AES 149TH CONVENTION PROGRAM

OCTOBER 27–30, ONLINE

AES 149th offered 80-plus hours of scheduled streaming content along with a slate of over 100 on-demand Paper, Engineering Brief sessions, as well as Workshops and Tutorials. The following is a listing of the events presented. Events without a date or time are On-Demand and will be available until November 30.

Winner of the 149th AES Convention Best Paper Award

Short-Range Rendering of Virtual Sources for Multichannel Loudspeaker Setups—*Juhani Paasonen, Ville Pulkki*, Aalto University, Espoo, Finland

Convention Paper 10401

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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 149th 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 149th AES Convention Student Paper Award is:

Individual Listening Zone with Frequency-Dependent Trim of Measured Impulse Responses—*Michele Ebri,[1] Nicolò Strozzi,[2] Filippo Maria Fazi,[3] Angelo Farina,[1] Luca Cattani[2]*

[1] University of Parma, Parma, Italy

[2] ASK Industries Spa, Reggio Emilia, Italy

[3] University of Southampton, Southapton, UK

Convention Paper 10409

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SPECIAL EVENT
OPENING CEREMONY
Tuesday, October 27, 2020
10:30 am – 11:30 am

Opening Remarks:
- Executive Director Colleen Harper
- President Agnieszka Roginska

Convention Chairs
- Valerie Tyler
- Paul Womack
- Jonathan Wyner

Welcome to AES's Fall convention. Leadership and program chairs will get you up to speed about what you can experience at this year's show!

The Audio Engineering Society has announced the winners of this year's AES Awards for distinguished achievement in the field of audio and service to the Society, presented during the Opening Ceremonies of the AES Show Fall 2020 Convention on October 27. The awards were presented in the online ceremony by AES President Agnieszka Roginska on behalf of Awards Committee Chair David Scheirman. The AES Show Opening Ceremonies are available on-demand for AES Show attendees at AESShow.com.

Awards presented at the AES Show Fall 2020 Convention i nclude:

AES Board of Governors Award, given to Members rendering exceptional service to the Audio Engineering Society, presented to:

• **Jorge Azama**, for chairing the highly successful AES Latin American Conference in September 2019 in Lima, Perú

AES Honorary Membership Award, given to individuals of outstanding repute and eminence in the science of audio engineering or its allied arts, presented to:

• **Imogen Heap**, for outstanding contributions to the world of music and technology

• **Karrie Keyes**, for her leadership in creating opportunities for girls and women in the field of live sound

AES Fellowship Award, given to Members who have rendered conspicuous service or are recognized to have made a valuable contribution to the advancement in or dissemination of knowledge of audio engineering, or in the promotion of its application in practice, presented to:

• **Peter Graham Craven**, for decades of commitment to the growth and ongoing health of the professional audio industry and for unflagging advocacy for, and dedication to, the Audio Engineering Society

• **Jackie Green**, for contributions to the sciences of microphone design and manufacture, microphone performance measurement and to microphone and wireless mic system education

• **Thomas Miller,** for valuable contributions to the development of miniature transducer design

• **Umberto Zanghieri**, for his work advancing the application of audio design and digital interconnectivity and for extraordinary service to the Audio Engineering Society

Distinguished Service Award, awarded in recognition of extraordinary service to the Society over a period of years, presented to:

• **Don Puluse**, for his leadership of the AES Educational Foundation as its longtime President, securing and awarding opportunity to deserving AES Student Members

ACOUSTICS AND PSYCHOACOUSTICS

• Cross-Talk Cancellation Systems: Past, Present, and Future

Presenters: **Filippo M. Fazi**, University of Southampton
Eric Hamdan
Jacob Hollebon, ISVR, University of Southampton

When binaural audio is mentioned we often think about headphone reproduction. The reproduction of binaural audio through two or more loudspeakers is, however, an attractive and widely-accepted alternative. Such systems are referred to as Cross-Talk Cancellation Systems (CTCS). In this tutorial we will cover the theoretical and practical fundamentals of CTCS, discussing their advantages and limitations. We will do that with the support of real-time practical demonstrations. We will also introduce some more advanced topics, like multichannel CTCS (i.e., loudspeaker arrays), multi-listener CTCS (i.e., for cinemas), and position-adaptive CTCS (with listener tracking).

• Digital-Electro-Mechano-Acoustical Transducers

Presenters: **Gregor Hohne,** Klippel GmbH, Dresden, Germany
Wolfgang Klippel, Klippel GmbH, Dresden, Germany

Developments like ongoing miniaturization and increased availability of DSP Processing power are allowing software algorithms to take over more and more tasks in loudspeaker systems that were traditionally solved by classic transducer and amplifier design. This can include compensation for significant distortion and undesired effects due to ageing and production variance as well as avoiding mechanical and thermal overload and amplifier clipping. With measured transducer and amplifier state variables and advanced modeling, the software achieves a sufficiently linear, time-invariant and thus predictable transfer function. This effectively merges the amplifier, software, and transducer together into a single module, needing only a digital input signal and a power supply for operation and generating acoustical output. The tutorial gives an overview of the principles of such systems like adaptive nonlinear control and overload protection but focuses on the practical impacts for transducer and amplifier design. Another focus is the consequence on applications like the design of small portable speakers and on the interaction with other techniques like acoustic echo cancellation and beam forming.

• Listening Test Design A to Z

Presenters: **Hyunkook Lee**, University of Huddersfield, Huddersfield, UK
Dale Johnson, University of Huddersfield, Huddersfield, UK

This workshop will first provide a comprehensive overview of the theoretical principles of various types of listening test methods with practical examples in audio engineering context. It will cover classic psychometric methods used for auditory threshold measurement (e.g., mAFC, staircase procedure, ABX, etc.) as well as audio quality rating methods from ITU-R and ITU-T. Important factors to consider for a successful listening test design will be pointed out. Some common misconceptions observed in subjective studies will also be discussed. Furthermore, an open-access universal listening test design framework called HULTI-GEN version 2 will be introduced.

• Listening Tests—Understanding the Basic Concepts

Presenter: **Jan Berg**, Luleå University of Technology, Luleå, Sweden

Listening tests are important tools for audio professionals as they assist our understanding of audio quality. There are numerous examples of tests, either formally recommended and widely used or specially devised for a single occasion. In order to understand listening tests and related methods, and also to potentially design and fully benefit from their results, some basic knowledge is required. This tutorial aims to address audio professionals without prior knowledge of listening test design and evaluation. The fundamentals of what to ask for, how to do it, whom to engage as listeners, what sort of results that may be expected, and similar issues will be covered.

• Sparse Sampling and Measurement: Black Magic or Beautiful Science?

Presenter: **Jamie Angus-Whiteoak**, University of Salford, Salford, UK

Many measurement tasks such as speaker polar pattern measurement, or PA coverage, require a large number of measurements to achieve accuracy due to the need to sample the measurement above the Nyquist/Shannon limit of two times the highest frequency. Sparse and Compressive Sampling allows one to sample the signal at apparently less than the Nyquist/Shannon limit of two times the highest frequency, this without losing any signal fidelity or even lower with some loss of accuracy! How can this be: is it black magic, or is it a beautiful science based method for achieving high accuracy with a few limited measurements? The purpose of this presentation is to give a non-mathematical introduction to Sparse/Compressive Sampling and measurement. We will examine the difference between “Sparse” and “Dense” signals. We will define what

is meant by "rate of innovation" and see how it relates to sample rate. We will then go on to see how we can create sparse signals either via transforms or filters to provide signals that can be sampled at much lower rates. We will then show how some of these methods are already used in audio, and suggest other areas of application, in particular audio and acoustic measurements. Finally we will finish off by showing how a commonly used audio system can be considered to be a form of compressive sensing.

• Time and Perception
Tuesday, October 27, 2020
Session Time: 11:00 am – 12:00 noon

Presenters: **Marina Bosi**, Stanford university, CA, USA, **Karlheinz Brandenburg**, TU Ilmenau, Ilmenau, Germany
Thomas Lund, Genelec Oy, Denmark

We summarize recent research on perception and active sensing from a pro audio perspective. Topics include overt and hidden reach-out features of listening and how time may influence what we hear on a number of different scales. A new understanding of human perception would not only result in a revision of textbooks, and influence some types of subjective testing. Between-listener variation might also be investigated more systematically, and possibly help guide improvements in recording, distribution and reproduction. The discussion will include a review of distribution based on perceptual coders, their usefulness and their limits.

APPLICATIONS IN AUDIO

• What's That Sound? An Introduction to the Field of Audio Forensic Analysis

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

Presenters: *Eddy B. Brixen*, EBB-consult / DPA Microphones / Danish National School of Performing Arts, Denmark
J. Keith McElveen, Wave Sciences
Gordon Reid, CEDAR Audio Ltd.
Jeff M. Smith, The MITRE Corp

This is a tutorial for those interested in learning about audio forensic analysis. Audio forensic analysis, a recognized field of forensic science, deals with the acquisition, evaluation, and interpretation of audio evidence that is likely to end up in a court of law. Audio evidence may come from police body cameras, smartphones, surveillance systems, emergency call centers, cockpit voice recorders, and many other devices. Audio examiners assess the authenticity of forensic evidence, enhance the quality and intelligibility of forensic recordings, and scientifically interpret the meaning and significance of sounds present in the recordings. Forensic examiners also inform courts and the public about what can and cannot be determined scientifically. The tutorial includes examples and contact information for follow-up questions.

ARCHIVING AND RESTORATION

• Software Solutions for Hardware Problems
Tuesday, October 27, 2020
Session Time: 3:30 pm – 4:30 pm

Presenter: **Alexey Lukin**
Jason Davies, Zynaptiq GmbH
Anna Frick

Find out about the latest advancements in software solutions for hardware problems, including development of noise reduction plugins, azimuth adjustment, and more.

• Archiving the 90s: Preservation of Early Digital Formats
Wednesday, October 28, 2020
Session Time: 3:00 pm – 4:30 pm

Presenters: **Kaylie Ackerman,** Harvard Library
Jason Bitner, Traffic Entertainment Group
Eddie Ciletti, Manhattan Sound Technicians Inc.
Kelly Pribble, Iron Mountain Entertainment Services
Pete Weiss, Verdant Studio

An encore presentation about the joys and technicalities of dealing with recording formats of the 90s: ADAT, DA-88 and DA-89, DTRS, 1630. Loads of great records were made on these formats, frequently in project studios with smaller budgets. Sadly, they are some of the most at-risk formats, both because the carriers are awful and because playback machines in working order are hard to find and maintain. Panelists will talk about playback and preservation of these formats, specific considerations in capturing audio, timecode and other data, sourcing and maintaining playback machines, and curating releases from this content.

• Archival Activism: The Intersection of Audio Archives and Social Justice
Thursday, October 29, 2020
Session Time: 3:30 pm – 4:30 pm

Presenters: **Yuri Shimoda**, UCLA Ethnomusicology Archive, Los Angeles, CA, USA
Regan Sommer McCoy, The Mixtape Museum/ Columbia University, New York City, NY, USA
Traci Mark, Metropolitan New York Library Council, New York City, NY, USA

Since the 1970s, the notion of passive, completely neutral archives has been questioned and challenged by activist archivists, those who wish to use their power in determining what materials are collected, preserved, and made accessible to future generations to address social issues and become involved with and create ways of supporting the communities that they document and serve. This panel examines what it means to be an activist audio archivist in the 21st century. Topics to be explored include community vs. institutional archives and custodial archiving; ethics and advocacy; collaborations with community partners and communities of origin; and the documentation and preservation of current events.

AUDIO BUILDERS WORKSHOP

• DIY Basics Series—Tools, Tips, and Builds For Your Home Studio

Presenters: **Jason Bitner**, Traffic Entertainment Group
Dereck Blackburn, Quiethouse Recording
Chris Kincaid, Indiana University Purdue University Indianapolis, IN, USA
Brewster LaMacchia, Clockworks Signal Processing

Topics aimed at homebound engineers interested in growing their skill sets. These videos will highlight useful aspects of the Do It Yourself approach; including tools, tips, and builds for the beginner and pro alike. Each "Basics" video will be presented by a veteran DIYer in the community and will present content from both the host and suggestions from our DIY community at AES. • Benchtop Basics: Tools and strategies from the community relating to

your workspace. • Breadboard Basics: Breadboard circuit builds for simple and interesting audio projects. • Home Studio Basics: Easy and low cost DIY solutions to working in your home studio. • DIY Resource Basics: Communities, DIY databases, and some of ABW's favorite places to get started.

• DIY Skills In The Job Market: How DIY Prepares You for Roles In Audio
Thursday, October 29, 2020
Session Time: 11:30 am – 1:00 pm

Presenters: **Erika Earl**, Earl Virtual Innovation Lab, Inc.
Laura Escudé, Laura Escudé Enterprises
Chris Kincaid, Indiana University Purdue University Indianapolis
EveAnna D. Manley, Manley Laboratories, Inc.
Angelica Tavella, Ableton

How can DIY skills change your career trajectory? In this panel we will explore how DIY skills have helped to prepare a variety of professionals for their current, past, and future roles in the audio world. Panelists from a diverse career field including programming, mixing, design, support, education, and more will dig into topics related to how DIY helped them to get their start in their careers all the way to leading successful companies.

• DIY Home Studio Acoustics: Science and Craft of Tuning Your Room At Home
Friday, October 30, 2020
Session Time: 1:00 pm – 2:00 pm

Presenters: **Sorgun Akkor**, STD Acoustics
Timothy Hsu, Indiana University, Purdue University Indianapolis
Chris Kincaid, Indiana University, Purdue University Indianapolis

This workshop/presentation will cover a bit of both the science and craft of building diffusion panels at home. We will begin with an introduction to the topic of home studio room acoustics, followed by diagnosing a room, and finally a look at several solutions based on our measurements, skill level, and access to tools. We will cover common questions and scenarios from the community as well as a live Q&A afterward about both the science and craft of diffusion.

• Roundtable Discussion: Audio Builders Workshop
Friday, October 30, 2020
Session Time: 8:00 pm – 9:00 pm

Presenters: **Jason Bitner**, Traffic Entertainment Group
Dereck Blackburn, Quiethouse Recording
Chris Kincaid, Indiana University Purdue University Indianapolis
Brewster LaMacchia, Clockworks Signal Processing

An open format discussion and hangout with live DIY builds, Q&A, and a DIY show and tell. Bring your one-of-a-kind build, a circuit you are currently building, and questions to ask other DIYers. A fun and laid back event to close out the technical program with Audio Builders Workshop! Ground Rules / Etiquette: 1. You are entering the Discussion on Mute. Please remain on Mute unless you would like to contribute with a comment or question. 2. There is both a chat box and a Raise Hand function—you are welcome to use either of those to indicate that you would like to speak. Please then wait for the Moderator to invite you to unmute. 3. By entering this chat, you have agreed to abide by the AES Code of Conduct, which can be found at https://www.aes.org/download.cfm?filename=AES_Member_Code_of_Conduct_Final.pdf

AUDIO FOR CINEMA

• Otis Osten on Finishing Wizards During a Pandemic

Presenter: **Otis Van Osten**
John F. Whynot, Berklee College of Music, Boston, MA, USA

Otis Van Osten and John Whynot discuss the vagaries of completing the sound and mix for the Netflix series "Wizards" after the COVID-19 pandemic interrupted production.

• Scoring Mixers Cage Match

Moderator: **John F. Whynot**, Berklee College of Music, Boston, MA, USA

Panelists: *Brad Haehnel*
Jason LaRocca
Phil McGowan

Scoring Mixers Jason LaRocca, Phil McGowan, Brad Haehnel, and John Whynot (moderator) discuss their craft, technology, collaboration, and the complications that have arisen in film scoring since the advent of COVID-19

• Simulating Movement in Spatial Audio: Sound Object Displacement Techniques in Ambisonics

Presenter: **Eduardo Luis Patricio,** Independent

This workshop is addressed to those who wish to learn more about sound design and mixing for immersive audio to picture. More specifically, this workshop will focus on strategies to simulate movement (rotation transformations) of one or multiple sound sources as well as whole Ambisonics sound fields in the context of linear visual media. Series of practical examples will be provided with basic workflows focused on achieving results based on the nature of specific visual changes (camera movement, characters displacement, sound effects, etc.). Participants of this workshop will also learn about creating sound fields by encoding individual sound objects, simulating reverberation, and basic track hierarchy.

• Indie Film Sound: Breaking into the Industry
Tuesday, October 27, 2020
Session Time: 1:30 pm – 2:30 pm

Moderator: **Jessica Green,** JAG 42

Panelists : *Miles Ito*, Vero Sound
Matt Kulewicz, Late Night Cartoons / CBS/ Showtime
Spencer G. Shafter, Hobo Audio
Andrew Tracy, One Thousand Birds

Indie Film Sound: How sound engineers adapt to the ever evolving world of film sound with independent filmmakers.

• Commercial Presentation: Source Elements: Remote Collaboration Solutions
Wednesday, October 28, 2020
Session Time: 1:30 pm – 2:30 pm

Presenters: **Robert Marshall**, Source Elements
Rebekah Wilson, Source Elements

Since 2005, Source Elements has been allowing studios engineers, musicians, producers, and actors to work remotely/from home/from anywhere. This year, they present a documentary in collaboration with Vienna's Synchron Stage Orchestra showcasing a global 2-week-long production demonstrating advancements and

updates in ADR, remote recording, review & approval, and more, using their comprehensive set of software and services including Source-Connect, Source-Live, Source-Nexus, and other exclusive technologies.

• Pandemic Practices—Recording VO from Home
Thursday, October 29, 2020
Session Time: 7:00 pm – 8:00 pm

Moderator: **Kiran Kumar,** Apple

Panelist: *Matt Kulewicz,* Late Night Cartoons / CBS/ Showtime
Spencer G. Shafter, Hobo Audio
Andrew Tracy, One Thousand Birds

Pandemic Practices—Recording VO from home; How sound engineers, studios and productions have adapted to remote/home recording during the COVID pandemic; Gear, acoustics, workflow, networks, etc.

• Cataloging SFX: The Universal Category System
Friday, October 30, 2020
Session Time: 6:00 pm – 7:00 pm

Presenter: **Tim Nielsen**, Skywalker Sound

How to catalog sound effects: The Universal Category System. A new initiative to assist in documenting and organizing sound effect libraries for users, vendors and facilities. This presentation will give an overview of the system and show some examples of it in use.

BROADCAST AND ONLINE DELIVERY

• A Century of Radio: What You May Not Know About The History of Broadcasting

Presenters: **Donna Halper**
Barry Mishkind

Inspecting the history of how and when broadcasting began is not only an interesting look at the work and efforts of the pioneers, but helps us understand how we arrived at the current state of the industry. During a discussion between Donna Halper, PhD, Associate Professor of Communication & Media Studies at Lesley University, Cambridge, MA, and Barry Mishkind, Editor/Publisher of the Broadcasters' Desktop Resource you will hear some surprising facts about broadcast history as well as some myths about things we think we know.

• Broadcasting from Home: The Tools, Connections, Hardware, Software, and Workflows that Will Outlive the COVID-19 Pandemic

Presenters: **Josh Agnew**
Rob Bertrand
Josh Bohn
Edwin Bukont
Shaun Dolan, Telos Alliance
Kirk A. Harnack, Telos Alliance
Jeff Heins, MyBridge Radio Network
Ted Johnson
Paul Montoya
Geary Morrill
Gibson Prichard, WTVF NewsChannel5
Douglas Rowe
Chris Tarr
Chris Tobin
Shane Toven, Educational Media Foundation
Charles Wooten

Many broadcasters moved quickly from their familiar and busy corporate studios to individual home-based studios. And while the viral transmission ground rules change every week or two, our radio transmission goal is ever the same: Stay on the air. Early in this rushed move to home-based broadcasting, engineers gathered mics, laptops, audio codecs, and mic-to-USB converters from wherever they could. They provided remote access to station automation systems and remote audio connections for their live shows. Remote voice tracking also ramped up significantly. We'll hear and see directly from broadcast engineers in large, medium, and small markets. The signal paths and workflows have morphed over time, and learn their predictions for the future of quality and timely content creation.

• Create ATSC 3.0 Conform MPEG-H with Common DAWs

Presenter: **Tom Ammermann**, New Audio Technology

MPEG-H is ATC 3.0 specified, so it is included in the TV broadcast standard in Korea, China, USA, and Brazil. Major end-user device manufacturers like Samsung and LG are equipped with MPEG-H; and Sony 360 Reality uses MPEG-H as a delivery format as well. So it's certainly good to know how to do MPEG-H. The presentation will show how to create, edit, monitor, and export MPEG-H content for broadcast applications quickly and easily in common production workflows with every DAW.

• Developing Versatile Performance Spaces for Podcast Production Studios

Moderators: **John Storyk**, WSDG Walters-Storyk Design Group
Joshua Morris, WSDG Walters-Storyk Design Group

Panelists: **Jimmy Buff**
Elliott Forrest, WQXR Radio
Austin Thompson, Spotify

Participants will illustrate their Live Performance Spaces with floor plan "walkthroughs" and video clips of on air podcast performances. Podcasting has been a broadcast staple since radio's earliest days. Recent years have seen this diverse format explode into a global "infotainment" phenomenon encompassing everything from one-on-one dialogues to full-up rock concerts. Podcast Studios size and reach are as varied as their programming formats. This Panel will explore the production facility landscape from a 6200 square foot 100+ year-old building in Kingston, NY, to a 20,000 square foot complex in Brooklyn, NY, and the input of users and engineers utilizing these spaces.

• Keeping The Late Show with Stephen Colbert on Air During COVID-19

Moderator: **Julie Middelburg,** Freelance

Panelist: *Bengt Erik Akerblom*, CBS

Keeping "The Late Show with Stephen Colbert" on air while sheltering-in-place due to COVID-19 was a monumental undertaking requiring creativity and out of the box thinking. Bengt Erik Akerblom, one of the key people who accomplished this feat, is with us to give some insight into how it all happened.

• Pass the Mic

Presenter: **John Holt**

As this is the 100th anniversary of radio broadcasting as we know it, it's a good time to reflect on the development of microphone technology which was spurred on by this new medium. Refinements of carbon and condenser microphones started off the 1920s and

the decade concluded with electromagnetic dynamics providing us with both the moving coil and ribbon velocity microphones. We will look at all of these microphones and their continued improvements into the mid 20th century.

• Podcasts: Telling Stories with Sound

Moderator: **Rob Byers**, American Public Media

Presenters: *Fernando Arruda*
Jim Briggs
Laura Starecheski
Ike Sriskandarajah

Narrative audio in all of its forms —podcasting, radio, audio books —is now a veritable playground for sound design, composition, and mixing techniques traditionally only found in other mediums. The demand for these skills continues to grow, workflows become more complicated—and audio engineers continue to prove their essential value.

Rob Byers (American Public Media, Final Final v2) leads a conversation about the craft of audio storytelling with the production team behind Reveal's investigative series "American Rehab." Sound designer/composers Jim Briggs and Fernando Arruda are joined by series reporter/producers Laura Starecheski and Ike Sriskandarajah to explore what it takes to pull together a multi-year, eight-part audio production. We'll discuss their creative process, workflows, and the techniques they used to tell this important investigative story.

Listen to American Rehab on your podcast app of choice or here: https://www.revealnews.org/american-rehab/

Listen to the score from American Rehab here: https://revealnews.bandcamp.com/album/american-rehab

• Practical Tips for Using Digital Audio in a 2110 Facility

Moderator: **Andy Butler**, Public Broadcasting Service

Presenters: *David Bialik*
Stephen Lampen, www.stevelampen.com
Kent Terry
Peter Wharton

The old adage that the Devil is in the details could never be more applicable. Not siding with Satan but understanding how to dance without getting burned is increasingly important for Audio Engineers in Media. The SMPTE 2110 family of standards is large and continues to grow with time. It incorporates a number of audio technologies to produce both stand alone audio projects and audio as part of other media projects. Understanding how to choose the best tools to support your work can be confusing. A panel of industry experts from Standards Experts to daily users share their tips for success and respond to some of the most challenging issues in this session.

• Radio War Stories from the Trenches

Moderator: **Scott Fybush**, Fybush Media

Presenters: *Tracy Teagarden*, Entercom Las Vegas
Shane Toven, Educational Media Foundation
Britny Williams

Gather a few experienced engineers in one place—even virtually—and before long, the war stories start to flow. That transmitter project that hit a snag, the studio rebuild that's become legendary, the audio processing that still has people talking about it decades later. Join Scott Fybush of NorthEast Radio Watch as he brings some of radio engineering's best storytellers together for an engaging session in which they share some of their favorite tales of their times in the radio trenches.

• Stay Safe: Disinfecting Microphones in the Time of Covid-19

Moderator: **David Prentice**, Dale Pro Audio

Presenters: *Bo Brinck,* DPA Microphones
Eddie Ciletti, Manhattan Sound Technicians Inc.
Ben Escobedo

So much of our media whether music, broadcast, or on line starts with an instrument or voice and a microphone. The biggest danger to artists and engineers used to be a mic with a stale beer smell or a large diaphragm capsule with moisture buildup from a singer's breath. Now the dangers of Coronavirus transmission through contact or or vapor droplets elevates microphone maintenance and sanitation to life and death importance. This program will examine how to keep the artists, engineers, and microphones healthy during recording and performance.

• The Home Radio Studio

Presenter: **Tom Ray**, Tom Ray Broadcast Consulting, LLC

"The Home Radio Studio" will take a took at a studio built at the home of Ron Ananian, host of the nationally syndicated *The Car Doctor* radio program. A network radio studio was built into the equivalent of a closet, with the control room being in the laundry room. We'll look at the equipment configuration, the way we deadened the room, and how we handled (and continue to handle) program production in the Covid-19 crisis.

