AES 143RD CONVENTION PROGRAM

OCTOBER 18–21, 2017

JAVITS CONVENTION CENTER, NY, USA

The Winner of the 143rd AES Convention Best Peer-Reviewed Paper Award is:

A Statistical Model that Predicts Listeners' Preference Ratings of In-Ear Headphones: Part 1—Listening Test Results and Acoustic Measurements—*Sean Olive, Todd Welti, Omid Khonsaripour,* Harman International, Northridge, CA, USA

Convention Paper 9840

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

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

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

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

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

(a) The paper was accepted for presentation at the AES 143rd 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 143rd AES Convention Student Paper Award is:

Modeling the Effects of Rooms on Frequency Modulated Tones—*Sarah R. Smith, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Convention Paper 9885

To be presented on Friday, Oct. 20, in Session 15—Posters: Applications in Audio

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Session P1 **Wednesday, Oct. 18**
9:00 am – 11:00 am **Room 1E11**

SIGNAL PROCESSING

Chair: **Bozena Kostek**, Gdansk University of Technology, Gdansk, Poland

9:00 am

P1-1 Generation and Evaluation of Isolated Audio Coding Artifacts—*Sascha Dick, Nadja Schinkel-Bielefeld, Sascha Disch*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Many existing perceptual audio codec standards define only the bit stream syntax and associated decoder algorithms, but leave many degrees of freedom to the encoder design. For a systematic optimization of encoder parameters as well as for education and training of experienced test listeners, it is instrumental to provoke and subsequently assess individual coding artifact types in an isolated fashion with controllable strength. The approach presented in this paper consists of a pre-selection of suitable test audio content in combination with forcing a specially modified encoder into non-common operation modes to willingly generate controlled coding artifacts. In conclusion, subjective listening tests were conducted to assess the subjective quality for different parameters and test content.
Convention Paper 9809

9:30

P1-2 Enhancement of Voice Intelligibility for Mobile Speech Communication in Noisy Environments—*Kihyun Choo,[1] Anton Porov,[2] Maria Koutsogiannaki,[3] Holly Francois,[3] Jonghoon Jeong,[1] Hosang Sung,[1] Eunmi Oh[1]*
[1]Samsung Electronics Co., Ltd., Seoul, Korea
[2]Samsung R&D Institute Russia, Moscow, Russia
[3]Samsung R&D Institute UK, London, UK

One of the biggest challenges still encounter with speech communication via a mobile phone is that it is sometimes

very difficult to understand what is said when listening in a noisy place. In this paper a novel approach based on two models is introduced to increase speech intelligibility for a listener surrounded by environmental noise. One is to perceptually optimize the speech when considering simultaneous background noise, the other is to modify the speech towards a more intelligible, naturally elicited speaking style. The two models are combined to provide more understandable speech even in a loud noisy environment, even in the case where we are unable to increase the speech volume. The improvements in perceptual quality and intelligibility are shown by Perceptual Objective Listening Quality Assessment and Listening Effort Mean Opinion Score evaluation.
Convention Paper 9810

10:00

P1-3 Application of Spectral-Domain Matching and Pseudo Non-Linear Convolution to Down-Sample-Rate Conversion (DSRC)—*Malcolm O. J. Hawksford*, University of Essex, Colchester, Essex, UK

A method of down-sample-rate conversion is discussed that exploits processes of spectral-domain matching and pseudo non-linear convolution applied to discrete data frames as an alternative to conventional convolutional filter and sub-sampling techniques. Spectral-domain matching yields a complex sample sequence that can subsequently be converted into a real sequence using the Discrete Hilbert Transform. The method is shown to result in substantially reduced time dispersion compared to the standard convolutional approach and circumvents filter symmetry selection such as linear phase or minimum phase. The formal analytic process is presented and validated through simulation then adapted to digital-audio sample-rate conversion by using a multi-frame overlap and add process. It has been tested in both LPCM-to-LPCM and DSD-to-LPCM applications where the latter can be simplified using a look-up code table.
Convention Paper 9811

10:30

P1-4 Detection of Piano Pedaling Techniques on the Sustain Pedal—*Beici Liang, György Fazekas, Mark Sandler*, Queen Mary University of London, London, UK

Automatic detection of piano pedaling techniques is challenging as it is comprised of subtle nuances of piano timbres. In this paper we address this problem on single notes using decision-tree-based support vector machines. Features are extracted from harmonics and residuals based on physical acoustics considerations and signal observations. We consider four distinct pedaling techniques on the sustain pedal (anticipatory full, anticipatory half, legato full, and legato half pedaling) and create a new isolated-note dataset consisting of different pitches and velocities for each pedaling technique plus notes played without pedal. Experiment shows the effectiveness of the designed features and the learned classifiers for discriminating pedaling techniques from the cross-validation trails.
Convention Paper 9812

Session P2 **Wednesday, Oct. 18**
9:00 am – 10:30 am **Room 1E12**

AUDIO EDUCATION

Chair: **Alex Ruthmann**, New York University, New York, NY, USA

9:00

P2-1 Audio Education: Audio Recording Production Students Report Skills Learned or Focused on in Their Programs—*Doug Bielmeier*, Purdue School of Engineering and Technology, IUPUI, Indianapolis, IN, USA

Previous research polled employers, new hires, and educators in the audio industry to identify what skills were most important, what skills new hires had, and what skills educators focused on in Audio Recording Production (ARP) Programs. The Skills Students Learned (SSL) Survey used in this study, polled 40 students from the U.S. and aboard to identify skills learned at ARP programs. Students reported their skill level before and after attending a formal ARP program via an online mixed methods survey instrument. In the quantitative section, students reported an improvement in all skill levels upon completing their ARP training. In the qualitative section, students reported communication skills and in-depth technical skills missing from their programs and personal skill sets. This study recommends infusion of these skills into existing ARP curriculum.
Convention Paper 9814

9:30

P2-2 Audio Archive Preservation Challenges and Pedagogical Opportunities: School of Music RePlayed—*Samantha Bennett*, Australian National University, Canberra, ACT, Australia

This paper considers the various challenges, implications and pedagogical opportunities presented via a small-scale audio archiving project: School of Music RePlayed. Housed in the Australian National University's School of Music, this historical archive of more than 1200 recital and concert tape recordings features multiple recordings of historical significance, yet presents with a number of issues pertaining to storage and tape deterioration. This paper first considers the challenges presented in the digitization of such an archive before focusing on the pedagogical opportunities afforded by such a unique project. Developed and run in conjunction with the National Film and Sound Archive of Australia, this unique project addresses both technological and pedagogical matters of preservation, heritage and digitization.
Convention Paper 9815

10:00

P2-3 The Education of the Next Generation of Pro-Audio Professionals—*Curig Huws*, University of South Wales, Cardiff, UK

Since the late 1990s and early 2000s, the changing nature of the music industry has led to the demise of recording studios, which have decreased dramatically in number. This decline has led to a corresponding disappearance of the "teaboy" route, the traditional route whereby engineers, producers, and mixers (EPM) learned their craft. In the training vacuum that the demise of recording studios creates, how do EPM professionals now learn the skills and knowledge necessary to succeed in the music industry? Through primary research and indepth interviews with leading EPM professionals and online education providers, this paper assesses the skills needed to become a successful EPM and explores whether the internet can ever replace the traditional teaboy route in educating the next generation of professionals. It concludes that there are currently significant limitations to internet learning of EPM skills, some of which might be overcome by new

technological developments such as virtual reality.
Convention Paper 9816

Broadcast/Media Streaming 1 **Wednesday, October 18**
9:00 am – 10:30 am **Room 1E08**

DESIGNING AND CONSTRUCTING A RADIO PERFORMANCE SPACE

Moderator: **David Prentice**, Dale ProAudio

Panelists: *Sam Cappas*, CBS Radio
Joshua Morris, Walters, Storyk Design Group
Steve Shultis, WNYC-NY Public Radio
Jeff Smith, iHeart Media

In baseball, a slash line is a player's statistics (batting average / on base percentage / slugging percentage) showing both performance and power at the plate. The slash line for a radio performance space would be live sound/video production/multi-use and occasionally radio showing the production versatility today's radio station requires. Increasingly, stations are adding just such spaces to reward their audience with exclusive performances, create new programming, add video content for their on-line and streaming channels, and increase the interaction with their audience.

Balancing the different requirements for various uses requires rethinking traditional designs for single-use spaces. The lighting and sound needs to be appropriate for both video and audio recording; control rooms will be required to service several different venues and functions; acoustics and aesthetics need to be appropriate for conversation, acoustic, or electric music; and everybody needs audio, video, and intercom monitoring. There's no set model for these rooms but our panelists are building the facilities and writing the new rules.

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

Sound Reinforcement 1 **Wednesday, October 18**
9:00 am – 10:30 am **Room 1E09**

SOUND SYSTEM OPTIMIZATION WITH BOB MCCARTHY

Presenter: **Bob McCarthy**

Bob will lead a discussion of methods of system optimization using speaker aiming, equalization, timing and level matching.

Spatial Audio 1 **Wednesday, October 18**
9:00 am – 10:00 am **Room 1E13**

CRASH COURSE IN 3D AUDIO

Presenter: **Nuno Fonseca**, ESTG/Polytechnic Institute of Leiria, Leiria, Portugal; Sound Particles, Leiria, Portugal

A little confused with all the new 3D formats out there? Although most 3D audio concepts already exist for decades, the interest in 3D audio has increased in recent years, with the new immersive formats for cinema or the rebirth of Virtual Reality (VR). This tutorial will present the most common 3D audio concepts, formats, and technologies allowing you to finally understand buzzwords like Ambisonics/HOA, Binaural, HRTF/HRIR, channel-based audio, object-based audio, Dolby Atmos, Auro 3D, among others.

Tutorial/Workshop 1 **Wednesday, October 18**
9:00 am – 10:30 am **Room 1E10**

AUDIO ENGINEERING WITH HEARING LOSS— A PRACTICAL SYMPOSIUM

Chair: **Jon Boley**, GN Advanced Science, Chicago, IL, USA

Panelists: *Richard Einhorn*, Einhorn Consulting, LLC, New York, NY, USA; Richard Einhorn Productions, Inc.
Larry Revit, Revitronix, Braintree, VT, USA
Michael Santucci, Sensaphonics, Chicago, IL USA

We have assembled a panel of experts—one to present an audiological perspective of hearing loss and others to focus on the more practical issues of working in the field of audio engineering with hearing loss and/or tinnitus (e.g., listening strategies, supplemental technologies, etc.).

This session is presented in association with the AES Technical Committee on Hearing and Hearing Loss Prevention

Spatial Audio 19 **Wednesday, October 18**
9:30 am – 10:30 am **Room 1E06**

SOUND DESIGN IN 3D FOR DANCE MUSIC

Presenter: **Lasse Nipkow**

Producers of Dance Music pursue the target to put nightclub audiences into trance. 3D audio includes a high potential to increase sound impressions significantly and engage the emotions, as is the case in film sound.

3D audio allows to create spectacular spatiality and sound sources around the listener. There are however some challenges: Clubbers on the dance floor are not oriented in the same way as listener of classical music in the concert hall because they are dancing. PA systems of music clubs are nearly exclusively designed for mono and stereo reproduction. Therefore, the impact by changing to a 3D audio system is significant.

During the presentation, the possibilities of musical designin consideration of psychoacoustic phenomena will be described and demonstrated with example recordings and videos.

Special Event
SE1: DIVERSITY TOWN HALL
Wednesday, October 18, 9:30 am – 10:45 am
Room 1E15/16

Moderator: **Leslie Gaston-Bird**, AES Vice President (Western Region) and D&I committee chair

Presenters: *Alex Case*, AES President
Leslie Ann Jones, Director of Music Recording and Scoring, Skywalker Sound
Karrie Keyes, Executive Director SoundGirls.org
Bob Moses, AES Executive Director
Piper Payne, Owner, Neato Mastering and D&I committee co-chair
Terri Winston, Founder & Executive Director, Women's Audio Mission

This Town Hall is meant to introduce the recently formed AES Diversity and Inclusion Committee and give AES members a chance to have a meaningful discussion about the committee's purpose. The committee has stated a number of goals intended to strengthen the AES as a whole:

• To work towards a membership of the AES that will best reflect the demographics of working audio professionals.

• To increase AES membership and broader participation in the audio industry by helping the Society become more diverse and inclusive.

These efforts will include women and other underrepresented groups, as well as students and young audio professionals working in newer music genres and/or audio fields.

Spatial Audio 2 — **Wednesday, October 18**
9:45 am – 11:15 am — **Room 1E14**

RIDING THE FADERS: HOW TO STAY AHEAD OF THE CURVE IN IMMERSIVE AUDIO

Presenters: **Brennan Anderson**, Pyramind
Steve Horelick, Steve Horelick Music
Paul Special, Founder SAS
Richard Warp, Intonic / Sound Designer, Pollen Music Group

VR, AR, and XR have highlighted the importance of sound for "presence" in all content types from live TV to film to games. This represents a unique opportunity for audio professionals. But making the transition to the new world of immersive realities is no easy task. Here you will learn from industry professionals who are creating new 3D audio workflows and growing professional networks while remaining firmly grounded in their craft. The Manhattan Producers' Alliance is a New York/San Francisco-based membership organization comprised of engineers, composers and producers. Our focus is on nurturing personal creativity within the art and craft of music making.

Game Audio & VR 1 — **Wednesday, October 18**
10:15 am – 11:15 am — **Room 1E13**

WILL VR BE A GAME CHANGER? CHALLENGES & OPPORTUNITIES IN 6DOF AUDIO

Chair: **Brooklyn Earick**, G'Audio Lab, Los Angeles, CA, USA

Panelists: *Andrew Grathwohl*, Director of Media Technology, Littlstar
Jean-Marc Jot, Distinguished Fellow, Magic Leap
Adam Levenson, VP of Business Development, Krotos

6DoF(degrees of freedom) type of VR content allows users the ultimate freedom to explore the virtual world, but it is a total pain in the neck for content creators including audio engineers. Compared to 360 videos, 6DoF experiences just require so much more to consider — room simulation, all possible interactive scenarios, and even spatialization that is specially crafted for full VR. This session will explore the major challenges audio engineers are facing in 6DoF content creation and how they are being addressed. Panels will share technologies they are embracing and where these applications are heading. The session intends to provide insights for spatial audio design in both gaming VR and non-gaming VR.

Project Studio Expo 01 — **Wednesday, October 18**
10:25 am – 11:05 am — **Stage 2**

A CONVERSATION WITH JOE CHICCARELLI

Presenter: **Joe Chiccarelli**, Producer, mixer, engineer, Boston, MA, USA

[Abstract not available]

Broadcast Audio Expo 01 — **Wednesday, October 18**
10:30 am – 11:15 am — **Stage 1**

TOTAL PRODUCTION NETWORKING

Presenters: **Mark Brunke,** Optocore
Maciek Janiszewski, Optocore
Vinnie Macri, Clear-Com

Recently very hot topic with modern broadcast workflows is the signal format and transport. Is SDI format going to survive or IP will take over completely? What about audio and intercom? The session will reveal the best practice in the production networking using fiber cabling and fiber networks. Industry experts will discuss the modern approach and the way to combine high-bandwidth video with multiple audio, comms and data.

Software@AES 01 — **Wednesday, October 18**
10:30 am – 10:50 am — **Stage 3**

BEST SERVICE

Session P3 — **Wednesday, Oct. 18**
10:45 am – 12:15 pm — **Foyer**

POSTERS: PERCEPTION

10:45 am

P3-1 Study on Objective Evaluation Technique for Small Differences of Sound Quality—*Yuki Fukuda,*[1] *Kenta Ueyama,*[1] *Shunsuke Ishimitsu*,[1] *Ryoji Higashi*,[2] *Seiji Yumoto,*[2] *Takashi Numanou*[2]
[1]Hiroshima City University, Hiroshima-shi, Japan
[2]Memory-Tech Corporation, Tokyo, Japan

In recent years, some results on different auditory impressions from differences of materials and media have been discussed. To check the causes of these differences, we analyzed the differences in the sound pressure levels and interaural time difference [1] between three different Compact Discs by using wavelet analysis. The results of these analyses detected objective differences in sound despite different materials having the same data, and the new Compact Disc called the "Ultimate Hi Quality Compact Disc" made of photopolymer, where a special alloy has been employed as a reflection film, reproduces more of the master sound than the conventional Compact Disc. We show the method for analyzing sound and evaluate these differences and consider their application on various sound quality evaluations.
Convention Paper 9817

10:45 am

P3-2 Alternative Weighting Filters for Multitrack Program Loudness Measurement—*Steven Fenton, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

The ITU-Recommendation BS.1770 is now established throughout most of the broadcast industry. Program loudness measurement is undertaken through the summation of K-weighted energy and this summation typically involves material that is broadband in nature. We undertook listening tests to investigate the performance of the K-weighting filter in relation to perceived loudness of narrower band stimuli, namely octave-band pink noise

and individual stems of a multitrack session. We propose two alternative filters based on the discrepancies found and evaluate their performance using different measurement window sizes. The new filters yield better performance accuracy for both pink noise stimuli and certain types of multitrack stem. Finally, we propose an informed set of parameters that may improve loudness prediction in auto mixing systems.
Convention Paper 9818

10:45 am

P3-3 An Audio Loudness Compression and Compensation Method for Miniature Loudspeaker Playback—*Ziran Jiang,*[1,2] *Jinqiu Sang,*[1,2] *Jie Wang,*[3] *Chengshi Zheng,*[1,2] *Fangjie Zhang,*[1,2] *Xiaodong Li*[1,2]
[1]Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, Beijing, China
[2]University of Chinese Academy of Sciences, Beijing, China
[3]Guangzhou University Guangzhou, China

Audio playback through miniature loudspeakers is bounded by the loudspeaker's limited dynamic range. How to compress the audio and simultaneously preserve the original artistic effect is worthy of study. Traditional peak-based and RMS-based dynamic range compression (DRC) methods do not consider the audio loudness characteristic that may influence the perceptual artistic effect. This paper proposes a novel compression and compensation method based on Zwicker's loudness model and equal-loudness contours. The proposed method aims to provide a high-quality audio playback by mapping the audio's loudness to a smaller range, while preserving the perceived spectral balance of the original audio. Subjective listening tests are performed to demonstrate the benefits of the proposed method.
Convention Paper 9819

10:45 am

P3-4 Assessing the Authenticity of the KEMAR Mouth Simulator as a Repeatable Speech Source—*Thomas McKenzie, Damian Murphy, Gavin Kearney*, University of York, York, UK

In audio engineering research, repeatability is paramount. Speech is a great stimulus to use when evaluating audio systems as it is a real world sound highly familiar to the human auditory system. With a view to the comparison of real and virtual sound fields, a repeatable speech source is therefore highly advantageous. This paper presents both an objective and subjective evaluation of the G.R.A.S. Knowles Electronic Manikin for Acoustic Research mouth simulator as a repeatable speech source, assessing its accuracy and perceptual authenticity.
Convention Paper 9820

10:45 am

P3-5 Pilot Experiment on Verbal Attributes Classification of Orchestral Timbres—*Ivan Simurra, Sr., Marcelo Queiroz*, University of São Paulo, São Paulo, Brazil

This paper presents a listening test of an ongoing research related to timbre perception, using a set of 33 orchestral music excerpts that are subjectively rated using quantitative scales based on 13 pairs of opposing verbal attributes. The aim of the experiment is to identify significant verbal descriptions potentially associated with timbre aspects of musical excerpts that explore technical aspects of contemporary music such as extended techniques and non-standard music orchestration. Preliminary results suggest that these scales are able to describe timbral qualities in a way that is consistent among different listeners.
Convention Paper 9821

10:45 am

P3-6 Precedence Effect Using Simultaneous High and Low-Passed Stimuli—*Austin Arnold, Wesley Bulla*, Belmont University, Nashville, TN, USA

This study was an exploration of interaural suppression in the context of two simultaneous auditory precedence scenarios. The experiment investigated the nature of aural precedence by presenting subjects with two sets of stimuli simultaneously. Combinations of lead-lag signals employed a series of low- and high-passed noise bursts presented as either leading on the same side or on opposite sides of the listener. Subjects were asked to localize each noise burst. Findings suggest that when signals originated at opposite loudspeakers, performance for both signals was degraded. However, degradation appeared to be dependent upon the frequency span between the two stimuli. This novel study of the precedence effect more broadly addresses the manner in which the brain resolves bilaterally conflicting information and provides evidence that binaural suppression is not band limited, is possibly object oriented, and may change with the content of the objects of interest.
Convention Paper 9822

Archiving/Restoration 1 — Wednesday, October 18
10:45 am – 12:15 pm — Room 1E12

GET INTO THE GROOVE: A PANEL DISCUSSION ON GROOVED MEDIA

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

Moderator: **Rebecca Y. Feynberg**, New York University, New York, NY, USA

Panelists: *Peter Alyes*
Dave Cawley
Melissa Widzinski

A panel on grooved media. Grooved media comes in may forms, cylinders, records, Dictaphone recordings and other bizarre shapes and sizes. If it has grooves then this panel of experts should be able to help. There will be four short presentations featuring George Blood on The Great 78rpm Project; Peter Alyea on IRENE update; Melissa Widzinski on Field Cylinder Digitization at Indiana University, and Dave Cawley on why All EQs are equal, but some EQs are more equal than others. There will be a 30 minute question and answer session where you can field even the most awkward questions. Our panel of experts can advise on everything from stylus size, speeds, EQ, turntables, cylinder players, optical reconstruction and almost everything else. Beginners and old hands are all welcome.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Audio for Cinema 1 — Wednesday, October 18
10:45 am – 12:15 pm — Room 1E07

3D AUDIO TOOLS FOR IMMERSIVE AUDIO

Presenters: **Rafael Duyos**, DSpatial
Nuno Fonseca, ESTG/Polytechnic Institute of Leiria, Leiria, Portugal; Sound Particles, Leiria, Portugal
Gael Martinet, Flux:: sound and picture development, Orleans, France
Connor Sexton, Avid, El Segundo, CA, USA

Immersive audio is here to stay, but should we continue to use the same ordinary mono/stereo tools? This tutorial will present four different approaches to 3D audio, covering immersive sound design and mixing, which will help you to take full advantage of the new immersive cinema formats.

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

Broadcast/Media Streaming 2 Wednesday, October 18
10:45 am – 12:15 pm Room 1E08

AUDIO CABLE PROBLEM SOLVING

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

Panelists: *David Carroll,* DCE Electronics
Ed Grogan
Christopher Payne, Advanced Systems Group LLC
Brad Pope, Belden Corporation
Tom Sahara, Turner Sports

This will be a panel discussion on common cable-related problems and solutions. This will feature prominent designers, installers, and end-users in the audio arena in sound reinforcement, recording, film sound and broadcast. Many of these problems, and their solutions, can be cross-pollinated between these industries and the hope is that the audience will add much to the discussion of problems encountered and possible solutions to those problems. Emerging technologies, such as Audio over IP, will be included on the menu.

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

Product Development 1 Wednesday, October 18
10:45 am – 12:15 pm Room 1E10

HEADPHONES, HEADSETS & EARPHONES: ELECTROACOUSTIC DESIGN & VERIFICATION

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

This tutorial reviews basic the electroacoustic concepts of gain, sensitivity, sound field correction, linear and non-linear response, and test signals for ear-worn devices. The insertion gain concept is explained and free and diffuse field target responses are shown. Equivalent volume and acoustic impedance are defined. Ear simulators and test manikins appropriate for circum-aural, supra-aural, intra-aural and insert earphones are presented. The salient portions of the ANSI/ASA S3.7 and IEC 60268-4 standards are reviewed. Examples of frequency response, insertion gain, left-right tracking, distortion, and impedance are shown. Issues with interfacing to digital devices over USB or Bluetooth are discussed. Evaluation of active and passive noise canceling devices is also presented.

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

Sound Reinforcement 2 Wednesday, October 18
10:45 am – 12:15 pm Room 1E09

RF BEST PRACTICES AND ADVANCED TECHNIQUES

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

Panelists: *Henry Cohen*, CP Communications
Jim Dugan, JetWave Wrieless
Peter Erskine, Best Audio
Vinny Siniscal, Firehouse Productions

Beyond the basics of accepted RF practices for wireless microphones, intercoms, IEMs, and IFBs is a plethora of facts about antennas, coax, and other passives not commonly understood by the production community at large. This session is comprised of an expert group of RF practitioners who will discuss the various types and performance characteristics of antennas, coax, filters, isolators/circulators, hybrid combiners, directional couplers, and other devices along with their own tips and tricks for dealing with difficult deployments.

Spatial Audio 16 Wednesday, October 18
11:00 am – 12:00 noon Room 1E06

THE BUTTERFLY FX—UNINTENDED CONSEQUENCES OF MASTERING TOOLS

Presenter: **Piper Payne**

[abstract unavailable]

Wednesday, October 18 11:00 am Room 1C03

Technical Committee Meeting on Hearing and Hearing Loss Prevention

Software@AES 02 Wednesday, October 18
11:00 am – 11:20 am Stage 3

ANTARES

Tutorial/Workshop 2 Wednesday, October 18
11:15 am – 12:30 pm Room 1E14

CREATING SOUNDS FROM SCRATCH

Presenters: **Scott B. Metcalfe**, Peabody Conservatory, Johns Hopkins, Severna Park, MD, USA
Andrea Pejrolo, Berklee College of Music, Boston, MA, USA

A live demonstration using synthesis to create unique sounds. Topics included a brief overview of how we arrived at the methods of synthesis commonly used today, why you should avoid using presets, and examples of working with techniques like physical modeling and wavetable synthesis to create novel sounds. Presented by co-authors of Creating Sounds from Scratch (Oxford University Press) Andrea Pejrolo, Chair, Contemporary Writing & Production, Berklee College of Music, and Scott B. Metcalfe, Director of Recording Arts and Sciences and Chair of Computer Music and Music for New Media at the Peabody Conservatory of Johns Hopkins University.

Broadcast Audio Expo 02 Wednesday, October 18
11:15 am – 12:00 noon Stage 1

AUDIO OVER IP (AOIP)

Presenter: **Mark Davies**, TSL Products

History of switching broadcast signals, packet switching vs crosspoints. Working through TDM solutions to the stage where Ethernet and IP become a viable solution for audio transport. A look at proposed standards to identify winners. Finally, AES67's interoperation with proprietary audio standards, ST-2110 and its use in a full, video, audio and data, essence based broadcast production.

Project Studio Expo 02 **Wednesday, October 18**
11:15 am – 12:00 noon **Stage 2**

MAKING THE BEST OF IT

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

In an ideal world we'd all be recording the Beatles in Abbey Road. In the real world we need to make a living, and that means recording anyone who walks through the studio door. And even if they'll never be the Fab Four, it's our job to make their music sound the best it can. In this seminar, SOS Features Editor Sam Inglis offers proven strategies to get the best from musicians who are inexperienced, unconventional, nervous or just plain bad!

Recording & Production 2 **Wednesday, October 18**
11:30 am – 12:30 pm **Room 1E11**

A GUIDE TO A HIGH-QUALITY VINYL RELEASE

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

Panelists *Dustin Blocker*, Hand Drawn Records, Dallas, TX, USA
Cameron Henry, Welcome to 1979, Nashville, TN, USA

An incredible knowledgebase has been assembled with experts on vinyl mastering, electroplating and record pressing. They will discuss not only how to get vinyl records pressed, but how to navigate the process with the goals of high fidelity, customer service and reduced turn-around times.

Spatial Audio 3 **Wednesday, October 18**
11:30 am – 12:30 pm **Room 1E14**

SPATIAL MUSIC: PRODUCING, PERFORMING, AND PUBLISHING MUSIC IN 3D

Presenter: **Kedar Shashidhar**, OSSIC, San Diego, CA, USA

Attendees will learn to produce immersive spatial music in an emerging market with familiar tools. Using examples of spatially mixed recordings from award winning artists, the tutorial will denote different approaches in various genres including EDM, Pop, and Jazz among others. Additional takeaways include various ways in which produced content can translate to a live performance setting while subsequently being released on various online platforms that support spatial formats.

Tutorial Will Cover:
• Basic Concepts in Spatial Audio (HRTFs & Ambisonics)
• Tools, Signal Flow, and Practical Implementation of 3D Audio technology
Creative Tips and Best Practices when producing a track in 3D

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

Software@AES 03 **Wednesday, October 18**
11:30 am – 11:50 am **Stage 3**

FABFILTER

Recording & Production 17 **Wednesday, October 18**
12:15 pm – 1:30 pm **Room 1E06**

RADICAL INTERPRETATIONS OF ICONIC MUSICAL WORKS

Presenter: **Morten Lindberg**, 2L

This event explanes the interdisciplinary research project at Norges musikkhøgskole with WOACT and 2L. Kjell Tore Innervik performing Morton Feldman's "The King of Denmark" and Iannis Xenakis' "Psappha," exploring the intimate performing space with large format recording techniques, engaging the listener in immersive audio. We recorded the Xenakis twice. Once from the perspective of an intimate listener and then literally "over-head" as the performer himself in first persona. The difference not only in microphone technique but also in state-of-mind from the performer and how he projects his playing is profound to the receiving experience of the music.

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

Software@AES 04 **Wednesday, October 18**
12:00 noon – 12:20 pm **Stage 3**

CELEMONY (MELODYNE)

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Wednesday, October 18, 12:30 pm – 2:00 pm
Room 1E15/16

Opening Remarks: • Executive Director Bob Moses
• President Alex Case
Convention Chairs • Paul Gallo & Agnieszka Roginska
Program: • AES Awards Presentation by Andres Mayo
• Introduction of Keynote Speaker
• Keynote Address by Edgar Choueiri

Awards Presentation

Please join us as the AES presents Special Awards to those who have made outstanding contributions to the Society in such areas of research, scholarship, and publications, as well as other accomplishments that have contributed to the enhancement of our industry. The awardees are:

CITATION
• Clair Krepps

BOARD OF GOVERNORS AWARD
• Linda Gedemer
• Michael MacDonald
• Andres Mayo
• Tim Ryan
• Valerie Tyler

FELLOWSHIP AWARD
• Daniel Mapes-Riordan
• Mandy Parnell

SILVER MEDAL AWARD
• Mark Davis
• Ville Pulkki

GOLD MEDAL AWARD
• Malcolm O. Hawksford

Keynote Speaker

This year's Keynote Speaker is Prof. Edgar Choueiri of Princeton University. The title of his address is, "Fooled by Audio".

How far are we from having reproduced or synthesized sound that is truly indistinguishable from reality? Is this laudable goal still the receding mirage it has been since the birth of audio, or are we on the cusp of a technical revolution - the VR/AR audio revolution? I will report on recent advances in virtual and augmented reality audio research from around the world, and focus on critical areas in spatial audio, synthesized acoustics, and sound field navigation in which recent breakthroughs are bringing us quicker and closer to being truly fooled by audio.

Software@AES 05 **Wednesday, October 18**
12:30 pm – 12:50 pm **Stage 3**

INTERNET CO.

Broadcast Audio Expo 03 **Wednesday, October 18**
1:00 pm – 1:45 pm **Stage 1**

WIRELESS UPDATE 2018

Presenters: **Mark Brunner**, Shure Incorporated, Niles, IL, USA
Joe Ciaudelli, Sennheiser, Old Lyme, CT, USA
Karl Winkler, Lectrosonics - Rio Rancho, NM, USA

The 600 MHz spectrum auction concluded last April. Join a panel of experts on spectrum matters, FCC regulations and the upcoming spectrum changes that affect all UHF wireless microphone, intercom, IEM and IFB operations in the core-TV bands.

