room, and we integrated the data with previous ones and analyzed them together. The result showed a tendency for both British students and Japanese students to prefer (or dislike) the same engineers' works. Furthermore, an analysis of verbal descriptions for mixing revealed that they gave most attention to similar listening points, such as "vocal," and "reverb."
Engineering Brief 541

3:30 pm

EB04-4 A Dataset of High-Quality Object-Based Productions— *Giacomo Costantini,[1] Andreas Franck,[1] Chris Pike,[2] Jon Francombe,[2] James Woodcock,[3] Richard J. Hughes,[3] Philip Coleman,[4] Eloise Whitmore,[5] Filippo Maria Fazi[1]*
[1] University of Southampton, Southampton, UK
[2] BBC Research and Development, Salford, UK
[3] University of Salford, Salford, UK
[4] University of Surrey, Guildford, Surrey, UK
[5] Naked Productions, Manchester, UK

Object-based audio is an emerging paradigm for representing audio content. However, the limited availability of high-quality object-based content and the need for usable production and reproduction tools impede the exploration and evaluation of object-based audio. This engineering brief introduces the S3A object-based production dataset. It comprises a set of object-based scenes as projects for the Reaper digital audio workstation (DAW). They are accompanied by a set of open-source DAW plugins—the VISR Production Suite—for creating and reproducing object-based audio. In combination, these resources provide a practical way to experiment with object-based audio and facilitate loudspeaker and headphone reproduction. The dataset is provided to enable a larger audience to experience object-based audio, for use in perceptual experiments, and for audio system evaluation.
Engineering Brief 542

3:30 pm

EB4-5 An Open-Access Database of 3D Microphone Array Recordings—*Hyunkook Lee, Dale Johnson*, University of Huddersfield, Huddersfield, UK

This engineering brief presents open-access 3D sound recordings of musical performances and room impulse responses made using various 3D microphone arrays simultaneously. The microphone arrays comprised OCT-3D, 2L-Cube, PCMA-3D, Decca Tree with height, Hamasaki Square with height, First-order and Higher-order Ambisonics microphone systems, providing more than 250 different front-rear-height combinations. The sound sources recorded were string quartet, piano trio, piano solo, organ, clarinet solo, vocal group, and room impulse responses of a virtual ensemble with 13 source positions captured by all of the microphones. The recordings can be freely downloaded from www.hud.ac.uk/apl/resources. Future studies will use the recordings to formally elicit perceived attributes for 3D recording quality evaluation as well as for spatial audio ear training.
Engineering Brief 543

Hip Hop & R&B
HH03 - STUDIO DESIGN FOR HIP-HOP AND R&B
Friday, October 18, 3:30 pm – 4:30 pm, Room 1E12

Moderators: **David Malekpour**, Professional Audio Design, Inc.
Paul "Willie Green" Womack, Willie Green Music, Brooklyn, NY, USA

Panelists: *IZ Avila*
Larrance Dopson
Dave Hampton
Frank Hendler
Rick St. Hilaire

The musical needs of different genres can require different tool sets in the studio. Here we will explore the tools needed to outfit Hip-Hop and R&B studios of all sizes and budgets.

Immersive & Spatial Audio
IS07 - SIX-DEGREES-OF-FREEDOM (6DOF) SOUND CAPTURE AND PLAYBACK USING MULTIPLE HIGHER ORDER AMBISONICS (HOA) MICROPHONES
Friday, October 18, 3:30 pm – 4:30 pm, Room 1E08

Presenters: **Lukasz Januszkiewicz**, Zylia Sp. z o.o., Poznan, Poland
Eduardo Patricio, Zylia Sp. z o.o., Poznan, Poland
Tomasz Zernicki, Zylia sp. z o.o., Poznan, Poland

This workshop is addressed to all who wish to learn about capturing immersive audio scenes for the purpose of virtual reality or music production. More specifically, this workshop will focus on a strategy for recording sound and enabling six-degrees-of-freedom playback, making use of multiple simultaneous and synchronized Higher Order Ambisonics (HOA) recordings. Such strategy enables users to navigate a simulated 3D space and listen to the six-degrees-of-freedom recordings from different perspectives. Additionally, during this workshop we will describe challenges related to creating a Unity based navigable 3D audiovisual playback system.

Audio Builders Workshop Booth Talks
ABT16 - LEARN TO SOLDER
Friday, October 18, 3:30 pm – 4:00 pm, Booth 266

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

Software@AES
S29 - BITWIG
Friday, October 18, 3:30 pm – 4:00 pm
Software@AES Pavilion

Archiving & Restoration
AR05 - PRESERVE THIS PODCAST
Friday, October 18, 4:00 pm – 4:30 pm, Room 1E11

Presenters: **Sarah Nguyen**, Metro
Molly Schwartz

Overview of the founding, funding, and development of Preserve This Podcast, a Mellon grant-funded project to develop a zine, podcast, and series of workshops to teach podcasters how to protect their work.

There is a big problem in the podcast universe: that, like all popular mass mediums, podcasts are at risk of disappearing in the face of rapid shifts in platform, delivery, and recording technology. This issue has been identified by those in the field of archiving and preservation as endemic to mass media technologies to date (such as reel-to-reels, VHS tapes and CDs). It is even more of a worry for digital content that can be easily wiped, corrupted, or replaced with a software update. Preserve This Podcast (PTP) is a 2-year Andrew W. Mellon Foundation grant-funded project whose goal is to create a podcast, zine and website, all which provide indie podcasters the tools and know-how to organize, backup and describe their digital files.

Broadcast & Online Delivery
B16 - IMMERSIVE AUDIO MIXING AND WORKFLOW FOR BROADCAST
Friday, October 18, 4:00 pm – 5:30 pm, Room 1E07

Moderator: **Sean Richardson**, Starz Entertainment, Denver, CO, USA

Panelists: *Cheryl Ottenriter*, OTT House Audio
Robert Reams, Psy(x) Research
Andrew Roundy, Dolby Laboratories

Consumer and corporate broadcast demand for 3D and personalized audio to augment next genera-tion video 4K, 8K, HDR has created both opportunities and challenges. Advanced audio production and distribution end-to-end specifications and best practices are coming into play for both linear broadcast and streaming paradigms. Come see four leading innovators in broadcast audio each present their sphere of expertise. What limitations or fine-tuning are desirable in an immersive mix to successfully transport through a broadcast plant and translate successfully for a typical consumer? Are interim distribution systems a way to jump start this impressive technology on a budget? The audience is listening!

Student Events & Career Development
SC19 - SOUND GIRLS MENTORING
Friday, October 18, 4:00 pm – 5:30 pm, South Concourse A

Please join SoundGirls for a Speed Mentoring Session with Industry Veterans. Get answers to the questions you have about working in professional audio. Sessions will be 30 minutes and we will rotate among mentors.

Recording Arts
Fela Davis: Recording and Live Sound Engineer, Co-Owner of 23db Sound - New York
Jessica Thompson: Audio mastering, restoration and archiving
Catherine Vericolli : owner, engineer and manager of Fivethirteen in Tempe, AZ

Live Sound
Michelle Sabolchick Pettinato FOH Engineer for Elvis Costello,Styx, Mr. Big, Goo Goo Dolls,
Michelle is Co-Founder of SoundGirls
Gil Eva Craig Live Sound Engineer and Partner in Western Audio New Zealand
Barbara Adams Live Sound Engineer and Audio Instructor
Karrie Keyes Monitor Engineer Pearl Jam and Eddie Vedder Karrie is a Co-Founder SoundGirls
Amanda Raymond Live Sound Engineer

Manufacturing
Sara Elliot VUE AudioTechnik VP of Operations

Mentors subject to change, more mentors TBA

AES Meeting
REGIONS & SECTIONS MEETING
Friday, October 18, 4:00 pm – 6:00 pm, Room 1C04

Technical Committee Meeting
SEMANTIC AUDIO ANALYSIS
Friday, October 18, 4:00 pm – 5:00 pm, Room 1C02

AES Standards Meeting
SC-02-12 WORKING GROUP ON AUDIO APPLICATIONS OF NETWORKS
Friday, October 18, 4:00 pm – 5:30 pm, Room 1C03

The scope of SC-02-12 includes the use of various network types for audio and audio-related applications in professional recording and broadcasting.

Sound Reinforcement
SR08 - YOUR NOISE ISN'T MY NOISE: IMPROVING SOUND EXPOSURE AND NOISE POLLUTION MANAGEMENT AT OUTDOOR EVENTS
Friday, October 18, 4:15 pm – 5:45 pm, Room 1E13

Chair: **Adam J. Hill**, University of Derby, Derby, Derbyshire, UK

Panelists: *Etienne Corteel*, L-Acoustics, Marcoussis, France
Bob McCarthy, Meyer Sound, New York, NY, USA
Elena Shabalina, d&b audiotechnik, Backnang, Germany
Andy Wardle, University of the Highlands & Islands, UK

Sound exposure and noise pollution due to outdoor entertainment events carry implications spanning both public and private life. This isn't a new issue. The AES library contains papers published over 50 years ago discussing these issues, although judging by the continued discussion and debate, it's clear that the industry has yet to reach a universally-accepted solution (or even understanding) of the relevant problems. This workshop is intended to present the current state of affairs surrounding the issue of outdoor event sound/noise, as identified by the AES Technical Committee on Acoustics and Sound Reinforcement (TC-ASR) working group that has been investigating these issues since 2018. The two principal areas of investigation are sound exposure on-site and noise pollution off-site. These issues are different in nature and require distinct approaches to mitigate negative short-term and long-term effects associated with event-related sound/noise exposure.

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

Archiving & Restoration
AR06 - YOU MEAN YOU WANTED THOSE TRACKS?!: CHALLENGES OF PRESERVING MULTITRACK RECORDINGS
Friday, October 18, 4:30 pm – 5:30 pm, Room 1E11

Chair: **Jeff Willens**, New York Public Library, Fair Lawn, NJ, USA

Panelists: *Bryan Hoffa*
Kelly Pribble, Iron Mountain Entertainment Services, Moonachie, New Jersey, USA

Whether on analog tape, digital tape, or born digital media, multitrack recordings make up an increasingly large percentage of archival content in need of preservation. How are various institutions dealing with the problem of multitracks? Do they see a difference between analog and born-digital sources? Do they require alternative workflows? This panel will cover practical and technical considerations of multitrack preservation, including tape degradation, the need for backups, metadata storage, and file-level tagging, as well as the need to develop industry-wide best practices for archiving multitrack recordings.