• The Importance of Loudness

Presenters: **David Bialik**
Robert A. Katz, Digital Domain, Inc.
John Kean, Kean Consultants LLC
Robert Orban
Jim Starzynski

Loudness continues to maintain its importance across multiple media platforms. Many groups are involved with standardizing loudness, while governments continue to show interest in regulations. The AES Technical Committee for Broadcast and Online Delivery is an industry leader in the creation of loudness documents for content creators and distributors of Over-the-Top Television (OTT) and streaming and on-demand audio. Please join our session where AES and Industry experts will discuss AES's recent loudness recommendations for OTT and also describe important work in progress for streaming audio and music.

• The Ins & Outs of Being an A2 Vol 1: Out of the Shadows

Co-Chairs: **Julie Middelburg**, Freelance
Jim Parente, Freelance

Panelists: *Monique Cook*, Freelance
Liz Ip, Freelance Broadcaster
Victor C. Smith, Willing & Able—Freelance

Industry professionals discuss various aspects of what it is to be an A2 within the field of broadcasting. Topics to be discussed will draw from the job description, "a day in the life," an A1's expectations, a digital future, "tricks of the trade," and our "best advice."

• The Ins & Outs of Being an A2 Vol 2: In the Time of COVID-19

Co-Chairs: **Julie Middelburg**, Freelance
Jim Parente, Freelance

Panelists: *Skip Kent*, Freelance
Victor C. Smith, Willing & Able—Freelance

With a pandemic, like any disaster, broadcast professionals must

learn and adapt as they brave their way to bring live events and information to the folks at home. We will take a look at some of the behind the scene changes as they apply to A2's in this continually evolving new reality.

• The Next Generation Audio CDN—What Is Needed

Moderator: **Sam Sousa**, Triton Digital

Presenters: *Malik Abdullah*
Michael Beach, NPR
Louis-Alex Bergeron
Antonio Calderon
Fernando Estigoy
Kirk A. Harnack, Telos Alliance
David Layer, National Association of Broadcasters
Gio Punzo
Marty Sacks
Bruce Wahl

Will encoders at stations still be relevant? If more "Playout in the Cloud" comes along, what does that mean for the station side engineering? How can we solve the need for Cues? Radio automation is trigger/Metadata based, but can we do better? What are the standards, or working standards, looking to resolve this and how can we "as AES, as Publishers, and as Technology providers solve this"? When or How does the Audio CDN distinguish itself from the Video solutions? Do we need to get closer to the video standards and insure they cover better the audio needs ?

• A Virtual Tour and Discussion: BBC Wales: Cardiff Central Square IP-Based Broadcast Facility
Thursday, October 29, 2020
Session Time: 10:00 am – 11:00 am

Moderator: **Adrian Wisbey**, BBC

Presenters: *Daniel Ashcroft*, BBC
Mike Ellis
Jamie Laundon, BBC
Phil Packer
Mark Patrick, BBC

An overview of BBC Wales' new headquarters in Cardiff which has just gone live and is built with ST-2110 and AES67 at its core. This presentation will look at the implementation of the radio facilities which make extensive use of networked audio between studios and the ST-2110 IP Live Core. We will discuss the design decisions, lessons learnt and obstacles that were overcome to get on air.

• A Virtual Tour and Discussion: BBC Broadcasting House Studio
Thursday, October 29, 2020
Session Time: 2:00 pm – 3:00 pm

Moderator: **Jamie Laundon**, BBC

Presenters: *Mark Patrick*, BBC
Paul D. Morgan, BBC Radio
Phil Packer

BBC Broadcasting House has been a London home of the BBC since 1932. The first studio on air in the building was Studio 8A, the "Marching Band Studio." Eighty-eight years later, we have reengineered this studio, now called Studio 80A, with networked audio and the implementation of modern design philosophies. As well as technical design, this presentation also looks at the challenges around training and taking complex systems into support, particularly in a COVID-19 world.

• AES Special Performance: Phantom Power: A Brief History of The Ghost In Our Machines
Thursday, October 29, 2020
Session Time: 8:00 pm – 10:00 pm

Moderators: **Sue Zizza**
David Shinn

Presenters: *Butch D'Ambrosio*
Annie Ellicott
Gary Francis Furlong
Alan Hawkins
Jeff Hays
Ahmed Mahmoud
Larry Oppenheimer
Laurie Winks

Given Halloween 2020 is only a few days away from this AES performance, "Phantom Power: A Brief History of the Ghost in Our Machines" is story of the morning show team at radio station WDMB, who have been quarantining together at the studios while continuing to broadcast since March 2020. On this 100th Anniversary of Commercial Broadcasting, "the ghosts" that live in the machines—that only Phantom Power can defeat—begin to "appear." Produced and directed by Sue Zizza of SueMedia Productions / Radio Waves Studios (suemediaproductions.com) and the HEAR Now Festival (hearnowfestival.org), Phantom Power features members of Soundbooth Theater (soundbooththeater.com), who will be performing live from each of their studios, over the internet. The production will also incorporate recorded and live sound effects and music during the performance. While its cast members hail from all across the country, Soundbooth Theater has been able to present live online performances weekly for their fans that feel as if you are in the studio with them.

EDUCATION

• Unlocking the Control Room: Equity Achievements in Audio
Wednesday, October 28, 2020
Session Time: 11:30 AM – 1:00 PM

Moderator: **Amandine Pras**, University of Lethbridge

Presenters: *Eliot Bates*, The Graduate Center, CUNY
Ky Grace Brooks, CIRMMT, McGill University
Daniel F. Fox, Tufts University, Wondersmith Audio
Mary Mazurek, DePaul University and Columbia College, Chicago
Kat Young, University of York

An international survey in partnership with AES showed that women who work in the recording studio experience more microaggressions than those who work in STEM academia. While gender is the most significant predictor of social discrimination in audio, age, sexual orientation, race/ethnicity, disability, and migrant status also have an impact. In response to these and other statistics regarding gender representation at AES conferences, this panel opens a dialogue based on the achievements of AES members who have applied their expertise to prompt actions towards equity. These contributions have promoted more diverse representation of invited presenters at AES events; the need for moderation of social media; the development of inclusive education programs; and the importance and effectiveness of access to role models, mentorship, and placement opportunities for women and other minorities.

• Clive Davis Institute of Recorded Music Virtual Technical Tour Workshop
Wednesday, October 28, 2020
Session Time: 7:00 PM - 8:00 PM

Moderator: **Alan Watson**, New York University, Clive Davis Institute of Recorded Music

Presenters: *Jim Anderson*, New York University, Clive Davis Institute of Recorded Music
Nick Sansano, New York University, Clive Davis Institute of Recorded Music

The Clive Davis Institute of Recorded Music moved to new facilities in fall of 2019. Four recording studios, 2 audio labs, rehearsal spaces, pre-production rooms, and a critical listening room were created by Francis Manzella. A virtual technical tour will take participants into the rooms. Participants, staff, and faculty from the construction and institute will participate. Presentation is moderated by Alan Watson with frequent interruptions and cogent comments by Clive faculty members Nick Sansano and Jim Anderson.

• Remote Sound Education in the Pandemic Age— A Panel Discussion
Thursday, October 29, 2020
Session Time: 12:00 noon – 1:00 pm

Presenters: **Amy Altadonna**, Artistic Associate with Colt Coeur Theatre company, USA829 member, MFA from Yale School of Drama, TSDCA executive board member, USITT Sound Commission Vice-Chair - UMass Amherst
Sun Hee Kil, State University of New York at New Paltz
Josh Loar, Michigan Technological University
Vincent Olivieri, University of California - Irvine
Christopher Plummer, Michigan Technological University
Joanna Lynne Staub, USA-829, IATSE Local #21, Independent Theatrical Sound Designer & Audio Engineer

Teaching sound usually requires a shared listening experience, which many achieve by gathering in classrooms, production studios, theaters, and the like. With the hit of Covid-19, all teaching went online in a hurry. While we all hope for an end to the pandemic, in the meantime, remote instruction will be with us, in one form or another, for a while. This panel gathers veteran sound educators to discuss the challenges of teaching sound remotely, the solutions they are finding and developing, the tools that are proving essential to this new paradigm, and the areas for improvement and more research going forward. Join this discussion and bring your questions, concerns, and ideas about how to successfully navigate this difficult practice.

• The Loop Lab: A Model for Equity, Access, and Change
Thursday, October 29, 2020
Session Time: 1:00 pm – 1:30 pm

Presenters: **Christopher Hope**, CTS, The Loop Lab
Jonathan Wyner, iZotope/Berklee College of Music

The Loop Lab is a BIPOC-led nonprofit specializing in media arts internships and digital storytelling. It's mission is to empower women and people of color in the media arts to develop careers in audio/video through job training and job placement. It is a successful model for creating access and driving toward equity and access through partnerships and education in audio technology—successfully serving an underrepresented community along with meaningful fundraising and partnership with tech companies, this program is an example of how audio tech and education can bring about change.

ELECTRONIC DANCE MUSIC

• Music Technology in Live Performance

Presenter: **Paul Hunter**, BIMM and SAE

This session will examine some of the ways to incorporate music technology into a live context. It will assess the various aspects of using hardware and software to plan and deliver performance from the studio to stage. Technology has become ubiquitous with today's modern performer through the use of live looping, triggering, manipulation of effects, and remixing on the fly. The tools are now available to enable the music producer to experimental with layers of sound and textures to create and perform within the live domain. This session will look at exporting files from Logic X and setting up Ableton for performing a track live.

• Samplers within Electronic Dance Music Production

Presenters: **Alexandra Bartles**, Altar Studios UK
Rick Snoman, Altar Studios UK

This session will look at the role of the sampler in Electronic Dance Music. In particular the Akai S1000, what its technology initially brought to Electronic Dance Music Production and its role in the modern production of this genre of music.

• Sound Design for Electronic Music with Bitwig
Tuesday, October 27, 2020
Session Time: 11:45 am – 12:30 pm

Presenter: **Thavius Beck**, Noiselab.io

During this course of this presentation, Thavius Beck (electronic musician, Ableton and Bitwig certified trainer) will discuss and demonstrate many techniques for creating specific types of sounds that are commonly used in electronic music. These sounds will be created using the PolyGrid device within Bitwig, which is essentially a modular environment that allows us to connect numerous modules in any way we see fit. By the end, viewers will have a much better understanding of certain sound design techniques that will help them design custom sounds from scratch.

• Eurorack Applications in Electronic Dance Music Production
Wednesday, October 28, 2020
Session Time: 10:00 am – 11:00 am

Presenters: **Alexandra Bartles**, Altar Studios UK
Rick Snoman, Altar Studios UK

Rick Snoman will explore Eurorack and its application in modern Electronic Dance Music production.

ELECTRONIC INSTRUMENT DESIGN AND APPLICATIONS

• Sequential Control in Electronic Instruments
Wednesday, October 28, 2020
Session Time: 4:00 pm – 5:00 pmM

Moderator: **Michael Bierylo**, Berklee College of Music

Panelists: *Richard Devine*
Gur Milstein, Tiptop Audio
Piotr Raczynski, Polyend
Tony Rolando, Make Noise

One of the unique capabilities found in electronic instruments is the ability to sequence events in various types of patterns. While step sequencers and arpeggiators present some of the most common implementations of these techniques, designers have found ways to get past the simple repeating patterns generally associated

with these devices and have developed new designs that incorporate more sophisticated algorithms to yield rich musical results. This session will explore some of the powerful and unique possibilities these designs offer.

• Women in Modular Synthesis
Wednesday, October 28, 2020
Session Time: 5:30 pm – 6:30 pm

Moderator: **Jennifer Hruska**, Berklee College of Music

Panelist : *Caterina Barbieri*, Caterina Barbieri
Lisa Bella Donna, Moog Music Inc.
Suzanne Ciani, Seventh Wave Productions

Some of the most important artist voices in modular synthesis come from women such as Suzanne Ciani, Caterina Barbieri, and Lisa Bella Donna. Moderated by synth designer, artist, and educator Jennifer Hruska this session will be a round table discussion among these artists discussing their art and approach to composing and performing with these intricate electronic instruments.

• Suzanne Ciani Livestream Performance
Wednesday, October 28, 2020
Session Time: 8:00 pm – 9:30 pm

Presenter: **Suzanne Ciani**, Seventh Wave Productions

Suzanne Ciani performs live quadraphonic modular synthesis from Envelop SF. This spatial audio stream allows us to experience Suzanne's immersive compositions online via normal headphones. Suzanne Ciani is a composer, recording artist, and pioneer in electronic music and sound design. She is best loved for her fifteen albums of original music, which feature her performances in a broad array of expressions: pure electronic, solo piano, piano with orchestra, and piano with jazz ensemble. Her music, renowned for its romantic, healing, and aesthetic qualities, has found a rapidly growing international audience, and her performances include numerous benefits for humanitarian causes. sevwave.com Watch on Eventbrite: https://www.eventbrite.com/e/suzanne-ciani-livestream-performance-aes-show-2020-envelop-stream-tickets-125401046907 Please note that AES Show registrants are welcome to use promo code AES2020ESF to view for free. Those not registered for AES Show who use the code will have their Livestream Performance registration rejected.

• Software for Open Source Hardware
Thursday, October 29, 2020
Session Time: 1:00 pm – 2:00 pm

Moderator: **Michael Bierylo**, Berklee College of Music

Panelists: *Astrid Bin*, Bela.io
Andrew Ikenberry, Electrosmith
Owen Osborn, Critter & Guitari

With the proliferation of open source hardware platforms being used in developing new instrument designs, the ability to code for these systems has been an important part of this type of development. While systems like Arduino using the Arduino language and Raspberry Pi using general purpose operating systems like Linux and Windows, newer systems like Daisy from Electrosmith focus on providing development platforms geared more specifically for instrument designers. This session will explore some of the software being used in open source hardware design examining the strengths each has to offer.

• Intuitive User Interface Design for Modern Synthesizers
Friday, October 30, 2020
Session Time: 11:00 am – 12:00 noon

Presenter: **Geert Bevin**, Moog Music

As electronic instruments become more powerful and offer ever expanding synthesis capabilities and control routings, designers are constantly challenged to make using them intuitive. In this session, engineers from Moog Music discuss how they approached the interface and experience design of the Moog One, a three-timbral polyphonic analog synthesizer with digital effects, thousands of parameters, three sequencers and arpeggiators, and a powerful modulation engine. The Moog One is a deep instrument providing some unique challenges for UI designers. Join lead software designers of the Moog One as they give concrete examples of design approaches that make this synthesizer immediately intuitive by harking back to familiar legacy instrument fundamentals while simultaneously embracing modern capabilities.

• Technical Issues in Reissuing Classic Synthesizers
Friday, October 30, 2020
Session Time: 5:30 pm – 6:30 pm

Moderator: **David S. Mash**, Bar of 2 Productions

Panelists: *Mark Crowley*, Moog Music
Eric Christopher, Moog Music
Rosser Douglas, Moog Music
Tim Johnson, Moog Music
Ibon Martinez Garcia, Korg/ARP
Ikuo Tanaka, Roland Corporation
Scott Tibbs, Roland Corporation
Hisakazu Yamasato, Roland Corporation
Yoshito Yamata, Korg/ARP

Early electronic instruments that were popular commercial releases in the1960s and 70s have, in any ways, become archetypes for future developments, inspiring generations of innovations and imitations. Synthesizers from Moog, ARP, and Sequential Circuits, to name few, have become sought after collector's items not only because of their scarcity, but due in no small way to the timeless quality of their sound. In recent years considerable effort has been made to bring these instruments back into production in their original form, closely adhering to the original specifications and components. In this session we'll explore some the technical challenges presented by this work as well as the impact on an already robust synthesizer marketplace.

ENGINEERING BRIEF SESSIONS: AUDIO APPLICATIONS AND TECHNOLOGIES

• **Derivation of Horn Parameters from Numerical Simulations, Analytical Methods, and Measurements—** *Alexander Voishvillo, Balazs Kakonyi, Brian McLaughlin*, Harman Professional Solutions, Northridge, CA, USA

In the authors' previous work, the input acoustical impedance of an axisymmetric horn was obtained through the combination of matrix analysis and dedicated electro-acoustical measurements. This work extends the approach to further refine the derivation of the acoustical impedance and transfer function of several different horns. The results are evaluated against COMSOL modeling and analysis made through the two-microphone method. A one-dimensional matrix model is also used for the horn impedance derivation. Acoustical transfer functions of two different horns are measured and compared with COMSOL modeling results.
Engineering Brief 614

• **Profiling Loudspeaker Nonlinearities Using Parametric Multitone Stimulus—** *Alexander Voishvillo, Balazs Kakonyi, Brian McLaughlin*, Harman Professional Solutions, Northridge, CA, USA

The previous work reported on the use of modulated

musical multitone (MMM) for the det ection of audible distortion in horn drivers. With an adjustable crest factor, MMM is designed to match the properties of music and is used to measure nonlinear distortion audibility thresholds. In this work, the signal is used to create a distortion profile of two-way systems for comparison. The MMM reaction is post-processed to obtain the Multitone Total Nonlinear Distortion response, which presents nonlinear distortion products averaged in a "sliding window" in the frequency domain. MTND responses are derived from a low level to a very high level. These MTND curves are compiled into a comprehensive spectral distortion profile which can be used to compare multiple systems.
Engineering Brief 615

• **A Survey of Current Music Technology & Recording Arts Curriculum Order—***Brett Leonard*, University of Indianapolis, Indianapolis, IN, USA

The rapid proliferation of audio education programs in a variety of institutional settings has led to curricular design that is highly adapted and variable between regions, focuses, and educational environments. While there have undoubtedly been numerous successful approaches to program/curriculum design, there is little published data on the approaches currently employed across the sector, nor the effect that these curricular decisions may have on student outcomes. This initial study presents information on the past and current state of curricular order collected from a voluntary survey including participants from 35 institutions across six countries and three continents. Observations are made on typical topic presentation order, as well as a case study regarding the evolution of curriculum order within one institution.
Engineering Brief 628

• **Cross-Modal Investigations for Improving Sound Localization Accuracy: A Mounted Vibrotactile Headset Design—***Oliver Wilde, Gavin Kearney*, University of York, York, UK

This engineering brief outlines the design and implementation of a vibrotactile headset for use in experiments related to cross-modal auditory localization. The device elicits vibration to the scalp through transducers mounted on a silicon cap and is considered for use in research for improving sound localization in non-individualizedbinauralrenderingthroughtheadditionofan extra modality. The system is optimized to create a somatosensory receptor sensitivity map to vibrational actuation in terms of perceptive directionality on the scalp surface. The paper documents the headset design and the characterization of eccentric rotation motors and linear resonance actuators in order to identify the most suitable vibration motor for use in the device. The motors have been characterized based on mounted and non-mounted dampening by a silicon cap for maximum malleability and comfort for test participants while taking into consideration vibrational frequency, acceleration amplitude (G), displacement amplitude (mm), bone conduction radius (BCR), and amplitude (dB/G). The design is applicable to audio for virtual and augmented reality applications, where the user is required to wear a video headset.
Engineering Brief 630

• **Immersive Network Music Performance: Design and Practical Deployment of a System for Immersive Vocal Performance—***Patrick Cairns, Helena Daffern, Gavin Kearney*, University of York, York, UK

Network Music Performance (NMP) systems provide a solution to remote musical performance over Wide Area Networks (WANs). Empirical NMP research is conducted largely on emulated or academic networks that differ significantly from typical internet connections. This paper presents the technical implementation of an NMP deployment to assess the capabilities of typical modern home internet connections for supporting NMP systems for Immersive Vocal Performance. Audio is streamed between performer home computers using Jacktrip and rendered locally to provide Binaural (virtual Ambisonics approach) playback. Preliminary testing is discussed.
Engineering Brief 632

ENGINEERING BRIEF SESSIONS: AUDIO EQUIPMENT

• **Loudspeaker Characterization by Indirect Measurement of Diaphragm Velocity—***Simon Bellonde, Thibaut Julienne, Antonin Novak*, Le Mans Université, Le Mans, France

A complete characterization of an electrodynamic transducers consists, among others, of measurement of the membrane velocity. Modern measurement devices, such as laser vibrometers and accelerometers, are coming with their set of disadvantages. The laser devices can be costly, the accelerometers, being attached to the device under test, are influencing the measurement and are unusable for small-sized loudspeakers. This paper explores the indirect measurement of the membrane velocity, using a single microphone, and the limits for measurement of loudspeaker parameters, including the frequency-dependent ones due to creep effects and eddy currents.
Engineering Brief 621

ENGINEERING BRIEF SESSIONS: AUDIO PROCESSING

• **An Approach for Implementing Time-Stretching as a Live Realtime Audio Effect—***Colin Malloy*, University of Victoria, Victoria, BC, Canada

This paper presents a novel buffer management technique that allows time-stretching to be implemented as a live realtimeaudio effect. Previously, implementing time-stretching as aneffect in a realtime audio system with live input and output was not practical due to technical and perceptual problems. This system is based on the circular buffer, but with a non-continuous, audio level triggered write methodology. This modification compensates for the fact that the output of a time-stretcher is longer than the input. This allows synchrony between a live input and output to be maintained as well as reducing the processing and memory requirements. The buffer management technique is paired here with the Paulstretch time-stretch algorithm, but can be adapted for use with other time-stretch algorithms as well. Also discussed are the implementation design details including live, manipulatable FFT window size and stretch factor parameters.
Engineering Brief 613

• **Sound Quality Improvement of MPEG-H 3D Audio Encoder—** *Akifumi Kono, Hiroyuki Honma, Toru Chinen*, Sony Corporation, R&D Center Tokyo Laboratory, Tokyo, Japan

In 2019 Sony launched 360 Reality Audio, which provides a new music experience using object-based spatial audio technology. Object-based audio contains information on time-varying object loudness and location and audio data, which are transmitted to playback devices, and then rendered and played back. It was reported in [148th AES Convention, eBrief 581] that object locations affect the subjective sound pressure perception depending on the direction of the sound source. In this e-brief we present an approach to increase the sound quality by considering the loudness and locations of objects. We perform a subjective listening test for three test items. The results indicate that two items had statistically significant differences in sound quality.
Engineering Brief 623

- **Generalized Digital Second Order Systems Beyond Nyquist Frequency—***Champ Darabundit,*[1,2] Russell Wedelich[2]
[1] Standord University, Stanford, CA, USA
[2] Eventide Inc., Little Ferry, NJ, USA

The Bilinear transform is an established design method for producing digital representations of analog filters. But due to frequency warping inherent in the transform, adequate representation in the digital domain is limited for analog filters designed near or above the Nyquist frequency. Previous work has demonstrated a method for surpassing this design limitation in the case of parametric filters. This paper is an extension of that previous work, and proposes a design methodology for digitally representing any second order system with a center frequency near or above the Nyquist frequency. By repositioning the system poles, this method compensates for the effects of frequency warping, and allows for a more exact digital replication of an analog system's magnitude response.
Engineering Brief 631

ENGINEERING BRIEF SESSIONS: IMMERSIVE AUDIO

- **Open Database of Spatial Room Impulse Responses at Detmold University of Music—**Sebastià V. Amengual Garí, Banu Sahin, Dusty eddy, Malte Kob, Detmold University of Music, Detmold, Germany

This manuscript presents an open source database of Spatial Room Impulse Responses (SRIR) captured at three different performance spaces of the Detmold University of Music. It includes one medium sized concert hall (Detmold Konzerthaus), one chamber music room (Brahmssaal), and one theater (Detmold Sommertheater). The collection contains approximately 600 multichannel RIRs corresponding to several source and receiver configurations. For each room we include measurement positions on stage and at the audience area captured with both an artificial head and an open microphone array compatible with the Spatial Decomposition Method (SDM). The Detmold Konzerthaus holds a large scale Wave Field Synthesis system, and SRIRs of an ensemble of focused sources on stage and conditions of increased reverberation are also included.
Engineering Brief 617

- **ArcheoEchi—A Virtual Reconstruction of a Medieval Cathedral in Southern Italy—***Gianluca Grazioli*, McGille University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

ArcheoEchi is an interactive virtual reality experience for Oculus Go based on the medieval cathedral of Saint Albert, located in the Southern Italian archeological site of Montecorvino (Foggia). The project saw the collaboration of the Department of Humanities of the University of Foggia and the AudioLab of the University of York, and it concerned the 3D modeling of the church, the auralization of its acoustics and the implementation of a binaural listening system. The development contributed to raise new questions about architectural acoustics of ancient heritage and implement new solutions for designing interactive systems. Thanks to both its educational value and academic background, ArcheoEchi shows how immersive technologies can enhance the offer of cultural heritage and inspire new academic studies.
Engineering Brief 618

- **Design and Application of a Native-D Recording Format for Optimal Dolby Atmos Reproduction—***Aron Kobaashi Ritch*, New York University, New York, NY, USA

In this paper I will propose a Native-D format immersive microphone system made specifically for mixing and reproduction within the Dolby Atmos framework. While there are a number of immersive recording systems in use today, most have been designed for reproduction within listening environments that use sub 90° angled height channels such as Auro-3D. The microphone system proposed in this paper aims to take advantage of the native 90° height-to-main layer angle found in Atmos speaker systems by combining a non-coincident main layer with a corresponding near-coincident, 90° directional microphone for each height channel. This system excels in its high-fidelity capture of both group and individual sources, and benefits from a high level of decorrelation from channel to channel. An additional advantage of the system is its capture of Native-D format signal, which ensures that no format conversion or complex matrixing must be done, avoiding a loss of fidelity from recording to mixing stages. A case study of this system, consisting of multiple recording sessions, has been done in order to establish the validity of a system of this type, ultimately resulting in a final mix using Dolby Atmos. Both a "native immersive" and a "non-native" recording approach were taken in order to exemplify the system's versatility.
Engineering Brief 619