Software@AES 06 **Wednesday, October 18**
1:00 pm – 1:20 pm **Stage 3**

ACCUSONUS

Project Studio Expo 03 **Wednesday, October 18**
1:00 pm – 1:45 pm **Stage 2**

CREATIVITY IN PRODUCTION

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

[Abstract not available]

Software@AES 07 **Wednesday, October 18**
1:30 pm — 1:50 pm **Stage 3**

LINE 6

Session P4 **Wednesday, Oct. 18**
2:00 pm–5:00 pm **Room 1E11**

TRANSDUCERS—PART 1

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

2:00 pm

P4-1 Estimation of Magnitude Response of Reflecting Loudspeaker System in Listening Area Using Near-Box Measurement—*Ge Zhu, Ziyun Liu, Yong Shen, Yuchen Shen*, Nanjing University, Nanjing, China

This paper presents a simple and robust method to estimate the general magnitude response in reflecting loudspeaker systems. The method utilizes statistical acoustics and is based on near-box impulse response measurement. This measurement holds information across the entire listening area after truncation post-processing. The estimation was investigated in different acoustic environments, which lead to that the more diffusive the room is, the more precise result can be achieved. Measured response can be a reliable reference for correction system in reflecting loudspeaker system.
Convention Paper 9823

2:30 pm

P4-2 Optimal Modulator with Loudspeaker Parameter Inclusion—*Nicolai J. Dahl, Niels Elkjær Iversen, Arnold Knott*, Technical University of Denmark, Kgs. Lyngby, Denmark

Today most class-D amplifier designs are able to deliver high efficiency and low distortion. However, the effect of parasitic component and speaker dynamics are not taken into account resulting in a degradation of the performance. This paper proposes a new PWM modulator that is able to capture an arbitrary amount of dynamics through optimization-based design methods. This makes it possible to include the parasitic components in the amplifier and the loudspeaker parameters in the design, thus creating a more linear response.
Convention Paper 9824

3:00 pm

P4-3 Fast Loudspeaker Measurement in Non-Anechoic Environment—*Christian Bellmann, Wolfgang Klippel,* Klippel GmbH, Dresden, Germany

The evaluation of the loudspeaker performance requires a measurement of the sound pressure output in the far field of the source under free field condition. If the available test room does not fulfil this condition, it is common practice to generate a simulated free field response by separating the direct sound from the room reflection based on windowing and holographic processing. This paper presents a new technique that performs a filtering of the measured sound pressure signal with a complex compensation function prior to other time and frequency analysis. The influence of room, nearfield and positioning error is compensated in the measured fundamental and nonlinear distortion characteristics. Different methods are presented for the generation of the compensation function based on a reference response measured under anechoic conditions and a test response measured under in-situ conditions. Benefits and particularities are demonstrated by practical measurements using different kinds of test signals.
Convention Paper 9825

3:30 pm

P4-4 Analog Circuit Model for Loudspeakers including Eddy Current Behavior and Suitable for Time Domain Simulation—*Stephen C. Thompson,*[1] *Daniel M. Warren*[2]
[1]Pennsylvania State University, State College, PA, USA
[2]GN Advanced Science, Glenview, IL, USA

This paper presents two analog circuit models for the blocked electrical impedance for a moving coil loudspeaker. The first includes an exact model of the effects of eddy currents as derived by Vanderkooy. The model is

implemented using a partial fraction expansion that allows an implementation using conventional electrical circuit components. An alternative circuit suggested by Leach uses fewer components and can model not only a purely semi-inductive behavior, but also other frequency variations that are sometimes observed. Because these eddy current models do not use frequency dependent components, they can be used in time domain simulations of loudspeaker behavior that are capable of modeling mechanical and magnetic nonlinearities.
Convention Paper 9826

4:00 pm

P4-5 Use of Repetitive Multi-Tone Sequences to Estimate Nonlinear Response of a Loudspeaker to Music—*Pascal Brunet,*[1] *William Decanio,*[1] *Ritesh Banka,*[1] *Shenli Yuan*[2]
[1]Samsung Research America, Valencia, CA USA; Audio Lab
[2]Center for Computer Research in Music and Acoustics (CCRMA), Stanford University, Stanford, CA, USA

Aside from frequency response, loudspeaker distortion measurements are perhaps the most commonly used metrics to appraise loudspeaker performance. Unfortunately the stimuli utilized for many types of distortion measurements are not complex waveforms such as music or speech, thus the measured distortion characteristics of the DUT may not typically reflect the performance of the device when reproducing usual program material. To this end, the topic of this paper will be the exploration of a new multi-tone sequence stimulus to measure loudspeaker system distortion. This method gives a reliable estimation of the average nonlinear distortion produced with music on a loudspeaker system and delivers a global objective assessment of the distortion for a DUT in normal use case.
Convention Paper 9827

4:30 pm

P4-6 Non-Invasive Audio Performance Measurement on Wireless Speakers—*Srinath Arunachalam, Douglas J. Button, Jay Kirsch, Meenakshi Barjatia*, Harman International, South Jordan, UT, USA

Wireless audio systems are gaining market share due to their portability, flexibility, and simply because users do not want to be entangled in wires. As with any technology, the advantages come with many challenges, one of which is creating a meaningful measurement of performance. In this paper we propose a non-invasive testing methodology for manufacturers to measure audio performance in their wireless speaker products [3]. The method begins with baseline acoustic measurements using electrical (line-in) inputs, which are used as a reference for measurements of other wireless input types such as Bluetooth and Wi-Fi. The results show the degradations due to the wireless transport.
Convention Paper 9828

Session P5 **Wednesday, Oct. 18**
2:00 pm – 5:00 pm **Room 1E12**

PERCEPTION—PART 2

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

2:00 pm

P5-1 Direct and Indirect Listening Test Methods—A Discussion Based on Audio-Visual Spatial Coherence Experiments—Cleopatra Pike,[1] Hanne Stenzel[2]
[1]University of St Andrews, Fife, UK
[2]University of Surrey, Guildford, Surrey, UK

This paper reviews the pros and cons of using direct measures (e.g. preference, annoyance) and indirect measures (e.g. “subconscious” EEG measures and reaction times, “RTs”) to determine how viewers perceive audio and audio-visual attributes. The methodologies are discussed in relation to spatial coherence testing (whether audio/visual signals arrive from the same direction). Experimental results in coherence testing are described to illustrate problems with direct measures and improvements seen with RTs. Suggestions are made for the use of indirect measures in testing, including more sophisticated uses of RTs. It is concluded that indirect measures offer novel insights into listener evaluations of audio-visual experiences but are not always suitable.
Convention Paper 9829

2:30 pm

P5-2 Identification of Perceived Sound Quality Attributes of 360º Audiovisual Recordings in VR Using a Free Verbalization Method—*Marta Olko, Dennis Dembeck, Yun-Han Wu, Andrea Genovese, Agnieszka Roginska*, New York University, New York, NY, USA

Recent advances in Virtual Reality (VR) technology have led to fast development of 3D binaural sound rendering methods that work in conjunction with head-tracking technology. As the production of 360° media grows, new subjective experiments that can appropriately evaluate and compare the sound quality of VR production tools are required. In this preliminary study a Free Verbalization Method is employed to uncover auditory features within 360° audio-video experiences when paired with a 3-degrees-of-freedom head-tracking VR device and binaural sound over headphones. Subjects were first asked to identify perceived differences and similarities between different versions of audiovisual stimuli. In a second stage, subjects developed bipolar scales based on their verbal descriptions obtained previously. The verbal constructs created during the experiment, were then combined by the authors and experts into parent attributes by means of semantical analysis, similar to previous research on sound quality attributes. Analysis of the results indicated that there were three main groups of the sound quality attributes: attributes of sound quality describing the general impression of the 360° sound environment, attributes describing sound in relation to the head movement, and attributes describing audio and video congruency. Overall, the consistency of sound between different positions in 360° environment seems to create the new fundamental aspect of sound evaluation for VR and AR multimedia content.
Convention Paper 9830

3:00 pm

P5-3 Tonal Component Coding in MPEG-H 3D Audio Standard—*Tomasz Zernicki, Lukasz Januszkiewicz, Andrzej Ruminski, Marcin Chryszczanowicz,* Zylia sp. z o.o., Poznan, Poland

This paper describes a Tonal Component Coding (TCC) technique that is an extension tool for the MPEG-H 3D Audio Standard. The method is used in order to enhance the perceptual quality of audio signals with strong and time-varying high frequency (HF) tonal components. At the MPEG-H 3D Audio Core Coder, the TCC tool exploits

sinusoidal modeling in order to detect substantial HF tonal content and transform it into the so-called sinusoidal trajectories. A novel parametric coding scheme is applied and the additional data are multiplexed into the bitstream. At the decoder side, the trajectories are reconstructed and merged with the output of the 3D Audio Core Decoder. The TCC was tested as an extension to MPEG-H Audio Reference Quality Encoder in low bitrate (enhanced Spectral Band Replication) and low complexity (Intelligent Gap Filling) operating mode. The subjective listening tests prove the statistical improvement of perceptual quality of signals encoded with proposed technique.
Convention Paper 9831

3:30 pm

P5-4 Lead-Signal Localization Accuracy for Inter-Channel Time Difference in Higher and Lower Vertical, Side, and Diagonal Loudspeaker Configurations—*Paul Mayo, Wesley Bulla*, Belmont University, Nashville, TN, USA

The effects of inter-channel time difference (ICTD) on a sound source's perceived location are well understood for horizontal loudspeaker configurations. This experiment tested the effect of novel loudspeaker configurations on a listener's ability to localize the leading signal in ICTD scenarios. The experiment was designed to be a comparison to standard horizontal precedence-effect experiments but with non-traditional loudspeaker arrangements. Examples of such arrangements include vertical, elevated, and lowered configurations. Data will be analyzed using sign and ANOVA tests with listeners' responses being visualized graphically. Outcomes are expected to follow a predicted precedence-based suppression model assuming localization will be concentrated at the leading loudspeaker.
Convention Paper 9832

4:00 pm

P5-5 Non-Intrusive Polar Pattern Estimation in Diffuse Noise Conditions for Time Variant Directional Binaural Hearing Aids—*Changxue Ma,*[1] *Andrew B. Dittberner,*[1] Rob de Vries[2]
[1]GN Resound Inc., Glenview, IL, USA
[2]GN Resound Inc., Eindhoven, The Netherlands

The directivity index is often used to represent the performance of beamforming algorithms on hearing aids. For binaural listening modeling, we also need to measure the directivity patterns. The common method to estimate the directivity patterns is using a single rotating sound source. With this method, it is difficult to obtain a good directivity pattern when the performance of the system dependents on the acoustic environment in an adaptive manner. The directivity pattern can be also confounded by other signal processing components. These processing include direction of arrival (DOA) based nonlinear post filtering. This paper proposes a method to extract the directivity patterns of a beamforming algorithm under diffuse noise conditions with a rotating probe signal. We are able to obtain the directivity pattern from the probe signal by spectral subtraction of diffuse noise sound field in a non-intrusive manner.
Convention Paper 9833

4:30 pm

P5-6 Training on the Acoustical Identification of the Listening Position in a Virtual Environment—*Florian Klein,*[1] *Annika Neidhardt,*[1] *Marius Seipel,*[1] *Thomas Sporer*[2]
[1]Technische Universität Ilmenau, Ilmenau, Germany
[2]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

This paper presents an investigation of the training effect on the perception of position dependent room acoustics. Listeners are trained to distinguish the acoustics at different listening positions and to detect mismatches between the visual and the acoustical representations. In virtual acoustic environments simplified representation of room acoustics are used often. That works well, when fictive or unknown rooms are auralized but may be critical for real rooms. The results show that 10 out of 20 participants could significantly increase their accuracy for choosing the correct combinations after training. The publication investigates the underlying processes of the adaptation effect and the reasons for the individual differences. The relevance of these findings for acoustic virtual/augmented reality applications is discussed.
Convention Paper 9834
[This paper is presented by Annika Neidhardt]

Archiving/Restoration 2 — Wednesday, October 18
2:00 pm – 3:30 pm — Room 1E07

THE EDISON KINETOPHONE

Presenters: **Jerry Fabris**, Thomas Edison National Historical Park, West Orange, NJ, USA
Brad McCoy, The Library of Congress, Culpeper, VA, USA
George Willeman, The Library of Congress, Culpeper, VA, USA

On February 17, 1913, Thomas A. Edison premiered his newest development in motion pictures—the Edison Kinetophone, a wonderful mechanical system for presenting talking pictures. Taking two of his most popular "inventions," the phonograph and the motion picture, Edison's engineers worked out a way to record live sound while simultaneously shooting film and then play back the two elements in sync. The Kinetophone was met with a standing ovation at the premiere and the showings went well for about two weeks. The downfall of the system was one of mechanical complexity and the reality that the human operator just could not keep the film and sound in sync. Although much time and development were put into the Kinetophone, within a year, production stopped and the system was quietly retired. Fast forward a century and with the collaboration of The Library of Congress and The Thomas Edison National Historical Park, the eight Kinetophones known to survive with both picture and sound have been reconstructed, using some of the newest digital applications for both picture and sound. The films now look better and sound better than they did even when they were new.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Audio for Cinema 2 — Wednesday, October 18
2:00 pm – 3:00 pm — Room 1E14

THE ART AND SCIENCE OF FOLEY

Chair: **Leslie Bloome**, Alchemy Post Sound

Panelists: *Andrea Bloome*, Alchemy Post Sound
Ryan Collison
Joanna Fang

Talk to the Emmy award winning Foley team from Alchemy Post

Sound about how they create organic sounds from a technical and artistic perspective, including: What is Foley; What techniques are commonly used to integrate Foley into the project's soundscape; What is the division of labor between Foley and sound design.

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

Broadcast/Media Streaming 3 **Wednesday, October 18**
2:00 pm – 3:30 pm **Room 1E08**

DECONSTRUCTING BINAURAL MACBETH

Presenters: **Sue Zizza**, Producer/Moderator
David Schinn, Master Engineer
Neil Hellegers, Co-Producer/Director
Amanda Rose Smith, Co-Producer/Director

Cast: *Dion Graham*, MacBeth
William Dufris, Additional Voices
Robin Miles, Additonal Voices

At the 2017 HEAR Now Festival we presented a special "reading" of Act IV, Scene 1 of Shakespeare's *MacBeth*.

The performance, inspired in part by the recent Tony winning Broadway show *The Encounter*, featured a Soundman dummy "head" with 2 DPA mono omni-directional capsules. The co-directors took special care to create the actor's staging around the binaural microphone head, which considered the proximity, intensity, and location arrangement. Sound design elements were added to the mix to compliment the "listening" experience. The result was a combination of live theater-going and full-cast audio drama, though a highly audio immersive sense, it sounded to the listener as if they were sitting center stage, around the very cauldron containing the witches' brew. In some ways similar to the Broadway show, while at the same time unique for the presentation of "audio fiction." As this program was broadcast/streamed live during the performance, care had to also be taken to maintain the performance for the broadcast stream.

This session will discuss the streaming aspects as well as discuss and demonstrate working with the various technologies to put the audience in the middle of the action of this famous moment in Shakespeare's play.

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

Product Development 2 **Wednesday, October 18**
2:00 pm – 3:30 pm **Room 1E13**

PARALLEL DEVELOPMENT: SPEED TIME TO MARKET

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

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

Spatial Audio 4 **Wednesday, October 18**
2:00 pm – 3:00 pm **Room 1E06**

SPATIAL AUDIO FOR MULTI-TRACKED RECORDINGS—WORKFLOWS, PHASE RELATIONSHIPS, EQUALIZATION, REVERBERATION AND DELIVERY SPECS

Presenter: **Albert Leusink**, Tumble and Yaw

VR and AR are making a big push onto the world's stage with projected revenues to exceed $120B by 2020. Spatial audio is an integral part of this, albeit in Ambisonics, Dolby Atmos, or other formats. In this tutorial we will learn about the differences and similarities between spatial and stereo workflows, illustrated by real-world examples. We will discuss binaural rendering engines and its impact on phase coloration and frequency response, phase relationships of stereo and mono sources in the Ambisonic soundfield, and loudness management for different delivery platforms.

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

Tutorial/Workshop 3 **Wednesday, October 18**
2:00 pm – 3:30 pm **Room 1E10**

DISRUPTION: MIDI—MACHINE LEARNING LOOKING BACK AND LOOKING AHEAD

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

Panelists: *Jonathan Bailey*, iZotope, Inc., Cambridge, MA, USA
Joshua D. Reiss, Queen Mary University of London, London, UK
Dave Smith, Dave Smith Instruments, San Francisco, CA, USA

AI, machine learning and deep learning are beginning to change audio products and audio production workflows. When new technologies threaten to disrupt existing practices and market, there's inevitable uncertainty and concern that accompanies these developments. By looking at historic examples of disruptive technology (eg MIDI) we might understand a little about the adaptation that audio production could undergo moving forward. By looking ahead we'll discuss what the new technological 'features' are that are coming out of machine learning development, describe how users are interacting with the new breed of tools and think about what's coming down the road.

Student Events/Career Development
EC1: OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Wednesday, October 18, 2:00 pm – 3:30 pm
Room 1E09

The first Student Delegate Assembly (SDA) meeting is the official opening of the Convention's student program and a great opportunity to meet with fellow students from all corners of the world. This opening meeting of the Student Delegate Assembly will introduce new events and election proceedings, announce candidates for the coming year's election for the North & Latin American Regions Vice Chair, announce the finalists in the Student Recording Competition categories and the Student Design Competition, and announce all upcoming student/education related events of the convention. Students and student sections will be given the opportunity to introduce themselves and their activities, in order to stimulate international contacts. The SDA leaders will then lead a dialog to discuss important issues significant to all audio students.

All students and educators are invited to participate in this meeting. Election results and Recording Competition and Design Competition Awards will be given at the Student Delegate Assembly Meeting–Part 2 on Saturday, October 21.

Standards Committee Meeting
SC-04-08 WORKING GROUP ON MEASUREMENT AND EQUALIZATION OF SOUND SYSTEMS IN ROOMS
Wednesday, October 18, 2:00 pm – 3:30 pm
Room 1C04

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.

Broadcast Audio Expo 04 **Wednesday, October 18**
2:00 pm – 2:45 pm **Stage 1**

MUSIC MIXING FOR BROADCAST

Presenter: **Paul Special**, Special Audio Sevices, Wantage, NJ, USA; ABC/Disney Television, New York, NY, USA

[Abstract unavailable]

Project Studio Expo 04 **Wednesday, October 18**
2:00 pm – 3:00 pm **Stage 2**

THE 5 MOST COMMON RECORDING MISTAKES

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

Tearing your hair out at mixdown? Then you've probably fallen into one of the classic project-studio traps during tracking. Learn what they are—and, more importantly, how to avoid them—in this down-to-earth workshop with Sound On Sound magazine's "Session Notes" and "Mix Rescue" columnist Mike Senior.

Software@AES **Wednesday, October 18**
2:00 pm – 2:20 pm **Stage 3**

BLUE CAT AUDIO

Session EB1 **Wednesday, Oct. 18**
2:15 pm – 3:45 pm **Poster Area**

POSTERS—PART 1

2:15 pm

EB1-1 Frontal Sound Localization with Headphone Systems Using Characteristics of Each Device—*Yuya Ono, Akira Saji, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

Generally, it is difficult to localize sounds on frontal positions with headphones because of the so called in-the-head localization. In this paper we proposed a method to create sound images on frontal positions by cancelling the frequency characteristics of sounds from headphones and adapting the frequency characteristics of sounds from frontal loudspeaker that are measured with HATS. We held auditory experiments that evaluating the similarity and naturalness between the sounds from frontal loudspeaker and headphones on the participants. As a result, we found the method can decrease the in-the-head localization
Engineering Brief 354

2:15 pm

EB1-2 Design of Loudspeaker Crossover Networks Using Genetic Algorithm—*Christian Claumarch, Kasper Kiis Jensen, Mikkel Krogh Simonsen*, Aalborg University, Aalborg, Denmark

This brief proposes a method which allows for automation of the design of crossover networks for loudspeakers. The algorithm utilizes a combination of Genetic Algorithm and Least Squared Error Frequency Domain Design to find optimal type 1 FIR-filters fitting a target. The algorithm takes magnitude response, group delay, directivity index and displacement into account. Based on a two-way loudspeaker, the algorithm is capable of finding a crossover network which results in the final loudspeaker having a flat magnitude response from 105 Hz– 16 kHz ± 1.6 dB on-axis, where the directivity index is minimized and displacement limits are not exceeded.
Engineering Brief 355

2:15 pm

EB1-3 Microphone Array Impulse Response (MAIR) Library for Spatial Audio Research—*Hyunkook Lee, Connor Millns*, University of Huddersfield, Huddersfield, UK

This engineering brief describes an open-access library of an extensive set of room impulse responses (RIRs) captured using numerous microphone arrays from 2-channel stereo to 9-channel surround with height. The RIRs were obtained for 13 loudspeakers placed in various positions on a stage in a reverberant concert hall. The library features five 2-channel stereo pairs, 10 main surround arrays, nine height microphone arrays for 3D main arrays and 15 4-channel configurations for surround and 3D ambience arrays, each with varied microphone polar patterns, directions, spacings, and heights. A dummy head and a first-order-Ambisonics microphone are also included. The library is provided with a rendering tool, with which the user can easily simulate different microphone combinations in both loudspeaker and binaural playback for 13 source positions. The library can be freely downloaded from the Resources section of the APL website: https://www.hud.ac.uk/apl
Engineering Brief 356

2:15 pm

EB1-4 A Database of Head-Related Transfer Functions and Morphological Measurements—*Rahulram Sridhar, Joseph G. Tylka, Edgar Choueiri*, Princeton University, Princeton, NJ, USA

A database of head-related transfer function (HRTF) and morphological measurements of human subjects and mannequins is presented. Data-driven HRTF estimation techniques require large datasets of measured HRTFs and morphological data, but only a few such databases are freely available. This paper describes an on-going project to measure HRTFs and corresponding 3D morphological scans. For a given subject, 648 HRTFs are measured at a distance of 0.76 m in an anechoic chamber and 3D scans of the subject's head and upper torso are acquired using structured-light scanners. The HRTF data are stored in the standardized "SOFA format" (spatially-oriented format for acoustics) while scans are stored in the Polygon File Format. The database is freely available online.
Engineering Brief 357

2:15 pm

EB1-5 Phase Continuity in Amplitude-Phase Spatial Audio Coding—*François Becker, Benjamin Bernard, Clément*

Carron, Coronal AudioParis, France

In the context of amplitude-phase spatial audio coding, we give a proof of a phase discontinuity problem that affects all previous tridimensional stereo-compatible schemes. We solve it by using a dynamic mapping of spherical coordinates to the Scheiber sphere, which ensures phase continuity.
Engineering Brief 358

2:15 pm

EB1-6 Evaluation of Binaural Renderers: A Methodology— *Gregory Reardon,*[1] *Agnieszka Roginska,*[1] *Patrick Flanagan,*[2] *Juan Simon Calle,*[1,2] *Andrea Genovese,*[1] *Gabriel Zalles,*[1] *Marta Olko,*[1] *Christal Jerez*[1]
[1]New York University, New York, NY, USA
[2]THX Ltd., San Francisco, CA, USA

Recent developments in immersive audio technology have motivated a proliferation of binaural renderers used for creating spatial audio content. Binaural renderers leverage psychoacoustic features of human hearing to reproduce a 3D sound image over headphones. In this paper a methodology for the comparative evaluation of different binaural renderers is presented. The methodological approach is threefold. A subjective evaluation of 1) quantitative characteristics (such as front/back and up/down discrimination, localization) ; 2) qualitative characteristics (such as timbre, naturalness); and 3) overall preference. The main objective of the methodology is to help to elucidate the most meaningful factors for the performance of binaural renderers and to provide indications on possible improvements in the rendering process.
Engineering Brief 359

2:15 pm

EB1-7 Simultaneous HRTF Measurement of Multiple Source Configurations Utilizing Semi-Permanent Structural Mounts—*Calum Armstrong, Andrew Chadwick, Lewis Thresh, Damian Murphy, Gavin Kearney*, University of York, York, UK

A compact HRTF measurement rig has been designed and erected within the anechoic chamber at AudioLab, University of York. Utilizing 24 discrete elevations the efficient simultaneous HRTF measurement of 11 popular source configurations, ideally suited for the binaural rendering of Ambisonics, is undertaken. An overlapped exponential swept sine technique is used to make optimal use of a subject's time. This report details the practical requirements, technical workflow and processing involved in the HRTF measurements, for inclusion to the SADIE database. The necessary modelling of low frequency cues is discussed
Engineering Brief 360

2:15 pm

EB1-8 Influence of Audience Noises on the Classical Music Perception on the Example of Anti-cough Candies Unwrapping Noise—*Adam Pilch, Bartłomiej Chojnacki, Teresa Makuch, Zuzanna Kusal, Marcjanna Czapla*, AGH University of Science and Technology, Kracow, Poland

A common problem in concert halls are people in the audience who distract other listeners by creating noises. Unwrapping anti-cough candies is an example of such undesirable behavior. The subject of the paper is to compare and analyze acoustic parameters of various candy wrappings in order to determine the discomfort they cause. The sounds generated while removing wrappings made of different materials were recorded in an anechoic chamber. The recordings were then analyzed in order to locate sounds in the audible frequency band in relation to musical sounds. Based on the results and a survey that was also carried out, an attempt was made to specify parameters of the noises perceived as most distracting.
Engineering Brief 361

2:15 pm

EB1-9 A Simple Evaluating Method of a Reproduced Sound Field by a Measurement of Sound Intensities Using Virtual Source Visualizer—Masataka Nakahara,[1,2] Akira Omoto,[1,3] *Yasuhiko Nagatomo*[4]
[1] ONFUTURE Ltd., Tokyo, Japan
[2] SONA Corp., Tokyo, Japan
[3] Kyushu University, Fukuoka, Japan
[4] Evixar Inc., Tokyo, Japan

Recently, many kinds of technologies for restoring/reproducing 3D sound fields are proposed. However, it is little opportunity to compare these acoustic performances under a common condition. Though a subjective evaluation is one of the most effective methods for evaluating reproduced sound fields, it requires careful effort. Therefore, the authors propose an alternative method which requires only a physical measurement of sound intensities. Because the method is based on the intensity analysis, "sound images" are assumed to be "amplitude-based phantom sound sources" here. In order to verify effectiveness of the method, various types of reproduced fields were measured and analyzed. As a result, it is ascertained that the method can evaluate proper features of reproduced sound fields, regardless of their restoring techniques.
Engineering Brief 362

2:15 pm

EB1-10 Physical Evaluations of Reproduced Sound Fields by a Measurements of Sound Intensities Using Virtual Source Visualizer—*Takashi Mikami,*[1] *Masataka Nakahara,*[1,2] *Akira Omoto* [2,3]
[1] SONA Corp., Tokyo, Japan
[2] ONFUTURE Ltd., Tokyo, Japan
[3] Kyushu University, Fukuoka, Japan

In order to evaluate acoustic properties of reproduced sound fields, sound intensities were measured and analyzed in various types of multichannel studios by using a Virtual Source Visualizer (VSV hereafter). First, two different methods to reproduce sound fields are examined; 24ch amplitude-based phantom sound sources and Kirchhoff-Helmholtz-integral-based Boundary Surface Control principle. Secondly, sound fields created by four different types of 3D panners are examined; Dolby Atmos, DTS:X, Auro-3D and 22.2ch. Through these measurements, it was demonstrated that the VSV analyzes acoustic features of reproduced fields well, and interchangeabilities and differences of acoustic properties among different reproduced fields can be understood clearly. The session discusses accuracy and features of various types of reproduced sound fields which we measured and analyzed by the VSV.
Engineering Brief 363

Software@AES 09 — Wednesday, October 18
2:30 pm – 2:50 pm — Stage 3

BEST SERVICE

Wednesday, October 18 3:00 pm Room 1C03

Technical Committee Meeting on Spatial Audio

Broadcast Audio Expo 05 Wednesday, October 18
3:00 pm – 3:45 pm Stage 1

BROADCAST MUSIC FOR GMA

Presenters: **Paul Special**, Special Audio Sevices, Wantage, NJ, USA; ABC/Disney Television, New York, NY, USA
Jim vanBergen

Top music acts, such as Green Day, Taylor Swift, Ed Sherran or Brad Paisley perform on Good Morning America for millions. GMA's award winning Broadcast Music Mixer Paul Special discusses what's involved mixing 'live to air,' providing a detailed, behind the scenes look at the day to day planning, the equipment, the setup and the music mix for this EMMY award winning morning show.

Project Studio Expo 05 Wednesday, October 18
3:00 pm – 3:45 pm Stage 2

FUTURE PROOFING AND EXPANDING YOUR RECORDING SET UP

This presentation will provide an overview of the benefits of a Dante-networking system, no matter what your workflow or DAW. The Dante protocol lets you run additional channels over great distances, with low latency and lower cost, without wasting money on technology or connectivity that will not be used. Focusrite will show you how you can future-proof your set up with its Red range of Dante / Pro Tools | HD interfaces, featuring superior quality and I/O flexibility. Red interfaces are the perfect balance of form and function, delivering the sound quality and versatility engineers and producers expect from Focusrite.

Software@AES Wednesday, October 18
3:00 pm – 3:20 pm Stage 3

SONARWORKS

Standards Committee Meeting
SC-02-01 WORKING GROUP ON DIGITAL AUDIO MEASUREMENT TECHNIQUES
Wednesday, October 18, 3:30 pm – 5:00 pm
Room 1C04

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.

Software@AES 11 Wednesday, October 18
3:30 pm – 3:50 pm Stage 3

ANTARES

Archiving/Restoration 3 Wednesday, October 18
3:45 pm – 5:15 pm Room 1E07

SOUNDSTREAM: THE FIRST COMMERCIAL DIGITAL AUDIO RECORDING SYSTEM

Presenter: **Paul R. Blakemore**, Concord Music Group, Cleveland, OH, USA

Paul Blakemore will present a workshop on the history of the Soundstream digital tape recording system including an overview of its operating principals, projects recorded with the system, and a playback demonstration of an original Soundstream master tape using a working Soundstream machine that dates from about 1980.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Audio for Cinema 3 Wednesday, October 18
3:45 pm – 5:15 pm Room 1E10

FUTURE OF AUDIO FOR CINEMA

Chair: **Nuno Fonseca**, ESTG/Polytechnic Institute of Leiria, Leiria, Portugal; Sound Particles, Leiria, Portugal

Panelists: *Kevin Collier*, Director of Engineering, Post Production, Warner Bros Studio Facilities
Douglas Greenfield, Dolby Labs, Burbank, CA, USA
Avi Laniado, Harbor
Brian A. Vessa, Sony Pictures Entertainment, Culver City, CA, USA; Chair SMPTE 25CSS standards committee

Technology is constantly changing and audio is not an exception. What would be the future of audio for cinema? What are the unresolved challenges that still need to be addressed? This session will discuss the future of audio for cinema with several experts on the field.

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

Game Audio & VR 3 Wednesday, October 18
3:45 pm – 4:45 pm Room 1E13

VR AUDIO: WORKFLOWS AND BACKGROUND FOR A NEW INTERACTIVE AUDIO PARADIGM

Presenter: **Sally-Anne Kellaway**, Zero Latency VR, Melbourne, Australia

Audio for Virtual Reality (VR) presents a significant array of challenges and augmentations to the traditional requirements of sound designers employed within the video games industry. The change in perspective and embodiment of the player, as well as the new technologies involved requires the employment of additional tools and consideration of object size, spacing, and spatial design as a more significant part of the sound design process. We will unpack some of the new technologies and potential approaches to these tasks from the perspective of developing audio for all scales of interactive VR applications. Focusing on the design considerations and processes required in this unique medium, the content of this presentation is designed to give insight in to the complexity of designing for this new medium, and potential pipeline and work flow approaches.

Recording & Production 3 **Wednesday, October 18**
3:45 pm – 5:15 pm **Room 1E15/16**

SPECIAL EVENT: RECORDING STUDIO DESIGN ADDRESSES IMMERSIVE AUDIO PRODUCTION

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

As with such successful past innovative audio format rollouts as stereo and Surround Sound, the rapid proliferation of Immersive Sound theatrical presentations has motivated an increasing number of global recording studio upgrades. An estimated 90% of today's feature films are produced with Immersive Sound. And, the format is ubiquitous with videogame producers/players. To meet the production/mixing requirements of this swiftly established innovation, a significant number of studio design issues including acoustics, aesthetics, ergonomics, master planning and future proofing must be addressed. This international panel of studio owners, engineers, designers, and acousticians will examine the requirements and realities associated with building or upgrading to meet Immersive Audio production needs.

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

Sound Reinforcement 3 **Wednesday, October 18**
3:45 pm – 5:15 pm **Room 1E09**

RF AND TV SPECTRUM—THE POST AUCTION UPDATE

Moderator: **Karl Winkler**, Lectrosonics

Panelists: *Mark Brunner*, Shure
Joe Ciaudelli, Sennheiser
Jackie Green, Alteros

The 600 MHz spectrum auction concluded in April of this year. Join a panel of experts on spectrum matters, FCC regulations and the upcoming spectrum changes that affect all UHF wireless microphone, intercom, IEM and IFB operations in the core-TV bands, along with some new band allocations to be available in the near future.

Spatial Audio 6 **Wednesday, October 18**
3:45 pm – 4:45 pm **Room 1E13**

PERCEPTUAL THRESHOLDS OF SPATIAL AUDIO LATENCY FOR DYNAMIC VIRTUAL AUDITORY ENVIRONMENTS

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

Generating the acoustic signals that reproduce the properties of natural environments through headphones remains a significant technical challenge. One hurdle is related to the time it takes to update the signal each time the observer moves. The end-to-end spatial audio latency (SAL) is the time elapsed between the listener assuming a new position and the updated sound being delivered to their ears. It is comprised of latencies in head-tracking, HRTF interpolation and filtering, operating system callback, audio driver and hardware (D/A conversion) buffering, and other parts of the signal processing chain. Because SAL is currently inevitable, it is important to know what SAL is detectable to set minimum thresholds for SAL in virtual auditory environments.

We used a 2-interval-forced-choice paradigm to measure SAL detectability at (10 and 60 degree) azimuths, both with and without the presence of co-located visual stimuli. Overall, mean SAL thresholds were between 128ms and 158ms. Consistent with results from minimum audible motion angle data, thresholds were greater at larger azimuthal positions. A retrospective analysis revealed that listeners who strategically varied the velocity, acceleration and rate of their head rotation were better able to perform the task. This suggests that thresholds for SAL will be lower for applications where users are expected to move their heads more rapidly and abruptly. Results are discussed in the context of prior research and the potential implications for rendering Virtual Reality audio.

Broadcast/Media Streaming 4 **Wednesday, October 18**
4:00 pm – 5:45 pm **Room 1E08**

EVOLVING BEST PRACTICES FOR STUDIO CONSTRUCTION AND REMODELING

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

Panelists: *Anthony Gervasi*, Former Senior Vice President Engineering & Technology - Nassau Broadcasting Partners, LP
Daniel Hyatt, Director of Engineering & IT - Max Media Denver, Principal Owner - DK Global Soucing
Gary Kline, Kline Consulting Group LLC - Broadcast Engineering & IT Consultant
Jason Ornellas, CBS RADIO Sacramento
Chris Tobin, Chief Engineer - Newark Public Radio, Owner - IP Codecs Consulting

Revolutionary changes in broadcast technology are impacting the way we plan, build, and remodel our broadcast studios. Both radio and television production spaces are affected by Audio and Video over IP, LED lighting, smaller yet more numerous cameras, and even more diverse connections to on-air talent. These technology advances inform our decisions about studio size, configuration, materials, and supporting infrastructure. That's the latest thinking about technical centers such as "rack rooms" and master control rooms? Is the in-house data center replacing or adding to traditional technical cores? How are backup systems changing? What new justifications are key when specifying main and auxiliary workflow and infrastructure systems? Leading broadcast engineers and system integrators will discuss the latest practices and predict what planning is critical for tomorrow's broadcast facilities.

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

Spatial Audio 5 **Wednesday, October 18**
4:00 pm – 5:30 pm **Room 1E13**

THE STATE OF THE ART OF BINAURAL AUDIO FOR LOUDSPEAKERS AND HEADPHONES

Presenter: **Edgar Choueiri**, Princeton University, Princeton, NJ, USA

I will describe the challenges of binaural audio through headphones (BAH) and loudspeakers (BAL), recent solutions to these challenges, and the state of the art of binaural processing and content development tools. In particular I will describe BACCH 3D Sound processing, which relies on optimal crosstalk cancellation filters, head tracking and automatic individualization to deliver accurate 3D imaging from binaural audio. I will then describe the recently developed BACCH-HP headphones processing, which significantly enhances the robustness of 3D imaging and the ability to head-externalize binaural audio. I will use the powerful BACCH-dSP software, which allows designing BACCH filters for BAL and BAH, processing binaural audio, translational and rotational head tracking, and 3D mixing, to illustrate the talk and demonstrate the technologies.

Student Events/Career Development
EC2: STUDENT RECORDING CRITIQUES
Wednesday, October 18, 4:00 pm –5:00 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.

Wednesday, October 18 4:00 pm Room 1C03

Technical Committee Meeting on Microphones and Applications

Broadcast Audio Expo 06 Wednesday, October 18
4:00 pm – 4:45 pm Stage 1

LIVE MUSIC MIXING FOR BROADCAST

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.

Project Studio Expo 06 Wednesday, October 18
4:00 pm – 4:45 pm Stage 2

A MUSICIAN'S GUIDE TO CARING FOR YOUR RECORDIST

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

Tape Op's editor/founder, Larry Crane, will be giving suggestions for musicians heading into the studio on how to get more out of their recording sessions by treating the recordist with mutual respect and setting them up to do their best work.

Software@AES 12 Wednesday, October 18
4:00 pm – 4:20 pm Stage 3

FABFILTER

Software@AES 13 Wednesday, October 18
4:30 pm – 4:50 pm Stage 3

CELEMONY (MELODYNE)

Game Audio & VR 2 Wednesday, October 18
5:00 pm – 6:00 pm Room 1E13

IMMERSIVE AUDIO & VR/AR: A TRAVELER'S GUIDE TO THE GROWING LANDSCAPE OF EXPERIENTIAL AUDIO

Presenters: **Josh Antonuccio**, Ohio University, Athens, OH, USA
Richard Warp, Intonic, Emeryville, CA, USA

Audio is going to be essential for creating immersive experiences, but what should the aspiring producer or engineer learn in order to excel, and how is higher education responding to the skills gap in this new field? This session is led by Josh Antonuccio, lecturer at the School of Media Arts and Studies at Ohio University and co-creator of the Immersive Media Initiative, and Richard Warp, founder of Intonic and VR audio professional. Topics include workflows and technology for immersive platforms, considerations for working with spatial audio, and the growing need for audio development expertise within augmented and virtual reality. Enjoy a fascinating "fireside" discussion covering educational, technical, and professional aspects of producing audio in the domain of "the 3 Rs"–VR, AR, and MR.

Wednesday, October 18 5:00 pm Room 1C03

Technical Committee Meeting on Recording Technology and Practices

Broadcast Audio Expo 07 Wednesday, October 18
5:00 pm — 5:45 pm Stage 1

REAL WORLD AES67 NETWORKING

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

Panelists: *Claudio Becker-Foss*, DirectOut
Will Hoult, Focusrite
Jeffrey Stroessner, Lawo
Nicolas Sturmel, Merging Technologies, Puidoux, Switzerland

The session, moderated by Andreas Hildebrand (ALC NetworX), focuses on practical aspects of using AES67 in the field. The panelist provide valuable insights experienced in real-world applications and share opinions on the importance of AES67 for the wider broadcast market. An outlook on the AES67 standard evolution and its potential influence on related emerging standards and other important industry work will round-up the discussion. Panelists include Nicolas Sturmel (Merging), Claudio Becker-Foss (DirectOut), Jeffrey Stroessner (Lawo) and Will Hoult (Focusrite).

Project Studio Expo 17 Wednesday, October 18
5:00 pm – 5:45 pm Stage 2

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

Presenters: **Miguel Chavez**, Analog Devices
David Thibodeau

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

Our demonstration will consist of developing an application using SigmaStudio starting from a blank file and coming up with a functioning solution and programming it into an EEPROM while you watch. Other more advanced concepts will be detailed using prepared projects that showcase capabilities and ease of programming.

Software@AES 14 — **Wednesday, October 18**
5:00 pm – 5:20 pm — **Stage 3**

ACCUSONUS

Networked Audio 1 — **Wednesday, October 18**
5:15 pm – 6:00 pm — **Room 1E09**

PRACTICAL BENEFITS THROUGH AES67 DEPLOYMENT IN THE BROADCAST INDUSTRY: A SYNOPSIS

Presenter: **Nestor Amaya**, COVELOZ

With migration to the network now firmly on the horizon as we look to leverage commodity infrastructure for real-time media distribution, we will discuss how AES67 opens the door for a broader diversity of networked-connected solutions in your facility. As an interoperability standard for high-performance audio over IP, learn about those manufacturers at the forefront of this technology and how AES67 provides the ability to combine products and solutions from multiple vendors.

HISTORICAL COMMITTEE MEETING
Wednesday, October 18, 5:15 pm – 6:00 pm
Room 1E11

The Historical Committee will hold a meeting during the convention. The meeting will be informal, with updates on Historical Committee activities. We will use most of the time for questions and comments from YOU, the vital members of the Society. Please come and learn what the AES Historical Committee is doing and planning. We hope to see you there!

Software@AES 15 — **Wednesday, October 18**
5:30 pm – 5:50 pm — **Stage 3**

LINE 6

Special Events
SE3: THE RICHARD C. HEYSER MEMORIAL LECTURE
Wednesday, October 18, 6:00 pm – 7:30 pm
Room 1E15/16

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

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

The Heyser Lecturer this year is **Leslie Ann Jones**.

