AES 136th Convention Program

April 26 – 29, 2014

Estrel Hotel & Convention Center, Berlin, Germany

At AES conventions, authors have had the option of submitting complete 4- to 10-page manuscripts for peer-review by subject-matter experts. The following paper has been recognized as winner of the AES 136th Convention Peer-Reviewed Paper Award.

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The Winner of the 136th AES Convention Best Peer-Reviewed Paper Award is:

Effect of Microphone Number and Positioning on the Average of Frequency Responses in Cinema Calibration—*Giulio Cengarle, Toni Mateos*, Dolby Laboratories, Barcelona, Spain
Convention Paper 9083

To be presented on Tuesday, April 29, in Session 11—Spatial Audio

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

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

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

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

(a) The paper was accepted for presentation at the AES 136th Convention.
(b) The first author was a student when the work was conducted and the manuscript prepared.
(c) The student author's affiliation listed in the manuscript is an accredited educational institution.
(d) The student will deliver the lecture or poster presentation at the Convention.

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The Winner of the 136th AES Convention Student Paper Award is:

Small-Signal Loudspeaker Impedance Emulator—*Niels Elkjær Iversen, Arnold Knott*, Technical University of Denmark, Kgs. Lyngby, Denmark
Convention Paper 9053

To be presented on Monday, April 28, in Session 7—Transducers—Part 1: Loudspeakers

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Student Events and Career Development
STUDENT DELEGATE ASSEMBLY MEETING PLACE
Saturday – Tuesday 9:00 – 18:30
Foyer

Come visit the SDA Booth to find out about AES student events at chapters around the world. This is also where you will see postings about finalists in the recording and design competitions as well as information regarding the Student Party..

All students and educators are invited to participate in this meeting. Election results and Recording Competition Awards will be given at the Student Delegate Assembly Meeting–Part 2 on the final day of the convention.

Saturday, April 26 09:00 Room Cannes

Technical Committee Meeting on Audio for Telecommunications

Tutorial 1 Saturday, April 26
09:30 – 11:00 Estrel Hall A

THE DRUM KIT: WHAT A RECORDIST OUGHT TO KNOW

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

Audio engineers, in order to serve the artists they record, must have deep knowledge of the instruments they track and mix. The greatest of tests is the pop/rock drum kit. Drummers obsess about every detail of their instrument. Engineers do the same with every detail of their studio. This tutorial merges those obsessions, so that a recording engineer can be more fully informed about the key drivers of sound quality for the drum kit. Know the instrument first, and let that drive your decisions for recording and mixing the instrument.

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

Session P1 Saturday, April 26
10:00 – 12:30 Room Paris

PERCEPTION—PART 1

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

10:00

P1-1 Consistency of High Frequency Preference Among Expert Listeners—*Richard King,*[1,2] *Brett Leonard,*[1,2] *Stuart Bremner,*[1,2] *Grzegorz Sikora*[3]
[1]McGill University, Montreal, Quebec, Canada
[2]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada
[3]Bang & Olufsen Deutschland GmbH, Pullach, Germany

Consistency is one of the most fundamental skills of the recording professional. This is particularly important in tasks that involve the shaping of timbre. A study was designed that allows expert listeners to control a simple shelving equalizer that alters the high-frequency content of high-quality stereo program material over repeated trials. Fifteen trained subjects performed the test. Results indicate that there was an expectedly large range of preference for high frequency content. This was, however, also accompanied by a somewhat large variance. Unlike previous studies, consistency of high frequency preference was shown to be less related to the subjects' experience than other balancing tasks.
Convention Paper 9018

10:30

P1-2 Subjective Evaluation of High Resolution Recordings in PCM and DSD Audio Formats—*Atsushi Marui,*[1] *Toru Kamekawa,*[1] *Kazuhiko Endo,*[2] *Erisa Sato*[2]
[1]Tokyo University of the Arts, Adachi-ku, Tokyo, Japan
[2]TEAC Corporation, Tokyo, Japan

High-resolution audio production and consumption are increasing attraction supported by releases of the relatively affordable audio recorders from multiple manufacturers and broader bandwidth of the Internet. However, differences in audio quality between high-resolution audio formats are still not well known, especially between the different audio formats available for the audio recorders. In order to evaluate the differences between subjective impression of the sounds recorded using high resolution audio formats, three audio formats—PCM (192 kHz/24 bits), DSD (2.8 MHz), and DSD (5.6 MHz)—recorded with multiple studio-quality audio recorders were evaluated in a double-blind A/B comparison listening test. Six sound programs evaluated by forty-six participants on eight attributes revealed statistically significant differences between PCM and DSD but not between the two sampling frequencies (2.8 MHz and 5.6 MHz) of DSD.
Convention Paper 9019

11:00

P1-3 The Acceptability of Speech with Interfering Radio Program Material—*Khan Baykaner,*[1] *Christopher Hummersone,*[1] *Russell Mason,*[1] *Søren Bech*[2,3]
[1]University of Surrey, Guildford, Surrey, UK
[2]Bang & Olufsen a/s, Struer, Denmark
[3]Aalborg University, Aalborg, Denmark

A listening test was conducted to investigate the acceptability of audio-on-audio interference for radio programs featuring speech as the target. Twenty-one subjects, including naïve and expert listeners, were presented with 200 randomly assigned pairs of stimuli and asked to report, for each trial, whether the listening scenario was acceptable or unacceptable. Stimuli pairs were set to randomly selected SNRs ranging from 0 to 45 dB. Results showed no significant difference between subjects according to listening experience. A logistic regression to acceptability was carried out based on SNR. The model had accuracy R2 = 0.87, RMSE = 14%, and RMSE* = 7%. By accounting for the presence of background audio in the target program, 90% of the variance could be explained.
Convention Paper 9020

11:30

P1-4 The Effect of Dynamic Range Compression on Loudness and Quality Perception in Relation to Crest Factor—
Mark Wendl, Hyunkook Lee, University of Huddersfield, Huddersfield, UK

Two listening tests were carried out to find the changes in perceived loudness and perceived quality as the crest factors were changed for three genres (rock, electronic, and jazz) as a result of limiting, a type of dynamic range compression. The stimuli ranged from crest factors of 15 dBFS to 9 dBFS with a 1 dBFS increment. Loudness and quality had significant differences between the crest factors suggesting that a change in crest factor affects both. Correlations between loudness and quality were present for rock and jazz however not for electronic suggesting that genres can affect how we perceive quality.
Convention Paper 9021

12:00

P1-5 Hyper-Compression in Music Production: Listener Preferences on Dynamic Range Reduction— *Robert W. Taylor,*[1] *William L. Martens*[2]
[1]University of Newcastle, Callaghan, NSW, Australia
[2]University of Sydney, Sydney, NSW, Australia

Achieving "loud" recordings as a result of hyper-compression is a prevailing expectation within the creative system of music production, sustaining a myth that has been developing since the mid-twentieth century as a consequence of the "louder is better" paradigm. The study reported here investigated whether the amounts of hyper-compression typical of current audio practice produce results that listeners prefer. The experimental approach taken in this study was to conduct a subjective preference test requiring listeners to make a forced choice between seven levels of compression for each of five musical programs that differed in musical genre. The presented seven versions of each musical program were carefully matched in loudness as the versions were varied in compression level, and so differences in loudness per se cannot account for the differences in preferences choices observed between musical programs. In addition, it was found that subject factors such as age group, and speculatively the amount of exposure to different genres, were of considerable influence on listener preferences.
Convention Paper 9022

Saturday, April 26 10:00 Room Cannes

Technical Committee Meeting on Fiber Optics for Audio

Workshop 1 Saturday, April 26
11:15 – 13:00 Estrel Hall A

CHALLENGES AND OPPORTUNITIES IN AUDIO FOR ULTRA-HIGH-DEFINITION TELEVISION

Chair: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia

Panelists: *Ton Kalker*, DTS, Mountain View, CA, USA
Frank Melchior, BBC Research and Development, Salford, UK
Kazuho Ono, NHK, Tokyo, Japan

Standards organizations are currently looking into the next generation of television—known as Ultra High Definition Television (UHDTV). 4K? 8K? Dolby Vision? High frame rate? All are interesting topics for picture. As sound engineers, we too want to bring our best work to the home. This panel of audio engineers will discuss the opportunities that await the next generation of television.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Project Studio Expo 1 | **Saturday, April 26**
11:00 – 12:00 | **Estrel Hall C**

LISTENING LIKE A PRODUCER

Presenter: **Stephen Webber**, Berklee College of Music, Valencia, Spain

Critical listening skills are essential for all audio professionals. Tools and strategies for setting up proper monitoring, deciding what speakers to purchase, and small room acoustics are crucial. Just as important are the development of listening skills on technical, emotional, and kinesthetic levels. Stephen Webber, the author of the online course "Music Production Analysis," will tune up your listening in this focused session.

Tutorial 2 | **Saturday, April 26**
11:30 – 13:00 | **Estrel Hall B**

AUDIO FORENSICS— WHAT'S IT ALL ABOUT

Presenters: **Eddy B. Brixen**, EBB-consult, Smørum, Denmark
Christopher Hicks, Cedar Audio Ltd., Cambridge, UK

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 another's work? Even building acoustics and psycho acoustics, 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. In this tutorial state-of-the-art sound cleaning will be demonstrated.

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

Project Studio Expo 2 | **Saturday, April 26**
12:00 – 13:00 | **Estrel Hall C**

BASIC MICROPHONE TECHNOLOGY

Presenter: **Ron Streicher**, Pacific Audio-Visual Enterprises, Pasadena, CA, USA

How do microphones work? What differentiates one operating type of transducer from another? How and why do they sound different? What are polar patterns, and how do they affect the way a microphone responds to sound? What is "proximity effect" and why do some mics exhibit more of it than others? What's the truth about capsule size—does it really matter? These are just a few of the topics covered in this brief overview of Basic Microphone Technology.

Saturday, April 26 | **12:00** | **Room Cannes**

Technical Committee Meeting on Network Audio Systems

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Saturday, April 26, 13:15 – 14:15
Estrel Hall A

Opening Remarks:
- Executive Director Bob Moses
- President Sean Olive
- Convention Chairs Sascha Spors & Umberto Zanghieri

Program:
- AES Awards Presentation by Jan Abildgaard Pedersen, Awards Chair
- Introduction of Keynote Speaker by Convention Chair
- Keynote Address by Wieslaw Woszczyk

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:

DISTINGUISHED SERVICE MEDAL AWARD
- Roger Furness
- Subir Pramanik

FELLOWSHIP AWARD
- Tom Ammerman
- Natanya Ford
- Dietrich Schüller

CITATION AWARD
- Christian Dittmar
- Gyorgy Fazekas

BOARD OF GOVERNORS AWARD
- Juha Backman
- Karlheinz Brandenburg
- Francis Rumsey
- Mark Sandler

Keynote Speaker
This year's Keynote Speaker is **Wieslaw Woszczyk.** He is an internationally recognized audio researcher and educator with leading expertise in emerging technology trends in audio. Woszczyk holds the James McGill Professor Research Chair position and a full professorship at McGill University, and is the founding director of the Graduate Program in Sound Recording (1978), and founding director of the CIRMMT Centre for Interdisciplinary Research in Music Media and Technology, an inter-university, inter-faculty, interdisciplinary research center established at McGill University in 2001. An AES member since 1976, Woszczyk is a Fellow of the Audio Engineering Society (1996) and the former Chair of its Technical Council (1996-2005), Governor (twice, in 1991-1993 and 2008-2010) and President (2006-2007). He also served on the Review Board of the AES Journal. Woszczyk received the Board of Governors Award in 1991 and a group Citation Award in 2001 for "pioneering the technology enabling collaborative multichannel performance over the broadband Internet."

"The world of audio is entering a period of renaissance. We are experiencing an unprecedented range of technologies serving artists, producers, and listeners of music and sound. Little remains the same for long in the face of our relentless drive for new discoveries. I would like to present my perspective on the value of human interaction, collaboration and continuing education as a way forward in the quickly transforming world. We should be wary of trying to predict the future but make our best effort to live fully in the present, observing, exploring, and building relationships. Together, we can refine the course and purpose of our industry, we can venerate the gems of the past yet remain imaginative, open and ready for a fascinating future."

