AES 142ND CONVENTION PROGRAM

MAY 20–MAY 23, 2017

MARITIM HOTEL, BERLIN, GERMANY

The Winner of the 142nd AES Convention Best Peer-Reviewed Paper Award is:

An Analytical Approach for Optimizing the Curving of Line Source Arrays—*Florian Straube, Frank Schultz, David Albenés Bonillo, Stefan Weinzier*l, Technical University of Berlin, Berlin, Germany

Convention Paper 9699

To be presented on Saturday, May 20, in *Session 2—Transducers 1*

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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 142nd-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.

The Winner of the 14nd AES Convention Student Paper Award is:

Joint Parameter Optimization of Differentiated Discretization Schemes for Audio Circuits—
Francois Germain,[1] *Kurt James Werner*[2]
[1]Stanford University Stanford, CA, USA
[2]Queen's Univerity Belfast, Belfast, UK

Convention Paper 9751

To be presented on Sunday, May 21, in *Session 13—Audio Processing and Effects*

Tutorial 1 **Saturday, May 20**
9:00 – 10:00 **Salon 7 Vienna**

THE USE OF NON-LINEAR SIGNAL PROCESSING IN MUSIC PRODUCTION

Presenter: **Austin Moore**, University of Huddersfield, Huddersfield, UK

When mixing music in the box it has become ever more important to apply subtle and sometimes not so subtle distortion and non-linear processing for coloration, changes to instrumental timbre, and creative effects. There currently exists a wide range of software emulations of classic hardware devices that can be used for non-linear processing and each have their own unique sonic signature. This tutorial investigates the fundamental principles behind distortion and coloration and explores how non-linear processing can be used by the mix engineer. The use of compressors for coloration, the effect of tape and the use of saturation plugins that emulate preamp clipping behavior will be demonstrated in the workshop and a full mix will be processed to provide audio examples that supplement the theory.

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

Tutorial 2 **Saturday, May 20**
9:00 – 10:00 **Salon 4+5 London**

OVERVIEW AUDIO AND VIDEO OVER IP STANDARDS

Presenters: **Andreas Hildebrand**, ALC NetworX, GmbH
Kevin Gross, AVA Networks

Since the publication of the AES67 Standard on High performance Streaming Audio-over-IP, further work has been started in various standardization organizations. Notably, the SMPTE is working on a suite of standards based on the work of the Joint Task Force for Network Media (JT-NM) and AES67 - SMPTE2110. This session will provide an overview on those standards, highlight their motivation, briefly explain the protocols and network mechanics chosen and how they relate to AES67.

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

Session P1 **Saturday, May 20**
9:30 – 12:00 **Salon 1 Moscow**

SPATIAL AUDIO—BINAURAL 1

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

9:30

P1-1 Perceptual Evaluation of Synthetic Early Binaural Room Impulse Responses Based on a Parametric Model—*Philipp Stade,*[1,2] *Johannes M. Arend,*[1,2] *Christoph Pörschmann*[1]
[1]TH Köln, Cologne, Germany
[2]Technical University of Berlin, Berlin, Germany

Binaural synthesis is often applied in the field of spatial audio to create a virtual acoustic environment using binaural room impulse responses (BRIRs). In the same area of research, spherical microphone arrays are gaining importance and allow for a spatio-temporal analysis. We present a new approach to describe the acoustical environment by a parametric model using sound field analysis. Combining spherical head related impulse responses (HRIRs) with this description, early BRIRs are synthesized and compared to the measured counterparts in a perceptual evaluation. The listening experiment revealed adequate performance of the approach, almost independently from room and test signal. Surprisingly the synthesis of direct sound and only diffuse reverberation yielded nearly the same results as for the entire parametric model.
Convention Paper 9688

10:00

P1-2 Implementation and Evaluation of a Low-Cost Headtracker for Binaural Synthesis——*Michael Romanov,*[1,2] *Paul Berghold,*[1,2] *Daniel Rudrich,*[2,3] *Markus Zaunschirm,*[2,3] *Matthias Frank,*[2,3] *Franz Zotter*[2,3]
[1]Graz University of Technology, Graz, Austria
[2]University of Music and Performing Arts Graz, Graz, Austria
[3]Institute of Electronic Music and Acoustics, IEM

Human auditory localization strongly relies on head movements. Thus, for plausible perception of virtual acoustic scenes the incorporation of head movements is mandatory. This is achieved via loudspeaker playback as a listener can move the head relatively to the scene. When using binaural synthesis the head movements need to be tracked and the scene needs to be rotated accordingly to achieve a stable perception of the acoustic scene. We present a low-cost, plug-and-play device (MrHeadTracker) to facilitate head-tracking based on the Arduino platform and the BNO055 sensor. Its performance is compared against another low-cost device (GY-85) and an optical tracking system (Optitrack Flex 13). The proposed MrHeadTracker outperforms the GY-85 device in terms of accuracy and latency and yields comparable results to the optical tracking system.
Convention Paper 9689

10:30

P1-3 Influence of Head Tracking on the Externalization of Auditory Events at Divergence between Synthesized and Listening Room Using a Binaural Headphone System—*Stephan Werner, George Götz, Florian Klein,* Technical University of Ilmenau, Ilmenau, Germany

This contribution presents an investigation on the influence of head tracking on the perceived externalization of auditory events using a binaural headphone system. Recordings of individual binaural room impulse responses of a five channel loudspeaker setup in two acoustic different rooms are conducted. Test persons are divided into two groups, while for the first group the listening and synthesized rooms do match (convergence), they do not for the second group (divergence). Moving the head during listening is mandatory and controlled by the test procedure. Perceived externalization of auditory events is used as a quality feature. The analysis of the ratings confirms that head tracking increases perceived externalization. Furthermore, the room divergence effect can be confirmed. Significantly lower externalization is observed if a divergence between the resynthesized and listening room occurs. However, the results clearly show that the benefit of head tracking on externalization does not overcome the room divergence effect.
Convention Paper 9690

11:00

P1-4 Laboratory Reproduction of Binaural Concert Hall Measurements through Individual Headphone Equalization at the Eardrum—*David Griesinger,* David Griesinger Acoustics, Cambridge, MA, USA

Progress relating measurements to perception of acoustics of all kinds has been stymied by the difficulty of accurately reproducing a room sound in a laboratory. Spatial aliasing above 1000 Hz, where most information in speech and music resides, severely limits the ability of multiple loudspeaker systems to reproduce proximity. We have developed a simple method of equalizing headphones that accurately reproduces the timbre of a frontal sound source at the eardrums. Combining individual headphone playback with Tapio Lokki's anechoic recordings makes hall research inexpensive, rapid, and accurate. We can easily test the effects of early reflections and other spatial properties. We find the earliest reflections, whether medial or lateral, are almost always detrimental. Examples from real halls will be presented.
Convention Paper 9691

11:30

P1-5 Approaching Immersive 3D Audio Broadcast Streams of Live Performances—*Giordano Jacuzzi,*[1] *Sofia Brazzola,*[1] *Johannes Kares*[2]
[1]Sennheiser (Schweiz) AG, Urdorf, Switzerland
[2]Sennheiser Austria GmbH

This paper explores the requirements and best practices of recording, mixing, and streaming broadcasts of live music performances in binaural and ambisonics formats. We outline the optimal workflows for incorporating and executing 3D audio streams with existing in-house infrastructure, as determined from our experience gathered testing and broadcasting public concert events in partnership with Moods jazz club in Zurich, Switzerland, and Vienna State Opera in Vienna, Austria. In addition, this paper discusses the current technological barriers for immersive audio content creation and consumption, areas for growth and improvement, and future projections for 3D immersive audio technology.
Convention Paper 9692

Session P2 **Saturday, May 20**
9:00 – 12:30 **Salon 2+3 Rome**

TRANSDUCERS 1

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

9:00

P2-1 Active Vibration Control of Breakup Modes in Loudspeaker Diaphragms—*William Cardenas*, Klippel GmbH, Dresden, Germany

One of the factors that contribute to the degradation of the sound quality of the loudspeakers are the breakup modes of the membrane since they cause complex directivity patterns, peaks, and deeps in the frequency response. This paper presents an active vibration control simulation applied to a 2D Finite Element model of a loudspeaker loaded by a fluid domain to demonstrate an innovative alternative to reduce the amplitude of the breakup modes of loudspeaker diaphragms, improving substantially the mechanical and acoustical performance. The benefits of the controlled system are demonstrated in terms of the acceleration response of the cone and the acoustic directivity.
Convention Paper 9693

9:30

P2-2 An Acoustic Radiator with Integrated Cavity and Active Control of Surface Vibration—*Arthur Berkhoff,*[1,2] *Farnaz Tajdari*[1]
[1]University of Twente, Enschede, The Netherlands
[2]TNO Acoustics and Sonar, The Hague, The Netherlands

This paper presents a method to realize an acoustic source for low frequencies with relatively small thickness. A honeycomb plate structure that is open on one side combines the radiating surface and the major part of the air cavity. The vibration of the plate is controlled with a decentralized feedback controller. The fundamental resonance is controlled, as well as higher-order bending modes, while avoiding possible instabilities due to the fluid-structure interaction. The smooth and well defined frequency response enables robust feedforward control for further response equalization. The influence of different actuation principles on the overall system efficiency is compared.
Convention Paper 9694

10:00

P2-3 The Acoustic Design of Minimum Diffraction Coaxial Loudspeakers with Integrated Waveguides—
Aki Mäkivirta, Jussi Väisänen, Ilpo Martikainen, Thomas Lund, Siamäk Naghian, Genelec Oy, Iisalmi, Finland

Complementary to precision microphones, creating an ideal point source monitoring speaker has long been considered the holy grail of loudspeaker design. Coaxial transducers unfortunately typically come with several design compromises, such as adding intermodulation distortion, giving rise to various sources of diffraction, and resulting in somewhat restricted maximum output performance or frequency response. In this paper we review the history of coaxial transducer design, considerations for an ideal point source loudspeaker, discuss the performance of a minimum diffraction coaxial loudspeaker and describe novel designs where the bottlenecks of conventional coaxial transducers have been eliminated. In these, the coaxial element also forms an integral part of a compact, continuous waveguide, thereby further facilitating smooth off-axis dispersion.
Convention Paper 9695

10:30

P2-4 Root Cause Analysis of Rocking Modes in the Nonlinear Domain—*Andreas Schwock,*[1] *William Cardenas,*[2] *Mattia Cobianchi,*[1] *Wolfgang Klippel*[2]
[1]B&W Group Ltd., West Sussex, UK
[2]Klippel GmbH, Dresden, Germany

Rocking modes are caused by small imbalances in the distribution of stiffness, mass, and force factor. A measurement technique to determine these root causes, using laser vibrometry, parameter identification, and root causes analysis has been presented in a previous paper. This paper focuses on the application of this technique to examine rocking modes in nonlinear domain. An incremental DC-offset is applied to the loudspeaker to examine changes of the root causes of the rocking throug hout the working range of the loudspeaker.
Convention Paper 9696

11:00

P2-5 Nonlinearity of Ported Loudspeaker Enclosures—*Juha Backman,* Hefio Oy, Espoo, Finland; Genelec Oy, Iisalmi, Finland

This paper presents the results of a computational fluid dynamics analysis of an unlined ported enclosure, focusing on the behavior around the tuning frequency. The work presents results for the amplitude dependence of the behavior and the time development of the sound field. The results indicate that the vortex formation around the port ends has a significant effect already at a relatively low flow velocities, and that the nonlinearity of the port is clearly visible in the acoustical load seen by the driver at the resonance frequency.
Convention Paper 9697

11:30

P2-6 Efficiency Investigation of Subwoofer Driven Around Resonance Frequency—*Tobias Thydal, Niels Elkjær Iversen, Arnold Knott,* Technical University of Denmark, Kgs. Lygby, Denmark

The need for efficient portable speaker systems has increased tremendously over the past 10 years. The batteries, amplifiers, and filtering has all seen great improvements in efficiency leaving the speakers' units as the most inefficient part of the system, mainly due to the large amounts of current drawn that ends up being dissipated as heat in the voice coil. This paper will look at how you can design a speaker system to take advantage of the resonance of a speaker unit, since that is where the unit is most efficient and draws the least current. A subwoofer speaker system will be designed with focus on only driving the speaker units near their resonance frequency. The tests found that with modern DSP it was rather simple to design a speaker system that operates in a very narrow frequency band around the speaker units' resonance frequencies, which in turn ensured a very small current draw. This greatest drawback of this method is the increase in components needed, which drives up cost and complexity.
Convention Paper 9698

12:00

P2-7 An Analytical Approach for Optimizing the Curving of Line Source Arrays—*Florian Straube, Frank Schultz, David Albenés Bonillo, Stefan Weinzier*l, Technical University of Berlin, Berlin, Germany

Line source arrays (LSAs) are used for large-scale sound reinforcement aiming at the synthesis of homogeneous sound fields for the whole audio bandwidth. The deployed loudspeaker cabinets are rigged with different tilt angles and/or electronically controlled in order to provide the intended coverage of the audience zones and to avoid radiation towards the ceiling, reflective walls or residential areas. This contribution introduces the analytical polygonal audience line curving (PALC) approach for finding appropriate LSA cabinet tilt angles with respect to the geometry of the receiver area, and the intended coverage. PALC can be previously applied to a numerical optimization of the loudspeakers' driving functions. The method can be used with different objectives, such as a constant interaction between adjacent cabinets with respect to the receiver geometry or by additionally considering amplitude attenuation. PALC is compared with typical standard LSA curving schemes. The advantages of the presented approach regarding sound field homogeneity and target-oriented radiation will be shown with the help of technical quality measures.
Convention Paper 9699

Session P3 **Saturday, May 20**
9:30 – 12:30 **Gallery Window**

POSTERS: ROOM, RECORDING, AND LISTENING

9:30

P3-1 Analysis of the Subgrouping Practices of Professional Mix Engineers—*David Michael Ronan,*[1] *Hatice Gunes,*[2] *Joshua D. Reiss*[1]
[1]Queen Mary University of London, London, UK
[2]University of Cambridge, Cambridge, UK

Subgrouping facilitates the simultaneous manipulation of a number of audio tracks and is a central aspect of mix engineering. However, the decision process of subgrouping is a poorly documented technique. This study sheds light on this ubiquitous but poorly defined mix practice and provides rules and constraints derived from a questionnaire that could be used in intelligent audio production tools. We prepared an online questionnaire consisting of 21 questions testing nine assumptions and identifying subgrouping decisions, such as why a mix engineer creates subgroups, when they subgroup and how many subgroups they use. We analyzed responses from 10 award winning mix engineers. Thematic analysis enabled us to discover five themes: Decisions, Subgroup Effect Processing, Organization, Exercising Control, and Analogue versus Digital. By analyzing the themes and each respondent's quantitative data we were able to show that eight out of nine assumptions could be accepted to be true.
Convention Paper 9700

9:30

P3-2 Combining Preference Ratings with Sensory Profiling for the Comparison of Audio Reproduction Systems—*Tim Walton,*[1,2] *Michael Evans,*[2] *Frank Melchior,*[2] David Kirk[3]
[1]Newcastle University, Newcastle upon Tyne, UK
[2]BBC Research & Development, Salford, UK
[3]Northumbria University, Newcastle upon Tyne, UK

One aim of perceptual audio evaluation is to understand the relationships between individual sensory attributes and overall quality of experience. This paper discusses one perceptual evaluation method by which this can be realized. Open Profiling of Quality (OPQ), a method first introduced in the field of visual and audiovisual evaluation, involves psychoperceptual evaluation, sensory profiling, and external preference mapping stages and is suitable for use with naive listeners. Here, a methodological case study is presented in which we discuss the implementation of this method and its adaptation for the comparison of audio reproduction systems.
Convention Paper 9701

9:30

P3-3 The Audience Effect on the Acoustics of Ancient Theaters in Modern Use—*Gino Iannace, Amelia Trematerra,* Università della Campania Luigi Vanvitelli, Aversa, Italy

Ancient theaters are used in modern contexts for different types of shows. When ancient theaters are used for musical performances, the audience criticizes the acoustics due to either not being able to understand what is spoken or the weakness of the music. An important aspect is the presence of the audience in the cavea, with it being important to understand whether it can have a negative role. Since it is not possible to take acoustic measurements during theater performances, the evaluation of the effects of the presence of the audience on the acoustics is carried out virtually through the software, "Odeon," in which the presence of the audience is simulated by changing the absorption coefficient value of the cavea.
Convention Paper 9702

9:30

P3-4 Evaluation of Training to Improve Auditory Memory Capabilities on a Mobile Device Based on a Serious Game Application—*Hunor Nagy, György Wersényi,* Széchenyi István University, Györ, Hungary

Capabilities of the auditory memory system were tested in a serious game application developed for the Android mobile platform. Participants played the well-known game of finding pairs by flipping and remembering objects on cards arranged in a matrix structure. Visual objects were replaced by iconic auditory events (auditory icons, earcons). Total time and different error rates were recorded and the effect of training was also evaluated. Results indicate that training contributes to a better performance and human voice samples are the easiest to remember.
Convention Paper 9703
Paper presented by György Wersényi

9:30

P3-5 Conversational Speech Quality in Noisy Environments—*Michal Soloducha,*[1] *Alexander Raake,*[1] *Stefan Bleiholder,*[2] *Frank Kettler*[2]
[1]Ilmenau University of Technology, Ilmenau, Germany
[2]HEAD acoustics GmbH, Herzogenrath, Germany

The present study reports on a conversation test conducted to reveal insights on how telephony users perceive speech transmission quality in a noisy environment. For this pur-

pose, a telephony setup has been built to simulate different degradations typical of real-life situations. A range of different conditions has been presented during the subjective test including usage of different terminals, environmental noises, and a noise suppression algorithm. A noise reproduction system has been installed on one side of the telecommunication channel to create an immersive noisy environment. Special focus of this paper is on the influence of different terminal devices and their signal processing on subjective quality. Moreover, more generic conclusions regarding conversational quality testing are provided.
Convention Paper 9704

9:30

P3-6 Acoustic Room Modelling Using a Spherical Camera for Reverberant Spatial Audio Objects—*Hansung Kim,*[1] *Richard J. Hughes,*[2] *Luca Remaggi,*[1] *Philip J. B. Jackson,*[1] *Adrian Hilton,*[1] *Trevor J. Cox,*[2] *Ben Shirley*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]University of Salford, Salford, UK

The ability to predict the acoustics of a room without acoustical measurements is a useful capability. The motivation here stems from spatial audio reproduction, where knowledge of the acoustics of a space could allow for more accurate reproduction of a captured environment, or for reproduction room compensation techniques to be applied. A cuboid-based room geometry estimation method using a spherical camera is proposed, assuming a room and objects inside can be represented as cuboids aligned to the main axes of the coordinate system. The estimated geometry is used to produce frequency-dependent acoustic predictions based on geometrical room modelling techniques. Results are compared to measurements through calculated reverberant spatial audio object parameters used for reverberation reproduction customized to the given loudspeaker set up.
Convention Paper 9705
Paper presented by Luca Remaggi

9:30

P3-7 Estimating the Diffuseness Level of the Acoustic Field—Reverberation Chamber Under Study—*Bartłomiej Chojnacki, Adam Pilch, Tadeusz Kamisinski, Artur Flach,* AGH University of Science and Technology, Krakow, Poland

A reverberation chamber is a widely used type of a laboratory room for widespread usage of standards like a ISO 354, ISO 17497-1 or ISO 3741. Chambers' shapes and types vary a lot around the world and so do the results of the sound absorption coefficient measurements, even though they meet the standard criteria. Verification of the ISO standards parameters is required, also introducing extra parameters describing the level of sound field diffuseness. Model studies have been conducted using the ray-tracing method in order to verify the level of sound field diffuseness in varying versions of irregular reverberation chamber, further rated with kurtosis of room normalized impulse response and sound field diffuseness coefficient, in order to assess the geometries under study.
Convention Paper 9706

9:30

P3-8 Environmental and Technical Problems in Acoustical Scaled Models—*Aleksandra Majchrzak, Bartłomiej Chojnacki, Katarzyna Baruch, Jaroslaw Rubacha,* AGH University of Science and Technology, Krakow, Poland

Using acoustical scaled models provides numerous theoretical and practical issues. After formulating theoretical requirements regarding the sound field in a given object, one should also conduct considerable amount of measurements of acoustical properties of materials to be used in a construction or the sound source. The paper discusses difficulties met during performing acoustical measurements in a scaled reverberation chamber in Technical Acoustics Laboratory AGH, such as utilized sound sources and the ways of adjusting atmospheric conditions (relative humidity and temperature). The paper particularly concerns the ideas of changing the conditions of measurement environment, sound sources being in use and the frequency range of the performed measurements.
Convention Paper 9707

9:30

P3-9 OpenAirLib: A JavaScript Library for the Acoustics of Spaces—*Kenneth Brown,*[1] *Matthew Paradis,*[2] *Damian T. Murphy*[1]
[1]Univerity of York, York, UK
[2]BBC Research and Development, London, UK

The possibilities for creating online sonic art and virtual acoustic environments have been increased by the introduction of audio Application Programming Interfaces, libraries, and online resources of acoustic impulse responses (which encapsulate the acoustics of real or imaginary spaces.) This paper presents the OpenAirLib JavaScript library that extends the capabilities of the World Wide Web Consortium Web Audio API, facilitating the incorporation of three-dimensional acoustics of spaces into web audio projects. It enables first-order ambisonic material from the Open Acoustic Impulse Response Library (OpenAIR) website to superimpose the acoustics of one space onto synthesized or recorded material from another space. An example is then described which uses these technologies to produce instances of generative online sonic art from OpenAIR data.
Convention Paper 9708
Paper presented by Damian Murphy

9:30

P3-10 Measurement and Visualization of Sound Intensity Vector Distribution in Proximity of Acoustic Diffusers—*Adam Kurowski, József Kotus, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

In this work, we would like to present analyses and visualizations of sound intensity distribution measured in proximity of an acoustic diffuser. Such distribution may be used for estimation of basic acoustic parameters of a diffuser. Measurement is performed with the use of a logarithmic sine sweep that allows for the analysis of waves scattered by the diffuser and rejecting the direct sound signal component. Pressure and sound intensity vector impulse responses are measured simultaneously. The measurement is carried out for a grid of 37 points arranged at equal intervals lying in a semicircle. To investigate the impact of objects evaluated on the sound wave propagation diffusion coefficients and sound intensity vector distributions are then compared.
Convention Paper 9709

Workshop 1 **Saturday, May 20**
9:30 – 10:30 **Berlin-A**

THE FUTURE OF MASTERING

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

Mastering as a discipline is entering a period of change. The disintermediation of delivery models and the arrival of assistive tools mean we have to re-think our practices and prepare for new techniques and workflows. This workshop will unpack what is changes now, how we are adapting to it, and what might be coming down the road.

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

Tutorial 3 **Saturday, May 20**
10:00 – 11:00 **Salon 4+5 London**

HISTORY OF DIGITAL SIGNAL PROCESSORS FOR AUDIO

Presenter: **Umberto Zanghieri**, ZP Engineering srl

The history of audio signal processing, starting from the 70s up to today, is marked by several interesting achievements. Signal processing for audio synthesis and computer music research was based on dedicated hardware specifically designed for those applications, but then evolved into the current solutions for general-purpose audio signal processing. The digital audio processing scenario has changed considerably in the last two decades, as new alternatives to traditional DSP devices are now available; furthermore, licensed cores now represent the majority of digital signal processors currently in use. An overview of digital signal processors for audio and options available in the last 40 years will be presented, along with considerations for selecting a suitable DSP device for new projects, and possibly for anticipating trends in future work opportunities.

Tutorial 4 **Saturday, May 20**
10:15 – 12:15 **Salon 7 Vienna**

AUDIO STUDIO DESIGN—AN EYE ON IMMERSIVE AUDIO

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

Presenters: *Renato Cipriano*, Walters-Storyk Design Group
Dirk Noy, Walters-Storyk Design Group

In an industry deluged by acronyms, Immersive Audio (IA) appears to have leapfrogged the trend. As with Surround Sound, Quad Sound and their various 3.1 - 5.1 - 7.1, etc., iterations, much of the noise made by these new innovations is focused on hype rather than on specific real world listener/viewer needs or actual science. That said, IA systems for producing, distributing and receiving this new sound experience do exist, they work and, they are proliferating. Fundamental design considerations for audio studios need to be maintained, but now with an eye on expanding mix configurations for the mixing engineer, the recording artist and the capturing microphone(s). This event will explore Immersive Audio Studio Design, including acoustics, aesthetics, ergonomics, master planning and future proofing.

Tutorial 5 **Saturday, May 20**
11:00 – 11:45 **Salon 4+5 London**

BULLETPROOF, HIGH QUALITY DIGITAL AUDIO TRANSMISSION: PRACTICAL DESIGN CONSIDERATIONS AND STANDARDS REVISIONS FOR RELIABLE HIGH QUALITY DIGITAL AUDIO

Presenter: **Jon D. Paul**, Vice President Scientific Conversion, Inc. , Founder, the Crypto-Museum

Engineers assume that all digital transmission must be perfect regardless of source, cable, destination or sample rate. However, in professional, cinema, and broadcast markets there are cases of signal dropouts, link failures, and even transient damage to the transceivers. The causes and solutions are in the design and choice of components and cables, as well as the digital audio physical layer standards (AES 3-4, AES 3id).

The tutorial summarizes 28 years of experience, with observations and test results for many types of cables, revealing huge variations in cable transmission and received spectra and eye- patterns. The cables, interface circuits, and components, such as transformers, and their effect on the quality and reliability of transmission are reviewed. The tutorial includes discussion of balanced to unbalanced conversion, shielding, transient protection and EMI susceptibility and compliance.

Finally, we suggest modifications to the standards, for instance frame sync, rise times, bandwidths, eye pattern masks, and reference designs.

Workshop 2 **Saturday, May 20**
10:45 – 11:45 **Berlin-A**

MASTERING WORKFLOWS

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

Panelists: *Eric Boulanger*, The Bakery, Los Angeles, CA, USA
Marc Ebermann, Music Factory
Mandy Parnell, Black Saloon, UK
Michael Romanowski, Coast Mastering, Bay Area, CA, USA

There are various workflows that come into play in mastering audio that range from self-mastering-while-mixing to traditional mastering in a proper mastering facility. Other variations involve working inside the box entirely vs using outboard gear, or a combination of the two. This event will represent various workflows in order to compare and contrast them, and highlight the strengths of each.

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

Pro Sound Expo **Saturday, May 20**
11:15 – 12:00 **PSE Stage**

THE INS AND OUTS OF MICROPHONES

Presenter: **John Willett**

Microphones are the very first link in the recording chain, so it's important to understand them to use them effectively. This presentation will explain the differences between different types of microphones; explain polar-patterns and directivity, proximity effect relative recording distances and a little about room acoustics. Many of these "golden nuggets" helped me greatly when I first understood them and I hope they will help you too.