Presented in collaboration with ARSC (Association for Recorded Sound Collections)

Hip Hop & R&B
HH04 - ENGINEERING FOR THE ABSTRACT: RECORDING Q-TIP AND A TRIBE CALLED QUEST
Friday, October 18, 4:30 pm – 5:30 pm, Room 1E12

Presenter: **Gloria Kaba**, Little Underground

As a member of legendary hip-hop group A Tribe Called Quest, emcee/producer Q-tip has changed the sonic landscape time and time again, most recently with the group's 2016 release "We Got It From Here... Thank You 4 Your Service."

Gloria Kaba is on Q-Tip's very short list of engineers, along with credits including Solange Knowles, Donnie McClurkin, and Madonna. In this session we will get a detailed look at the mak-ing of "We Got it From Here..." from the writing process to capturing a the wide range of perfor-mances and musicians.

Recording & Production
RP18 - COLOR IN MASTERING: THE CREATIVE SIDE OF THE DISCIPLINE
Friday, October 18, 4:30 pm – 5:30 pm, Room 1E17

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

Panelists: *Adam Ayan*, Gateway Mastering Studios, Portland, ME USA
Bob Katz, Digital Domain Mastering, Orlando, FL, USA
Alex Psaroudakis, Alex Psaroudakis Mastering, Quincy, MA, USA

The last several years have given rise to many resources to help people understand the core values and navigate the basic tasks of mastering, but, with few exceptions, there haven't been many discussions from credible sources about circuit topology, color and creative choices, how to evaluate them and how to think about their use. We include some objective measurements of emulations versus real to buttress up our subjective opinions of the topic.

Special Event
SE12 - THE PAST PRESENT AND FUTURE OF THE LEGENDARY QUAD BUILDING
Friday, October 18, 4:30 pm – 5:30 pm, Room 1E15+16

Moderator: **Prince Charles Alexander**, Berklee College of Music, Boston, MA, USA

Panelists: *DG*
Ricky Hosn
Dave Malekpour
Carla Springer

A panel moderated by producer/educator Prince Charles Alexander examines the history and current success of this great epicenter of music production.

Acoustics & Psychocoustics
AP08 - LISTENING TESTS— UNDERSTANDING THE BASIC CONCEPTS
Friday, October 18, 4:45 pm – 5:45 pm, Room 1E08

Presenter: **Jan Berg**, Luleå University of Technology, Piteå, Sweden

Listening tests are important tools for audio professionals as they assist our understanding of audio quality. There are numerous examples of tests, either formally recommended and widely used or specially devised for a single occasion. In order to understand listening tests and related methods, and also to potentially design and fully benefit from their results, some basic knowledge is required. This tutorial aims to address audio professionals without prior knowledge of listening test design and evaluation. The fundamentals of what to ask for, how to do it, whom to engage as listeners, what sort of results that may be expected, and similar issues will be covered.

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

Recording & Production
RP19 - SOUND REPORDUCTION ON WHEELS: HOW TO SERVE THE ART
Friday, October 18, 4:45 pm – 5:45 pm, Room 1E09

Chair: **Rafael Kassier**, Harman Becker Automotive Systems GmbH, Karlsbad, Germany

Panelists: *Frank Filipetti*, The Living Room
Richard King, McGill University, Montreal Quebec, Canada
Alan Norton, Ford Motor Company
Darcy Proper, Darcy Proper Mastering at Valhalla Studios, Auburn, NY, USA
Mark Ziemba, Panasonic Automotive US

Reproduction of audio in vehicles is a diverse topic. Certainly the automotive environment provides sometimes significant challenges to audio playback, but also provides specific advantages for system suppliers that are not available for traditional home HiFi manufacturers. Vehicles remain one of the most important environments within which to enjoy audio-only content, and probably the only environment where spatially immersive audio-only reproduction is widely enjoyed. Alongside the traditional challenges of the automotive environment, the advent of autonomous driving will provide further challenges and possibilities, and increasing e-mobility brings renewed focus on component weight and energy efficiency.

This workshop is the second in the series and will probe the connection between the expectations of the audio industry relating to how their art should be enjoyed, with the challenges and opportunities that automotive audio systems provide.

During Part 2, leading professionals from the recording and automotive industries will give their insights and discuss their strategies and approaches in a panel discussion format that should be interesting and entertaining for audience members from either branch of the industry.

Part 1 of this workshop series (which took place at the 2019 AES Automotive Audio Conference in Southern Germany) focused on audio reproduction aesthetics and challenges in automotive audio systems. The outcomes of Part 1 will be summarized and discussed during Part 2.

Broadcast & Online Delivery
B17 - THE CURRENT WAR: DIRECTOR'S CUT
Friday, October 18, 6:00 pm – 8:00 pm
Dolby Theater, 1350 6th Ave. at W. 55th St.

This event is limited to 70 people. Tickets are required (free) and can be obtained at the Registration area.

A special screening provided to the members of the Audio Engineering Society of the upcoming film *The Current War: Director's Cut*.

Three brilliant visionaries set off in a charged battle for the future in *The Current War*, the epic story of the cutthroat competition that literally lit up the modern world. Benedict Cumberbatch is Thomas Edison, the celebrity inventor on the verge of bringing electricity to Manhattan with his radical new DC technology. On the eve of triumph, his plans are upended by charismatic businessman George Westinghouse (Michael Shannon), who believes he and his partner, the upstart genius Nikolai Tesla (Nicholas Hoult), have a superior idea for how to rapidly electrify America: with AC current. As Edison and Westinghouse grapple for who will power the nation, they spark one of the first and greatest corporate feuds in American history, establishing for future Titans of Industry the need to break all the rules. Directed by Alfonso Gomez-Rejon (Me and Earl and the Dying Girl) with Producer Timur Bekmambetov, Basil Iwanyk and Executive Producer Martin Scorsese, *The Current War* also stars Katherine Waterston, Tom Holland, Matthew Macfadyen, and Tuppence Middleton.

Paper Session P15 Saturday, Oct. 19
9:00 am – 11:30 am Room 1E10

AUDIO EDUCATION

Chair: **Amandine Pras**, Digital Audio Arts, University of Lethbridge, Lethbridge, Alberta, Canada; School for Advanced Studies in the Social Sciences, Paris, France

9:00 am

P15-1 Production Processes of Pop Music Arrangers in Bamako, Mali—*Amandine Pras,*[1,2] *Kierian Turner,*[1] *Toby Bol,*[1] *Emmanuelle Olivier*[2,3]
[1] Digital Audio Arts, University of Lethbridge, Lethbridge, Alberta, Canada
[2] School for Advanced Studies in the Social Sciences, Paris, France
[3] National Centre for Scientific Research, Paris, France

Bamako, economic capital of Mali in West Africa, saw the recent multiplication of digital studios based on Cubase 5, FL Studio, cracked plugins, a MIDI keyboard, and a small cabin with a cheap condenser microphone and a pop-filter. From videos and screen captures of recording sessions in three of these studios, we analyzed the creative process of four DAW practitioners from the beginning of the beat production to the mastering of the track. We also examined their interaction with the singers and rappers. Our analyses showed that young Malian DAW practitioners constantly revisit their MIDI arrangement and vocal recordings with advanced editing techniques. Locally successful, they have quickly developed a notoriety that enables them to be directive with their clients.
Convention Paper 10296

9:30 am

P15-2 Towards a Pedagogy of Multitrack Audio Resources for Sound Recording Education—*Kirk McNally,*[1] *Paul Thompson,*[2] *Ken Scott*[2]
[1] University of Victoria, Victoria, BC, Canada
[2] Leeds Beckett University, Leeds, West Yorkshire, UK

This paper describes preliminary research into pedagogical approaches to teach and train sound recording students using multitrack audio recordings. Two recording sessions are described and used to illustrate where there is evidence of technical, musical, and socio-cultural knowledge in multitrack audio holdings. Approaches for identifying, analyzing, and integrating this into audio education are outlined. This work responds to the recent AESTD 1002.2.15-02 recommendation for delivery of recorded music projects and calls from within the field to address the advantages, challenges, and opportunities of including multitrack recordings in higher education teaching and research programs.
Convention Paper 10297

10:00 am

P15-3 Late Withdrawal [*Convention Paper 10298]*

10:30 am

P15-4 Mental Representations in Critical Listening Education: A Preliminary Study—*Stephane Elmosnino*, University of Technology Sydney, Sydney, New South Wales, Australia

This paper reports on a survey of critical listening training offered at tertiary education providers in the USA, UK, Australia, and Canada. The purpose of the investigation is to explore the concept of mental representations in educational contexts, as instructional materials do not always consider this aspect despite a rich research terrain in the field. The analysis shows a wide diversity of instructional methods used, seemingly influenced by course subject matter and institution business model. It also reveals a need to accurately define the concept of critical listening, depending on the context of its use. This study provides the background to a proposed evaluation of the effectiveness of mental representation models applied to new instructional designs.
Convention Paper 10299

11:00 am

P15-5 The Generation Gap—Perception and Workflow of Analog vs. Digital Mixing—*Ryland Chambers-Moranz,*[1] *Amandine Pras,*[1,2] *Nate Thomas*[1]
[1] Digital Audio Arts, University of Lethbridge, Lethbridge, Alberta, Canada
[2] School for Advanced Studies in the Social Sciences, Paris, France

Are sound engineers showing preference for the mixing technology of their generation? We interviewed producer Ezequiel Morfi who owns TITANIO in Buenos Aires and contrasted his opinions with those of four mixers based in Western Canada who were required to use analog-only or digital-only mixing tools when preparing stimuli for this study. To ascertain the myths about which technology sounds superior, 19 trained listeners of ages 17–37 compared analog and digital mixing versions of 8 pop-rock tracks in a double-blind listening test. The main results showed that the analog version of one track was significantly preferred by 79% of the listeners (p=.02), and we observed a slight trend towards the significance of age on preference for the analog format (p=.09).
Convention Paper 10300

Engineering Briefs Session EB05 Saturday, Oct. 19
9:00 am – 11:30 am Room 1E11

TRANSDUCERS

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

9:00 am

EB05-1 The Application of Graphene Oxide-Based Loudspeaker Membranes in 40mm Headphone Drivers—*William Cardenas, Robert-Eric Gaskell*, ORA Graphine Audio Inc., Montreal, Quebec, Canada

Graphene oxide-based materials have shown promise in loudspeaker membrane applications. The material allows the forming of highly stiff, low mass cones and domes for loudspeakers. The technology allows improvements in efficiency and linearity over other common loudspeaker membrane materials. This class of graphene material can be engineered to produce an excellent ratio of stiffness (Young's modulus) to density (g/cm3) and damping (tan ?). In a case study, acoustically optimized graphene materials were formed into membranes for headphone drivers. The performance of headphone drivers made with these membranes was analyzed and compared to standard polymer membrane headphone drivers. Relative to the polymer membrane drivers, the graphene membranes provide a significant reduction in both intermodulation and harmonic distortion while matching the sensitivity and producing a substantially smoother frequency response.
Engineering Brief 544
Presented by Robert-Eric Gaskell