Saturday, April 26 | **14:00** | **Room Nizza**

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

Session P2 | **Saturday, April 26**
14:30 – 18:30 | **Room Paris**

PERCEPTION—PART 2

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

14:30

P2-1 An Approach to Quantifying the Latency Tolerance

Range in Non-Collaborative Musical Performances— *Jorge Enrique Medina Victoria*, University of Applied Sciences, Darmstadt, Germany; Cork Institute of Technology, Cork, Ireland

Latency is a well-known issue in collaborative music performances over networks such as the Internet. The effects of latency in performances over networks has been researched for the last decade, however, relatively few researches deal with the question of how musicians cope with their own latency in non-collaborative performances (performing music solo). This paper introduces the new concept of Latency Tolerance Range (LTR) and describes a methodological approach in order to develop a listening test, the results of which may demonstrate the influence of the musicians' performed instruments (chordophones, aerophones, and membranophones) on latency perception.
Convention Paper 9023

15:00

P2-2 Emotional Impact of Different Forms of Spatialization in Everyday Mediatized Music Listening: Placebo or Technology Effects?—*Steffen Lepa*,[1] *Stefan Weinzierl*,[1] *Hans-Joachim Maempel*,[2] *Elena Ungeheuer*[3]
[1]Technical University of Berlin, Berlin, Germany
[2]Staatliches Institut für Musikforschung Preußischer Kulturbesitz, Berlin, Germany
[3]Julius-Maximilians-Universität, Würzburg, Germany

Do the spatial cues conveyed by different audio playback technologies alter the affective experience of music listening or is this rather a matter of quality expectations leading to "placebo effects"? To find out, we conducted a 2-factorial between-subjects design study employing "spatialization type" (stereo headphones / stereo loudspeakers / live concert simulation) and "spatial quality expectations" (yes / no) as independent experimental factors. Three-hundred-six subjects rated the perceived intensity of emotional expression when listening to four different musical pieces as well as the overall audio quality. While we observed significant effects of spatialization type on perceived affective expressivity of music and spatial audio quality, expectation-related placebo effects affected perceived spatial audio quality only. Results are discussed in terms of their significance for music and media research.
Convention Paper 9024

15:30

P2-3 Data-Driven Modeling of the Spatial Sound Experience— *Aki Härmä, Munhum Park, Armin Kohlrausch*, Philips Research Europe, Eindhoven, The Netherlands

Since the evaluation of audio systems or processing schemes is time-consuming and resource-expensive, alternative objective evaluation methods attracted considerable research interests. However, current perceptual models are not yet capable of replacing a human listener especially when the test stimulus is complex, for example, a sound scene consisting of time-varying multiple acoustic images. This paper describes a data-driven approach to develop a model to predict the subjective evaluation of complex acoustic scenes, where the extensive set of listening test results collected in the latest MPEG-H 3-D audio initiative was used as training data. The results showed that a few selected outputs of various auditory models may be a useful set of features, where linear regression and multilayer perceptron models reasonably predicted the overall distribution of listening test scores, estimating both mean and variance.
Convention Paper 9025

16:00

P2-4 Investigation into Vertical Stereophonic Localization in the Presence of Interchannel Crosstalk—*Rory Wallis, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Listening tests were carried out on 12 subjects, using stereophonic loudspeakers arranged vertically in the median plane, to determine the threshold at which the amplitude of a delayed upper loudspeaker had to be reduced in order for stimuli to be fully localized at a lower loudspeaker. The test stimuli used were seven octave bands of noise (125, 250, 500, 1000, 2000, 4000, and 8000 Hz) and one broadband source (125 ¬ 8000 Hz). The effect of frequency on the threshold was found to be significant (with the 1000 and 2000 Hz bands having the lowest thresholds) while the effect of delay time was non-significant. The threshold for the broadband stimulus was found to be significantly lower compared to each of the noise bands.
Convention Paper 9026

16:30

P2-5 The Perceptual Effects of Horizontal and Vertical Interchannel Decorrelation Using the Lauridsen Decorrelator—*Christopher Gribben, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

The perceptual effects of interchannel decorrelation, using a method proposed by Lauridsen, have been investigated subjectively, looking specifically at the frequency dependency of decorrelation. Twelve subjects graded the perceived auditory image width of a pink noise sample that had been decorrelated by a Lauridsen decorrelator algorithm, varying the frequency-band, time-delay, and decorrelation factor for each sample. The same test has been carried out in both the horizontal and vertical planes. Results generally indicate that decorrelation is more effective horizontally than vertically. For horizontal decorrelation, the higher the frequency, the more effective the decorrelation, with a longer time-delay required for lower frequencies. In contrast, the vertical width produced by vertical decorrelation is better perceived at lower frequencies than higher ones.
Convention Paper 9027

17:00

P2-6 The Effect of Auditory Memory on the Perception of Timbre—*Cleopatra Pike, Russell Mason, Tim Brookes*, University of Surrey, Guildford, Surrey, UK

Listeners are more sensitive to timbral differences when comparing stimuli side-by-side than temporally-separated. The contributions of auditory memory and spectral compensation to this effect are unclear. A listening test examined the role of auditory memory in timbral discrimination, across retention intervals (RIs) of up to 40 s. For timbrally complex music stimuli discrimination accuracy was good across all RIs, but there was increased sensitivity to onset spectrum, which decreased with increasing RI. Noise stimuli showed no onset sensitivity but discrimination performance declined with RIs of 40 s. The difference between program types may suggest different onset sensitivity and memory encoding (categorical vs non-categorical). The onset bias suggests that memory effects should be measured prior to future investigation of spectral compensation.
Convention Paper 9028

17:30

P2-7 Investigation of a Random Radio Sampling Method for

Selecting Ecologically Valid Music Program Material—
Jon Francombe,[1] *Russell Mason*,[1] *Martin Dewhirst*,[1] *Søren Bech*,[2,3]
[1]University of Surrey, Guildford, Surrey, UK
[2]Bang & Olufsen a/s, Struer, Denmark
[3]Aalborg University, Aalborg, Denmark

When performing subjective tests of an audio system, it is necessary to use appropriately selected program material to excite that system. Program material is often required to be wide-ranging and representative of commonly consumed audio, while having minimal selection bias. A random radio sampling procedure was investigated for its ability to produce such a stimulus set. Nine popular stations were sampled at six different times of day over a number of days to produce a 200-item pool. Musical and signal-based characteristics were examined; the items were found to span a wide range of genres and years, and physical similarities were found between items in the same genre. The proposed method is beneficial for collecting a wide and representative stimulus set.
Convention Paper 9029

18:00

P2-8 Criticality of Audio Stimuli for Listening Tests – Listening Durations During a Ranking Task—
Jonas Ekeroot, Jan Berg, Arne Nykänen, Luleå University of Technology, Luleå, Sweden

The process of selecting critical audio stimuli for listening tests is known from the literature to be both labor-intensive and time-consuming, and has been described as more of art than science. Explicit accounts of systematic procedures are not the most commonly encountered. In a previous study a ranking-by-elimination method was investigated, resulting in a rank order that could be used as a guide for critical stimuli selection. This paper presents a further exploratory analysis of data on the subjects' listening durations, both as a function of number of stimuli left on screen and individually per stimulus. A strong negative correlation was found between the rank order of criticality and playing duration.
Convention Paper 9030

Tutorial 3 **Saturday, April 26**
14:30 – 16:00 **Estrel Hall B**

TUNING YOUR STUDIO—ACOUSTIC ANALYSIS OF SMALL ROOMS FOR MUSIC

Presenter: **Lorenzo Rizzi**, Suono e Vita - Acoustic Engineering, Lecco, Italy

Music production nowadays is made in smaller and smaller rooms but the focus on audio technologies almost removes the importance of room acoustics. We'll deal with the fundamentals of sweet-spot response measurements for acoustic optimization of the listening system and speaker/dubbing/rehearsing rooms. Why small rooms for music do not follow Sabine's laws of acoustics. Are classical acoustical parameters (e.g., RT, and the others) of any relevance?

The tutorial will also cover measuring impulse and frequency responses and discuss the importance of spatial averaging and transients. A number of room measurement and system case studies will be presented that illustrate procedures such as balancing the channels, the influence of room modes in the frequency and in the time domain, stereo triangle optimization, microphone positioning, echogram analysis, the importance of scattering and diffraction—tips to improve the room quality and technical targets.

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

Workshop 2 **Saturday, April 26**
14:30 – 16:30 **Estrel Hall A**

WORLD-CLASS CINEMA SOUND MIXERS DISCUSS THEIR CRAFT—A MASTER CLASS

Chair: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia

Panelists: *Christian Beusch*, Tonstudio Beusch TSB, Zurich, Switzerland
Lars Ginzel
Branko Neskov, Loudness Films, Lisbon, Portugal
Martin Steyer, Hochschule für Film und Fernsehen "Konrad Wolf," Potsdam, Brandenburg, Germany

Leading cinema sound mixers discuss their craft—for both the professional doing Sound for Picture as well as students thinking about this for a career.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Project Studio Expo 3 **Saturday, April 26**
14:30 – 15:30 **Estrel Hall C**

TRACKING AND LEVELS

Presenter: **Carsten Kaiser**, META4S Creative Studio, Hattingen, Germany

In this basic recording workshop you will learn more about the interrelation between project studio gear and recording levels. Did you ever ask yourself: Which is the "best" recording level? Do I have to consider headroom in my DAW software? How important is SN-ratio in the digital age? Do I have to calibrate my plugins? Carsten Kaiser will show you how to use proper Gain Staging to get the most out of your DAW-productions.

Session EB1 **Saturday, April 26**
15:00 – 16:30 **Foyer**

ENGINEERING BRIEFS—POSTERS: PART 1

15:00

EB1-1 Remixing a Historic Film in Higher Order Ambisonics 3-D Audio—Workflow and Technical Solutions—
Tobias Falke,[1] *Johann-Markus Batke*,[2] *Thomas Görne*[1]
[1]Hamburg University of Applied Sciences, Hamburg, Germany
[2]Audio & Acoustics, Technicolor Research & Innovation, Hannover, Germany

The project investigates the suitability of Higher Order Ambisonics (HOA) 3-D audio for the redesign and remix of historic film soundtracks. Technical and aesthetic challenges and hurdles are studied with scenes of a 1953 French classic. In order to obtain a high degree of immersion and on the other hand also to fix technical limitations of the historical material, some parts of the soundtrack were redesigned. The resulting HOA 3-D mix is claimed to be format agnostic, i.e., it is designed to be used on all common 3-D and also 2-D loudspeaker layouts, given the appropriate rendering system is provided.
Engineering Brief 130

15:00

EB1-2 Moving the Room... Acoustics, Around the Beamformer Beam—*Georgios Flamis, Maria Platypodi*, Dialog Semiconductor, Patras, Achaia, Greece

As the demand for audio beamforming to commercial applications is increasing, the need for a robust evaluation method becomes apparent. In such applications, the beam is shaped by the microphone distribution and the directional response that the algorithm calculates. Thus, it will be located to the 3-D space with predefined acceptance angle. The performance of the beamforming system over the scenarios of noise reduction and echo control can be estimated once the beam location is properly defined. Presented are the methods of equalizing the sound production of background noise into the room acoustics, as well as the resulting conditions, under which the edges of the beam can be definable.
Engineering Brief 131

15:00

EB1-3 Spatial Sound for Mobile Navigation Systems—*Wataru Sanuki, Julian Villegas, Michael Cohen*, University of Aizu, Aizu Wakamatsu-shi, Fukashima-ken, Japan

We have developed a mobile navigation system featuring binaural spatial sound delivered via headphones. "Machibeacon" is intended to promote traffic and pedestrian safety: users select a destination relative to their current position, and the application renders both a visual map and an auditory earcon at the goal. The apparent location of this earcon is adjusted to reflect changes in orientation of the user by modulations of the interaural level difference. To disambiguate front and back directions, the earcon progressively changes between contrasting cues. By desaturating the visual modality, smartphone users can focus on their environment and its hazards and rewards.
Engineering Brief 132

15:00

EB1-4 Two Dimensional Gestural Control of Audio Processing—*Tom Wilson, Steven Fenton*, University of Huddersfield, Huddersfield, West Yorkshire, UK

This project investigates the design of a 2-D, single, and multi-touch gesture set for the control of audio processing commonly found within DAWS and mixing consoles. The recent popularity of touch pads has made wider application in this area possible. We describe the testing, analysis, and mapping of gestures to theorize the most efficient control over audio processing parameters. By improving the control interface, workflow efficiency could be greatly improved. A test was carried out that observed engineers as they carried out specific mixing tasks using standard Pro Tools plug-ins. In addition, a survey was constructed to determine the most popular gestures for common processing parameters. The workflow and recorded gestures were then analyzed and a set of optimized gesture based controls were produced.
Engineering Brief 133

15:00

EB1-5 Influence of Directional Differences of First Reflections in Small Spaces on Perceived Clarity—*Hidetaka Imamura,[1] Atsushi Marui,[1] Toru Kamekawa,[1] Masataka Nakahara[2]*
[1]Tokyo University of the Arts, Tokyo, Japan
[2]SONA Corp., Tokyo, Japan

The ultimate goal of the research is to propose an acoustic measure of perceived clarity for small spaces such as studio control rooms and listening rooms. While C_{80} is somewhat successful in predicting the perceived clarity of sound in performance spaces, detailed research of clarity in small spaces has not been conducted. An experiment was conducted to investigate the perceived clarity of reproduced sound in small spaces with a focus on the arrival direction and delay time of the first reflections. Seven participants were asked to evaluate the sounds with loudspeaker simulated wall reflections in author-constructed temporal quasi-anechoic chamber. Variation of the first reflections did significantly influence perceived clarity and spatial impressions such as ASW, LEV, and spatial definition.
Engineering Brief 134

15:00

EB1-6 Multidimensional Ability Evaluation of Participants of Listening Tests—*Tomasz Dziedzic, Piotr Kleczkowski*, AGH University of Science and Technology, Krakow, Poland

In this e-Brief, the problem of selecting proper participants for a listening test will be addressed. The authors will describe the idea of multidimensional ability evaluation of participants and present an initial version of application developed to perform preliminary ability evaluation tests. The software testing results and conclusions as well as a draft for future work are presented.
Engineering Brief 135

Saturday, April 26 15:00 Room Cannes

Technical Committee Meeting on Transmission and Broadcasting

Project Studio Expo 4 Saturday, April 26
15:30 – 16:30 Estrel Hall C

PROFESSIONAL MIXES FROM YOUR PROJECT STUDIO— COMMON MISTAKES, IMMEDIATE SOLUTIONS

Presenters: **Alex Case**, University of Massachusetts Lowell, Lowell, MA, USA
Carsten Kaiser, META4S Creative Studio, Hattingen, Germany
Stephen Webber, Berklee College of Music, Valencia, Spain

Mixing strategies for getting organized, staying creative, making decisions, and addressing the classic symptoms of a rookie mix. Come hear our experts share their wisdom on getting the perfect mix.