Tutorial 6 **Saturday, May 20**
12:00 – 12:45 **Salon 4+5 London**

THE SURPRISING ORIGINS OF DIGITAL AUDIO: SOME PALE GLEAMS FROM THE PAST

Presenter: **Jon D. Paul**, Vice President Scientific Conversion, Inc. , Founder, the Crypto-Museum

This tutorial explores the history of digital audio technology, and the contributions of the great inventors that led to modern digital media. Signal analysis began in 1822 with Fourier's transform. Audio compression started in 1938, with the VOCODER, Homer Dudley's landmark speech encoder. Dudley invented electronic speech analysis and synthesis, and achieved ten times speech compression.

The onset of World War II created an urgent need for speech encryption of strategic conferences via short wave radio links. In just six months, an unbreakable top secret speech scrambler SIGSALY, was designed at Bell Telephone Laboratories (BTL). Brilliant engineers including Claude Shannon, Henry Nyquist, and Homer Dudley, invented and implemented eleven funda- mental breakthroughs, including PCM, flash A/D conversion and spread spectrum.

The VOCODER and SIGSALY block diagrams, construction and operation are discussed, with their close links to modern audio codecs.

We describe the interesting background of the engineers, inventors and mathematicians who laid the foundations of digital audio; Dudley and Shannon's work on SIGSALY, Hedy Lamarr's spread spectrum, and Turing's Delilah scrambler. A reconstruction of the very first PCM codec ADC, using vintage tubes is described. This tutorial includes unusual photos, vintage music, early VOCODER speech and very rare SIGSALY decrypted speech.

Pro Sound Expo **Saturday, May 20**
12:15 – 13:00 **PSE Stage**

PRO TOOLS IN THE CLOUD

Presenter: **Dave Tyler**, Audio Application Specialist Manager, AVID

Dave Tyler looks at how cloud enabled workflows are already benefitting Pro Tools users and the possible future applications of cloud technology in digital media applications.

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Saturday, May 20, 13:00 – 14:30 **Berlin-A**

Opening Remarks: • Executive Director Bob Moses
• President Alex Case
Convention Chairs Sascha Spors & Nadja Wallaszkovitzs
Program: • AES Awards Presentation by John Krivit, Past President
• Introduction of Keynote Speaker
• Keynote Address by Alex Arteaga

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

BOARD OF GOVERNORS AWARD
• Philip Jackson
• Michael Williams

FELLOWSHIP AWARD
• Tapio Lokki (Juha Backman accepting)

DISTINGUISHED SERVICE AWARD
• Mark Yonge

SILVER MEDAL AWARD
• Wolfgang Klippel

Keynote Speaker

This year's Keynote Speaker is Dr. Alex Arteaga. Alex Arteaga's research integrates aesthetic and philosophical practices relating to aesthetics, the emergence of sense, meaning and knowledge, and the relationships between aurality, architecture and the environment through phenomenological and enactivist approaches. He studied piano, music theory, composition, electroacoustic music, and architecture in Berlin and Barcelona and received a Ph.D. in philosophy from the Humboldt University. After being an academic researcher at the Collegium for the Advanced Study of Picture Act and Embodiment at the Humboldt University he developed his own research projects at the Berlin University of the Arts among which Architecture of Embodiment as Einstein Junior Fellow. He currently heads the Auditory Architecture Research Unit and the Department of Auditory Architecture in the MA Sound Studies and Sonic Arts at the Berlin University of the Arts and is professor for contemporary philosophy and artistic research at the Research Master in Art and Design at EINA / Universitat Autonoma de Barcelona. The title of his address is "*Auditory Architecture: Environment, Sense, and Aurality*."

How do our environments emerge when we focus our activity on hearing and listening? How do these modalities of action condition the ongoing process of sense-making? Which practices and methods of research are appropriate to address these questions in a non-reductive way? In this lecture Alex Arteaga will present the conceptual framework and the main practices developed at the Auditory Architecture Research Unit (Berlin University of the Arts) for the research and design of "Klangumwelten": the "surrounding-aural-worlds."

Tutorial 7 **Saturday, May 20**
14:30 – 15:30 **Salon 4+5 London**

DIGITAL STREAMING: IMPLICATIONS FOR MIX AND MASTERING PRACTICE

Presenter: **Paul Tapper**

Streamed audio consumption is becoming an increasingly dominant factor in the music industry. This not only profoundly effects the music business, but also has deep implications for the creative choices made during mixing and mastering. Two major technical considerations are codec compression, and loudness normalization. Codecs produce a better result when the source audio does not have inter-sample clips. Loudness normalization dramatically effects the creative goals of mixing and mastering and the correct auditioning methodology. Producing a "hotter" louder master may well sound better in the studio but will just get attenuated during streaming so the consumer will not hear that increased loudness. Often-times head room and audio quality is sacrificed to achieve this greater studio loudness. These dramatic changes in music consumption patterns necessitate dramatic changes in mixing and mastering practices by audio professionals.

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

Pro Sound Expo **Saturday, May 20**
14:30 – 15:15 **PSE Stage**

GENELEC SMART ACTIVE MONITORS

Presenter: **Thomas Lund**, Genelec

Built for today's fast-paced studio environment, Genelec Smart Active Monitors (SAM) are designed to meet your workflow and help you improve your sound reproduction quality. As space becomes more limited, listening areas are more confined and room acoustic problems more prevalent.

SAM Systems draw on Genelec's decades of experience and expertise to create an intelligent, flexible network of monitors and subwoofers that can adapt to your requirements. Use Genelec

Loudspeaker Manager (GLMTM) 2.0 software to create monitoring systems ranging from traditional stereo to immersive audio setups relying on the proprietary power of GLM AutoCalTM to create an optimal monitoring environment.

Saturday, May 20 14:30 Salon 16 Riga

Standards Committee Meeting SC-02-02 Digital Input/Output Interfacing

Session P4 Saturday, May 20
14:45 – 16:15 Salon 1 Moscow

SPATIAL AUDIO—BINAURAL 2

Chair: **Ramona Bomhardt**, RWTH Aachen University, Aachen, Germany

14:45

P4-1 Comparison of Spatial Characteristics of Head-Related Transfer Functions between the Horizontal and Median Planes—*Xiaoli Zhong, Bosun Xie, Guangzheng Yu*, South China University of Technology, Guangzhou, China

Head-related transfer functions (HRTFs) vary with source direction and thus contain major localization cues. In this work, the directional variation of HRTFs in the horizontal and median planes are studied using directional Fourier expansion. Results indicate that up to 20 kHz, the preceding 8 or 9 order elevation harmonics account for 99% variation of HRTFs in the median plane; while the preceding 31 or 32 order azimuthal harmonics account for 99% variation of HRTFs in the horizontal plane. Therefore, the horizontal-plane variation of HRTFs is more complicated in comparison to the median-plane variation. Moreover, the front-back spatial symmetry calculated from expansion weights is compared between the horizontal and median planes.
Convention Paper 9710

15:15

P4-2 An Experiment to Evaluate the Performance of a Parametric Model for the Individualization of the HRTF in the Median Plane—*Pablo Gutierrez-Parera, Jose J. Lopez*, Universitat Politècnica de València, Valencia, Spain

Individualized HRTFs for headphone reproduction provide better immersion and natural localization of sounds than non-individualized, especially for elevated positions. This paper presents an experiment to determine the accuracy of individualized parametric modeled HRTFs. The modeling of the HRTFs was done with an algorithm that detects the main peaks and notches of the HRTF and models them with a chain of second order IIR peak filters. A subjective test was carried on to compare the perception and localization of the modeled versus measured HRTFs in the median plane. The analyzed data shows that simplified modeled versions of HRTFs with a few peaks simulated can obtain similar results than measured HRTF for elevation angles in the median plane.
Convention Paper 9711

15:45

P4-3 The Influence of Symmetrical Human Ears on the Front-Back Confusion—*Ramona Bomhardt, Janina Fels*, RWTH Aachen University, Aachen, Germany

Human beings have two ears to localize sound sources. At a first glance, the dimensions of the right and left ears are generally very similar. Nevertheless, the individual anthropometric dimensions and shape of both ears are disparate. These differences improve localization on the cone of confusion where interaural differences do not exist. To determine the influence of asymmetric ears, individual HRTF data sets are analytically and subjectively compared with their mirrored versions.
Convention Paper 9712

Session P5 Saturday, May 20
14:45 – 16:15 Salon 2+3 Rome

TRANSDUCERS 2

Chair: **Steve Temme**, Listen Inc., Boston, MA, USA

14:45

P5-1 Quantifying Consistency in Loudspeaker System Production—*Andrew Goldberg,* Georg Neumann GmbH, Berlin, Germany

This paper defines a new metric for production consistency of loudspeaker systems. The motivation is to improve stereo imaging of phantom sources. The measurement system’s consistency and three different loudspeaker designs are analyzed as examples to show how the metric can highlight inconsistencies. The analysis and summary metrics clearly highlighted (the artificially induced) inconsistencies. A simple classification system (from A to H) for loudspeakers is also proposed. It is hoped that this, or a similar, metric will be integrated into international standards as a way to help designers improve their products, production to improve their processes, and customers to better inform their purchase decisions.
Convention Paper 9713

15:15

P5-2 Loudspeakers Performance Variance Due to Components and Assembly Process—*Maria Costanza Bellini, Angelo Farina*, University of Parma, Parma, Italy

This paper presents an experimental study of the main causes of scrap during the production of a typical midrange loudspeaker. After analyzing the most critical components of a transducer, various samples with reference and defected components have been built and characterized in terms of frequency response and distortion. In addition, a second set of samples has been built using reference components but varying the assembly process parameters; these samples also have been characterized as the previous ones. Measurements have been performed both in an anechoic chamber and in a real production line and, by the analysis of acquired data, the authors have individuated the most influential components and assembly parameters in terms of required performance.
Convention Paper 9714

15:45

P5-3 Evaluation of Audio Test Methods and Measurements for End-of-the-Line Automotive Loudspeaker Quality Control—*Steve Temme,*[1] *Viktor Dobos*[2]
[1]Listen, Inc., Boston, MA, USA
[2]Harman/Becker Automotive Systems Kft., Székesfehérvár, Hungary

In order to minimize costly warranty repairs, automotive

manufacturers impose tight specifications and quality/reliability requirements on their part suppliers. At the same time, they also require low prices. This makes it important for automotive manufacturers to work with parts suppliers to define reasonable specifications and tolerances, and to understand both how the parts suppliers are testing and also how to carry out their own measurements for incoming QC purposes. Specifying and testing automotive loudspeakers can be very tricky since loudspeakers are inherently nonlinear, time-variant and effected by their working conditions and environment. This paper examines the loudspeaker characteristics that can be measured and discusses common pitfalls and how to avoid them on a loudspeaker production line. Several different audio test methods and measurements for end-of-the-line automotive speaker quality control are evaluated, and the most relevant ones identified. Speed, statistics, and full traceability are also discussed.
Convention Paper 9715

Workshop 3 **Saturday, May 20**
14:45 – 16:15 **Salon 7 Vienna**

PERCEPTION OF TEMPORAL RESPONSE AND RESOLUTION IN TIME DOMAIN

Chair: **Mike Turner**

Panelists: *David Griesinger*
Menno van der Veen
Hans van Maanen

Time response and temporal resolution are controversial topics in sound reproduction. In this workshop a theoretical analysis will be accompanied by the results of listening experiences on the lower end of the audio range and at the higher end. The background is that the temporal relation between the different tones, which make up complex sounds like attacks, need to be conserved to create a viable reproduction of the original sound. Many systems do not preserve a correct time relation (e.g., cross-over filters in loudspeakers, basereflex systems, and reconstruction filters in digital audio). The aim is to identify the audibility of such timing errors.

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

Special Event
BERLIN: CENTER OF ELECTRONIC MUSIC PRODUCTION, MIXING, SOUND DESIGN, AND THE COMMUNITY
Saturday, May 20, 14:45 – 16:15 **Berlin-A**

Moderator: **André Maletz**, Mixing Ambulance, consultant, Cologne/Berlin, Germany

Presenters: *David Miles Huber*, Artist, Mixer, Producer, Seattle/Berlin
Jan-Michael Kühn, aka DJ Fresh Meat, Author, DJ/Producer, Berlin, Germany
Richard Roloff, aka DJ Dickey, DJ, Mixer, Producer, Berlin, Germany
Andreas Schneider, Schneidersladen/Superbooth, Berlin, Germany
Brian Smith, Ableton User Group Berlin, Founder of big brain audio, Berlin, Germany

Since the eighties Berlin has grown to the center of electronic music, it developed to a vivid environment with clubs, studios, festivals, communities, companies, and much more, where artists (in general), musicians, DJs, and also music business people want to be part of.

The event will highlight aspects of this scene and their relevance and impact on professional audio, the panelist will discuss and present their view on this as authentic experts from different view angles.

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

Session P6 **Saturday, May 20**
15:00 – 18:00 **Gallery Window**

POSTERS: ANALYSIS, CODING, AND HEARING

15:00

P6-1 Determining Pronunciation Differences in English Allophones Utilizing Audio Signal Parameterization—*Bozena Kostek, Magdalena Piotrowska, T. Ciszewski, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

An allophonic description of English plosive consonants, based on audio-visual recordings of 600 specially selected words, was developed. First, several speakers were recorded while reading words from a teleprompter. Then, every word was played back from the previously recorded sample read by a phonology expert and each examined speaker repeated a particular word trying to imitate correct pronunciation. The next step consisted in partitioning by editing two recorded sets of words into allophones, then signals were analyzed and subsequently audio excerpts were parametrized. The comparison of two sets of allophones was reinforced by the phonology expert's assessment of produced speech sounds. Analyses presented in this paper allowed for determining a set of parameters describing an allophone pronunciation.
Convention Paper 9716

15:00

P6-2 Mathematical Model of the Acoustic Signal Generated by the Combustion Engine—*Michal Luczynski, Stefan Brachmanski*, Wroclaw University of Science and Technology, Wroclaw, Poland

Development of the technology of electric vehicles is progressing. The acoustic problem of no audible sensation like combustion engine is a significant problem. It is a desirable feature and safety problem as well. More and more automotive companies are working on engine sound synthesis systems. However there is no universal standard helping solve this problem. The aim of this paper is to define a universal form of mathematical model and general assumption to design engine sound synthesis systems for electric motorbikes.
Convention Paper 9717

15:00

P6-3 Whispered Speech Speaker Recognition. Listening Tests versus Speaker Recognition System—*Krzysztof Goliasz, Michal Luczynski*, Wroclaw University of Science and Technology, Wroclaw, Poland

In this study, efficiency of different methods of whispered speech speaker recognition have been compared. Whispered speech as one of the voice disguises is causing a lot of difficulties in speaker identification. During examination of both methods authors were looking for key factors of speaker recognition in both methods. Both methods were tested with and without normal voice as a reference

sample. Non-acoustic circumstances as time and costs were considered during comparison as well.
Convention Paper 9718

15:00

P6-4 A Study on Audio Signal Processed by "Instant Mastering" Services—*Magdalena Piotrowska,*[1] *Szymon Piotrowski,*[2] *Bozena Kostek*[1]
[1]Wroclaw University of Science and Technology, Wroclaw, Poland
[2]Learn How To Sound, Krakow, Poland

An increasing amount of music produced in home- and project-studios results in development and growth of "automatic mastering services." The presented investigation explores changes introduced to audio signal by various online mastering platforms. A music set consisting of 10 songs produced in small facilities was processed by eight on-line automatic mastering services. Additionally, some laboratory-constructed signals were tested. To determine, whether changes introduced to audio are invariable between trials, every music excerpt was submitted several times. For each sample, parameters related to music characteristics such as timbre, dynamics, and loudness were calculated before and after processing. Results obtained enable to discover some of the mechanisms underlying tested automatic mastering services as well as discern similarities and differences between various platforms.
Convention Paper 9719

15:00

P6-5 Car Infotainment Systems Capabilities vs. Customers' Needs and Expectations—*Bartlomiej Kukulski*, AGH University of Science and Technology, Krakow, Poland

To make the trip more enjoyable, decades ago vehicles gained radio systems, which evolved further into multimedia systems, and today into infotainment systems, which provide entertainment facilities, access to information (e.g., route guidance), and increasingly, network connectivity, which leads to the perception of those systems as smart devices. The paper presents an overview of modern car infotainment systems possibilities compiled with the results of survey conducted on representative group of car users, mainly drivers. The purpose of the survey was to obtain information about possessed car audio systems (loudspeakers, amplifiers), hardware and software capabilities of possessed infotainment systems, and to learn about user experience followed by satisfaction evaluation of quality of audio provided by systems installed in their cars.
Convention Paper 9804

Session EB1 **Saturday, May 20**
15:00 – 18:00 **Gallery Window**

POSTERS: ANALYSIS, CODING, AND HEARING

15:00

EB1-1 Source Separation in Action: Demixing the Beatles at the Hollywood Bowl—*James Clarke*, Abbey Road Studios, London, UK

Except for unamplified dramatic performances or concerts where the audience is either used to sitting quietly or can be persuaded to listen quietly, high levels of amplifications are required so that the speech and music can still be heard above the self-noise generated by the crowd. Crowd noise can reach surprisingly high sound levels when several thousand people are shouting at each other, all at the same time. In this paper a method is presented that was used to isolate the crowd noise non destructively, so the raw instrumentation can be targeted in isolation. These isolated sources now become available for re-mixing and balancing.
Engineering Brief 307

15:00

EB1-2 A 3D Sound Localization System Using Two Side Feature Selection for Real-Time Acoustic Drone Detection Using Genetic Algorithms—*Joaquín García-Gómez, Marta Bautista-Durán, Roberto Gil-Pita, Manuel Rosa-Zurera*, University of Alcala, Alcalá de Henares, Spain

Drones are taking off in a big way, but people sometimes use them in order to invade the privacy of others or to bypass the security systems, making their detection an actual issue. The objective of the proposed system is to design real-time acoustic drone detectors, able to distinguish them from objects that can be acoustically similar. A set of features related to the propeller sounds have been extracted, and genetic algorithms have been used to select the best subset. The classification error achieved with 30 features is below 13%, making feasible the real-time implementation of the proposed system.
Engineering Brief 308
eBrief presented by Marta Bautista-Durán

Student and Career Development Event
STUDENT RECORDING CRITIQUES
Saturday, May 20, 15:00 – 16:00
Salon 11 (Genelec Demo Room)

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

Students! Come and get tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system, and then receive feedback from a panel of renowned industry professionals. Students at any of their studies can sign up to participate. Students should sign up at the student (SDA) booth immediately on arrival at the convention, and deliver stereo 44.1 Khz, 24 bit AIFF or WAVE files to the SDA booth at that time. THE FIRST SESSION IS BEFORE THE FIRST SDA MEETING so come early and prepared! Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by Genelec.

Saturday, May 20 **15:00** **Salon 15 Paris**

Technical Committee Meeting on Audio for Cinema

Tutorial 8 **Saturday, May 20**
15:30 – 16:30 **Salon 4+5 London**

CREATING MPEG-H CONTENT WITH COMMON DAWS

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

MPEG-H is still the TV broadcast standard in Korea. Major end user device manufacturers like Samsung and LG are based there. So it's very likely that MPEG-H will be available soon on a lot of devices in the market worldwide which will likely increase the demand for MPEG-H content. The Tutorial will show how to create, edit and export content for that application quick and easy in common production workflows with any DAW using the Spatial Audio Designer.

Pro Sound Expo **Saturday, May 20**
15:30 – 16:15 **PSE Stage**

RIBBON MICS

Presenter: **Sammy Rothman**, AEA

What are ribbon microphones and how do you use them? Sammy Rothman answers these questions and more in "AEA Ribbon Mics: Fix It In the Mic" which delves into all things ribbons including best miking practices, how ribbons mics work, the differences between ribbons and condensers, and an exploration of AEA's diverse line of active and passive ribbon microphones.

Saturday, May 20 **16:00** **Salon 15 Paris**

Technical Committee Meeting on Audio for Games

Session P7 **Saturday, May 20**
16:30 – 17:30 **Salon 1 Moscow**

SPATIAL AUDIO—BINAURAL 3 AND OBJECTS

Chair: **Sascha Spors**, University of Rostock, Rostock, Germany

16:30

P7-1 **Augmented Reality to Improve Orchestra Conductors' Headphone Monitoring**—*Dimitri Soudoplatoff,*[1] *Amandine Pras*[1,2]
[1] Conservatoire National Supérieur de Musique et de Danse de Paris (CNSMDP), Paris, France
[2] University of Lethbridge, Lethbridge, AB, Canada

Conductors face challenges when conducting the orchestra with headphones to synchronize with a soundtrack or a click track. We sent a survey to 12 international conductors to identify and classify those challenges. They primarily reported on balance issues, aggressive click tracks, and the difficulty of hearing the acoustic sound of the orchestra, leading 70% of them to remove one ear out of the headphones. A solution using augmented reality monitoring through binaural rendering and head tracking was tested in various situations and showed that it could successfully reproduce the acoustic sound of the orchestra into the headphones. Another perceptual experiment evaluated the potential of realism of this solution when merging two binaural auditory scenes recorded in the same acoustic space together. Results encourage us to further develop immersive monitoring systems for conductors, with the soundtrack integrated in the real acoustic space.
Convention Paper 9720

Convention Paper 9721 was withdrawn

17:00

P7-2 **Wave Field Synthesis Driving Functions for Large-Scale Sound Reinforcement Using Line Source Arrays**—*Frank Schultz,*[1] *Gergely Firtha,*[2] *Peter Fiala,*[2] *Sascha Spors*[3]
[1]Technical University of Berlin, Berlin, Germany
[2]Budapest University of Technology and Economics, Budapest, Hungary
[3]University of Rostock, Rostock, Germany

Wave field synthesis (WFS) can be used for wavefront shaping using line source arrays (LSAs) in large-scale sound reinforcement. For that the individual drivers might be electronically controlled by WFS driving functions of a virtual directional point source. From the recently introduced unified 2.5D WFS framework it is known that positions of amplitude correct synthesis (PCS) only exist along an arbitrary shaped curve—the reference curve—in front of the LSA. However, its shape can be adapted with the so called referencing function. We introduce the adaption of the referencing function along the audience line of typical concert venues for optimized wavefront shaping. This yields considerable improvements with respect to sound field's homogeneity and more convenient setups compared to previous WFS-based sound reinforcement.
Convention Paper 9722

Session P8 **Saturday, May 20**
16:30 – 18:00 **Salon 2+3 Rome**

TRANSDUCERS 3

Chair: **Finn T. Agerkvist,** Technical University of Denmark, Kgs. Lygby, Denmark

16:30

P8-1 **Position Dependence of Fractional Derivative Models for Loudspeaker Voice Coils with Lossy Inductance**—*Alexander W. King, Finn T. Agerkvist*, Technical University of Denmark, Kgs. Lygby, Denmark

Commonly used models of moving-coil loudspeaker voice coils, which include effects from eddy current losses, are either inaccurate or contain an abundance of parameters and are difficult to extend to the nonlinear domain. On the contrary, fractional derivative models accurately describe the frequency and position dependence of the lossy inductance, with meaningful connections to the underlying physics, while keeping the number of parameters low. These fractional derivatives are also compatible with state-space polynomial methods of modeling nonlinear behavior. It is shown that the fractional order derivative approaches a value of 1, corresponding to an ideal inductance, when the voice coil is completely outside the magnetic system. Finally, the developed model reveals details about the effect of conductive voice coil formers.
Convention Paper 9723

17:00

P8-2 **Model for Evaluation of Power Consumption of Vented Box Loudspeakers**—*Filip Sommer Madsen, Søren Thorsen, Niels Elkjær Iversen, Arnold Knott,* Technical University of Denmark, Kgs. Lygby, Denmark

In the design of mobile sound systems an estimation of power consumption must be made in order to choose a battery of appropriate size and cost. However poor methods for power estimation tend to result in large and costly batteries. This paper aims to present a more precise method for estimating power consumption for a vented box sound system. Instead of simplifying a loudspeaker system as a purely ohmic resistance, its mechanical and acoustic parameters are used to create a state space model. Despite deviations at high frequencies, the state space model is at

least twice as accurate at estimating the power consumption than simplifying the speaker as a resistor.
Convention Paper 9724

17:30

P8-3 Assessment of the Radiation Mode Method for In Situ Measurements of Loudspeaker Systems—*Maryna Sanalatii,*[1,2,3] *Lucas Vindrola,*[2] *Clément Vasseur,*[2] *Philippe Herzog,*[1] *Manuel Melon,*[2] *Régine Guillermin,*[1] *Nicolas Poulain,*[3] *Jean-Christophe Le Roux*[3]
[1]Laboratoire de Mécanique et d'Acoustique (LMA), Marseille, France
[2]Laboratoire d'Acoustique de l'Université du Maine (LAUM), Le Mans cedex 9, France
[3]Centre de Transfert de Technologie du Mans (CTTM), Le Mans, France

In this paper the radiation mode method is used to measure the frequency response and the directivity pattern of loudspeaker systems. This method, which has been successfully applied to speaker measurement in free field conditions, is now tested in a large non-anechoic hall. Two closed box systems and a switchable bi-directive/cardioid subwoofer have been used. Each system is measured first in an anechoic chamber and then in the large hall. The radiation method is then applied to the two different measurement data sets. Results show a good agreement between both conditions. Finally, the influence of the mesh coarsening is studied.
Convention Paper 9725

Tutorial 9 — Saturday, May 20
16:30 – 18:00 — Salon 7 Vienna

PLANNING AND DEPLOYING THE AUDIO AND INTERCOM SYSTEMS FOR THE RIO 2016 OLYMPIC AND PARALYMPIC GAMES COMPETITION VENUES

Presenter: **Rosalfonso Bortoni**

The tutorial will present the "behind the scenes" of the audio and intercom systems planning and deployment for the Rio 2016 Games competition venues, analyzed from the technical, practical, feasible, and political perspectives. There were 39 competition venues in total, each one with its own dedicated FOH and BOH systems designed to fulfill the needs of the Sports, Results, Broadcast, Press, and overall operations. The close relationship between the audio and intercom systems became an elegant and efficient network, even carrying together data and video signals. The impact of the physical and acoustical aspects of the venues on the project will be discussed as well.

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

Workshop 4 — Saturday, May 20
16:30 – 18:00 — Salon 4+5 London

NEW DEVELOPMENTS IN STUDIO METADATA

Chair: **Paul Jessop**, County Analytics Ltd.

Panelists: *Julio Cesar Alvarez Fernandez*, SCAPR
Niels Rump, digital Data Exchange
John Sarappo, VeVa Sound
Konstantin Vogel, GVL

The importance of getting accurate metadata out of the studio (whatever that means these days) and into the delivery chain (and numerous other places) has never been greater. As streaming becomes dominant, performers will rely more and more on performance royalties and that requires accurate performer information. As consumers get more sophisticated in using streaming services (and others) they demand accurate metadata for discovery. How do we deliver this? The workshop will cover initiatives in DDEX (the recording industry messaging standards body) and elsewhere to make this happen.