- **Designing a 9-Channel Location Sound Microphone from Scratch—***Florian Camerer*, ORF – Austrian Broadcasting Corporation, Vienna, Austria

The design of a 9-channel microphone system for location recording of mainly atmospheres will be described. The key concept is matching the recording and reproduction angles of the individual sectors. The rig is designed for the AURO-3D 9-channel playback system (4 height speakers). An analysis of the reproduction angles will be included, as well as recording concepts like the Stereo Recording Angle (SRA), Williams curves, Scale Factors for different reproduction angles than 60°, and diffuse field decorrelation. Finally, practical aspects like microphone mounts and windshields for such a system will be presented.
Engineering Brief 622

- **Development of a 4-pi Sampling Reverberator, VSVerb—Application to VR Production—***Masataka Nkaahara,*[1,2] *Yasuhiko Nagatomo,*[3] *Akira Omoto*[1,4]
[1] ONFUTURE Ltd., Tokyo, Japan
[2] SONA Corporation, Tokyo, Japan

[3] Evixar Inc., Tokyo, Japan
[4] Kyushu Univerity, Fukuoka, Japan

The authors developed a 4-pi sampling reverberator, named "VSVerb," which restores a 4-pi reverberant field by using information of virtual sound sources that are captured in a target space. Acoustical properties of virtual sound sources are detected from sound intensities which are calculated from impulse responses measured by an Ambisonic A-format microphone. Spatial information of the virtual sound sources is translated into time responses, then the 4-pi spatial reverberation is obtained. Several schemes for detecting accurate acoustic properties of virtual sound sources have been developed so far, and their practicalities have been examined under various playback environments, e.g,, 5.1.4ch, 7.1.4ch, 22.2ch, 24ch, and 40ch. This time, the authors introduce an application example for implementation of the VSVerb into VR content production. Key features are 1) Binaural rendering by using an individual HRIR which is given as an AES69 sofa format and 2) Dynamic processing of changing reverberation with chasing listener's movement.
Engineering Brief 634

- **Multimodal Immersive Motion Capture (MIMiC): A Workflow for Musical Performance—***Cindy Bui, Andrea Genovese, Trey Bradley, Agnieszka Roginska*, New York University, New York, NY, USA

This paper introduces a workflow to record paired motion capture and sound of musical performances using an infrared light-based tracking system. This paper discusses example production cases ranging from short loops to full performances, addressing the production challenges and compromises, substantially different from a regular audio recording process. Motion capture is very versatile and can be applied in many disciplines including musical pedagogy, distributed performances, machine learning, and as creative assets. Paired audio-mocap data can be rendered into similar visual content for interactive virtual environments, live performances or fixed media. This extensible workflow can be adapted and extended for different purposes in musical performances to promote further research and creative content development.
Engineering Brief 635

ENGINEERING BRIEF SESSIONS: PERCEPTION

- **Investigation of a Real-Time Hearing Loss Simulation for Use in Audio Production—***Angeliki Mourgela*,[1] Trevor R. Agus,[2] *Joshua D. Reiss*[1]
[1] Queen Mary University of London, London, UK
[2] Queen's University, Belfast, UK

We present a perceptually motivated, real-time hearing loss simulation for use in audio production. The implementation builds on a previous simulation, but is now real-time, low latency, and available as a stereo audio effect plug-in with more accurate modelling of hearing loss. It offers the option of isolating and customizing high frequency threshold attenuations on each ear, corresponding to the audiogram information. The simulation also provides the option of incorporating additional suprathreshold effects such as spectral smearing, rapid loudness growth, and loss of temporal resolution on audio. The underlying psychoacoustic principles are described, and results are presented to show the simulation's performance.
Engineering Brief 620

- **Overview of Speech Quality Metrics in Terms of Automated Evaluation of Signal Denoising in a Presence of Non-Stationary Noise—***Karol J. Duzinkiewicz, Damian Koszewski, Kamila Pietrusinska, Pawel Trella*, Intel Technology Poland Gdansk, Poland

Recent developments in neural network-based speech enhancement ask for a robust subjective metric that can be used for comparing performance of a different noise suppression algorithms (for non-stationary noises), that would closely match costly and time-consuming subjective user tests. The article describes results of a comparison between subjective scores obtained using MUSHRA methodology vs. automated evaluation with objective metrics i.e., POLQA, 3QUEST, STOI, and ESTOI, on a set of recordings processed by two different denoising algorithms for close and far speaker distance. Correlation coefficient is calculated between subjective scores and examined metrics. The results are based on recordings simulated using an in-house simulation toolchain, based on impulse responses from an actual laptop device used in a low reverb quiet room.
Engineering Brief 625

- **Evaluating the Accuracy of Musicians and Sound Engineers in Performing a Common Drum Tuning Exercise—***Rob Toulson*,[1] *Marques Hardin*[2]
[1] RT60 Ltd., London, UK
[2] Anglia Ruskin University, Cambridge, UK

A listening test was conducted with particular relevance to the scenario of tuning a drumhead to have an equal or even overtone frequency response around the perimeter of the drum. If the sound of one drumhead location is identified as higher or lower frequency than another, then tuning adjustments can be made to achieve a uniform vibration profile. A subject group of 24 experienced musicians and sound engineers were asked to compare 18 pairs of A and B drumhead sounds. In total, only 45.4% of A-B comparisons were correctly identified as higher, lower or equal frequency.
Engineering Brief 629

- **Huddersfield Universal Listening Test Interface Generator (HULTI-GEN) Version 2—***Dale Johnson,, Hyunkook Lee*, University of Huddersfield, West Yorkshire UK

This paper describes a new version of HULTI-GEN (Huddersfield Universal Listening Test Interface Generator). The Max-based tool features a number of new improvements over the previous version, including the addition of psychometric testing methods (e.g., 2I-2AFC, ABX, Yes-No, etc., in adaptative and non-adaptive procedures) to the existing ITU methods; playback of multichannel stimuli (up to 64 channels); a SOFA-based, real-time binauralizer with headphone equalization filters; statistical-analysis-ready results files; and an open-source framework for experimenters to modify the existing tests or design their own test methods for HULTI-GEN. The tool is designed so that the setup, testing, and results gathering processes are effortless. The software is freely available from https://www.hud.ac.uk/apl.
Engineering Brief 633

- **Spatial Auditory Masking Caused by Phantom Sound Images—***Masayuki Nishiguchi, Kazuki Ishimoto, Kanji*

Watanabe, Koji Abe, Shouichi Takane, Akita Prefectural University, Akita, Japan

The spatial auditory masking effect caused by real sound source signals in a three-dimensional sound field has been previously examined [1]. However, in this study, spatial auditory masking caused by a phantom sound image generated by multiple sound sources in a 3D sound field was investigated. We have examined masking threshold levels for various phantom masker locations at maskee locations of 360° around a listener. The results were closely comparable to those where the real masker signals were employed. According to the test results, it was deduced that spatial auditory masking occurs in accordance with the sound image locations rather than the sound source device locations.
Engineering Brief 636

- **A Model to Predict the Impact of Dialog Enhancement or Mix Ratio on a Large Audience**—*Aaron Master, Hannes Müsch*, Dolby Laboratories, San Francisco, CA, USA

Increasing the speech-to-background mix ratio of content, either algorithmically through dialog enhancement (DE), or during production, is considered a means of reducing listening effort for an audience, some members of which have hearing impairments. But what exactly is the expected benefit? A portion of the audience can already follow the content effortlessly and dialog boosting will not improve their perception. Other parts of the audience are severely impaired, and their speech reception performance will improve until all background is removed. We introduce a model that predicts which parts of an audience benefit by how much from changing the speech-to-background mix ratio of a piece of content. The model is intended to allow decision makers to predict what impact changes in audio production guidelines or DE technologies will have on their audience.
Engineering Brief 637

ENGINEERING BRIEF SESIONS: RECORDING, PRODUCTION, AND REPRODUCTION

- **Design and Evaluation of a Binaural Chorus Effect**—*Sam Platt*, New York University, New York, NY, USA

A binaural chorus effect is designed in Max/MSP and Python, aiming to take the chorus effect beyond its current use in stereo mixes. Six separate iterations of the design are evaluated alongside a stereo version, focusing on how changes in azimuth position and at what point binaural synthesis is rendered in the signal processing affects the final product. Results are not found to be conclusive enough to set design standards, but are conclusive enough to provide a baseline methodology for future works.
Engineering Brief 616

- **"LDK-Cube" and "Time-Space-Energy-Equalization": A Case Study of Binaural 3D Recording and Production of an Orchestral Concert with a Sound Reinforcement System in Operation**—*Haiyan Xiang*,[1] *Xiang Ji*,[2] *Dakang Li*,[3] *Jie Zhu*,[4] *Peng Huang*,[5] *Shusen Wang*[6]
[1] Chengdu Radio and TV Station, Shengdu, China
[2] Shenzhen EZPro Sound & Light Technology Co., Ltd., Shenzhen, China;
[3] Communication University of China, Beijing, China
[4] Beijing Dreamula Digital Technology Co., Ltd., Beijing, China
[5] Beijing Film Academy Beijing, China
[6] AIYISI (Beijing) Science & Technology Col, Ltd., Beijing, China

As for the real-time immersive sound recording and production for an orchestral concert amplified through a sound reinforcement system, how can a TV program be produced binaurally with real 3D ambience and audience capture, and potentially for the purpose of multi-platform distribution? This engineering brief describes innovative solutions applied at the 2020 Chengdu New Year Concert: a largely-spaced 3D microphone array for capturing hall ambience, named the LDK-Cube, with a process of High Order Ambisonics encoding and binaural rendering; and Time-Space-Energy-Equalization (TSEEQ), a novel concept which means optimizing time-space-energy composition, through an integrated process of spectral and spatial equalization/delay, timeline EQ/delay, and group delay, is first presented in this paper as an overarching principle for the solutions. The perception that sound einforcement disarranges the recording, forcing broadcasting engineers to fall back on pre-recorded audience effects for years, has thus been subverted. Furthermore, binaural/transaural 3D encoded 2-channel recordings providing much better spatial perception, proved to be the future of the stereo, "bionic TSEEQ" is derivate thereof.
Engineering Brief 624

- **Snare Drum Data Set (SDDS): More Snare Drums than You Can Shake a Stick At**—*Matthew Cheshire*, Birmingham City University, Birmingham, UK

A comprehensive acoustic Snare Drum Data Set (SDDS) is recorded, providing extreme tonal variation of the snare drum. SDDS affords researchers a collection of recordings that mimic a wide range of real world studio practices, suitable for diverse research tasks. This is achieved by recording varied velocity snare strikes using 53 microphones, 10 tonally distinct snare drums, and several common dampening methods, recorded in an acoustically treated studio utilizing both close and far micing positions and 3 mic preamps. The resulting data set is 2,544 WAV audio files captured from 48 performances.
Engineering Brief 626

- **Design and Validation of a Low-Cost Acoustic Anechoic Chamber**—*Bram Cuyx*,[1,2] *Wim Buyens*,[2] *Wim Desmet*,[1,3] *Toon van Waterschoot*[1]
[1] KU Leuven, Leuven, Belgium
[2] SoundTalks NV, Leuven, Belgium
[3] Flanders Make, Leuven, Belgium

This engineering brief describes the design, construction and validation of a low-cost acoustic anechoic chamber. This anechoic chamber will be used in a production setting as well as for research and development of a microphone array system. Three chambers have been built using the same method. The performance of the rooms is evaluated in terms of sound isolation, deviation from inverse square law, frequency response function, and reverberation time. Results show that in the frequency range of interest (250 – 11025 Hz), the reverberation time for the three chambers is 19 ± 3 ms. The difference between the frequency response of the chambers is ± 4 dB in the frequency range of interest for the two latest built chambers, whereas the first chamber shows a larger deviation.
Engineering Brief 627

GAME AUDIO AND XR

• Being Black in Game Audio: IASIG Diversity Panel

Moderator: **Raheem J. Jarbo**

Panelists: *Marisa J. Ewing*, Hemlock Creek Productions
Joshua Matthews, Joe Sua Music
Devlon S. Samuels, Devlon Samuels Music
John H. Smith, Slide20XX

The IASIG recognizes that the game audio industry has a long way to go in terms of equal opportunity and access for minorities. As such we are making it a priority to elevate the voices of minorities to share their experiences with the wider audio community. We've brought together a panel of black composers and sound designers to AES to do just that, inform and inspire others through sharing their thoughts on the industry and struggles they have experienced and overcome. The goal of this panel is to recognize some shortcomings of the industry and inspire change through awareness and conversation.

• Synthesizing Sounds for Video Games and Films

Presenter: **Aleksandr Khilko**, Wargaming

My session will be about reverse engineering of any sounds and recreating it with synthesizers. This talk will be half theoretical / half practice. Some examples: https://www.youtube.com/watch?v=AisvxaCkAbI and https://www.youtube.com/watch?v=ITX2-W-ft_Q

• Mixing It up at Home
Tuesday, October 27, 2020
Session Time: 5:30 pm – 6:30 pm

Presenters: *Jason Kanter*, Avalanche Studios
Mark Petty, Gearbox Software Frisco
Lori Solomon, Dolby
Jay Weinland, Industrial Toys

Earlier this year audio professionals worldwide had to quickly transition their work environments from office to home, which presented many challenges—and opportunities. In this panel discussion you'll hear from game industry professionals about how they managed the change, what lessons were learned and what new rules and workflows they will adopt in the future.

• Roundtable Discussion: Game Audio & XR
Thursday, October 29, 2020
Session Time: 8:00 pm – 9:00 pm

Moderator: **Steve Martz**

Connect with your colleagues for lively discussion! Follow-up on sessions. Ask questions. Share your experience and perspectives. Network with friends and experts. Ask presenters and authors your questions. Get involved in a conversation on game audio & XR. Each Roundtable Discussion will be moderated by the Chair(s) of the Track. We hope you'll join us! Ground Rules / Etiquette: 1. You are entering the Discussion on Mute. Please remain on Mute unless you would like to contribute with a comment or question. 2. There is both a chat box and a Raise Hand function—you are welcome to use either of those to indicate that you would like to speak. Please then wait for the Moderator to invite you to unmute. 3. By entering this chat, you have agreed to abide by the AES Code of Conduct, which can be found at https://www.aes.org/download.cfm?filename=AES_Member_Code_of_Conduct_Final.pdf

HIP HOP AND R&B

• Deliverables—What To Do When You've Finished the Mix with Gloria Kaba

Moderator: **Paul Womack**, Audio Engineering Society

Presenter: **Gloria Kaba**

Your mix is finally finished; the artist loves it, the label loves it. It's time to assemble all the files for mastering and backup, but what exactly do you need? In this workshop we welcome engineer Gloria Kaba to explore exactly what assets you need to professionally deliver a mix.

• Black Music Round Table
Wednesday, October 28, 2020
Session Time: 12:00 noon – 1:30 pm

Moderator: **Paul Womack**, Audio Engineering Society

Panelists: *Prince Charles Alexander*, Berklee College of Music
Leslie Gaston-Bird, Audio Engineering Society
Jeriel Johnson, Recording Academy
Carolyn Malachi, Howard University

2020 has proven to be a landmark year in the struggle for social justice in America, though how that will affect the entertainment industry is yet to be determined. Black music has been central to the music, film, and TV industries for decades, but how does this current shift in cultural awareness impact opportunities for BIPOC artists and technicians? Moderated by Paul "Willie Green" Womack, this round table discussion will feature insight from industry veterans including: Prince Charles Alexander (Grammy Award Winning Producer/Engineer, Professor - Berklee College of Music) Leslie Gaston-Bird (Audio Engineer, Author, Educator, Governor at Large - Audio Engineering Society) Jeriel Johnson (Exec. Director - Recording Academy [Washington D.C. Chapter]) Carolyn Malachi (Grammy Nominated Artist/Songwriter, Professor - Howard University, Engineer)

• Roundtable Discussion: Hip Hop & R&B
Wednesday, October 28, 2020
Session Time: 2:30 pm – 3:30 pm

Moderator: **Paul Womack**, Audio Engineering Society

Connect with your colleagues for lively discussion! Follow-up on sessions. Ask questions. Share your experience and perspectives. Network with friends and experts. Ask presenters and authors your questions. Get involved in a conversation on Hip Hop & R&B. Each Roundtable Discussion will be moderated by the Chair of the Track. We hope you'll join us! Ground Rules / Etiquette: 1. You are entering the Discussion on Mute. Please remain on Mute unless you would like to contribute with a comment or question. 2. There is both a chat box and a Raise Hand function – you are welcome to use either of those to indicate that you would like to speak. Please then wait for the Moderator to invite you to unmute. 3. By entering this chat, you have agreed to abide by the AES Code of Conduct, which can be found at https://www.aes.org/download.cfm?filename=AES_Member_Code_of_Conduct_Final.pdf

• Sample Based Production and Mixing with Eddie Sancho & The Alchemist
Thursday, October 29, 2020
Session Time: 10:30 AM – 11:30 AM

Moderator: **Eddie Sancho**

Presenter: **The Alchemist**

With a career spanning four decades, The Alchemist is synonymous with the highest quality Hip-Hop. The producer, DJ, and rapper has collaborated with the most notable names in the history of the genre, such as Nas, Mobb Deep, Eminem, Snoop Dogg, and countless more. Engineer Eddie Sancho's career also stretches across four decades and has built an indisputable discography including The Notorious B.I.G., Jay-Z, DJ Premier, and Janet Jackson. These two veterans are not only legacy artists however. Over the past few years, the two have connected with modern luminaries like Freddie Gibbs, Action Bronson, Griselda, and Earl Sweatshirt in a tremendous streak of albums and singles, many of which releasing on Alchemist's own ALC Records. In this conversation we will explore their individual and collaborative processes and discuss how their sounds not only stand the test of time, but push the genre forward.

• The Sauce - Exploring Vocal Treatments for Modern R&B with John Kercy, D'Mile & Lucky Daye

Thursday, October 29, 2020
Session Time: 5:30 pm – 6:30 pm

Moderator: **Paul Womack**, Audio Engineering Society

Panelists: *Lucky Daye*
Dernst Emile
John Kercy

If the vocal is the most important part of a song, then the treatment of that vocal must be critical as well! In this session we will explore modern vocal effects through the lens of 4x Grammy nominated artist Lucky Daye's acclaimed album "Painted." Joining us will be recording and mix engineer John Kercy, who is responsible for mixing that album, along with special guests producer D'Mile and Lucky Daye himself!

HISTORICAL

• Jack Mullin Interview Reissued

Presenters: **Dan Mortensen**, Dansound Inc.
Bill Wray, AES

From 1877 with the birth of Edison's cylinder phonograph, how much time elapsed before home audio achieved hi-fi quality? How good was magnetic tape recording in the U.S.A.in 1948? Who built the first magnetic tape recorder? See answers to these questions and more with this fascinating video presentation by Jack Mullin (1913–1999), the man who brought the German tape recorder to the U.S.A. after the war, and helped start the professional audio tape recording industry in America. See Mullin's popular display of historical equipment, discussion of his extensive collection, and his life. First presented at the 1988 85th AES Convention in Los Angeles. This presentation features a new introduction by Ron Streicher, producer of the original video.

• Notes from the League of Young Musicologists

Presenters: **David Katznelson**, Birdman Recording Group, Inc.
Skip Walter, Marin Country Day School

For the past few years the 8th grade history department of Marin Country Day school has lead a class through a state-of-the-art musicology program, using recordings from the Great Depression to unlock the mysteries of that period of time. The premise: music can tell us the stories of our lives, we just have to learn the right questions to ask. Goals: As John Steinbeck wrote in his forward to *Hard Hitting Songs for Hard-Hit People*, "You can learn more about a people by listening to their songs than in any other way, for into the songs go all the hopes and hurts, the angers, fears, the wants and aspirations." No time was that more evident than in 1930s United States.

• Sigma Sound Studios—A History in Pictures

Presenter: **Arthur Stoppe**, Mountain Wind Media

Founded in 1968, Sigma Sound Studios was the center of the "Sound of Philadelphia" during the 1970s and beyond, and a facility was later opened in New York City as well. Sigma Sound engineer Arthur Stoppe, a veteran of 25 years at the studio, recounts its history while presenting an insider's view using photos he took during the heyday of the Philly Sound, including some images not previously shown publicly. The photos show not only the facilities, but provide a behind-the-scenes look at Sigma's staff and clients at work. Other Sigma engineers will add their recollections of life in the studio during the panel discussion that follows.

• Sylvia Massy's Mind-Blowing Microphone Collection!

Presenters: **Sylvia Massy**
Dan Mortensen, Dansound Inc.

Sylvia Massy shows some of the most compelling, interesting, and extremely rare mics in her collection while sharing stories about the first 100 years of microphone development. And, she lets us hear some of her classic mics in action. In addition to owning the largest microphone collection in the world, Sylvia maintains the world's largest library of original microphone documentation and supporting materials. She keeps over 1000 unique vintage microphone models in her studio and that's only a portion of her entire collection, which contains several historically significant and unusual microphones that simply don't exist anywhere else. This is a re-edit of a presentation from August 2020 for the PNW Section of the AES, which was her first showing of any part of the collection. We think it's worth sharing with a wider audience.

• The Development of Large Scale Concert Sound, and Bill Hanley's Role in Perfecting It
Wednesday, October 28, 2020
Session Time: 2:00 pm – 3:00 pm

Presenter: **John Kane**, Institute of Art and Design at New England College

In my presentation I examine how sound engineer pioneer Bill Hanley's contributions in outdoor live sound helped create entire music production industry. Through an extensive ten-year interview and archival process, I uncover how Hanley and his company—Hanley Sound—impacted a developing festival and concert marketplace.

• Roundtable Discussion: Historical
Wednesday, October 28, 2020
Session Time: 2:30 pm – 3:30 pm; 8:00 pm – 9:00 pm

Moderator: **Dan Mortensen,** Dansound Inc.

Join the Roundtable here: https://us02web.zoom.us/j/82717775563?pwd=bjdHa0RyTWE5Yk5PRDVhNUhUN2RJUT09 Passcode: 000012. These Round Tables were created to both to allow various audio interest groups to gather and share ideas, and also to allow the Presenters of On-Demand and Live presentations to interact face to face with attendees who are interested and may have questions about those topics. To that end, the Historical Track has scheduled (in no order) the following presenters to be present: Sigma Sound History: Arthur Stoppe Large Concert Sound/Hanley: John Kane League of Young Musicologists: David Katznelson Semantics of Reissue Mastering: Andreas Meyer/Jenn Nulsen (both TBD) Ground Rules / Etiquette: 1. You are entering the Discussion on Mute. Please remain on Mute unless you would like to contribute with a comment or question. 2. There is both a chat box and

a Raise Hand function – you are welcome to use either of those to indicate that you would like to speak. Please then wait for the Moderator to invite you to unmute. 3. By entering this chat, you have agreed to abide by the AES Code of Conduct, which can be found at https://www.aes.org/download.cfm?filename=AES_Member_Code_of_Conduct_Final.pdf

• The Semantics of Reissue Mastering while Remaining Truthful to the Original Bruno Walter Stereo Recordings
Friday, October 30, 2020
Session Time: 2:30 pm – 3:30 pm

Presenters: **Andreas Meyer**, Meyer Media LLC / Swan Studios NYC
Jennifer Nulsen, Swan Studios NYC

Historical reissue comes with challenges that go beyond physical handling. Technical understanding is given to ensure optimal transfer with no damage to original media. But once this is done, the job of remastering becomes a question of aesthetic, historical integrity and expectation. The latter is not just for the listener but also for the client. What about previous reissues that may or may not have adhered to the original first release? Should that even be considered? What about cross cultural considerations of listening perspective? How do we find balance among all these variables without distorting the truth? This presentation will focus on one of the most important catalogs of recording from the Sony Masterworks archives: Maestro Bruno Walter.

IMMERSIVE AND SPATIAL AUDIO

• Artistic Immersive Sound Contents Creation using the 22.2 Multichannel Sound—Part 2

Presenters: **Kimio Hamasaki**, ARTSRIDGE LLC
Misaki Hasuo, WOWOW Inc.
Miyu Hosoi, Freelance
Hideo Irimajiri, WOWOW Inc.
Toru Kamekawa ,Tokyo University of the Arts

Immersive sound becomes the new trend of the audio industry, and many musical productions have been already carried out in immersive audio. The 22.2 multichannel sound system was developed in 2004 and has become the reference 3D multichannel sound system for the researches, subjective evaluations, and various international standards such as ITU-R. This workshop introduces recent production examples of the artistic immersive sound contents creations using the 22.2 multichannel sound and explains its advantages and features that previous music productions could not achieve. Workshop panelists will discuss the benefits of immersive sound for the music creations and the several issues they encountered during their creations and productions concerning the immersive sound.

• Confusion in the Consumer Experience of Spatial Audio in Games

Moderator: **Steve Martz**

Panelists: *Patrick Flanagan*, Logitech
Adam Levenson, Waves Audio
Aaron McLeran, Epic Games

Spatial audio, 3D audio, binaurally rendered, object-based, higher order ambisonics, virtualizers. So many names, so much jargon and there is a lack of interoperability in spatial audio for games that means end-users may get a suboptimal listening experience—an experience that may be far from the game designer's artistic intent. A complex pipeline of processing brings game sound through audio engine rendering to game output, to platform processing, and even additional headset processing. Check out this engaging conversation moderated by Steve Martz to hear perspectives into how to unravel the complexities and potential ways to rework the spatial audio pipeline for games.