Saturday, April 26 16:00 Room Cannes

Technical Committee Meeting on High Resolution Audio

Tutorial 4 Saturday, April 26
16:30 – 18:00 Estrel Hall B

SPEECH TRANSMISSION INDEX (STI) MEASUREMENTS IN PRACTICE

Presenters: **Peter Mapp**, Peter Mapp Associates, Colchester, Essex, UK
Ben Kok, BEN KOK - acoustic consulting, Uden, The Netherlands

The Speech Transmission Index is today the most widely used international measure of potential speech intelligibility. In particular it is cited and performance requirements are incorporated in many national and international sound system and emergency sound system / voice alarm system standards and codes of practice. Few standards however state how STI performance should be measured and the equipment required to carry out such measurements. The tutorial will discuss measurement techniques, data analysis, measurement equipment (including smart phone apps), equipment calibration, and the capture and logging of measurement data. The eventp will be given by Dr. Peter Mapp, a leading authority on STI measurement and the current chairman of IEC 60268-16—the International Standard relating to STI and Ben Kok.

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

Project Studio Expo 5 **Saturday, April 26**
16:30 – 17:30 **Estrel Hall C**

MIXING: WHAT IS A PLATE REVERB AND WHY DOES IT STILL MATTER IN TODAY'S DIGITAL WORLD?

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

The power of our digital production environments is truly thrilling, rich with so much potential. But we don't just look forward. Significant audio innovations from the prior century—like Plate Reverb—still influence our craft and our sound. It's not a preset. It is a unique piece of hardware, with a signature sound so important, so useful, that we seek out vintage units, build our own and digitally archive, model, and emulate them. Find out why, and make Plate Reverb part of your production toolkit.

Workshop 3 **Saturday, April 26**
17:00 – 18:30 **Estrel Hall A**

PRACTICAL TECHNIQUES FOR RECORDING AMBIENCE IN SURROUND

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

Panelist: *Michael Williams,* Sounds of Scotland, Le Perreux sur Marne, France

In this workshop microphone recording techniques for ambience in 5.1 Surround are presented and discussed in theory and practice. Various simultaneous recordings were done in preparation of the workshop. These audio samples from six different techniques in five different venues are perfectly suitable for demonstrating the principal differences between the techniques and the perceptual consequences on immersion, localization, sound color, stability, etc. The differences are not only valid for ambience and for 5.1 Surround as they show the basic differences between level/time difference stereophony and as they confirm theories on correlation between channels and their consequence for the perceived spatial image.

During the workshop, the audio samples are compared in an A/B manner and differences are discussed. The audio samples as well as the full documentation can be downloaded for free use on www.hauptmikrofon.de. They are particularly useful in education but also for sound engineers that have to choose an ambience setup in practice.

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

Project Studio Expo 6 **Saturday, April 26**
17:30 – 18:30 **Estrel Hall C**

HOW DID THEY GET THAT SOUND?

Presenter: **Carsten Kaiser**, META4S Creative Studio, Hattingen, Germany

Ever wondered how to record, arrange, and mix the vocals for a hit record? In this event Carsten Kaiser explains the vocal production of David Guetta's *Titanium*—track by track. You will find out how to record a radio-ready vocal performance. You will learn how to lay out a professional sounding arrangement for backing vocals. And last but not least, you will get lots of info about how to edit and mix these vocals in the style of David Guetta.

Student Events and Career Development
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Saturday, April 26, 18:30 – 19:30
Estrel Hall B

Presenters: **Marija Kovacina**, Student Delegate Assembly Chair for European and International Regions
Andrea Pepper, Student Delegate Assembly Chair for North and Latin American Regions
Simon-Claudius Wystrach, Student Delegate Assembly Vice Chair for European and International Regions
Steven Van Dyne, Student Delegate Assembly Vice Chair for North and Latin American Regions
Kyle P. Snyder, AES Education Committee Vice Chair
Magdalena Plewa, AES Education Committee Vice Chair
John Krivit, AES Education Committee Chair

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 ayear's election for the European and International Regions, announce the finalists in the four new recording competition categories, and announce any upcoming 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. Students will also have the opportunity to hear from various AES officers about opportunities and scholarships available through the society.

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Saturday, April 26, 19:00 – 20:00
Estrel Hall A

Lecturer: **Dietrich Schüller**

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

The Richard C. Heyser distinguished lecturer for the 136th AES Convention is Dietrich Schüller. Schüller's rare combination of experience in physics, ethnomusicology, and cultural anthropology provided him with an ideal background for work in audio preservation. He joined the Phonogrammarchiv of the Austrian Academy of Sciences as a student assistant and, following graduation, became its Director in 1972. After concentration on methodological aspects of sound recording for research purposes, he became increasingly engaged in audiovisual preservation and re-recording. He was a member of the Executive Board of the International Association of Sound and Audiovisual Archives (IASA) from 1975 to 1987, founder of the IASA Technical Committee and its chair until 2001. He is a member of the Audio Engineering Society, was member of the Organizing Committee of the Vienna AES Conventions 1992 and 2007, ➡

and Vice-Chair of the AES Standards Subcommittee on Audio Preservation and Restoration until its closure in 2012. He became engaged in UNESCO's work as delegate of Austria for Communication and Information and as an expert for the Memory of the World Programme. He has worked, partly on behalf of UNESCO, as a consultant to a number of audiovisual archives in Europe, Asia, Africa, and America.

After retirement in 2008, he continued work for UNESCO where he presently is Vice-President of the Information for All Programme (IFAP), and chair of the IFAP Working Group on Information Preservation. An author of numerous publications on audiovisual preservation, he is also involved in national and international training courses and workshops for audiovisual archivists. The title of his lecture is, "Preserving Our Sound Recordings—25 Years since Everything Changed."

In 1989/1990 audio archivists started to understand that to pursue the classical paradigm of archives and museums, by preserving the original objects placed in their care, would be in vain. All audio carriers are vulnerable and most of them unstable, at least in comparison with traditional text documents and museum objects. Moreover, as machine readable documents their retrievability would always depend on the availability of replay equipment. By that time it had become clear that ever shorter life time cycles of digital carriers and their formats would confront archivists with the impossible task of keeping an ever-growing amount of sophisticated replay equipment in operable condition.

This lead to the change of the preservation paradigm: to concentrate on content preservation by retrieving signals from their original carriers, digitize analog signals, to automatically check digital files for data integrity, and to copy them losslessly, again using automated processes, from one digital preservation platform to the next. This new paradigm was not without dispute amongst traditional archivists at the time, but it became gradually accepted. Radio archives were the first to explore digital mass storage systems for audio archiving in 1992/93, during a first ARD pilot project in Baden-Baden. The incentive was not so much the "eternal" preservation of holdings, but rather the ease of program production from the desk of the radio producers. By the mid 1990s this approach was adopted by national and research archives. Video archives followed thereafter, and ultimately even film preservation took over this principle.

The lecture describes the situation that has lead to the change of paradigm and the role that AES and other organisations played in those days. It also surveys the established standards of audio preservation, specifically the ban on data reduction ("compression") for archival purposes, and the enormous challenges associated with the transfer of content from conventional carriers into digital repositories. Broadcast and national archives of wealthy countries have already safeguarded their holdings, or are doing so. Seen from a global perspective, however, the picture is not encouraging: less wealthy institutions, specifically in developing countries, notoriously suffer from underfunding, which may prevent them from safeguarding their holdings in time, before unavailability of replay equipment makes even well preserved carriers useless. The time window for that transfer has recently been estimated to be 10-15 years. For magnetic tape documents, however, this may already be overoptimistic, as spare part production, specifically of magnetic heads, is discontinuing.

Audio and video recordings are the documents proper of cultural and linguistic diversity of human kind. Their loss would destroy a significant part of the world's documentary heritage. Consequently, UNESCO is preparing a campaign to alert institutions, governments and the civil society of the unprecedented threat to this rich, important and diverse part of the collective memory of civilizations.

Student Event and Career Development
AES STUDENT MEET-UP!
Saturday, April 26, 22:00 – 24:00
TBA

Audio Students! Come and meet with other AES Students for a night on the town. Information will be available at SDA-1 meeting.

Session P3 — **Sunday, April 27**
09:00 – 11:30 — **Room Paris**

SIGNAL PROCESSING—PART 1

Chair: **Ville Pulkki**, Aalto University, Espoo, Finland; Technical University of Denmark, Denmark

09:00

P3-1 **Memory Requirements Reduction Technique for Delay Storage in Real Time Acoustic Cameras—** *Héctor A. Sánchez-Hevia, Inma Mohino-Herranz, Roberto Gil-Pita, Manuel Rosa-Zurera*, University of Alcalá, Alcalá de Henares, Madrid, Spain

Acoustic cameras are devices capable of displaying a visual representation of sound waves. Typically these devices rely on delay-based techniques, such as Delay and Sum Beamforming, being the calculation of the proper delay values a key component of the system. For real-time systems with a large amount of microphones it is not practical to perform such calculation being common to go for an offline strategy in which the pre-calculated values are stored in memory, allowing a faster dispatch of the data while increasing memory requirements. In this paper we present a technique for delay storage optimization based on various symmetries found within the pre-calculated values that allow a reduction up to almost 16 times over the initial memory requirements.
Convention Paper 9031

09:30

P3-2 **Introducing Waveform Distribution Moments for Audio Content Analysis—***Henrik von Coler*, SIM (Staatliches Institut für Musikforschung), Berlin, Germany; Technical University of Berlin, Berlin, Germany

This paper introduces *waveform distribution moments* as features for audio content analysis. Moments and central moments of distributions are directly calculated from the squared waveform, in order to extract information on the energy development of a signal. The feature trajectories thus obtained promise to be applicable in transient detection, onset detection, and related tasks and are more sensitive to rapid changes than root mean square based methods, as a qualitative analysis reveals. An evaluation of the proposed features is presented in a feature ranking experiment related to transient detection and in an onset detection experiment. In both applications the waveform distribution moments show promising results in comparison to other signal descriptors.
Convention Paper 9032

10:00

P3-3 **Efficient Low Frequency Echo Cancellation Using Sparse Adaptive FIR Filters—***Alexis Favrot, Christof Faller*, Illusonic GmbH, Uster, Switzerland

It is shown how finite impulse response (FIR) filtering and filter adaptation can be implemented with reduced computational complexity when applied to signals containing only low frequencies. A sparse adaptive filter (with only every M^{th} coefficient being non-zero) with reduced adaptation rate achieves a similar result as a conventional adaptive filter but with lower computational complexity. An echo control scheme based on a sparse adaptive filter is described. Low frequency echoes are cancelled followed by echo suppression over all frequencies.
Convention Paper 9033

10:30

P3-4 Computationally-Efficient Speech Enhancement Algorithm for Binaural Hearing Aids—*David Ayllón, Roberto Gil-Pita, Manuel Rosa-Zurera*, University of Alcalá, Alcalá de Henares, Madrid, Spain

The improvement of speech intelligibility in hearing aids is a complex and unsolved problem. The recent development of binaural hearing aids allows the design of speech enhancement algorithms to take advantages of the benefits of binaural hearing. In this paper a novel source separation algorithm for binaural speech enhancement based on supervised machine learning and time-frequency masking is presented. The proposed algorithm requires less than 10% of the available instructions for signal processing in a state-of-the-art hearing aid and obtains good separation performance in terms of WDO for low SNR.
Convention Paper 9035

11:00

P3-5 Two-Stage Impulsive Noise Detection Using Inter-frame Correlation and Hidden Markov Model for Audio Restoration—*Kwang Myung Jeon,*[1] *Dong Yun Lee,*[1] *Nam In Park,*[1] *Myung Kyu Choi,*[2] *Hong Kook Kim*[1]
[1]Gwangju Institute of Science and Technology (GIST), Gwangju, Korea
[2]Samsung Electronics, Gyeonggi-do, Korea

In this paper a two-stage impulsive noise detection method is proposed to improve the quality of audio signals distorted by impulsive noise. In order to reduce false alarms and missing detection errors, the proposed method first tries to detect whether a frame includes onsets on the basis of inter-frame correlation. Next, hidden Markov model-based maximum likelihood classification is carried out to decide if the onset has occurred from impulsive noise or not. It is shown from performance evaluation that the proposed method achieves higher detection accuracy than with conventional residual domain-based methods under various impulsive noise distributions.
Convention Paper 9036

Tutorial 5 **Sunday, April 27**
09:00 – 10:30 **Estrel Hall B**

HANDLING AND STORAGE OF AUDIOVISUAL CARRIERS

Presenter: **Dietrich Schüller**, Phonogrammarchiv, Vienna, Austria

This tutorial will provide a preview of the IASA technical recommendation TC-05 that will be published later this year. Special focus will be given on some less known facts, e.g., the role of the production process (and failures thereof) on the longevity of audiovisual carriers.

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

Workshop 4 **Sunday, April 27**
09:00 – 11:00 **Estrel Hall A**

CREATIVE DIMENSION OF IMMERSIVE SOUND —SOUND IN 3-D—A MASTER CLASS

Chair: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia

Panelists: *Lars Ginzel*
Branko Neskov, Loudness Films, Lisbon, Portugal
Rune Palving, Tonemestrene Studio, Copenhagen, Denmark

As more producers are interested in Immersive Sound mixes for their films, this new technique is starting to find a niche. Experienced Immersive Sound mixers discuss and demonstrate their techniques.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Spatial Audio Demos **Sunday, April 27 and Monday April 28**
09:00 – 18:00 **Room Lyon**

ARTISTIC APPROACHES TO WAVE FIELD SYNTHESIS

Chair: **Thomas Koch**

Demonstrations will take place on Sunday and Monday. Detailed program available on site.

Compositions, performances, and studies for a 96-channel wave field synthesis system by students of the postgraduate master's program "Sound Studies," Berlin University of the Arts and students of the Studio for electroacoustic Music (SeaM), HfM / BU Weimar.

Spatial Audio Demo 1 **Sunday, April 27**
09:00 – 11:00 **Room Strassburg**

THE NEW KIND OF SPATIAL AUDIO PRODUCTION FOR HEADPHONE APPLICATIONS

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

Three repeated 40 minute presentations at 09:00, 09:40, 10:20.