Student and Career Development Event
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Saturday, May 20, 16:30– 18:00 — Berlin-A

Presenters: **Michael France**, SDA Officer
Mitchell Graham, SDA Officer
Dave Moffatt, SDA Officer
Angelika Podola, SDA Officer

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

All students and educators are invited to participate in this meeting. Election results and Recording Competition and Design Competition Awards will be given at the Student Delegate Assembly Meeting—Part 2 on Tuesday, May 23.

Pro Sound Expo — Saturday, May 20
16:30 – 17:15 — PSE Stage

SIGMASTUDIO AND CURRENT DSP HW ARCHITECTURES FOR AUDIO APPLICATIONS

Presenter: **Miguel Chavez**, ADI

Graphical Audio DSP Programming Environments and newer "audio specific" DSP programming SW and HW architectures. How has SigmaStudio revolutionized how audio equipment is designed in today's competitive environment? What has been ADI (and others) response has been to new evolving customer needs from a software and hardware perspective? How have new algorithms emerged with the existence of different sensors and converters? How have algorithms evolved with the need for system-performance improvements requirements? What are the latency considerations within different audio applications: "headphones," "stage," "studio" among other listening environments?

Saturday, May 20 — 16:30 — Salon 16 Riga

Standards Committee Meeting SC-02-08 Audio File Tranfer and Exchange

Saturday, May 20 — 17:00 — Salon 15 Paris

Technical Committee Meeting on Network Audio Systems

Saturday, May 20 18:00 Salon 15 Paris

Technical Committee Meeting on Recording and Technology Practices

Session P9 Sunday, May 21
9:00 – 12:30 Salon 1 Moscow

SPATIAL AUDIO—AMBISONICS

Chair: **Franz Zotte**, University of Music nd Performing Arts Graz, Graz, Austria

9:00

P9-1 The Median-Plane Summing Localization in Ambisonics Reproduction—*Bosun Xie, Haiming Mai, Xiaoli Zhong,* South China University of Technology, Guangzhou, China

One aim of Ambisonics reproduction is to recreate the perception of virtual source in arbitrary directions. Practical Ambisonics reproduction is unable to recreate correct high-frequency spectra in binaural pressures that are known as front-back and vertical localization cue. Based on both a simple shadowless head model and KEMAR's HRTFs, the present work proves that the changes of ITD caused by head turning in Ambisonics match with these of a real source, and thus provide dynamic cue for vertical localization, especially in the median plane. In addition, the low-frequency virtual source direction can be approximately evaluated by using a set of localization equations or panning laws. The above analysis is validated by a median-plane virtual source localization experiment with Ambisonics reproduction.
Convention Paper 9726

9:30

P9-2 Exploring the Perceptual Sweet Area in Ambisonics—*Matthias Frank, Franz Zotter*, University of Music and Performing Arts Graz, Graz, Austria

A physical pressure-matching criterion to describe the size of the sweet area in Ambisonics does not quite match the generously large sweet area encountered in practice. To satisfy the need of a more practical characterization this paper comes up with a simple and systematic method to experimentally determine the perceptual sweet area. The method is not limited to assessing the localization of dry sounds as it also perm its plausibility assessment of more complex scenes. This contribution exemplarily presents results for playback of two different audio scenes (dry/reverberant) using different Ambisonic orders.
Convention Paper 9727

10:00

P9-3 Phantom Source Widening by Filtered Sound Objects—*Franz Zotter, Matthias Frank*, University of Music and Performing Arts Graz, Graz, Austria

Audio effects increasing the perceived source extent (width/distance) often employ frequency-dependent panning of a single virtual sound object or real-time-controlled design of stochastic multichannel filters. Both ways imply increased complexity required in the renderer or object representation. In this paper we present a frequency-dependent panning scheme to obtain constellations of 3, 4, 5, or 7 filtered sound objects, as a simplified object-based description of wide/distant sound for any renderer. We deal with the multichannel filter-design questions: Are the filters rather temporally compact or frequency-selective, zero-phase FIR vs. IIR or causal-sided FIR, how strictly power-complementary? By results of a listening experiment for selected examples, we can provide some answers and an effective design of useful width-/distance-increasing filtered sound objects.
Convention Paper 9728

10:30

P9-4 Ambisonic Spatial Blur—*Thibaut Carpentier*, IRCAM, Paris, France

This paper presents a technique for controlling the spatial resolution of an Ambisonic sound field while preserving its overall energy. The proposed method allows to transform a stream encoded in *N*-order Ambisonic to a lower order resolution. The transformation can be continuously operated, indeed simulating fractional order representation of the Ambisonic stream and varying the "bluriness" of the spatial image.
Convention Paper 9729

11:00

P9-5 Comparing Ambisonic Microphones—Part 2—*Enda Bates,*[1] *Sean Dooney,*[1] *Marcin Gorzel,*[2] *Hugh O'Dwyer,*[1] *Luke Ferguson,*[1] *Francis M. Boland*[1,2]
[1] Trinity College Dublin, Dublin, Ireland
[2] Google, Dublin, Ireland

This paper presents some further experiments devised to assess the performance of an expanded number of commercially available Ambisonic microphones. The subjective timbral and spatial quality of five microphones (Soundfield MKV, Core Sound TetraMic, Sennheiser Ambeo, MH Acoustics Eigenmike, and Zoom H2N) is assessed using listening tests and a recording of an acoustic quartet. Localization accuracy is assessed using an objective directional analysis and recordings from a spherical loudspeaker array. Intensity vectors are extracted from 25 critical frequency bands of the Bark scale and used to compute the angle to the source location. Significant differences were found between microphones with the Soundfield MKV and Eigenmike producing the best results in terms of timbral quality and localization respectively.
Convention Paper 9730

11:30

P9-6 Object-Based Reverberation Encoding from First-Order Ambisonic RIRs—*Philip Coleman,*[1] *Andreas Franck,*[2] *Dylan Menzies,*[2] *Philip J. B. Jackson*[1]
[1]University of Surrey, Guildford, Surrey, UK
[2]University of Southampton, Southampton, UK

Recent work on a reverberant spatial audio object (RSAO) encoded spatial room impulse responses (RIRs) as object-based metadata that can be synthesized in an object-based renderer. Encoding reverberation into metadata presents new opportunities for end users to interact with and personalize reverberant content. The RSAO models an RIR as a set of early reflections together with a late reverberation filter. Previous work to encode the RSAO parameters was based on recordings made with a dense array of omnidirectional microphones. This paper describes RSAO parameterization from first-order Ambisonic (B-Format) RIRs, making the RSAO compatible with

existing spatial reverb libraries. The object-based implementation achieves reverberation time, early decay time, clarity, and interaural cross-correlation similar to direct Ambisonic rendering of 13 test RIRs.
Convention Paper 9731

12:00

P9-7 Further Investigations on the Design of Radial Filters for the Driving Functions of Near-Field Compensated Higher-Order Ambisonics—*Nara Hahn, Sascha Spors*, University of Rostock, Rostock, Germany

Analytic driving functions for Near-field Compensated Higher-order Ambisonics (NFC-HOA) are derived based on the spherical harmonics expansions of the desired sound field and the Green's function that models the secondary sources. In the frequency domain, the radial part of the driving function is given as spherical Hankel functions and compensates the near-field effects of the secondary sources. By exploiting the polynomial expansion of the spherical Hankel functions, the radial filters can be implemented as cascaded biquad filters in the time domain, thereby reducing the computational complexity significantly. In this paper three practical issues regarding the design of the radial filters are addressed: pole-zero computation, pole-zero mapping, and gain normalization. Useful suggestions are given for improvements in terms of stability and numerical stability, which are demonstrated by numerical simulations.
Convention Paper 9732

Session P10 Sunday, May 21
9:00 – 11:30 Salon 2+3 Rome

AUDIO SYSTEMS, DESIGN, AMPLIFIERS

Chair: **Jamie Angus**, University of Salford, Salford, UK

9:00

P10-1 Supply Voltage Scaling Technique of Triode Tube Based on Harmonic Distortion Characteristics—*Kanako Takemoto, Shiori Oshimo, Toshihiko Hamasaki*, Hiroshima Institute of Technology, Hiroshima, Japan

In recent years, applying a triode tube to the guitar pedal as a signal modulator has been attracting considerable attention. The miniature tube 12AX7/ECC83 is still the key component for creating main sound properties of a guitar amplifier owing to its specific non-linear characteristics. Therefore, the attempts to incorporate the tube harmonic distortion sound to the pedal instead of solid-state circuit came out in the market. This paper describes a technique to generate the same harmonic spectrum focusing tube Crunch sound at 150V power supply voltage, which is half of conventional 300V. This voltage scaling contributes to reducing the battery power consumption of the tube guitar pedal to prolong the playing time.
Convention Paper 9733

9:30

P10-2 Linear Phase Crosstalk Cancellation Filters—*Arnaud Reymond,[1,2] Christof Faller, [1,3] Daniel Weiss[4]*
[1]EPFL Lausanne, Switzerland
[2]Sony Stuttgard Technology Center, Switzerland
[3]Illusonic GmbH, Uster, Switzerland
[4]Weis Engineering Ltd., Switzerland

The filter design approach presented here manages an attractive compromise that produces filters with good crosstalk cancellation, short impulse responses, and a linear phase. As a starting point, basic cancellation filters are considered, i.e., ideal pulses with delay and inversion. These are modified in the frequency domain with a gain applied so as to obtain a nearly spectrally flat total power at the ears. This modification allows to substantially reduce the coloration of the basic cancellation filters at the price of a small decrease of cancellation performance.
Convention Paper 9734

10:00

P10-3 GaN FETs Drive Fidelity and Efficiency in Class-D Audio Amplifiers—*Stephen Colino,[1] Skip Taylor[2]*
[1]Efficient Power Conversion Corp., Bear, DE, USA
[2]Elegant Audio Solutions, Austin, TX, USA

With the current maturity of Class-D audio amplifier architectures, amplifier fidelity and efficiency limitations are primarily at the device level. Silicon MOSFETs have been evolving for almost forty years, and their progress towards a perfect switch has slowed dramatically. There are some fundamental characteristics of MOSFETs that degrade sound quality and efficiency. In 2010, the enhancement mode Gallium nitride (GaN) power FET was introduced by Efficient Power Conversion (EPC), providing a large step towards the perfect switch.
Convention Paper 9735
Paper presented by Bhasy Nair

10:30

P10-4 Ultra Efficient Linear Amplifiers—*Jamie Angus*, University of Salford, Salford, UK

"Class-D" switching amplifiers are considered to be the most efficient amplifiers. However, designers must deal with supply rail, and radio frequency interference, and the need to switch power devices at high frequencies. Because of these, and other problems, not everyone wishes to use switching based technologies amplifiers. Unfortunately, linear amplifiers are significantly more inefficient than switching amplifiers under sine wave testing. However real audio signals spend much more time at low amplitudes than a sine wave. By changing the switch point for "Class-G" or "Class-H" they can have efficiencies that rival "Class-D" amplifiers producing the same output. The paper develops optimum switch points for both single and multiple switching points, with respect to the expected amplitude distribution of the audio.
Convention Paper 9736

11:00

P10-5 Evaluation of Audio Performance over Product Life—*Wolfgang Klippel*, Klippel GmbH, Dresden, Germany

Most measurements are performed on audio products during the development of the first prototype and at the end of the production line. Physical measurements and perceptional evaluation in the target application (e.g., car interior) are also required to define the target performance, to finish successfully the development and to evaluate the reliability and robustness of the product under the influence of climate, aging, and other external factors. This paper discusses evaluation techniques that are useful in the different phases of the product life cycle to generate a successful product that provides the maximum benefit to the end user.
Convention Paper 9737

Convention Paper 9738 was withdrawn

Workshop 5 **Sunday, May 21**
9:00 – 10:30 **Salon 7 Vienna**

BLUETOOTH MUSIC AND GAMING AUDIO IN THE CAR

Chair: **Jonny McClintock**, Qualcomm, San Diego, CA, USA

Wireless connectivity, in particular Bluetooth, enabling smart device owners to now be untethered. This session is designed to explore how the smart device can also be used for the car and ensure the occupants enjoy high quality audio streaming to the Infotainment system. It will also look at the potential of low latency applications for Rear Seat Entertainment and interactive gaming applications. The overall content will be relevant to both the consumer experience and the overall benefits to the automotive manufacturers.

Workshop 6 **Sunday, May21**
9:00 – 10:45 **Salon 4+5 London**

INTERCULTURAL INFLUENCES IN SUBJECTIVE AUDIO QUALITY EVALUATION

Chair: **Nadja Schinkel-Bielefeld**, Silvantos GmbH, Erlangen, Germany

Panelists: *Deborah Ebem*, University of Nigeria, Nsukka, Nigeria
Jan Holub, Czech Technical University, Prague, Czech Republic
Alexander Raake, TU Ilmenau, Germany
Qin Yili, Academy of Broadcast Planning SAPPRFT, Beijing, China

Listening tests, i.e., for the standardization of new coding technologies, are performed all over the world with listeners of different cultures rating items in various languages, including some they do not understand. However, it is not well understood how cultural differences of the listeners may affect these test results. In this workshop we want to discuss cultural influences in listening tests including: (a) the perception and judgement of audio quality; (b) the influence of native or foreign language material or music typical or atypical for the listener's culture; (c) how weighing of sound characteristics or artifacts correlates with cultural differences with respect to preference and audio quality; and (d) cultural dependencies of the use of scales and labels in listening tests.

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

Student and Career Development Event
EDUCATION FORUM: AUDIO EDUCATION—WHAT DOES THE FUTURE HOLD
Sunday, May 21, 9:00 – 10:30 **Berlin-A**

Moderators: **Nyssim Lefford, Kyle P. Snyder**

Panelists: *Thomas Bärtschi*
Sylvan Lambinet, Louis lumiere Film, Photography, and Sound Engineering School, Paris, France
Mandy Parnell
Tom Parnell
Joshua D. Reiss, Queen Mary University of London, London, UK

In this roundtable discussion featuring invited experts from across the continent, we will discuss the trajectory of audio education including challenges unique to both educators and students, suggestions for the future, and how employers expect audio education to adapt to their ever-changing needs.

Sunday, May 21 **9:00** **Salon 15 Paris**

Technical Committee Meeting on Broadcast and Online Delivery: AGOTTVS

Session EB2 **Sunday, May 21**
9:30 – 12:30 **Gallery Window**

POSTERS: SPATIAL AUDIO, ROOM, RECORDING, AND LISTENING

9:30

EB2-1 AKtools—An Open Software Toolbox for Signal Acquisition, Processing, and Inspection in Acoustics—*Fabian Brinkmann, Stefan Weinzierl*, Technical University of Berlin, Berlin, Germany

The acquisition, processing, and inspection of audio data plays a central role in the everyday practice of acousticians. However, these steps are commonly distributed among different and often closed software packages making it difficult to document this work. AKtools includes Matlab methods for audio playback and recording, as well as a versatile plotting tool for inspection of single/multichannel data acquired on spherical, and arbitrary spatial sampling grids. Functional blocks cover test signal generation (e.g., pulses, noise, and sweeps), spectral deconvolution, transfer function inversion using frequency dependent regularization, spherical harmonics transform and interpolation among others. Well documented demo scripts show the exemplary use of the main parts, with more detailed information in the description of each method. To foster reproducible research, AKtools is available under the open software European Union Public Licence (EUPL) allowing everyone to use, change, and redistribute it for any purpose: www.ak.tu-berlin.de/aktools.
Engineering Brief 309

9:30

EB2-2 Disagreement between STI and STIPA Measurements Due to High Level, Discrete Reflections—*Ross Hammond,*[1] *Peter Mapp,*[2] *Adam Hill*[1]
[1]University of Derby, Derby, UK
[2]Peter Mapp Associates, Colchester, UK

Objective measures of intelligibility, speech transmission index (STI), and speech transmission index for public address systems (STIPA) often form the basis for sound system verification. The reported work challenges the accuracy of both measures when encountering high level, discrete reflections. Tests were carried out in an anechoic environment with artificial reflections added between 0 and 500 ms. Discrepancies were found to occur above 80 ms due to synchronization between modulation frequencies and reflection arrival times. Differences between STI and STIPA of up to 0.1 were found to occur for the same delay condition. Results suggest STIPA should be avoided in acoustic environments where high level, discrete reflections occur after 80 ms and STI should only be used alongside other verification methods.
Engineering Brief 310

9:30

EB2-3 Soundscape Recording: Review of Approaches—*Katarzyna Sochaczewska, Pawel Malecki, Dorota Czopek, Jerzy Wiciak*, AGH University of Science and Technology, Krakow, Poland

In the soundscape analysis the collected data is crucial for possible relationships between the results of measurements of acoustic, psychoacoustic research results, and characteristics of the respondents. Such analysis shall verify that the physical characteristics of sound affect the subjective assessment. The article shows a review of commonly used approaches in soundscape recording both for analysis and archive purposes. Discussion on recording from one or several spots in the middle of the sound sources versus moving with the microphone towards or inside the acoustic environment is provided. Also, special attention is paid on traditional microphone techniques in sound engineering, binaural recordings, and sound field synthesis with spherical harmonics.
Engineering Brief 311

9:30

EB2-4 Ambience Recording for 3D Audio—*Marco Hanelt,*[1,2] *Andreas Ehret*[2]
[1]Ostbayerische Technische Hochschule Amberg-Weiden
[2]Dolby Germany, Nürnberg, Germany

3D audio is emerging as a production format to enable immersive consumer audio experiences, including the sensation of height. This eBrief focuses on the use of 3D microphone arrays for recording 3D ambiences. Multiple microphone arrays were evaluated, both in theory as well as in practice. The arrays were assessed for their perceptual performance such as spatial envelopment, location accuracy, and timbre. Furthermore, the practical usability of these recordings in a real world movie project and the handling in a common post-production environment have been tested. With the information gathered, a Best Practice guide for different use cases has been developed.
Engineering Brief 312

9:30

EB2-5 Do Microphone Angles Result in Audible Differences When Recording a Guitar Amplifier?—*Ellen Culloo, Malachy Ronan*, University of Limerick, Limerick, Ireland

Objective measurements using a sinusoidal sweep show that microphone angle has little effect on the frequency response of a guitar amplifier recording [1]. However, anecdotal evidence suggests that alterations to the microphone angle hold merit when recording ecologically valid sound sources. An ABX listening experiment was conducted with 20 participants to investigate whether microphone angles of 0, 30, and 60 degrees were audibly different to this cohort. Both dynamic and ribbon microphones were used and the loudness normalized guitar recordings were presented in solo and within a music mix. The experimental results suggest that microphone angles did not generate any perceivable changes to this cohort on this program material.
Engineering Brief 313

9:30

EB2-6 The Mixing Glove and Leap Motion Controller: Exploratory Research and Development of Gesture Controllers for Audio Mixing—*Jack Kelly, Diego Quiroz*, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canada

Digital musical instruments (DMIs) have been evolving over the past several decades, and much research has been done on the subject of capturing the gestures of performers in an effort to re-map them to digital instruments. The practice of audio mixing is no different from a musical performance in this respect. The gestures used by engineers are expressive and have complex metaphorical significance. In this paper two approaches to gesture-based mixing tools for audio engineers, the Mixing Glove and the Leap Motion Controller are explored. Both systems are designed to control volume, panning, solo/mute, and reverb, using hand gestures alone.
Engineering Brief 314

9:30

EB2-7 Discussion on Subjective Characteristics of High Resolution Audio—*Mitsunori Mizumachi,*[1] *Ryuta Yamamoto,*[2] Katsuyuki Niyada[3]
[1]Kyushu Institute of Technology, Kyushu, Japan
[2]Digifusion Japan, Tokyo, Japan
[3]Hiroshima Cosmopolitan University, Hiroshima, Japan

High resolution audio gains in popularity on both audio production and consumer markets. It is necessary to characterize the advantage of high resolution audio over legacy audio formats. The authors have already reported perceptual discrimination rates for high resolution audio, that is, 192 kHz/24 bits PCM format, against a 48 kHz/16 bits format and its compressed versions under a listening room and in-car environments, respectively. In this paper we also focus on perceptual discrimination concerning bit depth in between 24 and 16 bits. Participants were asked to judge either the same or not for each paired stimulus of 48 kHz/24 bits and 48 kHz/16 bits formats. The discrimination rates depend on the reproduction environments, although those subjects could discriminate the difference in between 192 kHz/24 bits and 48 kHz/16 bits formats. It is supposed that high resolution audio benefits more from the wide frequency range than from the bit depth.
Engineering Brief 315

9:30

EB2-8 Accurate Extraction of Dominant Reflections from Measured Sound Intensity Responses in a Room—*Masataka Nakahara,*[1] *Akira Omoto,*[1,2] *Yasuhiko Nagatomo*[3]
[1]ONFUTURE Ltd., Tokyo, Japan
[2]Kyushu Institute of Technology, Kyushu, Japan
[3]Evixar Inc., Tokyo, Japan

The intensity responses in three orthogonal directions, which are calculated from measured impulse responses, include information of dominant reflections in 4-pi space. This information is important for restoration of a sound field, e.g., a spatial reverberator. A method of extracting significant reflections from instantaneous intensities or their envelopes is, therefore, the key application element. To improve the accuracy of detecting reflections, the authors have introduced a new strategy, named "speed detection," which calculates the moving speed of instantaneous intensities at every sampled time. If the speed is lower than an assumed threshold, these intensities indicate reflections. On the contrary, faster speeds indicate the residual transient components of intensities. This "speed

detection" is verified with the measured data of several experiments.
Engineering Brief 316

9:30

EB2-9 Investigation of Interchangeability of Audio Objects' Spatial Sound Direction between 3D Audio Rendering Systems and Rooms by VSV (Virtual Source Visualizer)—*Takashi Mikami,*[1] *Masataka Nakahara,*[1,2] *Akira Omoto*[2,3]
[1]SONA Corpration, Tokyo, Japan
[2] ONFUTURE Ltd., Tokyo, Japan
[3] Kyushu Institute of Technology, Kyushu, Japan

Recently, some kinds of 3D sound rendering systems (such as Dolby Atmos, DTS:X, 22.2ch. etc.) are proposed and commercialized in audio industries. Similarity / difference in 3D sound localizations were examined by using sound intensity. Sound intensities are measured on different rendering systems in the same room, and also on the same rendering systems in different rooms. Subjective study to evaluate 3D sound rendering systems requires much time and labor. Evaluation by sound intensities, measured physically, is very useful. The session discusses interchangeability / difference of sound direction between rendering systems and between rooms obtained by analyzing the visual images and numerical data of sound localization.
Engineering Brief 317

9:30

EB2-10 Loudness Management in the Blu-ray Disc Ecosystem in the Context of Today's Playback Environments—*Andreas Ehret,*[1] *Sripal Mehta,*[2] *Mike Ward*[2]
[1]Dolby Germany, Nürnberg, Germany
[2] Dolby Laboratories, Inc., San Francisco, CA, USA

Loudness management within the Blu-ray Disc ecosystem has historically been less of a priority than in other media playback ecosystems. Instead, the industry has focused on delivering the highest fidelity and full dynamic range audio. As a result, the measured loudness of the content on Blu-ray Disc is generally not accurately indicated in the audio bitstreams carried on Blu-ray discs. However, as more use-cases emerge to connect Blu-ray Disc players to playback environments with limited dynamic range reproduction capabilities (such as TVs or Sound bars), loudness management is becoming more important to ensure optimal playback for these new device types. This brief explains the value of loudness management in the Bluray Disc ecosystem to address new playback environments and gives example workflows for correctly setting loudness values in audio bitstreams delivered on Blu-ray Disc.
Engineering Brief 318

Sunday, May 21 10:00 Salon 15 Paris

Technical Committee Meeting on Automotive Audio

Pro Sound Expo Sunday, May 21
10:15 – 11:00 PSE Stage

IMMERSIVE MIXING IN PRO TOOLS

Presenter: **Dave Tyler**, AVID

Dave Tyler talks about the latest Pro Tools release, which integrates Dolby Atmos immersive mixing into Pro Tools and takes a look at how immersive audio is shaping the future of the industry.

Student and Career Development Event
RECORDING COMPETITION—PART 1
Sunday, May 21, 10:30 – 12:15 Berlin-A

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

10:30: Category 1—Traditional Acoustic Recording

11:30: Category 2– Traditional Studio Recording

Workshop 7 Sunday, May 21
10:45 – 12:15 Salon 4+5 London

NEW DEVELOPMENTS IN LISTENING TEST DESIGN

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

Panelists: *Jan Berg*, Luleå University of Technology, Luleå, Sweden
Todd Welti, Harman International, Northridge, CA, USA

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

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

Sunday, May 21 11:00 Salon 15 Paris

Technical Committee Meeting on Loudspeakers and Headphones

Pro Sound Expo Sunday, May 21
11:15 – 12:00 PSE Stage

GENELEC SMART ACTIVE MONITORS

Presenter: **Aki Mäkivirta**, Genelec

Built for today's fast-paced studio environment, Genelec Smart

Active Monitors (SAM) are designed to meet your workflow and help you improve your sound reproduction quality. As space becomes more limited, listening areas are more confined and room acoustic problems more prevalent.

SAM Systems draw on Genelec's decades of experience and expertise to create an intelligent, flexible network of monitors and subwoofers that can adapt to your requirements. Use Genelec Loudspeaker Manager (GLMTM) 2.0 software to create monitoring systems ranging from traditional stereo to immersive audio setups relying on the proprietary power of GLM AutoCalTM to create an optimal monitoring environment.

Workshop 8 **Sunday, May 21**
11:30 – 12:30 **Salon 7 Vienna**

IMMERSIVE RECORDING AND MIXING TECHNIQUES

Presenter: **Daniel Shores**, Sono Luminus, Boyce, VA, USA

In this demonstration Dan Shores will discuss and demonstrate the recording and mixing techniques used to create 9.1 Auro-3D recordings of the Icelandic Symphony Orchestra, Los Angeles Percussion Quartet, ACME, Skylark, and others. The recordings discussed are all commercially available on Pure Audio Blu-ray. He will also demonstrate the mixing techniques used on the Auro-3D release of Platinum selling EDM Artist's album "Electronic Opus" that integrates electronics, dance, and live orchestra.

Sunday, May 21 **12:00** **Salon 15 Paris**

Technical Committee Meeting on Signal Processing

Pro Sound Expo **Sunday, May 21**
12:15 – 13:00 **PSE Stage**

RIBBON MICS

Presenter: **Sammy Rothmann**, AEA

What are ribbon microphones and how do you use them? Sammy Rothman answers these questions and more in "AEA Ribbon Mics: Fix It In the Mic" which delves into all things ribbons including best miking practices, how ribbons mics work, the differences between ribbons and condensers, and an exploration of AEA's diverse line of active and passive ribbon microphones.

Tutorial 10 **Sunday, May 21**
12:45 – 14:30 **Berlin-A**

AUDIO FORENSICS—WHAT'S IT ALL ABOUT

Chair: **Eddy B. Brixen**, ebb-consult, Smornum, Denmark

Panelists: **tba**

Working with audio forensics is serious business. Depending on the work of the forensics engineer, people may eventually end up in prison. This tutorial will present the kind of work related to the field. This covers fields as acoustics, when audio analysis can be a part of the crime scene investigation. Voice acoustics: Who was speaking? Electro acoustics: Checking data on tapes, discs or other data storage media. Recording techniques: Is this recording an original production or is it a copy of others' work. Even building acoustics and psychoacoustics, when the question is raised: Who could hear what? However, the most important "everyday work" of the audio forensics engineers is cleaning of audio recordings and providing transcriptions. This tutorial presented by practioners that have their ears deep in the matter.