9:15 am

EB05-2 MEMS Loudspeakers - A New Chip-Based Technology for Ultra-Small Speakers—*Fabian Stoppel,*[1] *Florian Niekiel,*[1] *Andreas Männchen,*[2] *Daniel Beer,*[2] *Bernhard Wagner*[1]
[1] Fraunhofer Institute for Silicon Technology ISIT, Itzehoe, Germany
[2] Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Due to the ability to combine exceptional functionality with a very small size and a low price, micro-electro-mechanical systems (MEMS) have become the state-of-the-art solution for many miniaturized components like microphones and inertial sensors. Lately, strongly increasing efforts are being made to exploit the miniaturization potential and the advantages of semiconductor manufacturing processes to create ultra-small loudspeakers. In this context, a new technology for integrated chip speakers is presented. The MEMS speakers utilize multiple piezoelectric bending actuators, which are able to generate high sound pressure levels at low power consumption. Based on the results of an in-ear speaker system, an insight into the technology is given. Moreover, possibilities and challenges for MEMS speakers in general are discussed.
Presentation only; not in E-Library

9:30 am

EB05-3 A Case Study on a Dynamic Driver: How Electromagnet Can Improve the Performance of a Micro Speaker—*Md Mehedi*, Carpenter Technology Corporation, Philadelphia, PA, USA

How can designers improve the sound quality of next-generation audio products when the market is demanding smaller devices? Bigger sound requires bigger speakers, right? Not necessarily. One approach is to re-evaluate the materials you are using. Alloy materials used within speakers to conduct sound has not changed drastically in the last 10-15 years. However, developments in the performance of electromagnet alloys can replace standard electrical iron/low carbon steel and provide higher efficiency and performance. The result is better sound quality, smaller devices, and extended battery life. We studied the performance of the transducer with different electromagnets in magnet assembly and reported the comparison to provide better insight for the next-gen audio dynamic drives.
Engineering Brief 545

9:45 am

EB05-4 Alignment of Triple Chamber Eighth-Order Band-Pass Loudspeaker Systems—*Hao Dong, Yong Shen, Rui Chen*, Nanjing University, Nanjing, China

An eighth-order band-pass loudspeaker system consisting of three vented chambers is analyzed. Since its frequency response has equal low-pass and high-pass cut-off slopes of fourth-order, the response function can be aligned to an eighth-order symmetric band-pass filter obtained by the frequency transformation method. For any desired frequency response, required system alignment parameters can be calculated by solving a system of equations. Design examples are presented and compared in terms of the mid-band attenuation factor and the diaphragm displacement.
Engineering Brief 546

10:00 am

EB05-5 Analysis of a Vented-Box Loudspeaker System via the Impedance Function—*James Lazar, Glenn S. Kubota*, Samsung Research America, Valencia, CA, USA

The vented-box loudspeaker system is studied through a small-signal equivalent circuit model via the impedance function. Some traditional models are found to inadequately represent the system losses, lumping them together in an effort to simplify the design process. Here, a low-frequency small-signal equivalent circuit model is proposed, incorporating five loss elements. The impedance function is derived, and system parameters are determined by curve-fitting the impedance function to measured impedance data. It is shown that the reactive elements determine the critical frequencies, and the lossy elements determine the Q-factors or contribute to the impedance level. Moreover, the lossy elements affect the curve-fit in a unique way, allowing their values to be quantified.
Engineering Brief 547

10:15 am

EB05-6 Designing Listening Tests of SR/PA Systems, A Case Study—*Eddy Bøgh Brixen*, EBB-consult, Smørum, Denmark; DPA Microphones, Allerød, Denmark

It is very common to arrange for comparisons of SR/PA-systems. However, often, these comparisons are organized in a way leaving procedures less transparent and results rather unclear. Standards for the assessment of loudspeakers do exist. The assessors basically must be trained for the purpose, and the set-up should support double-blind testing. However, in the test of big systems, the listening panel is not necessarily trained, and the practical problems of rigging huge arrays to some degree may weaken the procedures and the results. This paper describes considerations for the comparative assessment of SR/PA systems. The paper also reports the outcome of an experiment where considered principles were applied.
Engineering Brief 548

10:30

EB05-7 Noise and Distortion Mechanisms Encountered in Switching Audio Power Amplifier Design—*Robert Muniz*, Harmonic Power Conversion LLC, Doublas, MA, USA

When designing a switching power amplifier, many phenomena are encountered that leave the designer wonder-

ing why performance falls short of what theory predicts. While many sources of non-linearity and noise in the conversion process are known and intrinsic to the sub-systems involved, other sources of error are more subtle. The intent of this paper is to outline the noise, distortion, and error mechanisms commonly encountered in practice when designing a switching (Class-D) power amplifier. By understanding the root cause of these mechanisms, a more heuristic approach can be employed in switching power amplifier design. The focus will be on analog systems employing clocked, naturally sampled modulators, but the bulk of the material will be broadly applicable to any modulation scheme.
Engineering Brief 549

10:45 am

EB5-8 Acoustic Metamaterial in Loudspeaker Systems Design—*Letizia Chisari, Mario Di Cola, Paolo Martignon*, Contralto Audio srl, Parma (PR), Italy

Materials have been used to control waves propagation ever since, and optics is a prime example. In Loudspeaker Systems applications, there have also been approaches in the attempt of controlling waves by acoustic lenses. Metamaterials are artificial structures, typically periodic, composed of small meta-atoms that, in the bulk, behave like continuous material with unconventional effective properties without the constraints normally imposed by nature. This talk offers the opportunity to share what can be done with acoustic metamaterials in audio industry, especially in Loudspeaker Systems Design. The presentation brings back some approaches from the past that can be revisited using today's technologies. Moreover, this talk shows some of the already developed technologies that employ these extremely innovative materials.
Engineering Brief 550

11:00 am

EB5-9 Application of Matrix Analysis for Derivation of Acoustical Impedance of Horns—*Alexander Voishvillo, Balázs Kákonyi, Brian McLaughlin*, Harman Professional Solutions ,Northridge, CA, USA

The direct measurement of a horn's acoustical impedance requires knowledge of both the sound pressure and volume velocity at the throat of the horn. While measuring sound pressure is trivial, the measurement of volume velocity requires special equipment. This work proposes a new derivation method for the acoustical impedance of a horn. The method is based on matrix analysis and consists of two stages: derivation of the compression driver's square transfer matrix of A-parameters, and measurement of electrical impedance and sound pressure at the throat of the horn. These functions yield two matrix equations that relate measured sound pressure and electrical impedance, which allows for an acoustical impedance derivation of the horn. A comparison with COMSOL simulation is provided.
Engineering Brief 551

11:15 am

EB5-10 Application of Modulated Musical Multitone Signal for Evaluation of Horn Driver Sound Quality—*Alexander Voishvillo, Balázs Kákonyi, Brian McLaughlin*, Harman Professional Solutions, Northridge, CA, USA

This work introduces a new type of test signal called Modulated Musical Multitone (MMM): sinusoidal tones outlining E-minor triads in several octaves with amplitude modulation providing a variable crest factor which can match specific musical signals. Three different signals are used in the corresponding experiments including MMM, sinusoidal sweep, and music. An evaluation of sound quality is conducted for an FIR-filtered single horn driver. The effect of masking is observed when a matching linear low-pass channel is added to the signal. The multitone response is post-processed to obtain the distortion products spectrum.
Engineering Brief 552

Acoustics & Psychoacoustics
AP09 - TO PEAQ OR NOT TO PEAQ? - BS.1387 REVISITED
Saturday, October 19, 9:00 am – 10:00 am, Room 1E07

Presenters: **Pablo Delgado**, International Audio Laboratories Erlangen, a joint institution of Universität Erlangen-Nürnberg and Fraunhofer IIS, Erlangen, Germany; Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Thomas Sporer, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Carefully conducted listening tests are time consuming and expensive. Computerized, objective measurement schemes for the assessment of perceived audio quality seem to be an adequate replacement. The Recommendation ITU-R BS.1387 (PEAQ) is a standardized method to assess bit-reduced audio signals However in the last 20 years both audio coding and listening methods have evolved. In addition many authors use PEAQ for assessment of audio processing scheme not known and validated in 1998.

This tutorial will consist of the following parts: • explain what PEAQ is, how it was designed and validated; • show some examples where PEAQ fails to predict perceived quality; • summarizes the work since the standardization concerning newer audio coding tools, spatial audio and listening procedures; • gives an outlook of further developments; • give advice under which circumstance the current version of PEAQ should be used.

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

Archiving & Restoration
AR07 - ARCHIVING THE 90S!
Saturday, October 19, 9:00 am – 10:00 am, Room 1E21

Moderator: **Jason Bitner**, Traffic Entertainment Group, Somerville, MA, USA

Panelists: *Kaylie Ackerman*, Harvard University, Cambridge, MA, USA
Eddie Ciletti, Manhattan Sound Technicians, Inc., West Saint Paul, MN, USA
Kelly Pribble, Iron Mountain Entertainment Services, Moonachie, New Jersey, USA
Catherine Vericolli

Archival practice often spotlights the challenges of working with magnetic tape and grooved media. This panel shifts focus to the formats used frequently in 1990s recording production: ADAT, DA-88 and DA-89, DTRS, 1630. Loads of great records were made on these formats, frequently in project studios with smaller budgets. Sadly, they are some of the most at-risk formats, both because the carriers are awful and the because playback machines in working order are hard to find and maintain. The fact that most of the studios using these formats were smaller project studios with minimal budgets only heightens the urgency of preserving this content. Panelists will talk about playback and preservation of these formats, specific considerations in capturing audio, timecode and other data, sourcing and maintaining playback machines, and curating releases from this content.

Audio Builders Workshop
AB05 - DIY BUILD CLINIC: STARTING YOUR NEXT BUILD TODAY WITH AUDIO BUILDERS WORKSHOP
Saturday, October 19, 9:00 am – 12:00 noon, South Concourse B

Co-Chairs: **Owen Curtin**, Audio Builders Workshop, Lexington, MA, USA; Bridge Sound and Stage, Cambridge, MA, USA
Chris Kincaid, Audio Builders Workshop, Louisville, KY, USA

Panelists: *Peterson Goodwyn*, DIY Recording Equipment, Philadelphia, PA, USA
Brewster LaMacchia, Clockworks Signal Processing LLC ,Andover, MA, USA
Matthew McGlynn, MicParts.com, Sebastopol, CA, USA; Roswell Pro Audio

Sign up for this exclusive opportunity to build your very own gear from DIY Recording Equipment, MicrophoneParts.com, and Audio Builders Workshop module under the supervision of the designers, Peterson Goodwyn, Matthew McGlynn, and Brewster LaMacchia. This 3 hour session requires preregistration and a purchase of the kit. This build clinic is open to everyone, even beginners so if you've been on the fence to build this device or just get into DIY audio now is the perfect opportunity! Leave with a working tool that you can use in your own production. More information for signup can be found at AudioBuildersWorkshop.com.