• Recording and Post-Production in Live Virtual Acoustics: "The Lost Voices of Hagia Sophia"

Presenters: **Jonathan S. Abel**, CCRMA, Stanford University
Steve Barnett, Barnett Music Productions
Elliot K. Canfield-Dafilou, CCRMA, Stanford University
Preston Smith, Perfect Record

Acoustics are integral to the performance of music, affecting the tempo, phrasing, pitch contours, among a myriad of performance choices. Particularly in reverberant spaces, such as Hagia Sophia, Istanbul, musicians so thoroughly interact with the space that it becomes another instrument. Live simulation of acoustics in a studio setting allows recording of musical performances in spaces which are inaccessible or no longer exist. Here, we adapt these technologies to capture musical performances in a live simulated acoustic and describe the recording and post-production of "Lost Voices of Hagia Sophia" by Cappella Romana, for which we designed and synthesized the virtual acoustics, including positioning performers in the nave and rendering the immersive acoustic energy reflected from the main dome and colonnades.

• Roundtable Discussion: Immersive & Spatial Audio
Tuesday, October 27, 2020
Session Time: 2:30 pm – 3:30 pm

Moderator: **Steve Martz**

Connect with your colleagues for lively discussion! Follow-up on sessions. Ask questions. Share your experience and perspectives. Network with friends and experts. Ask presenters and authors your questions. Get involved in a conversation on immersive & spatial audio. Each Roundtable Discussion will be moderated by the Chair(s) of the Track. We hope you'll join us! Ground Rules / Etiquette: 1. You are entering the Discussion on Mute. Please remain on Mute unless you would like to contribute with a comment or question. 2. There is both a chat box and a Raise Hand function – you are welcome to use either of those to indicate that you would like to speak. Please then wait for the Moderator to invite you to unmute. 3. By entering this chat, you have agreed to abide by the AES Code of Conduct, which can be found at https://www.aes.org/download.cfm?filename=AES_Member_Code_of_Conduct_Final.pdf

• Abbey Road Spatial Audio Forum: The Role of Game Engines in Music Production Platforms of the Future
Friday, October 30, 2020
Session Time: 12:00 noon – 1:00 pm

Moderators: **Gavin Kearney**, University of York
Mirek Stiles, Abbey Road Studios

Panelists: *Stephen Barton*
Anastasia Devana, Hear XR
Ana Monte, DELTA Soundworks
Adam Smith

The lines are becoming increasingly blurred between traditional music production workflows and systems that enable content creation and mixing to take place directly in VR. In this panel we muse the question—What is the future role of game engines in the music production pipeline? How can game audio engines change the recording and mixing paradigms of commercial and home studios? What are the new creative possibilities? What could it mean for the consumer? In this open session of the Abbey Road Spatial Audio Forum, leading experts in music production, virtual and

augmented reality, composition and game audio will come together to discuss and explore the current challenges and new possibilities for music production using game engines.

IMMERSIVE MUSIC

• 360 Binaural Remix—"Superstition"

Presenters: **Brian (Bt) Gibbs**, AES SF Section Chair

A presentation on Stevie Wonder's "Superstition" utilizing Dolby Production Suite to demonstrate vs. the stereo "2-mix." Playback will also be shown through a separate platform to represent how varying renders can impact the sonics of a binaural mix. Additionally, this will demonstrate what's possible when expanding a 2-mix out to 12 channels (following a similar concept to Dolby Atmos layout of 7.1.4). A discussion of the current streaming platforms available, along with a demonstration of the SonyRA360 audio format.

• Dolby Atmos Music Sound Aesthetics

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

Dolby Atmos Music is becoming increasingly popular as a 3D music format thanks to various sales channels such as Blu-ray and streaming. Even though most consumers today listen to music using headphones, the majority of productions are mixed in certified studios on a speaker setup like 9.1.6. Such a large speaker setup offers many possibilities to achieve impressive results. However, most musicians are not yet familiar with it. The sound design of popular music in stereo has evolved for over half a century. It is therefore obvious to use creative ideas from stereophonic music productions and transmit them in a modified form into 3D audio. Lasse Nipkow will show how psychoacoustic phenomena from stereo can be used for Dolby Atmos.

• Mastering of Music for Immersive Audio Formats

Presenters: **George Massenburg**
Darcy Proper
Michael Romanowski,
Wilfried Van Baelen, Auro Technologies

After recording and mixing, the mastering process is another key element in the chain of music production for many decades. Since its existence, the mastering process was only in channel-based formats but with the recent addition of object-based technology, new ways of working are required. This workshop will highlight the experiences of multiple awarded Grammy winner mastering engineers who will give their approach of the mastering process in the most popular Immersive Sound formats for music production.

• New Creative Possibilities for Music Production in Immersive Sound

Moderator: **Rory S. Kaplan**, Rory Kaplan Productions

Presenters: *Chuck Ainlay*
Niko Bolas
Frank Filipetti
Rory S. Kaplan, Rory Kaplan Productions
George Massenburg
Elliot Scheiner
Al Schmitt

This session will be available very soon! Grammy winning engineers/producers discuss Immersive Audio mixing for music applications, including discrete and/or object based mixing. Deliveries, Mastering, Fold Down Mix Issues. Their experiences in balancing a musical mix, and how they approach utilizing height channels. Also, the benefits of 9.1 and 11.1 mixing. In addition discussing setting standards in Immersive Audio and setup for music application. They will share their thought on mastering and what they have been researching, as to getting the best results for fold down which is used in many cases.

• Immersive Music Production—Mixing Strategies and Observations
Wednesday, October 28, 2020
Session Time: 1:00 pm – 2:00 pm

Presenter: **Jeff Stone**, Artisyns Audio LLC

Immersive music has the potential introduce a new aesthetic to the consumer and open avenues of new expression to artists, but before that can happen engineers and producers will have to become fluent in this new aesthetic to satisfy the demands of the artist, consumer, and music industry. This workshop will focus on the production workflow, aesthetic considerations, and production techniques I used on one recent project that leverage traditional tools and methods in pursuit of the immersive aesthetic.

• Goodbye Stereo Part II
Friday, October 30, 2020
Session Time: 10:00 am – 11:00 am

Presenters: **Hyunkook Lee**, University of Huddersfield
Thomas Lund, Genelec OY

Technology has long been available to record and reproduce music in more perceptually satisfactory ways than stereo. With "immersive," excellent performances can be preserved more sentiently, also for future generations to enjoy. As "stereo" is becoming a playback option, we will discuss new recording techniques, discrimination between immersion and envelopment, and how the ideals of music reproduction may differ from those of film and drama. The workshop includes audio examples, so, for the best listening experience, please join wearing good around-ear headphones. AKG K240 are recommended for some of the examples.

• Roundtable Discussion: Immersive Music
Friday, October 30, 2020
Session Time: 2:30 pm – 3:30 pm

Moderator: **Michael Romanowski**, Coast Mastering

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NETWORKED AUDIO

• Future-Proof, Scalable Standards to Meet the Future of Networked Audio

Moderator: **Richard Bugg**, Meyer Sound

Presenters: *Florian Frick*,Neutrik
Henning Kaltheuner, d&b audiotechnik
Genio Kronauer, L-Acoustics

Today, the AV industry is starting to understand there cannot be a one-size fits all approach for applications; As network systems become the dominant structure for signals and control, there is a need for future-proof, user-friendly, and scalable standards. In the live production market, requirements for reliability, ease of use, and scalability have become increasingly critical, and manufacturers, production companies, end-users, system designers/integrators have not always found compatibility with networked devices. This presentation hosted by Milan founding members and end-users will provide an overview of Milan, key characteristics and features—with special attention paid to the benefits for all Pro AV stakeholders who wish to adopt a resilient, future-proof solution for the long term future of this industry.

• Interoperability Standards for IP Media Networking
Tuesday, October 27, 2020
Session Time: 1:30 pm – 2:30 pm

Presenters: **Claudio Becker-Foss**, DirectOut GmbH
Andreas Hildebrand, ALC NetworX
Terry Holton, AIMS (Alliance for IP Media Solutions)
Nicolas Sturmel, Merging Technologies

Two significant standards have emerged in the past several years to provide wide-ranging interoperability for professional media networking: AES67 and SMPTE ST 2110. This session will review the background and objectives behind the creation of each of these standards and explain the relationship between the two standards. Recent developments and the future roadmap for both of these important standards will also be explored. Terry Holton will begin the session with a brief overview of the relationship between AES67 and SMPTE ST 2110-30. He will be followed by Andreas Hildebrand, technology evangelist at ALC NetworX, who will review the status of relevant standards development work for AES67, SMPTE ST 2110 and ST 2059, and IEEE 1588-2019, as well as important work being done by related organizations. The next segment of the session, titled "AES67 on the WAN," will be presented by Nicolas Sturmel, who is senior technologist for Merging Technologies. Sturmel will discuss WANs, LANS, and their similarities and differences, and then address questions such as "Why would I need AES67 on the WAN?" and "What performance should I expect?" The final part of the presentation will look at a specific case study documenting the use of AES67 over WAN with standard DSL internet links. This segment, titled "Lockdown Rock — The Show Must Go WAN," will be presented by Claudio Becker-Foss, CTO/CEO at DirectOut GmbH.

• Roundtable Discussion: Networked Audio
Thursday, October 29, 2020
Session Time: 2:30 pm – 3:30 pm

Moderator: **Bob Lee**, Audio Engineering Society

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• The AES70 Revolution in System Control and Monitoring
Friday, October 30, 2020
Session Time: 10:30 am – 12:00 noon

Presenters: **Jeff Berryman**
Ethan Wetzell, Bosch Communication Systems

Presenter: Though long in development, AES70 is one of the newer AES standards. Our presenters will describe and demonstrate its usefulness for operating equipment from multiple manufacturers together on a common data network. This session will examine how AES70 can address different use cases as well as how AES70 is uniquely able to solve some common control challenges.

PAPER SESSIONS: AUDIO APPLICATIONS AND TECHNOLOGIES

• **Reverberation Slope Ratio Thesis**—*Michael Fay*, GraceNote Design Studio, San Diego, CA, USA

The Reverberation Slope Ratio (symbolically $T_{60}SR_6$) is a proposed standard for condensing six octaves (63 Hz–2 kHz) of reverberant decay data into a singular-quotient, qualitative score, for indoor performance, worship and entertainment facilities. It's a defining metric for scoring and grading the proportional relationship (i.e., ratio) between the longest and shortest of six reverberation (T_{60}) values, measured or predicted, and applied to fully-enclosed venues employing sound reinforcement systems. In practice, Bass Ratio (BR) and Slope Ratio (SR) goals are conflicting concepts. BR goals and calculations were developed to support the idea that acoustic instruments need a little extra assistance, via longer reverberation time, in the low-frequency range. SR goals and calculations support the notion that those same low frequencies do not require extra reverberation time, but rather need to be well contained. Longer low and very low-frequency reverberation is not needed, nor desirable, when an extended-range sound reinforcement system is used. The $T_{60}SR_6$ thesis is offered to advance and define a room's acoustic design objectives, and provide a simple numeric scoring scale and grading vocabulary, from which acoustical design specifications can be initiated and/or evaluated.
Convention Paper 10388

• **A Sonification Algorithm for Subjective Classification of Food Samples**—*Hsein Kew, Ryan Stables*, Birmingham City University, Birmingham, UK

Spectroscopic food analysis has been much studied over many years and is still an ongoing topic of research. In this paper we propose a spectroscopic diagnostic method to generate an audio output in order to discriminate between two classes of data, based on the features of spectral datasets. To do this, we first perform spectral pre-processing and extract appropriate features from the spectra, and then apply different selection criteria to narrow down the number of features selected. To optimize the process, we compare three selection criteria, as applied to two spectroscopy food datasets in order to evaluate the performance of sonification as a method for discriminating data. Lastly, the salient features are mapped to the parameters of a frequency modulation (FM) synthesizer, so as to generate audio samples. The results indicate that the model is able to provide relevant auditory information, and most importantly, allows users to discriminate consistently between two classes of spectral data. This approach using the sonification of spectroscopic data has been shown to be useful to a food analyst as a new method of investigating the purity of food.

Convention Paper 10393

• **Sonification of Spectroscopic Analysis of Food Data Using FM Synthesis**— *Hsein Kew, Ryan Stables*, Birmingham City University, Birmingham, UK

Food safety is a global concern, and with the rise of automation, novel methods of categorizing, sorting, and discriminating food types are being explored. These techniques require a reliable method for rapidly identifying food sources. In this paper we propose a method of spectroscopic food analysis, whereby audio data is generated from spectra to allow users to discriminate between two classes of a given food type. To do this, we developed a system which first extracts features and applies dimensionality reduction, then maps them to the parameters of a synthesizer. To optimize the process, we compare Amplitude Modulation (AM) and Frequency Modulation (FM) synthesis, as applied to two real-life datasets to evaluate the performance of sonification as a method for discriminating data. The results indicate that the model is able to provide relevant auditory information, and most importantly, allows users to consistently discriminate between two classes of spectral data. This provides a complementary tool to supplement current food detection methods.
Convention Paper 10394

• **Audience Effect on the Response of a Loudspeaker System in the Low Frequency Range, Part 1: Magnitude**
—*Thomas Mouterde*,[1] *Etienne Corteel*,[1] *Manuel Melon*[2]
[1] L-Acoustics, Marcoussis, France
[2] Laboratoire d'Acoustique de l'Université du Mans, Le Mans, France

The response of a loudspeaker system is affected by the presence of the audience. However, the loudspeaker system tuning is performed without an audience, applying equalization filters and delays for time alignment system components. The validity of these decisions with an audience is of primary importance. In this paper the magnitude response of a loudspeaker system is simulated at low frequencies using Finite Element Method over a flat listening area for multiple source heights and audience densities. The results show that the audience modifies notches due to the floor reflection for a flown source and creates a build-up associated with a low-pass behavior for ground-stacked sources. The implications on typical loudspeaker system configurations are presented and discussed.
Convention Paper 10398

• **A Study of Vibration Isolation for Floor Standing Loudspeakers**—*Bob Katz*, Digital Domain, Inc., Altamonte Springs, FL, USA

Sound is vibration. It can be the desired vibration of a musical instrument passing to the air and then to the listener's ears. It can be the wanted vibration of a loudspeaker transducer. It can be the unwanted vibration of a loudspeaker cabinet passing to the floor or supporting surface, which then vibrates in sympathy, producing interfering and unwanted additional signal paths. These phenomena can be measured and assessed including the potential psychoacoustic impact of the additional signal paths. In this study the author measured the sound and vibration produced by a floor-standing loudspeaker. Vibration induced into surfaces and sound produced in the air, with the loudspeaker mounted on either damped isolators or on solid wood blocks between the speaker cabinet and the floor. Analysis of the data reveals: • Vibration isolation under speakers produces measurable differences compared to wood blocks: more than 15 ms in reverberation time in some frequency ranges; low level artifacts in the waterfall; up to a dB in distortion in some ranges; up to a dB in frequency response. • The type of shell construction affects isolation performance—specifically solid barriers vs. limp mass. • In general, isolation produces an improvement in performance and potential audible benefits, minding the shell construction of the room.
Convention Paper 10405

• **Forensic Interpretation and Processing of User-Generated Audio Recordings**—*Robert C. Maher*, Montana State University, Bozeman, MY, USA

For audio forensic analysis, it is increasingly likely that multiple user-generated recordings (UGRs) may be presented as evidence in a criminal investigation. Audio evidence may come from handheld smartphones, private surveillance systems, police body cameras, and other unsynchronized recording devices. When multiple UGRs are available, the recordings could provide spatial and temporal information about the location and orientation of sound sources, and potentially a means to increase intelligibility of spoken utterances. However, UGRs generally start and stop at different times, differ in technical format specifications, and seldom have sufficiently reliable time stamp information for exact time and position synchronization. We study these analytical and practical constraints, and develop forensic recommendations for combining and synchronizing multiple UGRs.
Convention Paper 10419

• **Energy Limiter for Control of Diaphragm Displacement and Port Velocity**—*Pascal M. Brunet, Glenn S. Kubota, Yuan Li*, Samsung Research America, Valencia, CA, USA

In the audio industry, use of limiters is common practice to increase reproduction level while avoiding overload and ensuring system protection. We describe in this paper a novel energy-based limiter. Using a physical model, the total energy stored in a loudspeaker is controlled to keep the peak displacement within prescribed limits. This technique provides a smooth and precise control of the diaphragm maximum displacement, without change of the spectral balance. It will be presented first for a closed-box system and then extended to a bass-reflex system, to limit the air velocity in the port and avoid onset of port noise.
Convention Paper 10421

• **Audio in the New Millennium – Redux**—*James Moorer*, JAMMINPOWER.COM, Panacea, FL, USA

In 2000 I was invited to present the Heyser lecture, entitled "Audio in the New Millennium" [1]. As it was the turn of the Millennium, I chose to review the progress of digital audio over the preceding 20 years of my own career, then extrapolate forward to 2020. In the process of doing so, one of the trends identified in my lecture stood out as being of fundamental importance. This was that innovations in all the aspects of modern computer technology—computation, networking, disc storage—had come to be driven almost entirely by the entertainment industry—digital audio, digital video, and computer games. The situation of 1980 when audio and video fed on the scraps of technology developed for military and industrial uses has now been reversed: military, industrial, and scientific computing is now driven entirely by the needs for entertainment technology. This paper will review the prognostications of the

2000 lecture and take stock of the progress, noting specifically which predictions hit the mark (a few of them) and which did not (most of them). More strikingly, a number of trends and developments in entertainment that were entirely unanticipated in 2000 have moved to center stage as drivers of technology.
Convention paper 10438

- **Augmented Reality Tempo Control Tool for Conducting Students—***Diana Estefania Hernandez Martinez, Agnieszka Roginska*, New York University, New York, NY, USA

In this paper we discuss the implementation of a new tool that allows conducting students to practice in an augmented reality environment. The tool combines different hardware and software to provide a new tool with which beginner conducting students can practice their skills without the need for an ensemble or a stage. The tool is meant to be used as a practice tool to complement conducting lessons. It features an augmented reality headset, machine learning software, and MIDI sampling to provide a flexible environment for conducting. It is a novel addition to the conducting resources because it takes advantage of the latest technology to give not only audio but also visual feedback to the user.
Convention Paper 10444

PAPER SESSIONS: AUDIO CONTENT MANAGEMENT

- **Drum Sample Retrieval from Mixed Audio via a Joint Embedding Space of Mixed and Single Audio Samples—** *Wonil Kim, Juhan Nam*, Korea Advanced Institute of Science and Technology (KAIST), Daejeon, Korea

Sample-based music creation has become a mainstream practice. One of the key tasks in the creative process is searching desired samples in large collections. However, most commercial packages describe the samples using metadata, which is limited to explain subtle nuances in timbre and style. Inspired by music producers who often find instrument samples with a reference song, we propose a query-by-example scheme that takes mixed audio as a query and retrieves single audio samples. Our method is based on deep metric learning where a neural network is trained to locate single audio and their mixtures closely in the embedding space. We show that our model successfully retrieves single audio samples given mixed audio query in various evaluation scenarios.
Convention Paper 10390

PAPER SESSIONS: AUDIO EQUIPMENT

- **Low-Frequency Performance of a Woofer-Driven Flat-Panel Loudspeaker (Part 1: Numerical System Analysis and Small Signal Measurement)—***Benjamin Zenker, Robert Schurmann, Sebastian Merchel, M. Ercan Altinsoy*, Dresden University of Technology, Dresden, Germany

The low-frequency performance of loudspeakers is essential to the listener preference rating. To achieve the same low-frequency performance as conventional speakers, an optimized construction of a woofer-driven flat-panel loudspeaker was designed. The woofer radiates in a small air gap between the panel and the distance plate and excites the panel uniformly. At low frequencies, significant improvements compared to an exciter and a comparable performance to a conventional woofer system could be reached. This paper presents a numerical system analysis of the paneled woofer design, which provides a detailed understanding of the system's behavior. Furthermore, the radiation mechanism is compared to that of an exciter. All simulation models are comprehensively validated with the corresponding experimental data.
Convention Paper 10391

- **Cylinder Measurement Method for Directivity Balloons—***Maxime Bru*, NEXO-SA, Plaily, France

There are several ways to obtain the set of frequency responses all around a loudspeaker, the so-called *directivity balloon* which provides useful knowledge about the loudspeaker radiation behavior. In this paper, a novel method for collecting far-field directivity balloon datasets is presented. This alternative method builds on traditional polar measurements and proved to be practical both in terms of measurement time and equipment set-up. By cleverly combining datasets and by using far-field wave propagation, datasets in cylindrical coordinates are created and then projected on a sphere. Eventually, using a spherical interpolation, the directivity balloon is obtained.
Convention Paper 10400

- **Mapping of Theory of Reverse Recovery Losses to the Duty Cycle Distribution of Audio Signals with a Double Pulse Tester —***Andreas S. Petersen,*[1] *Niels E. Iversen,*[2] *Arnold Knott*[1]
[1] Technical University of Denmark, Lyngby, Denmark
[2] ICEpower, Søborg, Denmark;

Switch-mode power amplifiers have become the conventional choice for audio amplification due to their very high efficiency and negligible THD+N performance. This paper proposes a new method to determine the reverse recovery loss theoretically. The Spice model from the manufacture is simulated with a Double pulse tester in LTspice. Mapping the reverse recovery behavior for different currents and time periods allows it to be fitted to a model. The fitted model is used to estimate the reverse recovery loss in a Class-D audio amplifier power stage. The conventional model shows a 16 times larger loss compared to the fitted model for a worst case comparison.
Convention Paper 10407

- **Crossover Design Based on Median Level and Phase Correction within a Listening Window—***Victor Manuel Catalá Iborra*, DAS Audio Group, Fuente del Jarro, Spain

PA and sound reinforcement loudspeaker systems consist, many times, of two or more frequency bands reproduced by horn loaded transducers with phase plugs in front of the drivers and horn loaded compression drivers. The size, design, and arrangement of these elements inside the cabinet are restricted not only by acoustical reasons but by mechanical constraints, such as weight, gravity center, and the total available volume of the cabinet. Besides, the lack of directivity match at crossover frequencies is usual. Due to its own architecture, it is sometimes difficult or impossible to find a good reference point for crossover design that provides proper crossover and symmetric radiation around this point. In these cases, a crossover design based only on the on-axis responses cannot be optimum in regards to uniformity of coverage. A method based on the alignment at different angles within a listening window to get a representative median level and phase correction, and posterior phase optimization

based on maximum average level at crossover frequencies is presented. For certain designs, this method can provide better radiation pattern at crossover, better average level response within the intended listening window, and smoother directivity transitions between ways.
Convention Paper 10410

- **A Review of the Plasma Loudspeaker (Ionic Loudspeaker): Principles of Operation, General Model and Parts**—*Maria Carolina Rodriguez*, Pontificia Universidad Javariana, Bototá, Colombia; Center of Interdisciplinary Research in Music, Media, and Technology (CIRMMT) Montreal, Quebec Canada; McGill University, Montreal, Quebec, Canada

This is a review on the operation of the plasma/ionic loudspeaker. Although not commercially or functionally viable yet, the exploration of the plasma loudspeaker might bring the field of sound reproduction closer to a change in paradigm. In this paper the different discharge methods and regimes to generate the plasma discharge are described, the parts of some functional loudspeaker models are recounted, and the advantages and disadvantages of them are identified. Plasma loudspeakers do not have some of the drawbacks present in traditional loudspeakers, as they are massless systems thus no inertia, this results in transparent transient responses and more uniform frequency responses.
Convention Paper 10411

- **Balanced Electrostatic All-Silicon MEMS Speakers**—*Hermann A. G. Schenk,*[1] *Lutz Ehrig,*[1] *Franziska Wall,*[2] *Bert Kaiser,*[2] *Michael Stoltz,*[2] *Sergiu Langa,*[2,3] *David Schuffenhauer,*[2] *Jorge Mario Monsalve Guaracao,*[2] *Anton Melnikov,*[2] *Holger Conrad*[2]

[1] Arioso Systems GmbH, Dresden, Germany
[2] Fraunhofer Institute for Photonic Microsystems, Dresden, Germany
[3] Brandenburg University of Technology Cottbus-Senftenberg, Cottbus, Germany

MEMS based audio reproduction systems have recently become a vibrant field of research and development. Benchmarks for size, power consumption, and cost efficiency are very demanding. Electrostatic MEMS actuators offer a high potential to meet both commercial and technical targets. Macroscopic electrostatic headsets are among the best that audio markets offer today. Using MEMS technology, power consumption, and voltage requirements are reduced to the needs of portable devices. In this contribution we report for the first time a balanced MEMS µSpeaker, using two antagonistic electrostatic forces, improving electromechanical stability and harmonic distortion. The distinction to state-of-the-art MEMS push-pull actuators and the acoustic performance is presented.
Convention Paper 10414

- **Spatial Timbre Uniformity Analysis of Loudspeakers in Rooms Using MFCC**—*Takashi Minagawa,*[1] *Kazuhiko Kawahara,*[1] *Ryo Nakaie*[2]

[1] Kyushu University, Fukuoka, Japan
[2] TOA Corporation, Hyogo, Japan

Spatial timbre uniformity needs to be considered during public address system design using loudspeakers. In this paper an analysis based on Mel-frequency cepstrum coefficient (MFCC) was conducted on room impulse responses produced by different loudspeakers including conventional cone loudspeakers and a Distributed Mode Loudspeaker (DML). Histograms of Euclidean distances between MFCC vectors of each impulse response was plotted for each loudspeaker. The tail of the histogram that represents the frequency of larger distances was lighter in the histogram of the DML. The results imply that the DML can avoid extreme spatial timbre dissimilarity. The proposed method introduces concept of timbre uniformity, which could be helpful for designing acoustic spaces.
Convention Paper 10415