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

Workshop 9 **Sunday, May 21**
12:45 – 14:15 **Salon 7 Vienna**

RECORDING, MIXING, AND MASTERING FOR DIFFERENT IMMERSIVE AUDIO FORMATS

Chair: **Stefan Bock**

Panelists: *Tom Ammermann*, New Audio Technology GmbH, Hamburg, Germany
Artur House, Herold Studios Gmbh
David Miles Huber, Artist, Mixer, Producer, Seattle/Berlin
Morten Lindberg, 2L (Lindberg Lyd AS), Oslo, Norway
Darcy Proper, Wisseloord Studios, Hilversum, The Netherlands
Daniel Shores, Sono Luminus, Boyce, VA, USA
Tobias Wendl, Sound Alliance

Immersive audio formats have been around for some years now. This workshop gives insights about the latest experiences with recording, mixing, and mastering in immersive audio formats. With a number of competing formats on the market today, the productions have to take multiple aspects into consideration: different speaker layouts, channel counts, as well as object based approaches. The workshop will discuss the challenges for the engineers and studios to deal with the different standards for immersive audio. New workflows had to be established to adapt to the new requirements, technically, as well as musically. And depending on the music genre, the approaches might be totally different. The workshop will also explain, how immersive recordings can be delivered in multiple immersive audio formats and how consumers homes can be reached.

This session is presented in association with the AES Technical Committee

Session P11 **Sunday, May 21**
13:00 – 14:00 **Salon 1 Moscow**

AUDIO RECORDING

Chair: **Nadja Schinkel-Bielefeld**, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

13:00

P11-1 Is it Harder to Perceive Coding Artifact in Foreign Language Items? – A Study with Mandarin Chinese and German Speaking Listeners—*Nadja Schinkel-Bielefeld*,[1] *Zhang Jiandong*,[2] *Qin Yili*,[2] *Anna Katharina Leschanowsky*,[1] *Fu Shanshan*[3]
[1] Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]Academy of Broadcast Planning SAPPRFT, Beijing, China
[3]Pleasant Audio Consulting Ltd., Beijing, China

MUSHRA listening tests for the evaluation of audio coded material often also include speech stimuli in a language the listener does not understand. However, it is not clear to what extent the lack of understanding and the unfamiliarity with that language and its phonemes may influence the listener's behavior during the test

and his or her quality ratings. In a study containing German and Mandarin Chinese speaking listeners as well as items of these two languages we analyze how ratings and listening times are affected by the foreign language. Pooled over all conditions we find no significant differences in the ratings. However, for high quality items we find that compared to native listeners, non-native listeners need longer listening times and compare more between items.
Convention Paper 9739

13:30

P11-2 Parametric Joint Channel Coding of Immersive Audio —*Heidi-Maria Lehtonen, Heiko Purnhagen, Lars Villemoes, Janusz Klejsa, Stanislaw Gorlow*, Dolby Sweden AB, Stockholm, Sweden

This paper presents a parametric joint channel coding scheme that enables the delivery of channel-based immersive audio content in formats such as 7.1.4, 5.1.4, or 5.1.2 at very low bit rates. It is based on a generalized approach for parametric spatial coding of groups of two, three, or more channels using a single downmix channel together with a compact parametrization that guarantees full covariance re-instatement in the decoder. By arranging the full-band channels of the immersive content into five groups, the content can be conveyed as a 5.1 downmix together with the parameters for each group. This coding scheme is implemented in the A-JCC tool of the AC-4 system recently standardized by ETSI, and listening test results illustrate its performance.
Convention Paper 9740
Paper presented by Heiko Purnhagen

Workshop 10 — Sunday, May 21
13:00 – 14:30 — Salon 4+5 Vienna

EXPERT TRANSFER TECHNIQUES: A SPECIAL FOCUS ON MECHANICAL DISCS

Chair: **Nadja Wallaszkovits**, Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria; NOA GmbH, Vienna, Austria

Panelists: *Klaus Blasquiz*, Muselec, Paris, France
Stefano S. Cavaglieri, Fonoteca Nazionale Svizzera, Lugano, Switzerland
Jean-Hugues Chenot, Institut National de l'Audiovisuel, Bry-sur-Marne, France
Carl Haber

The workshop leads through the problem of transfer, digitization, and restoration of historical obsolete disc formats. Starting with the possibilities, advantages, and limitations of a conventional mechanical transfer, the discussion will outline some of the most proven and tested optical transfer methods and technologies and their special usability with broken/ delaminated/ damaged discs. The different approaches will be presented, including various audio examples.

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

Sunday, May 21 — 13:00 — Salon 15 Paris

Technical Committee Meeting on High Resolution Audio

Session EB3 — Sunday, May 21
14:00 – 14:45 — Salon 2+3 Rome

LECTURE: ROOM TRANSDUCERS, EDUCATION

Chair: **Bob Schulein**, RBS Consultants, Schaumburg, IL, USA

14:00

EB3-1 Optimization of the Overall Scattering Factor for the Acoustic Simulation of Classrooms—*Dragan Novkovic,*[1] *Stefan Dimitrijevic*[2]
[1]School of Electrical and Computer Engineering of Applied Studies, Belgrade, Serbia
[2] Structor Akustik AB, Stockholm, Sweden

An educational facility used for lectures and multimedia presentations was acoustically measured and simulated in the acoustic simulation software EASE. After processing the acquired data, large discrepancies between the results obtained by impulse response measurement and those obtained from the simulation were observed. All materials, whose data were entered into the simulation, were described solely by absorption coefficients without any information about the scattering, which is common in such situations. Overall simulated scattering factor was adjusted in such way to allow matching of measured and simulated results within a reasonable tolerance limits. As a result of this process, the authors have discussed the possible approaches for the optimization of this parameter in the process of software simulation of acoustically similar spaces..
Engineering Brief 319
This eBrief was withdrawn

14:15

EB3-2 High Frequency—Ultra Audio Band Mode Voice Coil Temperature Measurement—*Isao G. Anazawa*, NyWorks, Toronto, Ontario, Canada

As the power driving mobile devices loudspeaker increases for a better audio experience, an accurate measurement of the voice coil temperature becomes necessary in order to protect the loudspeaker from over-heating. Electrical solutions have been developed in the past to measure the temperature indirectly from the voice coil resistance using a low frequency probe tone or using main audio contents. This paper explains an ultra-audio band high frequency probe to measure the resistance. The test results show good accuracy without the known side effects that exist with current methods.
Engineering Brief 320

14:30

EB3-3 U 87—Microphone Development in the 1960s—*Martin Schneider*, Georg Neumann GmbH, Berlin, Germany

The *U 87* was introduced in 1967 as part of the first generation of transistorized condenser microphones. In the 1960s many aspects like dynamic range, powering, and the seemingly simple question of connectors had to be reconsidered differently from the preceding generations of tube microphones, and for the new recording environments of the time. A detailed look at this microphone and its many variants (for different countries, and different powering schemes) give an insight into the spectrum of development topics and recording technology of the 1960s and 1970s.
Engineering Brief 321

Student and Career Development Event
STUDENT RECORDING CRITIQUES
Sunday, 14:00 – 15:00 Salon 11 (Genelec Demo Room)

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

Students! Come and get tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system, and then receive feedback from a panel of renowned industry professionals. Students at any of their studies can sign up to participate. Students should sign up at the student (SDA) booth immediately on arrival at the convention, and deliver stereo 44.1 Khz, 24 bit AIFF or WAVE files to the SDA booth at that time. Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by Genelec.

Pro Sound Expo Sunday, May 21
14:00 – 14:45 PSE Stage

PRODUCING MUSIC IN 2.0, 5.1, AND 9.1

Presenter: **David Miles Huber**

There is practically nothing about the making of a DMH project that follows the normal recording process. For starters, it is created from multiple electronic and acoustic sound sources. Add to this the idea that each project is produced in multiple formats:

- Berlin dance (in both stereo and 5.1 surround)
- Original mix (in both stereo and 5.1 surround)
- Chill/relaxation mix (in both stereo and 5.1 surround)
- Bluray disc version (9.1 Auro3D)

Each mix version is carefully mixed and mastered at 96kHz/24bits. The berlin mix involves a special process that allows the original mix to be imported into Ableton Live, thereby allowing DMH to perform the project live on-stage in multiple output formats … these add up to a Grammy-nominated process that's totally unique.

Tutorial 11 Sunday, May 21
14:15 – 15:45 Salon 7 Vienna

RECORDING FOR AURO-3D/9.1—
IMMERSIVE & SURROUND FORMATS

Moderator: **Thor Legvold**, Sonovo A/S, Stavanger, Norway

Panelists: *Tom Ammermann*, New Audio Technology
Ralph Kessler, Pinguin Audio
Hyunkook Lee, University of Huddersfield, Huddersfield, UK
Daniel Shores, Sonoluminus

There are a number of competing formats to choose from in Immersive Audio: Auro-3D, Dolby Atmos, DTS-X, NHK 22.2, and more. This tutorial will focus on practical and theoretical considerations when producing content for immersive formats, especially with regards to recording and mixing. We will suggest methodologies as well as explore emerging standards and workflows in order to be positioned to take advantage of the ongoing developments in the field, and discuss market considerations and possible synergy between Immersive Audio and audio for Virtual/Augmented Reality. Panelists with research experience and published productions will discuss and present examples of their work for the various formats, provide valuable insight into the formats available and provide practical advice about how to be "Immersive Audio" ready.

Session EB4 Sunday, May 21
14:30 – 15:00 Salon 1 Moscow

LECTURE: SPATIAL AUDIO—BINAURAL 1

Chair: **Christoph Pörschmann**, TH Köln, Cologne, Germany

14:30

EB4-1 A Spherical Near-Field HRTF Set for Auralization and Psychoacoustic Research—*Christoph Pörschmann,*[1] *Johannes M. Arend,*[1,2] *Annika Neidhardt*[3]
[1]TH Köln, Cologne, Germany
[2]Technical University of Berlin, Berlin, Germany
[3]Technical University of Ilmenau, Ilmenau, Germany

Head-related transfer functions (HRTFs) describe the directional filtering caused by the head, pinna, and torso and are an essential component of binaural synthesis systems. Currently most of these systems are based on far-field HRTFs and thus do not consider acoustical specifics of nearby sound sources. One reason might be that full spherical near-field HRTF sets are rarely available. In this paper we present an HRTF set of a Neumann KU100 dummy head and a technical evaluation of the set. The set is freely available for download and contains post-processed impulse responses, captured on a circular and full spherical grid at distances between 0.25 m and 1.50 m. It can be used for psychoacoustic research and for applications where nearby virtual sound sources shall be auralized.
Engineering Brief 322

15:00

EB4-2 Binaural Recording System and Sound Map of Malaga— *Carmen Rosas, Salvador Luna-Ramirez*, University of Malaga, Malaga, Spain

This Engineering Brief describes part of the results obtained in the Master Thesis "Binaural Recording System and Sound Map" for the Masters in Acoustic Engineering at the University of Malaga. The aim of this project is the construction and the characterization of a pair of in-ear binaural microphones with high-quality capsules. A set of HRTF measurements was obtained and applied to different audio signals for the realization of a psychoacoustic experiment to assess the spatiality provided by the system. For the system assessment, another set of audio samples was generated from the MIT's HRTFs, and both results have been compared. Additionally, different soundscapes have been recorded with the binaural system, and a binaural sound map of Malaga has been developed, which aims to create an archive to collect and conserve the most distinctive sounds of the city using an immersive technology.
Engineering Brief 323

Tutorial 12 Sunday, May 21
14:30 – 16:30 Salon 4+5 London

STANDARD MEASUREMENT OF LOUDSPEAKER SYSTEMS AND COMPONENTS

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

This tutorial reports the development of new IEC standards addressing conventional and modern measurement techniques applicable to all kinds of transducers, active and passive loudspeakers, and other sound reproduction systems. The first standard proposal describes important acoustical measurements for evaluating the generated sound field and signal distortion based on black-box modeling. The second standard is dedicated to the measurement of electrical and mechanical state variables (e.g., displacement), the identification of lumped and distributed parameters (e.g., T/S), and long-term testing to assess power handling, thermal capabilities, product reliability and climate impact. The tutorial gives a deeper insight into loudspeaker modeling, which is the basis for modern measurement techniques, and shows the practical relevance of this knowledge for transducer and system design.

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

Workshop 11 **Sunday, May 21**
14:30 – 16:30 **Berlin-A**

MUSIC MIXING—PART 3

Chair: **Richard King**, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canadal

Panelists: *Michelle Desachy*, Estudio 19, Mexico City, Mexico
Phil Harding, Leeds Beckett University, UK
Rob Toulson, University of Westminster, London, UK

A panel of award-winning expert practitioners from varying backgrounds within the industry will spark interesting discussion and debate. Topics will include the process of mixing, techniques used, and proven methodologies that have yielded successful results over the years in a constantly changing industry. Focus will include the different ways to approach a mix, how to improve an existing mix, how to best interpret and address mix comments from the client. Balancing, use of processing, and listening levels will be addressed. Ample time will be reserved for a question period so that the audience will have a chance to solicit specific information from the panel members.

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

Session P12 **Sunday, May 21**
15:00 – 18:00 **Salon 1 Moscow**

ROOM ACOUSTICS:
SOUND FIELD SIMULATION AND GENERATION

Chair: **tba**

15:00

P12-1 Quantitative Investigation Artificial Room Simulations Reproduced by Channel-Based and Object-Based Surround Sound Systems—*Bernard Camilleri, Jakob Bergner, Christoph Sladeczek,* Fraunhofer Institute for Digital Media Technology (IDMT), Ilmenau, Germany

The introduction of object-based audio reproduction comes along with new challenges for the sound engineer to record, design, and synthesize reverberant sound fields due to the increased number of speakers and the placement of such. The aim of this paper is to show that several parameter settings from a digital reverberation unit produce contrasting reflectograms in a 5.0 channel-based setup and an object-based setup that can have effects on the perceived reverberant sound field. Conversely, established acoustical metrics derived from the measured room impulse responses (RIRs) in both multichannel reproduction setups do not highlight the differences noticed in the reflectograms. The potential consequences regarding individual system properties and the metrics themselves are discussed in this work.
Convention Paper 9741

15:30

P12-2 Comparative Perceptual Evaluation between Different Methods for Implementing Reverberation in a Binaural Context—*Lorenzo Picinali,*[1] *Alexander Wallin,*[2] *Yuli Levtov,*[2] *David Poirier-Quinot*[1]
[1]Imperial College London, London, UK
[2]Reactify Music, London, UK

Reverberation has always been considered of primary importance in order to improve the realism, externalization and immersiveness of binaurally spatialized sounds. Different techniques exist for implementing reverberation in a binaural context, each with a different level of computational complexity and spatial accuracy. A perceptual study has been performed in order to compare between the realism and localization accuracy achieved using five different binaural reverberation techniques. These included multichannel Ambisonic-based, stereo and mono reverberation methods. A custom web-based application has been developed implementing the testing procedures and allowing participants to take the test remotely. Initial results with 54 participants show that no major difference in terms of perceived level of realism and spatialization accuracy could be found between four of the five proposed reverberation methods, suggesting that a high level of complexity in the reverberation process does not always correspond to improved perceptual attributes.
Convention Paper 9742

16:00

P12-3 Data-Driven Granular Synthesis—*Sadjad Siddiq,* Square Enix Co., Ltd., Tokyo, Japan

Granular synthesis is a flexible method to create a wide range of complex sounds, like the sound of rain or water, using very short waveforms, called grains. To synthesize realistic, natural sounds appropriate grains are needed. In an earlier paper we already presented a method to extract grains from recordings of complex sounds. In this paper we describe an extension of the earlier method in which the end of incomplete grains is estimated to improve sound quality. Additionally synthesis parameters that allow us to recreate sound output very close to the original recordings are found automatically. A few seconds of audio input will provide enough data to synthesize sounds of arbitrary length. The necessary grains only require little memory and since synthesis parameters can also be varied to change the nature of the sound, this method is especially beneficial for video games. While empirical listening suggests that the synthesized waveforms sound natural, a formal listening test was not conducted. Sound samples are provided.
Convention Paper 9743

16:30

P12-4 Parametric Synthesis of Crowd Noises in Virtual Acoustic Environments—*Vincent Grimaldi,*[1] *Christoph Böhm,*[2] *Stefan Weinzierl,*[2] *Henrik von Coler*[2]

[1]IRCAM, Paris, France
[2] Technical University of Berlin, Berlin, Germany

This paper presents the design and evaluation of a parametric sound texture synthesis for the generation of crowd noise in virtual acoustic environments. It allows the control of the crowd size, its level of excitement, and its spatial distribution in real-time. A corpus-based concatenative approach is used to generate single streams of indistinct speech that are superimposed to create an unintelligible "babbling" texture. Speech material was recorded in semi-supervised group discussions in the anechoic chamber. The database is used in a real-time implementation with a subsequent rendering using dynamic binaural synthesis. Listening tests were conducted to evaluate the effect of different parameter settings, as well as the perceived "naturalness" of the simulation.
Convention Paper 9744

17:00

P12-5 Real or Illusion? A Comparative Study of Captured Ambiance vs. Artificial Reverberation in Immersive Audio Applications—*Richard King,*[1,2] *Brett Leonard,*[3] *Will Howie,*[1,2] *Jack Kelly*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2] Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canada
[3]BLPaudio, St. Louis, MO, USA

Spatial audio researchers and content producers agree that the best source material for immersive audio is provided by the capture of acoustic signals at various elevations in a room. Where music recording is concerned, this technique is generally preferred over signal processing, as it provides a more natural and realistic impression of immersion. The authors' previous work evaluated the content of rear height channels, which demonstrated that a group of listeners could not discriminate between real room sound and artificial reverberation, and showed no significant preference for either version. The current research investigates whether or not there is a preference for real source ambience over artificially generated reverberation in all four of the height channels (i.e., front and rear elevation) of a 9.1 immersive playback system. Results show some subjects can consistently discriminate between ambiences, but no consistent preference for ambience was observed.
Convention Paper 9745

17:30

P12-6 Investigating the Impact of a Music Stand on Stage Using Spatial Impulse Responses—*Sebastià Vicenç Amengual Gari, Malte Kob,* University of Music Detmold, Detmold, Germany

A measurement set-up replicating a trumpet solo concert situation is arranged on stage by means of a music stand, a directive loudspeaker, and a microphone array. Spatial Room Impulse Responses are measured and analyzed to evaluate the acoustic impact of the music stand at the musician position, depending on the stand location and orientation. Results show that when the stand is orientated towards the receiver the sound level at high frequencies increases up to 9 dB. In some cases, the level of the stand reflection at high frequencies is higher than the source itself, due to its radiation characteristics. The effects of a possibly perceivable timbre change on stage are discussed.
Convention Paper 9746

Session P13 **Sunday, May 21**
15:00 – 18:00 **Salon 2+3 Rome**

AUDIO PROCESSING AND EFFECTS

Chair: **Udo Zölzer**, Helmut-Schmidt-University, Hamburg, Germany

15:00

P13-1 The Perceptual Effect of Vertical Interchannel Decorrelation on Vertical Image Spread at Different Azimuth Positions—*Christopher Gribben, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Two subjective experiments have been conducted to investigate the effect of vertical interchannel decorrelation on the perception of vertical image spread (VIS). Pairs of vertically arranged loudspeakers, one at ear level and another elevated by 30°, were positioned at 0°, ±30°, and ±110° azimuth to the listener. The first experiment compared octave-band pink noise stimuli, consisting of two decorrelation methods with three levels of interchannel cross-correlation (ICC), a coherent sample and a monophonic sample. The effect of vertical ICC on VIS perception was found to be most effective for frequencies around 500 Hz and above, with little effect at lower frequencies. The second experiment judged the absolute lower and upper boundaries of perceived VIS, using stimuli from the first experiment, showing a potential association between VIS and vertical localization.
Convention Paper 9747

15:30

P13-2 Predictors for the Perception of "Wildness" and "Heaviness" in Distorted Guitar Timbre—*Koji Tsumoto, Atsushi Marui, Toru Kamekawa*, Tokyo University of the Arts, Adachi-ku, Tokyo, Japan

Predictors for the perception of *wildness* and *heaviness* in distorted guitar timbre were investigated. A pairwise comparison was conducted for the stimuli of five different amounts of distortion and three types of diodes. The result indicated that the perception of wildness and heaviness seemed to be compiled as one attribute associated with the "power" of the timbre. The ratings appeared to correspond to the threshold voltage of diodes and the amount of distortion. Also, the spectral kurtosis had a relatively high negative correlation with the ratings. The types of diodes, the amount of distortion, and the spectral kurtosis seemed to be appropriate predictors for the perception of *wildness* and *heaviness*.
Convention Paper 9748

16:00

P13-3 An Investigation into the Relationship between the Subjective Descriptor Aggressive and the Universal Audio of the 1176 FET Compressor—*Austin Moore, Jonathan Wakefield,* University of Huddersfield, Huddersfield, UK

In popular music productions, the lead vocal is often the main focus of the mix and engineers will work hard to impart creative coloration on this source. This paper conducts listening experiments to test if there is a correlation between perceived distortion and the descriptor "aggressive" which is often used to describe the sonic signature of the Universal Audio 1176. The results from this study show compression settings that impart audible distortion

are perceived as aggressive by the listener and there is a strong correlation between the subjective scores for distortion and aggressive. It was also shown there is a strong correlation between compression settings rated to have high aggressive scores and the audio feature roughness.
Convention Paper 9749

16:30

P13-4 Investigations Towards Plausible Blind Upmixing of Applause Signals—*Alexander Adami, Lukas Brand, Jürgen Herre,* International Audio Laboratories Erlangen, Erlangen, Germany

Blind upmix denotes the process of converting audio content into a higher number of output channels without the aid of any prior spatial information. This is often needed for upmixing legacy monophonic recordings into modern multichannel audio formats. Especially in live-recordings, applause plays a vital role. However, creating a convincing blind upmix of applause signals is a demanding task. Applause can be interpreted as a superposition of distinctive and individually perceivable foreground claps and a more noise-like background. While the background signal can be upmixed by applying decorrelation and distribution across channels, it is important that the foreground claps are spatially distributed in a perceptually meaningful and plausible manner. This paper investigates the effect of the spatial, temporal, and timbral structure of foreground claps on the perceived plausibility of applause signals. The assessment was done by means of two listening tests. Results show that especially for sparse applause, plausibility is significantly reduced if its natural timbral and temporal structure is corrupted.
Convention Paper 9750

17:00

P13-5 Joint Parameter Optimization of Differentiated Discretization Schemes for Audio Circuits—*Francois Germain,*[1] *Kurt James Werner*[2]
[1]Stanford University Stanford, CA, USA
[2]Queen's Univerity Belfast, Belfast, UK

We propose a new approach to discretizing audio circuits that involves applying differentiated discretization schemes among the elements of a linear circuit, or sub-circuit, rather than a single uniform scheme. The scheme coefficients are jointly optimized to minimize some frequency response error function for that linear circuit. We describe the mathematical framework for this optimization and apply it to the case of the parametric bilinear transform. Differentiated discretization coefficients are jointly optimized by minimizing the L^2 -norm error between the discretized frequency response and the frequency response of the original system. To demonstrate the validity of our approach, we apply our method to several examples and show a systematic reduction of the frequency response error in each case.
Convention Paper 9751

17:30

P13-6 Virtual Analog Modeling of Dynamic Range Compression Systems—*Felix Eichas, Etienne Gerat, Udo Zölzer,* Helmut-Schmidt-University, Hamburg, Germany

Dynamic range compression (DRC) systems reduce the dynamic range of an input signal by amplifying low amplitude levels and attenuating higher ones. This work describes a method to digitally model any analog dynamic range compression unit solely with the help of input/output measurements. For this purpose a generic dynamic range compression model is chosen and its structure is adapted to be able to recreate an analog reference device. The linear characteristic as well as the static curve of the reference device are extracted and directly used in the model. Afterwards the parameters of the digital model are adapted with an iterative optimization routine. Finally the output of the digital model and the analog reference system are compared to evaluate the quality of the emulation.
Convention Paper 9752

Session EB5 Sunday, May 21
15:00 – 18:00 Gallery Window

POSTERS: SPATIAL AUDIO—BINAURAL

15:00

EB5-1 Practical Method to Evaluate Noise Generation Systems—*Roksana Kostyk, Przemyslaw Maziewski, Dominik Stanczak,* Intel Technology Poland, Gdansk, Poland

The paper presents the method used to test the accuracy of a fully calibrated noise generation system. The method is based on a frequency response comparison of binaural recordings done in real and simulated environments. The paper will give examples coming from two different audio laboratories. It will also illustrate the influence of the head and torso unit's miss-position on the noise reproduction accuracy.
Engineering Brief 324

15:00

EB5-2 Free Database of Low Frequency Corrected Head-Related Transfer Functions and Headphone Compensation Filters—*Vera Erbes,*[1] *Matthias Geier,*[1] *Hagen Wierstorf,*[2] *Sascha Spors*[1]
[1]University of Rostock, Rostock, Germany
[2] Technical University of Ilmenau, Ilmenau, Germany

A database of publicly available head-related transfer functions (HRTFs) of a KEMAR manikin together with headphone compensation filters for various headphone types is presented. The HRTFs are based on previously published data from Wierstorf et al. (2011) that have additionally been corrected for low frequencies. This compensates for missing information due to low excitation energy in this frequency range during the measurement and allows for shorter impulse responses. A further benefit is demonstrated by the interpolation of HRTFs via magnitude and phase that is only possible with consistent phase information. Both the low-frequency correction as well as the generation of the headphone compensation filters are accompanied by Matlab code to document the processing.
Engineering Brief 325

15:00

EB5-3 Dataset of In-the-Ear and Behind-the-Ear Binaural Room Impulse Responses Used for Spatial Listening with Hearing Implants—*Florian Klein,*[1] *Stephan Werner,*[1] *Anja Chilian,*[2] *Maria Gadyuchko*[1]
[1]Technical University of Ilmenau, Ilmenau, Germany
[2]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

The contribution presents a dataset of binaural room impulse responses (BRIRs) using a KEMAR head and

torso simulator. Sixteen positions around the head are recorded in three rooms with differing room acoustics. The rooms represent a standardized listening lab, a room for rehabilitation of hearing diseases, and a large sized room. Additionally to the in-the-ear recordings, behind-the-ear BRIRs are recorded to simulate the microphone positions of hearing aid devices. The dataset is used in a research project to develop innovative methods and technologies for spatial listening and speech intelligibility using cochlear implants and bone conduction hearing aids. The dataset enables binaural resynthesis of different directions and rooms for research and rehabilitation.
Engineering Brief 326
eBrief presented by Stephan Werner