Leslie Ann Jones has been a recording and mixing engineer for over 35 years. Starting her career at ABC Recording Studios in Los Angeles in 1975, she moved to Northern California in 1978 to accept a staff position with David Rubinson and Fred Catero at the legendary Automatt Recording Studios. There she worked with such artists as Herbie Hancock, Bobby McFerrin, Holly Near, Angela Bofill, Frankie Beverly and Maze, Carlos Santana and Narada Michael Walden, and started her film score mixing career with *Apocalypse Now*.

From 1987 to 1997 she was a staff engineer at Capitol Studios located in the historic Capitol Records Tower in Hollywood. She recorded projects with Rosemary Clooney, Michael Feinstein, Michelle Shocked, BeBe & CeCe Winans, and Marcus Miller, as well as the scores for several feature films and television shows.

In February of 1997 she returned to Northern California to accept a position as Director of Music Recording and Scoring with Skywalker Sound, where she continues her engineering career recording and mixing music for records, films, video games, television, and commercials. And she now adds Record Producer to her list of credits.

Leslie has received four Grammy Awards and has been nominated multiple times including Best Engineered Recording, Non-Classical and Best Surround Sound Album.

Leslie is a past Chair of The Recording Academy's Board of Trustees. She serves on the Advisory Boards of Institute for Musical Arts, Ex'pression College for Digital Arts, G.A.N.G. (Game Audio Network Guild) and is an Artistic Advisor to the new Technology and Applied Composition degree program at the San Francisco Conservatory of Music. The title of her lecture is "Paying Attention."

No graphs, no charts, no powerpoint, okay maybe a few photos; but mostly a retrospective on how my life and career have been sustained and enriched by paying attention.

Broadcast/Media Streaming 5 — **Wednesday, October 18**
7:30 pm – 9:00 pm — **One World Trade Center**

THE BROADCAST FACILITY OF ONE WORLD TRADE CENTER

Moderator: **John Lyons**

On October 18th 2017 at 7:30pm Durst Broadcasting LLC will be hosting a short Panel discussion and tour of the New Preeminent Broadcast Facility in New York City. John Lyons, Assistant Vice President and Director of Broadcast Communications for The Durst Organization will be hosting the session along with a panel still TBA.

The discussion will include the origin of Durst involvement with the Port Authority on the project, the innovative technologies employed including green innovations and logistical innovations and the intricacies involved with developing and building a broadcast facility while still building the property in which the facility sits.

Following the panel discussion will be a tour of the broadcasting facilities and communications spaces. The tour will not include the roof as there is ongoing work there on the building maintenance systems and darkness will not make it conducive to touring.

Capacity is limited due to security issues, so be sure to book early to be a part of this experience.

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

Session P6 **Thursday, Oct. 19**
9:00 am – 12:00 noon **Room 1E11**

SPATIAL AUDIO—PART 1

Chair: **Ravish Mehra**, Oculus Research, Redmond, WA, USA

9:00 am

P6-1 Efficient Structures for Virtual Immersive Audio Processing—*Jean-Marc Jot,*[1] *Daekyoung Noh*[2]
[1]Magic Leap, Sunnyvale, CA, USA
[2]Xperi Corp., Santa Ana, CA, USA

New consumer audio formats have been developed in recent years for the production and distribution of immersive multichannel audio recordings including surround and height channels. HRTF-based binaural synthesis and cross-talk cancellation techniques can simulate virtual loudspeakers, localized in the horizontal plane or at elevated apparent positions, for audio reproduction over headphones or convenient loudspeaker playback systems. In this paper we review and discuss the practical design and implementation challenges of immersive audio virtualization methods and describe computationally efficient processing approaches and topologies enabling more robust and consistent reproduction of directional audio cues in consumer applications.
Convention Paper 9865

9:30 am

P6-2 Robust 3D Sound Capturing with Planar Microphone Arrays Using Directional Audio Coding—*Oliver Thiergart, Guendalina Milano, Tobias Ascherl, Emanuël A. P. Habets*, International Audio Laboratories Erlangen, Erlangen, Germany

Real-world VR applications require to capture 3D sound with microphone setups that are hidden from the field-of-view of the 360-degree camera. Directional audio coding (DirAC) is a spatial sound capturing approach that can be applied to a wide range of compact microphone arrays. Unfortunately, its underlying parametric sound field model is often violated which leads to a degradation of the spatial sound quality. Therefore, we combine the non-linear DirAC processing with a linear beamforming approach that approximates the panning gains in DirAC such that the required amount of non-linear processing is reduced while increasing the robustness against model violations. Additionally, we derive a DOA estimator that enables 3D sound capturing with DirAC using compact 2D microphone arrays, which are often preferred in VR applications.
Convention Paper 9866

10:00 am

P6-3 Frequency Bands Distribution for Virtual Source Widening in Binaural Synthesis—*Hengwei Su, Atsushi Marui, Toru Kamekawa*, Tokyo University of the Arts, Adachi-ku, Tokyo, Japan

The aim of this study is to investigate the perceived width in binaural synthesis. To synthesize sounds with extended source widths, monophonic signals were divided by 1/3-octave filter bank, each component was then distributed to different directions by convolution with head-related transfer function within the intended width. A subjective listening experiment was conducted by using pairwise comparison to evaluate differences of perceived widths between stimuli with different synthesis widths and distribution methods. The results showed that this processing method can achieve a wider sound source width in binaural synthesis. However, effectiveness may vary with spectrum characteristics of source signals. Thus, a further revision of this method is needed to improve the stability and the performance.
Convention Paper 9867

10:30 am

P6-4 Improving Elevation Perception in Single-Layer Loudspeaker Array Display Using Equalizing Filters and Lateral Grouping—*Julian Villegas, Naoki Fukasawa, Yurina Suzuki*, University of Aizu, Aizu Wakamatsu, Japan

A system to improve the perception of elevated sources is presented. This method relies on "equalizing filters," a technique that aims to compensate for unintended changes in the magnitude spectrum produced by the placement of loudspeakers with respect to the desired location. In the proposed method, when sources are on the horizon, a maximum of two loudspeakers are used for reproduction. Otherwise, the horizon spatialization is mixed with one that uses side loudspeakers grouped by lateral direction. Results from a subjective experiment suggest that the proposed method is capable of producing elevated images, but the perceived elevation range is somewhat compressed.
Convention Paper 9868

11:00 am

P6-5 Development and Application of a Stereophonic Multichannel Recording Technique for 3D Audio and VR—*Helmut Wittek,*[1] *Günther Theile*[2]
[1]SCHOEPS GmbH, Karlsruhe, Germany
[2]VDT, Geretsried, Germany

A newly developed microphone arrangement is presented that aims at an optimal pickup of ambient sound for 3D Audio. The ORTF-3D is a discrete 8ch setup that can be routed to the channels of a 3D Stereo format such as Dolby Atmos or Auro3D. It is also ideally suited for immersive sound formats such as wavefield synthesis or VR/Binaural, as it creates a complex 3D ambience that can be mixed or binauralized. The ORTF-3D setup was developed on the basis of stereophonic rules. It creates an optimal directional image in all directions as well as a high spatial sound quality due to highly uncorrelated signals in the diffuse sound. Reports from sound engineers affirm that it creates a highly immersive sound in a large listening area and still is compact and practical to use.
Convention Paper 9869

11:30 am

P6-6 Apparent Sound Source De-Elevation Using Digital Filters Based on Human Sound Localization—*Adrian Celestinos, Elisabeth McMullin, Ritesh Banka, William Decanio, Allan Devantier*, Samsung Research America, Valencia, CA, USA

The possibility of creating an apparent sound source elevated or de-elevated from its current physical location is presented in this study. For situations where loudspeakers need to be placed in different locations than the ideal placement for accurate sound reproduction digital filters are created and connected in the audio reproduction chain either to elevate or de-elevate the perceived sound from its physical location. The filters are based on head related transfer functions (HRTF) measured in human subjects. The filters relate to the average head, ears, and torso transfer functions of humans isolating the effect of elevation/de-elevation only. Preliminary tests in a movie theater setup indicate that apparent de-elevation can be achieved perceiving about –20 degrees from its physical location.
Convention Paper 9870

Session P7 **Thursday, Oct. 19**
9:00 am– 11:00 am **Room 1E12**

PERCEPTION—PART 2

Chair: **Dan Mapes-Riordan**, Etymotic Research, Elk Grove Village, IL, USA

9:00 am

P7-1 A Statistical Model that Predicts Listeners' Preference Ratings of In-Ear Headphones: Part 1—Listening Test Results and Acoustic Measurements—*Sean Olive, Todd Welti, Omid Khonsaripour*, Harman International, Northridge, CA, USA

A series of controlled listening tests were conducted on 30 different models of in-ear (IE) headphones to measure their relative sound quality. A total of 71 listeners both trained and untrained rated the headphones on a 100-point preference scale using a multiple stimulus method with a hidden reference and low anchor. A virtual headphone test method was used wherein each headphone was simulated over a high-quality replicator headphone equalized to match their measured magnitude response. Leakage was monitored and eliminated for each subject. The results revealed both trained and untrained listeners preferred the hidden reference, which was the replicator headphone equalized to our new IE headphone target response curve. The further the other headphones deviated from the target response, the less they were preferred. Part two of this paper develops a statistical model that predicts the headphone preference ratings based on their acoustic measurements.
Convention Paper 9840

9:30 am

P7-2 Perceptual Assessment of Headphone Distortion—*Louis Fielder*, Dolby Laboratories, Inc., San Francisco, CA, USA

A perceptually-driven distortion metric for headphones is proposed that is based on a critical-band spectral comparison of the distortion and noise to an appropriate masked threshold, when the headphone is excited by a sine wave signal. Additionally, new headphone-based masking curves for 20, 50, 100, 200, 315, 400, and 500 Hz sine waves are derived by subjective tests using bands of narrow-band noise being masked by a sine wave signal. The ratios of measured distortion and noise levels in critical bands over the appropriate masking curve values are compared, with the critical bands starting at the second harmonic. Once this is done the audibility of all these contributions are combined into a single audibility value. Extension to loudspeaker measurements is briefly discussed.
Convention Paper 9841

10:00 am

P7-3 The Adjustment / Satisfaction Test (A/ST) for the Subjective Evaluation of Dialogue Enhancement—
Matteo Torcoli,[1] *Jürgen Herre,*[1,2] *Jouni Paulus,*[1,2] *Christian Uhle,*[1,2] *Harald Fuchs,*[1] *Oliver Hellmuth*[1]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories Erlangen, Erlangen, Germany

Media consumption is heading towards high degrees of content personalization. It is thus crucial to assess the perceptual performance of personalized media delivery. This work proposes the Adjustment/Satisfaction Test (A/ST), a perceptual test where subjects interact with a user-adjustable system and their adjustment preferences and the resulting satisfaction levels are studied. We employ the A/ST to evaluate an object-based audio syst em that enables the personalization of the balance between dialogue and background, i.e., a Dialogue Enhancement system. Both the case in which the original audio objects are readily available and the case in which they are estimated by blind source separation are compared. Personalization is extensively used, resulting in clearly increased satisfaction, even in the case with blind source separation.
Convention Paper 9842

10:30 am

P7-4 Automatic Text Clustering for Audio Attribute Elicitation Experiment Responses—*Jon Francombe, Tim Brookes, Russell Mason*, University of Surrey, Guildford, Surrey, UK

Collection of text data is an integral part of descriptive analysis, a method commonly used in audio quality evaluation experiments. Where large text data sets will be presented to a panel of human assessors (e.g., to group responses that have the same meaning), it is desirable to reduce redundancy as much as possible in advance. Text clustering algorithms have been used to achieve such a reduction. A text clustering algorithm was tested on a dataset for which manual annotation by two experts was also collected. The comparison between the manual annotations and automatically-generated clusters enabled evaluation of the algorithm. While the algorithm could not match human performance, it could produce a similar grouping with a significant redundancy reduction (approximately 48%).
Convention Paper 9843

Audio for Cinema 4 **Thursday, October 19**
9:00 am – 10:30 am **Room 1E07**

PRODUCTION SOUND: CURRENT TRENDS AND PROVEN TRADITIONS

Presenter: **Glen Trew**, CAS

As audio technology advances rapidly, sound quality can improve as a result, but not always. Based on his 40 years as a production sound mixer for film and video, and owner of Trew Audio, Glen Trew will discuss the new technology and techniques of production dialog and music on the set, and when some long-established traditions are preferred. Choices regarding mixing and tracking, and boom mics vs. body mics, and music lip-sync vs. recording live on the set will be covered, followed by Q&A.

Co-organized with Cinema Audio Society (CAS). This session is presented in association with the AES Technical Committee on Audio for Cinema

Game Audio & VR 4 **Thursday, October 19**
9:00 am – 10:00 am **Room 1E13**

GATEWAYS TO THE EAST — MUSIC AND RECORDING TECHNIQUES FOR CHINESE VIDEO GAMES

Presenter: **Fei Yu**, Dream Studio, Los Angeles/ Beijing

As the Music Supervisor for NetEase online game Revelation, Fei Yu would love to discuss the challenge and triumphs of communicating and collaborating from across the world, the technology of recording for Chinese- Western style music, and the creative process, strategies for working with ethnic soloists to create authentic world sound and recording strategies for combining orchestra with soloists. The soundtrack became one of the first game scores to be released by the prestigious film music label Varese Sarabande and was Winner of the BSOSpirit Jerry Goldsmith Award and the Tracksounds Genius Choice Vote for Best Score: Video Game. Nominated for an International Film Music Critics Association Award, A Tracksounds Cue Award, A Reel Music Award, A Game Audio Network Guild Award and named one of the top ten scores of 2015 (game and film) by Cinematic Sound Radio.

Product Development 3 **Thursday, October 19**
9:00 am – 10:30 am **Room 1E09**

WHAT HAPPENS IN A PATENT LAWSUIT?

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

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

Tutorial/Workshop 4 **Thursday, October 19**
9:00 am – 10:15 am **Room 1E08**

PODCASTS: TELLING STORIES WITH SOUND

Chair: **Mary Nichols**, FuseBox Radio, Waldorf, MD USA

Panelists: *Jim Anderson*, Anderson Audio NY, New York, NY, USA; New York University, New York, NY, USA
Jeanne Montalvo Lucar, Audio Engineer, Producer

Podcasting is now the latest and greatest frontier for freedom of expression and the press with distribution networks that most of the average creator can gain equal access to with a (mostly) low cost of entry. This event is a jump off point about the many ways one can create, record, host, and distribute a podcast with high audio quality with the latest technologies out in the world today.

Thursday, October 19 **9:00 am** **Room 1C03**

Technical Committee Meeting on Archiving, Restoration, and Digital Libraries

Archiving/Restoration 4 **Thursday, October 19**
9:30 am – 11:00 am **Room 1E15/16**

SPECIAL EVENT: BEARING WITNESS: THE MUSIC OF STAR WARS—ARCHIVING ART AND TECHNOLOGY

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

When Sony Classics requested original vinyl masters for a new release of Star Wars soundtracks that started us on a mission to archive and preserve all the music of Star Wars. A tour of the Lucasfilm archives made us realize we were looking at a 40-year history of film scoring and at some point in the near future some of this material could not be played or accessed for reuse. We will discuss the format choice, transfer process, database creation, and editing process. The goal? .... how to future-proof accessibility of some of the greatest film music of all time.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Game Audio & VR 11 **Thursday, October 20**
9:30 am – 10:30 am **Room 1E10**

CREATING INTERACTIVE, REACTIVE, AND GENERATIVE MUSIC WITH PURE DATA

Presenter: **Yuli Levtov**, Reactify, London, UK

Since the advent of libpd, Pure Data, Max's open-source and slightly more "bare-bones" cousin, has found its way into many embedded production applications, from mobile apps to Web and VR experiences. Furthermore, bringing procedural and interactive music to platforms such as Unity and the Web has recently been made even more practical by a new and powerful tool, Heavy, which converts Pure Data patches into highly-performant code that can run on many different platforms. Reactify is a London-based creative production company focusing on the possibilities of interactive, reactive, and generative music, with Pure Data sitting at the core of almost all of its projects. In this presentation, co-founder Yuli Levtov will give an overview of how Pure Data can be used to interface with physical or virtual reality in creative ways, and provide a platform for collaboration between developers and artists, and how Heavy can close the gap between a rapid-prototyping tool and efficient production environment.

Recording & Production 4 **Thursday, October 19**
9:30 am – 10:30 am **Room 1E14**

METALLIANCE - YESTERDAY, TODAY AND TOMORROW: WHERE WE'VE BEEN, WHERE WE ARE, AND WHERE WE'RE GOING

Presenters: **Chuck Ainlay**, METAlliance, Nashville, TN, USA
Ed Cherney, Edward Cherney Company, Venice, CA, USA
Frank Filipetti
George Massenburg, Schulich School of Music, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Elliot Scheiner
Al Schmitt, Los Angeles, CA, USA

Established in 2005 by globally-recognized, award-winning audio engineers and producers Chuck Ainlay, Ed Cherney, Frank Filipetti, George Massenburg, Elliot Scheiner, Al Schmitt, and the late Phil Ramone, the METAlliance is a collaborative community that fosters relationships between producers, engineers, and manufacturers in order to ensure the highest standards of audio production. This group, which was deeply involved in establishing foundational music recording techniques and technical standards, has now turned its attention to providing education and inspiration to music creators while promoting excellence in engineering and production.

Accordingly, in conjunction with Hal Leonard, the METAlliance has begun publishing the collective knowledge amassed by these A-list producers and engineers under the METAlliance Academy brand. The product line includes A/V tutorial courses, print and ebooks, Power Learning Digital Print versions of these books, and in-person workshop events.

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

Spatial Audio 17 **Thursday, October 19**
9:30 am – 11:30 am **Room 1E06**

PMC: A 9.1 MUSICAL EXPERIENCE

Presenters: **Morten Lindberg**
Daniel Shores

[abtract unavailable]

Thursday, October 19 **10:00 am** **Room 1C03**

Technical Committee Meeting on Signal Processing

Software@AES 16 **Thursday, October 19**
10:00 am – 10:20 am **Stage 3**

BEST SERVICE

Broadcast/Media Streaming 6 **Thursday, October 19**
10:15 am – 12:15 pm **Room 1E08**

CASE STUDY—USING THE RIGHT WIRE FOR THE RIGHT JOB

Presenters: **Steve Lampen**, Belden, San Francisco, CA, USA
John Schmidt, Consultant

This presentation will focus on one or two major projects. The huge range of wire and cable in each will be outlined and detailed. The choice of one type of cable over another will also be addressed. The audience is encouraged to participate in the discussion of appropriate and inappropriate choices and applications for various cable types.

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

Broadway Sound Expo 01 **Thursday, October 19**
10:30 am – 11:15 am **Stage 1**

THEATRICAL VOCAL MIKING

Presenters: **Dominic Sack**, Sound Associates, Inc.
Karl Winkler, Lectrosonics, Rio Rancho, NM, USA

For theatrical applications, especially musicals, placement of the lav mic is critical for success. This workshop will cover the physiological and acoustic issues associated with this challenging subject. Once the lav mic is placed, next comes concealment to enhance the visual aspects of the performance. Tips, tricks and tools for this purpose will be covered.

Project Studio Expo 07 **Thursday, October 19**
10:30 am – 12:00 noon **Stage 2**

FINESSE THE MESS: MAXIMIZING YOUR CREATIVE PROCESS

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

We have best practices for miking techniques and mixing, and tips and hacks for using every piece of gear in the studio. However, the most essential item in every studio Is one that we rarely talk about: your brain. If you have struggled with "writers block," agonized over the final "finishing touches" of a project or had a conflict with a co-writer or collaborator this event is for you. Our group of creatives will discuss best practices for collecting, cultivating and manipulating ideas and provide a framework for expanding your personal creativity. In addition, we hope offer some tools for working with collaborators and navigating the creative process.

Software@AES 17 **Thursday, October 19**
10:30 am – 10:50 am **Stage 3**

SONARWORKS

Game Audio& VR 5 **Thursday, October 19**
10:45 am – 12:15 pm **Room 1E13**

STAR TREK: BRIDGE CREW VR—AUDIO POST MORTEM

Presenters: **Justin Drust**, Red Storm Entertainment, Cary, NC, USA
Matthew McCallus, Expert Sound Programmer, Red Storm Entertainment

Look, listen, and learn from the Red Storm Audio Team behind Star Trek: Bridge Crew as they show us new frontiers in VR game audio development from multiple perspectives—the Audio Director, Audio Programmer and others. Discover the tips, tricks, and techniques of the audio design process from conception to completion.

Product Development 4 **Thursday, October 19**
10:45 am – 12:15 pm **Room 1E09**

NEW AMPLIFIER REQUIREMENTS FOR SPEAKER PROTECTION AND CONTROL

Presenter: **Joachim Schlechter**

The rising demand for active speaker protection across all audio applications creates new challenges on the audio amplifier design. The amplifier is transforming more and more into a "smart" system integrating sensing capabilities, digital interfaces, and internal signal processing. To push nonlinear "green" loudspeakers in small enclosures to their physical limits, a higher peak power from the amplifier is desirable. At the same time, there is a growing need for higher overall efficiency and minimal battery power consumption. This tutorial is discussing current trends and coming requirements in amplifier design in conjunction with speaker protection and control.

Sound Reinforcement 5 **Thursday, October 19**
10:45 am – 12:15 pm **Room 1E07**

CORPORATE SOUND DESIGN

Moderator: **Lee Kalish**

Sound for corporate events can be lucrative but can also be very demanding. Complex matrixing or other unusual solutions may be required in signal routing to loudspeaker zones, recording devices, distant participants and web streaming. Amplifying lavalier mics strongly into a loudspeaker system is its own art. Client relations are of top importance. A discussion of how these factors shape our differing approaches to corporate sound systems.

Tutorial/Workshop 5 **Thursday, October 19**
10:45 am – 12:15 pm **Room 1E14**

NEW DEVELOPMENTS IN LISTENING TEST DESIGN

Chair: **Brecht De Man**, Queen Mary University of London, London, UK

Panelists: *Jan Berg*, Luleå University of Technology, Piteå, Sweden
Jürgen Herre, International Audio Laboratories Erlangen, Erlangen, Germany; Fraunhofer IIS, Erlangen, Germany
Todd Welti, Harman International Inc., Northridge, CA, USA

Listening tests are a key component in a wide range of audio research and development, from loudspeaker construction over source separation algorithms to emotion in music. New graphical user interfaces, transducer virtualization, and online tests have helped make the once tedious and expensive practice of perceptual evaluation of audio more accessible and efficient. However, these developments each bring their own challenges. Furthermore, while topics like audio codec design have an established set of practices, other types of evaluation are only slowly being standardized, if at all, or borrow from neigboring fields. In this workshop, some of the field's most prominent experts contribute different perspectives on advancements in the area of perceptual evaluation of audio, and offer their view on its future.

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

Student Events/Career Development
EC3: SPARS MENTORING
Thursday, October 19, 10:45 am – 12:45 pm
Open Area

This event is specially suited for students, recent graduates, young professionals, and those interested in career advice. Hosted by SPARS in cooperation with the AES Education Committee, career related Q&A sessions will be offered to participants in a speed group mentoring format. A dozen students will interact with 4–5 working professionals in specific audio engineering fields or categories every 20 minutes. Audio engineering fields/categories include gaming, live sound/live recording, audio manufacturer, mastering, sound for picture, and studio production. Mentors include: *Trevor Fletcher, Mike Mazzotta, Gina Zdanowicz, Matthew Rifino, Chuck Zwicky, Joel Hamilton, Tom Salta, Barry Cleveland, Anthony Schultz, David Amlen, Chris Mara, Leslie Mona-Mathus, Jamie Baker, Karrie Keyes, Leslie Ann Jones, Dann Michael Thompson.*

Audio for Cinema 5 **Thursday, October 19**
10:45 am – 12:15 pm **Room 1E10**

**LOUDNESS ISSUES IN CINEMA—
IS THE REFERENCE LOST?**

Chair: **Eelco Grimm**, HKU University of the Arts, Utrecht, Netherlands; Grimm Audio, Eindhoven, The Netherlands

Panelists: *John Fithian*, NATO
Tom Fleischman, CAS
Charles Q. Robinson, Dolby Laboratories, San Francisco, CA, USA

Better sound systems may reproduce louder movies. Louder movies generate audience complaints. Complaints press cinema owners to lower the playback level. Lower playback levels make dialogue intelligible. Dubbing stages are pressed to print louder.

Is the "sacred" reference level lost? This workshop will discuss the current issues with loudness in movies, with points-of-view from mixers, exhibitors, and technology providers.

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

Session P8 **Thursday, Oct. 19**
11:00 am – 12:30 pm **Foyer**

POSTERS: SIGNAL PROCESSING

11:00 am

P8-1 A Simplified 2-Layer Text-Dependent Speaker Authentication System—*Giacomo Valenti*,[1,2] *Adrien Daniel*,[1] *Nicholas Evans*[2]
[1]NXP Software, Mougins, France
[2]EURECOM, Biot, France

This paper describes a variation of the well-known HiLAM approach to speaker authentication that enables reliable text-dependent speaker recognition with short-duration enrollment. The modifications introduced in this system eliminate the need for an intermediate text-independent speaker model. While the simplified system is admittedly a modest modification to the original work, it delivers comparable levels of automatic speaker verification performance while requiring 97% less speaker enrollment data. Such a significant reduction in enrollment data improves usability and supports speaker authentication for smart device and Internet of Things applications.
Convention Paper 9844

11:00 am

P8-2 Binaural Sound Source Separation Based

on Directional Power Spectral Densities—
Joel Augusto Luft,[1,2] *Fabio I. Pereira*,[2] *Altamiro Susin*[2]
[1]Instituto Federal de Educação, Ciência e Tecnologia do Rio Grande do Sul, Canoas, RS, Brazil
[2]Universidade Federal do Rio Grande do Sul, Porto Alegre, RS, Brazil

Microphone arrays are a common choice to be used in spatial sound source separation. In this paper a new method for binaural source separation is presented. The separation is performed using the spatial position of sound source, the Head-Related Transfer Function, and the Power Spectral Density of fixed beamformers. A non-negative constrained least-squares minimization approach is used to solve the Head-Related Transfer Function based directivity gain formulation and the Power Spectral Density is used as a magnitude estimation of the sound sources. Simulation examples are presented to demonstrate the performance of the proposed algorithm.
Convention Paper 9845

11:00 am

P8-3 Improving Neural Net Auto Encoders for Music Synthesis—*Joseph Colonel, Christopher Curro, Sam Keene*, The Cooper Union for the Advancement of Science and Art, New York, NY, USA

We present a novel architecture for a synthesizer based on an autoencoder that compresses and reconstructs magnitude short time Fourier transform frames. This architecture outperforms previous topologies by using improved regularization, employing several activation functions, creating a focused training corpus, and implementing the Adam learning method. By multiplying gains to the hidden layer, users can alter the autoencoder's output, which opens up a palette of sounds unavailable to additive/subtractive synthesizers. Furthermore, our architecture can be quickly re-trained on any sound domain, making it flexible for music synthesis applications. Samples of the autoencoder's outputs can be found at http://soundcloud.com/ann_synth, and the code used to generate and train the autoencoder is open source, hosted at http://github.com/JTColonel/ann_synth.
Convention Paper 9846

11:00 am

P8-4 Comparative Study of Self-Organizing Maps vs Subjective Evaluation of Quality of Allophone Pronunciation for Non-Native English Speakers—
Bozena Kostek,[1] *Magdalena Piotrowska*,[1] *Tomasz Ciszewski*,[2] *Andrzej Czyzewski*[1]
[1]Gdansk University of Technology, Gdansk, Poland
[2]University of Gdansk, Gdansk, Poland

The purpose of this study was to apply Self-Organizing Maps to differentiate between the correct and the incorrect allophone pronunciations and to compare the results with subjective evaluation. Recordings of a list of target words, containing selected allophones of English plosive consonants, the velar nasal and the lateral consonant, were made twice. First, the target words were read from the list by nine non-native speakers and then repeated after a phonology expert's recorded sample. Afterwards, two recorded signal sets were segmented into allophones and parameterized. For that purpose, a set of descriptors, commonly employed in music information retrieval, was utilized to determine whether they are effective in allophone analysis. The phonology expert's task was to evaluate the pronunciation accuracy of each uttered allophone. Extracted feature vectors along with the assigned ratings were applied to SOMs.
Convention Paper 9847

11:00 am

P8-5 Automatic Masking Reduction in Balance Mixes Using Evolutionary Computing—*Nicholas Jillings, Ryan Stables*, Birmingham City University, Birmingham, UK

Music production is a highly subjective task, which can be difficult to automate. Simple session structures can quickly expose complex mathematical tasks which are difficult to optimize. This paper presents a method for the reduction of masking in an unknown mix using genetic programming. The model uses results from a series of listening tests to guide its cost function. The program then returns a vector that best minimizes this cost. The paper explains the limitations of using such a method for audio as well as validating the results.
Convention Paper 9813

Thursday, October 19 11:00 am Room 1C03

Technical Committee Meeting on Fiber Optics for Audio

Software@AES 18 Thursday, October 19
11:00 am – 11:20 am Stage 3

CELEMONY (MELODYNE)

Session EB2 Thursday, Oct. 19
11:15 am – 12:30 pm Room 1E12

RECORDING & PRODUCTION

Chair: **Palmyra Catravas**, Union College, Schenectady, NY, USA

11:15 am

EB2-1 Engineered Remote-Sensing Audio Power Amplifier for High-Fidelity Applications—*Peter Horowitz* Fourth Dimension Engineering, Columbia, MD, USA

The objective of this work is to minimize the deleterious effects of loudspeaker cable impedance when driving dynamic loudspeakers, accomplished primarily with a mathematical feedback analysis on the prominent role of the cables themselves within the audio baseband feedback loop. Presented are the measured system waveforms, along with computed root loci and transfer functions of a proof-of-principle remote-sensing (4-wire) 80 watt audio power amplifier. A single baseband feedback loop compares the incoming audio information (voltage) to the resultant voltage across the loudspeaker electrical terminals and minimizes the difference. Measured waveforms demonstrate notably superior replication of incoming information at the loudspeaker terminals over the audio band. The system is empirically robust for a wide range of dynamic loudspeaker and cable systems without any need for electronic adjustment. For example, with 35 meter 15/22 gauge cabling, a bandwidth of 72kHz, dynamic range of 110dB, phase linearity of <0.5°, and low impedance drive levels of <0.2Ω at the loudspeaker terminals are readily achieved simultaneously.
Engineering Brief 364

11:30 am

EB2-2 Building a Globally Distributed Recording Studio— *John Fiorello*, RecordME, Torrington, CT, USA

The internet has played a significant role in changing consumer behavior in regards to the distribution and consumption of music. Record labels, recording studios, and musicians have felt the financial squeeze as physical media delivery has been depreciated. However, the internet also enables these studios, musicians, and record labels to re-orient their business model to take advantage of new content creation and distribution. By developing a hardware appliance that combines high-resolution audio recording and broadcasting with real-time, two-way video communication across the web, we can expand the geographic area that studios can serve, increase revenue for musicians, and change the value proposition traditional record labels have to offer.
Engineering Brief 365

11:45 am

EB2-3 Simultaneous Audio Capture at Multiple Sample Rates and Formats for Direct Comparison and Evaluation— *Jordan Strum,*[1] ; *Richard King,*[2,3] *Oles Protsidym,*[1] *Ieronim Catanescu* [2]
[1] ProStudioMasters, Montreal, QC, Canada
[2] McGill University, Montreal, Quebec, Canada
[3] The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

In order to evaluate differences among recording formats and resolutions over a variety of classical, jazz, and popular musical material, a unique collection of audio assets was recorded. Live performances were captured using a single pair of microphones, positionally adjusted for each sound source. Preamplifier outputs were routed to 11 identical recording interfaces capturing various PCM and DSD formats simultaneously with 3 analogue tape recorders, the contents of which were then transferred to the above digital formats. These assets will be used to compare differences among recording formats and resolutions using identical performances, and to provide source m aterial for listening tests as well as further research. The design and execution of this project will be discussed.
Engineering Brief 366

12:00 noon

EB2-4 Undergraduate Curricular Development at the Electrical Engineering/Music Interface at Union College—*Palmyra Catravas*, Union College, Schenectady, NY, USA

Curricular development at the interface of electrical engineering and music will be described, with a focus on the pedagogical use of audio and acoustics to reinforce basic fundamentals in electrical engineering. The effort, which has taken place over more than a decade, seeks to reinforce the foundation provided by the traditional, rigorous engineering curriculum at Union College, an undergraduate liberal arts college in upstate NY. A related specialized research laboratory – Phasor Lab – is located in the Peter Irving Wold Science and Engineering Center at Union.
Engineering Brief 367

12:15 pm

EB2-5 Recording, Mixing and Mastering of Audio Using a Single Microphone Array and Audio Source Separation Algorithms—*Jakub Zamojski, Piotr Makaruk, Lukasz Januszkiewicz, Tomasz Zernicki*, Zylia sp. z o.o., Poznan, Poland

The authors present a new way of recording of music bands using a single microphone array and audio source separation algorithms. In contrast to the traditional recording process, this novel approach allows for capturing all of musical instruments simultaneously using only one recording device, avoiding a multiple sets of spot microphones, cables, and D/A converters. Moreover, using a single microphone array and dedicated algorithms gives the sound engineer a unique set of "audio processing tools" that can be used in the post-production stage. This paper describes step-by-step a recording process of a music band playing ukulele using a 19-capsules spherical microphone array and dedicated software. The process includes the following stages: recording, sound source separation, mixing, and mastering.
Engineering Brief 368

Broadway Sound Expo 02 — Thursday, October 19
11:15 am – 12:00 noon — Stage 1

THEATRICAL WIRELESS 3.0

Operating high-count multichannel wireless systems in the high-density RF environments of the future will require the application.

Software@AES 19 — Thursday, October 19
11:30 am – 11:50 am — Stage 3

FABFILTER

Student Events/Career Development
EC4: STUDENT RECORDING CRITIQUES
Thursday, October 19, 12:00 pm – 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.

Thursday, October 19 — 12:00 noon — Room 1C03

Technical Committee Meeting on Broadcast and Online Delivery

Broadway Sound Expo 03 **Thursday, October 19**
12:00 noon – 12:45 pm **Stage 1**

ANATOMY OF THEATER SOUND DESIGN

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

Based in New York, Jesse Stevens comes to L-Acoustics with 15 years of experience as a sound designer, engineer, and mixer for Broadway musicals, national tours and large-scale special events. He was an integral member of the sound design team for the Broadway productions of Rock of Ages, Motown the Musical, and After Midnight, all of which were nominated for a Tony Award for Outstanding Sound Design. Additionally, he's spent time with Blue Man Group at the Monte Carlo Las Vegas, the Radio City Spring Spectacular, and Waitress on Broadway.

Software@AES 20 **Thursday, October 19**
12:00 noon – 12:20 pm **Stage 3**

ANTARES

Special Events
SE4: LUNCHTIME KEYNOTE—ALEX CASE: AFTER THE LOUDNESS WARS—GET "LOUDER" WITHOUT THE FADER
Thursday, October 19, 12:30 pm – 1:15 pm
Room 1E15/16

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

Take this mixing challenge: place the faders at the lowest levels that make technical and musical sense. Try to make critical tracks easier to hear through more sophisticated means than simple fader rides. Turning it up is too easy and mucks up the rest of the mix. One loud track can make it harder to hear the other tracks in your mix. To get a track to rise up out of a crowded mix, leverage the strategic application of panning, ambience, reverb, compression, gating, EQ, distortion, and other unmasking effects. Tailoring non-fader properties of the sound to keep the track audible is more work and requires mastery of many subtle aspects of signal processing for mixing. However, resisting that urge to simply push up the fader means the other tracks won't have to play catchup.

Thursday, October 19 **12:30 pm** **Room 1C03**

Technical Committee Meeting on Audio for Games

Software@AES 21 **Thursday, October 19**
12:30 pm – 12:50 pm **Stage 3**

BLUE CAT AUDIO

Student Events/Career Development
EC5: SAUL WALKER STUDENT DESIGN EXHIBITION
Thursday, October 19, 1:00 pm – 3:00 pm
Poster Area

All accepted entries to the AES Saul Walker Student Design Competition are given the opportunity to show off their designs at this poster/tabletop exhibition. The session is free and open to all convention attendees and is an opportunity for aspiring student hardware and software engineers to have their projects seen by the AES design community. It is an invaluable career-building event and a great place for companies to identify their next employees. Students from both audio and non-audio backgrounds are encouraged to participate. Few restrictions are placed on the nature of the projects, which may include loudspeaker designs, DSP plug-ins, analog hardware, signal analysis tools, mobile applications, and sound synthesis devices. Attendees will observe new, original ideas implemented in working-model prototypes.