Today over 80% of the people in the world listen to their music over headphones. Mobile entertainment is exploding, so focusing audio production (for example music, game, and film) for headphone application is certainly a useful idea. The tutorial will show how this new content can sound, it will also introduce production tools and strategies to work in the headphone domain, as well as hopefully inspiring participants to conquer new soundscapes "where no man has gone before…"

Sunday, April 27 **09:00** **Room Cannes**

Technical Committee Meeting on Spatial Audio

Project Studio Expo 7 **Sunday, April 27**
10:00 – 11:00 **Estrel Hall C**

ADVANCED VOCAL EDITING

Presenter: **Carsten Kaiser**, META4S Creative Studio, Hattingen, Germany

Vocals are a crucial factor in Pop and Rock. A great vocal performance can really make the difference. Carsten Kaiser presents how to get the most out of a singer's performance with the help of some clever editing techniques. Learn how to comp "larger than life" performances. And find out how to support the "star factor" of a vocalist with the help of natural sounding pitch and correction tactics. All this in due consideration of preserving high signal quality as well as vocal authenticity.

Sunday, April 27 **10:00** **Room Cannes**

Technical Committee Meeting on Automotive Audio

Workshop 5 **Sunday, April 27**
10:45 – 12:15 **Estrel Hall B**

MASTERING OUR FUTURE MUSIC

Chair: **Rob Toulson**, Anglia Ruskin University, Cambridge, UK

Panelists: *Andreas Lubich*, Calyx Mastering, Berlin, Germany
Mandy Parnell, Black Saloon Studios, London, UK
Jonathan Shakhovskoy, Script, London, UK

Emerging technologies are impacting the way in which music is captured, packaged, and delivered to the listener. Communications and working practices are evolving, bringing new challenges and opportunities for producing a high quality final product. Technical initiatives including Mastering for iTunes, high resolution playback, streaming services, dynamic range control and advances in metadata require mastering engineers to continuously modernize their methods. Additionally, the methods and systems for music delivery are evolving, with artists exploring new avenues for engaging their audience. In particular the “Album App” format has been considered with regards to high resolution audio, secure digital content and the inclusion of album artwork and interactive features. Furthermore, we have seen a resurgence of vinyl as a preferred listening format for audio, which has reassured the commercial importance of high-fidelity audio and rich artistic content. Each of these contemporary initiatives has an impact on the way the audio is mastered, finalized, and distributed to the listener.

The Mastering Our Future Music workshop will involve presentations and interactive discussions with a panel of experts who are fundamentally engaged in these contemporary practices for audio mastering and delivery to the listener.

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

Project Studio Expo 8 Sunday, April 27
11:00 – 12:00 Estrel Hall C

EDM: LIVE PERFORMANCE MIXING TECHNIQUES

Presenter: **Stephen Webber**, Berklee College of Music, Valencia, Spain

DJ-ing is the live expression of today’s audio engineer and music producer. In a genre where the mixer IS the artist, a variety of new tools are available to expand creative expression. Stephen Webber, the author of the best selling books *Turntable Technique: The Art of the DJ*, and *DJ Skills: The Ultimate Guide to Mixing and Scratching*, will take us through current developments and creative concepts that can be applied to any production.

Sunday, April 27 11:00 Room Cannes

Technical Committee Meeting on Semantic Audio Analysis

Sunday, April 27 11:00 Room Nizza

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

Spatial Audio Demo 2 Sunday, April 27
11:15 – 12:45 Room Strassburg

3-D AUDIO: HOW AUDIO WILL BE EXPERIENCED IN A NEW DIMENSION

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

Three repeated 30 minute presentations at 11:15, 11:45, 12:15.

Even today where the common stereo production as well as reproduction reaches technical borders it’s a great chance to switch to multichannel productions to make the audio experience as real as possible and beyond. But the cinemas already switch to 3D as well as very soon the consumer application and headphone speaker virtualizations on standard headphones works better as ever before. So there is no need to do surround 5.1 because in 3D, where speakers also placed above the listener, make it so much more exiting. How this sound and how it can be produced with current studio equipment is the task of the session.

Sunday, April 27 12:00 Room Cannes

Technical Committee Meeting on Human Factors in Audio Systems

Project Studio Expo 9 Sunday, April 27
12:30 – 13:30 Estrel Hall C

SETTING UP YOUR OWN CHAMBER REVERB AT HOME

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

If your studio has a bathroom, your studio should have a Chamber Reverb. Acoustically generated reverb isn’t just for concert halls and cathedrals. Any sound reflective space—a bathroom, a basement, a garage—is a good candidate for generating acoustic resonance. Alex Case reviews the history, the science, the technology, and the art of setting up your own chamber. You have a reverb sound nobody else does. Are you ready to use it?

Workshop 6 Sunday, April 27
12:45 – 14:15 Estrel Hall A

DSD VS. DXD: EXTREME DSD AND PCM RESOLUTIONS COMPARED

Chair: **Dominique Brulhart**, Merging Technologies

Panelists *Morten Lindberg*, 2L (Lindberg Lyd AS), Oslo, Norway
Everett Porter, Polyhymnia

With the recent release of 11.2 MHz Quad-DSD production tools, more than a decade of DSD and DXD productions and the rapidly growing availability of DSD and DXD material available for download on the market, there is a constant debate in both the professional and the audiophile sector about the difference between DSD and PCM and ultimately which one “sounds better.” This panel would like offering the opportunity to two known specialists of these formats, Everett Porter from Polyhymnia and Morten Lindberg from 2L to present some of their recordings and discuss their experience making productions in DSD and DXD. Recent recordings in DSD and DXD will be presented and recording, editing, mixing, and mastering techniques and considerations using DSD and DXD will be discussed and compared.

Sunday, April 27 13:00 Room Cannes

Technical Committee Meeting on Coding of Audio Signals

Sunday, April 27 13:00 Room Nizza

Standards Committee Meeting SC-05-02, Audio Connectors

Project Studio Expo 10 Sunday, April 27
13:30 – 14:30 Estrel Hall C

MASTERING ENGINEERING—THE LINK TO YOUR AUDIENCE

Presenters: **Gavin Lurssen**, Lurssen Mastering, Los Angeles, CA, USA
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA
Michael Romanowski, Michael Romanowski Mastering, San Francisco, CA, USA; Owner Coast Recorders

No matter how much time and effort is put into your recordings and mixes, if your music isn't delivered to your audience properly, they will never realize the vision you intended. This is what makes mastering so vital. It's the bridge between your studio and the listener for whom you create. In this session world renowned Mastering Engineers discuss how to get the most out of the mastering process in an open discussion with attendees.

Session P4 **Sunday, April 27**
13:30 – 16:30 **Room Paris**

SIGNAL PROCESSING—PART 2

Chair: **Grzegorz Sikora**, Bang & Olufsen Deutschland GmbH, Pullach, Germany

13:30

P4-1 Creation of New Virtual Patterns for Emotion Recognition through PSOLA—*Inma Mohino-Herranz, Héctor A. Sánchez-Hevia, Roberto Gil-Pita, Manuel Rosa-Zurera*, University of Alcalá, Alcalá de Henares, Madrid, Spain

Human emotions can be recognized through speech analysis. One main problem of this discipline is the lack of databases with a sufficient number of patterns for a correct learning. This fact makes generalization in the learning process be more difficult. One possible solution is the creation of new virtual patterns, enlarging the training set. In order to carry out this enlargement, we modify the average pitch by using the technique known as Pitch Synchronous Overlap and Add combined with resampling, that allows to change the average pitch without altering neither the pitch variations nor the speech rate. Therefore, the emotion in the utterance is unaltered. Results over the original test set show that it is possible to achieve a significant reduction in the generalization effects with the proposed creation of new virtual training patterns.
Convention Paper 9037

14:00

P4-2 Extended Subtractive Synthesis of Harmonic Musical Tones—*Rémi Mignot, Vesa Välimäki*, Aalto University, Espoo, Finland

A new approach is presented for the digital sound analysis-synthesis of musical tones. Based on the Source-Filter principle, the Extended Subtractive Synthesis roughly consists of the real-time filtering of a source signal by a digital filter. Starting from the recorded notes of a given instrument, the time-varying fundamental frequency and the digital filters are jointly analyzed. First, one key point of this work is the use of new advanced tools for the filter identification, which allow a relative low-order approximation of the spectral envelopes with a perceptually based criterion. Second, we propose a particular filter chain, for the separated sine and noise parts, which significantly reduces the simulation cost in the case of polyphonic synthesis and facilitates the time-varying filtering.
Convention Paper 9038

14:30

P4-3 A Virtual Acoustic Environment for Automated Parameter Optimization of a Multichannel Downmix Algorithm—*Fabian Knappe*,[1] *Robert Mores*,[1] *Christian Hartmann*[2]
[1]Hamburg University of Applied Sciences, Hamburg, Germany
[2]Institut für Rundfunktechnik, Munich, Germany

This paper presents an environment for automated parameter-optimization of a multichannel downmix algorithm. Manual optimization of multiple parameters in audio signal algorithms is likely to deliver poor results, especially if many parameters mutually interfere with each other. Even professionals fail to control the correct adjustment of all the parameters. At the same time broadcast environments ask for automated and efficient handling. This paper approaches automated optimization of a 5.0 to 2.0 channel downmix algorithm by defining a virtual acoustic environment and using an optimization process based on the Levenberg-Marquardt algorithm. The aim of the study is to determine recommendations for the parameterization of the downmix algorithm that enable mixing engineers to employ the algorithm's potential without knowledge of all the parameters' dependencies. A listening test validates the results across various genres.
Convention Paper 9039

15:00

P4-4 An Approach to the Generation of Subharmonic Frequencies in Audio Applications—*Dieter Leckschat, Christian Epe*, University of Applied Sciences Düsseldorf, Düsseldorf, Germany

In recording studios it is common to use equipment and algorithms to enhance audio productions in the low frequency range. Today's methods use either a frequency-selective dynamic compression or focus on psychoacoustics to take advantage of the residuum effect. The basic subject of this paper is a method to generate sub-harmonics of an audio signal. The most interesting sub-harmonic is one octave below a signal's fundamental frequency. By implementing a mathematical formula it is possible to produce an oscillation at half the frequency of a given harmonic oscillation. The method works in the analog or digital domain and instantaneous, which makes it suited for real-time applications of musicians. Depending on the design the process can be optimized for stationary signals or for signals with transient components.
Convention Paper 9040

15:30

P4-5 True Peak Metering—A Tutorial Review—*Ian Dash*, Consultant, Marrickville, NSW, Australia

Along with the loudness algorithm, ITU-R Recommendation BS.1770 specifies a true-peak metering method using oversampling and interpolation. The need for such metering is discussed, along with considerations on its implementation and on its usage. Implementation issues include oversampling factor, the tradeoff between accuracy and processor load and the proportion of total processor load when combined with loudness measurement. Four sources of error are examined: timing of interpolated samples, ripple in the passband response of the interpolation filter, incomplete alias suppression in the stopband response of the filter and departure from linear phase response. Implications of filter topology and filter order are discussed. An example of implementation is given along with performance parameters.
Convention Paper 9041

16:00

P4-6 Drift and Wow Correction of Analogue Magnetic Tape Recordings in the Analogue Domain Using HF-Bias Signals—*Nadja Wallaszkovits, Tobias Hetzer*,[2] *Heinrich Pichler*[3]
[1]Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria;
[2]University of Applied Sciences FH Technikum Wein, Vienna, Austria;
[3]Audio Consultant, Vienna, Austria

Based on various existing ideas of using the high frequency (HF) bias signal of analogue magnetic tape recordings as a reference signal for irregular speed deviations, this paper discusses the approach to pre-process the bias signal in the analogue domain in a way that allows control of the playback speed of the replay machine in form of a servo loop. Therefore, the bias signal is captured via a sensor head and specifically preprocessed to match the reference frequency of the capstan motor of the tape machine. The machine is set to external vari-speed control mode and, thus, deviations of the bias signal act as external vari-speed reference, allowing automatic speed correction in real-time. The paper discusses the possibilities, problems and limits of the technical implementation of such a prototype.
Convention Paper 9042

Tutorial 6 **Sunday, April 27**
14:15 – 16:15 **Estrel Hall B**

LINEAR POWER AMPLIFIERS REVISITED: FROM PICOWATTS TO A KILOWATT—A PRACTICAL GUIDE TO DRIVING BOTH HEADPHONES AND LOUDSPEAKERS PROPERLY

Presenter: **John Dawson**, ARCAM, Waterbeach, Cambridge, UK; Jade Electronics, Landbeach, Cambridge, UK

This is a tutorial for students and working engineers covering classic and modern amplification design problems/solutions relating to interactions with a complex load and relating to in-ear balanced armature devices, over the ear headphones, and all types of speakers with 120 dB dynamic range. This would tackle class A, A/B, and related analog classes rather than class D.