Audio for Cinema
AC01 - THE PORTABLE SCORE MIX: CINEMATIC SCORES ON AN EPISODIC BUDGET
Saturday, October 19, 9:00 am – 10:00 am, Room 1E17

Presenter: **John Whynot**, Berklee College of Music, Los Angeles, CA, USA

Score mixer and Berklee Professor John Whynot will show his approach, developed over years of working on scores restrained by package deals, to bringing the size and sweep of a cinematic score to smaller productions such as the F/X series "Tyrant," the Amazon series "The Last Tycoon," and the Netflix animation series "3 Below." John will show how he has adapted his workflow to accommodate limited monitoring, the necessity of mixing in a composer's studio with a portable setup, collaborating on-the-spot with the composer during the mix, and the need for speed in package-deal projects.

Immersive & Spatial Audio
IS08 - AMBISONICS TOOLS FOR IMMERSIVE AUDIO CAPTURE AND POST-PRODUCTION
Saturday, October 19, 9:00 am – 12:00 noon, Room 1E13

Presenters: **Ianina Canalis**, National University of Lanús, Buenos Aires, Argentina
Brian Glasscock, Sennheiser
Andres A. Mayo, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina
Martin Muscatello, 360 Music Lab

Over the past 3 years, immersive audio production tools evolved considerably and allowed producers to combine them in many different ways. In this workshop we will provide an in-depth explanation of how Ambisonics works and how it can be a central piece of an immersive audio production workflow. Attendees will be able to experiment with dedicated hardware and software tools during the entire workshop. Bring your own laptop and headphones for a unique learning session!

Preregistration is required for this event. Tickets are $75 (member) and $125 (non-member) and can be purchased on-line when you register for the convention All Access Badge. Seating is limited.

Immersive & Spatial Audio
IS09 - PRODUCING HIGH-QUALITY 360/3D VR CONCERT VIDEOS WITH 3D IMMERSIVE AUDIO
Saturday, October 19, 9:00 am – 10:30 am, Room 1E08

Presenters: **Ming-Lun Lee**, University of Rochester, Rochester, NY, USA
Steve Philbert, University of Rochester, Rochester, NY, USA

Our 3D Audio Research Laboratory at the University of Rochester has recorded over 40 concerts at the Eastman School of Music since Fall 2017. We have used an Orah 4i 4K 360 VR Camera and a Kandao Obsidian R 8K 3D 360 VR Camera to make 360/3D video recordings, as well as two Neumann KU100 Binaural Microphones, a Sennheiser Ambeo Smart Headset, a 32-element mh acoustics em32 Eigenmike microphone array, a Sennheiser Ambeo VR Microphone, a Zoom H3-VR Handy Recorder, a Core Sound TetraMic, and a Core Sound OctoMic to make 3D immersive audio recordings. With Adobe Premiere, we have been able to edit and render high-quality 8K 360/3D concerts videos mixed with binaural recordings for head-locked binaural audio or Ambisonic recordings for head-tracking binaural audio.

This workshop aims to show our optimized workflows for making high-quality VR concert videos from recording, editing, rendering, and finally publishing on YouTube and Facebook. We plan to demonstrate some essential recording and editing techniques with practical examples for the attendants to hear binaural audio with headphones. Making long concert VR videos is much more challenging than making short VR music videos. We have encountered and investigated so many technical issues, including stitching, video/audio drifting, synchronization, and equalization. Therefore, we also want to share our experiences in resolving some critical A/V issues and improving the audio quality. Our session also welcomes the audience to join discussions and share their experiences.

Product Development
PD16 - HOW TO ADD AES70 CONTROL TO YOUR PRODUCTS
Saturday, October 19, 9:00 am – 10:30 am, Room 1E09

Presenters: **Jeff Berryman**, OCA Alliance
Simon Jones, Focusrite
Ethan Wetzell, OCA Alliance

AES70, also known as OCA, is an architecture for system control and connection management of media networks and devices. AES70 is capable of working effectively with all kinds of devices from multiple manufacturers to provide fully interoperable multi-vendor networks. In this session the presenters will provide a deep-dive on the architecture and implementation of the standard, demonstrating how to implement AES70 inside of a device and how AES70 controllers work. Aimed at developers, members of the OCA Alliance will cover topics such as the object model, class tree, protocol and implementation options, and practical advice for AES70 implementation.

Student Events & Career Development
SC20 - RESUME REVIEW (FOR STUDENTS, RECENT GRADS, AND YOUNG PROFESSIONALS)
Saturday, October 19, 9:00 am – 5:00 pm, SDA Booth

Moderator: **Alex Kosiorek**, Central Sound at Arizona PBS, Phoenix, AZ, USA

Students, recent graduates and young professionals… Often your resume is an employer's first impression of you. Naturally, you want to make a good one. Employer's often use job search websites to search for candidates. Some use automated software to scan your resume and in some cases, your LinkedIn/social media profiles as well. Questions may arise regarding formatting, length, keywords and phrases so it shows up in searches and lands on the desk of the hiring manager. No matter how refined your resume may be, it is always good to have someone else review your materials. Receive a one-on-one 20-25 minute review of your resume from a hiring manager who is in the audio engineering business. Plus, if time allows, your cover letter and online presence will be reviewed as well.

Sign up at the student (SDA) booth immediately upon arrival. For those who would like to have your resume reviewed on Wednesday, October 17th prior to SDA-1, please email the request to: aesresumereview@outlook.com. You may be requested to upload your resume prior to your appointment for review. Uploaded resumes will only be seen by the moderator and will be deleted at the conclusion of the 147th Pro Audio Convention.

This review will take place during the duration of the convention by appointment only.

Technical Committee Meeting
NETWORKED AUDIO SIGNALS
Saturday, October 19, 9:00 am – 10:00 am, Room 1C02

AES Standards Meeting
AESSC PLENARY
Saturday, October 19, 9:00 am – 11:00 am, Room 1C03

Summaries of all the individual working group meetings are presented.

Archiving & Restoration
AR08 - FINDING FUNDING: HOW TO CONNECT AUDIO ARCHIVAL COLLECTIONS, VENDORS, AND FUNDERS
Saturday, October 19, 10:00 am – 11:00 am, Room 1E21

Moderator: **John Krivit**, Professional Audio Design, Hanover, MA, USA; Emerson College, Boston, MA, USA

Panelists: *Joy Banks*, CLIR
Steve Rosenthal, MARS (MagicShop Archive and Restoration Studios), Brooklyn, NY, USA
Gerald Seligman
Derek Spencer, GRAMMY Museum

Representatives from major funding organizations will offer an inside look at the grant writing and review process for archival audio projects. Panelists will discuss how to prepare to write a grant, what kinds of projects are likely to get funded, red flags or other obstacles that can derail a grant application, alternatives to grant funding. The takeaway will be a better understanding of how to connect collections, funders, and vendors.

Technical Committee Meeting
PERCEPTION AND AND SUBJECTIVE EVALUATION OF AUDIO SIGNALS
Saturday, October 19, 10:00 am – 11:00 am, Room 1C02

Audio for Cinema
AC02 - BEST PRACTICES IN RE-RECORDING MIXING
Saturday, October 19, 10:15 am – 11:15 am, Room 1E12

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

This master class on re-recording will reprise some of Tom Fleischman's master class from AES145, with a primary focus on the use of developing technologies in cinema. 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.

Broadcast & Online Delivery
B18 - CONVERGENCE OF BROADCAST OVER-THE-AIR AND STREAMING DELIVERY
Saturday, October 19, 10:15 am – 11:45 am, Room 1E07

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

Panelists: *Sayon Deb*, Consumer Technology Association
Jeff Detweiler, Xperi Corp.

Radio broadcasters still reach the vast majority of their listeners with over-the-air (OTA) broadcast signals, but streaming audio is becoming an increasingly popular and important medium which broadcasters must take advantage of, especially when striving to reach younger listeners. New auto receivers are being designed today that make use of both OTA digital radio (HD Radio is the sys-tem used in the US) and streaming audio delivered by mobile broadband services. Smart speakers offer another big opportunity for delivery of broadcast content, both live and in other forms such as podcasts. This session will focus on the technologies and techniques being used to keep radio on the cutting edge.

Paper Session P16 **Saturday, Oct. 19**
10:30 am – 12:00 noon **South Concourse A**

POSTERS: SPATIAL AUDIO

10:30 am

P16-1 Calibration Approaches for Higher Order Ambisonic Microphone Arrays—*Charles Middlicott,*[1,2] *Bruce Wiggins*[1]
[1] University of Derby, Derby, Derbyshire, UK
[2] Sky Labs Brentwood, Essex, UK

Recent years have seen an increase in the capture and production of ambisonic material due to companies such as YouTube and Facebook utilizing ambisonics for spatial audio playback. Consequently, there is now a greater need for affordable high order microphone arrays due to this uptake in technology. This work details the development of a five-channel circular horizontal ambisonic microphone intended as a tool to explore various optimization techniques, focusing on capsule calibration & pre-processing approaches for unmatched capsules.
Convention Paper 10301

10:30 am

P16-2 A Qualitative Investigation of Soundbar Theory—*Julia Perla, Wesley Bulla*, Belmont University, Nashville, TN, USA

This study investigated basic acoustic principals and assumptions that form the foundation of soundbar technology. A qualitative listening test compared 12 original soundscape scenes each comprising five stationary and two moving auditory elements. Subjects listened to a 5.1 reference scene and were asked to rate "spectral clarity and richness of sound," "width and height," and "immersion and envelopment" of stereophonic, soundbar, and 5.1 versions of each scene. ANOVA revealed a significant effect for all three systems. In all three attribute groups, stereophonic was rated lowest, followed by soundbar, then sur-

round. Results suggest waveguide-based "soundbar technology" might provide a more immersive experience than stereo but will not likely be as immersive as true surround reproduction.
Convention Paper 10302

10:30 am

P16-3 The Effect of the Grid Resolution of Binaural Room Acoustic Auralization on Spatial and Timbral Fidelity—*Dale Johnson, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