Broadway Sound Expo 04 **Thursday, October 19**
1:00 pm – 2:00 pm **Stage 1**

THEATER SOUND INFRASTRUCTURE

Presenters: **Mark Brunke**, Optocore
Maciek Janiszewski, Optocore
Vinnie Macri, Clear-Com

Theater system integration is moving to the highest level. Manufacturers offer now combined solution for audio, video, intercoms, lights and control. This merge requires new technologies to fulfill high theater requirements like the lowest latency, redundancy, flexibility and control. It also requires multiple different worlds to connect - audio with video, audio with control, multi-vendor cooperation. The session uncovers possible challenges in the theater sound infrastructure design and explains how the market leaders deal with it.

Project Studio Expo 08 **Thursday, October 19**
1:00 pm – 1:45 pm **Stage 2**

MONITORING WORKSHOP

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

Reliable monitoring is the key to transforming your creative ideas to recordings that deliver your artistic intensions in all environments. Meet Aki Mäkivirta of Genelec and hear about the essentials of setting up monitoring that delivers in your room. Get an overview of the typical pitfalls and how to avoid them. From stereo to 3D immersive monitoring, systematic approach to setting up and calibrating monitoring systems pays off by making your work faster enabling you to concentrate on the essential – creating the content.

Session P10 **Thursday, Oct. 19**
1:30 pm – 3:30 pm **Room 1E12**

TRANSDUCERS—PART 2

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

1:30 pm

P10-1 Loudspeakers as Microphones for Infrasound—*John Vanderkooy*, University of Waterloo, Waterloo, ON, Canada

This paper shows that a sealed-box loudspeaker can be used as the sensor for a very high-performance infrasound microphone. Since the cone displacement essentially responds directly to infrasound pressure, the velocity-induced loudspeaker output must be electronically integrated to give a flat response. The undamped resonance peak of the loudspeaker is avoided by feeding the short-circuit voice coil current into the virtual ground input of an integrator op-amp. Design equations are given and a complete response analysis is presented. A prototype is compared with a conventional microphone used for infrasound measurement, showing the improved performance of the sealed-box loudspeaker design.
Convention Paper 9856

2:00 pm

P10-2 A Low-Cost, High-Quality MEMS Ambisonic Microphone—*Gabriel Zalles, Yigal Kamel, Ian Anderson, Ming Yang Lee, Chris Neil, Monique Henry, Spencer Cappiello, Charlie Mydlarz, Melody Baglione, Agnieszka Roginska*, New York University, New York, NY, USA; The Cooper Union for the Advanement of Science and Art, New York, NY, USA

While public interest for technologies that produce and deliver immersive VR content has been growing the price point for these tools has remained relatively high. This paper presents a low-cost, high-quality first-order ambisonics (FOA) microphone based on low-noise microelectromechanical systems (MEMS). This paper details the design, fabrication, and testing of a MEMS FOA microphone including its frequency and directivity response. To facilitate high resolution directivity response measurements a low-cost, automatic rotating microphone mount using an Arduino was designed. The automatic control of this platform was integrated into an in-house acoustic measurement library built in MATLAB allowing the user to generate polar plots at resolutions down to 1.8°. Subjective assessments compared the FOA mic prototype to commercially available FOA solutions at higher price points.
Convention Paper 9857

2:30 pm

P10-3 Automated Auditory Monitoring Using Evoked Potentials and Consumer Headphones—*Thomas Rouse, Loek Janssen*, Plextek, Great Chesterford, UK

Auditory Evoked Potentials (AEP) are electrical signals resulting from activity in the auditory system in response to stimuli. The characteristic waveforms can be indicative of cochlea and auditory brainstem function and may change after the onset of tinnitus or hearing threshold shifts, whether permanent or temporary. AEP measurement is currently used by audiologists for hearing assessment in infants and to aid the diagnosis of some diseases. Measurements were made using a variety of consumer headphones and integrated electrodes and compared with a reference audiology system. The results showed the ability to record a consistent response and indicated that AEPs can be reliably measured outside a clinical environment. This could be used to automatically monitor for changes in a user's hearing.
Convention Paper 9858

3:00 pm

P10-4 A Digital Class D Audio Amplifier with Pulse Density Modulation and Distortion Suppression Feedback Loop—*Robert McKenzie,*[1] *Xinchang Li,*[2] *Martin Snelgrove,*[3] *Wai Tung Ng*[1]
[1]University of Toronto, Toronto, ON, Canada
[2]Graduate University of the Chinese Academy of Sciences, Beijing, China
[3]Kapik Integration, Toronto, ON, Canada;

A novel fully digital Class D amplifier is presented in which the output stage error is digitized by a 10-bit ADC and fed back into the modulation path to suppress distortion. This technique attenuates the in-band noise introduced by the output stage, and can tolerate large latency. A fully digital Pulse Density Modulation (PDM) Class D amplifier with output stage noise shaping is implemented on a PCB prototype. Feedback loop functionality is verified experimentally, and a 10 dB improvement in Total Harmonic Distortion plus Noise (THD+N) is realized.
Convention Paper 9859

Archiving/Restoration 5 — **Thursday, October 19**
1:30 pm – 3:00 pm — **Room 1E14**

RESTORATION AUDIO: PRESERVATION OF YOUR ASSETS TODAY FOR TOMORROW

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

Panelists: *Bob Clearmountain*
Kelly Pribble
Andrew Scheps
Andy Skurow

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

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Audio for Cinema 6 — **Thursday, October 19**
1:30 pm – 3:00 pm — **Room 1E07**

DIALOG IN FILM AND TELEVISION

Presenters: **Bobby Johanson**, "Avengers," "Bridge of Spies"
Peter Schneider, Gotham Sound and Communications, Inc., Long Island City, NY, USA; Tisch School of the Arts, New York, NY, USA
Alexa Zimmerman

Storytelling is the main goal of movies, and storytelling is mainly done with dialogue. This session will focus on the work done with dialogue, from production sound to post production. Because the lack of dialogue intelligibility is the best way to ruin a movie.

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

Broadcast/Streaming Media 7 — **Thursday, October 19**
1:30 pm – 4:15 pm — **Room 1E15/16**

SPECIAL EVENT: AUDIO FOR ADVANCED VIDEO BROADCAST

Presenter: **Fred Willard**

Exciting and ground-breaking progress has been achieved over the past year in combining the next generation audio and video technologies into compelling and remarkable experiences to the consumer. Sports broadcasting has led the charge in parallel concurrent development of delivery infrastructure through MPEG-H / ATSC 3.0 for the 2018 Olympics in Korea, next year's world cup in Moscow, AC-4 / 3.0 in the States, and 8K / 22.2 in Japan for the 2020 games. Covering live capture, post, metadata handling, and transmission and delivery, we present to you the world's foremost experts in advanced audio for broadcast and streaming. Don't miss this popular session and the latest developments as standards are now becoming product and reality.

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

Game Audio & VR 6 **Thursday, October 19**
1:30 pm – 3:00 pm **Room 1E13**

AR/VR: NEW YORK CITY INITIATIVES AND DIRECTIONS

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

Panelists: *Kai Falkenberg*, First Deputy Commissioner of the NYC Mayor's Office of Media and Entertainment
Justin Hendrix, NYC Media Lab
Erik Hockman, MSG Ventures
Jeremy Rowe, NYU

The session focuses on the latest initiatives and developments of Augmented and Virtual Reality research and development centers in NYC. Panelists will discuss, share, and give specific examples of some of the projects and directions currently explored in AR & VR, including initiatives from NYC's Mayor's office, the NYC Media Lab, the NYU Holodeck project, and Madison Square Garden Ventures.

Game Audio & VR 7 **Thursday, October 19**
1:30 pm – 3:00 pm **Room 1E06**

SPATIAL SOUND FOR CONSOLE AND PC GAMES

Presenters: **Robert Ridihalgh**, Microsoft
Scott Selfon, Microsoft, Redmond, WA, USA
Mark Yeend, Microsoft

In Spring 2017, Xbox One and Windows added native platform-level support for spatial sound: games and other applications could now implement audio above, below, and around the listener once, and transparently render to multiple endpoints (speakers, headphones) and formats (Dolby Atmos, Windows Sonic for Headphones), while continuing to support all playback devices and mixing of "legacy" non-spatial content. This talk covers some of the early learning that has emerged from these efforts in terms of: spatial sound design and music best practices for both mixed reality and screen-based scenarios; audio creation and mixing pipelines for games; how games are combining channel-based and dynamic sound mixes; performing listening quality assurance with an often dizzying array of output and downmixing options; and more. Throughout we'll highlight some impactful and common use cases as exercised by early titles, from both technical and aesthetic perspectives.

Product Development 5 **Thursday, October 19**
1:30 pm – 3:00 pm **Room 1E09**

LOUDSPEAKER AND AMPLIFIER POWER RATINGS: IS IT TIME TO START OVER?

Presenters: **Klas Dalbjörn**, Powersoft S.P.A.
Charles Hughes, Excelsior Audio, Gastonia, NC, USA; AFMG, Berlin, Germany
Claudio Lastrucci, Powersoft S.P.A.

What are the relevant requirements and how can we move forward to a unified way to understand and specify loudspeakers and amplifiers? A system designer will want to have specifications for loudspeakers and amplifiers (or for powered loudspeakers) that answer the questions: • How loud can I play with a given solution with the program material? • How much mains current will it draw and what heat dissipation will I have with my desired SPL? The currently used methods for specifying loudspeakers and amplifiers leave room for improvement when it comes to simplifying this. From the loudspeaker side we will discuss the "power" consumed by a loudspeaker and what determines this. We will present a newer method, based on the output response of a loudspeaker, to quantify the maximum input level. Also discussed will be peak input level capability of the loudspeaker. From the amplifier side we will look what the real world demand is for an amplifier. How do some design concepts compare when it comes to the compromises made to keep the costs down? We will look at some existing amplifier measurement methods; how do they relate to the actual demand and the achievable SPL when aligned with the existing loudspeaker measurements standards? Examples will be shown regarding flaws in only looking at voltage. Loudspeakers vary in how reactive they are and many modern amplifiers have the ability to only consume the real power needed (i.e., not turn the apparent power into heat losses). This suggests that a future proof amplifier and loudspeaker method considers real power usage. An attempt to propose a way forward will be presented, which will likely lead to an interesting discussion in the last part of this session.

Software@AES 22 **Thursday, October 19**
1:00 pm – 1:20 pm **Stage 3**

ACCUSONUS

Recording & Production 5 **Thursday, October 19**
1:30 pm – 3:00 pm **Room 1E08**

RAW TRACKS: MODERN POP PRODUCTION

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

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

Paul "Willie Green" Womack is part of the newer generation of engineer/producer/artists who also writes about and teaches the art of musical production. This wide-ranging conversation will include the challenges and possibilities of modern music production, integration of acoustic instrumentation with sampling and synthesis, and in-depth investigation into the multi-track masters of one of his fantastic productions.

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

Tutorial/Workshop 6 **Thursday, October 19**
1:30 pm – 3:00 pm **Room 1E10**

LEVELS, LOUDNESS, PERCEPTION, AND THE AESTHETICS OF MUSIC PRODUCTION AND DISTRIBUTION

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

Panelists: *Eelco Grimm*, HKU University of the Arts, Utrecht, Netherlands; Grimm Audio, Eindhoven, The Netherlands
Paul Tapper, Nugen Audio, UK
Robert Taylor, University of Newcastle, Callaghan, NSW, Australia

This collection of presentations addresses the changing paradigms of music delivery. It will offer observations about market driven forces, the influence of technological changes over the years on the aesthetics of music production and delivery to the consumer. Included is some definitive research regarding perception and loudness. Suggestions and advice about levels for music delivery will be covered.

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

Software@AES 23 **Thursday, October 19**
1:30 pm – 1:50 pm **Stage 3**

LINE 6

Session P9 **Thursday, Oct. 19**
2:00 pm – 6:00 pm **Room 1E11**

RECORDING AND PRODUCTION

Chair: **Stephen Roessner**, University of Rochester, Rochester, NY, USA

2:00 pm

P9-1 Analysis and Prediction of the Audio Feature Space when Mixing Raw Recordings into Individual Stems— *Marco A. Martinez Ramirez, Joshua D. Reiss*, Queen Mary University of London, London, UK

Processing individual stems from raw recordings is one of the first steps of multitrack audio mixing. In this work we explore which set of low-level audio features are sufficient to design a prediction model for this transformation. We extract a large set of audio features from bass, guitar, vocal, and keys raw recordings and stems. We show that a procedure based on random forests classifiers can lead us to reduce significantly the number of features and we use the selected audio features to train various multi-output regression models. Thus, we investigate stem processing as a content-based transformation, where the inherent content of raw recordings leads us to predict the change of feature values that occurred within the transformation.
Convention Paper 9848

2:30 pm

P9-2 The Beat Goes Static: A Tempo Analysis of U.S. Billboard Hot 100 #1 Songs from 1955–2015— *Stephen Roessner*, University of Rochester, Rochester, NY, USA

The Billboard Hot 100 is a rich source of information for tracking musical trends. Using available data analysis tools, we devised a method to accurately track tempo throughout a song. In this paper we demonstrate through an analysis of all number one songs from the chart that tempo variation within a song has declined over a 60-year period. In the 5-year span from 1955–1959, the average standard deviation of tempo was 5.01 beats perminute, or about 4.8%. Conversely, from 2010–2014, the average standard deviation was less than 1 beat per minute, or only about 0.85% of the average tempo.
Convention Paper 9849

3:00 pm

P9-3 An Even-Order Harmonics Control Technique for Analog Pedal Effector—*Kanako Takemoto, Shiori Oshimo, Toshihiko Hamasaki*, Hiroshima Institute of Technology, Hiroshima, Japan

The primary distortion mechanism of the analog guitar pedal effector is saturating nonlinearity of a transfer function, which consists of operational amplifier and diode clippers with filters. The output spectrum of this system shows odd-order harmonics primarily, but it also contains even-order harmonics. We found that the intensity of this even-order harmonic varies depending on the power supply voltage and clarified the mechanism by analysis of the internal circuit topology of the operational amplifier. The analysis was justified compared with the conventional single-ended transistor pedal operation. Based on the analysis we proposed a new concept of even harmonic control technique, which was applied for analog "Distortion" pedals and demonstrated distinguished experimental results with a prototype.
Convention Paper 9850

3:30 pm

P9-4 Unified Modeling for Series of Miniature Twin Triode Tube—*Shiori Oshimo, Kanako Takemoto, Toshihiko Hamasaki*, Hiroshima Institute of Technology, Hiroshima, Japan

Unification of high precision SPICE modeling for the series of MT vacuum tube has been succeeful for the first time. The model formula was validated based on the comparison of electrode and space physical dimensions among 12AX7, 12AU7, 12AY7, and 12AT7 associated with various aspects of properties not limited to the general Ip-Vp family curves. As a result, the non-linear behavior of the grid current and the plate current as a function of plus/minus grid voltage were able to be expressed entirely by 17 parameters of the newly proposed SPICE model, in which 4 tube type specific parameters and 4 universal parameters are constant and matching of twin valves of each tube as well as product dispersion are fitted by 9 variable parameters.
Convention Paper 9851

4:00 pm

P9-5 Virtual Analog Modeling of a UREI 1176LN Dynamic Range Control System—*Etienne Gerat, Felix Eichas*, *Udo Zölzer*, Helmut-Schmidt-University, Hamburg, Germany

This paper discusses an application of block-oriented modeling to a popular analog dynamic range compressor using iterative minimization. The reference device studied here is the UREI 1176LN, which has been widely used in music production and recording. A clone of the circuit built in a previous project has been used as a reference device to compare the results of the implementation. A parametric block-oriented model has been designed, improved, and tuned using the Levenberg-Marquardt iterative error minimization algorithm. Only input/output measurements have been performed following a gray-box modeling approach. Finally the model has been evaluated with objective scores and a listening test. This work led to very convincing modeling results.
Convention Paper 9852

4:30 pm

P9-6 Amplitude Panning and the Interior Pan—*Mark R. Thomas, Charles Q. Robinson*, Dolby Laboratories, San Francisco, CA, USA

The perception of source location using multi-loudspeaker amplitude panning is considered. While there exist many perceptual models for pairwise panning, relatively few studies consider the general multi-loudspeaker case. This paper evaluates panning scenarios in which a source is panned on the boundary or within the volume bounded by discrete loudspeakers, referred to as boundary and interior pans respectively. Listening results reveal the following: (1) pans to a single loudspeaker yield lowest localization error, (2) pairwise pans tend to be consistently localized closer to the listener than single loudspeaker pans, (3) largest errors occur when the virtual source is panned close to the listener, (4) interior pans are accurately perceived and, surprisingly, in some cases more accurately than pairwise pans.
Convention Paper 9853

5:00 pm

P9-7 Recording in a Virtual Acoustic Environment—*Jonathan S. Abel, Elliot K. Canfield-Dafilou*, Stanford University, Stanford, CA, USA

A method is presented for high-quality recording of voice and acoustic instruments in loudspeaker-generated virtual acoustics. Auralization systems typically employ close-mic'ing to avoid feedback, while classical recording methods prefer high-quality room microphones to capture the instruments integrated with the space. Popular music production records dry tracks, and applies reverberation after primary edits are complete. Here a hybrid approach is taken, using close mics to produce real-time, loudspeaker-projected virtual acoustics, and room microphones to capture a balanced, natural sound. The known loudspeaker signals are then used to cancel the virtual acoustics from the room microphone tracks, providing a set of relatively dry tracks for use in editing and post-production. Example recordings of Byzantine chant in a virtual Hagia Sophia are described.
Convention Paper 9854

5:30 pm

P9-8 A Study of Listener Bass and Loudness Preferences over Loudspeakers and Headphones—*Elisabeth McMullin*, Samsung Research America, Valencia, CA USA

In order to study listener bass and loudness preferences over loudspeakers and headphones a series experiments using a method of adjustment were run. Listeners adjusted the bass and loudness levels of multiple genres of music to their personal preference in separate listening sessions over loudspeakers in a listening room and headphones equalized to simulate loudspeakers in a listening room. The results indicated that listeners who preferred more bass over both headphones and loudspeakers also tended to listen at higher levels. Furthermore the majority of listeners preferred slightly higher bass and loudness levels over loudspeakers than over headphones. Listener factors including musical preferences, hearing ability, and training level are also explored.
Convention Paper 9855

Session P11 **Thursday, Oct. 19**
2:00 pm – 3:30 pm **Foyer**

POSTERS: SPATIAL AUDIO

2:00 pm

P11-1 Deep Neural Network Based HRTF Personalization Using Anthropometric Measurements—*Chan Jun Chun,*[1] *Jung Min Moon,*[2] *Geon Woo Lee,*[2] *Nam Kyun Kim,*[2] *Hong Kook Kim*[2]
[1]Korea Institute of Civil Engineering and Building Technology (KICT), Goyang, Korea
[2]Gwangju Institute of Science and Technology (GIST), Gwangju. Korea

A head-related transfer function (HRTF) is a very simple and powerful tool for producing spatial sound by filtering monaural sound. It represents the effects of the head, body, and pinna as well as the pathway from a given source position to a listener's ears. Unfortunately, while the characteristics of HRTF differ slightly from person to person, it is usual to use the HRIR that is averaged over all the subjects. In addition, it is difficult to measure individual HRTFs for all horizontal and vertical directions. Thus, this paper proposes a deep neural network (DNN)-based HRTF personalization method using anthropometric measurements. To this end, the CIPIC HRTF database, which is a public domain database of HRTF measurements, is analyzed to generate a DNN model for HRTF personalization. The input features for the DNN are taken as the anthropometric measurements, including the head, torso, and pinna information. Additionally, the output labels are taken as the head-related impulse response (HRIR) samples of a left ear. The performance of the proposed method is evaluated by computing the root-mean-square error (RMSE) and log-spectral distortion (LSD) between the referenced HRIR and the estimated one by the proposed method. Consequently, it is shown that the RMSE and LSD for the estimated HRIR are smaller than those of the HRIR averaged over all the subjects from the CIPIC HRTF database.
Convention Paper 9860

2:00 pm

P11-2 The Upmix Method for 22.2 Multichannel Sound Using Phase Randomized Impulse Responses—*Toru Kamekawa, Atsushi Marui*, Tokyo University of the Arts, Tokyo, Japan

The upmix technique for 22.2 multichannel sound was studied using room impulse responses (RIRs) processed by phase randomized technique. From the result of the first experiment, the spatial impression of proposed method was close to the original sound, but the timbre differed. In the second experiment we divided the RIRs at the moment when the diffuse reverberation tail begins (mixing time) by two kinds of time, namely fixed to 80 msec and different mixing times for each frequency band. From the result, the similarity of proposed methods and the original sound was improved, however, it is suggested that the similarity of the timbre depends on the sound sources and the suitable mixing time of RIRs.
Convention Paper 9861

2:00 pm

P11-3 A 3D Sound Localization System Using Two Side Loudspeaker Matrices—*Yoshihiko Sato, Akira Saji, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

We have proposed a new 3D sound reproduction system that consists of two side loudspeaker matrices each with four loudspeakers. The 3D sound images that applied to this system were created by the amplitude panning method and convolution of head-related transfer function (HRTF). In our past research we used the loudspeaker

matrices arranged as a square shape, nevertheless the accuracy of sound image localization should be improved. We changed the shape of loudspeaker matrices from a square to a diamond by rotating 45 degrees to improve direction perception. As a result, we could be closer the localized sound images to intended directions than the square-shaped loudspeaker matrices by implementing the diamond-shaped loudspeaker matrices.
Convention Paper 9862

2:00 pm

P11-4 Optimization of Interactive Binaural Processing—
François Salmon,[1,2] *Matthieu Aussal,*[2] *Etienne Hendrickx,*[3,4] *Jean-Christophe Messonnier,*[3] *Laurent Millot*[1,3]
[1]Ecole Nationale Supérieur Louis-Lumière, Paris, France
[2]CMAP - Ecole Polytechnique, Paris, France
[3]Paris Conservatory (CNSMDP), Paris, France
[4]Université de Bretagne Occidentale, Brest Cedex, France

Several monitoring devices may be involved during a post-production. Given its lower cost and practical aspects, head-tracked binaural processing could be helpful for professionals to monitor spatialized audio contents. However, this technology provides significant spectral coloration in some sound incidences and suffers from its current comparison to a stereophonic signal reproduced through headphones. Therefore, different processing methods are proposed to optimize the binaural rendering and to find a new balance between externalization and timbral coloration. For this purpose, the alteration of the HRTF spectral cues in the frontal area only has been studied. In order to evaluate the accuracy of such treatments, listening tests were conducted. One HRTF processing method offered as much externalization as the original HRTFs while having a closer timbre quality to the original stereo signal.
Convention Paper 9863

2:00 pm

P11-5 A Direct Comparison of Localization Performance When Using First, Third, and Fifth Ambisonics Order for Real Loudspeaker and Virtual Loudspeaker Rendering—*Lewis Thresh, Calum Armstrong, Gavin Kearney*, University of York, York, UK

Ambisonics is being used in applications such as virtual reality to render 3-dimensional sound fields over headphones through the use of virtual loudspeakers, the performance of which has previously been assessed up to third order. Through a localization test, the performance of first, third, and fifth order Ambisonics is investigated for optimized real and virtual loudspeaker arrays utilizing a generic HRTF set. Results indicate a minor improvement in localization accuracy when using fifth order over third though both show vast improvement over first. It is shown that individualized HRTFs are required to fully investigate the performance of Ambisonic binaural rendering.
Convention Paper 9864

Project Studio Expo 09 — Thursday, October 19
2:00 pm – 2:45 pm — Stage 2

HOW TO MAKE YOUR VOCAL TWICE AS GOOD

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.

Standards Committee Meeting
SC-02-02 WORKING GROUP ON DIGITAL INPUT/OUTPUT INTERFACING
Thursday, October 19, 2:00 pm – 3:30 pm
Room 1C04

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.

Software@AES 24 — Thursday, October 19
2:00 pm – 2:20 pm — Stage 3

INTERNET CO.

Software@AES 25 — Thursday, October 19
2:30 pm – 2:50 pm — Stage 3

BLUE CAT AUDIO

Recording & Production 6 — Thursday, October 19
3:00 pm – 4:15 pm — Room 1E14

THE RECORDING ACADEMY P&E WING PRESENTS: HIGH RESOLUTION RECORD PRODUCTION AND WHY IT MATTERS

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

Panelists: *Chuck Ainlay*, METAlliance, Nashville, TN, USA
Marc Finer, Digital Entertainment Group, Los Angeles, CA, USA
Leslie Ann Jones, Skywalker Sound, San Rafael, CA, USA
Bob Ludwig, Gateway Mastering Studios, Inc., Portland, ME, USA

Many GRAMMY nominated and winning members of the Recording Academy Producers & Engineers Wing work in Hi-Res audio—lossless audio that's better than CD quality in both sample rate and bit depth. They're so serious about this topic they spent more than two years doing research and collaborating on a great set of guidelines titled "Recommendations for Hi-Res Music Production," available on the P&E website. Working at high resolution has many current and long term benefits, both artistically and commercially. They want the rest of the industry to join them. Passionate? You bet. Come here what they have to say about why they choose to work in Hi-Res.

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

Broadway Sound Expo 05 | **Thursday, October 19**
3:00 pm – 3:45 pm | **Stage 1**

SPATIAL REINFORCEMENT—NEW NORMAL FOR IMMERSIVE THEATER VOCALS AND FX

Presenters: **David Haydon**, TiMax & Outboard
Robin Whittaker, TiMax & Outboard

Explanation and simple demonstrations of spatial masking/unmasking and precedence, showing how real-time control of source-oriented delays in theatrical sound systems is essential for intelligibility, impact, and audience immersion. Dynamic delay-matrix processing hardware and its object-based control software will be explored, revealing how readily they can achieve transparent vocal mic amplification and realistic immersive sound effects for all audience members.

Project Studio Expo 10 | **Thursday, October 19**
3:00 pm – 3:45 pm | **Stage 2**

A CONVERSATION WITH ELLIOT SCHEINER

Presenter: **Elliot Scheiner**, Producer, USA

[Abstract not available]

Software@AES 26 | **Thursday, October 19**
3:00 pm – 3:20 pm | **Stage 3**

BEST SERVICE

Audio for Cinema 7 | **Thursday, October 19**
3:15 pm – 4:45 pm | **Room 1E10**

BEST PRACTICES IN RE-RECORDING MIXING

Presenter: **Tom Fleischman**, CAS

This session on re-recording will include how Tom Fleischman approaches a project from the beginning to the end, focusing on how the mix can enhance the storytelling, the importance of clarity of dialogue, and the various ways that music and sound effects are used to engage the audience emotionally and to keep the suspension of disbelief. He will also discuss some of the history of how re-recording has changed over the last 40 years and how the advent of digital sound recording and digital signal processing have changed the way that films are mixed. Although he uses advanced technological tools in the mix, and he will discuss the use of these tools, he is much more oriented toward the art of the process and how what he does affects how the story is told and how important it is to keep the audience engaged with what is happening dramatically on the screen.

Co-organized with Cinema Audio Society (CAS).

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

Game Audio & VR 8 | **Thursday, October 19**
3:15 pm – 4:15 pm | **Room 1E13**

IMMERSIVE AUDIO FOR VR WORKFLOW

Moderator: **Michael Kelly**, Senior Director, R&D, DTS, London, UK

Panelists: *Joel Douek*, ECCO VR
Sally-Anne Kellaway, Creative Director, Spatial Audio, OSSIC
Ken Usami

In the emerging field of AR/VR both the tools and techniques for creating content are still in a nascent state. Ways of thinking through the creative process and applications workflows are surfacing and evolving in real-time with new product development, making this a very exciting time to be a content creator (and a content consumer) but also includes potential gaps and problems as standard best practices are only developing.

Networked Audio 2 | **Thursday, October 19**
3:15 pm – 4:15 pm | **Room 1E08**

WHO OWNS THE AUDIO NETWORK, AV OR IT?

Presenter: **Patrick Kilianey**, Yamaha - Professional Audio Division, Buena Park, CA, USA

How would you feel if IT decided to mix your next show? Nervous? Well then, don't be surprised if IT is uneasy when we add computer network equipment in their facility. This session will cover some ideas on how to work with IT, whether your network is intended to connect to theirs or not. We'll offer insights to common IT concerns and how to address them with infrastructure design and network topology. Real-world examples will be given. We will also discuss differences in expectations between typical IT departments and AV personnel—addressing these early can prevent costly change orders and rework later.

Spatial Audio 7 | **Thursday, October 19**
3:15 pm – 4:45 pm | **Room 1E06**

PRACTICAL IMMERSIVE AUDIO AT HOME

Chair: **Chris Pike**, BBC Research and Development, Salford, UK; University of York, York, UK

Presenters: *Jon Francombe*, University of Surrey, Guildford, Surrey, UK
Hilmar Lehnert, Sonos
Alan Seefeldt, Senior Member of Research Staff, Dolby Laboratories

Current methods for immersive audio reproduction in the home include channel-based systems, object-based audio, different types of soundbars, multi-room wireless or Bluetooth loudspeakers, up-firing loudspeakers, ad hoc arrays of mobile phones, and so on. These very different approaches all unlock opportunities for creating immersive and personalizable listening experiences, and each has its own merits and limitations. This workshop will feature a panel of experienced industrial practitioners and academic researchers. It will provide a platform for discussion around different perspectives on the challenges and potential solutions for making engaging and personalizable spatial audio experiences available in the living room.

Student Events/Career Development
EC6: RECORDING COMPETITION—PART 1
Thursday, October 19, 3:15 pm – 5:15 pm
Room 1E07

The Student Recording Competition is a highlight at each convention. A distinguished panel of judges participates in critiquing finalists of each category in an interactive presentation and discussion. The top three finalists in each category, as identified by our judges, present a short summary of their production intentions and the key recording and mix techniques used to realize their goals. They then play their projects for all who attend. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2) on Sunday afternoon. The competition is a great chance to hear the work of your fellow students at other educational institutions. Everyone learns from the judges' comments, even those who don't make it to the finals, and it's a great chance to meet other students and faculty.

Standards Committee Meeting
SC-04-04 WORKING GROUP ON MICROPHONE MEASUREMENT AND CHARACTERIZATION
Thursday, October 19, 3:30 pm – 5:00 pm
Room 1C04

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 27 **Thursday, October 19**
3:30 pm – 3:50 pm **Stage 3**

SONARWORKS

Session EB3 **Thursday, Oct. 19**
3:45 pm – 5:00 pm **Room 1E12**

APPLICATIONS IN AUDIO

Chair: TBA

3:45 pm

EB3-1 Evolving the Audio Equalizer—*David Yonovitz*, Key 49, Del Mar, CA, USA

Current audio equalization techniques include Shelf, Parametric, and Graphic Equalizers. Each have inherent issues: dynamic spectrum input and the degradation of signal-to-noise ratio. The input spectrum is not static; yet, all the current equalizations are. To be effective, equalization must be dynamic, "tracking" the input signal spectrum. In the case of SNR, output noise is increased when no signal is present in the spectral band when adding gain. In an evolution of equalization, with input tracking capability, signal spectral components are identified and equalized; all other spectrum may be considered as noise and can be attenuated. The evolution of audio equalizers has progressed that negates the stated issues. Its implementation is realized in the Harmonic Tracking Equalizer (HTEq).
Engineering Brief 369

4:00 pm

EB3-2 Immersive Audio: Optimizing Creative Impact without Increasing Production Costs—*Connor Sexton*, Avid Technology, El Segundo, CA, USA

Since its introduction in 2012, Dolby Atmos has gained widespread adoption in theatrical distribution for films, with over 2,000 Dolby Atmos enabled theaters worldwide. Now expanding into TV and gaming, this unique audio mixing format provides a new dimension of creative control over the immersive listening experience. Numerous audio workstations and consoles have been retrofitted for Dolby Atmos, but without native support, workflows have become cumbersome and complex. This paper will present best practices for native immersive audio production, from sound design to mixing to distribution. It will demonstrate how the latest audio production tools and techniques enabled content creators to capitalize on the creative power of immersive audio while streamlining the parallel authoring of traditional formats.
Engineering Brief 370

4:15 pm

EB3-3 *Engineering Brief 371 was withdrawn*

4:30 pm

EB3-4 "Match Your Own Voice!": An Educational Tool for Vocal Training—*Evangelos Angelakis, Panayiotis Velianitis, Areti Andreopoulou, Anastasia Georgaki*, National and Kapodistrian University of Athens, Athens, Greece

In this paper, we discuss the development and preliminary evaluation of a new educational tool, intended for novice and advanced vocal students. The software, written in Max / MSP, aims to assist singing practice by providing users with a visual substitute to their subjective auditory feedback. Under the guidance of their professional vocal instructor, students can store in the software spectral representations of accurately produced sounds, creating personalized Reference Sound Banks (RSBs). When students practice on their own, the software can be put into practice, assisting them to match their current Voice Spectrum Harmonic Content to the stored RSBs one note at a time. Results of a preliminary evaluation showed that, when using this software, students achieve a larger number of accurately produced sounds in a smaller amount of time.
Engineering Brief 372

4:45 pm

EB3-5 Standardizing PSR, the Peak to Short-Term Loudness Ratio— *Ian Shepherd,*[3] *Eelco Grimm,*[1,2] *Paul Tapper,*[4] *Michael Kahsnitz,*[5] *Ian Kerr* [6]
[1]HKU University of the Arts, Utrecht, Netherlands
[2]Grimm Audio, Eindhoven, The Netherlands
[3]Mastering Media Ltd., Cambridge, Cambridgeshire, UK
[4]Nugen Audio, UK
[5]RTW, Cologne, Germany
[6]MeterPlugs Audio Inc., Canada

The "loudness war" still rages, but with major digital streaming services switching to loudness normalization by default, its end is near. Since absolute loudness is no longer effective at making music "stand out," engineers are finding it much more effective to optimize microdynamics instead. The overall PLR (Peak to Loudness Ratio) of an audio track is widely recognized as a useful metric to assess the overall microdynamics of a section of audio and the likely results of normalization. However, short-term variations are also important, especially when judging the results of compression and limiting on audio quality, and these can be usefully assessed by a real-time property known as PSR (Peak to ShortTerm Loudness Ratio). PSR is found to be straightforward and intuitive to use, and several popular meters are already reporting it. This paper proposes a standardization of the term, to encourage consistency and adoption.
Engineering Brief 373

Thursday, October 19 4:00 pm Room 1C03

Technical Committee Meeting on Acoustics and Sound Reinforcement

REGIONS AND SECTIONS MEETING
Thursday, October 19, 4:00 pm – 6:00 pm
Room 1E09

Broadway Sound Expo 06 Thursday, October 19
4:00 pm – 4:45 pm Stage 1

DIGITAL CONSOLE PERFORMANCE

Presenter: **Ron Lorman**, Cadac

An open discussion of digital console performance and how it relates to meaningful factors for retaining emotion in your mix, latency, ADDA conversion, clocking, and compression, 'ones and zeroes' vs analog, current audience audio quality expectations, new audio quality reference points, production values and priorities, meaningful industry-wide improvements.

Project Studio Expo 11 Thursday, October 19
4:00 pm – 4:45 pm Stage 2

MIXING LEAD VOCALS: POWER-USER TIPS FOR COMPETING WITH THE PROS

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

How can you maximize the hit potential of your lead vocals at mixdown? Join *Sound On Sound* magazine's "Mix Rescue" and "MixReview" columnist Mike Senior as he shares powerful insider techniques that can give your productions that competitive edge.

Software@AES 28 Thursday, October 19
4:00 pm – 4:20 pm Stage 3

CELEMONY (MELODYNE)

Broadcast/Streaming Media 8 Thursday, October 19
4:30 pm – 6:00 pm Room 1E08

AUDIO CONSIDERATIONS FOR PODCASTS

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

Panelists: *Malik Abdullah*, National Public Radio's Digital Media, Washington, DC, USA
Dan Jeselsohn, New York Public Radio, New York, NY, USA
Angelo Mandato, Blubrry Podcasting, Columbus, OH, USA
Samuel Sousa, Triton Digital, Montreal, QC, Canada

Podcasting, a system for packaging and distributing audio shows to listeners via the Internet, is growing exponentially. Currently, it is estimated that more than 40 million Americans listen to podcasts weekly - five times more than attend movies, and the numbers have risen more than 10% since 2016. The panel includes Angelo Mandatto of blubrry, whick specializes in business and demographic data on the industry. To cover podcast technology, audio metrics and workflows, panelists are Samuel Sousa, Senior Solutions Specialist at Triton Digital - providing technology services for the online audio industry, Dan Jeselsohn of New York Public Radio, and Chris Berry of National Public Radio's Digital Media - the largest global publisher of podcasts.