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

Tutorial 7 **Sunday, April 27**
14:30 – 16:30 **Estrel Hall A**

MUSIC PRODUCTION FOR FILM — A MASTER CLASS

Presenters: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia
Christine Aufderhaar, Christine Aufderhaar, Berlin, Germany

One of the largest attendance Master Classes in New York and Rome is back. Watch and hear a world-famous film composer develop the soundtrack for a feature film.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Sunday, April 27 **14:30** **Room Nizza**

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

Session P5 **Sunday, April 27**
15:00 – 16:30 **Foyer**

POSTERS: PERCEPTION / SPATIAL AUDIO / ROOM ACOUSTICS

15:00

P5-1 **Elicitation and Objective Grading of Punch within Produced Music**—*Steven Fenton, Hyunkook Lee, Jonathan Wakefield*, University of Huddersfield, Huddersfield, UK

This paper details the results from an investigation into the objective grading of punch within a complex musical signal. The term punch is a subjective term, which is often used to characterize music or sound sources that exhibit a sense of dynamic power or weight to the listener. In a novel reverse elicitation process, experts were asked to create audio samples that they perceived as having punch using a multi-band wave shaping process. Expert listeners then graded the generated punchy audio samples in a controlled listening test. Statistical analysis identified correlations between Mean Subject Scores and the parameters that created the punchy audio samples suggesting that an algorithm could be developed to objectively evaluate punch in produced music.
Convention Paper 9043

15:00

P5-2 **The Subjective Effect of BRIR Length on Perceived Headphone Sound Externalization and Tonal Coloration**—*Ryan Crawford-Emery, Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Binaural room impulse responses (BRIRs) of various lengths were convolved with stereophonic audio signals. Listening tests were conducted to assess how the length of BRIRs affected the perceived externalization effect and tonal coloration of the audio. The results showed statistically significant correlations between BRIR length and both externalization and tonal coloration. Conclusions are drawn from this and in addition, reasoning, a critical evaluation and suggested further work are suggested. The experiment provides the basis for further development of an effective and efficient externalization algorithm.
Convention Paper 9044

15:00

P5-3 **3-D Audio Object Rendering into 5.1 Surround System**—*Kangeun Lee, Seokhwan Jo, Do-Hyung Kim*, Samsung Advanced Institute of Technology, Yongin-si, Gyeonggi-do, Korea

Following the recent trend of employing UHD video for increasing reality, audio object-based representation is one of the candidates for UHD audio format. The current paper is concerned with an effective method for the rendering of audio objects into a conventional 5.1 surround system. In order to represent the 3-D objects onto the upper hemisphere of the listener, the proposed system introduces object localization and virtualization of height speakers. The object is mapped to the 10.1 channel by using the object localization, and the 10.1 channel is rendered to 5.1 surround layout by virtualization based on the mixed structure by HRTF and VBAP. Subjective impressiveness was compared with the 19.1 loudspeaker system, which demonstrated almost same performance on localization in the horizontal and vertical plane. Therefore, the proposed system is capable of delivering sound moving effects to listeners over the conventional surround system.
Convention Paper 9045

15:00

P5-4 **Elevation Localization Response Accuracy on Vertical Planes of Differing Azimuth**—*Tommy Ashby, Russell Mason, Tim Brookes*, University of Surrey, Guildford, Surrey, UK

Head movement has been shown to significantly improve localization response accuracy in elevation. It is unclear from previous research whether this is due to static cues created once the head has reached a new stationary posi-

tion or dynamic cues created through the act of moving the head. In this experiment listeners were asked to report the location of loudspeakers placed on vertical planes at four different azimuth angles (0°, 36°, 72°, 108°) with no head movement. Static elevation response accuracy was significantly more accurate for sources away from the median plane. This finding, combined with the statement that listeners orient to face the source when localizing, suggests that dynamic cues are the cause of improved localization through head movement.
Convention Paper 9046

15:00

P5-5 A New Method for the Determination of Acoustically Good Room Dimension Ratios— *John Sarris*, Aretaieio University Hospital, Athens, Greece

A new method for the determination of acoustically good room dimension ratios is presented. The method is based on the metric of variation of mean pressure defined as the variation of the mean levels of the sound pressure distribution within a room over a frequency range. This new metric quantifies the overall sound pressure variation within the room and is representative of the evenness of the frequency response among the various listening positions. Simulation results are presented for a small and a larger room where the new index is used to draw maps from which appropriate room proportions can be chosen.
Convention Paper 9047
[Poster was not presented but is available for purchase]

15:00

P5-6 A Novel Approach for Prototype Extraction in a Multipoint Equalization Procedure—*Stefania Cecchi*,[1] *Laura Romoli*,[1] *Francesco Piazza*,[1] *Balázs Bank*,[2] *Alberto Carini*[3]
[1]Universitá Politecnica della Marche, Ancona, Italy;
[2]Budapest University of Technology and Economics, Budapest, Hungary
[3]University of Urbino "Carlo Bo," Urbino, Italy

Multipoint equalization is a useful procedure used to enlarge the zone to be equalized in sound reproduction systems by measuring the room impulse responses in multiple locations and deriving a prototype function capable to represent the real environment. This paper deals with the introduction of a novel prototype function derived from the combination of quasi-anechoic impulse responses with the impulse responses recorded in the real environment to be equalized. This is motivated by the fact that at mid and high frequencies the timbre perception and localization is dominated by the direct sound, thus, the measurable, but mostly inaudible magnitude deviations due to reflections should not be equalized. Several experiments have been conducted in order to validate the proposed approach, considering a real environment and reporting objective and subjective measurements in comparison with the state of the art.
Convention Paper 9048

15:00

P5-7 Implementation of a Binaural Localization Algorithm in Hearing Aids: Specifications and Achievable Solutions— *Gilles Courtois*,[1] *Patrick Marmaroli*,[1] *Hervé Lissek*,[1] *Yves Oesch*,[2] *William Balande*[2]
[1]Swiss Federal Institute of Technology (EPFL), Lausanne, Switzerland
[2]Phonak Communciations AG, Murten, Switzerland

This paper introduces the constraints and issues related to the implementation of a binaural localization algorithm on a pair of hearing aids. This algorithm should improve the rendering of the spatial information available in the audio signals, which are usually distorted by the signal processing algorithms in the hearing devices, thus degrading localization cues. First, several reported algorithms achieving binaural sound localization in the frontal horizontal plane are reviewed. The way in which the standard methods and processes could be used within the context of hearing aids is then discussed. Finally, a solution that is suitable for a certain type of system is proposed.
Convention Paper 9034

Project Studio Expo 11 **Sunday, April 27**
15:00 – 16:00 **Estrel Hall C**

HOW DID THEY GET THAT SOUND?

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

Alex Case dissects popular recordings and reverse engineers some of their sonic secrets – sure to inspire you to create your own variations on the good ideas we hear in the mixes we love.

Sunday, April 27 **15:00** **Room Cannes**

Technical Committee Meeting on Hearing and Hearing Loss Prevention

Project Studio Expo 12 **Sunday, April 27**
16:00 – 17:00 **Estrel Hall C**

THE BUSINESS OF BEING IN THE MUSIC BUSINESS

Presenters: **David Miles Huber**
Carsten Kaiser, META4S Creative Studio, Hattingen, Germany
Gavin Lurssen, Lurssen Mastering, Los Angeles, CA, USA
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA
Stephen Webber, Berklee College of Music, Valencia, Spain

Our panel of seasoned experts discusses the ins and outs of making a career in today's music business.

Sunday, April 27 **16:00** **Room Nizza**

Standards Committee Meeting SC-04-04, Microhone Measurement and Characterization

Session P6 **Sunday, April 27**
16:30 – 18:30 **Room Paris**

ROOM ACOUSTICS

Chair: **Ben Kok**, Ben Kok Acoustic Consulting, Uden, The Netherlands

16:30

P6-1 Infrasound in Vehicles—Theory, Measurement, and Analysis—*John Vanderkooy*, University of Waterloo, Waterloo, ON, Canada

Infrasound (IS) in cars is quite strong and may be responsible for health effects. This paper presents measurements and simplified mechanisms for the production of IS in vehicles. Four mechanisms are proposed: (1) turbulence ➥

from the moving vehicle or other traffic, infusing through the vents; (2) flexing of the body causing volume changes; (3) acceleration of the vehicle, causing an inertial reaction from the enclosed and external air; and (4) pressure variations due to altitude changes. The acoustic pressure from these mechanisms can be simplified by the fact that IS wavelengths are much larger than the size of the vehicle. Measurements are interesting and analyzed to elucidate the acoustic contribution of each mechanism.
Convention Paper 9049

17:00

P6-2 Open Plan Office Acoustics and Computer Modeling: Theory versus Practice—*Lise W. Tjellesen*, Applied Acoustic Design (AAD), Staines, UK

The acoustics of open plan offices and offices in general has long been the subject of numerous studies looking at privacy levels and speech intelligibility. As acoustic consultants, office design is often dealt with on a daily basis, and various guidance's such as national / international standards, building regulations or general codes of practice are normally used as references when carrying out the design. In more complex designs of offices, especially open plan offices, computer modeling is used more and more as an integral and important tool alongside accumulated experiences from measurements. This paper explores the problems and challenges of using computer modeling in connection with open plan office design.
Convention Paper 9050
[Paper not presented but is available for purchase]

17:30

P6-3 On Low and Mid Frequencies Sound Absorption Characteristics of Porous Materials—*Elena Prokofieva*, Edinburgh Napier University, Edinburgh, UK

Porous materials are frequently used in sound insulation applications. In building acoustics they are installed inside the separating constructions to absorb unwanted mid and high frequencies propagating through them. Although it is not required by UK Building Regulations, low frequency noise transferred through separating partitions is considered as a nuisance by occupants and could and should be addressed. A laboratory study had been conducted to investigate the effect of various types of porous materials that may be used for partition fillings on the sound absorption over low and mid frequencies. The results suggest that open pore materials could improve low frequency range of absorption while have no detrimental effect on mid frequency absorption level.
Convention Paper 9051

18:00

P6-4 Optimizing Microphone Placement and Formats for Assistive Listening System Sound Pick-Up—*Peter Mapp*, Peter Mapp Associates, Colchester, Essex, UK

Approximately 10 – 14% of the general population (USA and Northern Europe) suffer from a noticeable degree of hearing loss and would benefit from some form of hearing assistance or deaf aid. However, many assistive listening systems do not provide the benefit that they should, as they are often let down by their poor acoustic performance. The paper investigates the acoustic and speech intelligibility requirements for ALS performance and examines a number of microphone pick-up scenarios in terms of their potential intelligibility and sound quality performance. The results of testing carried out in a number of rooms and venues are presented, mainly in terms of the resultant Speech Transmission Index (STI) measurements. The paper concludes by providing a number of recommendations and "rules of thumb" for successful microphone placement and testing.
Convention Paper 9052

Workshop 7 — Sunday, April 27
16:30 – 18:30 — Estrel Hall B

CINEMA SOUND—THEATER LOUDNESS AND STANDARDS ISSUES

Chair: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia

Panelists: *Eelco Grimm*, Grimm Audio, Utrecht, Netherlands
David Murphy, Krix Loudspeakers, Hackham, South Australia

Technical Committee Sound for Digital Cinema and Television has a subgroup on Cinema Loudness issues working under the direction of Brian McCarty and Eelco Grimm.

This workshop will present the current data gathered by the committee and prepare suggestions for continued Standards work by the Society.

This session is presented in association with the AES Technical Committee on Sound for Digital Cinema and Television

Project Studio Expo 13 — Sunday, April 27
17:00 – 18:30 — Estrel Hall C

FIX IT IN THE MIC—USING MICROPHONES AS YOUR EQ

Presenter: **Julian David**, AEA Ribbon Microphones, Cologne, Germany

Have you ever struggled with too many instruments fighting for real estate when mixing? Or have you ever wished you would have spotted an issue earlier, while EQ'ing a signal to death just to make it fit in the mix? Particularly in project studios, when recording time is more expensive than mix time, it's easy to postpone important decisions until later, opening the door to a myriad of avoidable problems. Producer/engineer and ribbon microphone expert Julian David will present microphone techniques and recording strategies for getting a great sound during tracking rather than "fixing it in the mix." A special focus will be on affordable and realistic solutions to improve your sound at home and in project studios.

Sunday, April 27 — 17:00 — Room Cannes

Technical Committee Meeting on Signal Processing

Workshop 8 — Sunday, April 27
17:30 – 18:30 — Estrel Hall A

MASTERING IN THE MODERN AGE

Chair: **Gavin Lurssen**, Lurssen Mastering, Los Angeles, CA, USA

Panelists: *Andrew Mendelson*, Georgetown Masters, Nashville, TN, USA
Mandy Parnell, Black Saloon Studios, London, UK
Darcy Proper, Wisseloord Studios, Hilversum, The Netherlands
Michael Romanowski, Michael Romanowski Mastering, San Francisco, CA, USA; Owner Coast Recorders

Despite the industrial infiltration of low-grade listening, the mastering field has maintained its relevance and progress of precision. Without this finalizing step from birth of an idea to its sonic realization, the energy and excitement that is most desired in music would be lost. Mastering's essentiality is heard in the refined grooves of a vinyl record and the restrained output of an MP3 format alike. This panel, comprised of world-class mastering engineers, explores how to get the most out of mastering in today's world of changing formats, music business models, and evolving technology.

Sunday, April 27 17:30 Room Nizza

Standards Committee Meeting SC-02-08, Audio-File Transfer and Exchange

Special Event
BANQUET/MIXER
Sunday, April 27
19:30-23:00

The banquet will be held at Solar (www.solar-berlin.de), which is located in the center of Berlin featuring a marvelous view from the 17th floor. Meet and chat with your friends in a relaxing atmosphere. A three course menu together with a selection of drinks is provided. More information and tickets are available at the registration desk. Bus will depart Estrel at 19:30.