15:00

EB5-4 Virtual Source Width in Binaural Synthesis with Frequency-Dependent Directions—*Hengwei Su, Atsushi Marui, Toru Kamekawa*, Tokyo University of the Arts, Adachi-ku, Tokyo, Japan

To control the perceived source width rendering by headphone, a method to distribute different frequency bands of a sound source to different directions by binaural synthesis was investigated. Three types of signals including two anechoic musical recordings and white noise were filtered and split into 1/3 octave bands, and each band was convolved with HRTFs from different directions within the intended source width range. Subjective listening tests were conducted to evaluate the performance of this process. There are no evident results showing that this method can successfully synthesis extended sound images. However, it suggests that the distribution of bands according to spectral characteristics of signals is necessary to synthesize sound image without displacement of localization.
Engineering Brief 327

15:00

EB5-5 Testing Babble Noise Reduction Performance of Headset Microphones—*Ergo Esken,*[1] *Antti Kelloniemi*[2]
[1]Skype, Microsoft Corporation, Tallinn, Estonia
[2] Skype, Microsoft Corporation, Redmond, WA, USA

Babble noise is a typical and specific problem in open offices and call centers, which is why workers in these environments use headsets. Babble noise causes disturbance in these spaces, and it easily leaks through typical send noise suppression processing to far end in telecommunication. To improve the send direction signal-to-noise ratio, headsets are equipped with microphone booms with acoustic noise cancelling microphones or microphone arrays. A method to evaluate their efficiency in reducing babble noise is described in this paper.
Engineering Brief 328

15:00

EB5-6 Binaural Spatialization Methods for Indoor Navigation
—*Sylvain Ferrand, François Alouges, Matthieu Aussal*, Ecole Polytechnique, Palaiseau, France

The visually impaired people are able to follow sound sources with a remarkable accuracy. They often use this ability to follow a guide in everyday activities or for practicing sports, like running or cycling. On the same principle, it is possible to guide people with spatialized sound. We have thus developed a navigation device to guide with sounds using binaural synthesis techniques. In this device we are using both localization information provided by a precise and low latency positioning system and heading data computed from an Inertial Measurement Unit. These positioning data are feeding an HRTF based binaural engine, producing spatialized sound in real-time and guiding the user along a way. The user follows the sound, quite naturally and without initial training. Experiments show that it is possible to guide a walker with enough precision.
Engineering Brief 329

15:00

EB5-7 The Two!Ears Database—*Fiete Winter,*[1] *Hagen Wierstorf,* [2] *Alexander Raake,*[2] *Sascha Spors*[1]
[1]University of Rostock, Rostock, Germany
[2]Technical University of Ilmenau, Ilmenau, Germany

TWO !EARS was an EU-funded project for binaural auditory modelling with ten international partners involved. Its main goal was to provide a computational framework for the modelling of active exploratory listening that assigns meaning to auditory scenes. As one outcome of the project, a database including data acquired by the involved partners as well as third-party measurements has been published. Among others, a large collection of Head Related Impulse Responses and Binaural Room Impulse Responses is part of the database. Further, results from psychoacoustic experiments conducted within TWO !EARS to validate the developed auditory model were added. For the usage of the database together with the TWO !EARS model, a software interface was developed to download the data from the database on demand.
Engineering Brief 330

15:00

EB5-8 Personalized HRTF Measurement and 3D Audio Rendering for AR/VR Headsets—*Woon-Seng Gan,*[1] *Santi Peksi,*[1] *JianJun He,*[2] *Rishabh Ranjan,*[3] *Nguyen Duy Hai,*[1] *Nitesh Kumar Chaudhary*[1]
[1]Nanyang Technological University, Singapore
[2]Maxim Integrated Products Inc., Singapore
[3]Immerzen Labs Pte. Ltd., Singapore

This e-Brief describes our recent work in acquiring a fast, personalized head related transfer function (HRTF) and a personalized 3D audio rendering headsets for augmented and virtual reality (AVR) headsets. Binaural signal acquisition and rendering are important tasks in capturing the idiosyncratic acoustics of the pinnae, head and torso, and playback via headphones to the left and right ears. We will highlight a personalized HRTF binaural acquisition cum 3D audio headphone playback system that can take advantage of our individual ear-head anthropometry information in 3D sound acquisition and rendering.
Engineering Brief 331

15:00

EB5-9 Effect of a Known Environment on the Estimation of Sound Source Distance—*Shashank Aswathanarayana*, University of California Santa Barbara, Santa Barbara, CA, USA

The estimation of sound source distance has been a topic of research interest for a number of decades now. Humans are known to be good at localizing sound in the azimuth and elevation but are poor at estimating the sound source distance. This project looks at examining the effect of a known environment on the estimation of sound source

distance. The project aims at initially testing the subjects perception of sound source in an unknown environment and then examining the effect of training the subject to the environment to see if training/learning the acoustics of the environment improves the estimation of the source distance.
Engineering Brief 332

15:00

EB5-10 The Effects of Decreasing the Magnitude of Elevation-Dependent Notches in HRTFs on Median Plane Localization—*Jade Raine Clarke, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

A binaural experiment was conducted to investigate whether a necessary magnitude of pinna related spectral notches in HRTFs exist. Individual HRIRs were measured at 0°, 30°, and 60° in the median plane for three subjects. The original HRTFs were manipulated so that dominant spectral notches between 5 and 10 kHz were filled in two different degrees. Localization tests were carried out with each subject judging each stimulus condition 15 times in a randomized order. It was found that for the 30° and 60° sources, two subjects tended to perceive the image to move upwards as pinna related notches were reduced. For 0°, however, an increase in front-back confusion occurred as a result of notch magnitude manipulation.
Engineering Brief 333

15:00

EB5-11 An Impulse Response Dataset for Dynamic Data-Based Auralization of Advanced Sound Systems—*Chris Pike,*[1,2] *Michael Romanov*[3,4]
[1]BBC Research & Development, Salford, UK
[2]University of York, York, UK
[3]University of Music and Performing Arts Graz, Graz, Austria
[4]Graz University of Technology, Graz, Austria

This engineering brief presents a freely-available binaural room impulse response (BRIR) dataset measured on a multichannel loudspeaker system. The 32-loudspeaker array includes all loudspeaker layouts specified in Recommendation ITU-R BS.2051. Measurements were carried out in an ITU-R BS.1116-compliant listening room using a Neumann KU100 dummy head microphone. BRIRs were measured at 2° steps of rotation of the dummy head. The dataset can be used for dynamic data-based auralization of multichannel loudspeaker signals, such as those generated by the so-called advanced sound systems described in ITU-R BS.2051, i.e., systems that can render surround sound with height signals from channel-based, object-based, and/or scene-based content representations. The dataset is made freely-available in the SOFA file format.
Engineering Brief 334

15:00

EB5-12 Equipment for Fast Measurement of Head-Related Transfer Functions—*Jose J. Lopez, Pablo Gutierrez-Parera*, Universitat Politècnica de València, Valencia, Spain

Binaural audio can become the future of spatial sound systems as more and more music is consumed on mobile devices through headphones. For a better experience, the binaural sound must be individualized for each subject through the use of their personal Head-Related Transfer Function (HRTF). The most straightforward way of personalization is to measure in-situ the HRTF. However, installations and set-ups for that purpose require anechoic chambers and complex motorized positioning systems. In this brief, we present an installation deployed in a non-anechoic room with multiple loudspeakers that provide a way of measuring the HRTF with an excellent resolution in the azimuthal plane and a sufficient resolution on elevation for common purposes.
Engineering Brief 335

Student and Career Development Event
STUDENT DESIGN EXHIBITION
Sunday, May 21, 15:00 – 17:00 Gallery Window

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

Pro Sound Expo Sunday, May 21
15:00 – 15:45 PSE Stage

THE INS AND OUTS OF MICROPHONES

Presenter: **John Willett**

Microphones are the very first link in the recording chain, so it's important to understand them to use them effectively. This presentation will explain the differences between different types of microphones; explain polar-patterns and directivity, proximity effect relative recording distances and a little about room acoustics. Many of these "golden nuggets" helped me greatly when I first understood them and I hope they will help you too.

Sunday, May 21 15:00 Salon 15 Paris

Technical Committee Meeting on Audio Forensics

Tutorial 13 Sunday, May 21
15:45 – 16:45 Salon 7 Vienna

KRAFTWERK 3D – THE CHALLENGE OF CREATING AN OUTSTANDING IMMERSIVE / 3D AUDIO PRODUCTION FOR A MAJOR MUSIC ACT

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

Music has not a cinematic approach where spaceships flying around the listener. Nonetheless music can become a fantastic spatial listening adventure on speakers as well as with common headphones. An outlook how to create such an adventure and how this could sound is the new Kraftwerk Blu-ray production "Kraftwerk 3D." Production strategies and workflows to create Dolby Atmos and Headphone Surround 3D in current workflows and DAWs will be shown and explained.

Sunday, May 21 **16:00** **Salon 16 Riga**

Standards Committee Meeting SC-02-12 Audio Applications of Networks

Tutorial 14 **Sunday, May 21**
16:30 – 18:00 **Salon 4+5 London**

EARPHONES, HEADPHONES, AND HEADSETS: ELECTROACOUSTIC DESIGN & VERIFICATION

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

This presentation reviews basic the electroacoustic concepts of gain, sensitivity, sound fields, linear and non-linear systems, and test signals for ear-worn devices. The Insertion Gain concept is explained and free and diffuse field target responses are shown. Equivalent volume and acoustic impedance are defined. Ear simulators and test manikins appropriate for Circum-, Supra-, and Intra-aural and insert earphones are presented. The salient portions of the ANSI/ASA S3.7 and IEC 60268-4 standards are reviewed. Examples of Frequency Response, Left-Right Tracking, Insertion Gain, Distortion, and Impedance are shown. The basic concepts of Noise Canceling devices are also presented.

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

Workshop 12 **Sunday, May 21**
16:30 – 18:00 **Berlin-A**

THIS IS A MIX! THIS IS A MASTER! V2.0

Chair: **Ian Corbett**, Kansas City Kansas Community College, Kansas City, KS, USA

Panelists: *Andreas Lubich*
Mandy Parnell
Magdalena Piotrowska

Whether you are a student, home studio or project studio user, or someone entering the professional industry, most of the music mixes you hear and try to emulate have been professionally mastered. Too many novices try to recreate a “mastered” sound in their mix. This is undesirable and limits what the mastering engineer can do. Join our panel of mastering engineers we continue the discussion from last year’s events, presenting some “off-the-console” mixes, discuss desirable qualities of mixes submitted to them, their mastering processes, play the resulting master, and discuss common issues they see in some of the material sent to them to master.

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

Tutorial 15 **Sunday, May 21**
17:00 – 18:00 **Salon 7 Vienna**

ALL YOU NEED TO KNOW ABOUT 3D AUDIO

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

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

Special Thanks: In this session we are using headphones from http://silentdisco.de

This session is presented in association with the AES Technical Committee

Pro Sound Expo **Sunday, May 21**
17:00 – 17:45 **PSE Stage**

SIGMASTUDIO AND CURRENT DSP HW ARCHITECTURES FOR AUDIO APPLICATIONS

Presenter: **Miguel Chavez**, ADI

Graphical Audio DSP Programming Environments and newer “audio specific” DSP programming SW and HW architectures. How has SigmaStudio revolutionized how audio equipment is designed in today’s competitive environment? What has been ADI (and others) response has been to new evolving customer needs from a software and hardware perspective? How have new algorithms emerged with the existence of different sensors and converters? How have algorithms evolved with the need for system-performance improvements requirements? What are the latency considerations within different audio applications: “headphones,” “stage,” “studio” among other listening environments?

Sunday, May 21 **17:00** **Salon 15 Paris**

Technical Committee Meeting on Acoustics and Sound Reinforcement

Special Event
THE RICHARD C. HEYSER MEMORIAL LECTURE
Sunday, May 21, 18:30 – 20:00 **Berlin-A**

Lecturer: **Jörg Sennheiser**, Sennheiser electronic GmbH & Co. KG, Wedemark, Germany

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

The Heyser Lecturer this year is **Jörg Sennheiser**.

Sennheiser became Sennheiser’s director of technology in 1976 before taking over management of the family business in 1982. He recently handed over the chairmanship to a successor. His many years of experience in the electroacoustics business, particularly in bringing digital technology to bear on the product range, will provide an interesting backdrop to his presentation entitled, *“A Historic Journey in Audio-Reality: From Mono to AMBEO.”*

The foundations of human audio perception, based on the “natural laws” of human hearing and their relative validity, will be be revisited at the outset. The interdependence of hearing with other human senses will be outlined, showing the high degree of subjectivity and uncertainty of our audio perception. Keeping this in mind, the evolution of audio capturing, processing and reproduction by technical means will be discussed. The achievable hearing pleasure and the audio quality level is reflected in the continuous development from single- to multichannel reproduction systems with loudspeakers and headphones, benchmarked against "sonic reality." The need for more emotionality in the hearing experience—especially together with visual presentations—leads to a requirement for new approaches and solutions throughout the workflow in the entertainment world and entertainment industry. Optimized solutions for capturing sound with action-cameras or smartphones are evolving, with prototypes and first products being already available and undergoing stringent end-user tests. Audio-visual presentations of sports events on TV call for the development of suitable high-quality audio formats in line with an individual choice of video perspective. The rapidly evolving field of gaming, augmented-reality (AR) and virtual-reality (VR) calls for "immersive" audio technologies currently under consideration. Multiple stakeholders in this market segment—from producers and development engineers to end-users—have to work together to imagine and design The Future of Audio.

BANQUET
Sunday, May 21, 20:00 – 22:00
Van Loon

Come and enjoy an evening in Berlin with your colleagues. The dinner will take place at the restaurant Van Loon (http://vanloon.de/van-loon-restaurantship/?lang=en) which is, in fact, a stationary ship. We will have a bus transfer departing at 20:00 from the hotel. The transfer will take about 15 min., the buffet dinner will start at 20:30. The menu includes a selection of starters (smoked fish, grilled vegetables, salads), a main course (picata milanese with tagliatelle), and a desert (pana cotta). A selection of beverages is included (soft drinks, coffee, beer, wine).Tickets will be available at the registration area. Spacing is limited

Session P14 **Monday, May 22**
9:00 – 12:30 **Salon 1 Moscow**

LISTENING TESTS AND PSYCHOACOUSTICS 1

Chair: **Russell Mason**, University of Surrey, Guildford, Surrey, UK

9:00

P14-1 Evaluation of Auditory Events with Projected Sound Sources Using Perceptual Attributes—*Tom Wühle, Sebastian Merchel, M. Ercan Altinsoy*, Dresden University of Technology, Dresden, Germany

The main aim of the projection of sound sources is to change the perceived direction of the auditory event from the direction of the real source to the direction of the projected source. However, the focusing capabilities of projecting sound sources are physically limited. Therefore, the perception of the listener is not only influenced by the projected sound but also by the sound that is directly radiated from the real source. In a scenario with projected sound sources a complex mixture of perceptual attributes change besides the direction of the auditory event. The present study describes this perceptual processes and investigates some of those attributes.
Convention Paper 9753

9:30

P14-2 The Evaluation of the Effect of Sound Directionality in Horizontal Plane on the Human Auditory Distance Perception in a Large Reverberant Room— *Tahereh Afghah,*[1] *Andrew Allen,*[2] *Peter Otto,*[1] *Aravindan Joseph Benjamin*[3]
[1]University of California San Diego, San Diego, CA, USA
[2]Google Inc., Mountain View, CA, USA
[3]Technical University of Ilmenau, Ilmenau, Germany

An evaluation of sound localization effect on the auditory distance estimation in a user study is presented. Binaural Room Impulse Responses of 60 positions were recorded in a reverberant space using a dummy head. The recordings were evaluated by the users in a headphone-based listening test to analyze the listeners' ability to perceive the distance with and without prior knowledge of direction of origin. When known, the distance estimation accuracy in left and right sides of the head in near field (2m, 4m) was improved and at some angles saw a significant improvement. However, known direction did not assist the users in determining the larger distance levels (6m, 8m, 10m). No improvements were seen in the front and back sides for all directions.
Convention Paper 9650
This paper was not presented but is available in the E-Libray

10:00

P14-3 Improvement of the Reporting Method for Closed-Loop Human Localization Experiments—*Fiete Winter,*[1] *Hagen Wierstorf,*[2] *Sascha Spors*[1]
[1]University of Rostock, Rostock, Germany
[2]Technical University of Ilmenau, Ilmenau, Germany

Sound Field Synthesis reproduces a desired sound field within an extended listening area using up to hundreds of loudspeakers. The perceptual evaluation of such methods is challenging, as many degrees of freedom have to be considered. Binaural Synthesis simulating the loudspeakers over headphones is an effective tool for the evaluation. A prior study has investigated whether non-individual anechoic binaural synthesis is perceptually transparent enough to evaluate human localization in sound field synthesis. With the used apparatus, an undershoot for lateral sound sources was observed for real loudspeakers and their binaural simulation. This paper reassesses human localization for the mentioned technique using a slightly modified setup. The results show that the localization error decreased and no undershoot was observed.
Convention Paper 9755

10:30

P14-4 Investigations on Perceptual Phenomena of the Precedence Effect Using a Bessel Sequence—*Florian Wendt,* University of Music and Performing Arts Graz, Graz, Austria

The precedence effect is typically investigated by presenting two instances of a sound with delay in between. Respective studies found various phenomena indicating that in human auditory localization the contribution of the first sound instance often prevails over a later sound or an acoustic reflection. In reverberant environments, the direct sound is typically followed by more than one reflection. Nevertheless, only little is known about the contribution of multiple reflections on the precedence effect. Understandably, a free number of sound instances increases the number of thinkable conditions drastically and an exhaustive systematic investigation appears infeasible. Directionally distributed impulses weighted by a Bessel sequence offer a neat set of free parameters. We chose this scheme to gain quantitative insights into the influence of multiple reflections on the precedence effect. Our study covers the transient precedence effect, the ongoing

precedence effect, and the onset capture effect, which we investigate using sounds of different envelope, frequency range, angular and temporal spread.
Convention Paper 9756

11:00

P14-5 Just Noticeable Difference in Apparent Source Width Depending on the Direction of a Single Reflection—*Dale Johnson*, *Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

An investigation on the just noticeable difference in angle of a single reflection in terms of apparent source width was performed using a staircase method to obtain two, single reflection angles between 0° and 180°. In the presence of a direct sound, subjects compared the apparent source width produced by a single 90° reference reflection, and a single test reflection ranging between 0° to 90° and 0° to 180° for each threshold. Subjects repeated this test for four delay times of 5 ms, 10 ms, 20 ms, and 30 ms. Reflection angles were found to be approximately 40° and 130° and, however, do not appear to vary with delay time. This implies that human hearing is not sensitive to changes in reflection angle in terms of apparent source width between the threshold angles.
Convention Paper 9757

11:30

P14-6 Modeling Horizontal Audio-Visual Coherence with the Psychometric Function—*Hanne Stenzel*, *Philip J. B. Jackson*, *Jon Francombe,* University of Surrey, Guildford, Surrey, UK

Studies on perceived audio-visual spatial coherence in the literature have commonly employed continuous judgment scales. This method requires listeners to detect and to quantify their perception of a given feature and is a difficult task, particularly for untrained listeners. An alternative method is the quantification of a percept by conducting a simple forced choice test with subsequent modeling of the psychometric function. An experiment to validate this alternative method for the perception of azimuthal audio-visual spatial coherence was performed. Furthermore, information on participant training and localization ability was gathered. The results are consistent with previous research and show that the proposed methodology is suitable for this kind of test. The main differences between participants result from the presence or absence of musical training.
Convention Paper 9758

12:00

P14-7 How Important Is Accurate Localization in Reproduced Sound?—*Russell Mason*, University of Surrey, Guildford, Surrey, UK

A meta-analysis was conducted on elicitation studies to examine the perceptual importance of localization-specific and localization-related attributes. It was found that the majority of attributes were localization-related, including (in order of commonality) extent, locatedness, distribution, spaciousness, and movement. The most common localization-specific attribute was distance, with only 2.6% of the attributes relating to the perceived lateral position. It is concluded that localization accuracy experiments may enable experimenters to make predictions about a reasonable proportion of the attributes found, though further research is needed to develop suitable analysis techniques. In addition, more research is required to develop subjective and objective methods for judging perceived distance.
Convention Paper 9759

Session P15 **Monday, May 22**
9:00 – 11:00 **Salon 2+3 Rome**

AUDIO ANALYSIS AND SYNTHESIS

Chair: **John Mourjopoulos***,* University of Patras, Patras, Greece

9:00

P15-1 Close Miking Empirical Practice Verification: A Source Separation Approach—*Konstantinos Drossos*,[1] *Stylianos Ioannis Mimilakis,*[2] *Andreas Floros*,[3] *Tuomas Virtanen*,[1] *Gerald Schuller*[2,4]
[1]Tampere University of Technology, Tampere, Finland
[2] Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[3]Ionian University, Corfu, Greece
[4]Technical University of Ilmenau, Ilmenau, Germany

Close miking represents a widely employed practice of placing a microphone very near to the sound source in order to capture more direct sound and minimize any pickup of ambient sound, including other, concurrently active sources. It is used by the audio engineering community for decades for audio recording, based on a number of empirical rules that were evolved during the recording practice itself. But can this empirical knowledge and close miking practice be systematically verified? In this work we aim to address this question based on an analytic methodology that employs techniques and metrics originating from the sound source separation evaluation field. In particular, we apply a quantitative analysis of the source separation capabilities of the close miking technique. The analysis is applied on a recording dataset obtained at multiple positions of a typical musical hall, multiple distances between the microphone and the sound source multiple microphone types and multiple level differences between the sound source and the ambient acoustic component. For all the above cases we calculate the Source to Interference Ratio (SIR) metric. The results obtained clearly demonstrate an optimum close-miking performance that matches the current empirical knowledge of professional audio recording.
Convention Paper 9760
Paper presented by Stylianos Ioannis Mimilakis

9:30

P15-2 Audio System Spatial Image Evaluation via Binaural Feature Classification—*Gavriil Kamaris*, *Stamatis Karlos*, *Stergios Terpinas*, *Dimitris Koutsaidis*, *John Mourjopoulos,* University of Patras, Patras, Greece

A method for evaluating different audio systems with respect to their spatial reproduction accuracy is described based on binaural auditory feature classification. The classifier is trained to act as expert listener judging system spatial quality and achieves high accuracy for a reference ideal system under anechoic conditions. The trained classifier is then employed to assess different suboptimal reproduction setups and listening conditions. The spatial accuracy was assessed with respect to this reference with respect to the panned image, image localization accuracy, and the sweet spot area spread. For 2 channel stereo reproduction, the study used loudspeakers of different directivity under anechoic or varying reverberant room

conditions. The work also assesses the relative effects of upmixing stereo for 5 channel reproduction.
Convention Paper 9761

10:00

P15-3 Long-Term Average Spectrum in Popular Music and its Relation to the Level of the Percussion—*Anders Elowsson, Anders Friberg,* KTH Royal Institute of Technology, Stockholm, Sweden

The spectral distribution of music audio has an important influence on listener perception, but large-scale characterizations are lacking. Therefore, the long-term average spectrum (LTAS) was analyzed for a large dataset of popular music. The mean LTAS was computed, visualized, and then approximated with two quadratic fittings. The fittings were subsequently used to derive the spectrum slope. By applying harmonic/percussive source separation, the relationship between LTAS and percussive prominence was investigated. A clear relationship was found; tracks with more percussion have a relatively higher LTAS in the bass and high frequencies. We show how this relationship can be used to improve targets in automatic equalization. Furthermore, we assert that variations in LTAS between genres is mainly a side-effect of percussive prominence.
Convention Paper 9762

10:30

P15-4 Efficient Music Identification Approach Based on Local Spectrogram Image Descriptors—*Massimiliano Zanoni, Stefano Lusardi, Paolo Bestagini, Antonio Canclini, Augusto Sarti, Stefano Tubaro,* Politecnico di Milano, Milan, Italy

The diffusion of large music collections has determined the need for algorithms enabling fast song retrieval from query audio excerpts. This is the case of online media sharing platforms that may want to detect copyrighted material. In this paper we start from a proposed state-of-the-art algorithm for robust music matching based on spectrogram comparison leveraging computer vision concepts. We show that it is possible to further optimize this algorithm exploiting more recent image processing techniques and carrying out the analysis on limited temporal windows, still achieving accurate matching performance. The proposed solution is validated on a dataset of 800 songs, reporting an 80% decrease in computational complexity for an accuracy loss of about only 1%.
Convention Paper 9763

Workshop 13 **Monday, May 22**
9:00 – 11:00 **Salon 7 Vienna**

CURRENT WORKFLOWS IN AUDIO FOR VR

Chair: **Chris Pike**, BBC Research & Development, Salford, UK

Panelists: *Nuno Fonseca*, ESTG/Polytechnic Institute of Leiria, Leiria, Portugal; Sound Particles, Leiria, Portugal
Henney Oh, G'Audio Lab, USA
Ferdinando Olivieri, Qualcomm, San Diego, CA, USA
Mandy Parnell
Jan Plogsties, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

In the recent past, several companies of various sizes have developed their own tools to meet the needs of VR-related audio production schemes. Even though we are still far from maturity in this field, it can now be seen how different approaches are becoming increasingly popular amongst the early adopters. This workshop intends to display the most up-to-date selection of tools with a detailed explanation of their workflows.