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

Game Audio & VR 9 Thursday, October 19
4:30 pm – 6:00 pm Room 1E13

MUSIC RECORDING FOR VIRTUAL REALITY

Presenters: **Gavin Kearney**, University of York, York, UK
Hyunkook Lee, University of Huddersfield, Huddersfield, UK
Hashim Riaz, University of York, York, UK
Mirek Stiles, Abbey Road Studios, London, UK

There is a growing market for innovative ways to appreciate and listen to music through new Virtual Reality (VR) experiences made accessible through smartphones and VR headsets. However, production workflows for creating immersive musical experiences over VR are still in their infancy. This workshop explores the effectiveness of different microphone configurations and recording techniques in a higher-order Ambisonic processing framework to deliver an engaging and hyper-real interactive VR music experience, with and without immersive 360 visuals. In particular, it will look at how traditional recording workflows can be adapted to immersive 360. Test material will be demonstrated of a live popular music recordings undertaken at Abbey Road with traditional music recording techniques such as spot and stereo microphone setups and advanced techniques using dedicated VR multichannel microphone arrays.

Recording & Production 7 Thursday, October 19
4:30 pm – 6:00 pm Room 1E14

RAW TRACKS: CANCELED

Sound Reinforcement 11 Thursday, October 19
4:30 pm – 6:00 pm Room 1E14

SOUND SYSTEM DESIGN FOR PUBLIC ADDRESS AND INTELLIGIBILITY

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

This Tutorial discusses what constitutes a difficult acoustic space and the effects of reverberation, echo, and noise on speech intelligibility. Sound system Intelligibility measurement methods and techniques will be reviewed, while system optimization techniques will also be examined. The tutorial includes numerous case histories and examples illustrating the techniques discussed. A fundamental question that will be raised will be "how intelligible do sound systems need to be and does this change with application or is intelligibility universal?" The case histories and examples will include Churches, Cathedrals and Houses of Worship, Concert Halls, Ceremonial Halls / Banqueting Halls, Shopping Malls, Sports Facilities and Arenas, Railway & Metro Stations, Road Tunnels and Airports. The challenges associated with each type of venue will be discussed, as will the thorny subject of aesthetics (i.e., Architects) and loudspeaker size and placement.

Special Events
SE5: PRODUCING ACROSS GENERATIONS:

NEW CHALLENGES, NEW SOLUTIONS
Thursday, October 19, 4:30 pm – 5:45 pm
Room 1E15/16

Moderator: **Nick Sansano**

Panelists: *Jesse Lauter*
George Massenburg
Bob Power
Kaleb Rollins
Hank Shocklee
Erin Tonkon

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

Software@AES 29 **Thursday, October 19**
4:30 pm – 4:50 pm **Stage 3**

FABFILTER

Recording & Production 1 **Thursday, October 19**
4:45 pm – 6:00 pm **Room 1E06**

MODERN CLASSICAL PRODUCTION

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

Panelists: *David Bowles*
Martha DeFrancisco
Susan DelGiorno
John Newton

Award-winning producers and engineers discuss the tools and evolving processes used in classical music production from recording chains to workflow to edit and mix.

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

Networked Audio 3 **Thursday, October 19**
5:00 pm – 6:00 pm **Room 1E12**

FIRST STEPS WITH DIGITAL AUDIO NETWORKS

Presenter: **Patrick Kilianey**, Yamaha - Professional Audio Division, Buena Park, CA, USA

From live sound systems, recording, broadcast, and distributed audio, everything runs on a network. But do you know why audio networks are powerful? This session will explain the value audio networks bring as it replaces massive analog and digital interconnects, splitters, routers, and computer interfaces. We will discuss how to keep systems simple enough for most audio technicians to manage, and we will touch on interfacing with an IT department. Finally, we will offer some perspective on popular networks, where you will find them and why.

Recording & Production 8 **Thursday, October 19**
5:00 pm – 6:00 pm **Room 1E10**

MICROPHONES - CAN YOU HEAR THE SPECS?

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

Panelists: *Jürgen Breitlow*, George Neumann, Berlin, Germany
Eddy B. Brixen, EBB-Consult, Smørum, Denmark
David Josephson, Josephson, Engineering, Inc. Santa Cruz, CA, USA

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

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

Thursday, October 19 **5:00 pm** **Room 1C03**

Technical Committee Meeting on Coding of Audio Signals

Broadway Sound Expo 07 **Thursday, October 19**
5:00 pm – 5:45 pm **Stage 1**

MIXING A MUSICAL

[Presenter and abstract not available]

Project Studio Expo 12 **Thursday, October 19**
5:00 pm – 5:45 pm **Stage 2**

JAMES MCKINNEY & SCOTT JACOBY—ANGUILLA MUSIC ACADEMY PROJECT

Presenters: **Scott Jacoby**, Producer, USA
James McKinney, Eusonia Studios, USA

[Abstract not available]

Software@AES 30 **Thursday, October 19**
5:00 pm – 5:20 pm **Stage 3**
LINE 6

Networked Audio 4 **Thursday, October 19**
5:30 pm – 6:00 pm **Room 1E07**

ST 2110 & AES67: THE AUDIO PARTS OF ST 2110

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

We will describe the basic principles of AES67 and the commonalities, differences, and constraints defined in ST2110 with respect to synchronization and transport of audio essence streams. Included is a brief outlook on how transport of non-linear audio formats (AES3) will most likely be defined in ST2110.

Software@AES 31 **Thursday, October 19**
5:30 pm – 5:50 pm **Stage 3**

ANTARES

Student Events/Career Development
EC7: AES STUDENT PARTY
Thursday, October 19, 7:00 pm – 10:00 pm
Jungle City Studios
520 West 27th St. #1002, New York, NY 10001

The AES Student Party is open to any 143rd Convention participant with an ALL ACCESS STUDENT BADGE—a great opportunity to meet fellow students from around the world. Tickets should be picked up at the Student Delegate Assembly desk on Wednesday 18th, at 2pm. http://www.aes.org/events/143/students/

Genelec invites you to come experience The Ones at Jungle City Studios for the official AES student social event of the show. Refreshments and food will be served, and guitarist extraordinaire Gary Hoey will be performing live in the studio.

All event attendees will be entered for a chance to win a pair of Genelec 8331's when they register at the door. As a secondary giveaway, there will be a drawing for a limited-edition pair of Genelec 8030's in red. This drawing is open only to U.S. residents who have subscribed to the Genelec USA Learning Channel on YouTube (https://www.youtube.com/genelecusa) by Nov. 30, 2017. Subscribers must share their YouTube subscriptions publicly to be included in the drawing.

Special Event
ORGAN CONCERT
Thursday, October 19, 7:15 pm – 9:00 pm
Church of the Ascension (5th Ave. at 10th St.)

Presenter: **Graham Blyth**

Graham Blyth's concerts are a highlight of every AES convention at which he performs. this year's concert will be held at the Church of the Ascension.

Graham Blyth's program will be:
J.S Bach (1685 - 1750)
1. Prelude & Fugue in E min BWV 548 (The Wedge)
2. Trio on "Allein Gott in der Höh' sei Ehr' " BWV 676
3. Fantasia & Fugue in G min BWV 542
Auguste César Franck (1822 - 1890)
Grand Pièce Symphonique

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

In 1973 he founded Soundcraft with Phil Dudderidge, and has been Technical Director from the beginning. Soundcraft has since grown to be one of the most significant designers and manufacturers of audio mixing consoles in the world. In 1988, the company was bought by Harman, whose professional audio companies now include JBL, DBX, Lexicon, AKG and Studer.

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

He has lived in Wantage, Oxfordshire, since 1984 where he is currently Artistic Director of Wantage Chamber Concerts and Director of the Wantage Summer Festival. In 1995 he built the Challow Park Recital Hall, an 80 seat auditorium with completely variable acoustics, using the Lexicon LARES system, designed by his then Harman colleague David Griesinger. This allows for performance and recording of music ranging from String Quartets through to Organ Recitals. Today he divides his time between audio engineering and organ design activities. In 2003 he founded the Veritas Organ Company to address the top end of the digital classical organ market, specializing in creating new organs by adding digital voices to existing pipes. He is a Fellow of the Royal Society of Arts and the Audio Engineering Society.

Session P12 **Friday, Oct. 19**
9:00 am – 11:30 am **Room 1E11**

TRANSDUCERS—PART 3

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

9:00 am

P12-1 Objective Testing of High-End Audio Systems—*Gregor Schmidle,[1] Gerd Köck,[2] Brian MacMillan[3]*
[1]NTi Audio AG, Schaan, Liechtenstein
[2]Art Déco Acoustics Kornwestheim, Germany
[3]NTi Audio Inc., Portland, OR, USA

The high-end audio equipment market is filled with extraordinary products. Although the engineering and the materials utilized are often of the finest available, the quality control of such systems is frequently done subjectively rather than objectively. This paper shows some best practice examples of how to deploy effective quality measurement systems through the complete life cycle (R&D, QC installation, and repair) of high-end audio systems.
Convention Paper 9835

9:30 am

P12-2 Theory of Constant Directivity Circular-Arc Line Arrays—*Richard Taylor,[1] D. B. (Don) Keele, Jr.[2]*
[1]Thompson Rivers University, Kamloops, BC, Canada
[2]DBK Associates and Labs, Bloomington, IN, USA

We develop the theory for a broadband constant-beamwidth transducer (CBT) formed by a continuous circular-arc isophase line source. Appropriate amplitude shading of the source distribution leads to a far-field radiation pattern that is constant above a cutoff frequency determined by the prescribed beam width and arc radius. We derive two shading functions, with cosine and Chebyshev polynomial forms, optimized to minimize this cutoff frequency and thereby extend constant-beamwidth behavior over the widest possible band. We illustrate the theory with simulations of magnitude responses, full-sphere radiation patterns and directivity index, for example designs with both wide- and narrow-beam radiation patterns.
Convention Paper 9836

10:00 am

P12-3 Constant Directivity Circular-Arc Arrays of Dipole Elements—*Richard Taylor,* [1]*Kurtis Manke,*[1]*D. B. (Don) Keele, Jr.*[2]
[1]Thompson Rivers University, Kamloops, BC, Canada
[2]DBK Associates and Labs, Bloomington, IN, USA

We develop the theory for a broadband constant-beam-width transducer (CBT) formed by a conformal circular-arc line array of dipole elements. Just as for CBT arrays of point sources, with suitable amplitude shading of the source distribution the far-field radiation pattern is constant above a cutoff frequency. This cutoff frequency is determined by the prescribed beam width and arc radius. We illustrate the theory with examples, including numerical simulations of magnitude responses, full-sphere radiation patterns, and directivity index. Unlike a circular-arc array of monopole elements, a dipole CBT maintains directivity control at low frequency. We give an example of one such array that achieves just 1 dB variation in directivity index over all frequencies.
Convention Paper 9837

10:30 am

P12-4 Voice Coil Temperature—Non Linearity Compensations for Ultra Audio Band Impedance Probing—*Isao Anazawa*, Ny Works, Toronto, ON, Canada

As loudspeaker output power of mobile devices increases for better audio experience, an accurate measurement or estimation of the voice coil temperature becomes necessary in order to protect the loudspeaker from over-heating. A voice coil designed with a short ring, a metal pole piece, or under hung voice coil will most likely exhibit impedance nonlinearity. A voice coil resistance based temperature measurement method that relies on the resistance value may be adversely affected by voice coil impedance nonlinearity when the resistance is measured using high frequency probing. For this reason, the nonlinearity must be known and compensated. This paper analyzes and explains voice coil high frequency impedance characteristics due to Eddy losses and impedance nonlinearity, and develops a method to compensate voice coil impedance nonlinearity to obtain an accurate voice coil temperature measurement.
Convention Paper 9838

11:00 am

P12-5 Variable Fractional Order Analysis of Loudspeaker Transducers: Theory, Simulations, Measurements, and Synthesis—*Andri Bezzola,*[1] *Pascal Brunet,*[1] *Shenli Yuan*[2]
[1]Samsung Research America, Valencia, CA USA
[2]Center for Computer Research in Music and Acoustics (CCRMA), Stanford University, Stanford, CA, USA

Loudspeaker transducer models with fractional derivatives can accurately approximate the inductive part of the voice coil impedance of a transducer over a wide frequency band, while maintaining the number of fitting parameters to a minimum. Analytical solutions to Maxwell equations in infinite lossy coils can also be interpreted as fractional derivative models. However, they suggest that the fractional order a cannot be a constant, but rather a function of frequency that takes on values between 1/2 and 1. This paper uses Finite Element (FEM) simulations to bridge the gap between the theoretical first-principles approach and lumped parameter models using fractional derivatives. The study explores the dependence of a on frequency for idealized infinite and finite cores as well as in four real loudspeaker transducers. To better match the measured impedances and frequency-dependent a values we propose to represent the voice coil impedance by a cascade of R-L sections.
Convention Paper 9839

Archiving/Restoration 6 **Friday, October 20**
9:00 am – 10:30 am **Room 1E15/16**

THE MUSIC NEVER STOPPED: THE FUTURE OF THE GRATEFUL DEAD EXPERIENCE IN THE INFORMATION AGE—PART 1

Chairs: **György Fazekas**, Queen Mary University of London, London, UK
Thomas Wilmering, Queen Mary University of London, London, UK
Panelists: *Jeremy Berg*, Cataloging Librarian, University of North Texas
Scott Carlson, Metadata Coordinator, Rice FondrenLibrary
Nicholas Meriwether, Director of Grateful Dead Archive, UCSC
John Meyer, CEO, Meyer Sound
Brian Pardo, Assoc. Professor Electrical Engineering & Computer Science, Northwestern University

Ever since the advent of recording, technology has been constantly shaping the way we interact with music as well as the relationship between artists and fans. For instance, file compression and broadband internet disrupted conventional music distribution, creating new opportunities for the formation of online fan communities. A growing number of bands allow taping their live performances while there are expanding online archives, such as Etree and the Live Music Archive, for trading audio freely with permission between an increasing number of fans. The Grateful Dead and their fans the Deadheads bestow a prominent example with their innovative use of this technology. Semantic technologies are next in line, with a premise of step change in how we access audio archives. This workshop explores how semantic technologies provide enriched experiences for fans, greater exposure for bands and new opportunities for archives to flourish. We demonstrate new ways of navigating concert recordings, particularly those of the Grateful Dead, discuss opportunities and requirements with audio archivists and librarians, as well as the broader social and cultural context of how new technologies bear on music archiving and fandom.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Archiving/Restoration 7 **Friday, October 20**
9:00 am – 10:30 am **Room 1E09**

THE PAST AND FUTURE OF ARCHIVING & PRESERVATION

Presenters: **Rebecca Yuri Feynberg**, New York University, New York, NY, 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
Brad McCoy, Library of Congress, Culpeper, VA, USA
Toby Seay, Drexel University, Philadelphia, PA, USA
Nadja Wallaszkovits, Vienna Phonogrammarchiv, Austria

This session will address important issues in archiving and preservation including audio preservation and higher education, preservation of multi-channel recordings for music and cinema, the process and procedures of archiving and preservation, and current work at the Library of Congress. The panelists will present issues that will be featured at the upcoming international AES conference on Audio Archiving Preservation and Restoration.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Game Audio & VR 10 **Friday, October 20**
9:00 am – 11:00 am **Room 1E13**

STATE OF THE ART: KICKSTART YOUR CAREER IN AUDIO FOR GAMES, VR, AND BEYOND

Chair: **Scott Looney**, Game Audio Institute

Panelists: *Brennan Anderson* Producer, Pyramind
Bonnie Bogovich, Schell Games
Damian Kastbauer
Larry Oppenheimer, Toys in the Attic

Make no mistake, game audio is a highly technical business. Creating compelling sound for games takes an incredible combination of skills that goes far beyond writing great music and designing awesome sounds. The best game audio designers possess deep expertise that highlights their dedication to the overall experience, and as the industry matures and moves into more immersive fields like VR and XR, an all-encompassing skillset like this will only become more sought after.

This panel will discuss how to succeed in this field by learning from industry professionals who have spent decades moving the needle in interactive audio and who have built their careers on a philosophy of lifelong learning and intellectual curiosity while breaking new ground at the bleeding edge of their art.

Product Development 6 **Friday, October 20**
9:00 am – 10:30 am **Room 1E14**

MODERN HYBRID AUDIO CODING

Presenters: **Jürgen Herre**, International Audio Laboratories Erlangen, Erlangen, Germany; Fraunhofer IIS, Erlangen, Germany
Sascha Dick
Andreas Niedermeier
Mark Vinton

During the past one and a half decades, recent audio coding schemes have significantly overcome traditional limits for compression efficiency by adopting techniques for semi-parametric (hybrid) coding of audio signals. By doing so, full-bandwidth stereo reproduction can today be achieved even at very low bitrates, such as 12kbit/s. The keys to this breakthrough achievement were two types of semi-parametric coding extensions: First, methods for bandwidth extension (BWE) allow full reproduced audio bandwidth even at low rates. Second, methods for parametric stereo (or multichannel) coding enable good reproduction of spatial sound under similar circumstances. The workshop will present the current state of development in these active areas, describe relevant technology and illustrate its performance by sound examples.

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

Recording & Production 9 **Friday, October 20**
9:00 am – 10:30 am **Room 1E08**

RAW TRACKS: STAYIN' ALIVE: RECORDING THE BEE GEES "SATURDAY NIGHT FEVER"

Moderator: **John Merchant**, Middle Tennessee State University, Murfreesboro, TN, USA

Panelists: *Albhy Galuten* (UMG, Sony; Bee Gees, Andy Gibb, Barbra Streisand, Eric Clapton, Diana Ross, Eagles, Cher, Jellyfish)
Karl Richardson (Bee Gees, Andy Gibb, Barbra Streisand, Eric Clapton, Diana Ross, Kenny Rogers, Aretha Franklin)

This panel features the engineers and producers of the classic soundtrack for *Saturday Night Fever*. The multitrack recordings of the songs will be available to illustrate points of discussion and the production techniques involved. The event is a guided discussion of the history of the recording, mixing and mastering, the critical and cultural response to the album, and includes a period for open questions and answers.

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

Sound Reinforcement 6 **Friday, October 20**
9:00 am – 10:30 am **Room 1E08**

CANCELED

Tutorial/Workshop 7 **Friday, October 20**
9:00 am – 10:30 am **Room 1E12**

TECH BEHIND THE TOOLS: PRODUCTION TO PLAYBACK

Chair: **Lisa Ferrante-Walsh**, iZotope

Panelists: *Marina Bosi*, Stanford University, CA, USA
Veronique Larcher, Sennheiser, Switzerland
Vanessa Li, Pandora
Hannah Robertson, iZotope

Have you wondered what's under the hood behind that favorite product in your audio tool belt? This 90 minute panel is made up of engineers and technologists - who are experts in the technology behind a sampling of the products, tools and services used in the audio production process. Each panel member will educate us about the technology behind a different aspect of the journey from production through playback.

• **Where Do Audio Plugins Come From?** - Hannah Robertson, DSP Engineer at iZotope

This talk will walk through several new features from recently released iZotope products to demystify the development cycle of a plugin: from the ideation stage to research, implementation, sound design, audio quality testing, and user feedback.

• **Perceptual Audio Coding** - Marina Bosi, Ph.D., Consulting Professor at Stanford University and a Founding Director of the Digital Media Project

Perceptual audio coders (e.g., MP3/AAC) have become part of our daily lives, living in mobile devices, DVDs, webcasting, etc. How did we get here and where are we going? This talk, presented by one of the early developers who helped advance the field of perceptual audio coding, will provide a brief overview, new developments and sound demonstrations to illustrate the impact of this technology.

• **The VR Production Workflow** - Veronique Larcher, Ph.D., Director AMBEO Immersive Audio at Sennheiser

This talk will review the constraints and tools for on-location VR audio capture, VR audio mixing and monitoring, and finally VR audio experience by the greater numbers on VR platforms or apps.

• **Content Delivery and Management at Pandora** - Vanessa Li, Software Engineer at Pandora

As demand for music consumption has shifted from physical albums to downloads to online streaming, new standards and protocols have emerged as the music industry business practices have adapted. This talk will address some of the technical challenges that Pandora and other on-demand streaming platforms face around content delivery, storage, metadata rights business logic and audio streaming.

Student Events/Career Development
EC8: WHISKEY TANGO FOXTROT—THE EVOLUTION OF AUDIO ENGINEERING IN MILITARY BANDS
Friday, October 20, 9:00 am – 10:30 am
Room 1E10

Moderator: **Brandie Lane**, United States Military Academy Band, West Point, NY, USA

Panelists: *Vince Magno*, Staff Sergeant, United States Military Academy Band
Noah Taylor, Staff Sergeant, United States Military Academy Band
Steven Van Dyne, Musician 1st Class, United States Navy Band

The United States military bands are considered public affairs resources and strategic tools that connect the United States Armed Forces with the world. Missions range from providing musical support for the President of the United States, bridging communication gaps in Middle Eastern communities, boosting the morale of troops overseas, inspiring America's future leaders, and creating a connection between the military and civilian population. Featuring a panel of audio engineers in different service branches, this workshop will discuss how audio techniques have evolved and adapted with the ever changing vision and goals of military music and how audio engineers support their organizations. Recording, live sound, and broadcast techniques will all be discussed, as well as projections for the future of audio in military bands.

Friday, October 20 9:00 am Room 1C03

Technical Committee Meeting on Networked Audio Systems

Standards Committee Meeting
SC-02-08 WORKING GROUP ON AUDIO-FILE TRANSFER AND EXCHANGE
Friday, October 20, 9:00 am – 10:30 am
Room 1C04

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.

Student Events/Career Development
EC15: THIS IS A MIX, THIS IS A MASTER
Friday, October 20, 9:30 am – 11:00 am
Room 1E06

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

Panelists: *Adam Ayan*, Gateway Mastering Studios, Inc.
Margaret Luthar, Chicago Mastering Service
Mandy Parnell, Black Saloon Studio

Most of the music mixes we hear and try to emulate have been professionally mastered. Too many novices try to recreate this "mastered" sound in their mix. This is undesirable and limits what the mastering engineer can do. Join our panel of mastering engineers at this listening session, as we compare "off-the-console" mixes with mastered versions, and discuss qualities desirable in mixes that allow the mastering engineer to do the best job possible.

Friday, October 20 10:00 am Room 1C03

Technical Committee Meeting on Perception and Evaluation of Audio Signals

Software@AES 32 Friday, October 20
10:00 am – 10:20 am Stage 3

ANTARES

Special Event
SE6: *PENSADO'S PLACE* LIVE FEATURING GREG WELLS
Friday, October 20, 10:15 am – 11:15 am
Room 1E15/16

Presenters: **Dave Pensado**
Herb Trawick

Join Dave and Herb for a special live version of their globally popular web TV series *Pensado's Place* with special guest Greg Wells! Producer, songwriter, musician, and mixer extraordinaire, Greg has produced such luminaries as Keith Urban, Katy Perry, Adele and One Republic and was recently on the cover of Modern Drummer. A staunch educational advocate, Greg is currently writing curriculum for his first book with the Strive/Hal Leonard imprint.

Aside from their discussion with Greg, which will be informative, musical, aspirational, and inspirational (with some laughs thrown in for good measure), Dave and Herb are bringing gifts and—perhaps—other guest stars! This AES-exclusive show is a blast you don't want to miss!

Live Sound Expo 00 Friday, October 20
10:15 am – 11:00 am Stage 1

POINT SOURCE OPTIMIZATION: 8 THINGS TO GET RIGHT

Presenter: **Kieran Eding**, KV2 Audio

Eight Things to Get Right with Point Source Systems: Height, Splay, Coupling, Fills, Delays, Amplification, Equalization and Rigging. Kieran Edling, KV2 AUDIO

Project Studio Expo 13 Friday, October 20
10:15 am – 12:00 noon Stage 2

NEW FRONTIERS IN PROJECT STUDIOS

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

Panelists: *Peter Hylenski*, Sound Designer, New York, NY, USA
Eddie Kramer, Remark Music Ltd., Woodland Hills, CA, USA
Scott M. Riesett, Producer/Engineer, New York, NY, USA

David Rosenthal, Musical Director, New York, NY, USA

Today's creative expectations, production methodologies, technology innovations, and business models has opened new frontiers for recording studios and particularly in the "Project Studio." In recent years, street buzz on the demise of the traditional recording studio has fostered a misperception of imminent extinction for the whole category. In fact, more studios are being built today than ever. What has and continues to change are the conventional business and production models and, the tools employed by artists, engineers and producers to record, mix, master and distribute their work in creative, cost effective and, sonically brilliant formats. This evolution continues to produce shifts in thinking about studio size, acoustic requirements, and architectural ergonomics. This 90-minute presentation will explore:

- Novel pre-fab acoustical treatments for low frequency control in small rooms;
- Creative design tips for maximizing limited space;
- Case Studies of recent project studio installations including previously unpublished photos and floor plans.

Standards Committee Meeting
SC-07-01 WORKING GROUP ON AUDIO METADATA
Friday, October 20, 10:30 am – 12:00 noon
Room 1C04

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.

Software@AES 33 **Friday, October 20**
10:30 am – 10:50 am **Stage 3**

LINE 6

Audio for Cinema 8 **Friday, October 20**
10:45 am – 12:00 noon **Room 1E07**

CONSTRUCTING A 5.1 HOME STUDIO FOR FILM, TV MIXING

Presenters: **Roger Guerin**, Ste. Adele, QC, Canada
Jean-Luc Louradour

Used to be a freelance mixer would come up to a mixing stage and after an hour or so he had a good take on the environment and was able to work. The room was calibrated, the equipment was serviced by trained personnel, and at least one resourceful assistant was available. Today everyone is looking to cut corners, the trained personnel has left, the calibration "was done two weeks ago," and the resourceful assistant has moved on. Thankfully the technology made giant leaps over the years, like the record business, with home recording; it's more and more viable to convert that extra room or garage into a mixing stage. This tutorial will demonstrate the ins and outs of a basic home mixing stage making use of off-the-rack materials with economical and ingenious solutions.

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

Broadcast/Streaming Medial 9 **Friday, October 20**
10:45 am – 12:15 pm **Room 1E08**

ADVANCES IN MICROPHONE CABLE TECHNOLOGY

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

This presentation outlines two new analog microphone cables from Belden.

Belden has been making cable for over 115 years. In that time, we've learned a thing or two about making cable. However, when it comes to audio cables, few significant advances have been made in more than a decade. It was felt that analog audio recording and reproduction has now reached a level of perfection that requires a new look at cable. Much of this surrounds the test and measurement of audio cables, and the results of our work will be shown compared to existing "standard" cables. Among the many tests made are those for resistance, capacitance, inductance all trackable as impedance. Tests will include the usual suspects, attenuation, frequency response (slope, tilt), distortion and shield effectiveness. Also shown will be common-mode rejection, group-delay (at audio frequencies) and other time effects, phase coherence (and other cancellation effects) of these new designs compared to existing "standard" designs. We will also delve into "listening tests" and similar controversial territory. This work has resulted in the first advanced analog cable. The cable itself will be shown and how best-ever performance was achieved.

The second analog audio microphone cable, Belden 8412P, is surprising mostly because nobody has produced anything like it before, as far as we know. It has two features which make it unique. First, to the best of our knowledge, it is the stiffest, LEAST flexible microphone cable made. It is also the first plenum-rated microphone cable. These two factors make this cable the perfect choice to solve some long-standing cable problems. These problems, and our solutions, will be discussed.

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

Product Development 7 **Friday, October 20**
10:45 am – 12:15 pm **Room 1E14**

AGILE AUDIO PRODUCT DEVELOPMENT

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

The days of designing everything from scratch are gone. Products are now too complicated and development times are too short to go it alone. This session focuses on recent advances in design tools for audio product developers. These tools streamline the development and tuning of real-time audio processing software. As a result, sophisticated audio features are now within reach of most companies, even those without large dedicated design teams. We walk you through a typical development cycle and highlight the benefits to design teams and organizations.

Recording & Production 10 **Friday, October 20**
10:45 am – 12:15 pm **Room 1E12**

COME TOGETHER: CONCEPTS IN AUDIO NORMALIZATION

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

Panelists: *Florian Camerer*, ORF, Vienna, Austria
Bob Katz
Bob Ludwig, Gateway Mastering Studios, Inc., Portland, ME, USA
George Massenburg, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
Gilbert Soulodre, Camden Labs

In order not to annoy listeners with audio levels jumping, an increasing number of delivery platforms now apply energy-based gain normalization of content by default. Loudness in broadcast is a well known example, the concept has become prevalent in music streaming, and soon personal players will be required to reduce the gain automatically to reduce the risk of causing hearing damage. This workshop includes an update from Europe and WHO concerning the latter, on Loudness in general, and on the two special case normalization strategies proposed for film (speech) and for music (album). The different concepts are discussed, and so is the potential influence these changes will have on our industry at large.

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

Sound Reinforcement 7 **Friday, October 20**
10:45 am – 12:15 pm **Room 1E09**

AC POWER, GROUNDING, AND SHIELDING

Presenters: **Bruce Olson**
Bill Whitlock, Whitlock Consulting, Oxnard, CA, USA

There is a lot of misinformation about what is needed for AC power for events. Much of it has to do with life-threatening advice. This panel will discuss how to provide AC power properly and safely and without having noise problems. This session will cover power for small to large systems, from a couple boxes on sticks up to multiple stages in ballrooms, road houses, and event centers; large scale installed systems, including multiple transformers and company switches, service types, generator sets, 1ph, 3ph, 240/120 208/120. Get the latest information on grounding and typical configurations the two leading experts in this field.

Networked Audio 7 **Friday, October 20**
10:45 am – 12:15 pm **Room 1E10**

HOW TO LEVERAGE AES67 TO INTEGRATE MULTIPLE NETWORKED AUDIO SYSTEMS

Presenters: **Andreas Hildebrand**, ALC NetworX GmbH, Munich, Germany
Will Hoult, Foxusrite
Greg Shay, The Telos Alliance, Cleveland, OH, USA

Experienced trainers will take participants through the workflow of configuring audio streaming from one networked audio platform to another using AES67. Attendees will see how AES67 complements the various AoIP solutions including Dante, RAVENNA, Q-LAN, and Livewire+ allowing for networked audio interoperability using standard IP networking techniques. Topics will include network configuration, stream setup in each of the platforms and troubleshooting.

Session P13 **Friday, Oct. 20**
11:00 am – 12:30 pm **Foyer**

POSTERS: TRANSDUCERS

11:00 am

P13-1 Equalization of Localized Sources on Flat-Panel Audio Displays—*Michael Heilemann, David Anderson, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

An equalization method is presented for sound sources rendered by eigenmode superposition on flat-panel audio displays. A filter is designed to provide a constant mechanical acceleration for each localized source region at all frequencies below the spatial aliasing frequency of the actuator array used to excite the panel's bending modes. Within this bandwidth, the vibration profile of the source remains consistent with the application of the equalization filter, preserving any spatial information conveyed to the listener from the source position. Directivity simulations and measurements show that these localized source regions do not exhibit the irregular directivity characteristic of single and multi-actuator distributed mode loudspeakers, but instead exhibit radiation characteristics similar to conventional piston loudspeakers within the array bandwidth.
Convention Paper 9871

11:00 am

P13-2 Loudspeaker 3D Directivity Estimation with First Order Microphone Measurements on a 2D Plane—*Lachlan Birnie, Thushara Abhayapala, Prasanga Samarasinghe*, Australian National University, Canberra, Australia

This paper proposes an efficient method to estimate the 3D directivity pattern of loudspeakers or portable devices with embedded speakers. We place the loudspeaker on a horizontal turntable and use a first order microphone located on the horizontal plane to measure pressure and pressure gradients along three orthogonal directions to construct equivalent virtual arrays of first order microphones on the horizontal plane. By exploiting the properties of the associated Legendre functions, we construct the 3D directivity pattern of the loudspeaker over frequencies. The method is equivalent to having a measurement setup consist of a dense spherical array encompassing the loudspeaker. The underlying theory and method are corroborated by simulations as well as measurements of the directivity of a physical loudspeaker.
Convention Paper 9872

11:00 am

P13-3 A Headphone Measurement System Covers both Audible Frequency and beyond 20 kHz (Part 3)—*Naotaka Tsunoda, Takeshi Hara, Koji Nageno*, Sony Video and Sound Corporation, Tokyo, Japan

New headphone frequency response measuring scheme was standardized as JEITA RC-8140B-1 in March 2016. The basic idea of the scheme is that the frequency response is to be measured by HATS and compensated by a free-field HRTF of HATS used in the measurement. One of the advantages of this measuring scheme is that obtained results have equivalent implication with the results of free-field frequency response of the loudspeakers. This report supplements the previous report that proposed the basic idea of above-mentioned scheme by adding topics regarding newly developed HATS to improve signal to noise ratio in high frequency areas above 20 kHz with ear simulators.
Convention Paper 9873

11:00 am

P13-4 Novel Type of MEMS Loudspeaker Featuring Membrane-Less Two-Way Sound Generation—*Fabian Stoppel,[1] Florian Niekiel,[1] Thorsten Giese,[1] Shanshan Gu-Stoppel,[1] Andreas Männchen,[2] Johannes Nowak,[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

In this paper a novel type of piezoelectric microelectromechanical loudspeaker is presented. The device concept is based on concentrically cascaded lead zirconate titanate actuators making it the first integrated two-way MEMS speaker reported. As a further novelty, the device is designed to operate without a closed membrane significantly improving the acoustic performance, energy efficiency, and manufacturability. Extensive finite element analysis studies have revealed a very high SPL of more than 79 dB in 10 cm distance at 500 Hz for a device 1 cm^2 in size operated at 30 V. At higher frequencies even larger SPL values are calculated enabling a flat frequency response with 89 dB for frequencies above 800 Hz. Based on this concept first speaker prototypes have been fabricated using MEMS technology.
Convention Paper 9874

11:00 am

P13-5 Analysis of the Mechanical Vibration and Acoustic Behavior of a Piezoelectric MEMS Microspeaker—
Andreas Männchen,[1] *Daniel Beer,*[1] *Florian Niekiel,*[2] *Johannes Nowak,*[1] *Fabian Stoppel,*[2] *Bernhard Wagner*[2]
[1]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[2]Fraunhofer Institute for Silicon Technology ISIT, Itzehoe, Germany

This paper investigates the performance of a piezoelectric MEMS-based microspeaker. The performance is compared to the state of the art in terms of electrodynamic microspeakers for mobile applications. The analysis is twofold: First, the mechanical behavior is evaluated using laser interferometry and discussed for different stimuli such as sine sweeps or static sinusoidal excitation. Second, the acoustic performance is assessed by way of measurements under anechoic conditions. Results show that the speaker performs well for its size while providing low power consumption. However, in order to achieve high broadband reproduction quality, further design improvements are necessary.
Convention Paper 9875

11:00 am

P13-6 Auditory-Based Smoothing for Equalization of Headphone-to-Eardrum Transfer Function—
Guangju Li,[1,2] *Ziran Jiang,*[1,2] *Jinqiu Sang,*[1,2] *Chengshi Zheng,*[1,2] *Renhua Peng,*[1,2] Xiaodong Li[1,2]
[1]Key Laboratory of Noise and Vibration Research, Institute of Acoustics, Chinese Academy of Sciences, Beijing, China
[2]University of Chinese Academy of Sciences, Beijing, China

Binaural headphone reproduction can be improved by equalization of headphone-to-eardrum transfer function (HETF) in an appropriate way. Direct inversion of HETF targeting at a flat frequency response cannot keep the peaks and valleys due to pinna and ear canal filtering that might help auditory perception. Moreover, Direct inversion might induce annoying high Q peak values due to variability across listeners. Smoothing the HETF before direct inversion can avoid over equalization. Two auditory-based spectral smoothing methods were studied in this research. One is based on roex filtering that can simulate human auditory filtering in the basilar membrane, and the other is cepstral smoothing that can simulate the auditory perception characteristic of frequency resolution. Subjective experiments show that, in comparison to direct inversion, the two proposed methods can improve binaural headphone reproduction.
Convention Paper 9876

11:00 am

P13-7 Interpolation and Display of Microphone Directivity Measurements Using Higher Order Spherical Harmonics—*Jonathan D. Ziegler*,[1,2] *Mark Rau*,[3] *Andreas Schilling*,[2] Andreas Koch[1]
[1]Stuttgart Media University - Stuttgart, Germany
[2]Eberhard Karls University Tübingen, Tübingen, Germany
[3]Center for Computer Research in Music and Acoustics, Stanford University, Stanford, CA, USA

The accurate display of frequency dependent polar response data of microphones has largely relied on the use of a defined set of test frequencies and a simple overlay of two-dimensional plots. In recent work, a novel approach to digital displays without fixed frequency points was introduced. Building on this, an enhanced interpolation algorithm is presented, using higher-order spherical harmonics for angular interpolation. The presented approach is compared to conventional interpolation methods in terms of computational cost and accuracy. In addition, a three-dimensional data processing prototype for the creation of interactive, frequency-dependent, three-dimensional microphone directivity plots is presented.
Convention Paper 9877

Student Events/Career Development
EC9: STUDENT RECORDING CRITIQUES
Friday, October 20, 11:00 am – 12:00 noon
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.