Session P7 Monday, April 28
09:00 – 13:00 Room Paris

TRANSDUCERS—PART 1: LOUDSPEAKERS

Chair: **Markus Koch**, Bang & Olufsen Deutschland GmbH, Pullach, Germany

09:00

P7-1 Small-Signal Loudspeaker Impedance Emulator—*Niels Elkjær Iversen, Arnold Knott*, Technical University of Denmark, Kgs. Lyngby, Denmark

Specifying the performance of audio amplifiers is typically done by playing sine waves into a pure ohmic load. However real loudspeaker impedances are not purely ohmic but characterized by the mechanical resonance between the mass of the diaphragm and the compliance of its suspension which vary from driver to driver. Therefore a loudspeaker emulator capable of adjusting its impedance to a given driver is in need for measurement purposes. This paper proposes a loudspeaker emulator circuit for small signals. Simulations and experimental results are compared and show that it is possible to emulate the loudspeaker impedance with an electric circuit and that its resonance frequency can be changed by tuning two resistors.
Convention Paper 9053

09:30

P7-2 Dynamic Measurement of Loudspeaker Suspension Parameters Using an Active Harmonic Control Technique
—*Antonin Novak,*[1,2] *Pierrick Lotton,*[1] *Laurent Simon*[1]
[1]Université du Maine, UMR CNRS 6613, Les Mans, France
[2]Orkidia Audio, Bidart, France

A new nondestructive technique to measure the nonlinear suspension parameters (stiffness K_{ms} and mechanical resistance R_{ms}) of a loudspeaker using an active harmonic control technique is presented. The goal of the active harmonic control is to eliminate the higher harmonics from the displacement signal so that a purely harmonic motion of the diaphragm is ensured. The nonlinear stiffness K_{ms} is then measured as a function of instantaneous and peak displacement; the mechanical resistance R_{ms} is measured as a function of velocity. A frequency dependence of these parameters is also discussed.
Convention Paper 9054

10:00

P7-3 Auralization of Signal Distortion in Audio Systems Part 2: Transducer Modeling—*Wolfgang Klippel*, Klippel GmbH, Dresden, Germany

A new method is presented for the auralization of selected distortion components generated by regular nonlinearities inherent in loudspeaker systems. Contrary to the generic approach presented in the first part the alternative approach presented here exploits the results of lumped parameter modeling in the state space. A mixing device generates a virtual output signal comprising nonlinear distortion attenuated or enhanced by a user-defined scaling factor. The auralization output can be used for systematic listening tests or perceptive modeling to determine audibility thresholds and to assess the impact on sound quality of the dominant nonlinearities in loudspeakers.
Convention Paper 9055

10:30

P7-4 Quantifying Acoustic Measurement Tolerances and Their Importance in the Loudspeaker Supply Chain—*Peter John Chapman*, Bang & Oufsen a/s, Struer, Denmark

Tolerances are attached to any type of measurement, and acoustical measurements are typically associated with relatively large tolerances. Despite this, measurement results are often quoted to a high degree of precision and test limits are regularly set without consideration of the measurement tolerances involved. Quantifying measurement tolerances in manufacturing in general is well documented; however the literature fails to describe the application of suitable analysis methods to the field of acoustical measurements. The paper presents the consequences of the presence of measurement tolerances in classifying parts and also describes the shortfalls of the Gauge R&R study. How to quantify a capable measurement system is described including a simple method for quantifying acoustical measurement tolerances. This is particularly relevant in quality assurance in loudspeaker production and relates strongly to the definition of test limits and loudspeaker specifications in the supply chain.
Convention Paper 9056

11:00

P7-5 An Investigation of Loudspeaker Simulation Efficiency and Accuracy Using a Conventional Model, a Near-to-Far-Field Transformation, and the Rayleigh Integral—*Ulrik Skov, René Christensen*, iCapture ApS, Roskilde, Denmark

Simulation on loudspeaker drivers require a conventional fully coupled vibroacoustic model to capture both the effect of the loading mass of the air on the moving parts and the geometric topology of the cone, dust cap, and surround. An accurate vibroacoustic model can be time-consuming to solve, especially in 3-D. In practical applications, this results in poor efficiency concerning the decision-making process to move on to the next simulation model. To overcome this the loudspeaker designer can use either a near-to-far-field transformation or post-process structural only results via the Rayleigh integral to reduce or totally eliminate the computationally demanding open air domain in front of the speaker. These simplifications come with the cost of a frequency dependent inaccuracy. This paper compares for three different drivers (a ➡

totally flat, a concave cone, and a convex dome) the efficiency and accuracy of a conventional fully-coupled vibroacoustic model where the measurement point is included in the computational FEA domain with respectively, a reduced air domain model having the measurement point outside the computational FEA domain obtained by a near-to-far-field transformation, and a model relying on the structural only Rayleigh integral post-processing.
Convention Paper 9057

11:30

P7-6 **Mechanical Nonlinearities of Electrodynamic Loudspeakers: An Experimental Study—***Balbine Maillou,*[1] *Pierrick Lotton,*[1] *Antonin Novak,*[1,2] *Laurent Simon*[1]
[1]Université du Maine, UMR CNRS 6613, Le Mans, France
[2]Orkidia Audio, Bidart, France

Spider and surround suspensions are at the origin of viscoelastic and nonlinear behaviors of loudspeakers because of their assembly geometry and their intrinsic materials. We propose here a new dynamic experimental method to characterize these properties. We drive the loudspeaker moving part with a shaker and measure the driving force, the acceleration, the velocity, and the displacement. Results are presented and discussed for a given loudspeaker, which surround suspensions exhibit viscoelastic behavior.
Convention Paper 9058

12:00

P7-7 **Active Loudspeaker Heat Protection—***Stéphan Tassart,*[1] *Simon Valcin,*[2] *Michel Menu*[2]
[1]STMicroelectronics, Paris, France
[2]STMicroelectronics, Grenoble, France

Loudspeakers are devices that accumulate heat during their transduction process. The rise of temperature is potentially harmful for the voice-coil and must be countered by the active heat control (AHC) process when other passive and mechanical dissipation schemes become inefficient. Known AHC aim at limiting the voice-coil temperature through a closed-loop approach and may lead to oscillations and audio artifacts when temperature measurements are available with latency. This paper establishes that an open-loop AHC relying on a dynamic range compressor configured as a brick-wall limiter whose threshold is modulated by the temperature of the magnetic components insures a bounded voice-coil temperature. The temperature of the magnetic assembly and the driving force of the loudspeaker can be both estimated in real-time, respectively by a linear quadratic observer (a Kalman filter) and by an envelope follower. The new AHC scheme is demonstrated and compared to closed-loop AHC on a simulation example.
Convention Paper 9059

12:30

P7-8 **A Novel Moving Magnet Linear Motor—***Claudio Lastrucci*, Powersoft S.p.a., Scandicci (FI), Italy

Electrical to acoustic conversion approach has not changed since the beginning of acoustics. New technologies in the electronic amplification domain and latest magnetic materials open a door in the field of alternative methods of acoustic transduction. A new electrodynamic device that considerably improves electrical to acoustical conversion efficiency, sound quality, robustness, and power handling has been developed. A fully balanced and symmetrical moving magnet motor design, along with anysotropic magnetic compound integration, delivers substantial performances in terms of acceleration, linearity, and efficiency providing additional degrees of freedom in high quality professional speaker design.
Convention Paper 9060

Tutorial 8 **Monday, April 28**
09:00 – 10:00 **Estrel Hall C1**

AURO 3D RECORDING AND MIXING OF CLASSICAL MUSIC

Presenter: **Malgorzata Albinska-Frank**, Tonstudio arton, Das Tonstudio für klassische Musik, Basel, Switzerland

In this tutorial, practical and aesthetic aspects of Auro 3D production for classical music will be discussed. These include:

- Discussing the design of the Auro 3D sound image
- Microphone setup
- Playback (construction of the speaker, influence of the room acoustics)
- Mixing (aesthetics of the design - effect or naturalness).

Tutorial 9 **Monday, April 28**
09:00 – 10:30 **Estrel Hall B**

ALL YOU NEED IS LUFS

Presenter: **Florian Camerer**, ORF-Austrian TV and chairman EBU group PLOUD

Loudness normalization is rapidly becoming the new standard for leveling audio. The transition has started in TV and is currently moving into other areas such as radio and the music world. In this session the basics will be summarized, and then a detailed view will be offered on some specific areas like the relative gate, the speech gate, recent developments in radio and more. Join the chairman of the EBU loudness group for a LUFSly ride along the river of audio harmony….

Workshop 9 **Monday, April 28**
09:00 – 10:30 **Estrel Hall A**

SEMANTIC AUDIO PRODUCTION

Chair: **Christian Uhle**, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Panelists: *Stefan Ledergerber*, Lawo Group, Zurich, Switzerland; LES Switzerland GmbH
Bryan Pardo, Northwestern University, IL, USA
Joshua D. Reiss, Queen Mary University of London, London, UK

Semantic audio production aims at developing tools to assist a creative person producing, editing, or mixing audio. Computational methods can perform more routine tasks, manage large amounts of data, and enable new functionalities. Examples of applications are editing single notes and objects in a mix, editing multitrack audio, live sound mixing, intelligent digital audio effects, DJ software for automatically syncing, and recommending tracks and score/audio alignment.

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

Workshop 10 **Monday, April 28**
09:00 – 10:00 **Estrel Hall C2**

THE IMMERSIVE SOUND FORMAT: REQUIREMENTS AND CHALLENGES FOR TOOLS AND WORKFLOW

Chair: **Bert Van Daele**, Auro Technologies NV, Mol, Belgium

Panelist: *Sven Mevissen*, Auro Technologies, London, UK

The Digital Cinema world has recently seen the advent of several new audio formats, each claiming to bring a new three-dimensional, immersive sound listening experience to the movie-going audience. As a result, several organizations, such as NATO (National Association of Theatre Owners) and DCI (Digital Cinema Initiative) have described a set of requirements, effectively requesting for a single, royalty-free, open interoperable standard for Immersive Sound, incorporating object-based audio capabilities.

The introduction of this new standardized format brings a few new challenges for content creators, such as movie studios and post-production facilities. Indeed, although the introduction of the standard simplifies the distribution of content with Immersive Sound, several challenges still remain, such as creating the standard 5.1 (and 7.1) Surround versions, as well as optimizing the object-based elements of the Immersive Sound standard for reproduction on a plethora of reproduction systems, each with a number of common as well as unique features.

This presentation discusses a unified workflow that enables the simultaneous creation of content in various audio formats for cinema, including the Immersive Sound format. Several attention points are highlighted such as the compatibility between deliverables, creative possibilities and limitations, playback in theaters and at home. Finally, a set of software tools is presented supporting this workflow.

Student Events and Career Development
STUDENT DESIGN COMPETITION/EXHIBITION
Monday, April 28, 09:00 – 11:00
Foyer

All accepted student entries to the AES Student Design Competition will have the opportunity to show off their designs at this poster/tabletop exhibition. This audio "science fair" 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's also 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 submit entries. Few restrictions are placed on the nature of the projects, but designs must be for use with audio. Examples include loudspeaker design, DSP plug-ins, analog hardware, signal analysis tools, mobile applications, and sound synthesis devices. Products should represent new, original ideas implemented in working-model prototypes.

Monday, April 28 09:00 Room Cannes

Technical Committee Meeting on Perception and Subjective Evaluation of Audio Signals

Tutorial 10 Monday, April 28
10:30 – 12:30 Estrel Hall C1

DITHER AND NOISE SHAPING IN DIGITAL AUDIO: HOWS AND WHYS

Presenter: **Jamie A. S. Angus**, University of Salford, Salford, Greater Manchester, UK

This tutorial investigates finite-word-length in digital audio: how it differs from analog audio, the side-effects of signal quantization, and how the adverse effects of quantization can be diminished via dither and/or noise shaping. The tutorial presents the theory of quantization along with a live interactive audio demonstration that illustrates the motivation behind the theory.

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

Tutorial 11 Monday, April 28
10:30 – 12:00 Estrel Hall B

SPEECH INTELLIGIBILITY

Presenter: **Askan Sigfried**, NDR, Hamburg, Germany

The higher awareness of audio in TV because of the loudness revolution has led to other areas in audio to step into the spotlight again. One of those is the intelligibility of speech, which parameters influence it and how it can be improved. After examining the potential causes for speech intelligibility problems, advice will be given on best practice how to ensure your audience doesn't have to strain to understand what is being said in a program.

Workshop 11 Monday, April 28
10:30 – 12:00 Estrel Hall C2

APPLICATIONS FOR OBJECT-BASED AUDIO

Chair: **Bert Van Daele**, Auro Technologies NV, Mol, Belgium

Panelists: *Ton Kalker, DTS*, Mountain View, CA, USA
Richard Furse, DTS, Inc., Blue Ripple Sound
Frank Melchior, BBC Research and Development, Salford, UK

Recently, the concept of object-based audio has received quite a lot of attention. Although not completely new—it has been used for gaming applications for many years—the use of audio objects in audio systems for digital cinema, home entertainment and broadcast is in its infancy. In this panel session, experts from various industries explain their view on the use of object-based audio in their field. Similarities and differences are highlighted, as well as considerations such as bandwidth, rendering, compatibility, creative options, etc. An overview is given of the various activities and the relations between them, as well as the value chain from creation to the home.

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

Spatial Audio Demo 3 Monday, April 28
10:30 – 12:00 Room Strassburg

AURO 3D—THE INVISIBLE SPACE

Presenter: **Malgorzata Albinska-Frank**, Tonstudio arton, Das Tonstudio für klassische Musik, Basel, Switzerland

Two repeated 45 minute presentations at 10:30 and 11:15.

How can we create an invisible space? This session provides the audience with an opportunity to listen to Auro 3D recordings of classical music, alongside a discussion of practical and aesthetic production aspects of Auro 3D recordings.

Workshop 12 Monday, April 28
10:45 – 12:45 Estrel Hall A

MASTERING TIPS FOR ENGINEERS AND STUDIO OWNERS

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

Panelists: *Stefan Bock*, msm-studios GmbH, Munich, Germany
Darcy Proper, Wisseloord Studios, Hilversum, The Netherlands

The presenters will "role play" the real situation with a client in a mastering session. All the issues that may come up in real life will be addressed in order to demonstrate how important it is as engineers (and/or facility managers) to make clients feel confident and at ease but still keeping control of the situation. In other words, how to develop interpersonal skills that are crucial to the music production business.