This session is presented in association with the AES Technical Committee

Session P16 **Monday, May 22**
9:30 – 12:30 **Gallery Window**

POSTERS: SPATIAL AUDIO

9:30

P16-1 MySofa—Design Your Personal HRTF—*Christian Hoene,*[1] *Isabel C. Patino Mejia,*[2] *Alexandru Cacerovschi*[3]
[1]Symonics GmbH, Tübingen, Germany
[2]Universität Tübingen, Tübingen, Germany
[3]DAS Solutions SRL, Chisinau, Moldova

Binaural auralizations are present in increasing numbers of applications and devices. Albeit most of the time they only use generic Head-Related Transfer Functions (HRTFs), the recent standardization of the HRTF format SOFA has paved the way to support individualized HRTFs broadly. We have developed and implemented MySofa: a web service that helps users to design a personal HRTF. In MySofa, based on anthropometric and user inputs, algorithms calculate and tune HRTFs. The result is displayed in the web browser and the user can listen to test renderings to verify, whether the personalized HRTF matches his expectations. In order to foster the use of individualized HRTF, we also implemented a light weight C-library called libmysofa, which helps programmers to read SOFA files and lookup FIR filters.
Convention Paper 9764

9:30

P16-2 Ecological Validity of Stereo UHJ Soundscape Reproduction—*Francis Stevens, Damian Murphy, Stephen Smith,* University of York, York, UK

This paper contains the results of a study making use of a set of B-format soundscape recordings, presented in stereo UHJ format as part of an online listening test, in order to investigate the ecological validity of such a method of soundscape reproduction. Test participants were presented with a set of soundscapes and asked to rate them using the Self-Assessment Manikin (SAM), and these results were then compared with those from a previous study making use of the same soundscape recordings presented in a surround-sound listening environment (a method previously shown to be ecologically valid). Results show statistically significant correlation between the SAM results for both listening conditions, indicating that the use of stereo UHJ format is valid for soundscape reproduction.
Convention Paper 9765

9:30

P16-3 Comparison of HRTFs from a Dummy-Head Equipped with Hair, Cap, and Glasses in a Virtual Audio Listening Task over Equalized Headphones—*György Wersényi,*[1] *József Répás*[1,2]
[1]Széchenyi István University, Györ, Hungary
[2] Obuda University, Budapest, Hungary

Head-Related Transfer Functions (HRTFs) are frequently

used in virtual audio scene rendering in order to simulate sound sources at different spatial locations. The use of dummy-head HRTFs (also referred as generic sets) is often criticized because of poor localization performance, leading to, e.g., lower spatial resolution, in-the-head localization, front-back reversals, etc. This paper presents results of horizontal plane localization obtained by digital filter representations of dummy-head HRTFs that were recorded normally and using additional cap, glasses, and hair on the head. Results of untrained subjects over equalized reference headphones showed no significant difference among the HRTF sets despite of large magnitude differences. This method for customization of generic HRTFs fails if improvement in localization is needed.
Convention Paper 9766

9:30

P16-4 Filter Design of a Circular Loudspeaker Array Considering the Three-Dimensional Directivity Patterns Reproduced by Circular Harmonic Modes—*Koya Sato, Yoichi Haneda*, The University of Electro-Communications, Chofu-shi, Tokyo, Japan

We propose a filter design method for a circular loudspeaker array. This method is based on extended three-dimensional (3-D) bases that are observed by driving each circular harmonic mode using a prototypical circular loudspeaker array. When a desired 3-D directivity pattern is expanded by the extended 3-D bases, the filter coefficients can be obtained by combining the circular harmonics with the expansion coefficients of the desired 3-D directivity pattern. Moreover, the proposed method can suppress large filter gains at low frequencies by limiting the gain at each order (mode) using L1-norm optimization. We evaluated the performance of directivity and sound distortion using an actual 8-element circular loudspeaker array of radius 0.054 m. These results showed that the proposed method could control the 3-D directivity with little distortion.
Convention Paper 9767

9:30

P16-5 Wearable Sound Reproduction System Using Two End-Fire Arrays—*Kenta Imaizumi, Yoichi Haneda*, The University of Electro-Communications, Chofu-shi, Tokyo, Japan

We propose a personal sound reproduction system that uses two wearable end-fire loudspeaker arrays instead of a headphone to present the sound image. The prototype arrays rest on the listener's chest so that the look direction of each array was directed to the listener's ears. To prevent sound leakage around the listener, we designed a narrow directivity for each array. Moreover, we used a crosstalk canceler for localizing the sound image with head-related transfer functions. We verified the performance of the prototype by using simulations. A difference of approximately 15 dB of sound pressure was obtained between the look direction and the other directions. The crosstalk was suppressed from approximately 10 dB to 20 dB. Additionally, we also conducted a subjective listening test of the sound image localization. The right and left correct answer rate was approximately 90%, and the exact match was approximately 40%.
Convention Paper 9768

9:30

P16-6 Normalization Schemes in Ambisonic: Does it Matter?—*Thibaut Carpentier,* IRCAM, Paris, France

In the context of Ambisonic processing, various normalizations of the spherical harmonic functions have been proposed in the literature and there is yet no consensus in the community about which one should be preferred (if any). This is a frequent source of confusion for the end users and this may lead to compatibility issues between rendering engines. This paper reviews the different conventions in use, presents an extension of the FuMa scheme to higher orders, and discusses possible pitfalls in the decoding stage.
Convention Paper 9769

9:30

P16-7 Perceptually Motivated Amplitude Panning (PMAP) for Accurate Phantom Image Localization—*Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

This paper proposes and evaluates a new constant-power amplitude-panning law named "Perceptually Motivated Amplitude Panning (PMAP)." The method is based on novel image shift functions that were derived from previous psychoacoustic experiments. The PMAP is also optimized for a loudspeaker setup with an arbitrary base angle using a novel phantom image localization model. Listeni ng tests conducted using various sound sources suggest that, for the 60° base angle, the PMAP provides a significantly better panning accuracy than the tangent law. For the 90° base angle, on the other hand, both panning methods perform equally good. The PMAP is considered to be useful for intelligent sound engineering applications, where an accurate matching between the target and perceived positions is important.
Convention Paper 9770

9:30

P16-8 Full-Sphere Binaural Sound Source Localization by Maximum-Likelihood Estimation of Interaural Parameters—*Benjamin Hammond, Philip J. B. Jackson,* University of Surrey, Guildford, Surrey, UK

Binaural recording technology offers an inexpensive, portable solution for spatial audio capture. In this paper a full-sphere 2D localization method is proposed that utilizes the Model-Based Expectation-Maximization Source Separation and Localization system (MESSL). The localization model is trained using a full-sphere head related transfer function dataset and produces localization estimates by maximum-likelihood of frequency-dependent interaural parameters. The model's robustness is assessed using matched and mismatched HRTF datasets between test and training data, with environmental sounds and speech. Results show that the majority of sounds are estimated correctly with the matched condition in low noise levels; for the mismatched condition, "cone of confusion" arises with albeit effective estimation of lateral angles. Additionally, the results show a relationship between the spectral content of the test data and the performance of the proposed method.
Convention Paper 9771

9:30

P16-9 Spatial Quality and User Preference of Headphone Based Multichannel Audio Rendering Systems for Video Games: A Pilot Study—*Joe Rees-Jones, Damian T. Murphy,* University of York, York, UK

This paper presents a pilot experiment comparing the perceived spatial quality and preference of virtualized 7.0 sur-

round-sound video game audio with a stereo down-mix of the same material. The benefits of multichannel audio in gaming are clear in that spatialized sound effects can be used to create immersive and dynamically reacting virtual environments, whilst also offering competitive advantages. However, results from this study suggest that the spatial quality of virtual 7.0 surround-sound is not perceived to be significantly different to that of a stereo down-mix and neither rendering method is preferred, based on a feedback from 18 participants. These results are interesting but surprising, as they bring into question the current methods used for spatial game audio presentation over headphones.
Convention Paper 9772

Workshop 14 **Monday, May 22**
9:30– 11:00 **Berlin-A**

EMBRACING COLLABORATION IN THE MODERN PRODUCTION STUDIO

Chair: **Rob Toulson**, University of Westminster, London, UK

Panelists: *Phil Harding*, Leeds Beckett University, UK
Mandy Parnell
Ken Scott, Producer/Engineer
Paul Thompson, Music Producer and Educator

Collaboration takes many forms in contemporary music production. Building effective professional relationships can be the secret to success in modern music production, even in a world where autonomous working is more possible than ever. For example, we see engineers collaborate and co-produce with artists, electronic producers working remotely with session musicians, and self-producing artists nurturing their product through the recording mixing and mastering chain. In this workshop we explore the contemporary practices of collaboration in music production, particularly reflecting on modern technologies and tools that have enabled new frameworks for communication and co-working. We will look at methods of the past that have perhaps been lost owing to new technologies and working methods, and evaluate the education needs to enable new artists and producers to be successful in their careers.

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

Tutorial 16 **Monday, May 22**
9:45 – 10:45 **Salon 4+5 London**

PRACTICAL AES67

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

The AES67 Standard on High performance Streaming Audio-over-IP Interoperability was published in September 2013. Since then, first applications with AES67-compatible devices have been projected and put into operation. This session will demonstrate a working AES environment live and give insight on device and network configuration, management and monitoring.

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

Student and Career Development Event
EDUCATION AND CAREER/JOB FAIR
Monday, May 22, 10:00 – 12:00 **Gallery Window**

The combined AES 142nd Education and Career Fair will match job seekers with companies and prospective students with schools.

Companies

Looking for the best and brightest minds in the audio world? No place will have more of them assembled than the 141st Convention of the Audio Engineering Society. Companies are invited to participate in our Education and Career Fair, free of charge. This is the perfect chance to identify your ideal new hires!

All attendees of the convention, students and professionals alike, are welcome to come visit with representatives from participating companies to find out more about job and internship opportunities in the audio industry. Bring your resume!

Schools

One of the best reasons to attend AES conventions is the opportunity to make important connections with your fellow educators from around the globe. Academic Institutions offering studies in audio (from short courses to graduate degrees) will be represented in a “table top” session. Information on each school’s respective programs will be made available through displays and academic guidance. There is no charge for schools/institutions to participate. Admission is free and open to all convention attendees.

Monday, May 22 **10:00** **Salon 15 Paris**

Technical Committee Meeting on Semantic Audio Analysis

Monday, May 22 **10:30** **Salon 16 Riga**

Standards Committee Meeting SC-04-08 Measurement of Sound Systems in Rooms

Tutorial 17 **Monday, May 22**
10:45 – 12:15 **Salon 4+5 London**

DEVELOPING NOVEL AUDIO ALGORITHMS AND PLUGINS - MOVING QUICKLY FROM IDEAS TO REAL-TIME PROTOTYPES

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

High-level programming languages are frequently used by DSP and Audio Engineers for developing audio products and plugins for use in music production. These languages allow designers to rapidly create and evaluate new audio processing ideas which later are targeted for implementation in commercial audio products. In this workflow, fine-tuning the algorithm design is an important component. In this tutorial, we will use our industry knowledge to summarize the best programming practices adopted by audio companies to enable the reuse of research code directly for real-time prototypes. We will show a number of examples, tips, and tricks to minimize latency, maximize efficiency, run in real-time on a PC, and generate native VST plugins for testing and prototyping.

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

Pro Sound Expo **Monday, May 22**
10:15 – 10:45 **PSE Stage**

STUDIO MONITORS INTEGRATED WITH AOIP—WHY AOIP MAKES SENSE IN LOUDSPEAKER TECHNOLOGY

Presenters: **Frederick Knop, Klaus Heinz**, HEDD

[unavailable]

Session P17 Monday, May 22
11:00 – 12:00 Salon 2+3 Rome

HEARING AIDS AND PRESERVATION

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

11:00

P17-1 Do In-Ear Monitors Protect Musicians' Hearing?—*Arne Nykänen, Magnus Löfdahl, Tomas Johannesson, Jan Berg*, Luleå University of Technology, Luleå, Sweden

In-ear monitors for live performances are commonly considered to give better sound quality than loudspeaker monitors. They are also often assumed to reduce sound exposure. Because of lack of evidence for this, sound exposure for pop/rock/jazz musicians was compared between performances with in-ear and loudspeaker monitors. Equivalent sound pressure levels at the musicians' ears were 94 to 105 dBA with loudspeaker and 86 to 108 dBA with in-ear monitors. Many participants used earplugs when using loudspeaker monitors. Therefore, the recommendation, from a pure hearing protection perspective, is to use loudspeaker monitors and earplugs. However, the large spread in levels between musicians using in-ear monitors suggests that with better training and measurements of sound exposure, in-ear monitors could be used safely.
Convention Paper 9773

11:30

P17-2 An Open-Source Audio Renderer for 3D Audio with Hearing Loss and Hearing Aid Simulations—*Maria Cuevas-Rodriguez,*[1] *Daniel Gonzalez-Toledo,*[1] *Ernesto de La Rubia-Buestas,*[1] *Carlos Garre,*[1] *Luis Molina-Tanco,*[1] *Arcadio Reyes-Lecuona,*[1] *David Poirier-Quinot,*[2] *Lorenzo Picinali*[2]
[1]University of Malaga, Malaga, Spain
[2]Imperial College London, London, UK

The EU-funded 3D Tune-In (http://www.3d-tune-in.eu/) project introduces an innovative approach using 3D sound, visuals, and gamification techniques to support people using hearing aid devices. In order to achieve a high level of realism and immersiveness within the 3D audio simulations, and to allow for the emulation (within the virtual environment) of hearing aid devices and of different typologies of hearing loss, a custom open-source C++ library (the 3D Tune-In Toolkit) has been developed. The 3DTI Toolkit integrates several novel functionalities for speaker and headphone-based sound spatialization, together with generalized hearing aid and hearing loss simulators. A first version of the 3DTI Toolkit will be released with a non-commercial open-source license in Spring 2017.
Convention Paper 9774

Monday, May 22 **11:00** **Salon 15 Paris**

Technical Committee Meeting on Broadcast and Online Delivery

Pro Sound Expo Monday, May 22
11:15 – 11:45 PSE Stage

HOW AOIP CAN BE APPLIED IN THE PROJECT STUDIO ENVIRONMENT

Presenter: **Kieran Walsh**, Regional Manager of Global Support Services, Audinate

Audio Over IP has found many applications in Commercial, Live, and Broadcast sound, where existing infrastructures and deployment methods are of immediate benefit. Audio Over IP is also highly compelling in project studios where the benefits of BYOD (Bring your own device) and the flexibility of a production workflow "independent of cabling" are being increasingly realised by many artists and project studio owners.

Workshop 15 Monday, May 22
11:30 – 13:00 Salon 7 Vienna

3D RECORDING

Chair: **Gregor Zielinsky**, Sennheiser electronic GmbH & Co. KG, Wedemark, Germany

Panelists: *Malgorzata Albinsha-Frank*, Arton, Switzerland
Stephan Thyssen, TVN, Germany

This workshop will have two parts: musical aspects of 3D recording and 3D recording techniques in different situations.

The history of recording has produced great results—no matter at which time of history recordings had been done. Even some old Caruso recordings, played through original Grammophones produce great musical results. However, two dimensional recordings have big problems in transmitting all the magic, detail, and music impact of the music itself. The presentation will show several examples and comparisons between 2 dimensional and 3 dimensional recordings. Beethoven, Mahler, as well as Holst examples will be played and discussed. Also, aspects of upmixing from Stereo to 3D will be discussed and presented by audio examples.

Different situations need different answers. The presentation takes you through different places and different music or sport situations. 3D setups for large orchestra including digital mics are shown. Also, some jazz recordings, chamber music, and even Premier league recordings will be played and explained. Finally a live recording with stars like Joe Walsh, Alice Cooper, and Abe Laboriel will be played and explained.

Pro Sound Expo Monday, May 22
12:00 – 12:30 PSE Stage

AES67 – WHAT IS IT, WHAT CAN IT DO AND HOW CAN IT BE USED

Presenter: **Andreas Hildebrand**, Ravenna/ALC Networks

AES67 is an AES Standard for Audio over IP interoperability, published in 2013. The presentation will explain, how it works and what it can do in certain application scenarios. It also explains its short-comings and how to get along with them.

Monday, May 22 **12:00** **Salon 15 Paris**

Technical Committee Meeting on Audio for Telecommunications

Monday, May 22 **12:00** **Salon 16 Riga**

Standards Committee Meeting SC-04-04 Microphone Measurement and Characterization

Pro Sound Expo Monday, May 22
12:30 – 13:00 PSE Stage

STATE OF THE ART AUDIO OVER IP SOLUTIONS IN PROFESSIONAL AUDIO APPLICATIONS

Presenter: **Jeff Barryman, Nico Lewis**, Bosch/RTS

- When is a standard really an open public standard?
- Positioning of AES 67
- How important is control in professional audio applications?
- Positioning of AES 70
- What is Bosch´s / RTS´s approach to deliver state of the art technology?

Session P18 **Monday, May 22**
13:00 – 15:30 **Salon 1 Moscow**

LISTENING TESTS AND PSYCHOACOUSTICS 2

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

13:00

P18-1 Sensory Profiling of High-End Loudspeakers Using Rapid Methods—Part 2: Projective Mapping with Expert and Naïve Assessors—*Davide Giacalone,*[1] *Maciej Nitkiewicz,*[1] *Samuel Moulin,*[2] *Torstein Boðason,*[3] *Jakob Lund Laugesen,*[4] *Søren Bech*[2,3]
[1]University of Southern Denmark, Odense, Denmark
[2]Bang & Olufsen, Struer, Denmark
[3]Aalborg University, Aalborg, Denmark
[4]University of Copenhagen, Frederiksberg, Denmark

This is the second of a series of papers evaluating the efficiency of rapid sensory profiling methodologies in the audio field [1]. The present paper introduces projective mapping [2] as a method for perceptual audio evaluation and demonstrates its application for discrimination and description of a set of high-end loudspeakers. Additionally, the suitability of the method with both experts and naïve assessors was evaluated. The results showed a successful discrimination between the loudspeakers with the main differences primarily associated to bass strength and bass depth. A high degree of agreement was observed between perceptual configurations obtained separately by the expert and the naïve assessors, though the former outperformed the latter in the descriptive part of the method.
Convention Paper 9775

13:30

P18-2 Potential Audibility and Effects of Ultrasonic Surveillance Monitoring of PA and Life Safety Sound Systems—*Peter Mapp*, Peter Mapp Associates, Colchester, UK

Ultrasonic surveillance monitoring, to check the operational integrity of PA and Emergency Communication Systems, has been in existence for over 40 years—particularly in Europe. Since its inception, there has been debate as to the potential audibility that these systems may have. As the vast majority of PA systems engineers and designers have not heard or experienced any effects, is has generally been assumed that the general public do not either. Recently however, concern has been raised and claims of ill effects have been reported. There is however, little or no data as to the ultrasonic sound levels that PA systems actually emit. The paper discusses the results of an initial survey of ultrasound radiated by a sample of some 50 PA systems and compares the results with a number of international standards—there currently being little or no specific guidance. The paper reviews the technology involved, typical emission levels and concludes by making a number of recommendations to assist with the control of ultrasonic emissions from PA systems that should help to mitigate unintended side effects.
Convention Paper 9776

14:00

P18-3 Pink Noise Formant Bandwidth Discrimination— *Tomira Rogala*, Fryderyk Chopin University of Music, Warsaw, Poland

This paper presents the results of the third part of an experiment aimed to determine discrimination thresholds for timbre of pink noise modified by a formant. The investigated parameter was the Q factor ($Q=f/\Delta f$). The $Q=3$ was used as a reference and the comparison stimuli had $Q>3$. A 3AFC test paradigm was used. The listeners, who were tonmeisters and non-musicians, were asked to indicate which noise burst in each group of three was a different one. The results indicate that: (1) the Q discrimination threshold as a function of formant frequency has a U shape, (2) tonmeisters better discriminate Q changes than non-musicians, and (3) all listeners improved their scores with practice. Above results are consistent with those reported previously.
Convention Paper 9777

14:30

P18-4 The Influence of Program Material on Sound Quality Ratings of In-Ear Headphones—*Sean Olive, Todd Welti, Omid Khonsaripour*, Harman International, Northridge, CA, USA

A listening test was conducted to identify music programs that provide sensitive, discriminating, and reliable ratings for in-ear (IE) headphone evaluations. Ten trained listeners gave sound quality ratings for eight models of IE headphones using ten different music programs. A virtualized headphone method was used to provide double blind, controlled presentations in which headphone leakage effects were monitored and eliminated. The main effect on the sound quality ratings was due to headphones while the program produced no significant effects or interactions. However, certain programs produced more discriminating and reliable ratings than other programs, the key factor being the bandwidth of the program's spectral content, and the subject's familiarity with it. As expected, the amount of bass content in each program tended to influence the ratings of headphones that had too much or too little bass output in their measured frequency response.
Convention Paper 9778

15:00

P18-5 Audio Quality Evaluation in MUSHRA Tests—Influences between Loop Setting and a Listeners' Ratings—*Nadja Schinkel-Bielefeld*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

In many listening tests for audio quality evaluation the listeners have the possibility to set loops, meaning they can focus on a smaller part of the audio signal and listen to that repeatedly. In previous papers we already showed that experienced listeners set more loops and that learning to set loops increases the ability of the listener to perceive artifacts. Now we analyze to what extent these loops chosen by the listener vary from listener to listener and whether the ratings are influenced by the choice of loops of the listener. We show that—depending on the stimulus—listeners who set different loops may also rate significantly different.
Convention Paper 9779

Workshop 16 **Monday, May 22**
13:00 – 14:00 **Salon 4+5 London**

AES67 INTEROPERABILITY TESTINGS— THE PLUG FEST REPORTS

Chair: **Nicolas Sturmel**, Merging Technologies

Panelists: *Bruce Olson*, Olson Sound Design
Greg Shay, TELOS
Peter Stevens, BBC

In order to assess the state of implementation of the AES67 Standard on High performance Streaming Audio-over-IP, the AES has organized a series of Interoperability testings (Plug Fests) in Munich (IRT, 2014), Washington (NPR, 2015), and London (BBC, 2017). This session will present the results as well as the progression in the overall interoperability between the plug fests. Specific issues that appeared during those event will also be highlighted: network setup, PTP, usual bugs and standard interpretation errors.

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

Workshop 17 **Monday, May 22**
13:00 – 15:00 **Salon 7 Vienna**

CAPTURING SOUND FOR 360VR

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

Panelists: *Enda Bates*, Trinity College Dublin, Dublin, Ireland
Gavin Kearney, Univerity of York, York, UK
Henrik Oppermann, Visualise
Tom Parnell
Ulli Scuda, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

The rapid rise of VR is increasing the need for 360-deg binaural audio as well as video in order to provide users with a fully immersive and realistic experience. Although the theory and workflow for binaural object rendering is relatively well understood now, optimal microphone techniques for capturing acoustic sound for VR have not been fully discussed yet. This workshop invites recording experts from the industry and academia to discuss the theories and practices of various microphone techniques for capturing 360-deg VR audio, including First-Order Ambisonics (FOA), Higher-Order Ambisonics (HOA), Equal Segment Microphone Array (ESMA), and some of the currently available proprietary techniques. The panels will discuss the pros and cons of each technique and its suitability for different applications. The session will also provide practical examples of binaural recordings made using the techniques to be discussed.

This session is presented in association with the AES Technical Committee

Pro Sound Expo **Monday, May 22**
13:00 – 13:30 **PSE Stage**

DANTE DOMAIN MANAGER, THE NEXT STEP IN AOIP

Presenter: **Julian Carro**, EMEA System Solution Account Director, Audinate

[abstract unavailable]

Monday, May 22 **13:00** **Salon 15 Paris**

Technical Committee Meeting on Coding of Audio Signals

Session EB6 **Monday, May 22**
13:30 – 14:45 **Salon 2+3 Rome**

RECORDING, LIVE SOUND, EFFECTS, OTHER

Chair: **Alfred J. Svodobnik**, MVOID Group, Karlsruhe, Germany

13:30

EB6-1 **The DFA Fader: Exploring the Power of Suggestion in Loudness Judgments**—*Jack Haigh, Malachy Ronan*, University of Limerick, Limerick, Ireland

Anecdotal evidence suggests that when performers request loudness increases in their on-stage monitoring device, feedback regarding task completion is sometimes sufficient for the performer to perceive a loudness change. This is colloquially known as a DFA fader. Given the dearth of empirical evidence, qualitative interviews were conducted with live sound engineers to investigate the type of feedback required to successfully deliver a suggestion of a loudness change. Following this, 22 participants completed a paired comparison listening experiment to determine whether verbal suggestions produce perceived loudness changes. The experimental results demonstrate a significant difference between participants receiving a verbal suggestion and those that did not in 12 out of 20 presentations. These results support the use of verbal suggestions to convey loudness increases in live sound contexts.
Engineering Brief 336

13:45

EB6-2 **Quantization Noise of Warped and Parallel Filters Using Floating Point Arithmetic**—*Balázs Bank,*[1] *Kristóf Horváth*[2]
[1]Budapest University of Technology and Economics, Budapest, Hungary
[2]Prolan Process Control Co., Budapest, Hungary

For audio filter and equalizer design it is desirable to take into account the frequency resolution of hearing. Therefore, various specialized filter design methodologies have been developed, from which warped and parallel filters are particularly appealing options due to their simple design and good approximation properties. This paper compares the quantization noise of two different warped IIR implementations with that of fixed-pole parallel filters in single-precision floating point arithmetic. It is shown by simulations that the parallel filter provides the best compromise between quantization noise and computational complexity, since it significantly outperforms the series second-order warped IIR implementation in terms of noise performance, while requires less computational resources compared to the original warped IIR structure.
Engineering Brief 337

14:00

EB6-3 **Warped Implementation of Parallel Second-Order Filters with Optimized Quantization Noise Performance**—*Balázs Bank,*[1] *Kristóf Horváth*[2]
[1]Budapest University of Technology and Economics, Budapest, Hungary
[2]Prolan Process Control Co., Budapest, Hungary

Fixed-pole second-order parallel filters provide an effi-

cient way of implementing IIR filters with a logarithmic frequency resolution. However, the fine frequency resolution needed at low frequencies can only be achieved by poles near the unit circle. This may lead to large roundoff noise at low frequencies when the filters are implemented using bit-depths of 24 bits or lower in fixed-point arithmetic. This paper investigates the performance improvement when the parallel second-order sections are implemented as warped IIR filters. In addition, an analytical expression is given for computing the warping parameter as a function of the pole location of the original second-order section so that the quantization noise power is minimized.
Engineering Brief 338

14:15

EB6-4 Power Out of Thin Air: The Harvesting of Acoustic Energy—*Charalampos Papadokos, John Mourjopoulos*, University of Patras, Patras, Greece

Recent evolution in Acoustic Energy Harvesting (AEH) indicate that beyond its communication function, sound can be a potential energy resource for powering contemporary and future applications operating in the range nW - mW. Acoustic energy can be either ambient or produced via speech and music reproduction, portable and mobile devices, jet and automobile engines, means of transport, electroacoustic transducers, etc. This work provides a short review of relevant studies in the art and focuses on AEH inside closed-box loudspeaker enclosures.
Engineering Brief 339

14:30

EB6-5 Fully Digital Development of Automotive Audio Systems—*Alfred Svobodnik,*[1] *Marc Levasseur,*[1] *Christof Faller*[2]
[1]MVOID Group, Karlsruhe, Germany
[2]Illusonic GmbH, Uster, Switzerland

This paper describes the building blocks of a fully digital development environment for automotive audio systems. The whole development process, including all major engineering disciplines, has been virtualized—up to the realistic audibility of the sound systems by means of auralizations. All building blocks are based on simulations, and thus fully digital prototypes can be used already in the early concept phase. Hence, product quality, i.e., reproduced sound performance, can be assessed, and improved, long before any hardware exists.
Engineering Brief 340

Pro Sound Expo **Monday, May 22**
13:30 – 14:00 **PSE Stage**

UNDERSTANDING LATENCY FROM MICROPHONE TO HEADPHONE IN AOIP SOLUTIONS

Presenter: **Jan Lykke**, NTP

It is a common misconception that AoIP systems add too much latency for it to be used in latency-critical recording situations. The presentation will explain the latency in an AoIP-based recording chain all the way from microphone to headphone.