This paper investigates the effect of the grid resolution of binaural room acoustic auralization on spatial and timbral fidelity. Binaural concert hall stimuli were generated using a virtual acoustics program utilizing image source a nd ray tracing techniques. Each image source and ray were binaurally synthesized using Lebedev grids of increasing resolution from 6 to 5810 (reference) points. A MUSHRA test was performed where subjects rated the magnitudes of spatial and timbral differences of each stimulus to the reference. Overall, it was found that on the MUSHRA scale, 6 points were perceived to be "Fair," 14 points "Good," and 26 points and above all "Excellent" on the grading scale, for both spatial and timbral fidelity.
Convention Paper 10303

10:30 am

P16-4 A Compact Loudspeaker Matrix System to Create 3D Sounds for Personal Uses—*Aya Saito, Takahiro Nemoto, Akira Saji, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

In this paper we propose a new 3D sound system in two-layers as a matrix that has five loudspeakers on each side of the listener. The system is effective for sound localization and compact for personal use. Sound images in this system are created by extended amplitude panning method, with the effect of head-related transfer functions (HRTFs). Performance evaluation of the system for sound localization was made by auditory experiments with listeners. As the result, listeners could distinguish sound image direction localized at any azimuth direction and high elevation direction with small biases.
Convention Paper 10304

10:30 am

P16-5 Evaluation of Spatial Audio Quality of the Synthesis of Binaural Room Impulse Responses for New Object Positions—*Stephan Werner, Florian Klein, Clemens Müller*, Technical University of Ilmenau, Ilmenau, Germany

The aim of auditory augmented reality is to create an auditory illusion combining virtual audio objects and scenarios with the perceived real acoustic surrounding. A suitable system like position-dynamic binaural synthesis is needed to minimize perceptual conflicts with the perceived real world. The needed binaural room impulse responses (BRIRs) have to fit the acoustics of the listening room. One approach to minimize the large number of BRIRs for all source-receiver relations is the synthesis of BRIRs using only one measurement in the listening room. The focus of the paper is the evaluation of the spatial audio quality. In most conditions differences in direct-to-reverberant-energy ratio between a reference and the synthesis is below the just noticeable difference. Furthermore, small differences are found for perceived overall difference, distance, and direction perception. Perceived externalization is comparable to the usage of measured BRIRs. Challenges are detected to synthesize more further away sources from a source position that is more close to the listening positions.
Convention Paper 10305

10:30 am

P16-6 Withdrawn

10:30 am

P16-7 An Adaptive Crosstalk Cancellation System Using Microphones at the Ears—*Tobias Kabzinski, Peter Jax*, RWTH Aachen University, Aachen, Germany

For the reproduction of binaural signals via loudspeakers, crosstalk cancellation systems are necessary. To compute the crosstalk cancellation filters, the transfer functions between loudspeakers and ears must be given. If the listener moves the filters are usually updated based on a model or previously measured transfer functions. We propose a novel architecture: It is suggested to place microphones close to the listener's ears to continuously estimate the true transfer functions and use those to adapt the crosstalk cancellation filters. A fast frequency-domain state-space approach is employed for multichannel system tracking. For simulations of slow listener rotations it is demonstrated by objective and subjective means that the proposed system successfully attenuates crosstalk of the direct sound components.
Convention Paper 10307

10:30 am

P16-8 Immersive Sound Reproduction in Real Environments Using a Linear Loudspeaker Array—*Valeria Bruschi, Nicola Ortolani, Stefania Cecchi, Francesco Piazza*, Universitá Politecnica della Marche, Ancona (AN), Italy

In this paper an immersive sound reproduction system capable of improving the overall listening experience is presented and tested using a loudspeaker linear array. The system aims at providing a channel separation over a broadband spectrum by implementing the RACE (Recursive Ambiophonic Crosstalk Elimination) algorithm and a beamforming algorithm based on a pressure matching approach. A real time implementation of the algorithm has been performed and its performance has been evaluated comparing it with the state of the art. Objective and subjective measurements have con?rmed the effectiveness of the proposed approach.
Convention Paper 10308

10:30 am

P16-9 The Influences of Microphone System, Video, and Listening Position on the Perceived Quality of Surround Recording for Sport Content—*Aimee Moulson, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

This paper investigates the influences of the recording/reproduction format, video, and listening position on the quality perception of surround ambience recordings for sporting events. Two microphone systems—First Order Ambisonics (FOA) and Equal Segment Microphone Array (ESMA)—were compared in both 4-channel (2D) and 8-channel (3D) loudspeaker reproductions. One subject group tested audio-only conditions while the other group was presented with video as well as audio. Overall, the ESMA was rated significantly higher than the FOA for all quality attributes tested regardless of the presence of video. The 2D and 3D reproductions did not have a significant difference within each microphone system. Video had a significant interaction with the microphone system and listening

position depending on the attribute.
Convention Paper 10309

10:30 am

P16-10 Sound Design and Reproduction Techniques for Co-Located Narrative VR Experiences—*Marta Gospodarek, Andrea Genovese, Dennis Dembeck, Corinne Brenner, Agnieszka Roginska, Ken Perlin*, New York University, New York, NY, USA

Immersive co-located theatre aims to bring the social aspects of traditional cinematic and theatrical experience into Virtual Reality (VR). Within these VR environments, participants can see and hear each other, while their virtual seating location corresponds to their actual position in the physical space. These elements create a realistic sense of presence and communication, which enables an audience to create a cognitive impression of a shared virtual space. This article presents a theoretical framework behind the design principles, challenges and factors involved in the sound production of co-located VR cinematic productions, followed by a case-study discussion examining the implementation of an example system for a 6-minute cinematic experience for 30 simultaneous users. A hybrid reproduction system is proposed for the delivery of an effective sound design for shared cinematic VR.
Convention Paper 10287

Immersive & Spatial Audio
IS10 - MUSIC PRODUCTION FOR DOLBY ATMOS
Saturday, October 17, 10:30 am – 11:30 am, Room 1E06

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

Multichannel music productions are increasingly using Dolby Atmos, the audio format originally conceived for the cinema. The speaker setup provides ceiling speakers, which are located in very different positions on the ceiling depending on the number of speakers. This requires special strategies for recording and mixing, so that the result is great regardless of the speaker configuration in the listening room. On the one hand, caution is advised when reproducing direct sound vertically from the ceiling to the listener, as musical instruments are nearly never located directly above the listener during live performances. On the other hand, the ceiling speakers should be used, as they make a significant contribution to the immersive effect.

For larger speaker configurations, additional speakers are added to the side. These make it possible to localize stable sound sources on the side. Sideways positioned instruments lead to largely undesirable results for music because the interaural level differences in this case can be very large and thus appear uncomfortable for the listener. It is therefore recommended to use signals for those speakers that do not lead to such large level differences. One possibility to avoid this unwanted effect can be reached by simultaneously reproducing signals from both side speakers that are so short that their localization from the side does not lead to a disbalance of the mix.

During the workshop Lasse Nipkow will explain the most important phenomena that can be used for music in Dolby Atmos and explain how they can best be used for a loudspeaker setup in a large auditorium. During the presentation various sound and video examples will be shown.

Networked Audio
NA03 - WHAT AUDIO ENGINEERS NEED TO KNOW ABOUT MEDIA NETWORKS
Saturday, October 19, 10:45 am – 12:15 pm, Room 1E08

Presenter: **Patrick Killianey**, Audinate, 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 Audinate. 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
PD17 - USING GAGE R&R TOOLS TO VALIDATE AND CHARACTERIZE ACOUSTIC MEASUREMENT SYSTEMS
Saturday, October 19, 10:45 am – 12:15 pm, Room 1E09

Presenter: **Emily Wigley**, Acoustical Engineer, Shure Inc., Chicago, IL, USA

Product performance is validated through measuring, but how is the system used to take the measurement validated? This is of particular concern in developing end-of-line production measurement systems that are customized to meet the needs of a specific product. A Gage Repeatability and Reproducibility (Gage R&R) study is a set of statistical analysis tools that can be used to characterize the performance of a measurement system and is the first step in determining product performance limits. This session will cover the different types of gage studies, when and how to perform them, and how the use the results to improve the measurement system. Example studies from Shure product development projects will be included.

Student Events & Career Development
SC21 - STUDENT DELEGATE ASSEMBLY, PART 2
Saturday, October 19, 10:45 am – 12:15 pm, Room 1E15+16

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.

Immersive & Spatial audio
IS17 - GENELEC PLAY IMMERSIVE
Saturday, October 19, 11:00 am — 12:00 noon, Room 1E17

A wide collection of excellent immersive music, nature and film recordings, conveyed via a point source 7.1.4 reproduction system.

Technical Committee Meeting
RECORDING AND TECHNOLOGY PRACTICES
Saturday, October 19, 11:00 am – 12:00 noon, Room 1C02

Recording & Production
RP20 - LIVE CONCERT SOUND
Saturday, October 19, 11:15 am – 12:15 pm, Room 1E21

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

Panelists: *Fela Davis*, Christian McBride, KRS One

Liz Ip, MTV, MLB Network
Michelle Sabolchick Pettinato, Gwen Stefani, Jewel

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

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

Sound Reinforcement
SR09 - IMPROVING EDUCATION AND KNOWLEDGE TRANSFER IN SOUND REINFORCEMENT
Saturday, October 19, 11:30 am – 12:30 pm, Room 1E12

Chair: **Elena Shabalina**, d&b audiotechnik, Backnang, Germany

Panelists: *Etienne Corteel*, L-Acoustics, Marcoussis, France
Adam J. Hill, University of Derby, Derby, Derbyshire, UK
Malle Kaas, Women in Live Music
Manuel Melon, Le Mans Université, Le Mans, France

This panel will focus on how people populating the field of sound reinforcement can work together more efficiently. Tasks in sound reinforcement have a wide range: From using a sound system, designing sound system to product development to applied research, and fundamental research. All those tasks involve musicians, mixing engineers, electrical and communication engineers, acousticians and noise control engineers, software developers, signal processing and scientific computing specialists, physicists and mathematicians, to name just a few. They require different educational backgrounds, different professional experience, and often very different ways of thinking. These differences can make the communication and knowledge transfer very challenging in everyday work.

However, in order to create the best possible experience for the audience, all the parts of the chain should work together smoothly, making sure nothing gets lost on the way and no important part is missing. The panelists will address the topic from their unique perspectives and will try to identify what works well in our field and what needs improvement in future.

How do we make sure that all parts of the whole picture get enough attention? Is there enough research? Are we missing some important new ideas being developed in other fields? What to study to work in sound reinforcement? What background is needed to work in fundamental research, R&D, sound system design or to become a mixing engineer? Can users of sound systems get all the knowledge they need? Are there any specific knowledge gaps? What can we do to improve the situation?

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

Student Events & Career Development
SC22 - STUDENT RECORDING CRITIQUES
Saturday, October 19, 12:30 pm – 1:30 pm, Room 1E06

Presenters: **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.