- **Degradation Effects of Water Immersion on Earbud Audio Quality**—*Scott Beveridge,*[1] *Steffen A. Herff,*[2,3] *Estefanía Cano*[1]

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[2] Ècole Polytechnique Fédérale de Lausanne, Lausanne, Switzerland
[3] Western Sydney University, Sydney, NSW, Australia

Earbuds are subjected to constant use and scenarios that may degrade sound quality. Indeed, a common fate of earbuds is being forgotten in pockets and faced with a laundry cycle (LC). Manufacturers' accounts of the extent to which LCs affect earbud sound quality are vague at best, leaving users to their own devices in assessing the damage caused. This paper offers a systematic, empirical approach to measure the effects of laundering earbuds on sound quality. Three earbud pairs were subjected to LCs spaced 24 hours apart. After each LC, a professional microphone as well as a mid-market smartphone were used to record i) a test tone ii) a frequency sweep and iii) a music signal played through the earbuds. We deployed mixed effects models and found significant degradation in terms of RMS noise loudness, Total Harmonic Distortion (THD), as well as measures of change in the frequency responses of the earbuds. All transducers showed degradation already after the first cycle, and no transducers produced a measurable signal after the sixth LC. The degradation effects were detectable in both, the professional microphone as well as the smartphone recordings. We hope that the present work is a first step in establishing a practical and ecologically valid method for everyday users to assess the degree of degradation in their personal earbuds.
Convention Paper 10418

- **Acoustic Mass and Resistance as Function of Drive Level for Straight, Bent, and Flared Loudspeaker Ports**—*Andri Bezzola, Glenn S. Kubota, Allan O. Devantier*, Samsung Research America, Vanencia, CA, USA

Nonlinear control of bass-reflex loudspeakers requires accurate knowledge of acoustic mass and acoustic resistance of the port. The values of these parameters are hard to measure and previous research indicates that they are both functions of turbulence levels in the port airflow. Well-designed ports are known to accept higher drive levels before flow separation and vortex shedding causes unwanted port noise. This work investigates the relationship between dependence of port variables and port noise as functions of drive level. Measurements show that port acoustic mass is essentially a constant independent of drive level, and that port acoustic resistance is related to drive level much in the same way port compression and port noise are. Absence of proper flaring of ports and introduction of bends in ports can both exacerbate the increase of port acoustic resistance with drive level, along with port compression and port noise. A properly flared port—even if it is bent—performs better than a straight unflared port. Straight and optimally flared ports have val-

ues of port acoustic mass and resistance that change less than 2 dB from lowest to highest drive levels.
Convention paper 10423

• **Reluctance Force Compensation for the Nonlinear Control of a Loudspeaker**—*James F. Lazar, Pascal M. Brunet*, Samsung Research America, Valencia, CA, USA

The motor structure of a conventional electrodynamic loudspeaker driver relies on a Lorentz force to move the loudspeaker diaphragm and generate sound. The reluctance force is an undesired consequence of this very motion, causing the displacement of the diaphragm to be other than that intended based on the Lorentz force acting alone. In other words, the reluctance force generates distortion in the audio output. Reluctance force compensation provides a method for controlling the motion of the diaphragm, taking into account the excursion dependent variation of the voice-coil inductance that generates the reluctance force. Based on a target diaphragm displacement, a nonlinear control algorithm utilizing reluctance force compensation calculates a corrected drive signal to drive the loudspeaker, producing a more ideal trajectory of the diaphragm, and reducing distortion in the audio output.
Convention Paper 10425

• **Omnidirectional Sound Sources Based on Multiple Transducers Design Method**—*Bartlomiej Chojnacki, Tadeusz Kamisinski, Artur Flach*, AGH University of Science and Technology, Cracow, Poland

The current state of the art provides many ways of delivering the omnidirectional sound source for acoustic modeling. They are complicated and unreachable for most scientists to recreate, which is a common problem, while constructing optimal, high-quality omnidirectional sound source created with multiple transducers placed uniformly on the sphere is still unsolved and not described in detail. This paper provides a quick review of existing omnidirectional sound source ideas and compares them with conventional, multi-speakers design. Some principles on traditional speaker design concepts are presented, like the dependency between the number of transducers or their size on overall source performance. A novel approach to the complex and detailed omnidirectional sound source design process will be discussed.
Convention paper 10426

• **In-Room Low-Frequency Sound Power Optimization Using Near Field Response**—*Adrian Celestinos, Ritesh Banka, Pascal Brunet*, Samsung Research America, Valencia, CA, USA

The total sound power (TSP) produced by a loudspeaker can be severely affected when placed in typical living rooms. General approaches consider an equalization filter designed toward a desired target. The computation of the TSP requires measurements on a number of microphones spaced in the room. In this work an automatic method to estimate the TSP without the use of numerous measurements is proposed. The proposed method includes a static microphone configured to measure the near-field sound pressure of the driver, and a controller to determine its velocity to automatically adjust the sound power levels to an acoustic environment. Results in typical living rooms show an average standard deviation error of 2.6 dB on the estimation of the TSP.
Convention paper 10429

• **Boundary Element Subdomain Modeling for Electroacoustics** —*Joerg Panzer*, R&D Team, Salgen, Germany

The boundary element method (BEM) is a flexible tool for modeling interior and exterior acoustics. However one of the prerequisites for this mathematical procedure to work is that the domain forms a proper volume. For example a sheet or a narrow slit would pose a problem. This so-called thin shape break-down effect can be over-come. The technique described here divides the acoustical domain into subdomains. Each subdomain should then be suitable for the boundary element method. The subdomain technique proofs to be robust and efficient. This paper demonstrates the subdomain technique for modeling of a typical vented loudspeaker enclosure.
Convention paper 10433

• **Low-Frequency Performance of a Woofer-Driven Flat-Panel Loudspeaker (Part 2: Numerical System Optimization and Large Signal Analysis)**—*Benjammin Zenker, Martin Heinl, Sebastian Merchel, M. Ercan Altinsoy*, Dresden University of Technology, Dresden, Germany

The paneled woofer design is a new concept to improve the low-frequency behavior of flat-panel loudspeakers. This paper focuses on the numerical optimization of the transducer resources, the panel properties, and the system construction. The low-frequency optimization is based on an electro-mechanical network model, by which the modal depending air compliance is introduced. However, various parameters are modified and validated with measurements. Furthermore, the maximum sound pressure level is considered and the limiting distortions are differentiated from the panel and the woofer itself. The results are compared with the large-signal behavior of an exciter mounted on the same panel. Summarized, additional advice for the optimization of the frequency response is given and examples for unsuitable constructions are presented.
Convention Paper 10443

PAPER SESSIONS: AUDIO PROCESSING

• **Overlapping Acoustic Event Detection via Perceptually Inspired the Holistic-based Representation Method**—*Hyeonsik Choi,[1] Keunsang Lee,[1] Minseok Keum,[2] David Han,[3] Hanseok Ko[4]*
[1] LG Electronics, Seoul, South Korea
[2] SELVAS AI, Seoul, Korea
[3] Army Research Laboratory, Adelphi, MD, USA
[4] Korea University, Seoul, Korea

A novel dictionary learning approach that utilizes Mel-scale frequency warping in detecting overlapped acoustic events is proposed. The study explored several dictionary learning schemes for improved performance of overlapping acoustic event detection. The structure of NMF for calculating gains of each event was utilized for including in overlapped signal for its low computational load. In this paper we propose a method of frequenca warping for better sound representation and apply dictionary learning by a holistic-based representation, namely nonnegative K-SVD (NK-SVD) in order to resolve a basis sharing problem raised by part-based representations. By using Mel-scale in a dictionary learning, we show that the information carried by low frequency components more than high frequency components and dealt with a low complexity.

Also, the proposed holistic-based representation method avoids the permutation problem between another acoustic events. Based on these benefits, we confirm that the proposed method of Mel-scale with NK-SVD delivered significantly better results than the conventional methods.
Convention paper 10395

- **Exploring Quality and Generalizability in Parameterized Neural Audio Effects**—*William Mitchell, Scott H. Hawley*, Belmont University, Nashville, TN, USA

This work expands on prior research published [1] on modeling nonlinear time-dependent signal processing effects by means of a deep neural network with parameterized controls, with the goal of producing commercially viable, high quality audio, i.e., 44.1 kHz sampling rate at 16-bit resolution. These results highlight progress in modeling these effects through architecture and optimization changes, towards increasing computational efficiency, lowering signal-to-noise ratio, and extending to a larger variety of nonlinear audio effects. Most of the presented methods provide marginal or no increase in output accuracy over the original model, with the exception of dataset manipulation. We found that limiting the audio content of the dataset provided a significant improvement in model accuracy over models trained on more general datasets.
Convention paper 10397

- **Automatic Classification of Live and Studio Audio Recordings**—*Tejas Manjunath, Jeet Pawani, Alexander Lerch*, Georgia Institute of Technology, Atlanta, GA, USA

We present a study on the automatic classification of live and studio audio recordings, an important meta-information for music catalogue browsing and music recommendation systems. Several possible input representations (MFCCs, Mel spectrograms, VGGish) are combined with the classifiers GMM, SVM, and CNN to identify the most powerful approach. The results show that a CNN with VGGish input clearly outperforms other approaches and that its detection accuracy is high enough to be useful in practical applications.
Convention Paper 10399

- **Short-Range Rendering of Virtual Sources for Multichannel Loudspeaker Setups**—*Juhani Paasonen, Ville Pulkki*, Aalto University, Espoo, Finland

A method for rendering virtual sources is proposed where the sources are perceived to be closer than the loudspeaker array radius. The development is based on an informal finding about closer perception of the range of virtual sources rendered coherently over multiple loudspeakers. To avoid sound quality issues inherent with coherent rendering, the input is split to two streams, one with more transients and the other with smoother temporal envelope. The transient stream is rendered over coherent reproduction, and the continuous stream is processed with time-frequency-domain spreading technique. The results from localization tests with moving sources show that the proposed method produces the perception of closer distances on both sweet-spot and off-sweet-spot listening.
Convention Paper 10401

- **Sparse Audio Inpainting: A Dictionary Learning Technique to Improve Its Performance**—*Georg Tauböck, Shristi Rajbamshi, Peter Balazs*, Austrian Academy of Sciences, Vienna, Austria

The objective of audio inpainting is to fill a gap in a signal, either to be meaningful or even to reconstruct the original signal. We propose a novel approach applying sparse modeling in the time-frequency (TF) domain. In particular, we develop a dictionary learning technique which deforms a given Gabor frame such that the sparsity of the analysis coefficients of the resulting frame is maximized. A suitable modification of the SParse Audio Inpainter (SPAIN) allows to exploit the obtained sparsity gain and, hence, to benefit from the learned dictionary. Our experiments demonstrate that our methods outperform several state-of-the-art audio inpainting techniques in terms of signal-to-noise ratio (SNR) and objective difference grade (ODG).
Convention Paper 10402

- **Low Latency Timbre Interpolation and Warping Using Autoencoding Neural Networks**—*Joseph T. Colonel,*[1] *Sam Keene*[2]
[1] Queen Mary University of London, London UK
[2] The Cooper Union, New York, NY, USA

A lightweight algorithm for low latency timbre interpolation of two input audio streams using an autoencoding neural network is presented. Short-time Fourier transform magnitude frames of each audio stream are encoded, and a new interpolated representation is created within the autoencoder's latent space. This new representation is passed to the decoder, which outputs a spectrogram. An initial phase estimation for the new spectrogram is calculated using the original phase of the two audio streams. Inversion to the time domain is done using a Griffin-Lim iteration. A method for avoiding pops between processed batches is discussed. An open source implementation in Python is made available.
Convention paper 10406

- **Revisiting Harmonic Change Detection**—*Pedro Ramoneda,*[1,2] *Gilberto Bernardes*[2]
[1] University of Zaragoza, Zaragoza, Portugal
[2] University of Porto, Porto, Portugal

In this paper we advance an enhanced method for computing Harte et al.'s (2006) Harmonic Change Detection Function (HCDF), which aims to detect harmonic transitions in musical audio signals. Each of the HCDF component blocks is revisited in light of recent advances in harmonic description and transformation. To evaluate our proposal, we compute an exhaustive grid search to compare the multiple proposed algorithms and a large set of parameterizations across four large style-specific musical datasets. Our results show that the newly proposed methods and parameter optimization improve the detection of harmonic changes by 5.57% (f-score) with respect to previous methods. Furthermore, while guaranteeing recall values at >99%, our other method improves precision by 6.28%.
Convention Paper 10408

- **Individual Listening Zone with Frequency-Dependent Trim of Measured Impulse Responses**—*Michele Ebri,*[1] *Nicolò Strozzi,*[2] *Filippo Maria Fazi,*[3] *Angelo Farina,*[1] *Luca Cattani*[2]
[1] University of Parma, Parma, Italy
[2] ASK Industries Spa, Reggio Emilia, Italy
[3] University of Southampton, Southapton, UK

Acoustic Contrast Control (ACC) has been widely used to achieve individual audio delivery in shared environments. The effectiveness of this method is reduced when the

control is performed in reverberant environments. Even if control filters are computed using measured transfer functions, the robustness of the system is affected by the presence of reverberation in the plant matrix. In this paper a new optimization method is presented to improve the ACC algorithm by applying a frequency-dependent windowing of the measured impulse response used for the filter computation, thus removing late reflections. The effects of this impulse response optimization are presented by means of sound zoning results obtained from experimental measurements performed in a car cabin.
Convention paper 10409

- **Estimating Nonlinear Impulse Response Length Using Time-Delayed Mutual Information—***Ethan R. Hoerr, Robert C. Maher*, Montana State University, Bozeman, MT, USA

Using impulse responses measured from various audio systems has become common in audio signal processing. However, determining the length of an impulse response captured from a nonlinear system becomes problematic, as these systems violate the linearity principle from signals and systems theory. A "lumpy" or "spiky" tail in the measured impulse response of a nonlinear system is a telltale symptom of this issue. In this in-progress work, we investigate the use of time-delayed mutual information (TDMI), a concept from the field of information theory, to identify the useful portion of a recovered nonlinear impulse response. Initial test systems include linear, time-invariant FIR and IIR filters, and IIR filters with a static nonlinear distortion added.
Convention Paper 10416

- **Dialog Enhancement via Spatio-Level Filtering and Classification—***Aaron S. Master,*[1] *Lie Lu,*[1] *Heidi-Maria Lehtonen,*[2] *Harald Mundt,*[3] *Heiko Purnhagen,*[2] *Daniel P. Darcy*[1]
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[3] Dolby Germany GmbH, Nürberg, Germany

Dialog enhancement (DE) is a feature that allows a listener to increase the level of dialog in a content item relative to backgrounds. DE is "unguided" if only the finished mix is available, meaning that a DE system must estimate the dialog. Spatio-Level Filtering (SLF) is a source separation technology that, when combined with dialog classification, allows for high-quality unguided DE for typical entertainment content in a stereo or higher channel count format. SLF exploits spatial and level information and requires little lookahead, memory, computation, and training data. To evaluate results, we conduct two subjective listening experiments which indicate favorable performance.
Convention Paper 10427

- **Design of Audio Processing Systems with Autogenerated User Interfaces for System-on-Chip Field Programmable Gate Arrays—***Trevor Vannoy,*[1,2] *Dylan Wickham,*[2] *Dustin J. Sobrero,*[2] *Connor Dack,*[2] *Ross K. Snider,*[1,2] *Tyler B. Davis*[2]
[1] Montana State University, Bozeman, MT, USA
[2] AudioLogic Inc., Bozeman, MT, USA

System-on-Chip (SoC) Field Programmable Gate Arrays (FPGAs) are well-suited for real time audio processing because of their high performance and low latency. However, interacting with FPGAs at runtime is complex and difficult to implement, which limits their adoption in real-world applications. We present an open source software stack that makes creating interactive audio processing systems on SoC FPGAs easier. The software stack contains a web app with an autogenerated graphical user interface, a proxy server, a deployment manager, and device drivers. An example design comprising custom audio hardware, a delay and sum beamformer, an amplifier, filters, and noise suppression is presented to demonstrate our software. This example design provides a reference that other developers can use to create high performance interactive designs that leverage the processing power of FPGAs.
Convention Paper 10428

- **Applying Signal Decorrelation to a Guitar Modulation Effect—***Alessandro Terenzi, Valeria Bruschi, Stefano Nobili, Stefania Cecchi*, Università Politecnica delle Marche, Ancona, Italy

In this paper a new algorithm for a guitar modulation effect for a stereophonic reproduction is presented and tested. The algorithm is based on a signal decorrelation with its application on a vibrato effect. The decorrelation is achieved through an all-pass biquad IIR filters, while the vibrato effect consists of a periodic variation of the signal time delay. A real-time implementation of the system has been realized and several tests have been performed. An objective and subjective analysis of the algorithm is performed to verify the feasibility of the proposed approach.
Convention Paper 10430

- **Effects of Software Tuning Programs on Vocal Recordings—***Chandler Raymond Bridges, Jr.*, Indiana University, Bloomington, IN, USA

The primary purpose of this study was to investigate the effect of tuning software on pitch correction for a vocal example: a professional tenor singing the melody from the chorus section of a popular song. Three commonly used pitch-shifting tuning programs were used to create experimental stimuli. All tuned examples were created with the programs' automatic tuning settings. The examples were analyzed with Praat software to determine the amount of pitch change and the remaining distance to the desired target pitch for each programs' resulting output. One program provided more aggressively tuned results and had closest condition to the desired target pitch; however, there is much debate over which software program is preferred.
Convention paper 10436

- **Rapid Approximation of Weakly-Nonlinear Mode Parameters—***Travis Skare, Jonathan Abel*, CCRMA, Stanford University, Stanford, CA, USA

Modal processors and synthesizers require determining resonant frequencies and decay rates of a system. We introduce a simple, efficient, and flexible approach to compute these parameters when speed is of the essence, for example modeling a user's sample folder in a realtime mobile synthesizer application. The approach compares DFT peaks in two impulse response windows at different points in time to obtain the modal frequencies and decay rates. The approach trades precision for speed, and should be considered an approximation. For a selection of samples across tonal/atonal instruments and reverb responses, we consider sensitivity to window choice and window size, and compare results to traditional modeling methods, which operate on the entire impulse response. We propose extensions of the algorithm to more than two windows, and to capture non-

linear, time-varying behavior such as the pitch dive after striking a tom drum at high velocity.
Convention Paper 10437

- **A Fexible Numerical Optimization Approach to the Design of Biquad Flter Cascades—***Peter Dobbs*, Facebook Reality Labs Research, Redmond, WA, USA

The design of recursive filters with specified magnitude and phase responses is a common problem in audio signal processing. This paper presents a flexible methodology for the design of digital filter cascades with specified magnitude and phase responses based on a non-linear least squares fitting methodology. The proposed methodology utilizes the Levenberg-Marquardt algorithm to derive optimal parameters in the z-plane for a given number of second order recursive filters using a cost function that can accommodate arbitrary frequency-weighting curves in addition to any hard or soft constraints on the filter behavior. The cost function being minimized is based on the L2 distance in the complex frequency domain and incorporates additional terms to impose constraints on filter shape, stability, and distortions.
Convention paper 10439

- **Perceptually-Informed Features for Room Reverberation Modeling and Prediction—***Kimberly Kawczinski, Jonathan S. Abel*, Center for Computer Research in Music and Acoustics, Stanford University, Stanford, CA, USA

A room reverberation model based on normalized echo density and bandwise energy decay is presented. The model produces psychoacoustically transparent and statistically independent impulse response copies by sampling the parameters at various rates. These parameters are further considered as a feature set for prediction within an acoustic space and tested using a standard data set with three measured spaces. The numerical error introduced in the prediction procedure is evaluated, revealing space-dependent trends.
Convention Paper 10446

PAPER SESSIONS: IMMERSIVE AUDIO

- **Are You There? : A Literature Review of Presence for Immersive Music Reproduction—***Jack Kelly, Wieslaw Woszczyk, Richard King*, McGill University, Montreal, Quebec, Canada, The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

Immersive audio purportedly enhances the listener's sensation of presence within virtual experiences. However, providing a concrete definition of "presence" in relation to immersive media can be difficult. In this paper a multidisciplinary review is given of various conceptualizations of presence, including physical, social, spatial, and environmental. Literature pertaining to factors that influence the sensation of presence in immersive music production is presented. Aspects of reverberation that are hypothesized to contribute to the sensation of presence are explored in depth. Key characteristics of the sound-source and the listener's perception are highlighted. A discussion of related perceptual attributes (immersion, realism, and naturalness) is included in an effort to disentangle them from presence. An overview is presented covering methods used to measure presence, focusing on techniques that show applicability to immersive music research. Future work is discussed.
Convention paper 10389

- **Efficient HRTF Representation Using Compact Mode HRTFs—***Jacob Hollebon, Filippo Maria Fazi*, University of Southampton, Southampton, UK

This paper proposes a new Head-Related Transfer Function (HRTF) representation, which utilizes a rotation to reorder the energy of the HRTF in the spherical harmonic domain into a minimal number of spherical harmonics and then discards those with low energy content. This new representation is titled the compact mode HRTF. The rigid sphere and Neumann KU100 HRTFs are considered to a truncation order of N = 30 which requires 961 spherical harmonics. The KU100 compact mode HRTF using only 178 spherical harmonics is shown to match the full KU100 HRTF to within just a 5% error bound. Thus the compact mode scheme is proposed as an efficient method for representing, transmitting and utilizing HRTFs to high orders.
Convention paper 10404

- **A Formula for Low-Frequency Interaural Level Difference—***Rahulram Sridhar, Edgar Y. Choueiri*, Princeton University, Princeton, NJ, USA

A formula for low-frequency interaural level difference (LF-ILD) as a function of source distance and direction is derived from rigid-sphere head-related transfer function (RS-HRTF) theory. Since ILD at low frequencies (typically $f < 700$Hz) is a dominant cue for near-field distance perception in the free-field, a simple formula akin to the Woodworth formula for high-frequency interaural time difference, may be beneficial to easily and efficiently implement distance- and direction-dependent LF-ILDs for real-time spatial audio applications on devices with limited computational resources. By evaluating the limit as $f \to 0$ of the infinite series representation of the RS-HRTF for finite source distances, an exact, closed-form expression for the "DC gain" as a function of source distance and direction is derived. For a given source location, using this expression to compute the DC gain at the left and right "ears," and then taking the ratio of the two quantities gives the desired LF-ILD. As an example, it is shown that the derived formula may be used to extrapolate far-field ILD spectra of a rigid sphere to the near-field exactly for low frequencies and with sufficiently high accuracy for higher frequencies. Furthermore, the derived formula, like the Woodworth formula, is well-suited to individualization.
Convention paper 10412

- **A Framework for Shared Sound Reproduction Across a Connected TV and Soundbar for Sound Expansion—** *Liam Kelly, Yoonjae Lee, Jongwoo Kim, Hackwant Park*, Samsung Electronics, Gyeonggi-do, Korea

Connected accessory audio devices such as soundbars are often used in substitution of built in TV audio because of their superior hardware. In this paper a framework is introduced in which the audio hardware of the two devices can be utilized simultaneously in TVs where amenable hardware is available. In such a configuration audio tasks are additionally assigned to top firing TV loudspeakers allowing for vertical sound expansion. An audio rendering algorithm is utilized in which additional audio channels for high quality sound expansion are synthesized based on input and analysis of content. Listening tests reveal a preference for the

expanded sound compared with a soundbar only reference.
Convention Paper 10420

- **Comparison of Methods to Control Early Reflections for Acoustic Enhancement Systems**—*Dai Hashimoto, Takayuki Watanabe, Hideo Miyazaki*, Yamaha Corporation, Shizuoka, Japan

  Acoustic enhancement systems have gained popularity in recent years and are broadly used in various types of facilities because of their acoustic naturalness and system stability. To effectively enhance several acoustic parameters of a room, general early reflections and reverberation are separately controlled in acoustic enhancement systems. In particular, early reflections play a vital role in determining the spatial impressions of the listeners. This study was performed to investigate an effective method for controlling early reflections. Herein, three methods for controlling early reflections using finite impulse response filters based on acoustic measurement results in an auditorium were constructed in the same reproduction space. The characteristics of these systems were assessed via acoustic measurements and subjective evaluation experiments.
  Convention Paper 10422

- **The Sense of Auditory Presence in a Choir for Virtual Reality**—*Louise Bryce,*[1] *Mark Sandler,*[1] *Lars Koreska Andersen,*[2] *Ali Adjorlu,*[2] *Stefania Serafin*[2]
  [1] Queen Mary University of London, London, UK
  [2] Aalborg University, Copenhagen, Denmark

  This paper investigates which auditory parameters influence the sense of presence in an immersive environment of children with low functioning autism (LFA) and social anxiety. The auditory parameters investigated were Spatialized, Authoritarian (e.g., the ability to hear a conductor), and Ambisonic. A 360-degree video of the Danish National Children's choir shown in the VE enables the participants to sing and dance together with them, while simultaneously being exposed to the different parameters. We discuss how the different audio parameters affect the plausibility of the environment to children. While asking questions in relation to presence, the four participating children were having difficulties grasping the meaning, even when adjusted to their preference and level of understanding. However, the observational data yielded a change in their behavior and participation with the virtual choir, while differentiating with the changing parameters. This could indicate that presence through auditory stimuli can be observed through the behavioral patterns of children with LFA and social anxiety.
  Convention Paper 10440