Student Events and Career Development
EDUCATION/CAREER FAIR
Monday, April 28, 11:00 – 12:30
Foyer

The combined AES 136th 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 136th 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, April 28 11:00 Room Nizza

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

Monday, April 28 12:00 Room Cannes

Technical Committee Meeting on Acoustics and Sound Reinforcement

Tutorial 12 Monday, April 28
12:15 – 13:15 Estrel Hall C2

BEETHOVEN'S 9TH IN 3-D AUDIO WITH DIGITAL MICROPHONES

Presenter: **Gregor Zielinsky**, Sennheiser Electronic GmbH & Co. KG, Germany

In Part 1 of this tutorial the difficulties and specialities of recording Beethoven's 9th Symphony in Auro 3D, with the Seoul Philharmonic Orchestra under Maestro Myung-whun Chung, will be presented. This will include an explanation of the digital microphone setup.

Part 2 of the tutorial will introduce a "one-size fits all" algorithm that can be used by consumers for upmixing from 2-D to 3-D automatically. The algorithm can be used in a professional version with the option of parameter tuning for professional upmix mastering. The algorithm was used for the "David Bowie is..." exhibition, which will be presented in Berlin 2014 as well as in other worldwide locations such as London, Chicago, Sao Paulo, and more.

The recording and the algorithm upmixing technique can be listened to in the "Recording Beethoven's 9th in 3D plus Upmix Techniques" session.

Student Events and Career Development
RECORDING COMPETITION—PART 1
Monday, April 28, 13:00 – 15:00
Estrel Hall A

Moderators: **Marija Kovacina**, SDA Chair for European and International Regions
Andrea Pepper, SDA Chair for North and Latin American Regions
Simon-Claudius Wystrach, SDA Vice Chair for European and Intenational Regions
Steven Van Dyne, SDA Vice Chair for North and Latin American Regions

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). 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 if your project isn't one of the finalists, and it's a great chance to meet other students and faculty members.

Monday, April 28 13:00 Room Cannes

Technical Committee Meeting on Sound for Digital Cinema and Television

Session P8 Monday, April 28
13:30 – 16:00 Room Paris

TRANSDUCERS—PART 2

Chair: **Hans Riekehof-Boehmer**, SCHOEPS Mikrofone, Karlsruhe, Germany

14:30

P8-1 Ambient Atmospheric Conditions and Their Influence on Acoustic Measurements—*Peter John Chapman*, Bang & Oufsen a/s, Struer, Denmark

The paper describes an area that is often not considered by those who are involved in performing acoustic measurements. Specifically, the measured sound pressure level is directly influenced by the ambient atmospheric conditions in which the measurement is performed and by the raw condition of the device under test. The influence of temperature, atmospheric pressure, and humidity are described. Different strategies for removing these influences are presented. Furthermore, consequences of ignoring these influences in the laboratory and on the production line are illustrated in terms of measurement error, falling yield, and misalignment of sensitivity in active systems. The paper focuses on indoor acoustic measurements but the subject is equally valid for outdoor measurements.
Convention Paper 9061

15:00

P8-2 Numerical Simulation of Microphone Wind Noise, Part 3: Wind Screens and Shields—*Juha Backman*, Nokia Corporation, Espoo, Finland

This paper discusses the use of the computational fluid dynamics (CFD) for computational analysis of microphone wind noise. This final part of the study discusses the effect of typical wind shields and pop screens on the flow around the microphone body. The simulations show that although all types of screens share common action mechanisms, moving the vortices generated by the flow further away from the microphone diaphragm and absorbing the vortex energy, the importance of these mechanisms varies with the type and size of the shield. In addition to the flow properties also the effect of different wind shields on the acoustical response of the microphone is discussed.
Convention Paper 9062

15:30

P8-3 **Graphene Microphone**—*Dejan Todorovic,*[1] *Iva Salom,*[2] *Djordje Jovanovic,*[3] *Aleksandar Matkovic,*[3] *Marijana Milicevic,*[3] *Mirjana Radosavljevic*[1]
[1]Dirigent Acoustics Ltd., Belgrade, Serbia
[2]Institute Mihajlo Pupin, Belgrade, Serbia
[3]University of Belgrade, Belgrade, Serbia

This paper analyses recent trends in graphene research and applications in acoustics and audio-technology and attempts to identify future directions in which the field is likely to develop. The possibilities of application of single or multi-layer graphene as membranes in transducers are the scope of the research of the graphene group. FEM and experimental analysis of single and multi-layer graphene, as well as realization of the first samples of acoustic transducers, are in progress.
Convention Paper 9063

16:00

P8-4 **How Far Do Microphones Reach? A Comparison between Dynamic and Analog / Digital Condenser Microphones**—*Jürgen Breitlow, Dominic Haul*, Georg Neumann GmbH, Berlin, Germany

The feedback of recording engineers to condenser microphones occasionally refers to them as "hearing too far." Unwanted background noises such as fans, clocks, air conditioners that are not heard when recording with a dynamic microphone are being reproduced by a condenser microphone. In the present study this phenomenon is examined in regard to its physical cause. The whole signal chain from the microphone to the recorder has to be considered. Therefore it needs to be determined how the masking effect and the perceptibility of quiet sources are influenced by the self-noise of the signal chain.
Convention Paper 9064

16:30

P8-5 **A Condenser Microphone for Close-Miking and Very High Sound Pressure Levels–Revisited**—*Martin Schneider*, Georg Neumann GmbH, Berlin, Germany

In the late 1960s, tube condenser microphones were superseded by their transistorized counterparts. Close-miking was adopted for pop and rock music. Microphones able to handle these very high SPLs, but keeping the familiar sonic characteristics, needed to be developed. One model, that has now become a classic of its own, will be described in detail, taking advantage of well-sorted archives and the original documents.
Convention Paper 9065

Tutorial 13 **Monday, April 28**
13:30 – 14:30 **Estrel Hall B**

PERCEPTUALLY MOTIVATED FILTER DESIGN WITH APPLICATIONS 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 13 **Monday, April 28**
13:30 – 15:00 **Estrel Hall C2**

HOW DO WE MAKE THE SOUND FOR HEADPHONES?

Chair: **Toru Kamekawa**, Tokyo University of the Arts, Adachi-ku, Tokyo, Japan

Panelists: *Bob DeMaa*, DTS Inc., Scotts Valley, CA, USA
Sean Olive, Harman International, Northridge, CA, USA
Ville Pulkki, Aalto University, Aalto, Finland
Bob Schulein, RBS Consultants, Schaumburg, IL, USA

Although headphone listening became popular with the advent of portable audio playback systems such as Walkman and iPod, most of the music production is done using traditional loudspeaker systems. With this discrepancy in mind, the workshop will discuss following points: What is the difference between headphone listening and loudspeaker listening? Can we make the suitable sound for headphone listening through loudspeaker playback? Whether we should change the recording technique to adapt headphone listening? How a dummy head or HRTF technique used in actual recording? Panelists will introduce actual examples and discuss how we make the sound for headphones playback.

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

Tutorial 14 **Monday, April 28**
14:00 – 15:30 **Estrel Hall B**

LOCATION SOUND 101

Presenter: **TBA**

[abstract unavailable]

Spatial Audio Demo 4 **Monday, April 28**
14:00 – 15:30 **Room Strassburg**

RECORDING BEETHOVEN'S 9TH IN 3-D PLUS UPMIX TECHNIQUES

Presenter: **Gregor Zielinsky**, Sennheiser Electronic GmbH & Co. KG, Germany

Three repeated 30 minute presentations at 14:00, 14:30, 15:00.

Part 1

In this session, examples from the original recording of Beethoven's 9th Symphony, with the Seoul Philharmonic Orchestra under Maestro Myung-whun Chung, will be played (as introduced in the accompanying tutorial).

Part 2: Upmix of Stereo to 3-D

In order to prepare the audience for 3-D sound, sources need to be upmixed from 2-D to 3-D. A "one-size fits all" algorithm is presented, which is able to upmix 2D to 3D automatically. Furthermore, the ➡

algorithm can be used in a professional version with the option of parameter tuning for professional upmix mastering.

The algorithm was used for the “David Bowie is...” exhibition, which will be presented in Berlin 2014 as well as in other worldwide places like London, Chicago, Sao Paulo, and more.

Many sound examples from Bowie to Pink Floyd will be played.

Session P9 **Monday, April 28**
15:00 – 16:30 **Foyer**

POSTERS: AUDIO SIGNAL PROCESSING / TRANSDUCERS / RECORDING / NETWORK AUDIO

15:00

P9-1 **Adaptive Digital Oscillator for Virtual Acoustic Feedback** —*Leonardo Gabrielli*,[1] *Marco Giobbi*,[1] *Stefano Squartini*,[1] *Vesa Välimäki*[2]
[1]Universitá Politecnica delle Marche, Ancona, Italy
[2]Aalto University, Espoo, Finland

In the domain of Virtual Acoustics research, the emulation of acoustic feedback, such as the so-called guitar howling, has been scarcely addressed. This paper takes pace from this peculiar effect to introduce a computational technique aimed at its emulation and extension to possible new scenarios of Virtual Acoustics. A nonlinear digital oscillator for real-time operation with good stability properties and low computational cost is employed to emulate guitar feedback (or guitar howling). The oscillator frequency is tuned according to a pitch detection system that adaptively tracks pitch changes in real-time. A real-time implementation of the algorithm in the Puredata environment has been developed to provide guitar howling emulation.
Convention Paper 9066

15:00

P9-2 **A Psychoacoustic-based Vocal Suppression for Enhanced Interactive Service Using Spatial Audio Object Coding—** *Tung Chin Lee*,[1] *Young-cheol Park*,[2] *Dae Hee Youn*[1]
[1]Yonsei University, Seoul, Korea
[2]Yonsei University, Wonju, Kwangwon-do, Korea

In this paper we present a new vocal suppression algorithm that can enhance the quality of music signal coded using Spatial Audio Object Coding (SAOC) in Karaoke mode. The remained vocal component in the coded music signal is estimated and suppressed by using a spectral subtraction method. Using the fact that the level of the remained vocal components is varied depending on the input object power, we propose a psychoacoustic rule where the suppression level is adapted according to the auditory masking property. Objective and subjective test were performed and the results confirm that the proposed algorithm offers an improved quality.
Convention Paper 9067

15:00

P9-3 **Application of Common-Pole Parallel Filters to Nonlinear Models Based on Orthogonal Functions—***Laura Romoli*,[1] *Stefania Cecchi*,[1] *Balázs Bank*,[2] *Michele Gasparini*,[1] *Francesco Piazza*[1]
[1]Universitá Politecnica della Marche, Ancona, Italy
[2]Budapest University of Technology and Economics, Budapest, Hungary

Different nonlinear models are exploited to model real-world devices. Among them, an effective technique is based on the combination of orthogonal nonlinear functions and frequency-domain adaptive filtering algorithms for nonlinear system identification. In this paper first the independence of the model from the orthogonal basis is demonstrated by complementing previously obtained results. Then, a highly efficient model implementation is presented by taking advantage of fixed pole parallel filters for the linear filtering part. The efficiency comes both from using common-pole modeling and from applying a warped filter design that takes into account the frequency resolution of human hearing. Experimental results prove the effectiveness of the proposed approach showing its suitability in real-time digital simulation of nonlinear audio devices.
Convention Paper 9068

15:00

P9-4 **Multiphysic Modeling and Heuristic Optimization of Compression Driver Design—***Michele Gasparini*,[1] *Stefania Cecchi*,[1] *Francesco Piazza*,[1] *Emiliano Capucci*,[2] *Romolo Toppi*[2]
[1]Universitá Politecnica della Marche, Ancona, Italy
[2]Faital S.P.A., Milan, Italy

The use of finite element analysis is quite common in modern design techniques. Modeling allows to save time and efforts, especially when complex phenomena have to be considered. A compression driver is an example of a product with a problematic design, because the great number of variables and the different physics that are involved in the sound generation process makes the direct solution of mathematical models not trivial. In this paper an algorithm that optimizes the design parameters of the driver through an evolution strategies based procedure, taking advantage of the accuracy of the results from finite elements simulation, is presented. The method has been tested by optimizing a real compression driver and the results are reported.
Convention Paper 9069

15:00

P9-5 **Properties of Gradient Loudspeakers—***Sigmund Gudvangen*, Buskerud and Vestfold University College, Kongsberg, Norway

The radiation pattern of loudspeakers play a crucial role for how the acoustic power is distributed in the room. There is mounting evidence that early reflections from the side walls are beneficial, while early reflections parallel to the sagittal plane appears to be less desirable. Gradient loudspeakers provide a means of producing unidirectional radiation patterns. Moreover, their radiation patterns are frequency-independent. In view of these very desirable properties the characteristics of first-order gradient loudspeakers are analyzed. General expressions for sound pressure and particle velocity are derived and the distortion of the radiation patterns in the high-frequency region is reviewed.
Convention Paper 9070

15:00

P9-6 **A Guide to the Design and Evaluation of New User Interfaces for the Audio Industry—***Christopher Dewey*, *Jonathan Wakefield*, University of Huddersfield, Huddersfield, UK

This paper starts from the viewpoint that the audio industry should take advantage of the possibilities offered by new visual and interactive interfaces in order to provide the best tools for audio tasks. Audio industry products have moved toward better displays in modern digital mixers and digital audio workstations but haven’t fully embraced the possibilities of current interface technology and remain largely traditional in interface design. In order for audio

engineers to develop new visual and interactive audio products an understanding of existing Human Computer Interface (HCI) design and evaluation methodology is required. This paper presents a design and evaluation process that is tailored to audio industry product development and was used in developing a new EQ interface.
Convention Paper 9071

15:00

P9-7 **An Open-Source Dynamic Networked Audio System—** *Michelle Daniels*, University of California San Diego, La Jolla, CA, USA