Audio for Cinema
AC03 - YOU DON'T HAVE TO GO THERE: RECORDING CINEMA AUDIO REMOTELY
Saturday, October 19, 1:30 pm – 2:30 pm, Room 1E17

Presenter: **Robert Marshall**, Source Elements

A master class by Robert Marshall from Source Elements, the creator of Source-Connect, Source-Live, Source-Nexus, and other ubiquitous production tools. Robert will walk through the procedures of using real-time internet connections to produce, capture, refine, and transfer performances by actors, musicians, and other talent in cinema productions.

Archiving & Restoration
AR09 - AUDIO REPAIR AND RESTORATION FOR MUSIC AND POST: BUILD YOUR SKILLS
Saturday, October 19, 1:30 pm –4:30 pm, Room 1E13

Presenters: **Dave Barber**, Juniper Post, Burbank, CA, USA
Alexey Lukin, iZotope, Boston, MA, USA
Jessica Thompson, Jessica Thompson Audio, Berkeley, CA, USA
Jonathan Wyner, M Works Studios/iZotope/Berklee College of Music, Boston, MA, USA; M Works Mastering

Single-ended noise reduction and audio repair tools have evolved during the past 35 years to the point that they have become an integral part of the work and workflows across audio disciplines. During this workshop attendees will be lead through an overview of the various sorts of technologies, techniques, and strategies used to solve audio challenges in music and audio post. Attendees will be guided through exercises that will help them develop their skills in audio repair and restoration

Preregistration is required for this event. Tickets are $75 (member) and $125 (non-member) and can be purchased on-line when you register for the convention All Access Badge. Seating is limited.

Broadcast & Online Delivery
B19 - EMERGENCY PREPAREDNESS AND SAFETY FOR BROADCASTERS
Saturday, October 19, 1:30 pm – 2:45 pm, Room 1E07

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

Panelists: *Kirk Harnack*, Telos Alliance
Jim Leifer, American Tower, Boston, MA, USA
Howard Price

For broadcast engineers at all levels, emergency preparedness has never been as important as it is today. Natural disasters are more frequent and longer lasting, stations face ever-increasing security threats, and there are an ever-increasing number of transmission and distribution paths that have to be kept resilient and fully functional.

In this session we will examine multiple aspects of emergency preparedness, including working with local emergency management officials, preparing for the ongoing needs of your staff and their families, hardening facilities to better survive disaster, and current best practices in designing and building backup and disaster recovery facilities for stations large and small.

This session is co-presented with SBE, the Society of Broadcast Engineers. This session is presented in association with the AES Technical Committee on Broadcast and Online Delivery

Game Audio & XR
GA15 - BINAURAL AUDIO - JUST A 360/VR GEEK THING OR FUTURE AUDIO ENTERTAINMENT?
Saturday, October 19, 1:30 pm – 3:00 pm, Room 1E08

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

Panelists: *David Miles Huber*, Seattle, WA, USA
Andres A. Mayo, Andres Mayo Mastering & Audio Post, Buenos Aires, Argentina
Agnieszka Roginska, New York University, New York, NY, USA
Thilo Schaller, SUNY Buffalo , Buffalo, NY, USA

Is it just a fashion that over 85% of the people listen to their music on headphones? Is 360/VR/AR the only application where binaural audio is applicable? Could the mobile entertainment gain the need for new binaural audio productions in music, film, interactive, and game? And how to create and deliver binaural audio to the market? This and a lot more questions will be discussed by a panel of experienced binaural audio persons.

Networked Audio
NA04 - NETWORK AUDIO STRATEGIES: CHARTING SOFTWARES, HARDWARES, INTERNET REQUIREMENTS
Saturday, October 19, 1:30 pm – 2:30 pm, Room 1E21

Presenter: **Sarah Weaver**, NowNet Arts, Inc.

This workshop will address network audio strategies based on internet bandwidth, quality of service, network type (public or private/research), and administration configurations. The strategies are matched with current network audio software and hardware utilizing varieties of channel numbers and qualities to provide a comprehensive chart for usage of the technology. The chart illustrates strategies including and beyond Gigabit Ethernet for broader access, efficiency, and sustainability in network audio across a wide scope of public and private networks.

Product Development
PD18 - ACOUSTIC METAMATERIAL IN LOUDSPEAKER SYSTEMS DESIGN
Saturday, October 19, 1:30 pm – 3:00 pm, Room 1E09

Presenters: **Letizia Chisari**, Contralto Audio srl, Casoli, Ital
Mario Di Cola, Contralto Audio srl, Casoli, Italy
Paolo Martignon, Contralto Audio srl, Casoli, Italy

This talk offers the opportunity to share what can be done with acoustic metamaterials in audio industry and especially in the Loudspeaker Systems Design. The presentation brings back some approaches from the past that can be revisited using today's technologies. Moreover, this talk offers a state-of-the-art of current research in the field of acoustic metamaterials and exhibits the already developed technologies that employ these extremely innovative materials.

Engineering Briefs Session EB06 Saturday, Oct. 19
2:00 pm – 3:00 pm Room 1E11

SPATIAL AUDIO

Chair: **Andreas Franck**, University of Southampton, Southampton, UK

2:00 pm

EB06-1 Study of the Effect of Tikhonov Regularization on the Low Frequency Performance of Cross-Talk Cancellation Systems—*Filippo Maria Fazi, Eric Hamdan, Marcos Simón, Andreas Franck*, University of Southampton, Southampton, Hampshire, UK

Tikhonov regularization is widely applied to the design of cross-talk cancellation (CTC) systems to ensure stability by limiting the loudspeaker array effort. The effect of regularization is significant especially at low frequencies, where the underlying inverse system is generally ill-posed. Previous work by the authors demonstrated that regularization leads to a distortion of the auditory scene, known as stage compression. In this work an analytical formula is derived to calculate, for a given loudspeaker arrangement, the minimum amount of Tikhonov regularization required to ensure that the array effort does not exceed the desired limit. The analytical derivation is presented also of the low frequency limit below which the effect of regularization has a significant effect on the CTC system performance.
[Presentation only; not available in E-Library]
Presented by Marcos Simón

2:15 pm

EB06-2 Subjective Comparison of Binaural Audio Rendering through Headphones and CTC—*Jonathan Phillips, Marcos Simón*, University of Southampton, Southampton, UK

This work compares the subjective performance of a compact cross-talk cancellation (CTC) soundbar prototype compared to headphone reproduction. Preference scores were obtained from listening to different video content with binaural audio. A graphical user interface (GUI) was designed to show the video content, switch between systems and elicit preference scores and ratings of several spatial audio attributes. The study aimed to determine how these two methods compare in terms of relative preference and attribute performance by combining four different video content. Emphasis was placed on the perception of envelopment (or immersion) alongside clarity, depth of field, horizontal width, realism, and spatial naturalness. The results show a preference towards the CTC soundbar over headphones over most of the content/attributes configurations.
[Presentation only; not available in E-Library]

2:30 pm

EB06-3 Tetra-Speaker: Continual Evaluation of the Immersive Experience of a Single-Point Reproduction System— *Parichat Songmuang*, New York University, New York, NY, USA

In the first phase of experimentation, a tetra-speaker had been built as an efficient and compact system for the reproduction of individual sound sources. The reproduction process was based on the relationship of a single sound source, such as an instrument, and an acoustic space. This relationship focused on the radiating behavior of the source. In this paper the tetra-speaker is further evaluated in a case-like study on the immersive experience of the system with additional discussion of how the system may expand its usage to improve this experience within virtual environments. Professionals within the audio field were asked to give an expert's opinion based on defined attributes of spatial impression and realism.
Engineering Brief 553

2:45 pm

EB06-4 Tetrahedral Microphones: An Effective A/B Main System—*Alexander Dobson, Wieslaw Woszczyk*, McGill University, Montreal, QC, Canada

A simple approach to produce an effective stereo main audio recording system using tetrahedral microphones is described, capturing a full array of close and distant sound with a substantial amount of early reflections. This allows for the easy possibility of later surround or 3D reproduction. Furthermore, the implementation of a pragmatic and simple two-microphone set-up can lead to an efficient capture of the stereo soundfield that has many applications. Particular attention was paid to binaural mixing of this microphone system to demonstrate an easy first step into immersive reproduction of the sound.
Engineering Brief 554

Engineering Briefs Session EB07 Saturday, Oct. 19
2:30 pm – 4:15 pm Room 1E10

AUDIO SIGNAL PROCESSING

Chair: **Dave Moffat**, Queen Mary University London, London, UK

2:30 pm

EB07-1 Realistic Procedural Sound Synthesis of Bird Song Using Particle Swarm Optimization—*Jorge Zúñiga, Joshua D. Reiss*, Queen Mary University of London, London, UK

We present a synthesis algorithm for approximating bird song using particle swarm optimization to match real bird recordings. Frequency and amplitude envelope curves are first extracted from a bird recording. Further analysis identifies the presence of even and odd harmonics. A particle swarm algorithm is then used to find cubic Bezier curves which emulate the envelopes. These curves are applied to modulate a sine oscillator and its harmonics. The synthesized syllable can then be repeated to generate the sound. Thirty-six bird sounds have been emulated this way, and a real-time web-based demonstrator is available, with user control of all parameters. Objective evaluation showed that the synthesized bird sounds captured most audio features of the recordings.
Engineering Brief 555

2:45 pm

EB07-2 Multi-Scale Auralization for Multimedia Analytical Feature Interaction—*Nguyen Le Thanh Nguyen,[1] Hyunhwan Lee,[1] Joseph Johnson,[1] Mitsunori Ogihara,[1] Gang Ren,[1] James Beauchamp[2]*
[1] University of Miami, Coral Gables, FL, USA
[2] University of Illinois at Urbana-Champaign, Urbana, IL, USA

Modern human-computer interaction systems use multiple perceptual dimensions to enhance intuition and efficiency of the user by improving their situational awareness. A signal processing and interaction framework is proposed for auralizing signal patterns and augmenting the visualization-focused analysis tasks of social media content analysis and annotations, with the goal of assisting the user in analyzing, retrieving, and organizing relevant information for marketing research. Audio signals are generated from video/audio signal patterns as an auralization framework, for example, using the audio frequency modulation that follows the magnitude contours of video color saturation. The integration of visual and aural presentations will benefit the user interactions by reducing the fatigue level and sharping the users' sensitivity, thereby improving work efficiency, confidence, and satisfaction.
Engineering Brief 556

3:00 pm

EB07-3 Perceptually Motivated Hearing Loss Simulation for Audio Mixing Reference—*Angeliki Mourgela,[1] Trevor Agus,[2] Joshua D. Reiss[1]*
[1] Queen Mary University of London, London, UK
[2] Queens University Belfast, Belfast, UK