- **Click-to-Music Ratio: Using Active Headphones to Increase the Gap**—*Leonard Menon*, University of Lethbridge, Lethbridge, Alberta, Canada

  The volume of a clicktrack relative to the music playback through a monitoring system is an artifact that may damage the performance and the performers' aural health. This user study compared two monitoring methods across four recording situations with five performer participants. The first method served as a control test using traditional headphones. The second introduced Active Headphones (AH) that binauralized the signals captured from microphones fastened around the head, mixed with a stationary clicktrack. Participant feedback unanimously revealed that the AH design contributed positively to the overall recording experience. I also found an overall average difference of 18.42 LU between the clicktrack and the music playback when using the AH, vs. 7.84 LU with the controlled method.
  Convention Paper 10441

- **Exploratory Research into the Suitability of Various 3D Input Devices for an Immersive Mixing Task. Part II**—*Diego Quiroz, Denis Martin*, McGill University, Montreal, Quebec, Canada, Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada

  This study evaluates the suitability of one 2D (mouse and fader), and three 3D (LeapMotion, SpaceMouse, Novint Falcon) devices for an immersive mixing task. A test, in which subjects were asked to pan a monophonic sound object (probe) to the location of a pink noise burst (target), was conducted in a custom 3D loudspeaker array. The objectives were to determine how quickly and accurately the subjects were able to perform the task using each device, which was most appropriate, and which was most preferred overall. Results show significant differences in response time between devices. Furthermore, it was found that block order had a significant influence over the subject's response time, as well as trial number, pointing towards a familiarization effect.
  Convention Paper 10442

- **Comparison of Spatialization Techniques with Different Music Genres**—*Shashank Aswathanarayana*, University of California, Santa Barbara, CA, USA

  Sound spatialization has interested artists and researchers alike for a number of decades. While the researchers have focused on the ability to effectively spatialize sound in different loudspeaker configurations as well as bring in an immersive environment into headphone listening, artists have been focusing on trying to create various compositions and installations, which utilize these loudspeaker configurations to create interesting effects and pieces. This study aims to understand if there are any preferences or biases for certain spatialization techniques with certain genres of music and what the underlying reason could be for any such preferences. This is done with a user study in a listening room over loudspeakers. Results showed that subjects preferred the Vector Base Amplitude Panning (VBAP) configuration for all genres except Rock music where they preferred Ambisonics method of delivery.
  Convention Paper 10447

PAPER SESSIONS: PERCEPTION

- **Non-Classical, *Bounded* Fechnerian Integration for Loudness: Contrary to Luce and Edwards, Initial Loudness-Difference-Size Stipulations Are Only Recouped for Linear Loudness Growth**—*Lance Nizami*, Independent Research Scholar, Palo Alto CA, USA

  A major question in sensory science is how a sensation of magnitude F (such as loudness) depends upon a sensory stimulus of physical intensity I (such as a sound-pressure-wave of a particular root-mean-square sound-pressure-level). An empirical just-noticeable sensation difference $(\Delta F)_j$ at F_j specifies a just-noticeable intensity difference $(\Delta I)_j$ at I_j. Intensity differences accumulate from a stimulus-detection threshold I_{th} up to a desired intensity I. Likewise, the corresponding sensation differences are classically presumed to accumulate, accumulating up to $F(I)$ from

$F(I_{th})$, a non-zero sensation (as suggested by hearing studies) at I_{th}. Consequently, sensation growth $F(I)$ can be obtained through Fechnerian integration. Therein, empirically-based relations for the Weber Fraction, $\Delta I/I$, are individually combined with either Fechner's Law $\Delta F = B$ or Ekman's Law $(\Delta F/F) = g$; the number of cumulated steps in I is equated to the number of cumulated steps in F, and an infinite series ensues, whose higher-order terms are ignored. Likewise classically ignored are the integration bounds I_{th} and $F(I_{th})$. Here, we deny orthodoxy by including those bounds, allowing hypothetical sensation-growth equations for which the differential-relations $\Delta F(I) = F(I + \Delta I) - F(I)$ or $(\Delta F(I)/F(I)) = (F(I + \Delta I) - F(I))/F(I)$ do indeed return either B or g, for *linear* growth of sensation F with intensity I. Also, 24 sensation-growth equations $F(I)$, which had already been derived by the author likewise using bounded Fechnerian integration (12 equations for the Weber Fraction ($\Delta I/I$), each combined with either Fechner's Law or with Ekman's Law), are scrutinized for whether their differential-relations return either B or g respectively, particularly in the limits $(\Delta I/I) << 1$ and the even-more-extreme limit $(\Delta I/I) \rightarrow 0$, both of which seem unexplored in the literature. Finally, some relevant claims made by Luce and Edwards (1958) are examined under bounded Fechnerian integration: namely, that three popular forms of the Weber Fraction, when combined with Fechner's Law, produce sensation-growth equations that subsequently return the selfsame Fechner's Law. Luce and Edwards (1958) prove to be wrong.
Convention Paper 10387

- **Identification of Virtual Receiver Array Geometries that Minimize Audibility of Numerical Dispersion in Binaural Auralizations of Finite Difference Time Domain Simulations**—*Julie Meyer*,[1] *Tapio Lokki*,[1] *Jens Ahrens*[2]
[1] Aalto University, Espoo, Finland
[2] Chalmers University of Technology, Gothenburg, Sweden

This paper presents a perceptual evaluation of numerical dispersion in free-field headphone-based head-tracked binaural auralizations of finite difference time domain (FDTD) simulations. The simulated pressure, captured by virtual volumetric receiver arrays, is used to perform a spherical harmonics decomposition of the sound field and generate binaural signals. These binaural signals are compared perceptually to dispersion error-free binaural signals in a listening experiment designed using a duo-trio paradigm. The aim of the present work is to identify the size and density of the receiver array minimizing the audibility of numerical dispersion in the generated binaural signals. The spherical harmonics order was chosen to be 12 for the spatial decomposition. The overall reconstruction error, defined as the absolute value of the difference between the dispersion error-free and FDTD-simulated left-ear magnitude spectrum, was used as an objective metric to measure the spectral differences between the two signals. The listening experiment results show that this error does not correlate with the discrimination rates of the subjects. These results therefore suggest that this error does not suffice to describe the perceptual aspects introduced by numerical dispersion in the free-field dynamic binaural auralizations presented in the listening experiment. The results also show that increasing the receiver density for a fixed array size does not necessarily render numerical dispersion inaudible in the auralizations. Five out of 27 volumetric arrays led to FDTD-simulated binaural auralizations indistinguishable from the dispersion error-free binaural auralizations.
Convention Paper 10392

- **A Flexible Software Tool for Perceptual Evaluation of Audio Material and VR Environments**—*Stefan Gorzynski*,[1,2] *Neofytos Kaplanis*,[1] *Søren Bech*[1]
[1] Bang & Olufsen, Struer, Denmark
[2] University of Huddersfield, Huddersfield, UK;

Audio evaluation makes use of listening tests that allow the presentation of audio stimuli and gather data on their influence on human perception. Due to the rapid technological advancement and application-specific protocols, such work typically requires the experimenter to develop custom software that suits the experimental design and/or incorporates technological advancements, i.e., spatial audio. This article describes a publicly available listening test suite (BO-LTS) that allows standardized and sensory analysis evaluation protocols to be built with a high degree of flexibility, while supporting multichannel audio processing and the use of Virtual Reality (VR) environments.
Convention Paper 10396

- **I'm All Ears: What Do Untrained Listeners Perceive in a Raw Mix versus a Refined Mix?**—*Kelsey Taylor*, University of Lethbridge, Lethbridge, Alberta, Canada

Creative professionals spend time learning rules just to then break them. The standards established in the audio industry function under the collective understanding that there is not a single "correct" way to produce music, but rather there exists a general consensus of what the "wrong" way may be. To explore whether average music listeners hear details that audio engineers notice, I conducted semi-directed interviews with ten untrained listeners who compared a '"aw mix" and an industry standard "refined mix" of two different songs. Results show that these untrained listeners do have a certain level of understanding of what they hear in a mix; the main difference compared with trained listeners is the vocabulary that they use to describe sound.
Convention Paper 10403

- **Native High-Resolution versus Red Book Standard Audio: A Perceptual Discrimination Survey**—*Mark Waldrep*, California State University, Dominguez Hills, Carson, CA, USA; AIX Media Group/AIX Records, Los Angeles, CA, USA

The perceptibility of high-resolution versus CD standard audio has been the subject of research and debate since the introduction of hi-res audio distribution formats 20 years ago. The author conducted a large survey to determine whether experienced listeners could differentiate between a diverse set of 20 native high-resolution PCM stereo recordings and downconversions of the same masters at 44.1 kHz/16-bits fidelity. Participants were encouraged to audition the files using their own systems, which ranged from modest, headphone-based personal setups to audiophile quality rooms costing in excess of $50,000 to professional studio environments. They were not allowed to use analytical tools or other non-listening means to assist in their observations. Over 400 responses were received from professional audio engineers, experienced audiophiles, casual music enthusiasts, and novices aged 11 to 81 years. The online survey submissions show that high-resolution audio was undetectable by a substantial majority of the respondents regardless of experience level, equipment cost, or process with almost 25% choosing "No

Choice." However, some evidence exists that specific genres and recordings produced moderately higher positives.
Convention Paper 10417

- **Listener-Perspective-Dependent Variation of Interaural Cross-Correlation Coefficient in a Reverberant Space—** *Bogdan Ioan Bacila Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

  This paper presents an extensive set of IACC measurements using BRIRs obtained at 16 different positions in a reverberant concert hall, each with 100 rotation angles. The results show the variation of IACC relative to the listening positions and head rotations, for both early and late reflections. It was found that the variation of $IACC_{E3}$ (< 80ms) depending on the position had some consistent patterns with the relative head orientation to the sound source. However, it was observed that, near the side wall, the variation of $IACC_{E3}$ showed a higher consistency with the head rotation relative to the direction of the wall. The measurements of $IACC_{L3}$ (> 80ms) revealed that values recorded outside the central axis of the room were influenced neither by the head rotation nor the position in the room.
  Convention Paper 10435

- **Using ITU-R BS.1770 to Measure the Loudness of Music versus Dialog-Based Content—***Scott G. Norcross*, Dobly Laboratories San Francisco, CA, USA

  ITU-R BS.1770 was developed for "typical" broadcast program material and was created using that material. While BS.1770 has been shown to be effective at predicting the loudness of "typical" broadcast material, questions around its use for music content has surfaced. More specifically, how music content should be normalized with respect to broadcast content that is speech based. A loudness matching test similar to the one used in the development of ITU-R BS.1770 was carried out, where subjects were tasked with matching audio sequences that were exclusively music based to a speech-based reference signal. Results show that subjects on average tend to match music content 3–4 dB higher than the speech-based reference signal when measured similar in loudness by ITU-R BS.1770.
  Convention Paper 10448

PAPER SESSIONS: RECORDING PRODUCTION, REPRODUCTION

- **Single-Channel Sound Power Estimation for Reverberation Effects—***Samuel D. Bellows, Timothy W. Leishman*, Brigham young University, Provo, UT, USA

  The reverberant field of a room is driven by the sound power of a radiating source (i.e., the sound radiating in all directions). For audio applications, the signal driving reverberation effects is typically based on a microphone positioned somewhere that does not actually capture the sound power of the source and its associated spectrum. This paper describes how one can find an optimal microphone placement position such that the mean-squared pressure spectrum best matches the shape of the sound-power spectrum of the source. The technique is based on measured directivity functions. The results will help recording and sound-reinforcement engineers produce more realistic reverberation effects.
  Convention Paper 10413

- **The Reality of the Loudness War in Japan—The Case Study on Japanese Popular Music—***Kazuma Watanabe,*[1] *Kawahara Kazuhiko,*[1] *Hiriko Nishida,*[1] *Kosuke Okusa*[2]
  [1] Kyushu University, Fukuoka, Japan
  [2] Yokihama City University, Yokohama, Japan

  This paper focused on the loudness statistics of the songs, especially commercially successful albums of popular music in Japan during the years 1989 to 2018. The purpose of this paper was to verify the existence of the "loudness war" in Japan. The statistical results showed there was a leap of loudness in the middle 1990s, which implies the existence of the loudness war was on in Japan. Moreover, the authors employed the time series analysis and estimated the average loudness from 2019 to 2028. The authors proposed the loudness estimation by using the ARIMA model. The model estimated the future average loudness would rise to –4 LUFS, in case there were no regulations in place.
  Convention paper 10424

- **Audio Education: Proposed Course Project and Process-Based Assessment for Learning Remote Recording Workflows—***Doug Bielmeier*, Northeastern University, Boston, MA, USA

  Aspiring recording engineers need to train in situated learning environments to develop technical skills and the ability to communicate effectively with clients and co-workers, according to previous audio education research. However, establishing situated learning environments that reflect emerging audio production industry workflows that incorporate solely digital and remote production modalities can be challenging in traditional classrooms. Thus, there is a need to develop authentic learning environments, projects, and assessments for classrooms. This paper outlines a specific outcome-based audio production and engineering course project and a process-based assessment for educators teaching an audio production course. The course project balances technical learning objectives while incorporating the acquisition of communication skills in remote production workflows. The development of authentic learning environments, projects, and assessments in audio education is paramount for program and institutional efficacy and relevance to the increasingly remote audio production industry.
  Convention Paper 10431

- **Composing, Recording and Producing with Historical Equipment and Instrument Models—***Clive Mead*, University of Plymouth, Plymouth, UK

  This paper describes how models were created and employed (as accurately as possible, using available resources), to simulate the recording technology and instruments, available at different points in recorded music history, initially with two models, based on 1955 and 1965. A series of explorative experiments in composing, performing, and recording music were conducted with these models in an effort to understand how the strengths and inherent limitations of the tools available affect both the composition and production process, and also the stylistic identity of the music created. How do influences from different musical genres familiar to the composer inspire and inform their creative decisions? How much of this influence is dictated by the limitations in the instruments and equipment used?
  Convention Paper 10432

- **The Effects of the Depth and Width of an Ambience Microphone Array on Perceived Depth, Width and Listener Envelopment—***Hyunkook Lee*, University of

Huddersfield, Huddersfield, UK

This paper presents a subjective study that investigated the effects of the depth and width dimensions of a 4-channel ambience microphone array on perceived environmental depth (ED), environmental width (EW), and listener envelopment (LEV). Test stimuli were reverberant recordings made using four backward-facing cardioid microphones with varying depth and width dimensions (0.5m, 1m, and 2m). Listening tests conducted using four loudspeakers ±30° and ±120° found that the array depth had no significant effect on either ED, EW or LEV. The array width effect was significant on all of EW, ED and LEV (1m = 2m > 0.5m), with ED significantly dependent on the type of sound source. From this, a new microphone array for 3D ambience recording is proposed.
Convention Paper 10434

• **ODESSA: An Interdisciplinary Symphonic Recording for the Study of Orchestral Sound Blending**—*Martha de Francisco,*[1,2] *Malte Kob,*[3] *Jack Kelly,*[1,2] *Diego Quiroz,*[1,2] *Stefanos Ioannou*[3]
[1] McGill University, Montreal, Canada
[2] Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Canada
[3] Detmold Univesity of Music, Detmold, Germany

The Analysis, Creation, and Teaching of Orchestration (ACTOR) Partnership brings together a varied group of experts to realize interdisciplinary research on topics surrounding musical timbre and orchestration. In support of ACTOR research activities, recording engineers/producers (Tonmeisters) offer their expertise in the recording of complex acoustic music. For the ODESSA (Orchestral Distribution Effects in Sound, Space and Acoustics) Project and complemented with concurrent acoustic research activities, the Tonmeisters realize an exploratory multitrack recording of a symphony orchestra, capturing sound components from different perspectives and mixing them with the intention of producing a convincing sonic representation of the music. The mixes as well as the disassembled components of the sound are later presented to the ACTOR researchers for evaluation, enabling them to further examine aspects of timbral blending in symphonic sound.
Convention Paper 10445

PRODUCT DEVELOPMENT

• Convolution—A Practical Review

Presenter: **Ricardo Losada**, MathWorks

This tutorial session focuses on practical aspects of convolution and FIR filtering for audio research and audio product development, from algorithm exploration through to real-time signal processing. We discuss the tradeoffs between time and frequency domain implementations, including latency and computational cost. For both domains we review the best-known implementation variants and some of the hardware technologies most used for performance acceleration. Throughout the session, we use practical MATLAB examples mostly drawn from spatial audio applications. You should expect a mix of introductory and advanced topics—some signal processing experience will be beneficial, but it is not required.

• Evolution of Audio Products—Demystifying Innovation

Friday, October 30, 2020
Session Time: 12:00 noon – 1:00 pm

Moderator: **Lisa Ferrante-Walsh**, iZotope, Inc.

Presenters: *Emma Azelborn*, iZotope, Inc.
Katrina Lui, Descript
Elpiniki Pappa, Native Instruments

As innovative products and features come to market, do you find yourself wondering about the genesis of these novel ideas and how they evolved into marketable solutions? What process do companies go through to provide novel solutions to existing problems, or to problems that have no solution? How are these ideas iterated on until a viable solution emerges that provides the most customer value? What is the role of the user during product development? How do companies embrace risk and learn from failure while innovating? This 60-minute panel will leave you with concrete and actionable steps to apply in your organization. The panel consists of technologists from companies that employ these techniques and will be followed by Q&A.

• Workshop on Computational Platforms for Automotive Audio

Friday, October 30, 2020
Session Time: 4:30 pm– 5:30 pm

Moderator: **Dilip Warrier**, Bose Corporation

Presenters: *Paul Beckmann*, DSP Concepts
Daniel L. Bernal, Arm, Inc.
Samir K. Gupta, Qualcomm
John Redford, Analog Devices Inc

This presentation discusses the features that are common to all kinds of processing, and ones that are unique to audio processing. Its most distinctive characteristic is its need for determinism, which is made more difficult by a lot of standard computer hardware techniques, such as caching and virtual memory.

RECORDING AND PRODUCTION

• Building World Class Audio Production Facilities in Emerging Global Communities

Moderators: **Sergio Molho**, Walters-Storyk Design Group
John Storyk, Walters-Storyk Design Group

Panelists: *Mateo De Los Rios*, Universidad ICESI
Marc Viadiu, WSDG, LLC
Mills Xu, 55TEC STUDIOS

This panel will explore the unique and challenging aspects of planning, designing, and constructing state-of-the art recording studios internationally. The "human" aspect of this complex journey is critical. Familiarity with local laws and regulations and insights into qualified contractors and systems integrators are invaluable assets. Challenges include console and equipment acquisition and maintaining positive relationships with local authorities. Studio assignments ranging from Doha, Qatar to Beijing and Mongolia, China, Mexico City, Rio de Janeiro, Brazil, Cali, Colombia, Santa Domingo, Dominican Republic and Zurich, Switzerland have provided these panelists with keen insights into these complex and challenging missions.

• Chamber Reverb Challenge

Presenter: Alex U. Case, University of Massachusetts Lowell

Challenging times call for inspired solutions. Out-of-the-box thinking suggests out-of-the-plug-in reverb. If your studio has a bathroom, it should have a Chamber Reverb. Reverberation doesn't have to come from a digital algorithm, a concert hall, or a cathedral. Any sound reflective space—a bathroom, a basement, a garage—is a good candidate for generating acoustic resonance. Alex Case

reviews the history, sound, science, technology, and art of setting up your own chamber, with ease. Engineers at all levels will welcome this basic advice on selecting the space, setting up easily available gear, and fine-tuning your sound. The result: a reverb uniquely yours. Pandemic constraints can produce epic effects. Accept the challenge and create your own chamber reverb.

• Elvis Is Back in the Building

Moderator: **Jim Kaiser**, Belmont University

Panelists: *Tony Castle*
Andy Childs
Ed Seay

This talented production team was chosen for the task of returning to the roots of Elvis' music, and with the help of skilled musicians and modern techniques, put a new "twist" on his classic recordings in "Where No One Stands Alone." The audience will be treated to the original 1960s Elvis songs and the "new version" focusing on the original and refined Elvis vocals. Engineers Ed Seay and Tony Castle, along with producer Andy Childs will discuss what it took to transform them, the obstacles encountered, and the process of extending songs using alternate takes that no one has heard before.

• Remote Human Mastering vs The Algorithm

Moderator: **Andres Mayo**, Andres Mayo Mastering & Audio Post

Presenters: *David Gould*
Shachar Gilad, SoundBetter
Michael Romanowski, Coast Mastering

The lock down only reinforced what mastering engineers have been experimenting for the last decade: mastering is now done remotely in over 90% of the cases worldwide. Even though it has many advantages, remote mastering presents a really tough challenge because the market is completely different and the required skill set is much larger. Besides, engineers are now strongly competing against algorithms. In this panel, two experienced mastering engineers and the founder of SoundBetter will discuss all these topics in depth.

• Secrets of Near-Field Monitoring

Moderator: **Michael Rodriguez**, Nowcast Network

Presenter: **Carl Tatz**, Carl Tatz Design LLC

"Secrets Of Near-Field Monitoring"—Insight into the world of professional monitoring, finding unique solutions to real-world problems that are not ordinarily discussed. Michael Rodriguez moderates an in-depth conversation with award-winning studio designer and monitoring expert Carl Tatz. Topics to be discussed include monitor positioning using the Null Positioning Ensemble protocol; the Allison Effect (What no one talks about —and should); calculating the listening position relative to room modes; subwoofers; backstage peak at the Phantom Focus tuning protocols.

• The Best Hall for Beethoven is in Japan

Presenters: **Akira Fukada**
Ulrike K. Schwarz, Anderson Audio New York

Conductor Mariss Jansons declared that only in Suntory Hall, Tokyo, a modern Beethoven symphonic cycle could be recorded. The team of Bayerische Rundfunk spent three years recording the cycle at Herkulessaal, Munich, and then flew the Bavarian Radio Symphony Orchestra and Bavarian Radio Choir to Tokyo to rerecord the complete cycle in four nights at Suntory Hall. Engineers Ulrike Schwarz and Akira Fukada will explore the the two cycles regarding hall layout, recording technique and inter cultural aspects. Excerpts from the BR Klassik production and the BRKlassik-NHK co-production will be played. This presentation is in honor of Beethoven's 250th anniversary celebration.

• Vinyl Mastering Demonstration

Presenters: **Jim Kaiser**, Belmont University
Margaret Luthar, Welcome To 1979

Maggie Luthar (along with assistant mastering and cutting engineer, Anna Clark), will give a basic overview of the Neumann VMS-70 lathe, as well as demonstrate cutting into a lacquer master. Topics covered will include: overview of the basic functions, buttons, and features of the Neumann lathe, what a "lacquer master" is, best practices for delivering files to your mastering engineer, and more. The demonstration will be technical, but conversational so everyone—even those with little to no experience mastering for vinyl —can learn!

• The Evolution of the Project / Home Studio— Today's "e-Studio"
Tuesday, October 27, 2020
Session Time: 4:30 pm – 5:30 pm

Moderators: **Joshua Morris**, WSDG Walters-Storyk Design Group
John Storyk, WSDG Walters-Storyk Design Group

Panelists: *Pete D. Hofmann*, milo music ltd t/as Miloco Studios / Miloco Builds
Sam Inglis
Robert Margouleff, Independent Producer/ Engineer/Technology Maven
PK Pandey, AVN/SYS engineering
Clay Sheff, Smash Studios

The project or home studio has been a familiar term for nearly 50 years but what does it mean? The argument can be made that personal studios have always been project intensive. However, a continually evolving generation of smaller, less expensive and more powerful gear, as well as a constantly expanding universe of applied acoustic techniques and materials have totally democratized the field and will continue to do so. Certainly the events of the past year have presented social conditions unlike anything we have previously experienced in our lifetime. Every facet of Broadcast TV, Cable, and Online studio development has been affected, including audio studio design and installation. This panel will explore design, technology, social and legal aspects of this seemingly never-ending discussion, and will introduce the possibility of an entirely new studio format, we are calling the "e-studio"—at the moment a "working title" for the future.

• Post-Covid Recording Challenges
Tuesday, October 27, 2020
Session Time: 6:00 pm – 7:00 pm

Moderators: **Peter H. Doell**, Audio Engineering Society
Warren Huart, Produce Like A Pro

Panelists: *JJ Blair*
Alan Meyerson
Piper Payne
Cameron Webb

We will discuss whether Online Interaction, Live Streaming, and/ or Apps, can truly replace all or any part of the record making process. Reduced budgets restricting travel mean we've been making music remotely for years—now, however, the ability to get together has been completely removed for most people. Some studio owners, producers, engineers, and mixers say the Pandemic has helped them streamline their process,. What are the pros and cons that you've experienced?

• Platinum Latin Engineers & Producers
Tuesday, October 27, 2020
Session Time: 7:00 pm – 8:00 pm

Moderator: **Andres Mayo** ,Andres Mayo Mastering & Audio Post

Presenters: *Francisco J. Botero,* Francisco Botero/ Studio G Brooklyn
Tweety Gonzalez
Sacha Trijueque, SOGA Records

Since 2010, the Platinum Latin Panel brings together some of the most renowned Producers and Engineers from Latin America. This Panel will be no exception, with a roster of 5 super talented, Grammy-winning professionals who will share their secrets and open up their mixing sessions . This event will be held in Spanish.