Friday, October 20 11:00 am Room 1C03

Technical Committee Meeting on High Resolution Audio

Software@AES 34 Friday, October 20
11:00 am – 11:20 am Stage 3

CELEMONY (MELODYNE)

Live Sound Expo 01 **Friday, October 20**
11:15 am – 12:00 noon **Stage 1**

DIGITAL VS. ANALOG WIRELESS ON THE STAGE

Presenters: **Christopher Evans**, The Benedum Center, Pittsburgh, PA, USA
Jason Glass, Clean Wireless
Vinny Siniscal, Firehouse
Karl Winkler, Lectrosonics, Rio Rancho, NM, USA

The majority of wireless systems in use today are analog, but digital systems are gaining. This panel explores the strengths and weaknesses of each, and issues arising from using them both together on the stage. Specifically, this all-star team will discuss bandwidth, latency, potential interference and intermods, and other challenges in the field, due to the imminent loss of 600 MHz and eventual spectrum crowding.

Recording & Production 16 **Friday, October 20**
11:30 am –12:30 pm **Room 1E13**

AL SCHMITT & KAT EDMONSON: THE ENGINEER/ARTIST STRATEGY

Moderator: **Mr Bonzai**

Panelists: *Kat Edmonson*
Al Schmitt

Grammy Award-winning recording engineer Al Schmitt and top jazz singer/songwriter Kat Edmonson will join moderator Mr. Bonzai for an exploration of the intense relationship between engineer and recording artist. Schmitt has won over twenty Grammy Awards for his work with Henry Mancini, Steely Dan, George Benson, Toto, Natalie Cole, Quincy Jones, and others. The genre-defying Kat Edmonson recently performed in Woody Allen's 2016 film set in the 1930s, "Café Society." Her most recent record debuted at #1 on the Billboard Heatseekers Albums Chart, #1 on Contemporary Jazz Chart and #2 on the Total Jazz Chart. Edmonson will explain her songwriting process and how she works up material before joining Schmitt in the studio. Together they will explore the connection between the art and science of capturing a performance at its peak, and offer personal anecdotes about the chemistry in the studio, including mic choices and vocal techniques, live room acoustics and overdubs, all leading up to the grand finale of the mix.

Software@AES 35 **Friday, October 20**
11:30 am – 11:50 am **Stage 3**

BLUE CAT AUDIO

Networked Audio 6 **Friday, October 20**
11:45 am – 12:15 pm **Room 1E11**

AV NETWORKING—A ONE-SIZE-FITS-ALL PROSPECT?

Presenter: **Henning Kaltheuner**

Discussions about media network solutions are ongoing within the professional AV industry and its corresponding applications and business fields. As new standard proposals continue to arise, the issue of "the one right network" is becoming increasingly debatable. This event outlines the different user and customer requirements for media networking solutions for a variety of applications and contexts such as live event production, broadcast, performance, and commercial installations. These requirements are matched with key characteristics of the most relevant currently discussed network standards, and why differing market demands drive network standards now and in the future.

Spatial Audio 18 **Friday, October 20**
12:00 noon – 1:00 pm **Room 1E06**

PMC: MIXING LIKE A PRODUCER

Presenter: **Chris Tabron**

[abstract not available]

Friday, October 20 **12:00 noon** **Room 1C03**

Technical Committee Meeting on Automotive Audio

Live Sound Expo 02 **Friday, October 20**
12:00 noon – 1:00 pm **Stage 1**

CHOOSING THE RIGHT VOCAL MIC

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

Software@AES 36 **Friday, October 20**
12:00 noon – 12:20 pm **Stage 3**

ACCUSONUS

Special Events
SE7: LUNCHTIME KEYNOTE: A COOKBOOK APPROACH TO SYSTEM OPTIMIZATION
Friday, October 20, 12:30 pm – 1:15 pm
Room 1E15/16

Presenter: **Bob McCarthy**

A master chef starts with carefully constructed recipes that combine together to make a great meal. This approach can be applied to system optimization, which is a combination of discrete operations that prepare each subsystem before all are combined into a whole. Lunch will be served along with the recipes for speaker aiming, crossover combination, splay angle setting, front fill spacing, cardioid sub arrays and more.

Student Events/Career Development
EC10: EDUCATION AND CAREER/JOB FAIR
Friday, October 20, 12:30 pm – 2:30 pm
Open Area

The combined AES 143rd 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 143rd 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.

Software@AES 37 **Friday, October 20**
12:30 pm – 12:50 pm **Stage 3**

SONARWORKS

Friday, October 20 **1:00 pm** **Room 1C03**

Technical Committee Meeting on Loudspeakers and Headphones

Standards Committee Meeting
SC-02-12 WORKING GROUP ON AUDIO APPLICATIONS OF NETWORKS
Friday, October 20, 1:00 pm – 3:00 pm
Room 1C04

The scope of SC-02-12 includes the use of various network types for audio and audio-related applications in professional recording and broadcasting.

Project Studio Expo 14 **Friday, October 20**
1:00 pm – 1:45 pm **Stage 2**

CREATIVE PROCESSING FOR AMBIENT MUSIC PRODUCTION

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

Covering the creation of sound beds from instruments, vocals, and "found sounds" with the use of processing, ambient treatments for conventional instruments such as guitar and piano, the creative abuse of pitch correction, favorite processing tools, utilizing mechanical and domestic sounds, creating "ear candy" or sonic punctuation through manipulation of short sounds extracts. Techniques for the mixing of ambient music will be covered, as well as specific treatments for bass and rhythm sounds.

Software@AES 38 **Friday, October 20**
1:00 pm – 1:20 pm **Stage 3**

CELEMONY (MELODYNE)

Session P14 **Friday, Oct. 20**
1:30 pm – 4:00 pm **Room 1E12**

PERCEPTION—PART 3

Chair: **Brecht de Man**, Queen Mary University of London, London, UK

1:30 pm

P14-1 A Statistical Model that Predicts Listeners' Preference Ratings of In-Ear Headphones: Part 2—Development and Validation of the Model—*Sean Olive, Todd Welti, Omid Khonsaripour*, Harman International, Northridge, CA, USA

Part 1 of this paper presented the results of controlled listening tests where 71 listeners both trained and untrained gave preference ratings for 30 different models of in-ear (IE) headphones. Both trained and untrained listeners preferred the headphone equalized to Harman IE target curve. Objective measurements indicated the magnitude response of the headphone appeared to be a predictor of its preference rating, and the further it deviated from the response of the Harman IE target curve the less it was generally preferred. Part 2 presents a linear regression model that accurately predicts the headphone preference ratings (r = 0.91) based on the size, standard deviation and slope of the magnitude response deviation from the response of the Harman IE headphone target curve.
Convention Paper 9878

2:00 pm

P14-2 Comparison of Hedonic and Quality Rating Scales for Perceptual Evaluation of High- and Intermediate Quality Stimuli—*Nick Zacharov, Christer Volk, Tore Stegenborg-Andersen*, DELTA SenseLab, Hørsholm, Denmark

In this study four rating scales for perceptual evaluation of Preference were compared: 9-point hedonic, Continuous Quality Scale (CQS) (e.g., used in ITU-R BS.1534-3 [1], "MUSHRA"), Labelled Hedonic Scale (LHS) [2], and a modified version of the LHS. The CQS was tested in three configurations to study the role and impact of the reference and anchor stimuli, namely: A full MUSHRA test with anchors and references, a test without references, and a test with neither references nor anchors. The six test configurations were tested with two groups of AAC codec qualities: High and Intermediate quality ranges. Results showed that the largest difference in scale usage were caused by having a declared reference, but also that the scale range usage is not strongly related to stimuli discrimination power.
Convention Paper 9879

2:30 pm

P14-3 Perceptual Evaluation of Source Separation for Remixing Music—*Hagen Wierstorf, Dominic Ward, Russell Mason, Emad M. Grais, Chris Hummersone, Mark D. Plumbley*, University of Surrey, Guildford, Surrey, UK

Music remixing is difficult when the original multitrack recording is not available. One solution is to estimate the elements of a mixture using source separation. However, existing techniques suffer from imperfect separation and perceptible artifacts on single separated sources. To investigate their influence on a remix, five state-of-the-art source separation algorithms were used to remix six songs by increasing the level of the vocals. A listening test was conducted to assess the remixes in terms of loudness balance and sound quality. The results show that some source separation algorithms are able to increase the level of the vocals by up to 6 dB at the cost of introducing a small but perceptible degradation in sound quality.
Convention Paper 9880

3:00 pm

P14-4 Adaptive Low-frequency Extension Using Auditory Filterbanks—*Sunil G. Bharitkar, Timothy Mauer, Charles Oppenheimer, Teresa Wells, David Berfanger*, HP, Inc., San Francisco, CA, USA

Microspeakers used in mobile devices and PCs have

band-limited frequency response, from constraining small drivers in tight enclosures, resulting in the loss of low-frequency playback content. The lack of low-frequencies in turn degrades the audio quality and perceived loudness. A method to overcome this physical limitation is to leverage the auditory phenomena of the missing fundamental; where by synthesizing the harmonic structure decodes the missing fundamental frequency. The proposed approaches employs side-chain processing for synthesizing the harmonics with only dominant portions of the low-frequency signal using critical-band filters. Additionally a parametric filter is used to shape the harmonics. Listening tests reveal that the proposed technique is preferred in terms of both the overall sound quality and the bass-only quality.
Convention Paper 9881

3:30 pm

P14-5 The Bandwidth of Human Perception and its Implications for Pro Audio—*Thomas Lund, Aki Mäkivirta*, Genelec Oy, Iisalmi, Finland

Locked away inside its shell, the brain has ever only learned about the world through our five primary senses. With them, we just receive a fraction of the information actually available, while we perceive far less still. A fraction of a fraction: The perceptual bandwidth. Conscious perception is furthermore subject to 400 ms of latency, and associated with a temporal grey-zone that can only be tapped into via reflexes or training. Based on a broad review of physiological, clinical and psychological research, the paper proposes three types of listening strategies we should distinguish between; not only in our daily lives, but also when conducting subjective tests: Easy listening, trained listening, and slow listening.
Convention Paper 9882

Broadcast/Streaming Media 10 — Friday, October 20
1:30 pm – 3:00 pm — Room 1E08

PODCAST PRODUCTION

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

Panelists: *Ramtin Arablouei*, NPR
Kate Bilinski, Homecoming, 36 Questions
Austin Gimlet, Gimlet Media
Jonathan Mitchell, The Truth

"Podcasting." Does it make you think of high-quality production, engrossing storytelling, and immersive field recordings? It should! Podcast audiences are growing and producers are investing, hiring and pushing boundaries. That means there is more opportunity than ever for engineers to create engaging audio experiences. This session brings together a number of experienced audio engineers — who are actively creating, sound designing, editing, and mixing podcast content at the top of the charts—to discuss the intricacies of their craft. Learn about their creative techniques, technical workflows, and strategies for working in the burgeoning industry.

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

Game Audio & VR 12 — Friday, October 20
1:30 pm – 3:00 pm — Room 1E06

THE "HORRIFIC" SOUND OF RESIDENT EVIL 7: BIOHAZARD

Presenters: **Akiyuki Morimoto**, Lead coomposerCapcoom
Ken Usami, Sound Designer, Capcom

The audio team behind Capcom's Resident Evil 7: Biohazard will discuss their methods to create realistic high-quality sound and the systems that allowed more flexibility and efficiency in sound implementation by introducing various automated tools. The team will also discuss about how the music design achieved the horror aspect of sound design and the game. The team chose to use a technique called "music-concrete" to cover a wide variety of sound texture, tones, and noises to deliver an uncomfortable and disturbing atmosphere in the game. They will also share their workflows of crafting the horror scene with sound by collaborating with level designers.

Product Development 8 — Friday, October 20
1:30 pm – 3:00 pm — Room 1E14

REUSING AND PROTOTYPING TO ACCELERATE INNOVATION IN AUDIO SIGNAL PROCESSING

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

Innovative devices like the Amazon Echo are disrupting big segments of the audio market and shifting consumer expectations on performance and capabilities of audio and voice interfaces. As new products are driven to deliver increasingly complex features, successful manufacturers and IP providers need to reuse more design assets, deliver original innovation more efficiently, and prototype more quickly than ever before. In this session, you will learn about different techniques to integrate existing code and IP into early simulations of algorithms and system designs, ranging from embeddable code to cloud-based services. You will also be exposed to quick prototyping workflows, including methods for running in real-time and validating ideas on live real-world signals. The presentation will go through practical worked examples using MATLAB, while discussing some early-stage challenges in the design of voice-driven connected devices.

Sound Reinforcement 8 — Friday, October 20
1:30 pm – 3:00 pm — Room 1E09

LIVE SOUND SUBWOOFER SYSTEM DESIGN

Presenter: **Adam J. Hill**, University of Derby, Derby, Derbyshire, UK; Gand Concert Sound, Elk Grove Village, IL, USA

There is little reason this day in age to accept undesirable low-frequency sound coverage in live sound reinforcement. The theories behind subwoofer system optimization are well-known within academia and various branches of industry, although this knowledge isn't always fully-translated into practical terms for end-users. This tutorial provides a comprehensive overview of how to achieve desirable low-frequency sound coverage including: subwoofer polar response control, array and cluster configuration, signal routing/processing options, performance stage effects, source decorrelation, acoustic barriers and perceptual considerations. The tutorial is suitable for practitioners, academics, and students alike, providing practical approaches to low-frequency sound control and highlighting recent technological advancements.

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

Tutorial/Workshop 8 — Friday, October 20
1:30 pm – 3:00 pm — Room 1E07

MACHINE LEARNING AND MUSIC RECOMPOSITION:

A GLIMPSE AT THE FUTURE

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

Panelists: *Jonathan Bailey*, iZotope
Tony Elfers, Sonixphere, Chicago, IL, USA
Patrick Stobbs, Jukedeck
Alex Tsilfidis, Accusonus
Charles Van Winkle, Adobe Systems Incorporated, Minneapolis, MN, USA

Machine learning, artificial intelligence… buzzwords or reality? In this session, learn how machine learning algorithms are impacting music composition, editing, and mixing. You'll hear from experts from Adobe, iZotope, Accusonus and Jukedeck as they share what's real now – and what's to come..

Special Event
SE8: MASTERING FOR THE NEW PARADIGM: HIGH RESOLUTION AUDIO, STREAMING AND HIGH RESOLUTION STREAMING
Friday, October 20, 1:30 pm – 3:00 pm
Room 1E15/16

Presenters: **Bob Ludwig**, Gateway Mastering Studios, Inc., Portland, ME, USA
Ian Shepherd, Mastering Media Ltd., Cambridge, Cambridgeshire, UK
Bob Stuart

This years Platinum Mastering Panel explores the new audio paradigms of high resolution audio and streaming. It is still so new that A&R departments are still screaming for maximum level when, in reality, streaming is starting to pay their bills and the big streaming companies will automatically turn your mastered level down so all music plays at a similar level. More consumers are listening to higher resolution audio than ever and streaming of the highest PCM sampling rates has been working successfully for 10 months now. Mastering engineers need to know about this seismic audio shift and they need to educate their clients, artists and record label associations about this new playing field.

We have 3 distinguished panelists to guide you through the new world order:

Bob Ludwig will speak about mastering high resolution audio and the differences between Mastered for iTunes and full high resolution media, downloads, streaming and of course vinyl.

Bob Stuart, co-founder of MQA will discuss how MQA can offer the highest guaranteed-quality streaming of even 352.8 kHz masters and tell you the basics of how it all works.

Ian Shepherd will discuss the various streaming services, how Loudness Normalization is the new paradigm, and the opportunity it offers mastering engineers.

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

Student Events/Career Development
EC11: TEACHING ELECTRONIC MUSICAL INSTRUMENT DESIGN: TECHNOLOGIES AND TECHNIQUES
Friday, October 20, 1:30 pm – 3:00 pm
Room 1E11

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

Panelists: *Ben Houge*, Berklee College of Music
Denis Labrecque, Analog Devices, San Jose, CA, USA
Matt Moldover
Teresa Marrin Nakra
Joe Paradiso

Designing musical instruments is an area that appeals to students in many disciplines across engineering and the arts. Schools and colleges have implemented programs in the field that encourage students to cross over into unfamiliar areas and collaborate with others with whom they otherwise would have little contact. It is constantly evolving as the technologies available to these programs get smaller, cheaper, and more user-friendly. Students in the field get to learn about the newest technologies in hands-on environments. The tangible results from these programs are not only highly rewarding, they sometimes become commercial products. Students go on to careers in the audio industry as software and hardware designers, product specialists, consultants, and entrepreneurs. The panel will include university faculty who direct these programs, instrument makers, and makers of the enabling technologies. The focus will be on how the programs will meet the industry's needs in the future.

Software@AES 39 **Friday, October 20**
1:30 pm – 1:50 pm **Stage 3**

INTERNET CO.

Spatial Audio 8 **Friday, October 20**
1:45 pm – 3:15 pm **Room 1E13**

IMMERSIVE AUDIO FOR MUSIC—WHY DO IT?

Presenters: **Stefan Bock**, msm Studios, Munich, Germany
Morten Lindberg, 2L Lindberg Lyd AS, Oslo, Norway
Daniel Shores, Sono Luminus, Boyce, VA, USA; Shenandoah Conservatory Music Production and Recording Technology, Winchester, VA, USA

The panel will discuss concepts in the future of immersive music. They will explore ideas in how to reach the masses, explain the current efforts, and the challenges of reaching the consumers. But most of all, they will examine the question; "why are we doing it?"

Project Studio Expo 15 **Friday, October 20**
1:45 pm – 3:00 pm **Stage 2**

FAB DUPONT AND SPECIAL GUEST ANN MINCIELI: THE SPECIAL SAUCE FOR MIXING A HIT RECORD

Presenters: **Fab Dupont**, Flux Studios, New York, NY, USA; Flux Studios
Ann Mincieli, Jungle City Studios, New York, NY, USA

Producer Fab Dupont (Mark Ronson, Jennifer Lopez) talks with Ann Mincieli (Beyoncé, Alicia Keys) as they walk through one of today's hottest tracks. Hear how the pros approach crafting a hit with the same tools available to you and what that "special sauce" is too.

Session P15 **Friday, Oct. 20**
2:00 pm – 3:30 pm **Foyer**

POSTERS: APPLICATIONS IN AUDIO

2:00 pm

P15-1 Noise Shaping Scheme Suppressing Quantization Noise Amount—*Akihiko Yoneya*, Nagoya Institute of Technology, Nagoya, Aichi-pref., Japan

A noise shaping scheme for multi-bit pulse-code-modulation suppressing the amount of the quantization noise is proposed. In the case with ordinary digital processing or digital-to-analog converters, noise is added over the whole frequency range and the shaped quantization noise of the source signal may only make the total signal-to-noise ratio worse. Therefore the amount of the quantization noise is preferable to be small even if the noise spectrum is shaped. In the proposed method, magnitude of the quantization noise is restricted at each sample and the optimal additional quantization pattern over a receding horizon with respect to the specified perception filter is searched in the look-ahead sigma-delta modulator manner. The amplitude of the quantization noise may be about 0.72 LSB regardless of the perception filter with the proposed method but a higher order perception filter requires a wide horizon of the optimization and a huge amount of the computation. An example is presented.
Convention Paper 9883

2:00 pm

P15-2 Evaluation of the Acoustics of the Roman Theater in Benevento for Discreet Listening Points—*Gino Iannace, Amelia Trematerra*, Universitá della Campania "Luigi Vanvitelli," Aversa, Italy

This work reports the acoustics of the Roman Theater in Benevento evaluated for discreet listening points positioned in the cavea in three radial directions. The theater, built in the second century A.D., was abandoned due to historical reasons and natural events. The recovery work ended in 1950. The theater is the center of important social activities. The theater acoustic measurements were taken by placing an omnidirectional spherical sound source on the stage and in the orchestra, with the microphone along three distinct radial directions on the steps of the cavea. The acoustic properties in the various seating areas were measured. The aim of the work is to evaluate in which sectors of the cavea the acoustic parameters are optimal for listening to different types of theatrical performances.
Convention Paper 9884

2:00 pm

P15-3 Modeling the Effects of Rooms on Frequency Modulated Tones—*Sarah R. Smith, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

This paper describes how reverberation impacts the instantaneous frequency tracks of modulated audio signals. Although this effect has been observed in a number of contexts, less work has been done relating these deviations to acoustical parameters of the reverberation. This paper details the instantaneous frequency deviations resulting from a sum of echoes or a set of resonant modes and emphasizes the conditions that maximize the resulting effect. Results of these models are compared with the observed instantaneous frequencies of musical vibrato tones filtered with the corresponding impulse responses. It is demonstrated that these reduced models may adequately reproduce the deviations when the signal is filtered by only the early or low frequency portion of a recorded impulse response.
Convention Paper 9885

2:00 pm

P15-4 New Research on Low-Frequency Absorption Using Membranes—*John Calder*, Acoustic Geometry, Minneapolis, MN, USA

Room modes are one of the greatest concerns when considering accurate sound recording and reproduction. Low-frequency (LF) absorbers are used to mitigate modes, however, most independent testing laboratories are only large enough to measure accurate absorption results above 160 Hz but not below. One lab is large enough to be accurate down to 40 Hz. A new LF absorber was designed to complement the capabilities of an original LF absorber. Summary: the type of absorber, and its location and orientation in a room, are all critical to LF absorber effectiveness. Without standardized laboratory absorption testing in a lab capable of accurately testing down to 40 Hz, it is difficult to state conclusively that low-frequency absorber products perform as claimed.
Convention Paper 9886

2:00 pm

P15-5 Analysis of Drum Machine Kick and Snare Sounds—*Jordie Shier, Kirk McNally, George Tzanetakis*, University of Victoria, Victoria, BC, Canada

The use of electronic drum samples is widespread in contemporary music productions, with music producers having an unprecedented number of samples available to them. The development of new tools to assist users organizing and managing libraries of this type requires comprehensive audio analysis that is distinct from that used for general classification or onset detection tasks. In this paper 4230 kick and snare samples, representing 250 individual electronic drum machines are evaluated. Samples are segmented into different lengths and analyzed using comprehensive audio feature analysis. Audio classification is used to evaluate and compare the effect of this time segmentation and establish the overall effectiveness of the selected feature set. Results demonstrate that there is improvement in classification scores when using time segmentation as a pre-processing step.
Convention Paper 9887

2:00 pm

P15-6 Dynamic Range Controller Ear Training: Description of a Methodology, Software Application, and Required Stimuli—*Denis Martin, George Massenburg, Richard King*, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

Several successful spectral ear training software applications are now available and being used by individuals and audio institutions around the world. While some listener training applications address other audio attributes, they have not received the same level of development and refinement. A methodology, software application, and the required stimuli for a dynamic range controller ear training program are described herein. This program, based on ideas developed for spectral ear training, addresses several limitations of earlier dynamic range controller ear training programs. It has been designed for web access, making use of the Web Audio API for audio processing, a custom audio compressor design, and a wide range of musical stimuli.
Convention Paper 9888

2:00 pm

P15-7 An "Infinite" Sustain Effect Designed for Live Guitar

Performance—*Mark Rau, Orchisama Das*, Center for Computer Research in Music and Acoustics (CCRMA), Stanford University, Stanford, CA, USA

An audio effect to extend the sustain of a musical note in real-time is implemented on a fixed point, standalone processor. Onset detection is used to look for new musical notes, and once they decay to steady state the audio is looped indefinitely until a new note onset occurs. To properly loop the audio, pitch detection is performed to extract one period and the new output buffer is written in a phase aligned manner.
Convention Paper 9889

Live Sound Expo 03 **Friday, October 20**
2:00 pm – 2:45 pm **Stage 1**

MIXING MONITORS FOR LARGE SCALE EVENTS

Presenter: **Jason Spence**, President-J Sound Services, Monitor Engineer (Keith Urban, Megadeth, SNL40th, CMT, CMA, ACM Billboard Awards), Nashville, TN, USA

Live events take significant amount of preparation, expertise, and finesse to pull off successfully. Learn the best practices about the art of mixing monitors from a seasoned professional in order to stay on the top of your game.

Software@AES 40 **Friday, October 20**
2:00 pm – 2:20 pm **Stage 3**

BEST SERVICE

Software@AES 41 **Friday, October 20**
2:30 pm – 2:50 pm **Stage 3**

ANTARES

Broadcast/Streaming Media 11 **Friday, October 20**
3:00 pm – 4:30 pm **Room 1E08**

METADATA FOR RADIO AND STREAMING: THE DIGITAL DASHBOARD

Moderator: **Glynn Walden**, CBS Radio Consultant, Marlton, NJ, USA

Panelists: *Stuart Buck*, Artic Palm
Mike Englebecht, NextRadio + TagStation
David Julian Gray, NPR
Mike Raide, DTS

The session is a panel discussion where each panelist will have 10-15 minutes for opening comments to be opened up to a panel / audience participation. This panel will discuss how the digital dashboard contributes rich textural support for audio for analog/ HD radio delivered over-the-air or via a hybrid radio and streaming services. The emphasis should be on creating a total audio and visual experience that can engage with the audience and increase listening time.

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

Friday, October 20 **3:00 pm** **Room 1C03**

Technical Committee Meeting on Semantic Audio Analysis

Standards Committee Meeting
SC-04-03 WORKING GROUP ON LOUDSPEAKER MODELING AND MEASUREMENT
Friday, October 20, 3:00 pm – 4:30 pm
Room 1C04

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.

Live Sound Expo 04 **Friday, October 20**
3:00 pm – 3:45 pm **Stage 1**

"IMMISSION MISSION"—NOISE PREDICTION FOR OUTDOOR EVENTS

Presenters: **Daniel Belcher**, d&b audiotechnik, Backnang, Germany
Nick Malgieri, d&b audiotechnik, Asheville, NC USA

The prediction of noise from outdoor events in neighboring residential and office areas is becoming more important. NoizCalc is a tool for accurately predicting noise immissions from sound reinforcement systems.

Software@AES 42 **Friday, October 20**
3:00 pm – 3:20 pm **Stage 3**

FABFILTER

Audio for Cinema 9 **Friday, October 20**
3:15 pm – 4:45 pm **Room 1E10**

BRIDGING THE GAP BETWEEN CREATIVITY & TECHNOLOGY: WORKING WITH COMPOSERS ON FILM AND MEDIA PROJECTS

Presenters: **Joe Carroll**, Manhattan Producers Alliance, New York, NY, USA
Frank Ferrucci, Manhattan Producers Alliance: VP, New York, NY, USA; Leenalisa Music: Composer/Producer

This session presented by NYC Composer/Producer and Manhattan Producers Alliance VP Frank Ferrucci gives a behind the scenes look into the technological challenges composers and engineers face when collaborating on film, television, and other visual media projects. The presentation addresses some less obvious but no less important ways that Music Engineers and Film Mixers can work best with composers and how technology can be used to help this collaboration be as seamless as possible.

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

Game Audio & VR 13 **Friday, October 20**
3:15 pm – 4:15 pm **Room 1E13**

VR ANALYTICS DEVELOPED FROM PERCEPTUAL AUDIO RESEARCH TO IMPROVE TOOLS AND PROCESSES FOR DEVELOPERS

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

Panelists: *Simon Calle*, Independent Research Engineer
Sean Olive, Harman International
Nils Peters, Qualcomm Technologies, Inc.
Nick Zacharov, Delta SenseLab

Leading experts in perceptual audio testing will discuss perceptual methods used to analyze user behavior in VR experiences. The panel will also discuss ongoing research at NYU that aims to quantize localization behavior that is useful feedback to Developers. They have created a Citizen Science game in a VR environment that can compare HRTF filtersets and spatial audio renderers in order to provide analytics on source accuracy and acquisition speed.

Networked Audio 5 **Friday, October 20**
3:15 pm – 4:15 pm **Room 1E11**

AES67 INTEROPERABILITY—NETWORK AUDIO SYSTEMS: A REPORT ON THE STATE OF THE TECHNOLOGY

Chair: **Nicolas Sturmel**, Digigram S.A., Montbonnot, France

Panelists: *Andreas Hildebrand*, ALC NetworX GmbH, Munich, Germany
Chris Roberts, BBC
Greg Shay, The Telos Alliance, Cleveland, OH, USA

The AES67 standard for High-performance streaming audio-over-IP interoperability was released in 2013. However, while the standard defines what protocols and functions need to be supported, it still leaves various choices open to the implementer. Interoperability between AES67 devices has been practiced many times at AES67 plug fests and IP Showcases since 2013. This workshop will mainly focus on how to achieve interoperability within the standard. The panelists will share their own experience in preparing for interoperability showcase or tests, designing products, configuring the network and accommodating the various choices open in the standards. Practical advice and tools to facilitate interoperability will also be given to the audience.

Product Development 9 **Friday, October 20**
3:15 pm – 4:45 pm **Room 1E14**

FRONT END AUDIO PROCESSING FOR VOICE ENABLED PRODUCTS

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

Voice recognition has become a sought-after feature in consumer and automotive audio products. Many OEMs are now scrambling to add these features to their products with little or no experience with microphone processing and many are struggling. This session focuses on the front end audio processing needed by a device to properly interface to a cloud based ASR engine. We cover beamforming, echo cancellation, direction of arrival estimation, and noise reduction. We show how the algorithms must be designed to work in concert for far field voice pickup and the difficult to achieve “barge in” feature. Performance metrics and evaluation procedures for the various algorithms are presented. Particular emphasis is given to the design of the microphone arrays and beamforming. We also present a novel metric that is correlated with performance and allows easy comparison of beamformer designs.

Sound Reinforcement 9 **Friday, October 20**
3:15 pm – 4:15 pm **Room 1E09**

MICROPHONE DRESSING FOR THEATER

Presenters: **John Cooper**, Local 1 IATSE, New York, NY. USA; Les Miserables Sound Dept
Mary McGregor, Freelance, Local 1, New York, NY, USA

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

Spatial Audio 9 **Friday, October 20**
3:15 pm – 4:15 pm **Room 1E06**

KRAFTWERK AND BOOKA SHADE —THE CHALLENGE TO CREATE ELECTRO POP MUSIC IN IMMERSIVE / 3D AUDIO FORMATS LIKE DOLBY ATMOS

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

Music has not a cinematic approach where spaceships are flying around the listener. Nonetheless, music can become a fantastic spatial listening adventure in immersive / 3D. How this sounds will be shown with the new Kraftwerk and Booka Shade Blu-ray releases this year. Production philosophies, strategies and workflows to create immersive / 3D in current workflows and DAWs will be shown and explained.

Special Events
SE9: DTV Audio Group AES FORUM
Friday, October 20, 3:15 pm – 6:00 pm
Room 1E15/16

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

Presenters: *Tim Carroll*, Senior Director, Office of the CTO, Dolby Laboratories
Roger Charlesworth, Executive Director, DTV Audio Group
Michael Englehaupt, Vice President and Chief Technology Officer, Graham Media Group
Stacey Foster, President Production Services, Broadway Video, Coordinating Producer, Saturday Night Live, Technical Consultant, Tonight Show with Jimmy Fallon
Jackie Green, President and Chief Technology Officer, Alteros
Scott Norcross, Manager Sound Platform Group, Office of the CTO, Dolby Laboratories
Jeffrey Riedmiller, Vice President, Sound Group, Office of the CTO, Dolby Laboratories
Sean Richardson, Executive Director and Principal Audio Engineer, Starz Entertainment
Tom Sahara, Vice President Operations and Engineering, Turner Sports, Chairman Sports Video Group
Steve Silva, Consultant Technology Strategy, Fox Networks Engineering and Operations
Jim Starzynski, Director and Principal Audio Engineer, NBC Universal, Chairman DTV Audio Group
James Stoffo, Chief Technology Officer and Principal Radio Active Designs

Television Audio in the Streaming Age: Mobile Takes the Forefront and the Spectrum Crunch Arrives

The explosion of television streaming is rapidly eclipsing traditional over-the-air broadcasting and MVPD distribution. While streaming opens the door to new features of personalization, accessibility, and immersive sound, the growing proliferation of mobile and fixed devices challenge us to adapt content to a range of listening conditions while attempting to maintain consistency and protect the creative intent of content providers.

"*The impact of streaming has up-ended the entire television industry. The migration from traditional broadcasting to an IP stream-based model will continue to accelerate the uptake of advanced encoding solutions with sophisticated audio services while creating new challenges of providing quality and consistency across an ever-widening range of device and environments*."~ Roger Charlesworth, Executive Director, DTV Audio Group

Please join the DTVAG for a discussion of these and other important television audio issues.

Discussion topics will include:

The Big Convergence: Streaming technology, social media, and IP infrastructure are beginning to converge to enhance the personalization of television content. To what degree will proliferating smart devices and aps connect with the growing intelligent media cloud to enable "self-driving" enhanced audio features? How are mobile video players and technology providers coming together with content creators to evolve our television listening experience?

The Challenges of Loudness and DRC Management in Mobile: As the center of gravity for television viewing shifts to mobile experiences, are the tools in place to appropriately manage target playback loudness and dynamic range for a variety of listening scenarios? Can existing metadata tools be better leveraged to address current challenges? What are the prospects for new CODEC-independent loudness and DRC metadata approaches?

Phase Zero: The Wireless Spectrum Crunch Starts Now: With the spectrum auction completed and carriers already rolling out services in their newly acquired 600 MHz blocs, production using wireless is already getting a lot trickier in some places. As the rollout in open blocs continues, and as stations begin to exit their existing allocations, things are about to get much more crowded. How bad will the crunch get and what emerging technologies or evolving practices can help to ease the inevitable crowding?

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

Student Events/Career Development
EC12: RECORDING COMPETITION—PART 2
Friday, October 20, 3:15 pm – 5:15 pm
Room 1E07

The Student Recording Competition is a highlight at each convention. A distinguished panel of judges participates in critiquing finalists of each category in an interactive presentation and discussion. The top three finalists in each category, as identified by our judges, present a short summary of their production intentions and the key recording and mix techniques used to realize their goals. They then play their projects for all who attend. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2) on Sunday afternoon. The competition is a great chance to hear the work of your fellow students at other educational institutions. Everyone learns from the judges' comments, even those who don't make it to the finals, and it's a great chance to meet other students and faculty.

Project Studio Expo 16 — Friday, October 20
3:15 pm – 4:15 pm — Stage 2

GEAR CLUB PODCAST SPECIAL EVENT

Presenters: **John Agnello**, Kurt Vile, Sonic Youth, New York, NY, USA
Stewart Lerman, Producer, New York, NY, USA

John Agnello is a New York City based music producer, engineer and mixer. His career started at the legendary Record Plant, where he worked on acclaimed albums by Cyndi Lauper, Aerosmith, and John Mellencamp. Since then he has worked with alt rock legends, Dinosaur Jr, Sonic Youth, Son Volt and Patti Smith and up and comers such as Phosphorescent, Kurt Vile and Twin Peaks.

Stewart Lerman is a 2x Grammy winning music producer, engineer and mixer based out of New York City. He has worked with Elvis Costello, Shawn Colvin, Neko Case, Sharon Van Etten, Patti Smith, The Roches, Antony and the Johnsons, and David Byrne. His film and tv credits include The Royal Tenenbaums, The Aviator, Cafe Society, and HBO's Vinyl, Grey Gardens and Boardwalk Empire.

Jack Douglas is a Grammy Award-winning producer/engineer, who worked with John Lennon to engineer Imagine, as well as many other artists including Aerosmith, Cheap Trick and Patti Smith.

Friday, October 20 — 4:00 pm — Room 1C03

Technical Committee Meeting on Audio Forensics

Recording & Production 12 — Friday, October 20
4:15 pm – 5:45 pm — Room 1E12

MUSIC MIXING, PART 4

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

Panelists: *Jim Anderson*, New York University, NYC
Michelle Desachy, Estudio 19, Mexico City
George Massenburg, McGill University, Montreal, Quebec, Canada
Ronald Prent, Wisseloord Studios, Holland

A continuation of the successful "Mixing Music" workshops at AES 139 (New York), AES 140 (Paris), and AES 142 (Berlin). A panel of award-winning expert practitioners from varying backgrounds and genres within the industry will spark interesting discussion and debate. Topics will include the process of mixing, techniques used, and proven methodologies that have yielded successful results over the years in a constantly changing industry. Focus will include the different ways to approach a mix, how to improve an existing mix, how to best interpret and address comments from clients. Balancing, use of processing, and listening levels will be addressed. Ample time will be reserved for questions, so that the audience will have a chance to solicit specific responses from the panel members.