Monday, May 22 **13:30** **Salon 16 Riga**

Standards Committee Meeting SC-04-03 Loudspeaker Modeling and Measurement

Special Event
LOUDNESS WAR II: THE STREAMING BATTLE
Monday, May 22, 13:45 – 15:15 **Berlin-A**

Moderator: **Florian Camerer**, ORF – chairman of EBU PLOUD

Panelists: *Leslie Gaston-Bird*
Eelco Grimm
Matthieu Parmentier, francetélévisions, Paris, France

Optimum loudness normalization for music streaming: Eelco Grimm, HKU University of the Arts and Grimm Audio)

Major pop music releases have suffered from the Loudness War for almost two decades. Now music streaming surpassed CD as major platform for music sales, there is a big opportunity to end this 'war', because the music of streaming services can be normalized in one central location. As soon as all music is played at an equal level, it makes no sense to issue loud masters anymore and the loudness war ends. Eelco Grimm cooperates with streaming service Tidal to design an optimal loudness levelling algorithm that does not harm the artistic intentions and opens the door to high sound quality for all major artists. He will present the results of his research into all 4.3 million albums of the Tidal database, plus a subject test. The project focusses on two questions: can album normalization also be used succesfully outside the album context? And: which target level should be choosen?

Audio Guidelines for OTT TV and Video Streaming: Leslie Gaston-Bird, University of Colorado - AES administrator
Status of the collaborative work done within the AGOTTVS group chaired by Jim Starzynski (NBC Universal).

Strategies to stream broadcast contents on various platforms: Matthieu Parmentier, France TV – chairman of EBU Audio, co-chair of AES TC-Broadcast and Online Delivery
Production of broadcast contents is now fully constrained by loudness guidelines. Masters are usually aligned to –23 LUFS/-24 LKFS to feed either broadcast, broadband and streaming platforms. Due to the growth of mobile listening, where the overall dynamic needs to be reduced to face the background noise, several solutions arise and affect the The aim of this presentation is to present various strategies to allow the distribution of a same content on various networks and devices.

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

Student and Career Development Event
STUDENT RECORDING CRITIQUES
Monday, May 22, 14:00 – 15:00 Salon 11 (Genelec Demo Room)

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

Students! Come and get tips, tricks, and advice to push your skills to the next level! The Student Recording Critiques are non-competitive listening sessions, in which students get to listen to their recordings and productions on a world-class playback system, and then receive feedback from a panel of renowned industry professionals. Students at any of their studies can sign up to participate. Students should sign up at the student (SDA) booth immediately on arrival at the convention, and deliver stereo 44.1 Khz, 24 bit AIFF or WAVE files to the SDA booth at that time. THE FIRST SESSION IS BEFORE THE FIRST SDA MEETING so come early and prepared! Finalists in the Recording Competition are excluded from submitting to these events so that as many students as possible can benefit from this type of feedback. (Recording competition finalists get this feedback as part of the competition process). These events are generously supported by Genelec.

Pro Sound Expo **Monday, May 22**
14:00 – 14:30 **PSE Stage**

3D IN-EAR MIXING AND MONITORING WORKFLOW

Presenters: **Pascal Dietrich, Phil Kamp**, KLANG:technologies GmbH

Sound engineer Phil Kamp of KLANG:technologies will talk and answer questions about: • binaural sound, binaural monitoring and its advantages; • live and studio applications; • creative aspects of working with binaural sound; • possible future usecases in music production; • walkthrough of KLANG:fabrik and examples of usecases with Dante and MADI.

Monday, May 22 **14:00** **Salon 15 Paris**

Technical Committee Meeting on Microphones and Applications

Workshop 18 **Monday, May 22**
14:30 – 15:30 **Salon 4+5 London**

A UNIVERSAL LANGUAGE FOR AUDITORY INTERACTION—SOUND DESIGN FOR ELECTRONIC DEVICES

Presenter **Myoung woo Nam**, Samsung Electronics, Seoul, Korea

Humans talk. Birds sing. And electronic devices all around us generate sound. The user experience on today's mobile devices provides feedback for action, delivering a message, requesting attention, or confirming an action. it is typically expressed by short beeps or melodies - auditory icons, or earcons. As devices grow more capable, they might sometimes borrow human language directly, providing detailed information such as "power on," "power off." But the Auditory icon must provide a basic level of communication(power on/off, volume up/down, and etc.), that everybody can understand, regardless of their language, race, gender, and age. In this workshop, we will show how the sound design process for Auditory Interaction utilizes musical language, tonal cliche, and real-metaphor to deliver an appropriate message to the user. Examples are quoted from the author's latest work on the newest Samsung Galaxy smartphone, Digital Appliances, and apply to any mobile device where the sound is an essential part of the user experience.

Pro Sound Expo **Monday, May 22**
14:30 – 15:00 **PSE Stage**

FOCUSRITE REDNET—OUR INTERFACES FOR THE DANTE NETWORK AND HOW IT CAN WORK FOR YOU

Presenter: **Dankmar Klein**, Focusrite

RedNet is Focusrite's range of interfaces for the Dante network. Coupling the tried and tested Dante network to Focusrite's enviable audio heritage. RedNet offers a growing number of interfaces and bridges that link analogue and digital I/O – along with additional complex audio topologies like MADI and Pro Tools – to the Dante network. Attendees will have the chance to learn about how RedNet can work for them.

Session P19 **Monday, May 22**
15:00 – 18:00 **Gallery Window**

POSTERS: TRANSDUCERS, SYSTEMS, AND EFFECTS

15:00

P19-1 **Audio Time Stretching with Controllable Phase Coherence**—*Nicolas Juillerat*, University of Fribourg, Fribourg, Switzerland

This paper presents a hybrid audio time stretching technique in which the trade-off between vertical and horizontal phase coherence can be freely controlled by a single parameter. Depending on that parameter, the proposed technique sounds like a time domain technique at one extreme, like a phase-locked vocoder at the other extreme, or anywhere in between. By properly choosing the value of the control parameter, it is possible to manually adjust the algorithm to the characteristics of the audio signal being transformed in order to get an optimal result. Furthermore, appropriate middle values yield good results for a wide range of audio signals with mixed content.
Convention Paper 9780

15:00

P19-2 **Modelling Nonlinearities on Musical Instruments by Means of Volterra Series**—*Lamberto Tronchin,*[1] *Vanna Lisa Coli,*[2] *Francesco F. Gionfalo*[1]
[1] University of Bologna, Bologna, Italy
[2]University of Modena and Reggio Emilia, Modena, Italy

The behavior of the soundboard of electroacoustic tools and musical instruments has being investigated for several years. The modelling of such instruments is fundamental in order to determine their acoustic characterization. The determination of nonlinear features of the sound production and propagation allows the definition of acoustical aspects that can't be reproduced with methods based on linear impulse response. A method that allows approximating nonlinear distortions of musical instruments by exploiting the Volterra series model is presented. A Matlab code has been developed in order to test the method on real world audio signals. Results of applications are presented on a series of different wind instruments. Some sound examples are provided.
Convention Paper 9781
Paper presented by Vanna Lisa Coli

15:00

P19-3 **The Influence of Source Spectrum and Loudspeaker Azimuth on Vertical Amplitude Panning**—*Maksims Mironovs*, *Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Listening tests were conducted to examine the influence of source spectrum and loudspeaker azimuth on the accuracy of vertical amplitude panning. Subjects judged the perceived elevation of the phantom images created using vertical loudspeaker pairs placed at 0° and 30° azimuths. Six sound sources with different spectral characteristics were used: broadband, low-passed and high-passed pink noises as well as speech, bird and tank shot recordings. Results generally indicated that the localization accuracy was poor, however, lower or upper response biases observed in the results were found to be significantly dependent on the target panning angle, the stimuli and the loudspeaker azimuth angle. In particular, the low-passed noise presented from the loudspeakers at 30° azimuth was perceived to be significantly elevated.
Convention Paper 9782

15:00

P19-4 Efficient Natural Sample Calculation for Digital Pulse Width Modulation—*Carsten Wegner*,[1] *Robert Schwann*,[1] *Dietmar Ehrhardt*[2]

[1]CAMCO Produktions- und Vertrieb-GmbH, Wenden, Germany
[2]Universität Siegen, Siegen, Germany

In this paper, an improved algorithm for natural sampling is presented that is suitable for digitally controlled fixed frequency PWM modulators. With only 5 MAC operations, 4 multiplications, and 2 additions, the algorithm calculates both switching times for double-sided 3-level PWM, and offers more than 100 dB between signal and PWM related distortion products for high fidelity audio applications. These features compare well with results published [2–6]. The algorithm can be combined with a noise shaped local feedback for quantized pulse lengths and the digital modulator can be integrated into a global feedback loop.
Convention Paper 9783

15:00

P19-5 Construction of Lightweight Loudspeaker Enclosures—*Herle Bagh Juul-Nyholm*, *Jonas Corfitz Severinsen*, *Henrik Schneider, Niels Henrik Mortensen, Michael A. E. Andersen*, Technical University of Denmark, Kgs. Lygby, Denmark

On the basis of bass cabinets, this paper deals with the problem of reducing loudspeaker enclosure weight. An introductory market analysis emphasizes that lighter cabinets are sought, but maintenance of sound quality is vital. The problem is challenged through experiments and simulations in COMSOL Multiphysics, which indicate that weight reduction and sound quality maintenance is possible by reducing wall thickness and using adequate bracing and lining.
Convention Paper 9784

15:00

P19-6 LAMI: A Gesturally Controlled Three-Dimensional Stage Leap (Motion-Based) Audio Mixing Interface—*Jonathan Wakefield*, *Christopher Dewey*, *William Gale*, University of Huddersfield, Huddersfield, UK

Interface designers are increasingly exploring alternative approaches to user input/control. LAMI is a Leap (Motion-based) AMI that takes user's hand gestures and maps these to a three-dimensional stage displayed on a computer monitor. Audio channels are visualized as spheres whose Y coordinate is spectral centroid and X and Z coordinates are controlled by hand position and represent pan and level respectively. Auxiliary send levels are controlled via wrist rotation and vertical hand position and visually represented as dial-like arcs. Channel EQ curve is controlled by manipulating a lathed column visualization. Design of LAMI followed an iterative design cycle with candidate interfaces rapidly prototyped, evaluated, and refined. LAMI was evaluated against Logic Pro X in a defined audio mixing task.
Convention Paper 9785

15:00

P19-7 OSPW (Open Signal Processing Workstation)—Development of a Stand-Alone Open Platform for Signal-Processing in AV-Networks—*Holger Stenschke*,[1] *Thomas Resch*,[1] *Peter Glaettli*,[2] *Roman Riedl*,[1] *Clemens Fiechter* [1]

[1]FHNW Fachhochschule Nordwestschweiz Musikhoch schulen/Hochchule für Musik, Basel, Switzerland
[2]Studer Professional Audio GmbH, Regensdorf, Switzerland

This paper presents the concept and design of a newly developed stand-alone, fully programmable signal processing platform for networked audio and music applications. In recognition of one of the first successful music DSP computation platforms, the ISPW [1], this prototype was named OSPW | Open Signal Processing Workstation. The first part of this paper describes the project's main objectives. The second part provides an overview of the OSPW system components, along with the technologies in use. The third part outlines proof-of-concept demo applications and gives an outlook as to potential user scenarios.
Convention Paper 9786

15:00

P19-8 Extending Temporal Feature Integration for Semantic Audio Analysis—*Lazaros Vrysis, Nikolaos Tsipas, Charalampos Dimoulas, George Papanikolaou*, Aristotle University of Thessaloniki, Thessaloniki, Greece

Semantic audio analysis has become a fundamental task in contemporary audio applications; consequently, further improvement and optimization of classification algorithms has also become a necessity. During the recent years, standard frame-based audio classification methods have been optimized and modern approaches introduced additional feature engineering steps, attempting to capture the temporal dependency between successive feature observations. This type of processing is known as Temporal Feature Integration. In this paper, the enhancement of statistical feature integration is proposed by extending and extensively evaluating the measures that can be deployed. Under this scope, new functions for capturing the shape of a texture window are introduced and evaluated. The ultimate goal of this work is to highlight the best performing measures for early temporal integration, focusing on simple feature engineering, avoiding complexity, and forming a compact and robust set of meta-features that can improve performance in audio classification tasks.
Convention Paper 9808

Session EB7 — Monday, May 22
15:00 – 18:00 — Gallery Window

POSTERS: TRANSDUCERS, SYSTEMS, AND EFFECTS

15:00

EB7-1 Establishing the Performance of a DIY Tapped Horn Loudspeaker—*Andy Wardle*, University of the Highlands and Islands, Perth College, Perth, UK

A DIY Tapped Horn subwoofer was constructed and driven using modestly priced hardware. The position of the acoustic center, frequency and polar responses were established under pseudo-free field conditions. Polar responses and results were compared to a commercially available device. The DIY product performed comparatively poorly with respect to amplitude and frequency response but displayed comparable polar response. Very low frequencies (<80 Hz) displayed no directionality, with effective pattern control beginning at 100 Hz in line with the cutoff frequency predicted by its mouth area. It was established that if processed appropriately, multiple units would provide additional

SPL and lower the frequency of pattern control providing a viable alternative to more expensive products for small to medium scale outdoor events.
Engineering Brief 341

15:00

EB7-2 The Influence of the Passive Electronic Components Quality on the Electroacoustic Parameters of the Audio Devices—*Maciej Sabiniok*, Wroclaw University of Science and Technology, Wroclaw, Poland

A large group of young and inexperienced electronics engineers interested in building audio devices asked how the quality of the passive components such as resistors and capacitors affected on the electroacoustic parameters of designed circuits. This group also includes students who are the members of the Polish Student Section of the Audio Engineering Society at Wroclaw University of Science and Technology willing to work with audio electronics and obtain the best possible quality of the constructed equipment. The aim of this paper is to investigate the impact of the varying passive components quality into the audio circuits performance. The results will allow students to know the limitations related to the choice of passive components.
Engineering Brief 342

15:00

EB7-3 Design of a Digitally Controlled Graphic Equalizer—
Marcelo Herrera Martinez,[1] *Dario Alfonso Páez Soto,*[2] *Jonnathan Montenegro Niño,*[2] *Carlos Mauricio Betancur Vargas,*[3] *Vladimir Trujillo Olaya*[3]
[1]University of Iceland, Reykjavik, Iceland
[2]Universidad de San Buenaventura sede Bogotá, Bogotá, Colombia
[3]Universidad de San Buenaventura sede Cali, Cali, Colombia

This article deals with the design of a digitally audio controller for use in general applications. The goal is to create a 10-band graphic equalizer of which the signal gain or attenuation in every octave band is controllable by a smartphone /tablet application. The application provides a user interface to enhance perceptive audio quality intuitively. Making the equalizer digitally controllable by an app eliminates the necessity of manually adjusting the equalizer faders, thus the need of the presence of a musician/engineer at the location of the equalizer is removed. Preset configurations are easily activated in the equalizer hardware with only one touch within the app. Further testing and optimization efforts are required for the validation of the system.
Engineering Brief 343

15:00

EB7-4 Design of an Algorithm for VST Audio Mixing Based on Gibson Diagrams—*Belman Jahir Rodriguez Nino,*[1] *Marcelo Herrera Martinez*[2]
[1]Universidad de San Buenaventura sede Bogotá, Bogotá, Colombia
[2]University of Iceland, Reykjavik, Iceland

This project consists on the creation of a plugin on the Ableton Live platform, with the aim of providing visually the audio mixing process in real-time. The software programming is developed on Max for Live—a program to establish the link between Max Msp and Ableton Live. The plugin is assigned for each channel with the aim of visualizing the correspondent sound to a "sphere" object on a 3D window and there to observe the variations in real time of loudness, panning, and frequency analysis based on David Gibson's interpretation in his book *The Art of Mixing*.
Engineering Brief 344
eBrief presented by Marcelo Herrera Martinez

Tutorial 18 **Monday, May 22**
15:00 – 16:00 **Salon 7 Vienna**

USING BINAURAL AUDIO TO INCREASE ACCESSIBILITY TO FILM AND TELEVISION

Presenters: **Mariana Lopez**
Gavin Kearney, University of York, York, UK

Audio Description (AD) is a pre-recorded verbal commentary that is added to a film or television program to make visual elements clearer to visually impaired audiences. One of the disadvantages of AD is that the addition of a layer of verbal commentary means that elements from the original soundtrack are masked and valuable information on the film as well as part of the intended engagement is lost. The Enhancing Audio Description project proposes to use binaural audio to reduce the number of verbal descriptions used for accessibility by using accurately placed sound elements to give audiences information on the position of characters and objects in space as well as information on cinematic elements such as high and low camera angles and shots.

Special Thanks: In this session we are using headphones from http://silentdisco.de

Monday, May 22 **15:00** **Salon 16 Riga**

Standards Committee Meeting SC-02-01 Digital Audio Measurement Techniques

Session P20 **Monday, May 22**
15:30 – 16:30 **Salon 1 Moscow**

EDUCATION IN AUDIO

Chair: **John Krivit**

15:30

P20-1 Audio Education: Audio Recording Production Students Report Skills Learned or Focused on in Their Programs—*Doug Bielmeier*, Indiana University-Purdue University, Indianapolis, IN, USA

Previous research polled employers, new hires, and educators in the audio industry to identify what skills were most important, what skills new hires had, and what skills educators focused on in Audio Recording Production (ARP) programs. This study, the Skills Students Learned (SSL) survey, polled 40 students from the U.S. and aboard to identify skills learned at ARP programs. Via an online mixed methods survey instrument, students reported their skill level before and after attending a formal ARP program. In the quantitative section, students reported an improvement in all skill levels upon completing their ARP training. In the qualitative section, students reported job specific communication skills and in-depth technical skills missing from their programs and personal skill sets. This paper recommends the infusion of these skills into existing ARP curriculum.
Convention Paper 9787
This paper has been withdrawn

16:00

P20-2 Facilitating Online International Student Collaborations Through Sound Design—*Kenneth B. McAlpine,[1] Robert Steel[2]*

[1]Abertay University, Dundee, UK
[2]DePaul University CDM, Chicago, IL, USA

Cultural exchange and internationalization have grown hugely in significance within higher education in the last few years. In the broadest sense, this agenda is about preparing students for living in and contributing to an increasingly connected global society. At a time when the political and social trend seems to be towards exclusionism, exposing students to a vibrant blend of ideas, opinions and experiences within the stimulating yet safe space of university resonates all the more strongly. Historically, however, it has been difficult to encourage students to participate fully, particularly with regard to student mobility and studying abroad. Socio-economic and cultural factors play an important role here. Abertay and DePaul are both committed to widening participation and have a high proportion of first-generation students from the lowest socio-economic groups. Consequently, less than five percent of students at DePaul study abroad, and Scotland has one of the lowest student mobility rates in Europe, thus limiting opportunities for students to situate their learning within a global context. Recent developments in digital communications and platform sharing technologies have allowed universities to explore online collaboration and virtual exchange, but that raises new challenges. In particular, how do you embed a sense of genuine cultural exchange between students who are geographically remote and still enmeshed in their local culture? This paper explores one response to the problem, using a collaborative sound design project to build a strong sense of cultural exchange between students located at two universities, Abertay University in Scotland and DePaul in the USA.
Convention Paper 9788
Paper presented by Robert Steel

Convention Paper 9789 was withdrawn

Session P21 Monday, May 22
15:30 – 17:30 Salon 2+3 Rome

MISCELLANEOUS 1

Chair: **Elena Shabalina**, d&b audiotechnik GmbH, Backnang, Germany

15:30

P21-1 Far-Field Noise Prediction for Open-Air Events. Part 1: Background and Propagation Models—
Matthias Christner,[1] Jochen Schaal,[2] Dieter Zollitsch,[2] Elena Shabalina,[1] Daniel Belcher [1]

[1]d&b audiotechnik GmbH, Backnang, Germany
[2] SoundPLAN International LLC, Backnang, Germany

In the past the main focus of loudspeaker manufacturers and sound system designers was to provide the best possible sound quality for the listeners. With the number of these events increasing along with the number of the affected inhabitants and their complaints, the focus is shifting towards including predicting and minimizing the noise in the neighborhood in the planning of an open air event. The presented calculation method is designed to close the gap between the environmental noise propagation models and complex loudspeaker system models. The implementation of the Nord2000 and ISO 9613-2 propagation models were extended to include complex loudspeaker setups. This paper presents the motivation and the theoretical background of the new prediction method.
Convention Paper 9790
Paper presented by Elena Shabalina

16:00

P21-2 Noise Prediction Software for Open-Air Events Part 2: Experiences and Validation—*Daniel Belcher, Matthias Christner, Elena Shabalina*, d&b audiotechnik GmbH, Backnang, Germany

The prediction and minimization of noise in the neighborhood during the planning phase of open-air events is becoming more important. The common available software for calculating environmental noise did not consider complex summation of sound because typical noise sources in traffic or industry are not coherent. State of the art sound systems with arrays of loudspeakers and subwoofers effectively use coherence in order to achieve their high directivity. The propagation models were not only extended for complex summation, but also for import of complex data from a system design tool (see Part 1 for details). This paper presents experiences with the simulation software NoizCalc in the field since its launch, its validation by means of a comparison with accompanying measurements and a derivation of uncertainty, in order to set the informative value of a prediction into context
Convention Paper 9791

16:30

P21-3 Development and Evaluation of an Interface with Four-Finger Pitch Selection—*Henrik von Coler,[1] Gabriel Treindl,[1] Hauke Egermann,[2] Stefan Weinzierl[1]*

[1]Technical University of Berlin, Berlin, Germany
[2]University of York, York, UK

This paper presents the development and evaluation of an interface for electronic musical instruments, designed for controlling monophonic synthesizers. The hand-held device allows the pitch selection with one hand, using four valve-like metal mechanics and three octave switches. Note events are triggered with a wooden excitation pad, operated with the second hand. The sensors are equipped with an advanced aftertouch, which enables expressive playing. In a user experiment, the controller is compared to a MIDI keyboard, regarding the reaction time and error rate in simple tasks. Results show no significant difference in the response time but a higher error rate for the novel interface. Outcome of this work is a list of necessary improvements and a plan for further experiments.
Convention Paper 9792

Tutorial 19 Monday, May 22
15:30 – 16:30 Salon 4+5 London

CONTROL SYSTEM AND ELECTROACOUSTICAL CONSIDERATIONS FOR LARGE-SCALE LOUDSPEAKER ARRAYS: PAST, PRESENT & FUTURE

Presenter: **David Scheirman**

A timeline of live-performance system design evolution over four decades will be highlighted. In addition to a historical review of control and monitoring processes, this presentation bridges the gap from control-only networks to network digital audio, noting migration paths to beam-steerable line array elements that are now described as network endpoint devices. Tutorial also presents various loudspeaker enclosure and multi-box array topologies in use over time, as a broad-spectrum overview of technical developments taking place since the AES 6th International Conference (Sound Reinforcement, Nashville, 1988) and the AES 13th International Conference (Computer-Controlled Sound Systems, Dallas, 1994). Each of these landmark events included content that foreshadowed the development of today's modern high-powered loudspeaker arrays that incorporate beam-steering technology. Recently-emerging trends will be examined, and potential future developments contemplated. Of potential interest to sound reinforcement technicians and system operators, installed-system designers, rental sound service company providers, and live-sound equipment product development engineers.

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

Student and Career Development Event
RECORDING COMPETITION—PART 2
Monday, May 22, 15:30 – 18:00 Berlin-A

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

15:30 Category 3— Sound for Visual Media

16:30: Category 4—Modern Studio Recording & Electronic Music

Tutorial 20 Monday, May 22
16:00 – 18:00 Salon 7 Vienna

CREATING AUDIO FOR VIRTUAL REALITY APPLICATIONS

Presenters: **Tom Ammermann**, New Audio Technology GmbH, Hamburg, Germany
Robert Schulein, RBS Consultants, Schaumburg, Il, USA

Audio has always been an integral element in the creation of more realistic audio-visual entertainment experiences. With the evolution of personal motion tracking 3D imaging technologies, entertainment experiences are possible with a higher degree of cognition, commonly referred to as virtual reality. The quest for more engaging user experiences has raised the challenge for more compelling audio. Elements of binaural hearing and sound capture have come to play a central role in existing and evolving production techniques. This tutorial will cover the elements of binaural audio as they relate to producing compelling entertainment and educational content for virtual reality applications. Specific areas to be covered with support audio and 3D anaglyph video demonstrations include: audio for games, music entertainment, radio drama, and music education. Audio production tools including binaural and ambisonic capture microphone systems, with and without motion capture will be presented and demonstrated. The tutorial will also cover aspects relating to creating high-quality content with professional workflows using common tools and DAWs.

Special Thanks: In this session we are using headphones from http://silentdisco.de

Monday, May 22 16:00 Salon 15 Paris

Technical Committee Meeting on Perception and Subjective Evaluation of Audio Signals

Workshop 19 Monday, May 22
16:30 – 18:00 Salon 4+5 London

MODERN HYBRID AUDIO CODING

Chair: **Jürgen Herre**, International Audio Laboratories Erlangen, Erlangen, Germany

Panelists: *Sascha Dick*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Andreas Niedermeier, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Heiko Purnhagen, Dolby Sweden AB, Stockholm, Sweden

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

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

Monday, May 22 16:30 Salon 16 Riga

Standards Committee Meeting SC-07-01 Metadata for Audio

Session EB8 Monday, May 22
17:00 – 18:00 Salon 1 Moscow

LECTURE: SPATIAL AUDIO—BINAURAL 2

Chair: **Juha Backman**, Hefio Oy, Espoo, Finland

17:00

EB8-1 Motion-to-Sound Latency Measurement Procedure for VR Sound Reproduction—*Jorgos Estrella, Jan Plogsties*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

In the last couple of years, virtual auditory displays have finally reached the consumer market as part of emerging VR technologies. One of the challenges VR technology providers have to face is to reach affordable low motion-to-sound latency. Low latency is a very important factor while aiming towards immersive spatial sound reproduction. In this e-Brief a motion-to-sound latency measurement approach is proposed. This method employs a simplified parallel system running externally as reference.