• Behind the Mix
Wednesday, October 28, 2020
Session Time: 10:30 am – 11:30 am

Presenters: **Peter H. Doell**, Audio Engineering Society
Vance Powell

Vance Powell is a six-time Grammy award winning producer, engineer and mixer, whose impressive list of credits includes Chris Stapleton, Stray Cats, Elle King, The Raconteurs, The Dead Weather, The White Stripes, Arctic Monkeys, and Wolfmother among many others. Powell recently won two Grammy awards in 2018 for mixing and engineering Chris Stapleton's "From A Room: Volume 1" (Best Country Album), and mixing and engineering various songs on CeCe Winans' "Let Them Fall In Love" (Best Gospel Album). He was also nominated for mixing Robert Randolph & The Family Band's "Got Soul" (Best Contemporary Blues Album). Vance will detail some of his most successful mixing and recording techniques with participation from the audience.

• The Day School Came Home
Wednesday, October 28, 2020
Session Time: 4:30 pm – 5:30 pm

Moderator: **Ulrike K. Schwarz**, Anderson Audio New York

Presenters: *Jim Anderson*, New York University
Alex U. Case, University of Massachusetts Lowell
Agnieszka Roginska, New York University
Nick Sansano, New York University

In this workshop educators Jim Anderson, Alex Case, and Agnieszka Roginska will give a hands-on report on how their concepts of teaching audio theory, mixing, music performance, critical listening to immersive audio in class rooms changed to an online university approach within days in March 2020. Stories of teaching over different time zones with students scattered around the world (students in Australia, Korea and Europe being taught from a dining room in Brooklyn, NY) will be told and why going outside or a haircut are overrated. Instruments include twitch.tv, audiomovers.com, binaural audio encoding and how to choose the right Zoom license to avoid unwanted guests.

• It's All About The Vocal
Wednesday, October 28, 2020
Session Time: 6:00 pm – 7:00 pm

Moderator: **Terri Winston**, Women's Audio Mission

Panelists: *Natalia Ramirez*, Petite Boss LLC
Simone Torres

Hear from platinum vocal producers that have crafted the vocal tracks on some of the biggest hit records, across multiple genres. This panel provides an overview of the vocal production process and explores everything from tracking techniques, microphones and recording chains used, to editing, processing and headphone mixes and everything in between. Hear from multiple-Grammy winning, multi-platinum vocal producers Natalia Ramirez (Camila Cabello, Jennifer Lopez, Ricky Martin) and Simone Torres (Normani, Cardi B, Sia, Dua Lipa).

• Are We in Sync? Audio for Video in Classical Music Production
Thursday, October 29, 2020
Session Time: 1:30 pm – 2:30 pm

Presenters: **David Fros**, Metropolitan Opera
John Kerswell, Metropolitan Opera
Alex S. Kosiorek, Arizona PBS & Audio Engineering Society
Benjamin A. Maas, Fifth Circle Audio
Mark Schubin, Metropolitan Opera
Roberto Toledo, Jr. The New World Symphony

Whether for television, movie theater, outdoor projection, on-demand or live video streaming, audio is always there to accompany it. For classical music, each type of production requires special coordination with video technicians, performance production staff, and the performers themselves. Now more than ever, audio engineers are performing dual roles in audio and video production—understanding the importance of artistic temperament while addressing technical considerations for a successful event. We will discuss these and other aspects of media production as we explore a few case studies including those of Camerata Pacifica, the Metropolitan Opera, and The New World Symphony among others.

• Encore Replay of Goodbye Stereo (from Virtual Vienna)
Thursday, October 29, 2020
Session Time: 3:00 pm – 4:00 pm

Presenters: **Hyunkook Lee**, University of Huddersfield
Thomas Lund, Genelec OY

Technology has long been available to record and reproduce music in more perceptually satisfactory ways than stereo. Formats known as "surround" was a first step, but with "immersive," excellent music performances can be preserved more sentiently, also for future generations to enjoy. As "stereo" is fast becoming just a playback option, we will discuss immersive recording techniques, and which mixing principles to put behind us, for instance level panning.

• The Producers: From Melodies to Masters
Thursday, October 29, 2020
Session Time: 4:30 pm – 5:30 pm

Moderator: **Peter H. Doell**, Audio Engineering Society

Panelists: *Eric Boulanger*, The Bakery
Jack Douglas
CJ Vanston
Shelly Yakus

This panel will trace the progress of producing a record all the way from the writing of a song, it's pre-production, basic recording and "sweetening," through the mixing and the mastering. Rather than focusing on "the Guy Wearing the Producer's Hat," our panel discussion will pay special attention to the skill sets necessary for each link in the chain, and how technology has dramatically blurred these elements—and how a growing number of people are mastering the Art of Recording to produce their own material.

• Old Tracks, New Tricks: Classic Studios in the 21st Century
Friday, October 30, 2020
Session Time: 7:00 pm – 8:00 pm

Moderator: **Mark Rubel**, The Blackbird Academy

Panelists: *Ken Caillat*, Sausalito Record Plant
Daniel Crockett, Sun Studio
Rodney Hall, FAME
Teresa Knox, Leon Russell's The Church
Zach Ozburn, Sun Studio
Debbie Wilson, Muscle Shoals Sound

Classic studios are carrying their sonic legacies into the 21st Century, while repurposing and diversifying. Representatives from such historic facilities as Sun Studio, FAME, Muscle Shoals Sound, The Sausalito Record Plant, and Leon Russell's The Church Studio will discuss their methods, challenges, opportunities and strategies.

SOUND REINFORCEMENT

• 3D Audio for Live Events

Presenters: **Phil Kamp,** KLANG:technologies GmbH
Hugo Larin, FLUX:: Immersive
Scott A. Sugden, L-Acoustics
Ralf Zuleeg

An emerging trend in the live sound industry is the rapid adaption of 3D sound technology. 3D sound is conceived to better connect and immerse the audience in the audio and or visual experience. Increasing the sweet spot for the live audience and giving the artist onstage a greater pallet of sonic tools to create and engage with the patrons. The change to 3D audio in live, broadcast, and stage monitoring will present challenges in mix and system designs for both creative and objective reasons. This workshop seeks to provide good advice on how to move into the world of 3D audio from experts who have been a part of the transition.

• An Evolutionary View on Audio and Control Distribution for Live Sound Reinforcement

Presenter: **Etienne Corteel**, L-Acoustics

From analog to the adoption of digital audio with AES 3 and MADI and the emergence of network audio, the distribution of audio has rapidly evolved in the last twenty years. With installations growing in scale and equipment list, network has also been introduced to monitor and control electronic units throughout the audio chain. In this tutorial, an evolutionary view on audio and control distribution in the context of live sound is presented. The pros and cons of multiple formats are outlined from a technological and practical perspective, explaining why more and more manufacturers are now turning to AVB MILAN as a preferred solution for simultaneous transmission of audio and control data on a single, possibly redundant, network infrastructure.

• Audio Signal Decorrelation for Live Sound

Presenter: **Adam J. Hill**, University of Derby

A "democracy of sound" is typically the goal in live sound, where every member of the audience receives the same high-quality listening experience. One of the most significant challenges in achieving this ideal comes from comb-filtering between coherent low-frequency sound sources and/or early reflections. This tutorial will describe the cutting edge in audio signal decorrelation, namely Diffuse Signal Processing (DiSP), demonstrating how such a decorrelation procedure, if implemented properly, can significantly limit the effects of comb-filtering without perceptually coloring the audio signal. Multiple potential applications will be explored, ranging from large-scale live event subwoofer systems to single subwoofer home theater systems.

• Immersive Audio Production and Reproduction for Industry Applications

Presenter: **Tom Ammermann**, New Audio Technology

Immersive and interactive audio applications in live events, full domes, art installations, and theaters are coming up fast. But dealing with unique loudspeaker configuration becomes to be a real challenge. Furthermore, the kind of input formats increase rapidly, and the reproduction of these input sources must be flexible and interactively as well. The workshop will show strategies of today rendering processors on-site and how to feat such engines with content from regular DAW workflows.

• Live Concert Sound: Mixing a Major World Tour

Moderator: **Terri Winston**, Women's Audio Mission

Panelists: *Loreen Bohannon*, Star Phoenix Productions LLC
Amanda Davis
Fela Davis, 23dB Productions

Learn what life is like on a major world tour as a Front of House (FOH) or Monitor Engineer with the artists Lizzo, Janelle Monae, Tegan and Sara, Ella Mai and Christian McBride. From the various roles on tour, what a typical day looks like, to favorite gear, most important skills, biggest challenges, and how to take care of yourself with 18 hour+ days, etc. Featuring: Loreen Bohannon: Monitor Engineer for Lizzo, Michael Bolton, FOH Engineer for Rusted Root Amanda Davis, FOH engineer for Janelle Monae, Tegan and Sara, Ella Mai Fela Davis, FOH engineer for Christian McBride, Jose Feliciano, Meshell Ndegeocello.

• Multi-Zone Alignment Strategies for Immersive Sound System Designs

Presenters: **Javier Frutos-Bonilla**, d&b audiotechnik
Boris Rehders, d&b audiotechnik

The design of sound reinforcement systems entails the segmentation of the venue into different zones that need to be addressed individually and aligned together to play as a complete system. For the usual stereo reinforcement scenario, we focus our efforts in choosing an alignment position that maximizes the performance for most listener positions given two sources to align, i.e., outfill and/or subwoofer left to stereo left. However, when doing immersive stage audio, the position of objects on the stage adds a new variable into the game, the space-time. On this workshop we will review different design examples, explain the factors to take into account for the alignment process and go through different strategies, especially when dealing with object based audio.

• Remote System Optimization: Tuning a Sound System across the World from Home

Presenters: **James Anderson**, Rational Acoustics LLC
Bob McCarthy, Meyer Sound
Merlijn van Veen, Meyer Sound Laboratories

How to remotely tune a sound system in Covid times using dual-channel FFT transfer function analysis programs. An introductory presentation on the challenges of remote tuning is followed by panel discussion and Q & A with remote participants.

• Wireless Microphone and Intercom Spectrum Update

Presenters: **Mark Brunner**, Shure Incorporated
Joe Ciaudelli, Sennheiser
Karl Winkler, Lectrosonics, Inc.

The results of the 600 MHz spectrum auction are upon us, as the new owners of the 616–698 MHz range have turned on their services and the TV channel re-pack is complete. Questions remain

about the potential T-Band auctions for the spectrum between 470–512 MHz. Join a panel of experts covering these changes and the affects on all UHF wireless microphone, intercom, IEM, and IFB users. Wireless microphone system best practices will also be covered briefly, including the important topics of band planning, frequency choice and coordination, RF gain structure, and body absorption of RF signals.

• Sound Engineering for Safe Listening
Tuesday, October 27, 2020
Session Time: 6:30 pm – 8:00 pm

Presenters: **Shelly Chadha**, World Health Organization
Adam J. Hill, University of Derby
Kaloyan Kamenov, World Health Organization
Mark Laureyns, AEA European Association of Hearing Aid Professionals
Michael Santucci, Sensaphonics
Ian Wiggins, University of Nottingham

WHO estimates that 1.1 billion young adults are at risk of hearing loss from dangerous sound exposure. This is partially due to unsafe use of personal audio devices and sound exposure in entertainment venues. Under the "Make Listening Safe" initiative, WHO developed tools to raise awareness of and change behavior towards safe listening, including: • WHO-ITU standard for safe listening devices and systems. • Regulatory framework for control of sound exposure in entertainment venues (currently under development). • Communication materials for journalists, governments, music makers, and schools. This session gathers experts in fields from sound reinforcement to audiology to share WHO's vision, provide insight into current actions and discuss the important role sound engineers can play in this global effort.

• Roundtable Discussion: Sound Reinforcement
Thursday, October 29, 2020
Session Time: 2:30 pm – 3:30 pm; 8:00 pm – 9:00 pm

Moderator: **Dan Mortensen**, Dansound Inc.

Join the Roundtable here: https://us02web.zoom.us/j/88028784140?pwd=MTlOTi9WcU9OZElWakkvOEVzOHZzdz09 Meeting ID: 880 2878 4140 Passcode: 570387 These Round Tables were created to both to allow specific audio interest groups to gather and share ideas, and also to allow the Presenters of On-Demand and Live presentations to interact face to face with attendees who are interested and may have questions about those topics. To that end, the Sound Reinforcement Track has scheduled (in no order) the following presenters (possibly with more panel members) to be present: 1. 3D Audio For Live Events - Scott Sugden 2. An Evolutionary View on Audio & - Etienne Corteel 3. Immersive Audio Production - Tom Ammerman 4. Multi-Zone Alignment - Javier Frutos-Bonilla 5. Research, Education, Knowledge - Elena Shabalina 6. Safe Listening Levels - Michael Santucci/Mark Laureyns/Marcel Kok Ground Rules / Etiquette: 1. You are entering the Discussion on Mute. Please remain on Mute unless you would like to contribute with a comment or question. 2. There is both a chat box and a Raise Hand function – you are welcome to use either of those to indicate that you would like to speak. Please then wait for the Moderator to invite you to unmute. 3. By entering this chat, you have agreed to abide by the AES Code of Conduct, which can be found at https://www.aes.org/download.cfm?filename=AES_Member_Code_of_Conduct_Final.pdf

• Sound Reinforcement in a Covid World: An International Panel of P.A. Company Representatives and Production Professionals Summarize the State-of-the-Art of Getting On With It
Thursday, October 29, 2020
Session Time: 6:30 pm – 8:00 pm

Presenters: **Victor Arko**, Eighth Day Sound
Tim McCulloch, Pro Audio Systems, Inc.
Gordon McGregor, GBR Scotland
Dan Mortensen, Dansound Inc.
Owen Orzack, Eighth Day Sound
Jonathan Stoverud-Myers
Denise Woodward, IATSE Local 16, San Francisco Symphony, NARAS Member

An ad-hoc panel talks about their preparations for getting back to work when the pandemic subsides enough to allow large gatherings. They will examine the necessary pro-active steps to protect their crews, their artists/clients, and the audience in this weird time. They will discuss how the best advice has changed over time and what seems like the current best advice. They will not tell you what you should do, nor will they purport to have the best advice. They will describe what they are doing.

• Research, Education and Knowledge Transfer in Sound Reinforcement
Friday, October 30, 2020
Session Time: 3:30 pm – 5:00 pm

Presenters: **Finn T. Agerkvist,** Technical University of Denmark
Etienne Corteel, L-Acoustics
Adam J. Hill, University of Derby
Manuel Melon, Le Mans University / LAUM
Martin Møller, Bang & Olufsen a/s
Elena Shabalina, d&b audiotechnik GmbH & Co. KG

This panel will address the link between research and industry in the field of sound reinforcement. In many other fields both industry and research benefit from a close contact and collaboration. What can we learn from them? The panellists will address the topic from their unique perspectives and will try to identify what works well and what needs improvement. What are the ways for industry and research institutions and universities to work together? To name a few, there are Masters thesis, industrial PhD, collaboration agreements, industry-funded research, larger research projects with several parties. What are the obstacles on the way? What are the steps we all can take to improve the situation? Join us and help us find out!

SPECIAL EVENTS

• Platinum Partner Session: Transforming the Performance of Your Room
Tuesday, October 27, 2020
Presentation Time: 10:30 am – 11:30 am

Presenters: **Paul Stewart**
Will Eggleston

The W371 seamlessly complements our 8341, 835, and 8361 monitors to create a series of free-standing full-range monitoring solutions. Will and Paul will discuss why we created the W371A, who it's designed for, and how it can transform the LF performance of your room—helping you to produce mixes that translate beautifully to other loudspeaker systems.

• Heyser Lecture
Tuesday, October 27, 2020
Session Time: 1:00 pm – 2:30 pm

Lecturer: **John M. Chowning**, Stanford University

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 Spring and Fall 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 award-

ed nine patents in audio and communication techniques and was widely known for his ability to clearly present new and complex technical ideas. Heyser was also an AES governor and AES Silver Medal recipient.

The Richard C. Heyser distinguished lecturer for the 148th AES Convention is John M. Chowning.

Realizing a Dream, a Discovery, and Lissajous Figures

While studying music in Paris in 1959, I heard electronic music composed for four loudspeakers surrounding the audience—what today we refer to as "quad." Struck by the feeling of space in this quad format, I imagined composing sounds that could move freely in more complicated paths, changing in azimuth and distance as an addition to the traditional parameters of music, pitch, loudness, and timbre. As a graduate student at Stanford my idea of sound in space remained but a dream. Then in November 1963 Max Mathews writes, "There are no theoretical limitations to the performance of the computer as a source of musical sounds." Max saw the future transformation toward digital technology: any sound wave can be approximated based on the sampling theory. And so, at 29, with years of music studies behind me, but never having even seen a computer and with no background in technology, but with good ears and a musical goal, I began! While looking for synthesized sounds having internal dynamism that would be perceptually distinguishable from their reverberant field, I experimented with extreme vibrato and with the computer there was no practical limit to either the depth or the rate. I produced a number of different timbres, having both harmonic and inharmonic spectra. That is another story that I will present at the 2021 New York lecture, as FM synthesis. I continued building programs to allow arbitrary sound trajectories in a quad space. I realized that I could increase the effectiveness of the distance cue, by localizing the reverberation with increasing distance. Finally, why do Lissajous figures have anything to do with moving sound sources? At the end of this presentation I will show convincing sound synchronous animations that demonstrate their improbable association with spatialization.

• Roundtable Discussion: Diversity & Inclusion
Tuesday, October 27, 2020
Session Time: 2:30 pm – 3:30 pm

Moderator: **Leslie Gaston-Bird**, Audio Engineering Society

Connect with your colleagues for lively discussion! Follow-up on sessions. Ask questions. Share your experience and perspectives. Network with friends and experts. Ask presenters and authors your questions. Get involved in a conversation on industry diversity and inclusion. Each Roundtable Discussion will be moderated by the Chairof the Track. We hope you'll join us! Ground Rules / Etiquette: 1. You are entering the Discussion on Mute. Please remain on Mute unless you would like to contribute with a comment or question. 2. There is both a chat box and a Raise Hand function – you are welcome to use either of those to indicate that you would like to speak. 3. By entering this chat, you have agreed to abide by the AES Code of Conduct, which can be found at https://www.aes.org/download.cfm?filename=AES_Member_Code_of_Conduct_Final.pdf

• Seven Audio Wonders of the World Encore: Skywalker Studios
Tuesday, October 27, 2020
Session Time: 2:30 pm – 3:30 pmM

With origins based in Ben Burtt's landmark work on 1977's *Star Wars*, Skywalker Sound specializes in sound design, mixing, and audio post-production across multiple mediums. From the gathering of real-life, organic sounds to developing new techniques in sound presentation, Skywalker Sound remains one of the world's most innovative facilities—eager to explore, create, and venture into the unknown.

• Seven Audio Wonders of the World Encore: Galaxy Studios
Tuesday, October 27, 2020
Session Time: 3:30 pm – 5:00 pm

The Galaxy Studios Group is a cluster of companies dedicated to the fine arts of film, music, and sound technology. Galaxy Studios is situated in Belgium and for more than 35 years the no-compromise studio complex has been a world-class beacon of advanced technology for sound recording and film post-production.

• Seven Audio Wonders of the World: Blackbird Studios
Tuesday, October 27, 2020
Session Time: 5:00 pm – 6:00 pm

Presenter: **Mark Rubel**, The Blackbird Academy

Founded in 2002 by sound engineer John McBride and his wife, country artist Martina McBride, Blackbird has become one of Nashville's preeminent sound studios. The Blackbird complex now includes nine studios and houses The Blackbird Academy, a post-secondary audio engineering school. Driven by a passion for great audio, Blackbird boasts an attentive, professional staff and a gear inventory second to none.

• Marquee Event: An Intimate and Compelling Journey to the Amazing World of Imogen Heap
Wednesday, October 28, 2020
Session Time: 11:00 am – 12:00 noon

Presenter: **Imogen Heap**

Known for her use of cutting-edge technologies in production and musical applications, Heap will expound upon her current projects and her aim to help music makers regain empowerment and visibility in both virtual and actual worlds.

• Seven Audio Wonders of the World: The Village
Wednesday, October 28, 2020
Session Time: 2:30 pm – 3:30 pm

A full-service recording, mixing, and live performance complex housed in a vintage 1920's Masonic Temple in West L.A., the Village Studios is famous for landmark sessions by music's legends across six decades. The Village is where "vintage gear heaven" meets state-of-the-art technology, accommodating orchestras, film scores, rock bands, hip-hop, & pop artists—all the way down to single-mic recordings such as voiceovers, audio books, and podcasts.

• Platinum Mastering Panel
Wednesday, October 28, 2020
Session Time: 5:00 pm – 6:00 pm

Moderator: **Jonathan Wyner**, iZotope/Berklee College of Music

Panelists: *Eric Boulanger*, The Bakery
Bob Ludwig, Gateway Mastering Studios
Herb Powers

Join Bob Ludwig, Eric Boulanger, Herb Powers, and Jonathan Wyner for an engaging conversation about the state of the art and the future of Mastering. We'll have a special focus on artist relationships, communication, and collaboration

• Seven Audio Wonders of the World: Abbey Road
Thursday, October 29, 2020
Session Time: 11:00 am – 12:00 noon

Presenters: **Brian Kehew**
John Krivit

ONE TIME ONLY!! The Abbey Road tour will take place on Thursday, October 29 at 11:00am ET ONLY. It will NOT be available On Demand. We hope you can join us for the live streamed tour! Abbey Road Studios is one of the most famous recording studios in the world and a global music icon (a band named the Beatles

had something to do with building that reputation). Originally a nine-bedroom house built in 1829, it was purchased in 1928 by the Gramophone Company, who went on to build the world's first purpose-built recording studio. The St John's Wood address was chosen for its large garden and ideal location—close enough to the performance spaces of the time, but away from the noise and vibrations of the traffic and trains.

• Keynote: "What They'll Say About the Music"—A Conversation with Finneas
Thursday, October 29, 2020
Session Time: 4:00 pm – 5:00 pm

Keynote Speaker: **FINNEAS**

Moderator: *Steve Baltin,* Forbes

In this AES Show Special Event, Keynote Speaker FINNEAS unites us through his candid look at the past year—from his "bedroom beginnings" to his amazing sweep of GRAMMY Awards including "Best Engineered Album, Non-Classical," "Record of the Year," "Album of the Year," and "Song of the Year" for his collaborations with Billie Eilish, along with the honor of being the youngest person to win the GRAMMY Award for "Producer of the Year, Non Classical." FINNEAS will talk about where we are now, how to be authentic in the recording process, and how to move forward, technically and creatively. And in the end, it's all about "What They'll Say About the Music."

• Seven Audio Wonders of the World: United Recording
Thursday, October 29, 2020
Session Time: 5:00 pm – 6:00 pm

Bill Putnam opened United Recording Studios in 1957, and it quickly became one of the most legendary recording facilities in the world. With his uncompromising standards, technical brilliance, and unparalleled knowledge of acoustics, Putnam built studios that sound like no other. Located on Sunset Boulevard, these magnificent studios have attracted the biggest names in recording history.

• Platinum Producers Panel: Mysteries of the Formidable Hook —A Producers View on Creating Impact Through Songwriting, Technology, and Sonic Textures
Thursday, October 29, 2020
Session Time: 6:00 pm – 7:00 pm

Moderator: **Michael Molenda,** Mi5 Recordings/Universal Music Group

Panelists: *Fab Dupont*, FLUX Studio
*Julian Raymon*d, Big Machine John Varvatos Records
Ebonie Smith, Gender Amplified, Inc.

Catching the ears of an often-distracted audience is the superpower of a successful producer. In this panel renowned producers Fab Dupont, Julian Raymond, and Ebonie Smith reveal how they deploy songwriting chops, audio-production strategies, fashion and style savvy, hook smarts, and other devices to make songs as compelling, commercial, and seductive as possible.

• Platinum Engineers Panel: Recording—Then, Now and Future
Friday, October 30, 2020
Session Time: 1:00 pm – 2:30 pm

Moderator: **Jimmy Douglass**

Panelists: *Marcella Araica*
Russell Elevado
Tony Visconti
Joe Zook

Tony Visconti and his engineering pals discuss recording in the new world!

• Platinum Partner Session: A Complete Systems Approach to Immersive Monitoring Solutions
Friday, October 30, 2020
Session Time: 2:00 pm – 3:00 pm

Presenters: **Will Eggleston**
Paul Stewart

In order to produce high quality immersive content that translates faithfully to all playback devices, you'll need a neutral, transparent monitoring system that is carefully optimized for your room. To demonstrate, Paul Stewart uses this webinar to guide you through the configuration of a complete immersive monitoring system. Using Genelec Smart Active Monitors and our GLM calibration software, Paul covers all aspects of setup, connectivity, and integration of both hardware and software. You'll learn how GLM is able to configure, calibrate and control an entire immersive system to produce mixes that you can rely on every time.

• Keynote: Jackson Browne—Let the Rhythm Lead: How the Chemistry of People and the Recording Process Fosters Inspiration
Friday, October 30, 2020
Session Time: 3:30 pm – 4:30 pm

Keynote Speaker: **Jackson Browne**

Moderator: **Scott Goldman**, GRAMMY Museum

Panelists: *Paul Beaubrun*
Jenny Lewis
Jonathan Wilson

Our inspiration, encouragement, influence, and strengths very often come from the people we surround ourselves with—who we connect with, and who help make us who we are. Music is such a powerful force that can connect friends, an audience, encapsulate a moment, trigger a memory—it unites the world with its universal language. When we bring together the right people for a project, our creative strengths and thoughts get pleasantly pushed further and further and we all shine. When you bring your best, you become a strength and an inspiration to all around you. This is exactly what happened when Jackson Browne, Jenny Lewis, Jonathan Wilson, Paul Beaubrun, and friends went to Haiti to collaborate on the World Music album *Let the Rhythm Lead: Haiti Song Summit, Vol. 1*, with the proceeds benefiting Artists For Peace And Justice and the Artists Institute of Jacmel, Haiti.