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

Session EB4 — Friday, Oct. 20
4:30 pm – 5:45 pm — Room 1E11

TRANSDUCERS

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

4:30 pm

EB4-1 The Resonant Tuning Factor: A New Measure for Quantifying the Setup and Tuning of Cylindrical Drums—*Rob Toulson*, University of Westminter, London, UK

A single circular drumhead produces complex and in-harmonic vibration characteristics. However, with cylindrical drums, which have two drumheads coupled by a mass of air, it is possible to manipulate the harmonic relationships through changing the tension of the resonant drumhead. The modal ratio between the fundamental and the batter head overtone therefore provides a unique and quantified characteristic of the drum tuning setup, which has been termed as the Resonant Tuning Factor (RTF). It may be valuable, for example, for percussionists to manipulate the RTF value to a perfect musical fifth, or to simply enable a repeatable tuning setup. This research therefore considers a number of user interfaces for analyzing the RTF and providing a tool for quantitative drum tuning.
Engineering Brief 374

4:45 pm

EB4-2 Design and Implementation of a Practical Long-Throw High-Q CBT Array—*D. B. (Don) Keele, Jr.*,[1] *Marshall Kay*[2]
[1]DBK Associates and Labs, Bloomington, IN, USA
[2]Keysight Technologies, Apex, NC, USA

This paper describes the design and construction of a very-tall 5m experimental passive long-throw high-Q CBT array that provides coverage in a large church general-purpose activity room with a full-size basketball court. The room is 7.8 x 20 x 30 m (H x W x L). The 5 m tall 20° circular- arc array contains 80 ea 63.5 mm (2.5") full-range drivers, and provides a tight 15° vertical beamwidth. The mechanically aimed no-DSP passive segmented design is composed of five straight-front boxes each containing 16 drivers. Series-parallel connections, resistive attenuators, and two power amplifiers provide the frequency-independent four-bank CBT shading. This paper also provides detailed simulation data of the array's predicted beamwidth vs. frequency, directivity, vertical polar response, axial foot prints and predicted frequency response at three different downward tilt angles. The array provides very-even coverage along the entire length of the 30 m room.
Engineering Brief 375

5:00 pm

EB4-3 Effects of Acoustic Center Position in Subwoofers—*Mario Di Cola*,[1] *Paolo Martignon*,[1] *Merlijn van Veen*[2]
[1]Audio Labs Systems, Casoli, Italy
[2]Merlijn van Veen, Soest, Utrecht, The Netherlands

As explained by J.Vanderkooy [1] the acoustic center of a direct radiating subwoofer unit is placed ahead respect to the driver membrane, at a distance depending on driver and cabinet dimensions. This has effects on acoustic simulations and it deserves some attention to avoid errors. Measurements are shown which confirm acoustic center position theoretical calculation and a discussion is made about its effect on the definition of models for accurate simulations.
Engineering Brief 376
[eBrief presented by Paolo Martignon]

5:15 pm

EB4-4 Design and Implementation of a Constant-Directivity Two-Way 12" Woofer Wedge Loudspeaker System—*D. B. (Don) Keele, Jr.*,[1] *Hugh Sarvis*[2]
[1]DBK Associates and Labs, Bloomington, IN, USA
[2]KPresonus Audio Electronics-Worx Audio Technologies, Baton Rouge, LA, USA

This paper describes the design and implementation of a two-way constant-directivity wedge loudspeaker system that houses a single 12" woofer and eight 2" drivers in a 20° circular arc mounted on a curved baffle that covers the LF driver. An individual system comprises a 20° wedge box with a four-channel plate amplifier with two bridged channels driving the woofer, and the two other channels individually driving each half of the eight-driver array. This basic wedge box is then used in multiples to form larger circular-arc arrays of one up to six boxes making arrays that provide various vertical beamwidths in the range of 15° to 90°. Appropriate amplifier gains are chosen to smooth the polar coverage for each array size.
Engineering Brief 377

5:30 pm

EB4-5 A Tutorial on the Audibility of Loudspeaker Distortion at Bass Frequencies—*James Larson*,[1] *Gene DellaSaia*,[1] *D. B. (Don) Keele, Jr.*[2]
[1]Audioholics—Online A/V Magazine, South Elgin, IL, USA
[2]DBK Associates and Labs, Bloomington, IN, USA

This tutorial paper goes into detail concerning the audibility and perception of loudspeaker distortion at low frequencies. It draws on many past references and publications to summarize many of the factors that contribute to low-frequency loudspeaker distortion. Items covered include: types of distortion and audibility: linear vs. nonlinear, THD vs. IMD, auditory masking and distortion thresholds, measurements methods including continuous sine wave, two-tone IM, and tone-burst, among others. In conclusion, this paper does observe that distortion does occur, but by identifying the point at which distortion does become audible, one can be prudent in choosing which distortions to ignore.
Engineering Brief 378

Archiving/Restoration 8 — Friday, October 20
4:30 pm – 6:00 pm — Room 1E08

THE MUSIC NEVER STOPPED: THE FUTURE OF THE GRATEFUL DEAD EXPERIENCE IN THE INFORMATION AGE—PART 2

Chairs: **György Fazekas**, Queen Mary University of London, London, UK
Thomas Wilmering, Queen Mary University of London, London, UK

Panelists: *Jeremy Berg*, Cataloging Librarian, University of North Texas
Scott Carlson, Metadata Coordinator, Rice FondrenLibrary
Nicholas Meriwether, Director of Grateful Dead Archive, UCSC
John Meyer, CEO, Meyer Sound
Bryan Pardo, Assoc. Professor Electrical Engineering & Computer Science, Northwestern University

Ever since the advent of recording, technology has been constantly shaping the way we interact with music as well as the relationship between artists and fans. For instance, file compression and broadband internet disrupted conventional music distribution, creating new opportunities for the formation of online fan communities. A growing number of bands allow taping their live performances while there are expanding online archives, such as Etree and the Live Music Archive, for trading audio freely with permission between an increasing number of fans. The Grateful Dead and their fans the Deadheads bestow a prominent example with their innovative use of this technology. Semantic technologies are next in line, with a premise of step change in how we access audio archives. This workshop explores how semantic technologies provide enriched experiences for fans, greater exposure for bands and new opportunities for archives to flourish. We demonstrate new ways of navigating concert recordings, particularly those of the Grateful Dead, discuss opportunities and requirements with audio archivists and librarians, as well as the broader social and cultural context of how new technologies bear on music archiving and fandom.

These sessions are presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Game Audio & VR 14 **Friday, October 20**
4:30 pm – 6:00 pm **Room 1E13**

IMMERSIVE HEADPHONE AUDIO REPRODUCTION IN VR/AR

Chair: **Juergen Peissig**, Sennheiser Electronics, Wedemark, Germany

Panelists: *Linda Gedemer*, Technical Director / VR Audio Evangelist, Source Sound VR, Woodland Hills, CA USA
Jean-Marc Jot, Magic Leap, Inc. Plantation, FL, USA
Veronique Larcher, Sennheiser, Switzerland
Qing Zhang, Director Lightfield & Soundfield Reproduction Technology Lab, HUAWEI European Research Center, Munich, Germany

Major applications of virtual and augmented reality today are presenting with head mounted displays. This allows for mobility and unconstrained movements and is important especially for augmented reproduction to immersively embed the virtual into the real environment. Here headphone audio reproduction is mandatory to reproduce location dependent personalized 3D sound.

Audio rendering has to be of high perceptual quality to yield a high degree of spatial resolution, externalization and congruence of the virtual with the real acoustical environment. Fitting the augmented audio content to the real acoustical environment and the headphone being transparent for the real acoustical environment are important features.

The panelists will discuss various aspects of these challenges for signal processing and headphone acoustics under different AR & VR application requirements.

Sound Reinforcement 10 **Friday, October 20**
4:30 pm – 6:00 pm **Room 1E09**

INTERCOM SYSTEMS

Moderator: **Pete Erskine**, Best Audio, Mt. Vernon, NY, USA

Whether a simple a 2-wire intercom system with only a few stations or a complex digital matrix based system with a mixture of 2-wire, wireless, IP, and other technologies and scores of user stations, intercom is a mission critical part of any show and traditionally been the domain of the sound company/department. Intercom veterans share design and operational aspects for designing and deploying all types and sizes of intercom systems while minimizing system noise, controlling audio levels and assigning individual audio paths.

Spatial Audio 10 **Friday, October 20**
4:30 pm – 5:30 pm **Room 1E06**

NATIVE IMMERSIVE RECORDINGS

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

In this tutorial Sono Luminus head engineer will demonstrate and discuss the techniques and planning process, as well as play examples from numerous albums including Los Angeles Percussion Quartet, ACME, Lorelei Ensemble, Skylark, Iceland Symphony Orchestra, and others.

Standards Committee Meeting
SC-05-05 WORKING GROUP ON GROUNDING AND EMC PRACTICES
Friday, October 20, 4:30 pm – 6:00 pm
Room 1C04

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 crosstalk 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.

Archiving/Restoration 9 **Friday, October 20**
5:00 pm – 6:00 pm **Room 1E10**

THE ROOTS OF STEREOPHONY

Presenter: **Thomas Fine**, Sole proprietor of private studio, Brewster, NY, USA

A review of the roots and developments that led from 2-channel telephony in France and England in the late 1800s to the stereo LP in 1958. Reviewed will be key developments from Bell Labs, EMI/Blumlein, Magnetophone/German radio, Magnecord, Emory Cook and the beginnings of stereo recording by the major record labels. A brief examination of the related "aside" topic of "accidental stereo" recordings from the 78 era will also take place.

These sessions are presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Broadcast/Streaming Media 12 **Friday, October 20**
5:00 pm – 6:30 pm **Room 1E14**

THE AUDIO CREW OF THE LATE SHOW WITH STEPHEN COLBERT…. A SPECIAL BROADCAST EVENT

Presenters: **Final panel will be determined based on their production schedule**

Come and hear, even join in the conversation, about "a day in the life" of these unsung heroes in this rare opportunity as they are always on the job. Fascinating backgrounds and expertise come together in this truly stellar, award winning, audio team with a legacy of amazing music to create the sound of "The Late Show with Stephen Colbert;" a multi-layered, recorded live, broadcast television production complete with comedy monologues, roaming house band, comedy sketches, live animation, celebrity interviews, and guest musical performances.

Learn how they put together a cohesive, clear sound in The Ed Sullivan Theater while simultaneously running a recorded live broadcast and show in front of a live studio audience all in the vicinity of the neighboring Broadway Theaters' wireless signals! Not to mention, there have been a few occasions where they did it all Live On-Air!

Conversation Topics: •Overview of the audio design; •Integration of the various areas including PL & Coms; •Mixing perspectives for the different positions; •Mic placement/use considerations due to multiple uses per mic; • Specific challenges for "The Late Show" (roaming band, periodic live broadcast, live animation); •Specific challenges to The Ed Sullivan Theater (RF & proximity to Times Square, new challenges due to the re-opening of the dome ceiling, working within a NYC Historical Landmark Interior); •RF frequency coordination & techniques; •Post production (editing & systems); •Actual broadcast; •Director's needs; •Plans for future technology and work flow; •Q & A

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

Friday, October 20 5:00 pm Room 1C03

Technical Committee Meeting on Audio for Cinema

Networked Audio 8 Friday, October 20
5:30 pm – 6:00 pm Room 1E07

AES67 PRACTICAL

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

The AES67 Standard on High performance Streaming Audio-over-IP Interoperability was published in September 2013. Since then, first applications with AES67-compatible devices have been projected and put into operation. This session will give insight and provide tips on system planning, device and network configuration, management, and monitoring.

Special Events
SE10: HEAR NOW FESTIVAL PRESENTATION:
AUDIO TONIGHT: A STORY-TELLERS CABARET
Friday, October 20, 7:30 pm – 9:00 pm
Dolby Laboratories NY Screening Room
1350 Ave. of the Americas Main Floor
(Doors open at 7:00 pm – show begins at 7:30)
Seating is limited to 60 persons; tickets required

Join the HEAR Now Festival on Friday, October 20, 2017, at the Dolby Theater (1350 6th Ave, New York) beginning at 7:30pm as they celebrate the art of audio story-telling at this 90 minute live show.

From original radio plays and sketch comedy to narration **AUDIO Tonight!**, and host Barbara Rosenblat (*Orange Is the New Black, Bug Diaries*, Audie winner) along with Sue Zizza (SueMedia Productions/ HEAR Now Festival), take you on a fun and entertaining tour of what you might hear and see throughout the annual HEAR Now Festival weekend.

Meet "Mark Twain" as premiere voice actor Bob Kaliban introduces you to HEAR Now's Mark Twain narrators.

NYC's VoiceScapes Audio Theater presents new and original audio plays with award winning voice artists Barbara Rosenblat, Robin Miles and more—and don't forget the live SFX and MUSIC!!

Binaural Shakespeare, directed by Amanda Rose Smith and Neil Hellegers, features Robert Fass as MacBeth, with Lisa Flanagan, Robin Miles, and Sarah Mollo-Christensen as the three witches, and the voices of Bob Kaliban.

From Producer William Dufris comes a special "sneak preview" of the newest installment of Audible's *The X-Files* featuring David Duchovny and Gillian Anderson which won't be available till HALLOWEEN!!

Sponsored by the Broadcast and Streaming Media Track

Session P16 Saturday, Oct. 21
9:00 am – 12:30 pm Room 1E11

SPATIAL AUDIO—PART 2

Chair: **Jean-Marc Jot**, Magic Leap, Sunnyvale, CA, USA

9:00 am

P16-1 On Data-Driven Approaches to Head-Related-Transfer Function Personalization—*Haytham Fayek*,[1,2] *Laurens van der Maaten*,[2] *Griffin Romigh*,[1] *Ravish Mehra*[1]
[1]Oculus Research, Redmond, WA, USA
[2]Facebook AI Research, New York, NY, USA

Head-Related Transfer Function (HRTF) personalization is key to improving spatial audio perception and localization in virtual auditory displays. We investigate the task of personalizing HRTFs from anthropometric measurements, which can be decomposed into two sub tasks: Interaural Time Delay (ITD) prediction and HRTF magnitude spectrum prediction. We explore both problems using state-of-the-art Machine Learning (ML) techniques. First, we show that ITD prediction can be significantly improved by smoothing the ITD using a spherical harmonics representation. Second, our results indicate that prior unsupervised dimensionality reduction-based approaches may be unsuitable for HRTF personalization. Last, we show that neural network models trained on the full HRTF representation improve HRTF prediction compared to prior methods.
Convention Paper 9890

9:30 am

P16-2 Eigen-Images of Head-Related Transfer Functions—*Christoph Hold, Fabian Seipel, Fabian Brinkmann, Athanasios Lykartsis, Stefan Weinzierl*, Technical University of Berlin, Berlin, Germany

The individualization of head-related transfer functions (HRTFs) leads to perceptually enhanced virtual environments. Particularly the peak-notch structure in HRTF spectra depending on the listener's specific head and pinna anthropometry contains crucial auditive cues, e.g., for the perception of sound source elevation. Inspired by the eigen-faces approach, we have decomposed image representations of individual full spherical HRTF data sets into linear combinations of orthogonal eigen-images by principle component analysis (PCA). Those eigen-images reveal regions of inter-subject variability across sets of HRTFs depending on direction and frequency. Results show common features as well as spectral variation within the individual HRTFs. Moreover, we can statistically de-noise the measured HRTFs using dimensionality reduction.
Convention Paper 9891

10:00 am

P16-3 A Method for Efficiently Calculating Head-Related Transfer Functions Directly from Head Scan Point Clouds—
Rahulram Sridhar, Edgar Choueiri, Princeton University, Princeton, NJ, USA

A method is developed for efficiently calculating head-related transfer functions (HRTFs) directly from head scan point clouds of a subject using a database of HRTFs, and corresponding head scans, of many subjects. Consumer applications require HRTFs be estimated accurately and efficiently, but existing methods do not simultaneously meet these requirements. The presented method uses efficient matrix multiplications to compute HRTFs from spherical harmonic representations of head scan point clouds that may be obtained from consumer-grade cameras. The method was applied to a database of only 23 subjects, and while calculated interaural time difference errors are found to be above estimated perceptual thresholds for some spatial directions, HRTF spectral distortions up to 6 kHz fall below perceptual thresholds for most directions.
Convention Paper 9892

10:30 am

P16-4 Head Rotation Data Extraction from Virtual Reality Gameplay Using Non-Individualized HRTFs—*Juan Simon Calle, Agnieszka Roginska*, New York University, New York, NY, USA

A game was created to analyze the subject's head rotation during the process of localizing a sound in a 360-degree sphere in a VR gameplay. In this game the subjects are asked to locate a series of sounds that are randomly placed in a sphere around their heads using generalized HRTFs. The only instruction given to the subjects is that they need to locate the sounds as fast and accurate as possible by looking at where the sound was and then pressing a trigger. To test this tool 16 subjects were used. It showed that the average time that it took the subjects to locate the sound was 3.7±1.8 seconds. The average error in accuracy was 15.4 degrees. The average time that it took the subjects to start moving their head was 0.2 seconds approximately. The average rotation speed achieved its maximum value at 0.8 seconds and the average speed at this point was approximately 102 degrees per second.
Convention Paper 9893

11:00 am

P16-5 Relevance of Headphone Characteristics in Binaural Listening Experiments: A Case Study—*Florian Völk,[1,2] Jörg Encke,[1] Jasmin Kreh,[1] Werner Hemmert[1]*
[1]Technical University of Munich, Munich, Germany
[2]WindAcoustics, Windach, Germany

Listening experiments typically target performance and capabilities of the auditory system. Another common application scenario is the perceptual validation of algorithms and technical systems. In both cases, systems other than the device or subject under test must not affect the results in an uncontrolled manner. Binaural listening experiments require that two signals with predefined amplitude or phase differences stimulate the left and right ear, respectively. Headphone playback is a common method for presenting the signals. This study quantifies potential headphone-induced interaural differences by physical measurements on selected circum-aural headphones and by comparison to psychoacoustic data. The results indicate that perceptually relevant effects may occur, in binaural listening experiments, traditional binaural headphone listening, and virtual acoustics rendering such as binaural synthesis.
Convention Paper 9894

11:30 am

P16-6 Evaluating Binaural Reproduction Systems from Behavioral Patterns in a Virtual Reality—A Case Study with Impaired Binaural Cues and Tracking Latency—*Olli Rummukainen, Sebastian Schlecht, Axel Plinge, Emanuël A. P. Habets*, International Audio Laboratories Erlangen, Erlangen, Germany

This paper proposes a method for evaluating real-time binaural reproduction systems by means of a wayfinding task in six degrees of freedom. Participants physically walk to sound objects in a virtual reality created by a head-mounted display and binaural audio. The method allows for comparative evaluation of different rendering and tracking systems. We show how the localization accuracy of spatial audio rendering is reflected by objective measures of the participants' behavior and task performance. As independent variables we add tracking latency or reduce the binaural cues. We provide a reference scenario with loudspeaker reproduction and an anchor scenario with monaural reproduction for comparison.
Convention Paper 9895

12:00 noon

P16-7 Coding Strategies for Multichannel Wiener Filters in Binaural Hearing Aids—*Roberto Gil-Pita,[1] Beatriz Lopez-Garrido,[2] Manuel Rosa-Zurera[1]*
[1]University of Alcalá, Alcalá de Henares, Madrid, Spain
[2]Servicio de Salud de Castilla la Mancha (SESCAM), Castilla-Mancha, Spain

Binaural hearing aids use sound spatial techniques to increase intelligibility, but the design of the algorithms for these devices presents strong constraints. To minimize power consumption and maximize battery life, the digital signal processors embedded in these devices have very low frequency clocks and low amount of available memory. In the binaural case the wireless communication between both hearing devices also increases the power consumption, making necessary the study of relationship between intelligibility improvement and required transmission bandwidth. In this sense, this paper proposes and compares several coding strategies in the implementation of binaural multichannel Wiener filters, with the aim of keeping minimal communication bandwidth and transmission power. The obtained results demonstrate the suitability of the proposed coding strategies.
Convention Paper 9896

Session P17 **Saturday, Oct. 21**
9:00 am – 12:00 noon **Room 1E12**

APPLICATIONS IN AUDIO

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

9:00 am

P17-1 Challenges of Audio Forensic Evaluation from Personal Recording Devices—*Robert C. Maher*, Montana State University, Bozeman, MT, USA

Typical law enforcement audio forensic investigations

involve audio evidence recorded under less-than-ideal circumstances by mobile phones, surveillance systems, and personal audio recorders. Moreover, the audio information is often transmitted and stored using a data compression algorithm such as a speech coder (e.g., VSELP) or a wideband audio coder (e.g., MP3). There are few systematic studies of the signal behavior of these systems for forensically-relevant audio material, and this may discourage a forensic examiner from using such acoustic evidence to draw reliable conclusions. This paper includes simulation and evaluation of personal audio recording systems in the context of audio forensics. The results indicate areas of strength and weakness in the forensic realm.
Convention Paper 9897

9:30 am

P17-2 An Acoustic Study of Airbag Deployment in Vehicles—*John Vanderkooy, Kevin Krauel,*
University of Waterloo, Waterloo, Ontario, Canada

This study shows the acoustic pressures produced in typical airbag deployments and introduces the topic to the AES. Two representative vehicles were tested: a 2005 Pontiac Montana SV6 minivan and a 2006 Mazda 3 hatchback. Microphones were placed at the left driver ear, right passenger ear, and rear seat positions. Wideband pressure data was obtained for each of the steering wheel, passenger, and any optional side airbags. Our data agrees with the plethora of studies of earlier work. Weighted and unweighted peak SPL levels are calculated for various deployment scenarios. The influence of the cabin volume and the vents of the vehicles are discussed. Concerns over hearing loss, possible eardrum perforation, and other hearing-related symptoms are considered, gleaned mainly from important earlier studies. Some aspects are counterintuitive.
Convention Paper 9898

10:00 am

P17-3 CLEAR: Conditionally Lossless Encoding under Allowed Rates for Low-Delay Sound Data Transmission—
Ryosuke Sugiura, Yutaka Kamamoto, Noboru Harada, Takahito Kawanishi, Takehiro Moriya, NTT Communication Science Labs, Atsugi-shi, Kanagawa-ken, Japan

We present in this paper a near-lossless full-band stereo compression scheme, Conditionally Lossless Encoding under Allowed Rates (CLEAR), aiming at its use in real-time transmission of sound data, sounds to be mixed or processed after transmitted. Using uniform quantizer with MPEG-4 Audio Lossless Coding (ALS) and adaptive pre- and post-processing, CLEAR controls the encoding bit rate with maximum fidelity of reconstructed signals. Objective experiments show an enhancement in signal to noise ratio (SNR) and from conventional low-delay codecs with compatible perceptual quality. Additionally, companding-based perceptual weighting designed for CLEAR is shown to make an improvement in Perceptual Evaluation of Audio Quality (PEAQ).
Convention Paper 9899

10:30 am

P17-4 A New THD+N Algorithm for Measuring Today's High Resolution Audio Systems—*Alfred Roney,*[1] *Steve Temme*[2]
[1]Mathworks Inc., Framingham, MA, USA (formerly at Listen Inc.)
[2]Listen, Inc., Boston, MA, USA

We present a mathematical definition of Total Harmonic Distortion + Noise suitable for testing high-resolution digital audio systems. This formal definition of the "distortion analyzer" mentioned in AES17 defines THD+N as the RMS error of fitting a sinusoid to a noisy and distorted sequence of measurements. We present the key theoretical result that under realistic conditions a modern THD+N analyzer is well-described by a Normal probability distribution with a simple relationship between relative error and analysis dwell time. These findings are illustrated by comparing the output of a commercial distortion analyzer to our proposed method using Monte Carlo simulations of noisy signal channels. We will demonstrate that the bias of a well-designed distortion analyzer is negligible.
Convention Paper 9900

11:00 am

P17-5 Influences of a Key Map on Soundwalk Exploration with a Textile Sonic Map—*Alessia Milo, Nick Bryan-Kinns, Joshua D. Reiss,* Queen Mary University of London, London, UK

Sonic maps are an increasingly popular form of exploring soundscapes and are a possible means of communicating the experience of a soundwalk. We describe how a printed key influenced exploration of an interactive textile sonic map. We explain the technology behind the map, employing capacitive sensing and real-time audio processing. The sonic map contained 18 binaural recordings extracted from a soundwalk. Thirty participants explored the map. The strengths and limitations of the interfaces were established, and participants' modes of exploration were identified. Results show how the use of the key map levelled the location preference. The participants' experience with the interface suggested possible uses of e-textiles for soundscape awareness promotion and studies and in the field of interactive audio.
Convention Paper 9901

11:30 am

P17-6 Challenges of IoT Smart Speaker Testing—
Glenn Hess,[1] *Daniel Knighten*[2]
[1]Indy Acoustic Research, Indianapolis, IN, USA
[2]Listen, Inc., Boston, MA, USA

Quantitatively measuring the audio characteristics of IoT (Internet of Things) smart speakers presents several novel challenges. We discuss overcoming the practical challenges of testing such devices and demonstrate how to measure frequency response, distortion, and other common audio characteristics. In order to make these measurements, several measurement techniques and algorithms are presented that allow us to move past the practical difficulties presented by this class of emerging audio devices. We discuss test equipment requirements, selection of test signals, and especially overcoming the challenges around injecting and extracting test signals from the device.
Convention Paper 9902

Session EB5 **Saturday, Oct. 21**
9:00 am – 10:30 am **Poster Area**

POSTERS—PART 2

9:00 am

EB5-1 Impulse and Radiation Field Measurements for Single Exciter versus Exciter Array Flat-Panel Loudspeakers—
David Anderson, Michael Heilemann, Mark F. Bocko,
University of Rochester, Rochester, NY, USA

Flat-panel loudspeakers with single exciters exhibit significant directivity shifts and many discrete resonances in frequency regions of low modal density. These phenomena are demonstrated through mechanical and acoustic measurements on an acrylic and a glass prototype panel, both with single exciters. The measurements are repeated for the acrylic panel using an array of exciters where the force magnitude of each exciter is specified to actuate only the lowest-index bending mode. The mechanical measurements demonstrate that no modes in the array-addressable frequency region above the first mode are actuated. Acoustically, measurements show omnidirectional radiation with a single low-frequency resonance, showing how the exciter array enables the flat panel to behave similarly to a conventional loudspeaker within the array-addressable frequency region.
Engineering Brief 379

Engineering Brief 380 was withdrawn

9:00 am

EB5-2 Implementation of a Dipole Constant Directivity Circular-Arc Array—*Kurtis Manke,[1] Richard Taylor,[1] Mark Paetkau,[1], D. B. (Don) Keele, Jr.[2]*
[1]Thompson Rivers University, Kamloops, Canada
[2]DBK Associates and Labs, Bloomington, IN, USA

We briefly present the theory for a broadband constant-beamwidth transducer (CBT) formed by a conformal circular-arc array of dipole elements previously developed in seminal works. This technical report considers a dipole CBT prototype with cosine amplitude shading of the source distribution. We show that this leads to a readily-equalizable response from about 100 Hz to 10 kHz with a far-field radiation pattern that remains constant above the cutoff frequency determined by the beam-width and arc radius of the array, and below the critical frequency determined by discrete element spacing at which spatial aliasing effects occur. Furthermore, we show that the shape of the radiation pattern is the same as the shading function, and remains constant over a broad band of frequencies.
Engineering Brief 381

Engineering Brief 382 was withdrawn

9:00 am

EB5-3 Flexible Control of the Transducer and the Duct Resonance of a Speaker System Ducted to the Exterior of a Vehicle Cabin—*Takashi Kinoshita,[1] John Feng[2]*
[1]Bose Automotive G.K. - Tokyo, Tokyo, Japan
[2]Bose, Framingham, MA, USA

In order to reproduce lower frequency sound in a vehicle cabin efficiently, Zeljko Velican proposed a speaker system, where the backside of a transducer unit communicates with the exterior of a vehicle cabin via a tuned acoustic appliance. [1] Since this speaker system couples the interior and the exterior of a vehicle cabin, the efficiency and the frequency range of internal and external noise transmission are both important considerations. These two characteristics are strongly correlated with the two dominant resonances of the system. One is the mechanical resonance of the transducer which defines the lower limit of the sound reproduction frequency range. Another one is the Helmholtz resonance of the back-side acoustic appliance (enclosure and duct), which defines the frequency where, for example, noise transmission through the appliance is optimized. Choosing the appropriate acoustic parameters to balance those two dominant resonances is the key to optimal design this speaker system. But with the existing configuration [1], these two dominant acoustic resonances have strong mutual interaction via coupled design parameters, it can be difficult to find a good compromise between them. In this paper, a new speaker system configuration, consists of a transducer, an enclosure ducted to the exterior of the vehicle cabin, and a passive radiator to cover the duct, will be proposed and discussed. With this configuration, the two dominant resonances of the system can be controlled quasi-individually, therefore enhancing design flexibility for the practical use of such systems on a vehicle.
Engineering Brief 383

9:00 am

EB5-4 Multichannel Microphone Array Recording for Popular Music Production in Virtual Reality—*Hashim Riaz,[1] Mirek Stiles,[2] Calum Armstrong,[1] Andrew Chadwick,[1] Hyunkook Lee,[3] Gavin Kearney[1]*
[1]University of York, York, UK
[2]Abbey Road Studios, London, UK
[3]University of Huddersfield, Huddersfield, UK

There is a growing market for innovative ways to appreciate and listen to music through new Virtual Reality (VR) experiences made accessible through smartphones and VR headsets. However, production workflows for creating immersive musical experiences over VR are still in their infancy. This engineering report documents different microphone configurations and recording techniques applied in a higher-order Ambisonic processing framework to deliver an engaging and hyper-real interactive VR music experience. The report documents a live popular music recording undertaken at Abbey Road with traditional music recording techniques such as spot and stereo microphone setups and advanced techniques using dedicated VR multichannel microphone arrays.
Engineering Brief 384

9:00 am

EB5-5 *Engineering Brief 385 was withdrawn*

9:00 am

EB05-6 Consonant Perception and Improved S/N Ratio Using Harmonic Tracking Equalization—*Al Yonovitz,[1] Silas Smith,[1] David Yonovitz[2]*
[1]University of Montana, Missula, MT, USA
[2]Key 49, Del Mar, CA, USA

Audio equalization techniques are often used to enhance signals and reduce noise. These include Shelf, Parametric, and Graphic Equalizers. These techniques modify spectral components within specified bands by applying gain or attenuation. Another promising technique utilizes the tracking of harmonics and sub-harmonics (HTEq). These harmonics may be individually changed in intensity. This study utilized 21 Consonant Vowel (CV) stimuli with a white noise masker (+6 dB S/N). Each stimulus was randomly presented to listeners. Confusion matrices determined consonant intelligibility and information transmission for distinctive features. Perceptually, after HTEq, the noise was minimally audible and required considerably less effort to identify consonants. The results indicated the distinctive feature transmission was not altered. Comparisons were made for consonants at various levels of noise reduction
Engineering Brief 386

9:00 am

EB5-7 Bridging Fan Communities and Facilitating Access to Music Archives through Semantic Audio Applications—

Thomas Wilmering, Florian Thalmann, György Fazekas, Mark B. Sandler, Queen Mary University of London, London, UK

Semantic Audio is an emerging field in the intersection of signal processing, machine learning, knowledge representation, and ontologies unifying techniques involving audio analysis and the Semantic Web. These mechanisms enable the creation of new applications and user experiences for music communities. We present a case study focusing on what Semantic Audio can offer to a particular fan base, that of the Grateful Dead, characterized by a profoundly strong affinity with technology and the internet. We discuss an application that combines information drawn from existing platforms and results from the automatic analysis of audio content to infer higher-level musical information, providing novel user experiences particularly in the context of live music events.
Engineering Brief 387

9:00 am

EB5-8 The ANU School of Music Recording Studios: Design, Technology, Research, and Pedagogy—*Samantha Bennett, Matt Barnes*, Australian National University, Canberra, Australia

This engineering brief addresses the refurbishment process of the School of Music, Australian National University recording studios to include focus on the historical, pedagogical and research requirements of a 21st Century studio facility. The brief will first address issues of space, heritage and purpose before considering the acoustic (re)design process. Furthermore, the brief examines issues of technological integration and facilitation of analogue, digital and hybrid workflows. Finally, the brief considers the research and pedagogical remit of the refurbished facilities.
Engineering Brief 397

Networked Audio 9 — Saturday, October 21
9:00 am – 10:30 am — Room 1E08

HOW TO MAKE AN AES70 DEVICE

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

Making an AES70-controllable device is not unduly difficult, once you have a basic understanding of the AES70 device model. This session will present the recommended steps in designing a device's AES70 control interface, and discuss realities of implementing it. One or more examples will be given.

Product Development 10 — Saturday, October 21
9:00 am – 10:30 am — Room 1E14

IS THE ANECHOIC CHAMBER OBSOLETE?

Presenter: **Christian Bellmann**, Klippel GmbH, Dresden, Germany

Anechoic rooms have been considered as an expensive but optimum way for assessing loudspeakers under free field conditions. This tutorial discusses the physical limitations of the room and gives practical advice how to avoid systematic measurement errors. In the second part alternative techniques are discussed that allow to provide measurement results under simulated free field conditions. The tutorial also explains common errors generated when a relatively short window is applied to non-anechoic measurement data to extract the direct sound and harmonic distortion (Farina). The tutorial also shows the new opportunities of the holographic measurements based on near field scanning applied to drivers operated in a baffle and complete system. This technique provides highly accurate reference data which can be used for correcting other measurements performed at a single measurement point under non-anechoic conditions by applying inverse filtering of the microphone signal prior to the signal analysis. This tutorial illustrates the conventional and new method on practical examples.

Sound Reinforcement 4 — Saturday, October 21
9:00 am – 10:15 am — Room 1E09

THEATRICAL SOUND DESIGN

Moderator: **Nevin Stein**, Modern Projects Inc., New York, NY, USA

Panelists: *Kai Harada*
Scott Lehrer
Nicholas Pope

Since the late 1980's and the introduction of digital signal processing, Broadway sound designers have explored ways of closely associating the amplified signal with its acoustic source in time and space. Localization techniques have developed around increasingly sophisticated technology. In a panel discussion, four leading Broadway sound designers discuss both their localization approach and recent productions to illustrate the state of the art.

Spatial Audio 11 — Saturday, October 21
9:00 am – 10:00 am — Room 1E13

CREATING AUDIO FOR VIRTUAL REALITY APPLICATION

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

Audio has always been an integral element in the creation more realistic audio-visual entertainment experiences. With the evolution of personal 3D audio and imaging technologies, entertainment experiences are possible with a higher degree of cognition, commonly referred to as virtual reality. The quest for more engaging user experiences has raised the challenge for more compelling audio. Elements of binaural hearing and sound capture have come to play a central role in existing and evolving production techniques. Of particular importance is the value of images related to audio content as a means of improving realism and minimizing binaural recording and reproduction artifacts. This tutorial will cover the elements of binaural audio as they relate to producing compelling entertainment and educational content for virtual reality applications. Specific areas to be covered with support audio and 3D anaglyph video demonstrations include: audio for games, music entertainment, radio drama, and music education. Audio production tools including binaural and higher order ambisonic capture microphone systems, with and without motion capture will be presented and demonstrated.

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

Tutorial/Workshop 9 — Saturday, October 21
9:00 am – 10:30 am — Room 1E07

PATHS TO BEING/BECOMING A PRO

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

Panelists: *Peter Auslan*, Manhattan Center Studios, New York, NY, USA
Lisa Ferrante-Walsh, iZotope
John Krivit, Audio Engineering Society, Marblehead, MA, USA; Emerson College, Boston, MA, USA
Ann Mincieli, Jungle City Studios, New York, NY, USA
Mike Wells, Mike Wells Mastering, Los Angeles, CA, USA

The Audio Professional focused on music production is looking more and more like a unicorn, particularly with regard to specialization. The panel will describe paths taken to becoming professionals who derive their income from music production, the skills they feel are most important that allowed them to succeed, and their current approaches to maintaining their status. Designed for students and the young engineers.