Here, a second head-orientation sensor is used to modulate a signal generator. Correlation analysis between the generated signal and the output signal of the device under test are used to assess latency.
Engineering Brief 345

17:15

EB8-2 Flexible Python Tool for Dynamic Binaural Synthesis Applications—*Annika Neidhardt, Florian Klein, Niklas Knoop, Thomas Köllmer*, Technical University of Ilmenau, Ilmenau, Germany

In this report we present an open source tool for real-time dynamic binaural synthesis implemented in Python on top of PyAudio. The core is an efficient implementation of the uniformly partitioned convolution with the overlap-save approach. The dynamic and interactive reproduction of spatial audio scenes has become a common requirement in science and industry. Use cases are various, reaching from listening tests considering head rotation to complex reproduction scenarios for augmented or virtual reality, e.g., in combination with head mounted displays. With Python as a flexible and easy to learn programming language, PyBinSim offers great value in research and teaching of binaural synthesis. Source code, examples and documentation are available online.
Engineering Brief 346

17:30

EB8-3 A Self-Calibrating Earphone—*Juha Backman, Tom Campbell, Jari Kleimola, Marko Hiipakka*, Hefio Oy, Espoo, Finland

A self-calibrating system estimates the acoustical transfer function from sound pressures at the entrance of the ear canal to sound pressures at the eardrum: An earphone plays a broadband sound into the auditory meatus and an in-ear microphone then receives the sound at the entrance of the ear canal. Parenthetically, assessing calibration, results showed that spectral analysis of recordings of this signal is replicable to within 3 dB from 0.5 to 22 kHz for each given ear. A digital signal processing unit calculates an individualized filter from that signal. The calculated filter neutralizes the transfer function via software, which controls the digital signal processing unit's output into the earphones whilst playing media.
Engineering Brief 347

17:45

EB8-4 End-To-End Process for HRTF Personalization—*Tomi Huttunen, Antti Vanne*, OwnSurround Ltd., Kuopio, Finland

The personalization of the head-related transfer functions (HRTFs) improves externalization and spatialization in headphone listening. The accurate measurement of an individual HRTF is time-consuming and complicated that has led to increased interest towards simulation based HRTF acquisition. The main challenge for simulations has been the lack of the fast and simple method to generate the three-dimensional (3D) geometry of the head and pinnae. On the other hand, a numerical solution of the 3D wave equation that characterizes the HRTF has been considered computationally demanding. We introduce an end-to-end process from the acquisition of the geometry to use of the personalized HRTFs in several applications. Results from the preliminary listening tests and future improvements are also discussed.
Engineering Brief 348

Monday, May 22 17:00 Salon 15 Paris

Technical Committee Meeting on Spatial Audio

Special Event
ORGAN CONCERT—"JAZZ MEETS CLASSICAL"
Monday, May 22, 20:00 – 21:30
St. Matthias
Goltzstraße 29 (am Winterfeldtplatz), Berlin

Performers: *Francis Rumsey, Sigrid Erbe-Sporer*

AES conventions have entertained delegates with an organ concert for many years, but at the 142nd in Berlin there'll be a new twist. This time classical themes will morph into jazz style on one of the city's finest pipe organs.

Included in Francis' first half program: Mozart's Fantasia in F minor and Widor's Finale from the 6th Organ Symphony. Morphing into jazz style for the second half, Sigrid starts with "Mozart Changes" composed by Zolt Gárdonyi, which starts in classical style and gradually shifts into jazz mode. Sigrid continues with the "Suite Jazzique" of Johannes Matthias Michel, inspired by the well-known "Suite Gothique" of Leon Boellmann.

The organ at St. Matthias was built in 1958 by the firm of Romanus Seifert & Son. From 1972–4 it was enlarged by Seifert to become what was then the largest organ in Berlin, containing 109 ranks and 74 stops. In 1993 it was subject to a general overhaul during the church renovation, and a new console was built by Stockmann. Thanks to recent additions in 2008–9 it now has an extensive combination system and a few more ranks, bringing the specification to 111 ranks and 76 stops, arranged on four manuals and pedal. More information can be found at http://www.die-orgel-seite.de/specials/stmatthias/stmatthias_e.htm, and http://st-matthias-berlin.de/musik/die-st-matthias-orgel.html

Session P22 Tuesday, May 23
9:00 – 10:30 Salon 1 Moscow

SPATIAL AUDIO—CHANNEL BASED

Chair: **Akio Ando**, University of Toyama, Toyama, Japan

9:00

P22-1 Optimization of Temporally Diffuse Impulses for Decorrelation of Multiple Discrete Loudspeakers—*Jonathan B. Moore, Adam Hill*, University of Derby, Derby, UK

Temporally diffuse impulses (TDIs) were originally developed for large arrays of distributed mode loudspeakers to achieve even radiation patterns. This initial investigation evaluates the performance of TDIs in terms of the reduction of low frequency spatial variance across an audience area when used with conventional loudspeakers. A novel variable decay windowing method is presented, allowing users control of TDI performance and perceptibility. System performance is modelled using an anechoic and an image source acoustic model. Results in the anechoic model show a mean spatial variance reduction of 42%, with a range of source material and using the optimal TDI generation methodology. Results in the image source model are more variable, suggesting that coherence of source reflections reduces static TDI effectiveness.
Convention Paper 9794

9:30

P22-2 Seamless Spatial Calibration of Multichannel Sound

Systems—*Antoine Peillot*, Gibson Innovations, Leuven, Belgium

In a multichannel audio setup, spatial calibration aims at delivering an optimal sound experience at the listening position. Since the listener is not expected to stand at the focal point between the surrounding speakers, usually arbitrarily placed, it is needed to focus the sweet spot at the listening position. To do so, proper gains and delays need to be applied to each channel composing the audio setup. The work presented in this paper provides a solution to automatically estimate and apply these parameters. It is based on joint user and speaker localization in a seamless way thanks to microphones embedded in surrounding speakers. A patent is currently pending for the spatial calibration method described in this paper.
Convention Paper 9795

10:00

P22-3 Extraction of Interchannel Coherent Component from 3D Multichannel Audio—*Yuta Hashimoto, Hiroki Tanaka, Akio Ando*, University of Toyama, Toyama, Japan

Extraction of interchannel coherent component is a useful method that is applicable to the improvement of blurry sound image and setting up an upmix system. In this paper we propose a new method that extracts the component from three-dimensional (3D) multichannel audio signal. Such a signal sometimes has a negative cross correlation among channels because it includes independent sounds propagated from different directions. To handle this problem, a new method is proposed to estimate the component of one channel signal by the other channel signals having positive correlations with the signal in each subband. The experimental result showed that the estimation of the component by selected channel signals brought better performance than that by all channel signals.
Convention Paper 9796

Tutorial 21 **Tuesday, May 23**
9:00 – 10:30 **Salon 4+5 London**

SYSTEM DESIGN USING ADAPTIVE LOUDSPEAKER CONTROL

Presenter: **Gregor Höhne**

Progressing miniaturization and portability introduce new challenges for loudspeaker design. However, demands like higher efficiency and reduced weight can be achieved by combining digital signal processing and optimized transducer design. The tutorial gives an introduction into nonlinear adaptive loudspeaker control and how it can be used to equalize, stabilize, linearize, and actively protect transducers. This covers the physical background of nonlinear loudspeaker behavior and the resulting demands on a control algorithm with a strong focus on the practical implementation. The discussion includes resulting requirements for amplifier and transducer design, like power demands and advantages of dc-coupling, as well as techniques to evaluate the system performance.

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

Workshop 20 **Tuesday, May 23**
9:00 – 10:30 **Salon 7 Vienna**

VIRTUAL RADIO PROJECTS

Chair: **Jamie Laundon**, BBC Design and Engineering

Panelists: *Fredrik Bergholtz*
Martin Dutasta, Digigram
Chris Roberts, BBC Design and Engineering

Audio networking allows new virtualization use-cases in broadcast operations reducing costs and allowing new workflows for content contribution and distribution in radio. This workshop will present 3 virtual radio projects at Swedish Radio, BBC, and Digigram, all enabled by the use of Virtualization, IT infrastructures and software based systems.

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

Monday, May 22 **9:00** **Salon 16 Riga**

Standards Committee Plenary Meeting

Session P23 **Tuesday, May 23**
10:30 – 12:30 **Salon 1 Moscow**

RECORDING AND LIVE SOUND

Chair: **Will Howie**, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canada

10:30

P23-1 Subjective Evaluation of Orchestral Music Recording Techniques for Three-Dimensional Audio—*Will Howie,*[1,2] *Richard King,*[1,2] *Denis Martin,*[1,2] *Florian Grond*[2,3]
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canada
[3] The Input Device and Music Interaction Laboratory, Montreal, Quebec, Canada

A double-blind study was conducted to evaluate a recently developed microphone technique for three-dimensional orchestral music capture, optimized for 22.2 Multichannel Sound. The proposed technique was evaluated against a current 22.2 production standard for three-dimensional orchestral music capture, as well as a coincident, higher order ambisonics capture system: the Eigenmike. Analysis of the results showed no significant difference in listener evaluation between the proposed technique and the current production standard in terms of the subjective attributes "clarity," "scene depth," "naturalness" "environmental envelopment," and "quality of orchestral image."
Convention Paper 9797

11:00

P23-2 Formal Usability Evaluation of Audio Track Widget Graphical Representation for Two-Dimensional Stage Audio Mixing Interface—*Christopher Dewey*, *Jonathan Wakefield*, University of Huddersfield, Huddersfield, UK

The two-dimensional stage paradigm (2DSP) has been suggested as an alternative audio mixing interface (AMI).

This study seeks to refine the 2DSP by formally evaluating graphical track visualization styles. Track visualizations considered were text only, circles containing text, individually colored circles containing text, circles color coded by instrument type with text, icons with text superimposed, circles with RMS related dynamic opacity, and a traditional AMI. The usability evaluation focused on track selection efficiency and included user visualization preference for this micro-task. Test subjects were instructed to click five randomly selected tracks for a six, sixteen, and thirty-two track mix for each visualization. The results indicate text only visualization is best for efficiency however test subjects preferred icons and traditional AMI.
Convention Paper 9798

11:30

P23-3 In-Ear vs. Loudspeaker Monitoring for Live Sound and the Effect on Audio Quality Attributes and Musical Performance—*Jan Berg, Tomas Johannesson, Magnus Löfdahl, Arne Nykänen*, Luleå University of Technology, Luleå, Sweden

A successful performance of live music is dependent on how well musicians can hear themselves and the other members of the ensemble. Sound reinforcement systems can offer monitoring either by on-stage loudspeakers or in-ear headphones. These two monitoring conditions were compared to search for perceived auditory differences that affect parts of musical performance. Four jazz/pop/rock bands made live performances where monitor sound was provided to the musicians. Each band repeated their performance, changing from one monitoring condition to the other. After every performance, the musicians responded to questionnaires covering musical performance and audio quality. Experts also assessed recordings of the performances. Results show that perceived differences exist in audio quality and musical performance between loudspeaker monitors and in-ear headphone monitors.
Convention Paper 9799

12:00

P23-3 Using a Speech Codec to Suppress Howling in Public Address Systems—*David Ditter,*[1,2] *Edgar Berdahl*[3]
[1]Technical University of Berlin, Berlin, Germany
[2]Jünger Audio GmbH, Berlin, Germany
[3] Louisiana State University

Acoustical feedback is present whenever a loudspeaker signal gets redirected to a microphone that feeds its input signal directly or indirectly back into the loudspeaker. If the gain around such a closed feedback loop is close to or higher than unity, unpleasant acoustical artifacts will typically occur and will nearly always lead to a periodic howling sound. Most readers are probably familiar with this noise, which can for example set in when a microphone is accidentally pointed at a speaker. This research project aims to suppress these unwanted effects of acoustical feedback by the insertion of a modified speech coder and decoder into the signal path of the feedback loop. It is demonstrated that the Speex open-source speech codec can be successfully tweaked to increase the maximum stable feedback gain by as much as 3 dB to 7 dB through adjustment of the codec's quality parameter. This enhancement outperforms the simple introduction of shaped noise into the feedback loop and is compared with the performance of a frequency shifter. Tests are conducted using an automated experimental framework for determining the maximum stable gain of a public address system.
Convention Paper 9800

Session P24 **Tuesday, May 23**
10:30 – 12:00 **Salon 2+3 Rome**

MISCELLANEOUS 2

Chair: **Annika Neidhardt**, Technical University of Ilmenau, Ilmaneu Germany

10:30

P24-1 Usability and Effectiveness of Auditory Sensory Substitution Models for the Visually Impaired—*Adam Csapo,*[1] *Simone Spagnol,* [2] *Marcelo Herrera Martinez,* [2] *Michal Bujacz,*[3] *Maciej Janeczek,*[3] *Gabriel Ivanica,*[4] *György Wersényi,*[1] *Alin Moldoveanu,*[4] *Runar Unnthorsson*[2]
[1]Széchenyi István University, Györ, Hungary
[2] University of Iceland, Reykjavik, Iceland
[3]Lodz University of Technology, Lodz, Poland
[4] Politechnica University of Bucharest, Bucharest, Romania

This paper focuses on auditory sensory substitution for providing visually impaired users with suitable information in both static scene recognition and dynamic obstacle avoidance. We introduce three different sonification models together with three temporal presentation schemes, i.e., ways of temporally organizing the sonic events in order to provide suitable information. Following an overview of the motivation and challenges behind each of the solutions, we describe their implementation and an evaluation of their relative strengths and weaknesses based on a set of experiments in a virtual environment.
Convention Paper 9801

11:00

P24-2 Adaptive Audio Engine for EEG-Based Horror Game—*Jordan Craig*, New York University, New York, NY, USA

This paper documents the design and play-testing of a videogame that incorporates electroencephalography (EEG) technology to augment traditional controls. A survival horror game was created using Unity3D. The player navigates the game using conventional keyboard and mouse movement, however, they also wear an Emotiv EPOC headset that transmits their level of calm to the game via OSC. In order to complete the game, the player must remain as calm as possible. An adaptive audio engine was developed to act as an auditory display for this complex parameter in lieu of a distracting visual indicator. Every element of the audio was designed to adapt to the constantly fluctuating value. Procedural audio modules were created in Max, where player EEG data was simultaneously mapped to a myriad of modulators. FMOD Studio was used for non-procedural elements due to its facilitation of real-time control parameters, as well as its integration with Unity3D.
Convention Paper 9802

11:30

P24-3 Real-Time Reverb Reduction for Improved Automatic Speech Recognition in Far-Field—*Adam Kupryjanow, Przemyslaw Maziewski, Lukasz Kurylo, Piotr Lasota*, Intel Technology Poland, Gdansk, Poland

In the paper, methods of real-time reverb reduction based

on Generalized Weighted Prediction Error (GWPE) were presented. It was shown that usage of the proposed audio processing routines highly improve the accuracy of Automatic Speech Recognition (ASR) system namely word error rates (WERs) are reduced 11.36% when the user stands 5 meters from the microphone array. The obtained results are close to the ones that are achieved by the offline GWPE implementation (12.06%). Thanks to optimizations and parameters tuning, computational complexity of the proposed realization of GWPE was highly reduced and it achieves RTFs lower than 1.0 (computation time is shorter than signal duration) when using one core of CPU.
Convention Paper 9803

Tutorial 22 **Tuesday, May 23**
10:30 – 12:15 **Salon 4+5 London**

GO ON! SURPRISE ME! AN INTRODUCTION TO AUDIO AND VIDEO CODING

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

Coded Audio is an essential part of modern audio distribution, such as the internet, film, etc. But how does it work? What is it about a signal that can allow you to reduce its data rate without loss, as in "Lossless Coding"? How can one take advantage of human perception when one does lossy coding such as mpeg? This tutorial will use both video and audio coding to explain the characteristics that allow one to encode such signals at a reduced data rate without any loss of fidelity. It will then go on to explain how one can go about reducing the data rate with the minimum of perceptible distortion.

Workshop 21 **Tuesday, May 23**
10:45 – 11:15 **Berlin-A**

OBJECT-BASED AUDIO: NOW & NEXT

Chair: **Chris Pike**, BBC Research & Development, Salford, UK

Panelists: *Dave de Roure*, University of Oxford, Oxford, UK
Philip J. B. Jackson, University of Surrey, Guildford, Surrey, UK
Paul Morgan, BBC Radio
Michael Weitnauer

After many years of discussion in the AES about object-based audio, we are now seeing these concepts being adopted into mainstream industry standards and services, including the next generation of digital television systems within DVB and ATSC. Developments in IP-based production technology also allow broadcasters to take advantage of the benefits of object-based audio in the production environment. This workshop will give the audience an insight into the opportunities and challenges faced by broadcasters when implementing object-based end-to-end systems. It will also discuss the potential developments of object-based audio systems beyond this current generation, with presentations from two large-scale collaborative research projects investigating the future of object-based audio, as applied to spatial audio and music.

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

Tutorial 23 **Tuesday, May 23**
11:00 – 11:45 **Salon 7 Vienna**

CANCELED

Student and Career Development Event
STUDENT DELEGATE ASSEMBLY MEETING—PART 2
Tuesday, May 23, 12:15 – 14:30 **Berlin-A**

At this meeting the SDA will elect a new vice chair. One vote will be cast by the designated representative from each recognized AES student section in the Europe and International Regions. Judges' comments and awards will be presented for the Recording Competitions and Design Competitions. Plans for future student activities at local, regional, and international levels will be summarized.

Session P25 **Tuesday, May 23**
13:00 – 14:30 **Salon 1 Moscow**

SOUND ZONES

Chair: **Jan Abildgaard Pedersen**, Dynaudio A/S, Skanderborg, Denmark

13:00

P25-1 Amplitude Panning between Beamforming-Controlled Direct and Reflected Sound—*Franck Zagala,*[1,2] *Julian Linke*,[1,2] *Franz Zotter,* [2]*Matthias Frank*[2]
[1]University of Music and Performing Arts Graz, Graz, Austria
[2]Institute of Electronic Music and Acoustics, Graz, Austria

Loudspeaker beamformers such as commercial sound bars can be used to produce narrow beams of sound that mainly reach the listener on distinct reflection paths or the direct path in the room. What happens if such variable directivity loudspeakers create two simultaneous beams of sounds with the same signal, each of which pointing to another acoustic path in the room? What is the resulting perceived direction of such a phantom source, and how do changes of time and level differences in the signal pair affect the result? This paper investigates these questions by a listening experiment that employs an auralized 3rd order source.
Convention Paper 9805

13:30

P25-2 Sound Zones: On the Effect of Ambient Temperature Variations in Feed-Forward Systems—*Martin Olsen,*[1] *Martin Bo Møller*[2,3]
[1]Harman Lifestyle Audio, Struer, Denmark
[2]Bank & Olufsen A/S, Struer, Denmark
[3]Aalborg University, Aalborg, Denmark

The precondition for realizing personal sound zones, relying on multichannel feed-forward control, is the robustness in the characterization of the sound field inside the control regions. Achieving high separation depends on the ability to accurately estimate the acoustic transfer functions from a set of control loudspeakers to the zones. In this paper the assessment of ambient temperature variations is based on a front-to-rear scenario at low frequencies in a car cabin. Experimental studies in a production vehicle show significant performance decrease, when the temperature conditions in the playback situation differ from those present during the setup procedure. The main cause of the mismatch in the two sets of acoustic transfer functions is analyzed and potential compensation strategies are discussed accordingly.
Convention Paper 9806

14:00

P25-3 Assessing the Influence of Loudspeaker Driver Nonlinear Distortion on Personal Sound Zones—*Xiaohui Ma,*[1,2] *Patrick J. Hegarty,*[1] *Jan Abildgaard Pedersen,*[1] *Lars G. Johansen,*[2] *Jakob Juul Larsen*[2]
[1]Dynaudio A/S, Skanderborg, Denmark
[2]Aarhus University, Aarhus, Denmark

The impacts of loudspeaker nonlinear distortion on sound zones are measured in an anechoic chamber. Two loudspeaker arrays, each with four equally spaced drivers, are used to generate two sound zones, one bright and one dark. Acoustic contrast control (ACC) and planarity control (PC) are employed as control methods. A 250 Hz sinusoidal signal is used as stimulus, and the target sound pressure level for the bright zone is 82 dB. Simulations based on measured transfer functions give acoustic contrast of 43.1 dB between the two zones whereas the experimentally measured acoustic contrast is only 32.1 dB for ACC, and 29.3 dB for PC. Nonlinear distortion contributes to this contrast loss according to spectrum measurements. Experiments also reveal that the nonlinear distortion can be controlled through regularization of the control effort; the regularization parameter has an optimal value which can balance the acoustic contrast and nonlinear distortion.
Convention Paper 9807

Tutorial 24 **Tuesday, May 23**
13:00 – 14:30 **Salon 7 Vienna**

OBJECT-BASED AUDIO FOR BROADCASTERS

Presenters: **Matthieu Parmentier**
Lidwine Hô

A tutorial to consider the opportunities of Object-Based Audio for broadcasters: new contents, formats, workflows and broadcasting strategies.

Object-Based Audio for broadcasters: what, why, how? New story-tellings, renewed tools, this tutorial will focus on the whole production chain to better serve the next generation of contents and enhance the end-user experience.

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

Workshop 22 **Tuesday, May 23**
13:00 – 14:30 **Salon 4+5 London**

MICROPHONES - CAN YOU HEAR THE SPECS?

Chair: **Helmut Wittek**, Schoeps

Panelists: *Jürgen Breitlow*, Sennheiser/Neumann
Hans Riekehof-Böhmer, Schoeps
Martin Schneider, Sennheiser/Neumann

There are lots and lots of microphones available to the audio engineer. The final choice is often made on the basis of experience or perhaps just habits. (Sometimes the mic is chosen because of the Looks...). Nevertheless, there is valuable information in the microphone specifications. This tutorial demystify the most important microphone specs and provide the attendee with up-to-date information on how these specs are obtained and understood and how the numbers relate to the perceived sound. It takes a critical look on how specs are presented to the user, what to look and listen for, and what to expect.

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

Session EB9 **Tuesday, May 23**
14:30 – 15:00 **Salon 1 Moscow**

LECTURE: SPATIAL AUDIO, LISTENING TESTS, SYSTEMS

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

14:30

EB9-1 Object-Based Audio in Large Scale Live Sound Reinforcement Controlled by Motion Tracking—*Mario Seideneck*, *Jakob Bergner, Christoph Sladeczek*, Fraunhofer Institute for Digital Media Technology (IDMT), Ilmenau, Germany

This work shows a detailed application of an optical tracking system to control the positioning of sound sources in an object-based audio reproduction system for live sound reinforcement. This need is brought up by live performances with moving actors like operas, musicals or spoken theater. With state-of-the-art object-based audio reproduction systems it is possible to distribute virtual sound sources for improved sound localization within the audience area. To cope with applications of high complexity automated auxiliary systems like motion tracking provide valuable control data and thus enhance the usability of such systems. The presented approach shows a solution with focus on interfaces between systems and devices.
Engineering Brief 349
This e-Brief is presented by Jakob Bergner

14:45

EB9-2 A Basic Study of the Upmix Method for 22.2 Multichannel Sound—*Toru Kamekawa, Atsushi Marui*, Tokyo University of the Arts, Adachi-ku, Tokyo, Japan

The upmix technique to 22.2 multichannel sound from 9 channel, 5 channel, and 3 channel IRs (impulse responses) were studied. The two upmix techniques were used. One is the IRs made from the original IRs converted to phase randomized signals with the same time envelope to original signal and the other is the IRs obtained by simply adding signals of adjacent channels. The experimental stimuli were obtained from these impulse responses convolved to the sound of a xylophone and a female voice recorded in an anechoic room. From the results, there is a tendency of different impression between these two methods, and it is suggested that the phase randomized method may be effective with the case using from less channels.
Engineering Brief 350

15:00

EB9-3 The Dawn of Audiophile Quality Audio on Your Smartphone—*Stefan Gustavsson*, Qualcomm, San Diego, CA, USA

The mobile phone has become the primary device for personal music and multimedia consumption. This increases the focus on audio quality, especially when listening through headphones, providing impetus for a transition from mobile audio being a low quality, best effort music player to one that can be compared to dedicated high audio quality playback systems. The task of delivering audiophile quality music to mobile phone users provides

a unique challenge requiring extremely low power. The latest, highly integrated audio solutions for smartphone chipsets use technologies, architectures, and algorithms that can deliver HIFI audio while still providing attractive power consumption and cost. With this fast improvement in mobile phone audio the system level design and testing methodologies need to keep up.
Engineering Brief 351

15:15

EB9-4 Practical Loudness Measurement and Management for Immersive Audio—*Scott G. Norcross, Marvin Pribadi, Sachin Nanda*, Dolby Laboratories, Inc., San Francisco, CA, USA

Loudness management is an essential and often mandatory aspect for content providers and broadcasters. Regional requirements/guidelines based on Recommendation ITU-R BS.1770 form the basis for the loudness practice in broadcasting. It has recently been revised to support new immersive-channel formats, but not explicitly for object-based audio formats. Object-based audio is currently being delivered over-the-top (OTT) and loudness management must be addressed to meet requirements and provide a good user experience. These new audio formats allow the content to be played back over a larger range of playback configurations, which has the potential for loudness variations. This brief describes loudness measurement and management for currently deployed object-based delivery and shows how legacy playback of this content meets the current broadcasting recommendations.
Engineering Brief 352

15:30

EB9-5 An Analog Audio Sensor Board for Microcontrollers—*Colin Zyskowski, Mauricio de Oliveira*, University of California San Diego, San Diego, CA, USA

This paper presents work on the Audio Sensor Board (ASB), a circuit board designed to serve as an analog interface between audio signals and the digital/analog inputs of common micro-controllers. The ASB allows for low-level manipulation of audio signals so that those signals can be easily used as control parameters, in essence creating a two-channel analog/digital sensor for sound sources. In this paper we layout the various functions of the board, its design, and describe practical purposes for which it has successfully been used.
Engineering Brief 353

Tutorial 25 **Tuesday, May 23**
14:30 – 15:15 **Salon 4+5 London**

PERCEPTUALLY MOTIVATED FILTER DESIGN WITH APPLICATION TO LOUDSPEAKER-ROOM EQUALIZATION

Presenter: **Balázs Bank**, Budapest University of Technology and Economics, Budapest, Hungary

Digital filters are often used to model or equalize acoustic or electroacoustic transfer functions. Applications include headphone, loudspeaker, and room equalization, or modeling the radiation of musical instruments for sound synthesis. As the final judge of quality is the human ear, filter design should take into account the quasi-logarithmic frequency resolution of the auditory system. This tutorial presents various approaches for achieving this goal, including warped FIR and IIR, Kautz, and fixed-pole parallel filters, and discusses their differences and similarities. It also shows their relation to fractional-octave smoothing, a method used for displaying transfer functions. With a better allocation of the frequency resolution and filtering resources, these methods require a significantly lower filter order compared to straightforward FIR and IIR designs at a given sound quality.

This session is presented in association with the AES Technical Committees on Loudspeakers and Headphones and Signal Processing

Workshop 23 **Tuesday, May 23**
14:45 – 15:45 **Salon 7 Vienna**

MIXING MUSIC IN DOLBY ATMOS

Chair: **Cristian Stefanescu**

Panelist: *Alex Koller*

Probably the first musical album mixed and presented to the audience in a Dolby Atmos-equipped cinema hall. A listening session without any accompanying visuals, images or lights. Music has become a soundtrack to mundane daily activities and it lost its place as an art form and a form of reflection, of enjoyment or of entertainment. Listeners do not care too much about the quality of the sound and we definitely lost the idea of listening. Friends meeting around a turntable, excitedly getting out a record and playing it back is a thing of the past. I've challenged my audience to forget about their smart phones for 50 minutes, forget about drinks and conversations and enjoy the music in an immersive, emotional way.

Special Event
BERLIN: CENTER OF ELECTRONIC MUSIC (2)— THE SOUND AND THE MIX
Tuesday, May 23, 15:00 – 16:30 **Berlin-A**

Moderator: André **Maletz**, Mixing Ambulance, consultant, Cologne/Berlin, Germany

Panelists: *Martin Eyerer*, Artist, Mixer, Producer, Riverside Studios, Berlin, Germany
David Miles Huber, Artist, Mixer, Producer, Seattle/Berlin

What are the practical approaches in sound design and mixing/producing Electronic Music? What are the challenges in differentiation and getting high-end audio results? How do they work and produce and what are their production environments? Is there a Berlin way? The panelists will present their approaches and audio/production examples from their actual work, explain their ways of mixing and producing.

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