This paper proposes the development of a hearing loss simulation for use in audio mix referencing, designed according to psychoacoustic and audiology research findings. The simulation proposed in this paper aims to reproduce four perceptual aspects of hearing loss; threshold elevation, loss of dynamic range, reduced frequency and temporal resolution, while providing an audio input/output functionality.
Engineering Brief 557

3:15 pm

EB07-4 Modeling between Partial Components for Musical Timbre Imitation and Migration—*Angela C. Kihiko,[1] Mitsunori Ogihara,[2] Gang Ren,[2] James Beauchamp[3]*
[1] Spelman College, Atlanta, GA, USA
[2] University of Miami, Coral Gables, FL, USA
[3] University of Illinois at Urbana-Champaign, Urbana, IL, USA

Most musical sounds have strong and regularly distributed spectral components such as harmonic partials. However, the energy distribution patterns between any two such sonic partials, the in-between low-energy signal patterns such as performance articulation or instrument signatures, are also important for characterizing musical sounds. This paper presents a timbre-modeling framework for detecting and modeling the between-partial components for musical timbre analysis and synthesis. This framework focuses on timbre imitation and migration for electronic music instruments, where timbral patterns obtained from acoustical instruments are re-interpreted for electronic instruments and new music interfaces. The proposed framework will help musicians and audio engineers to better explore musical timbre and musical performance expressions for enhancing the naturalness, expressiveness, and creativeness of electronic/computer music systems.
Engineering Brief 558

3:30 pm

EB07-5 Coherence as an Indicator of Distortion for Wide-Band Audio Signals such as M-Noise and Music—*Merlijn van Veen, Roger Schwenke*, Meyer Sound Laboratories, Berkeley, CA, USA

M-Noise is a new scientifically derived test signal whose crest factor as a function of frequency is modeled after real music. M-Noise should be used with a complementary procedure for determining a loudspeaker's maximum linear SPL. The M-Noise Procedure contains criteria for the maximum allowable change in coherence as well as frequency response. When the loudspeaker and microphone are positioned as prescribed by the procedure, reductions in coherence are expected to be caused by distortion. Although higher precision methods for measuring distortion exist, coherence has the advantage that it can be calculated for wide-band signals such as M-Noise as well as music. Examples will demonstrate the perceived audio quality associated with different amounts of distortion-

induced coherence loss.
Engineering Brief 559

3:45 pm

EB07-6 Fast Time Domain Stereo Audio Source Separation Using Fractional Delay Filters—*Oleg Golokolenkao, Gerald Schuller*, Ilmenau University of Technology, Ilmenau, Germany

Our goal is a system for the separation of two speakers during teleconferencing or for hearing aids. To be useful in real time, we want it to work online with as low delay as possible. Proposed approach works in time domain, using attenuation factors and fractional delays between microphone signals to minimize cross-talk, the principle of a fractional delay and sum beamformer. Compared to other approaches this has the advantage that we have lower computational complexity, no system delay and no musical noise like in frequency domain algorithms. We evaluate our approach on convolutive mixtures generated from speech signals taken from the TIMIT data-set using a room impulse response simulator.
Engineering Brief 560

4:00 pm

EB07-7 Line Array Optimization through Innovative Multichannel Filtering—*Paolo Martignon, Mario Di Cola, Letizia Chisari*, Contralto Audio srl, Casoli, Italy

Element dependent filtering offers the possibility to optimize the sound coverage of vertical line arrays: distance dependent frequency response, as well as mid-low frequency beaming and air absorption can be partially compensated. Simulation of array elements contributions to venue acoustics is normally the input data for filters calculation, but some phenomena exist in the real world that are hardly addressed by simulations: for example, the dispersion of transducers responses, as well as the acoustic paths atmospheric conditions, among different array elements. This awareness induced us to develop an algorithm with the aim of being robust against these inaccuracies.
Engineering Brief 561

Paper Session P17 — Saturday, Oct. 19
3:00 pm – 5:30 pm — Room 1E09

PRODUCT DEVELOPMENT

Chair: **Phil Brown**, Dolby Laboratories, San Francisco, CA, USA

3:00 pm

P17-1 Summed Efficiency-Method for Efficient Vented Box Speaker Design—*Niels Elkjær Iversen,*[1,2] *Theis Christensen,*[2] *Anders Bjørnskov,*[2] *Lars Petersen*[2]
[1] Technical University of Denmark, Kogens Lyngby, Denmark
[2] ICEpower, Søborg, Denmark

Loudspeakers are inefficient and conventional design methods do not consider the efficiency in the design process. With the rise of Digital Signal Processing (DSP) the frequency response can be corrected with pre-filtering. Based on a frequency domain analysis of vented box enclosures this paper proposes a new method, the Summed efficiency-method (S?-method), for designing vented box enclosures. The method focuses on mapping the efficiency and SPL output vs. volume and tuning frequency enabling the designer to perform knowledge-based trade-off decisions in the design process. A design example shows how the method can be used to improve the efficiency with over 70% compared to a conventional maximum flat alignment for a compact Public Address (PA) subwoofer application.
Convention Paper 10310

3:30 pm

P17-2 Loudspeaker Port Design for Optimal Performance and Listening Experience—*Andri Bezzola, Allan Devantier, Elisabeth McMullin*, Samsung Research America, Valencia, CA USA

Bass reflex ports produce noise at high sound-pressure levels due to turbulence and vortex shedding. Flared ports can reduce port noise compared to straight ports, but the optimal flare rate in ports has remained an unsolved problem. This work demonstrates that there is in fact an optimal amount of flare, and it proposes a design method based on acoustic Finite Element simulations to efficiently predict the optimal flare rate for given port dimensions. Optimality of the flare rate is confirmed with noise and compression measurements as well as double-blind listening tests. At onset of unwanted port noise, optimally flared ports can be played 1 to 3 dB louder than slightly under-flared or over-flared ports, and 10 to 16 dB louder than straight ports.
Convention Paper 10311

4:00 pm

P17-3 A Method for Three-Dimensional Horn Geometry Optimization—*Christoper Smolen, Jerome Halley*, QSC Audio Products LLC, Costa Mesa, CA, USA

A method for three dimensional (3D) horn geometry optimization is introduced. The method uses 3D Computer Aided Design (CAD) combined with Finite Element Analysis (FEA), the Boundary Element Method (BEM) and scientific programming where: the acoustical properties of horn geometry parametrized in CAD are analyzed using FEA and BEM, and scientific programming is used to manipulate the parametrized geometry and optimize the horn according to specified objective functions. The example of a horn design using this method is presented together with measurements of the resulting geometry.
Convention Paper 10312

4:30 pm

P17-4 A Perceptually-Motivated Headphone Transparency Algorithm—*Josh Lando, Alex Brandmeyer, Phil Brown, Alan Seefeldt, Andy Jaspar*, Dolby Laboratories, San Francisco, CA, USA

Many modern closed-back wireless headphones now support a user-selectable "hear-through" or "transparency" feature to allow the wearer to monitor their environment. These products typically work by passively mixing the signals from external microphones with the primary media being reproduced by the headphone's internal speakers. When there is no media playing back, that approach works reasonably well. However, once media is playing, it tends to mask the passthrough of the external audio and the wearer can no longer hear the outside world. Here we describe a perceptually motivated algorithm for improving audibility of the external microphone signals without compromising the media playback experience. Subjective test results of this algorithm as implemented in a consumer headphone product are presented.
Convention Paper 10313

5:00 pm

P17-5 Temporal Envelope-Based Psychoacoustic Modelling for Evaluating Non-Waveform Preserving Audio Codecs—*Steven van de Par*,[1,2] Sascha Disch,[3,4] *Andreas Niedermeier*,[3] *Elena Burdiel Pérez*,[3] *Bernd Edler*[4]
[1] University of Oldenburg, Oldenburg, Germany
[2] Fraunhofer HSA, Oldenburg, Germany
[3] Fraunhofer IIS, Erlangen, Germany
[4] Friedrich Alexander University, Erlangen-Nürnberg, Germany

Masking models that evaluate the audibility of error signals have a limited validity for assessing perceptual quality of parametric codecs. We propose a model that transforms the audio signal into an Internal Representation (IR) consisting of temporal-envelope modulation patterns. Subsequently, the IR of original and encoded signals are compared between both signals. Even though the audio signals compared may be uncorrelated, leading to a large error signal, they may exhibit a very similar IR and hence are predicted to sound very similar. Additional post-processing stages modeling higher-level auditory perceptual phenomena such as Comodulation Masking Release are included. Predictions are compared against subjective quality assessment results obtained with encoding methods ranging from parametric processing methods up to classic waveform preserving codecs.
Convention Paper 10314

Networked Audio
NA05 - NETWORK MEDIA OPEN SPECIFICATIONS (NMOS)—HOW THEY RELATE TO AUDIO NETWORKING
Saturday, October 19, 2:45 pm – 4:15 pm, Room 1E21

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

With AES67, a proven and well-accepted standard for real-time transport of studio quality audio signals has found its way into nearly any networked audio product in the market. But solving interoperability issues for synchronization and transport formats does not automatically result in fully working systems when they combine devices from various brands. From an application perspective, system management and device control are also essential to system becoming "usable."

While device control, particularly when it comes to specific features of individual devices, may remain device-specific and maybe even proprietary, it is highly desirable to include common functionality that would allow basic system management, including discovery of devices and their resources on a network and basic connection management as well. AES67 provides hints on these "upper layer" functionalities but doesn't mandate any particular scheme.

The Advanced Media Workflow Association (AMWA), an industry-wide organization defining business needs for interoperability of networked media systems, has identified this gap and a few years ago initiated the NMOS (Networked Media Open Specifications) project.

Meanwhile, a number of specifications have been developed that are currently adopted by a large number of manufacturers. Chief among them in addressing the basic needs beyond AES67 are IS-04 (discovery and registration) and IS-05 (connection management). This session provides an overview on these and other important NMOS specifications currently in the works and explains how they fit with AES67 to result in enhanced system-level interoperability.