Special Events
SE11: LATIN PRODUCERS PANEL:
HOW LATIN MUSIC TURNED INTO MAINSTREAM
Saturday, October 21, 9:00 am – 11:00 am
Room 1E15/16

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

Panelists: *Carlos Beguerie*, Studio Instrument Rentals, NYC
Mariano Bilinkis, Silent Sounds
Héctor Castillo, @perroconsed
Ariel Lavigna, Nómade Estudios
Juan Cristobal Losada, NYU Steinhardt School of Culture, NYC
Ezequiel Morfi, Titanio Es Arte
Daniel Sanint, Flux Studios, New York, NY, USA

This panel will discuss how Latin music evolved from an isolated cultural expression into a world-wide mainstream trend

Student Events/Career Development
EC13: IRONS IN THE FIRE: CAREER DEVELOPMENT AND BUSINESS MENTORING FOR INDEPENDENT MUSIC CREATIVES
Saturday, October 21, 9:00 am – 10:30 am
Room 1E10

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

Panelists: *Harold Chambers*, Principal Recording Engineer, Pittsburgh Symphony
Andy Schwartz, Manhattan Producers Alliance, New York, NY, USA; Local 802, AFM
Carl Tatz, Carl Tatz Design, Nashville, TN, USA
Richard Warp, Intonic, Emeryville, CA, USA

Bring your energy, enthusiasm, business ideas, and questions. At this event the focus is on YOU! Members of the Manhattan Producers Alliance from New York, Nashville, San Francisco, and Los Angeles will offer fresh insights into career development, networking and entrepreneurship in an ever-changing industry. We will begin with a brief brief talk about developing your brand, your business, and functioning as a creative talent. After the formal session stay for a consultation with one our ManhatPro member mentors. The second half of this event will include breakout sessions where you will discuss your personal career goals one on one and get a chance to meet some ManhatPro members. Manhattan Producers Alliance is a New York/San Francisco-based membership organization comprised of engineers, composers, and producers. Our focus is on nurturing personal creativity within the art and craft of music making. Our relationship with AES includes a long list of seminars and special events at conferences dating back to our inception in 2005.

Spatial Audio 12 **Saturday, October 21**
10:15 am – 12:15 pm **Room 1E13**

BINAURAL LISTENING EXPERIENCE

Moderator: **Marta Olko**, New York University, New York, NY, USA

This is a listening session that will feature a selection of binaural recordings, mixes, and listening experiences from various artists, composers, recording, and mixing engineers.

Archiving/Restoration 10 **Saturday, October 21**
10:45 am – 12:15 pm **Room 1E10**

TALES OF ASSET MANAGEMENT: THE GOOD, THE BAD, AND "V2_FINAL_MIX_MASTER_VOX UP_FINAL.WAV.AIFF"

Moderator: **Margaret Luthar**, Sonovo Mastering, Hellertown, PA, USA

Panelists: *Adam Gonsalves*, Telegraph Mastering
Paul Jessop, Founder, County Analytics
Brian Losch, Freelance recording/mixing/post engineer
Mark Yeend, Creative Director, ATG at Xbox, Microsoft

Whether you're a studio engineer or a location sound recordist —at some point, the work you do will be handed off to someone else. Not only is it important to be diligent in organization for one's own personal project management, but it is important when dealing with clients, labels, production houses, and last but not least, other audio engineers! In this event, we discuss the needs of individual branches of the audio world (film sound, game audio, interactions between the tracking engineer and mixing engineer, and mastering needs) as well as archival standards put forth by AES/EBU/NARAS. We will also discuss maintaining good written records while working, project workflow, and what is actually crucial to doing good work from start to finish. Whether you're a novice or a seasoned engineer, this workshop will help you learn good organizational routines—or break some bad habits (we all have them)!

Archiving/Restoration 11 **Saturday, Oct. 21**
10:45 am – 12:15 pm **Room 1E08**

30TH ST. STUDIO

Presenter: **Dan Mortensen**, Dansound Inc., Seattle, WA, USA

Chronological history of the studio from its construction as a church through its eras as a studio used to create many phenomenal sounding recordings in many musical genres by the country and world's leading musical, social, artistic, and theatrical artists. It was converted from a church in 1948 and demolished in 1982. The presenter has been researching the studio for the last 9 years and is in the midst of a long-term project to recreate the life of the studio with as much detail as can be found. CBS's chief recording engineer Fred Plaut was also a semi-professional photographer, and the presenter has scoured Fred's photographic archive of over 30,000 pictures to find those relevant to the studio and its occupants; many pictures from that collection will be in the presentation.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Product Development 11 **Saturday, October 21**
10:45 am – 12:15 pm **Room 1E14**

3D FIR-BASED BEAM SHAPING—REVOLUTION OR EVOLUTION

Presenters: **Stefan Feistal**
Scott Leslie

In this session we discuss backgrounds and trends of FIR-based coverage optimization. After the era of conventional beam steering, for modern line arrays it is nowadays almost a standard to tailor radiation patterns precisely to the venue. This technology uses simulated FIR filters in order to optimize SPL uniformity, create avoidance zones, and improve speech intelligibility. However, the next level of sound field control is already looming: Horizontal arrays as well as modular planar systems with steering capabilities are being designed and tested. The talk outlines important aspects of horizontal and 3D coverage control using FIR filter optimization and demonstrates their capabilities using case studies. We compare these new concepts with the known approaches of classical beam steering and 2D vertical beam shaping. Finally we discuss off-the-shelf availability, hardware and software requirements, as well as computational effort and needed measurement accuracy.

Recording & Production 11 **Saturday, October 21**
10:45 am –12:15 pm **Room 1E07**

192DB DR, 32BIT/DXD ULTRA HIGH RESOLUTION CAPTURE, PROCESSING AND MASTERING WORKFLOW FOR MQA, DSD AND FUTURE HIGH RESOLUTION ARCHIVING AND DELIVERY

Presenter: **Michal Jurewicz**, Mytek, New York, NY, USA

This presentation will include a complete back to back working example utilizing Mytek Brooklyn ADC and DAC mastering converters for the modern 21st century mastering approach: 32bit fixed/192dB/352.8k DXD capture and playback and 64bit double precision DAW signal processing. World highest dynamic range with flat noise floor to 200kHz and precision processing allow for new opportunities to create masters for all new high resolution formats including MQA and DSD256 as well as incorporation of new restoration techniques such as Plangent tape process unachievable with regular 24/192k conversion. George Massenburg, Bob Ludwig, Alan Silverman and few other mastering engineers are currently testing system suitability for MQA(R) mastering.

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

Recording & Production 13 **Saturday, October 21**
10:45 am – 12:15 pm **Room 1E09**

MASTERING 201: BEYOND THE BASICS

Moderator: **Alan Silverman**, Owner/Mastering Engineer, Arf! Mastering, NYC, NY USA

Panelists: *Adam Ayan*
Heba Kadry
Darcy Proper
Sarah Register

In perfect world mastering would amount to a flat transfer. As recorded music shifts to production done in project studios and personal spaces, mixes are often sent off to mastering with any number of difficult issues that were not apparent until heard on full-range, high-resolution monitors. A revised mix is the ideal solution, but revisions are not always possible. This tutorial looks at recurring mix issues and explores mastering approaches made possible by recent advancements in digital audio processing tools. The panel will also explore creative possibilities made available to us via the new tool sets. An issue faced by independent mastering engineers working solo is the limited opportunity for exchanging new ideas and technical solutions. This tutorial is an ideal forum for attendees to discover fresh approaches through demonstrations by a panel of seasoned professional mastering engineers.

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

Spatial Audio 13 **Saturday, October 21**
10:45 am – 11:45 am **Room 1E06**

3D AMBEO LIVE AND STUDIO RECORDINGS

Presenters: **Gregor Zielinsky**, Sennheiser Electronic GmbH & Co. KG, Germanyl
Jim Andrson, Anderson Audio New York, New York, NY, USA
Ulrike Schwarz, Anderson Audio New York, New York, NY, USA

Live recording experts Jim Anderson and Ulrike Schwarz of Anderson Audio New York captured this year's Chelsea Music Festival for delivery via Sennheiser's immersive Ambeo 3D audio format, and will present this at the session.

Host, Gregor Zielinsky of Sennheiser, will present further examples of live and studio Ambeo productions. Also, the MLH 800 Twin plug in, will be explained and presented in a live session. This free Plug In makes the work with the double capsule/two way output MKH 800 Twin much easier and more flexible.

Special Events
SE12: LUNCHTIME KEYNOTE: EMILY LAZAR— MORE COWBELL? TRUSTING YOUR SONIC "GUT"
Saturday, October 21, 12:15 pm – 1:15 pm
Room 1E15/16

Presenter: **Emily Lazar**, The Lodge, New York, NY, USA

A discussion about the start to finish process of making recorded music today. How does one prevent getting lost in the lush landscape of infinite digital possibilities and potentially endless sonic decisions? Is staying focused on enhancing the vibe while upholding the overall sonic integrity an audio paradox? Do you trust your sonic "gut"?

Spatial Audio 15 **Saturday, October 21**
1:15 pm – 2:45 pm **Room 1E06**

AFTERNOON LISTENING SESSION IN 9.1

Moderators: **David Bowles**
Paul Geluso

Presenter: *Tom Ammerman*

Please join us for a 90 minute immersive listening journey on Saturday afternoon. This session will be dedicated to experiencing recent recorded works created specifically for multi-channel loudspeaker listening environments. The program will include classical, pop, electronic, jazz, and world music recordings created by a variety of engineers and producers who are dedicated to the art of spatial audio.

Session EB6 Saturday, Oct. 21
1:30 pm – 3:15 pm Room 1E12

SPATIAL AUDIO

Chair: **Mattieu Parmentier**, francetélévisions, Paris, France

1:30 pm

EB6-1 How Streaming Object Based Audio Might Work—*Adrian Wisbey*, BBC Media Services, London, UK

Object based media is being considered as the future platform model by a number of broadcasting and production organizations. This paper is a personal imagining of how object based broadcasting might be implemented with IP media as the primary distribution whilst still supporting traditional distributions such as FM, DAB and DVB. The examples assume a broadcaster supporting a number of linearly scheduled services providing both live (simulcast) and on-demand (catch-up) content. An understanding of the basics of object based audio production and broadcasting by the reader is assumed. Whilst this paper specifically discusses audio or radio broadcasting many of the components and requirements are equally valid in a video environment.
Engineering Brief 398

1:45 pm

EB6-2 DIY Measurement of Your Personal HRTF at Home: Low-Cost, Fast and Validated—*Jonas Reijniers, Bart Partoens, Herbert Peremans*, University of Antwerp, Gent, Belgium

The breakthrough of 3D audio has been hampered by the lack of personalized head-related transfer functions (HRTF) required to create realistic 3D audio environments using headphones. In this paper we present a new method for the user to personalize his/her HRTF, similar to the measurement in an anechoic room, yet it is low-cost and can be carried out at home. We compare the resulting HRTFs with those measured in an anechoic room. Subjecting the participants to a virtual localization experiment, we show that they perform significantly better when using their personalized HRTF, compared to a generic HRTF. We believe this method has the potential of opening the way for large scale commercial use of 3D audio through headphones.
Engineering Brief 399

2:00 pm

EB6-3 Audio Localization Method for VR Application—*Joo Won Park*, Columbia University, New York, NY, USA

Audio localization is a crucial component in the Virtual Reality (VR) projects as it contributes to a more realistic VR experience to the users. In this paper a method to implement localized audio that is synced with user's head movement is discussed. The goal is to process an audio signal real-time to represent three-dimensional soundscape. This paper introduces a mathematical concept, acoustic models, and audio processing that can be applied for general VR audio development. It also provides a detailed overview of an Oculus Rift- MAX/MSP demo.
Engineering Brief 400

2:15 pm

EB6-4 Sound Fields Forever: Mapping Sound Fields via Position-Aware Smartphones—*Scott Hawley, Sebastian Alegre, Brynn Yonker*, Belmont University, Nashville, TN, USA

Google Project Tango is a suite of built-in sensors and libraries intended for Augmented Reality applications allowing certain mobile devices to track their motion and orientation in three dimensions without the need for any additional hardware. Our new Android app, "Sound Fields Forever," combines locations with sound intensity data in multiple frequency bands taken from a co-moving external microphone plugged into the phone's analog jack. These data are sent wirelessly to a visualization server running in a web browser. This system is intended for roles in education, live sound reinforcement, and architectural acoustics. The relatively low cost of our approach compared to more sophisticated 3D acoustical mapping systems could make it an accessible option for such applications.
Engineering Brief 401

2:30 pm

EB6-5 Real-time Detection of MEMS Microphone Array Failure Modes for Embedded Microprocessors—*Andrew Stanford-Jason*, XMOS Ltd. ,Bristol, UK

In this paper we describe an online system for real-time detection of common failure modes of arrays of MEMS microphones. We describe a system with a specific focus on reduced computational complexity for application in embedded microprocessors. The system detects deviations is long-term spectral content and microphone covariance to identify failures while being robust to the false negatives inherent in a passively driven online system. Data collected from real compromised microphones show that we can achieve high rates of failure detection.
Engineering Brief 402

2:45 pm

EB6-6 A Toolkit for Customizing the ambiX Ambisonics-to-Binaural Renderer—*Joseph G. Tylka, Edgar Choueiri*, Princeton University, Princeton, NJ, USA

An open-source collection of MATLAB functions, referred to as the SOFA/ambiX binaural rendering (SABRE) toolkit, is presented for generating custom ambisonics-to-binaural decoders for the ambiX binaural plug-in. Databases of head-related transfer functions (HRTFs) are becoming widely available in the recently-standardized "SOFA format" (spatially-oriented format for acoustics), but there is currently no (easy) way to use custom HRTFs with the ambiX binaural plug-in. This toolkit enables the user to generate custom binaural rendering configurations for the plug-in from any SOFA-formatted HRTFs or to add HRTFs to an existing ambisonics decoder. Also implemented in the toolkit are several methods of HRTF interpolation and equalization. The mathematical conventions, ambisonics theory, and signal processing implemented in the toolkit are described.
Engineering Brief 403

Engineering Brief 404 was not presented

Archiving/Restoration 12 Saturday, October 21
1:30 pm – 3:00 pm Room 1E14

IS THERE LIFE ON MARS? PRESERVATION TECHNIQUES FOR THE 21ST CENTURY

Moderator: **Rebecca Yuri Feynberg**, NYU

Chair: **Steve Rosenthal**, MARS (MagicShop Archive and Restoration Studios), Brooklyn, NY, USA

Panelists: *Wally De Backer (Gotye)*
Michael Graves
Nora Guthrie
Stephen Maucci
Alex Slohm
Regan Sommer McCoy

Owner of the MagicShop and MARS Steve Rosenthal and his colleagues will present the work they have been doing with remastering and archiving significant, historical recordings from Lou Reed to Woody Guthrie. The presenters will describe their history and path as they moved forward from the MagicShop to the MagicShop Archive and Restoration Studios (MARS).

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Game Audio & VR 15 **Saturday, October 21**
1:30 pm – 2:30 pm **Room 1E13**

CREATE SOUND AND MUSIC FOR VR AND 360 USING COMMON DAW WORKFLOWS

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

This session focuses on the latest initiatives and developments of Augmented and Virtual Reality research and development centers, including the first publicly funded VR/AR Lab in the country – in New York City. Panelists will discuss, share, and give specific examples of some of the projects and directions currently explored in AR/VR.

Networked Audio 10 **Saturday, October 21**
1:30 pm – 3:00 pm **Room 1E09**

HOW TO MAKE AN AES70 CONTROLLER

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

Once you have a feeling for the basics of AES70 classes, commands, and events, making a basic AES70 controller is straightforward. This session will present two approaches to designing AES70 controllers and discuss realities of implementing them. Examples will be given.

Recording & Production 14 **Saturday, October 21**
1:30 pm – 3:00 pm **Room 1E07**

EVOLUTION OF ALBUM PRODUCTION FROM START TO FINISH

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

Panelists: *Ann Mincieli* (Alicia Keys)
Kim Rosen (Bonnie Raitt, Aimee Mann)
Angie Teo (Madonna)

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

Tutorial/Workshop 10 **Saturday, October 21**
1:30 pm – 3:00 pm **Room 1E10**

TOWARDS THE NEW HORIZON OF TECHNICAL EAR TRAINING

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

Panelists: *Jason Corey*, University of Michigan, Ann Arbor, MI, USA
Kazuhiko Kawahara, Kyushu University, Fukuoka, Japan
Doyuen Ko, Belmont University, Nashville, TN, USA
Sean Olive, Harman International, Northridge, CA, USA
Timothy Ryan, Webster University

Recently, various technical ear-training programs have been introduced to audio and acoustic engineering communities. In the previous workshops, the panels have discussed necessary features and methods for efficient and effective training (AES131, 132, and 141). The current workshop aims to (1) let workshop attendees experience and compare the characteristic functions of various ear-training programs through hands-on demonstrations by the panelists, and (2) discuss the latest development trends and future applications. While the workshop locally aims to provide the attendees with chance to experience theoretical and empirical matters of ear training programs around the world, it also globally aims to consider the importance of "listening" in the current video-oriented society.

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

Special Events
SE13: AESX TALKS
Saturday, October 21, 1:30 pm – 4:45 pm
Room 1E15/16

1:30 pm – 2:00 pm

Close Miking Strings—Navigating Bridges, Fingerboards, Sound Holes and Sound Boards—*Alex Case*, Sound Recording Technology, University of Massachusetts Lowell

Engineers must be prepared to record a wide range of string instruments, from guitar to ukulele. Fortunately, musical acoustics dictates that they have much in common, so we can form general strategies to help us the next time we track a piano, or the first time we encounter a harp.

2:00 pm – 2:30 pm

Recording Orchestra—*Richard King*, McGill University, Montreal

With a focus on the orchestra as an instrument and sound source, this tutorial will cover learning how to listen, understanding microphones, concert halls and orchestral seating arrangements. How to set up the monitoring environment, and how to approach each section of the orchestra as part of the recording process.

Sat. 2:30pm – 3:00pm

Urban Sound Capture using MEMS Microphones—*Charlie Mydlarz*, New York University, New York, NY, USA

Sensing urban sound using MEMS microphones is a new and promising venture. Recent advances in these technologies has made MEMs microphones a viable solution for urban sound capture and analysis. The pros and cons of this approach will be discussed, with some prototype applications to see and hear.

3:00 pm – 3:30 pm

Polarity vs. Phase: What Should We Call That Button and What Does It Do?— *Eric Ferguson*, New England School of Communications, Hussan University, Bangor, ME USA

Professionals and amateurs alike refer to polarity reversal as "flipping the phase". Are they wrong? While a semantical debate may be pedantic, it highlights a misunderstanding of phase by many in the audio industry. This short presentation will investigate the polarity button and its impact on phase and sound quality.

3:30 pm – 4:00 pm

Music Metadata and Credits: A Defining Moment—*Gebre Waddell, Connor Reviere*, Soundways, Memphis, TN USA

This talk covers the revolution happening in music metadata and how we have reached this defining moment. The industry has shifted beyond dialog, and moved forward to action and application. Discover how this shift occurred and the steps that can be taken to help producers, engineers, and musicians to receive credit where credit is due.

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

Student Events/Career Development
EC14: STUDENT DELEGATE ASSEMBLY MEETING—PART 2
Saturday, October 21, 1:30 pm – 3:00 pm
Room 1E08

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 Europe and International Regions. Judges' comments and awards will be presented for the Recording Competitions and Design Competitions. Plans for future student activities at local, regional, and international levels will be summarized.

Session P18 **Saturday, Oct. 21**
2:00 pm – 4:30 pm **Room 1E11**

SOUND REINFORCEMENT AND ACOUSTICS

Chair: **Joerg Panzer**, R&D Team, Salgen, Germany

2:00 pm

P18-1 Dynamic Diffuse Signal Processing for Low-Frequency Spatial Variance Minimization across Wide Audience Areas—*Jonathan Moore, Adam J. Hill*, University of Derby, Derby, Derbyshire, UK

Diffuse signal processing (DiSP) is a method of decorrelating coherent audio signals that is applicable to various components of sound reinforcement systems. Previous tests have indicated that DiSP can successfully decorrelate multiple low-frequency sources, leading to the reduction of comb filtering effects. However, results also show that performance is variable with source material and that effectiveness is reduced in closed acoustic spaces. In this work a dynamic variant of DiSP is examined where the decorrelation algorithm varies over time. The effectiveness of the processing is analyzed and compared to static DiSP and unprocessed systems. Results show that dynamic DiSP provides superior low-frequency spatial variance reduction to static DiSP due to improved decorrelation between direct sounds and early reflections.
Convention Paper 9903

2:30 pm

P18-2 A Novel Procedure for Direct-Method Measurement of the Full-Matrix Speech Transmission Index—*Jan A. Verhave, Sander van Wijngaarden*, Embedded acoustics BV, Delft, The Netherlands

When measuring the Speech Transmission Index (STI), until now one had to choose between two alternatives: impulse-response based full STI measurements (indirect method), or measurements based on modulated STIPA signals (direct method). Limitations apply when using either method. A novel procedure is proposed to measure the full STI through the direct method. The procedure combines advantages of indirect full STI measurements and direct STIPA measurements, completing a full STI measurement in 65.52 seconds. Similar to STIPA, the test signal is simultaneously modulated with 2 modulation frequencies per octave band. However, a rotation scheme is applied that uses a different set of modulation frequencies during different stages of the measurement, ending up with a full matrix (7 octaves x 14 modulation frequencies).
Convention Paper 9904

3:00 pm

P18-3 Blind Estimation of the Reverberation Fingerprint of Unknown Acoustic Environments—*Prateek Murgai,*[1] *Mark Rau,*[1] *Jean-Marc Jot*[2]
[1]Center for Computer Research in Music and Acoustics (CCRMA), Stanford University, Stanford, CA, USA;
[2]Magic Leap, Sunnyvale, CA, USA

Methods for blind estimation of a room's reverberation properties have been proposed for applications including speech dereverberation and audio forensics. In this paper we study and evaluate algorithms for online estimation of a room's "reverberation fingerprint," defined by its volume and its frequency-dependent diffuse reverberation decay time. Both quantities are derived adaptively by analyzing a single-microphone reverberant signal recording, without access to acoustic source reference signals. The accuracy and convergence of the proposed techniques is evaluated experimentally against the ground truth obtained from geometric and impulse response measurements. The motivations of the present study include the development of improved headphone 3D audio rendering techniques for mobile computing devices.
Convention Paper 9905

3:30 pm

P18-4 Microphone Selection Based on Direct to Reverberant Ratio Estimation—*Alexis Favrot, Christof Faller*, Illusonic GmbH, Uster, Zürich, Switzerland

Microphone recording in a room is ideally carried out by using a close distance microphone to prevent reverberation and noise annoyances, but this restricts the flexibility and the surface covered by the recording. When using multiple distant microphones, a microphone selection algorithm is needed for selecting the momentarily best microphone, namely the one with the least reverberation. Given several microphones arbitrarily distributed in a room, this paper describes an algorithm which, based on an estimation of the direct-to-reverberation ratio for each microphone, switches to the best microphone. The algorithm allows prioritizing a microphone and compensation of different directivity patterns.
Convention Paper 9906

4:00 pm

P18-5 Experimental Investigation on Varied Degrees of Sound Field Diffuseness in Full Scale Rooms—*Alejandro*

Bidondo, Sergio Vazquez, Javier Vazquez, Universidad Nacional de Tres de Febrero, Buenos Aires, Argentina

Sound field diffusion in enclosures should be experimentally quantified based on measured room impulse responses, at least to know how many scattering surfaces produce a sufficiently diffuse sound field for each application. To achieve this a parameter, the Sound Field Diffusion Coefficient (SFDC) which is still under development, was applied. SFDC expresses the reflection's amplitude control and temporal distribution gaussianity, using third octave-band energy-decay compensated impulse responses and taking reference with SFDC average results from a set of impulse responses synthesized with Gaussian white noise. In an attempt to demonstrate the quantification capability of the SFDC, a systematic investigation was conducted whereby varied room configurations using carefully designed scattered interior surfaces were examined with the hypothesis that varied degrees of surface scattering will ultimately lead to varied degrees of sound field diffusion inside two, full scale, rooms. To this end, each room´s floor was covered with different configurations ranging from no diffusers to 16.74 m2 of diffusely reflecting surfaces, in 3 steps. This paper discusses the experimental design and evaluates the results of data collected using systematic modifications of varied degrees of surface scattering, each with combinations of different source orientations and microphone positions.
Convention Paper 9907

Session EB7 — Saturday, Oct. 21
2:00 pm – 3:30 pm — Poster Area

POSTERS—PART 3

2:00 pm

EB7-1 Cycle-Frequency Wavelet Analysis of Electro-Acoustic Systems—*Daniele Ponteggia*, Audiomatica Srl, Firenze (FI), Italy

A joint time-frequency analysis of the response of electro-acoustic systems has been long sought since the advent of PC based measurement systems. While there are several available tools to inspect the time-frequency response, when it comes to inspect resonant phenomena, there are always issues with time-frequency resolution. With a rather simple variable substitution in the Wavelet Analysis it is possible to switch one of the analysis axes from time to cycles. With this new cycle-frequency distribution it is then possible to analyze very easily resonances and decays. A PC based measurement tool capable of Cycle-Frequency analysis will be introduced.
Engineering Brief 388

2:00 pm

EB7-2 Sharper Spectrograms with Fast Local Sharpening — *Robin Lobel*, Divide Frame, Paris, France

Spectrograms have to make compromises between time and frequency resolution because of the limitations of the short-time Fourier transform (Gabor,1946). Wavelets have the same issue. As a result spectrograms often appear blurry, either in time, frequency, or both, A method called Reassignment was introduced in 1978 (Kodera et al.) to make spectrograms look sharper. Unfortunately it also adds visual noise, and its algorithm does not make it suitable for realtime scenarios. Fast Local Sharpening is a new method that attempts to overcome both these drawbacks.
Engineering Brief 389

2:00 pm

EB7-3 An Interactive and Intelligent Tool for Microphone Array Design—*Hyunkook Lee, Dale Johnson, Maksims Mironovs*, University of Huddersfield, Huddersfield, West Yorkshire, UK

This engineering brief will present a new microphone array design app named MARRS (microphone array recording and reproduction simulator). Developed based on a novel psychoacoustic time-level trade-off algorithm, MARRS provides an interactive, object-based workflow and graphical user interface for localization prediction and microphone array configuration. It allows the user to predict the perceived positions of multiple sound sources for a given microphone configuration. The tool can also automatically configure suitable microphone arrays for the user's desired spatial scene in reproduction. Furthermore, MARRS overcomes some of the limitations of existing microphone array simulation tools by taking into account microphone height and vertical orientations as well as the target loudspeaker base angle. The iOS and Android app versions of MARRS can be freely downloaded from the Apple App Store and the Resources section of the APL website: https://www.hud.ac.uk/apl , respectively.
Engineering Brief 390

2:00 pm

EB7-4 Real-Time Multichannel Interfacing for a Dynamic Flat-Panel Audio Display Using the MATLAB Audio Systems Toolbox—*Arvind Ramanathan, Michael Heilemann, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

Flat-panel audio displays use an array of force actuators to render sound sources on a display screen. The signal sent to each force actuator depends on the actuator position, the resonant properties of the panel, and the source position on the screen. A source may be translated to different spatial locations using the shifting theorem of the Fourier transform. A real-time implementation of this source positioning is presented using the MATLAB Audio Systems Toolbox. The implementation includes a graphical interface that allows a user to dynamically position the sound source on the screen. This implementation may be combined with audio source separation techniques to align audio sources with video images in real-time as part of a multimodal display.
Engineering Brief 391

2:00 pm

EB7-5 Perceived Differences in Timbre, Clarity, and Depth in Audio Files Treated with MQA Encoding vs. Their Unprocessed State—*Mariane Generale, Richard King*, McGill University, Montreal, QC, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

The purpose of this engineering brief is to detail a planned experiment in examining any perceived differences in timbre, clarity, and depth between WAV and Master Quality Authenticated (MQA) audio files. A study proposes examining the responses of engineers, musicians, and casual listeners on whether any changes to timbre, clarity, and depth are perceived between WAV and MQA. A blind listening test is considered in a controlled environment using both professional and consumer level loudspeakers and headphones. Additional interests include a comparison of responses between the target groups on different listening mediums.
Engineering Brief 392

2:00 pm

EB7-6 The BACH Experience: Bring a Concert Home—*Sattwik Basu, Saarish Kareer*, University of Rochester, Rochester, NY, USA

Inverse filtering of rooms to improve their frequency response or reverberation time is a well-researched topic in acoustical signal processing. With the aim of giving music lovers the experience of a concert hall in their own homes, we describe a system that employs signal processing techniques, including inverse filtering, to accurately reproduce concert hall acoustics in a home listening space. First, binaural impulse responses were measured at a few chosen seating positions in the concert hall. Next, the listening location along with its loudspeaker configuration is acoustically characterized and inverse filtered using MINT and Cross-talk Cancellation algorithms to produce a flat-frequency response. We observed that speech and music, after our inverse filtering method showed near-anechoic qualities which allowed us to subsequently impress the acoustical response of a wide range of concert halls upon the original audio. A demonstration will be provided using 4 loudspeakers for a quadraphonic sound reproduction at the listening area. In continuing work, to produce a sufficiently wide listening area, we are combining head tracking with adaptive inverse filtering to adjust to the listeners' movements.
Engineering Brief 393

2:00 pm

EB7-7 Early Reflection Remapping in Synthetic Room Impulse Responses—*Gregory Reardon*, New York University, New York, NY, USA

In audio-visual augmented and virtual reality applications, the audio delivered must be consistent with the physical or virtual environment, respectively, in which the viewer/listener is located. Artificial binaural reverberation processing can be used to match the listener's/viewer's environment acoustics. Typical real-time artificial binaural reverberators render the binaural room impulse response in three distinct section for computational efficiency. Rendering the response using different techniques means that within the response the early reflections and late reverberation may not give the same room-acoustic impression. This paper lays the theoretical foundation for early reflection remapping. This is accomplished by acoustically characterizing the virtual room implied by the early reflections renderer and then later removing that room-character from the response through frequency-domain reshaping.
Engineering Brief 394

2:00 pm

EB7-8 Acoustic Levitation—Standing Wave Demonstration—*Bartłomiej Chojnacki, Adam Pilch, Marcin Zastawnik, Aleksandra Majchrzak*, AGH University of Science and Technology, Krakow, Poland

Acoustic levitation is a spectacular phenomenon, perfect for standing waves demonstration. There are a few propositions for such a construction in scientific literature, however they are often expensive and difficult to build. The aim of this project was to create a functional stand—easy to construct, with no need for much expensive software or hardware. Piezoelectric transducers, typical for ultrasonic washing machines, were used as a sound source; their directivity pattern and frequency characteristics have been measured. The final result of the project was a stand-alone acoustic levitator with very little need for calibration, and with no walls, so the effect can be observed easily. The paper presents whole design process and describes all functionalities of the final stand.
Engineering Brief 395

2:00 pm

EB7-9 Developing a Reverb Plugin; Utilizing Faust Meets JUCE Framework—*Steve Philbert*, University of Rochester, Rochester, NY, USA

Plug-ins come in many different shapes, sizes and sounds, but what makes one different from another? The coding of audio and the development of the graphical User Interface (GUI) play a major part in how the plugin sounds and how it functions. This paper details methods of developing a reverb plugin by comparing different programming methods based around the Faust meets JUCE framework launched in February of 2017. The methods include: Faust direct to a plugin, Faust meets JUCE compiled with different architectures, and C++ with JUCE Framework. Each method has its benefits; some are easier to use while others provide a better basis for customization.
Engineering Brief 396

Broadcast/Streaming Media 13 Saturday, October 21
3:00 pm – 5:00 pm Room 1E13

WHAT'S THIS? DOCTOR WHO WITH SPATIAL AUDIO!

Presenter: **Chris Pike**, BBC Research and Development, Salford, Greater Manchester, UK; University of York, Heslington, York, UK

If there is one constant to the more than 50-year legacy of the mysterious Doctor, it is change. Now Whovians on both sides of the pond can immerse themselves in the next level of fright and terror. BBC lead engineer Chris Pike takes us into the magic that happens in post to transform a normally recorded episode into a magnitude enhanced experience. Chris will highlight the tools, the process, and how BBC maintains compatibility with existing workflows. Was that creak behind me? No, it's the floor below me! Follow us into the next realm with our favorite Time Lord.

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

Spatial Audio 13 Saturday, October 21
3:00 pm – 4:00 pm Room 1E06

CAPTURING HEIGHT: RECORDING TECHNIQUES THAT INCLUDE THE VERTICAL DIMENSION

Presenter: **David Bowles**, freelance classical music producer/engineer
Paul Geluso, New York University, New York, NY, USA
Sungyoung Kim, Rochester Institute of Technology, Rochester, NY, USA

Height sound contains crucial information about sound sources, the recording space, and ambient sounds. As object oriented and dedicated height channel playback system become more common, recording engineers need to be familiar with effective height sound capturing techniques to create true three-dimensional sound recordings. Height sound can be captured using specialty microphones and decoders, or by adding dedicated height microphones to an existing mono, stereo or surround microphone system. At this workshop, the panelists will give a comprehensive overview of current height sound recording techniques and share their personal experiences working with height sound. Recent music recordings made with some of the techniques discussed at the workshop will be played during a technical tour of the studio facilities at NYU Steinhardt.

Archiving/Restoration 13 — **Saturday, October 21**
3:15 pm – 4:45 pm — **Room 1E14**

NEW DEVELOPMENTS IN STUDIO METADATA

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

Responsibilities of studio experts now extend well beyond great sound into accurate data about the recordings they enable. This workshop will look at the evolution of standards, tools, expectations, and opportunities in the field and will bring together experts to discuss the whats, the ways, and the hows of collecting and providing studio metadata.

This session is presented in association with the AES Technical Committee on Archiving, Restoration and Digital Libraries

Recording & Production 15 — **Saturday, October 21**
3:15 pm – 4:45 pm — **Room 1E08**

RAW TRACKS: THE SOUND OF PHILADELPHIA (TSOP)

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

Panelists: *Dirk Devlin*, former engineer Sigma Sound Studio, Philadelphia, PA, USA
Toby Seay
Arthur Stoppe
Joe Tarsia

Philadelphia has been an incubator for all kinds of great music, and in the '70s and '80s funk, soul, pop and Latin styles combined to create some phenomenally great, danceable and far-reaching music. Mark Rubel will interview Philadelphia engineer Dirk Devlin (and possibly some other special guests) about the stories, techniques, and style of recording, plus an examination of an 11-minute multitrack recording.

Tutorial/Workshop 11 — **Saturday, October 21**
3:15 pm – 5:00 pm — **Room 1E10**

WHAT AUDIO STUDENTS ARE LEARNING, WHAT COULD THEY BE LEARNING, WHAT SHOULD THEY BE LEARNING

Co-Chairs: **Richard King**, McGill University, Montreal, Quebec, Canada; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
Enrique Gonzalez-Muller, Berklee College of Music, Boston, MA, USA

Panelists: *Owen Curtin*, Audio Builders Workshop, Lexington, MA, USA
Scott B. Metcalfe, Peabody Conservatory, Johns Hopkins, Severna Park, MD, USA

Through a survey the world's audio education programs, one would observe a great variance in approaches to subjects, course content, and expected outcomes.

The workshop panelists representing technical colleges, undergraduate, graduate, and online programs will describe their respective program structures, highlighting the strengths of each and where the greatest challenges lie. Discussion (and friendly debate) of what comprises the skill set that students need to acquire as part of their academic track, and how best to "package" learning modalities into a selection of courses that provide the required knowledge across programs of different lengths and intensities. Educators and students are invited to attend, and questions/comments will be welcomed from the audience as we open up the discussion to the room.

This workshop is intended to complement TW10 which seeks to identify desirable skill set and competencies for those entering the world of professional audio from the prospective of the working professional.

Tutorial/Workshop 12 — **Saturday, October 21**
3:15 pm – 4:45 pm — **Room 1E08**

PRACTICAL 3D ACOUSTIC MEASUREMENTS

Presenter: **Malcolm Dunn**, Marshall Day Acoustics, Auckland, New Zealand

This tutorial will illustrate how 3D impulse response measurements can be utilized in practical applications. An introduction will be provided to 3D acoustic measurement technology with a focus on b-format signal and compact microphone arrays. The types of signals gathered will be described and methods of "visualizing" the results will be presented. Limitations of the information gathered will also be discussed. Examples would be provided for a range of applications where the use of directional information has provided worthwhile insight into the acoustic performance of a space.

Tutorial/Workshop 13 — **Saturday, October 21**
3:15 pm – 4:45 pm — **Room 1E07**

INSIGHTS ON COLLABORATION IN COMMERCIAL MUSIC PRODUCTION

Chair: **Rob Toulson**, Music Producer and Educator

Panelists: *Steve Baughman*, Mix and Mastering Engineer
Trevor Gibson, Circle Studios, Birmingham, UK
Adam Gonsalves, Tracking, Mixing and Mastering Engineer
Mandy Parnell, Mastering Engineer

Collaboration takes many forms in contemporary music production. Building effective professional relationships can be the secret to success in modern music production, even in a world where autonomous working is more possible than ever. For example, we see engineers collaborate and co-produce with artists, electronic producers working remotely with session musicians, and self-producing artists nurturing their product alone through the recording mixing and mastering chain. In this workshop we explore the contemporary practices of collaboration in music production, particularly reflecting on technologies and tools that have enabled new frameworks for communication and co-working. We will look at methods of the past that have perhaps been lost, and evaluate the education needs to enable new artists and producers to be successful in their careers.