• Seven Audio Wonders of the World: Capitol Studios
Friday, October 30, 2020
Session Time: 5:00 PM – 6:00 PM

Since its completion in 1956, Capitol Studios has been a staple of the recording industry. Iconic artists like Frank Sinatra, Nat King Cole, and The Beach Boys first made musical history in its rooms, and to this day, major icons of popular music continue in their wake. Capitol Studios was recently fully refurbished to ensure that it remained a cutting edge, state-of-the-art facility where artisan craft can continue to thrive as it has for the past 60 years.

STUDENT WEEK

STUDENT EVENTS AND CAREER DEVELOPMENT

• SDA-1 Meeting

The first Student Delegate Assembly (SDA) meeting is the official opening of the Convention's student program. This opening meeting of the Student Delegate Assembly will introduce new events and election proceedings, 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.

• Student Mix Critiques 1

Moderator: **Ian Corbett**

Come and get tips, tricks, and advice to push your skills to the next level! The Student Mix Critiques are non-competitive listening sessions in which students receive feedback on their work from a panel of industry professionals. Students (with student registration for the AES Show) at any stage of their studies can submit a mix—something you recorded and mixed, or something you mixed. This will be a Zoom session, with the music examples played back via an AudioMovers webpage for better audio quality (links will be provided via the AES Show platform to registered attendees). Anyone registered for the AES Show can watch the session.

If you sign up, please make sure you log in to the start of the session, otherwise alternates will be placed on the schedule in your place. (Finalists in the Recording Competition please refrain from submitting because you will get this feedback as part of the competition process.)

• Education Fair
Session Time: 12 noon–2:00 pm

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, each in a virtual "room." Information on each school's respective programs will be made available and there will be an opportunity for you to interview and be interviewed. There is no charge for schools/institutions to participate.

• Student Recording Competition Cat. 1
Tuesday, October 20

Traditional Acoustic, Traditional Studio, Modern Studio

• Student Design Competition
Wednesday, October 21

• Student Recording Competition Cat. 2, 3, & 4 [on demand]

Sound for visual Media, Remix, Immersive

• MATLAB Plugin Competition [on demand]

Moderator: **Gabriele Bunkheila**, MathWorks

MathWorks continues to support the AES MATLAB Plugin Student Competition and Showcase, which invites students to design a new kind of audio production VST plugin using MATLAB Software. The competition provides students with the opportunity to challenge both their signal processing skills and creativity, and to share their results with the audio engineering community.

• Student Mix Critiques 2
Session Time: 12 noon –

Moderator: **Ian Corbett**

Come and get tips, tricks, and advice to push your skills to the next level! The Student Mix Critiques are non-competitive listening sessions in which students receive feedback on their work from a panel of industry professionals. Students (with student registration for the AES Show) at any stage of their studies can submit a mix – something you recorded and mixed, or something you mixed. This will be a Zoom session, with the music examples played back via an AudioMovers webpage for better audio quality (links will be provided via the AES Show platform to registered attendees). Anyone registered for the AES Show can watch the session.

If you sign up, please make sure you log in to the start of the session, otherwise alternates will be placed on the schedule in your place. (Finalists in the Recording Competition please refrain from submitting because you will get this feedback as part of the competition process.)

STANDARDS COMMITTEE MEETINGS

• SC-02-02: Digital InputOutput Interfacing
Monday, October 5, 2020
Session Time: 10:00 am – 11:30 am (all times listed are Eastern Daylight Saving times)

• SC-02-08: Audio File Transfer and Exchange
Monday, October 5, 2020
Session Time: 11:30 am - 12:30 pm

• SC-03-12: Forensic Audio
Tuesday, October 6, 2020
Session Time: 10:00 am - 11:00 am

• SC-03-06: Digital Archive Library Systems
Tuesday, October 6, 2020
Session Time: 11:00 am – 12:00 noon

• SC-02-01: Digital Audio Measurement Techniques
Tuesday, October 6, 2020
Session Time: 12:00 noon – 1:00 pm

• SC-04-03: Loudspeaker Modeling and Measurement
Wednesday, October 7, 2020
Session Time: 10:00 am – 11:30 am

• SC-04-09: Loudness and Annoyance
Wednesday, October 7, 2020
Session Time: 11:30 am – 12:30 pm

• SC-04-08: Measurement of Sound Systems in Rooms
Thursday, October 8, 2020
Session Time: 10:00 am – 11:00 am

• SC-05-05: Grounding and EMC Practices
Thursday, October 8, 2020
Session Time: 11:00 am – 12:00 noon

• SC-07-01: Metadata for Audio
Thursday, October 8, 2020
Session Time: 12:00 noon – 1:00 pm

• SC-02-12: Audio Applications of Networks
Friday, October 9, 2020
Session Time: 10:00 am – 11:30 am

• SC-04-04: Microphone Measurement and Characterization
Friday, October 9, 2020
Session Time: 11:30 am – 12:30 PM

• SC-05-02: Audio Connectors
Friday, October 9, 2020
Session Time: 12:30 pm – 1:00 pm

• AESSC Plenary
Saturday, October 10, 2020
Session Time: 10:00 am – 12:00 noon

TECHNICAL COMMITTEE MEETINGS

• Technical Committee Meeting on Coding for Audio Signals
Monday, October 26, 2020
SessionTime: 11:00 am – 12:00 noon

• Technical Committee Meeting on Loudspeakers and Headphones
Monday, October 26, 2020
SessionTime: 11:00 am – 12:00 noon

• Technical Committee Meeting on Signal Processing
Monday, October 26, 2020
SessionTime: 12:00 noon – 1:00 pm

• Technical Committee Meeting on Perception and Subjective Evaluation of Audio Signals
Monday, October 26, 2020
SessionTime: 1:00 pm – 2:00 pm

• Technical Committee Meeting on Automotive Audio
Tuesday, October 27, 2020
SessionTime: 11:00 am – 12:00 noon

• Technical Committee Meeting on Acoustics and Sound Reinforcement
Tuesday, October 27, 2020
SessionTime: 12:00 noon – 1:00 pm

• Technical Committee Meeting on Archiving Restoration and Digital Libraries
Tuesday, October 27, 2020
SessionTime: 1:30 pm – 2:30 pm

• Technical Committee Meeting on Coding for Audio Forensics
Wednesday, October 28, 2020
SessionTime: 12:00 noon – 1:00 pm

• Technical Committee Meeting on Recording Technology Practices
Wednesday, October 28, 2020
SessionTime: 12:30 pm – 1:30 pm

• Technical Committee Meeting on Spatial Audio
Thursday, October 29, 2020
SessionTime: 11:00 am – 12:00 noon

• Technical Committee Meeting on High Resolution Audio
Thursday, October 29, 2020
SessionTime: 12:00 noon – 1:00 pm

• Technical Committee Meeting on Fiber Optics for Audio
Thursday, October 29, 2020
SessionTime: 2:00 pm – 3:00 pm

• Technical Committee Meeting on Broadcast and Online Delivery
Friday, October 29, 2020
SessionTime: 11:00 am – 12:00 noon

• TLC Plenary Meeting
Thursday, November, 2020
SessionTime: 12:00 noon – 1:00 pm

AES SHOWCASE: EXHBITORS AND SPONSORS

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* * *

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For over 20 years, **AEA**'s goal has been to create the best microphones and preamps that we can make, to bring the imaginations of musicians and engineers to life. Since the creation of the R44C, AEA has expanded into designing and building new ribbon microphones, each with a unique application and function, using the same RCA traditions. Over the years, AEA has advanced ribbon technology, using new materials and updated manufacturing techniques. We now make a whole line of ribbon mics that each give musicians a unique sonic signature.

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Amphion offers near-field monitoring solutions, which are characterized by high resolution, imaging, and phase coherency. Point-source like reproduction and controlled dispersion delivers predictable results in wide range of acoustic environments. Amphion´s neutral balance and midrange detail work equally well for recording, mixing or mastering applications in stereo and multichannel alike.

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Analog Devices is a world leader in DSP, wireless, and power components for professional audio. Our private Demo Room will showcase new solutions for signal processing; speaker linearization; echo/noise cancellation; and networked audio, including the A2B digital audio bus and Dante support. Featured products include the SHARC Audio Module; SigmaDSP processors; Class-D amplifiers and more. Please contact AES@analog.com to schedule a meeting in our private Demo Room during AES.

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www.APIAudio.com

In 1969, **Automated Processes Inc.** was formed by engineers and musicians who shared a common vision: to create the highest possible quality professional audio gear and then back it up with excellent customer service. 50 years later, that vision is stronger than ever and remains an integral part of API's extraordinary success. We're grateful for the inspired genius of API Founder Saul Walker, and we're grateful for an ever-expanding number of users that recognize the unique value of API. That's why from the largest recording console to the smallest internal components, API continues our tradition of manufacturing the world's best analog recording equipment and offering the audio community our unmistakably warm analog sound.

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Audinate is the leading provider of professional digital audio networking technologies globally. Audinate's Dante platform distributes digital audio signals over computer networks, and is designed to bring the benefits of IT networking to the professional AV industry. Using Dante-enabled products ensures interoperability between audio devices and allows end users to enjoy high quality, flexible solutions—typically with a lower total cost of ownership.

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For over thirty-five years, **AP** has offered high-performance audio analyzers, accessories, and applications to help engineers worldwide design, validate, characterize, and manufacture consumer, professional, and industrial audio products. Ongoing innovation has been a key theme for the organization since its founding, with market-leading THD+N performance, a wide range of digital interfaces, software that combines power with ease-of-use, and tools for electro-acoustic and perceptual audio testing. Headquartered in Beaverton, Oregon, AP is part of the high-tech "Silicon Forest" of the greater Portland metropolitan area.

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www.audio-technica.com

Established in 1962, **Audio-Technic**a is a leading innovator in the design and manufacture of high-quality audio products. The company offers a complete range of wired and wireless microphones, professional headphones and audio accessories for every live-sound, recording, broadcast and installed-sound need

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https://austrian.audio/

Following the closure of the AKG offices in Vienna, we set out to create something new, challenging, and respectful to our heritage. Opening our doors in July 2017, **Austrian Audio** was born. Armed with a wealth of experience and a mission to make passion heard, our team of experts were ready to take on the industry giants by striving to produce products that embody the sound and engineering excellence one would expect from a company with our history. We may be relatively new, but the skill, knowledge, and tradition lives on at Austrian Audio, backed by over 350 years of cumulative engineering experience.

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time code to low-jitter word clock and video sync, and now PTP and GPS, Brainstorm products have been used and trusted for over 30 years by engineers worldwide in the post, broadcast and live environments.

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Clear-Com, an HME company, is a trusted global provider of professional real-time communications solutions and services since 1968. We innovate market proven technologies that link people together through wired and wireless systems. Clear-Com was first to market portable wired intercom systems for live performances. Since then, our history of technological advancements and innovations has delivered significant improvements to the way people collaborate in professional settings where real-time communication matters. For the markets we serve -- broadcast, live performance, live events, sports, military, aerospace and government—our communication products have consistently met the demands for high quality audio, reliability, scalability and low latency, while addressing communication requirements of varying size and complexity.

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Manufacturer of the original award-winning Cloudlifter line of microphone and instrument activators as well as high-quality ribbon microphones. **Cloud** products are 100% made in the USA and carry a limited lifetime warranty.

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Dale Pro Audio is a leading provider of solutions for the Broadcast, Live Sound, Install, Contractor, and Recording markets since 1956. We boast a top-notch sales staff, carry hundreds of the industry's best brands, and stock thousands of products in our Queens, NYC warehouse.

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Award-winning automixing technology from **Dan Dugan Sound Design** makes it easy to manage microphones in live settings, providing fast, transparent fades without upcutting, choppy sound or shifts in background noise. Transitions between talkers are always smooth. Models are available for analog, AES digital, ADAT, MADI, and Dante I/O.

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ETS-Lindgren provides quality acoustic chambers: anechoic, reverberation, predictable field, and small device—in both standard and custom designs—to meet your requirements. Our chambers are used by leading companies to test products, including loudspeakers, cell phones, microphones, IT equipment, and handheld devices, among other products. We also offer acoustic test services at our NVLAP 100286-0/ISO accredited Acoustic Research Lab. The lab includes a reverberant chamber for measuring sound transmission loss and a hemi-anechoic chamber for sound power measurements. These chambers are acoustically isolated on floating concrete floors and structurally isolated from the parent building. Our chambers are designed for fast specimen throughput, and can easily accommodate multiple test specimens of various sizes.

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Since 1971 **Eventide** has remained at the forefront of recording technology. In 1975 they revolutionized the audio industry by creating the world's first commercially available digital audio effects unit, the H910 Harmonizer. Since then, their legendary studio processors, effects pedals and plug-ins have been heard on countless hit records.

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Flux:: was founded the 1990s by Gaël Martinet, back then a sound engineer in the audio industry in Paris, on a mission to create the software tools he needed himself but couldn't find on the market. After over a decade, with a close collaboration as a subcontractor for Merging Technologies being deeply involved in the creation of Merging's well renowned products, in 2006 the first exquisite suite of audio plugins, tailored for professional sound engineers, was presented to the world. Today, two decades later the Flux:: product line presents a wide range of products for Immersive Audio, Audio Analysis and Audio Processing, used by sound engineers and producers in the music, broadcast, post production, mastering and live audio industry all over the world.

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The **Focusrit**e brand offers audio interfaces and other solutions for recording musicians, producers, podcasters, and audio professionals alike. Today the company is famous for offering unprecedented sonic performance at every price point, notably the ubiquitous Scarlett range of USB interfaces. Focusrite relentlessly pursues opportunities to inspire creativity through technology, constantly seeking new ways to eliminate technological barriers, without compromising on sound quality.

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Focusrite Audio Engineering has pioneered professional audio recording technology spanning almost three decades. Focusrite Pro, the company's professional and commercial division, meets the demands of recording, post-production, live sound, and broadcast professionals. It consists of RedNet, a fully modular audio-over-IP solution, and the Red range, Focusrite's flagship multi-format interfaces, along with the heritage ISA range of microphone preamplifiers and analogue signal processors. The solutions have been developed to meet the needs of the most demanding applications through a relentless focus on ease of use, quality, and reliability. Focusrite is based in High Wycombe, Buckinghamshire, with offices in Los Angeles and Hong Kong.

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www.iis.fraunhofer.de/audio

For over 30 years, the institute's Audio and Media Technologies division has been shaping the globally deployed standards and technologies in the fields of audio and moving picture production. Starting with the creation of mp3 and continuing with the co-development of AAC and the Digital Cinema Initiative test plan, almost all consumer electronic devices, computers and mobile phones are equipped with systems and technologies from Erlangen today. Meanwhile, a new generation of best-in-class media technologies—such as MPEG-H Audio, xHE-AAC, EVS, LC3/LC3plus, Symphoria, Sonamic and upHear—is elevating the user experience to new heights. Always taking into account the demands of the market, **Fraunhofer IIS** develops technology that makes memorable moments.

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GIK Acoustics manufactures and sells bass traps, acoustic panels, and diffusors direct to customers around the world. GIK Acoustics is the #1 choice in acoustic treatments for recording studios, listening rooms, home theaters, churches, restaurants, and live auditoriums. Founded in Atlanta, GA, in 2004, GIK Acoustics understands the importance of great sound. Dollar for dollar, GIK Acoustics' products absorb more sabins (sound) than any other product on the market. We proudly offer a large selection of highest quality products at affordable prices as well as provide clients with a professional design and support staff to achieve a the best sounding space possible.

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KLIPPEL is an innovation leader in providing unique test equipment dedicated to electro-acoustical transducers and audio systems. The novel techniques are the result of fundamental research over 30 years providing more accurate physical models of loudspeakers, micro-speakers, and headphones valid in small and large amplitudes. These tests reveal the root causes of signal distortion and give practical indications for further improvements in design and manufacturing. This cutting-edge technology changed the way loudspeakers have been developed and manufactured in the audio industry. The theory verified in practice is the basis for the Controlled Sound Technology. KLIPPEL GmbH was founded in 1997 by Prof. Dr. Wolfgang Klippel in Dresden,Germany. Acoustic engineers, software and hardware development experts, technicians, sales, and administration staff joint the KLIPPEL team.

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Lectrosonics is a U.S. company located in the city of Rio Rancho in the heart of New Mexico. With a continued focus on quality and innovation since 1971, Lectrosonics is well respected within the film, broadcast, music, and theater technical communities. We have a strong history of delivering products that satisfy your needs for quality wireless technology with excellent customer support and service. Lectrosonics wireless microphone systems and audio processing products are used daily in mission-critical applications by audio engineers familiar with the company's dedication to quality, customer service, and innovation. Lectrosonics provides consulting to its customers as part of its customer service for product sales. Lectrosonics will also assist you with sound system component selection and design.

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MICPARTS is the leading provider of DIY studio microphone kits and custom built microphones. We offer three large-diaphragm capsule voicings, two distinct FET circuits, multipattern mic kits, and two different tube mic kits. Be sure to log in to the show planner to see our upcoming live webinar on DIY mic building.

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There's a reason why **Mogami Cable** is called "the cable of the pros." Virtually every major recording facility is wired with Mogami, which means that just about any music you listen to has passed through Mogami somewhere in the recording chain. Mogami Cable is unmatched for accuracy, flexibility, and quality.

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Mojave Audio manufactures microphones designed by Technical Grammy Winner David Royer.

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Neutrik, the world's most recommended AV connector brand, is the leading supplier of professional connectivity solutions for the audio, video, lightin, & broadcast markets. Neutrik's opticalCON ADVANCED and opticalCON LITE provide up to 24 fibers in the compact D-size. For copper connectivity, Neutrik's etherCON Cat 5, Cat 6, and Cat 6A solutions suit virtually all rugged Ethernet connectivity requirements, offering ingress protection ratings up to IP 65. And an extensive range of UHD-capable BNC connectors. Active devices include the NA2-IO-DLINE and NA2-IO-DPRO 2-in/2-out Dante interfaces that can also be deployed in racks, podiums, and trusses using Neutrik's optional mounting accessories. NPS-30W is a compact, rugged PoE injector—a perfect companion for these and other products requiring PoE in a rugged, small form factor.

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NTI Audio will be showing its range of products for audio testing including audio analyzers, sound level meters (SLM), sound sources and signal generators. Powerful handheld products for testing analog and digital audio systems. Benchtop systems for R&D and production testing of audio systems and components: Smartphones, Speakers, Amplifiers and Microphones. New in 2019 is our Room Acoustics Reporter. Measure a room's reverberation time (RT60), compare to target values, and model the effects of adding sound absorbing material.

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NTP Technology (www.ntp.dk) produces high-reliability audio routing and signal processing systems for major international broadcasters, audio mastering, editing, and post-production studios under its Penta brand. DAD—Digital Audio Denmark (www.digitalaudio.dk),

NTP Technology's second brand, is renowned worldwide for analog-to-digital and digital-to-analog audio converters delivering outstanding sonic performance.

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NUGEN Audio creates intuitive professional audio software for high-end music producers, post-production engineers, and broadcasters. Our tools for audio analysis, loudness metering, mixing/mastering, and tracking are used worldwide by the world's top names to create leading TV, films, music and games. For more information, visit nugenaudio.com

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Software Publisher for High Accuracy Audio Tools. Focus on: Reference Metering for Mastering and Ingest / QC broadcast dep. 2nd Focus: Virtual Acoustics (immersive OEM libraries & systems). for post pro studios, premium cars, museums, multipurpose performance rooms & halls.Custom Implementations. Upmix algorithms (1D to 3D). AUDIO CI (Corporate Audio Identity) building blocks for post-pro studios. R&D Services & Technology for OEM business. Custom Worskshops. Consultancy for Virtual Acoustic systems. Consultancy of working groups for tech standards. legal audio expertise. compliance verification office.Scientific research.

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Pulse Techniques produces the original PULTEC designs of Pulse Techniques, Inc., including the EQP-1A, EQP-1A3, MEQ-5, and EQP-1S3 equalizers as well as mastering versions and 500-series models. We manufacture our EQs using the same methods and to the same exacting specifications as the original units. Top recording engineers, producers, and artists from around the world have discovered that our products actually sound exactly like the vintage units they have been using for most of their careers. Please visit our AES 2020 virtual booth (and our Website) to learn how you can own a brand new pair of Pultec EQs, complete with warranty, for what you might spend on a single vintage unit.

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Radial Engineering is the industry leader in direct injection boxes and Reamp devices used for live touring and studio applications. Radial manufactures studio and live tools for musicians, technicians, engineers, and producers across all levels of industry. Whether you need to convert signal levels for processing, split, switch or audition multiple source materials or instruments, drive signals long distances, or any other challenge you may face, Radial is sure to have a solution to help. In addition to electronic solutions, Primacoustic acoustic treatment offers ready-made solutions to treat your studio or space which may be subject to flutter echo, room modes, or other reflections that harm your listening experience. Room kits designed as a one-stop solution are available for a variety of room sizes.

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Resonado is an American technology company that designs and licenses proprietary speaker driver technologies to brands and manufacturers. The company's core pursuit is to develop speaker driver architecture that achieves both optimal form and function, without compromise. The company's flagship product, Flat Core Speaker (FCS) technology, is the first manifestation of that pursuit. FCS technology is a novel type of electrodynamic speaker driver engineered with a low depth structural profile and high diaphragm aspect ratio as an alternative to the conventional conical driver type to meet the ever-expanding demand for innovative, space-efficient product designs without experiencing tradeoffs in sound performance. Developed to be completely scalable, FCS technology can be implemented in products ranging from consumer electronics, to vehicles, to architectural spaces.

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"Why does the world need another mic?" Someone posted that to our Facebook page recently. I answered: "If this one sounded like all the rest, I'd agree with you." I founded the company and design the products, and I have no interest in selling "me, too" microphones or fake "vintage recreations." Instead, I apply many years' research (see RecordingHacks.com) to creating unique and distinctive microphones, with premium capsules and components and solid engineering. It's science, not fiction. LOG IN to the Show Planner to see our upcoming livestreamed studio demo / Virtual Showroom event.

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Rupert Neve Designs is nestled in the picturesque Texas Hill Country, where all of our modules are crafted and painstakingly tested. Our small team of dedicated engineers have designed a new range of professional audio gear and systems based on both proven classic design concepts and modern innovations. This new range of equipment utilizes custom-designed transformers and single-sided amplification to provide sound quality that rivals the best recording gear —vintage or modern—available anywhere in the world.

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Designed for the highest level of transparent, accurate, and natural sounding recordings, the **Sanken** Chromatic Series of 10 distinct microphones can handle anything from the roughest sounds of a metalcore band to the supernatural realm of symphonic recording. Sanken will feature the new CUX-100K super wide-range 3-mode mic at the show. They are designed and meticulously engineered in Japan using the latest technology and include many innovative features such as membranes impervious to humidity and temperature change. The result is a set of mics that have phenomenal frequency response, accurate off axis response, exact patterns, minimal proximity effect, low noise and high SPL tolerance.

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SCHOEPS Microphones is a family business that, since its inception in 1948, has earned itself a reputation as one of world's finest manufacturers of microphones. Our entire product range is designed and created in Durlach, an old district in the German city of Karlsruhe. We are a team of 50 co-workers who develop our own original products, and manufacture all important their parts including the capsules.

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Our range of premium products is made up of wireless microphones and monitoring systems, wired microphones, conference and information technology, headphones, audiology products, and streaming and 3D audio technologies. Today the name of **Sennheiser** brings together a number of strong brands including Georg Neumann in Berlin, the world's leading producer of studio microphones and monitors.

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We're a company of engineers. But we're also a company of performers, artists, presenters, concertgoers, and sometimes yeah, fans. We know the excitement before going on stage, starting a critical conference call or listening to your favorite music. We understand what you're going through, and what you need. This year at AES, we're excited to unveil DuraPlex lavalier and headworn waterproof microphones. This new line brings legendary **Shure** sound and quality to new applications and environments requiring exposure to the elements. But we're not stopping there. We're announcing ANOTHER product on Monday, October 26, and hosting a release event exclusively for AES attendees. To get details on how to attend, register at p.shure.com/AES-shurenext.

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Sound Particles is a software for epic sound design, capable of generating thousands of sounds in a virtual 3D audio world. This immersive audio application will enable you to create highly complex sounds on the fly, which will ultimately enable you to create sound better and faster than ever. he software is used by all major Hollywood studios to create epic sounds for movies such as *Aquaman* or *Ready Player One* and TV series such as "Game of Thrones."

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Digigram develops mission-critical solutions dedicated to the remote broadcast, production, and safe distribution of audio content. Digigram sound cards, IP audio codec solutions, cloud applications, and networking infrastructure are used in thousands of broadcast, AV, and industrial applications.

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From the very beginning, unsurpassed performance has been one of the cornerstones of **RME**'s product design, and this is even more evident today. RME were the first to deliver professional performance over USB 2.0 and have recently paved the way for multichannel audio on USB 3.0 and Thunderbolt technology for Windows. RME's refusal to compromise on any aspect of product design or manufacture has resulted an unrivaled reputation for quality, performance and reliability.

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The Blackbird Academy – Professional School of Audio: Founded in 2013, The Blackbird Academy has quickly risen to a top-tier school for professional audio (consistently ranked in the top 5 audio/

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Voyage Audio was founded in San Diego, CA, to design, build and market innovative microphones and software for audio recording and production. With a rich history in electronics, music, and over 25 years of combined experience designing and building microphones, the Voyage Audio co-founders have brought dozens of products to market for major mic manufacturers, winning both Pro Audio Review and TEC awards. Spatial Mic, the first product from Voyage Audio, begins a journey to create unique recording tools, opening new frontiers in music production while supporting emerging technology in Virtual and Augmented Reality.