Paper Session P18 **Saturday, Oct. 19**
3:00 pm – 4:30 pm **South Concourse A**

POSTERS: PERCEPTION

3:00 pm

P18-1 Comparison of Human and Machine Recognition of Electric Guitar Types—*Renato Profeta, Gerald Schuller*, Ilmenau University of Technology, Ilmenau, Germany

The classification of musical instruments for instruments of the same type is a challenging task not only to experienced musicians but also in music information retrieval. The goal of this paper is to understand how guitar players with different experience levels perform in distinguishing audio recordings of single guitar notes from two iconic guitar models and to use this knowledge as a baseline to evaluate the performance of machine learning algorithms performing a similar task. For this purpose we conducted a blind listening test with 236 participants in which they listened to 4 single notes from 4 different guitars and had to classify them as a Fender Stratocaster or an Epiphone Les Paul. We found out that only 44% of the participants could correctly classify all 4 guitar notes. We also performed machine learning experiments using k-Nearest Neighbours (kNN) and Support Vector Machines (SVM) algorithms applied to a classification problem with 1292 notes from different Stratocaster and Les Paul guitars. The SVM algorithm had an accuracy of 93.9%, correctly predicting 139 audio samples from the 148 present in the testing set.
Convention Paper 10315

3:00 pm

P18-2 Preference for Harmonic Intervals Based on Overtone Content of Complex Tones—*Benjamin Fox, Wesley Bulla*, Belmont University, Nashville, TN, USA

This study investigated whether or not overtone structure generated preferential differences for harmonic intervals. The purpose of this study was to determine if the structure of a complex tone affects the perception of consonance in harmonic intervals. Prior studies suggest harmonicity as the basis for so-called "consonance" while others suggest exact ratios are not necessary. This test examined listener responses across three tonal "types" through a randomized double-blind trinomial forced-choice format. Stimuli types used full, odd, and even overtone series at three relative-magnitude loudness levels. Results revealed no effect of loudness and a generalized but highly variable trend for the even overtone series. However, some subjects exhibited a very strong preference for certain overtone combinations, while others demonstrated no preference.
Convention Paper 10316

3:00 pm

P18-3 Just Noticeable Difference for Dynamic Range Compression via "Limiting" of a Stereophonic Mix—*Christopher Hickman, Wesley Bulla*, Belmont University, Nashville, TN, USA

This study focused on the ability of listeners to discern the presence of dynamic range compression (DRC) when applied to a stereo recording. Past studies have primarily focused on listener preferences for stereophonic master recordings with varying levels of DRC. A modified two-down one-up adaptive test presented subjects with an increasingly "limited" stereophonic mix to determine the 70.7% response threshold. Results of this study suggest that DRC settings considered "normal" in recorded music production may be imperceptible when playback levels are loudness-matched. Outcomes of this experiment indicate the use of so-called "limiting" for commercial purposes, such as signal chain control, may have no influence on perceived quality; whereas, uses for perceived aesthetic advantages should be reconsidered.
Convention Paper 10317

3:00 pm

P18-4 Discrimination of High-Resolution Audio without Music—*Yuki Fukuda, Shunsuke Ishimitsu*, Hiroshima City University, Hiroshima, Japan

Nowadays, High-Resolution (Hi-Res) audio format, which has higher sampling frequency (Fs) and quantization bit number than the Compact disc (CD) format, is becoming extremely popular. Several studies have been conducted to clarify whether these two formats can be distinguished. However, most of the studies were conducted by only using music sources to reach a conclusion. In this paper we will try to bring out the problems due to the primary use of music sources for experimental purposes. We will also answer the question related to discrimination between hi-Res and CD formats using sources other than music, such as noise.
Convention Paper 10318

3:00 pm

P18-5 Subjective Evaluation of Multichannel Audio and Stereo on Cell Phones—*Fesal Toosy, Muhammad Sarwar Ehsan*, University of Central Punjab, Lahore, Pakistan

With the increasing trend of using smart phones and other handheld electronic devices for accessing the internet, playback of audio in multichannel format would eventually gain popularity on such devices. Given the limited options for audio output on handheld electronic devices, it is important to know if multichannel audio offers an improvement in audio quality over other existing formats. This paper shows a subjective assessment test of multichannel audio versus stereo while played on a mobile phone using headphones. The results show that multichannel audio improves on perceived audio quality as compared to stereo.
Convention Paper 10319

Audio for cinema
AC04 - AMBISONICS IN CINEMA
Saturday, October 19, 3:00 pm – 4:00 pm, Room 1E08

Presenter: **John Escobar**, Berklee College of Music, Boston, MA, USA

Berklee Professor and cinema audio maven John Escobar explores the uses of Ambisonics in cinema post-production to enhance existing audio as well as audio captured by soundfield microphones. Ambisonics can be used to "spatialize" non-soundfield recordings for better localization and potential further use in interactive media. Mr. Escobar will also demonstrate the use of Ambisonics in film score production. Audio examples using the technology will be played.

Student Events & Career Development
SC23 - ESSENTIAL ELEMENTS OF EAR TRAINING FOR CONTEMPORARY AUDIO ENGINEERS
Saturday, October 19, 3:00 pm – 4:30 pm, Room 1E07

Presenters: **Jason Corey**, University of Michigan, Ann Arbor, MI, USA
Kazuhiko Kawahara, Kyushu University, Fukuoka, Japan
Sungyoung Kim, Rochester Institute of Technology, Rochester, NY, USA
Doyuen Ko, Belmont University, Nashville, TN, USA
Sean Olive, Harman International, Northridge, CA, USA
Timothy Ryan, Webster University

Since 2011 workshop panelists have enthusiastically shared their experiences for efficient and effective methods of critical listening and technical ear training (AES 131st, 132nd, 141st, & 143rd Conventions). We are experiencing new technical ear training tools and observing how fast those tools are evolving for recent audio technologies. Do we need a paradigm shift in technical ear training and critical listening for future listeners? What are the essential elements for the training of today's junior audio engineers and students? In this workshop, panelists will consider the aforementioned questions. In addition, additional topics will be discussed, including (but not limited to): (1) influence of sound stimuli and associated influence on training performance; (2) verbal elicitation of spectral modification; (3) beyond spectral ear training: spatial, dynamic, and more; (4) a case study: longitudinal analysis of last decade's ear training data; (5) features required for future auditory training; and (6) gamification. The workshop will demonstrate listening and training examples. While the workshop aims to provide the attendees with a chance to experience theoretical and empirical matters of ear training programs around the world, it also aims to consider the importance of "listening" in the current video-oriented society.

Engineering Briefs Session EB8 Saturday, Oct. 19
3:30 pm – 4:30 pm Room 1E11

APPLICATIONS IN AUDIO

Chair: **Sunil G. Bharitkar**, HP Labs., Inc., San Francisco, CA, USA

3:30 pm

EB08-1 Vibrary: A Consumer-Trainable Music Tagging Utility
—*Scott Hawley,[1] Jason Bagley,[2] Brett Porter,[2] Daisey Traynham[2]*
[1] Belmont University, Nashville, TN, USA
[2] Art+Logic, Pasadena, CA, USA

We present the engineering underlying a consumer application to help music industry professionals find audio clips and samples of personal interest within their large audio libraries typically consisting of heterogeneously-labeled clips supplied by various vendors. We enable users to train an indexing system using their own custom tags (e.g., instruments, genres, moods), by means of convolutional neural networks operating on spectrograms. Since the intended users are not data scientists and may not possess the required computational resources (i.e., Graphics Processing Units, GPUs), our primary contributions consist of (i) designing an intuitive user experience for a local client application to help users create representative spectrogram datasets, and (ii) "seamless" integration with a cloud-based GPU server for efficient neural network training.
Engineering Brief 562

3:45 pm

EB08-2 Casualty Accessible and Enhanced (A&E) Audio: Trialling Object-Based Accessible TV Audio—*Lauren Ward,[1,2] Matthew Paradis,[3] Ben Shirley,[1] Laura Russon,[4] Robin Moore,[2] Rhys Davies[4]*
[1] University of Salford, Salford, UK
[2] BBC R&D, North Lab, Salford, UK
[3] BBC R&D, South Lab, London, UK
[4] BBC Studios, Cardiff, Wales, UK

Casualty Accessible and Enhanced (A&E) Audio is the first public trial of accessible audio technology using a narrative importance approach. This trial allows viewers to personalize the audio of an episode of the BBC's "Casualty" drama series based on their hearing needs. Using a simple interface the audio can be varied between the broadcast mix and an accessible mix containing narratively important non-speech sounds, enhanced dialogue, and attenuated background sounds. This paper describes the trial's development, implementation, and it's evaluation by normal

and hard of hearing listeners (n=5209 on 20/8/2019). 299 participants also completed a survey, rating the technology 3.6/5 stars. 73% reported the technology made the content more enjoyable or easier to understand.
Engineering Brief 563

4:00 pm

EB08-3 Generative Modeling of Metadata for Machine Learning Based Audio Content Classification—*Sunil G. Bharitkar*, HP Labs., Inc., San Francisco, CA, USA

Automatic content classification technique is an essential tool in multimedia applications. Present research for audio-based classifiers look at short- and long-term analysis of signals, using both temporal and spectral features. In this paper we present a neural network to classify between the movie (cinematic, TV shows), music, and voice using metadata contained in either the audio/video stream. Towards this end, statistical models of the various metadata are created since a large metadata dataset is not available. Subsequently, synthetic metadata are generated from these statistical models, and the synthetic metadata is input to the ML classifier as feature vectors. The resulting classifier is then able to classify real-world content (e.g., YouTube) with an accuracy ˜ 90% with very low latency (viz., ˜ on an average 7 ms) based on real-world metadata.
Engineering Brief 564

4:15 pm

EB08-4 Individual Headphone Equalization at the Eardrum with New Apps for Computers and Cellphones—*David Griesinger*, David Griesinger Acoustics, Cambridge, MA, USA

Ear canal resonances that concentrate energy on the eardrum are highly individual, and headphones alter or eliminate them. The result is inaccurate timbre and in-head localization. We have developed computer apps that use an equal loudness test to match the sound spectrum at the eardrum from a pair of headphones to the spectrum at the eardrums from a frontal loudspeaker. The result is precise timbre and frontal localization. The improvement in sound is startling. In this presentation we will demonstrate the process and the easy to use software that is now available for VST, AAX, Windows, MAC, Android and IOS cellphones.
[Presentation only; not available in E-Library]

Sound Reinforcement
SR10 - SEVEN STEPS TO A SUCCESSFUL SOUND SYSTEM DESIGN
Saturday, October 19, 3:30 pm – 5:00 pm, Room 1E12

Presenter: **Josh Loar**, Michigan Technological University, Houghton, MI, USA; The Producers Group, Burbank, CA, USA

Sound systems are getting more complex every day. In the era of digital audio, network-enabled devices, and complex interdependent show control, developing a sound system design can feel like a daunting task. Josh Loar (author of The Sound System Design Primer available from Focal Press/Routledge) presents a systematic, seven step process for designing any sound system that demystifies the process, and allows the prospective designer to make logical choices, and maximize the elegance and efficiency of the technical solutions chosen. Loar introduces a question and answer process for the designer that allows them to parse the system needs in any gear category—identifying what specs are most relevant to the work of the designer, and what questions a designer must ask and answer in the process. Additionally, Loar discusses the differing needs of various classes of system, from live theater and concert systems, to studio and production systems, to theme parks, casinos, and other installed systems.