This paper presents an open-source networked audio system for managed networks that consists of a single Streaming Audio Manager (SAM) and an arbitrary number of clients that can be dynamically added to or removed from the system. Clients stream multichannel uncompressed audio to SAM using the Real-Time Transport Protocol. Inside of SAM, clients can be muted and soloed, and their volume can be adjusted. Additionally, client streams can be delayed to compensate for static differences in latency between audio and video playback in a multimedia environment. Basic SAM setups mix all incoming streams to a specified set of output channels. However, in advanced setups, SAM can send discrete outputs for each client to an external audio rendering system, which communicates with SAM using Open Sound Control (OSC). Third party developers can create their own renderers for advanced audio processing and can also implement user interfaces to remotely control and monitor SAM and its clients using additional OSC messages.
Convention Paper 9072

Session EB2 **Monday, April 28**
15:00 – 17:00 **Estrel Hall C1**

ENGINEERING BRIEFS—PAPERS: PART 1

Chair: **Christopher Kling**, Klangkantine, Darmstadt, Germany

15:00

EB2-1 **Auto Adaptation of the Mobile Device Characteristics for Various Acoustic Conditions —***Jozef Kotus, Andrzej Ciarkowski, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

The proposed methodology of auto adaptation of the mobile device characteristics for various acoustic conditions was presented in the paper. The main purpose of this study was to determine the parameters of the acoustic path of the mobile device for both transmitting (speaker) and receiver (microphone). Results of the measurement characteristics of mobile devices were presented. Information about characteristics of the particular partials of the sound path were used to design and develop a technique of linearization of the device frequency response characteristics. Preliminary results obtained with the proposed methodology are presented. The performed research evolved into the design of an adaptive self-linearization method that compensates for the changing of acoustic conditions through continuous monitoring and regulating the audio settings.
Engineering Brief 136

15:15

EB2-2 **The Sonic Characteristics of the Jazz Style Electric Bass Guitar—***Bryan Martin*, McGill University, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, QC, Canada

The electric Jazz-style bass has long been established as a staple in jazz and popular music. This investigation seeks to measure and map the resonant, sonic, acoustic, and electromagnetic characteristics of the instrument and its constituent parts. The characteristics and constituents to be examined are the electromagnetic pickups, resonances in the body and neck, passive electronics, and electrical impedances. An analysis will examine their confluence and contributions to the resultant sound of the instrument. While much has been published concerning the electric guitar and its pickups, there has been very little published concerning solid-body bass guitars.
Engineering Brief 137

15:30

EB2-3 **New Studio Strategies in Music Production— The Disappearing Gap between Engineer and Producer—** *Christopher Kling*, Klangkantine, Darmstadt, Germany

In an industry of shrinking music production budgets, it is common that the audio engineer and music producer happen to be the same person. New technologies as well as new system designs in hardware and software further enforce this transition—not only in project/home studios. In addition to the risks of this development, there are also benefits: New production and studio strategies enable engineers to be more flexible to the changing demands of clients and artists while working more cost-effectively. This involves changes in common studio architecture and setup and a different recording workflow through the use of new technologies in DAWs, as well as less distinction between recording and mixing. A few specific examples and perspectives will be given.
Engineering Brief 138

15:45

EB2-4 **Compensation of Crossover Region Overshoot in Multiband Compression—***David Traore, Joshua Atkins, Andrei Krishkevich, Adam Strauss,* Beats Electronics, LLC, Santa Monica, CA, USA

Overshoot in the crossover region of multiband dynamic range compression (DRC) systems is an issue that is encountered in several audio applications such as hearing aids, audio mastering tools, and portable loudspeaker systems. This overshoot translates into a loss of overall loudness due to reduced post scale headroom, digital clipping, or allowable output above a chosen threshold. This paper introduces a gain compensation filter in the limiter gain computation path in each band, thus preserving the audio quality and loudness of the system. Furthermore, a couple of methods for calculating the optimal compensation filter is presented along with analysis of the two and three band DRC systems with and without the proposed solution.
Engineering Brief 139

16:00

EB2-5 **Multiphysical Simulation Methods for Loudspeakers—A (Never-)Ending Story?—***Alfred Svobodnik*, Konzept-X GmbH, Karlsruhe, Germany

Multiphysical simulations of loudspeakers have been investigated by scientific and industrial researchers for more than 40 years. At a first glance an electrodynamic loudspeaker seems to be a fairly simple assembly—a simple (sub-)system compared to typical applications of modern CAE methods. So where is the challenge? In detail! Besides strongly coupled different physical domains (electromagnetics, mechanics, acoustics, thermal transport, fluid dynamics ...), we also have to deal with path dependent ➥

dynamic effects and nonlinearities (including instabilities) in each domain. Additionally, materials with totally different behavior (and thus totally different material models to be used) and different joining techniques for each component are used as well. This paper will summarize challenges for realistic simulations and will discuss efficient solutions for daily usage in the industrial work flow of product development.
Engineering Brief 140

16:15

EB2-6 A Method for Comparison of Nonlinearities of Consumer Earphones Using Equalized Stimuli —*Felix Fleischmann, Jorgos Estrella, Jan Plogsties*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Comparison of nonlinearities between different earphone models is not directly possible due to their different sensitivities and the high dynamics in the transducer's frequency response. Also, nonlinearities are highly dependent on the level of the excitation signal. An approach to overcome these differences and allow for fair comparison is proposed. The method is based on filtering commonly used stimuli like pink sweeps with a linear correction filter. This filter is obtained by a measurement at low input levels where the transducer shows linear behavior. The nonlinear response is then measured at different levels and THD is computed. In this way the non-linearity or different transducers can be compared directly. Some examples demonstrating the performance of consumer grade earphones are presented and discussed. The results show that nonlinearities mainly appear for low frequency excitation.
Engineering Brief 141

16:30

EB2-7 Design and Development of Auralization Room at Edinburgh Napier University—*Elena Prokofieva, C. Luciani, I. McGregor*, Edinburgh Napier University, Edinburgh, Scotland, UK

The auralization room was designed as a joint project between School of Engineering and Built Environment and the School of Computing of Edinburgh Napier University. The interior acoustic design specialists conducted the acoustic design of the auralization room, while the expert in computing provided the electronic systems. The room is planned to be used for various purposes, for example on the simulation of sound insulation of the partitions for proposed developments where the trial process is unrealistic under the real circumstances.
Engineering Brief 142

16:45

EB2-8 Comparative Results between Loudspeaker Measurements Using a Tetrahedral Enclosure and Other Methods—*Geoff Hill*, Hill Acoustics Limited, Leigh on Sea, Essex, UK

A major problem for the loudspeaker and transducer industries throughout the world, is an inability to rely upon measurements routinely exchanged between suppliers and customers. This paper updates "Consistently Stable Loudspeaker Measurements Using a Tetrahedral Enclosure"—EB4-7 published in 2013—with comparative measurements using results by other people, equipment, and methods: Small IEC Baffle in Anechoic Chamber and Large IEC Baffle Outside vs a TTC 350. These Test Chambers give us the capability to approach "Design Quality" measurements easily throughout the entire supply chain, reducing errors and improving quality while driving down the cost of measurement.
Engineering Brief 143

Monday, April 28 15:00 Room Cannes

Technical Committee Meeting on Electro Magnetic Compatibility

Monday, April 28 15:00 Room Nizza

Standards Committee Meeting SC-02-12, Audio Applications in Networks

Student Events and Career Development
RECORDING COMPETITION—PART 2
Monday, April 28, 15:30 – 17:30
Estrel Hall 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). 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 if your project isn't one of the finalists, and it's a great chance to meet other students and faculty members.

Tutorial 15 Monday, April 28
16:00 – 18:00 Estrel Hall B

FUNDAMENTAL KNOWLEDGE ABOUT MICROPHONES

Presenter: **Joerg Wuttke**, Joerg Wuttke Consultancy, Pfinztal, Germany

Even a professional with many years of experience might enjoy reviewing the basics of acoustics and the operating principles of microphones. This tutorial also includes a discussion of technical specifications and numerous practical issues.

- Introduction: Vintage technology and the future; physics and emotion; choosing a microphone for a specific application
- Basic acoustics: Sound waves; frequency and wavelength; pressure and velocity; reflection and diffraction; comb filter effects; direct and diffuse sound
- Basic evaluations: Loudness and SPL; decibels; listening tests; frequency/amplitude and frequency/phase response; frequency domain and time domain
- How microphones work: Pressure and pressure-gradient transducers; directional patterns; some special types (boundary layer microphones and shotguns)
- Microphone specifications: Frequency responses (plural!); polar diagrams; free-field vs. diffuse-field response; low- and high-frequency limits; equivalent noise, maximum SPL and dynamic range
- Practical issues: Source and load impedance; powering; wind and breath noise.

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

Tutorial 16 Monday, April 28
16:00 – 17:30 Estrel Hall C2

SPECIAL CHALLENGE METADATA: PRESERVING THE COLLECTION OSKAR SALA OR … HOW TO SAFEGUARD HITCHCOCK'S *BIRDS*

Presenter: **Nadja Wallaszkovits**, Phonogrammarchiv, Austrian Academy of Science, Vienna, Austria

Oskar Sala (1910 – 2002) was a German musician, scientist, and a pioneer of electronic music. He played and further developed the trautonium, a predecessor of the synthesizer. By enhancing and modifying this instrument Sala was able to create totally new sounds and effects. He composed the scores for more than 300 films and created the soundtrack effect for Alfred Hitchcock's film The Birds, receiving many awards for his works. After his death he left, among others, a collection of about 1200 analog magnetic audio tapes which are stored in the archives of Deutsches Museum in Munich.

Oskar Sala fully exploited all the possibilities of the analog tape technology, using impressive experimental approaches. His tapes have become artworks themselves, as they comprise a unique richness of very special and specific metadata: most of the tapes are cut up to 200 times per reel, and he used them as a (more or less readable) notebook. Such and many more surprises made the adequate safeguarding and digitization of the collection a unique undertaking. The collection has been successfully digitized, financed by KUR – Programme for the Conservation of Moveable Cultural Assets (Germany) and under the consultancy of the Phonogrammarchiv Vienna.

The workshop outlines the various challenges of this project and discusses the parameters and practical problems of the audio transfer, as well as the strategy of safeguarding the richness of metadata by use of high definition video recording. Finally a database is shown that merges the complete digitized works of Oskar Sala and provides comprehensive access to the material for the first time.

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

Spatial Audio Demo 5 — **Monday, April 28**
16:00 – 17:30 — **Room Strassburg**

ROOM SIGNALS—PROPERTIES AND INFLUENCE ON THE AESTHETIC OF AURO 3D RECORDINGS

Presenter: **Lasse Nipkow**, Silent Work GmbH, Zurich, Switzerland

Three repeated 30 minute presentations at 16:00, 16:30, 17:00.

The relevance of room signals in a recording, especially of classical music for 3-D reproducing systems like Auro 3D, is essential: The listener has a spatial impression of being in the recorded room, and generally, perceiving a natural sounding room.

There are also some other important parameters for 3-D reproducing systems:

- Energy spreading of the room signal by the involved loudspeakers;
- Proportion of direct sound in the miked room sound;
- Content correlation: similarity of the content of the different loudspeaker signals;
- Sense of depth of room sound reproduced .

During the presentation, the above-mentioned aspects will be shown by demonstrative examples. Recordings using different room microphone systems will be demonstrated.

Monday, April 28 — **16:00** — **Room Cannes**

Technical Committee Meeting on Audio Forensics

Monday, April 28 — **16:30** — **Room Nizza**

Standards Committee Meeting SC-07-01, Metadata for Audio

Session P10 — **Monday, April 28**
17:00 – 18:30 — **Room Paris**

HUMAN FACTORS

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

17:00

P10-1 A New Algorithm for Vocal Tract Shape Extraction from Singer's Waveforms—*Rebecca Vos,*[1] *Jamie A. S. Angus,*[2] *Brad H. Story*[3]
[1]University of York, Heslington, York, UK
[2]University of Salford, Salford, Greater Manchester, UK
[3]University of Arizona, Tucson, AZ, USA

This paper presents a new algorithm for extracting vocal tract shape from speech or singing. Based on acoustic sensitivity functions it removes the ambiguity that conventional methods suffer from. We describe acoustic sensitivity functions and how we extract the necessary formant frequencies from the acoustic waveform. Results are presented for a variety of singers both male and female singing a variety of vowels and notes. The results are good and the system not only has applications in voice training but could also be used for control of games or music synthesis.
Convention Paper 9073

17:30

P10-2 Participatory Amplitude Level Adjustment of Gesture Controlled Upper Body Garment Sound in Immersive Virtual Reality—*Erik Sikström, Morten Havemøller Laursen, Kasper Søndergaard Pedersen, Amalia de Götzen, Stefania Serafin*, Aalborg University Copenhagen, Copenhagen, Denmark

Gesture-controlled sounds from virtual clothes in immersive virtual environments, is a relatively unexplored topic. In this paper an experiment aiming to find a range between a highest acceptable amplitude level and a lowest acceptable level for the sounds of an upper-body garment was conducted. Participants were asked to set the two amplitude levels of the sound from the virtual clothes that were generated by the subjects' gesture input, in relation to other sound sources with already predefined levels (footsteps and ambient sounds). This task was performed while walking around in a virtual park area. The results yielded two dynamic ranges that were differently placed depending on if the sound was initially presented at a loudest possible level, or the lowest possible level.
Convention Paper 9074

18:00

P10-3 Audio Information Mining—Pragmatic Review, Outlook, and a Universal Open Architecture—*Philip J. Duncan, Duraid Y. Mohammed, Francis F. Li*, University of Salford, Salford, Greater Manchester, UK

There is an immense amount of audio data available currently whose content is unspecified and the problem of classification and generation of metadata poses a significant and challenging research problem. We present a review of past and current work in this field; specifically in the three principal areas of segmentation, feature extraction, and classification and give an overview and critical appraisal of techniques currently in use. One of the major impediments to progress in the field has been specialism and the inability of classifiers to generalize, and we propose a non exclusive generalized open architecture framework for classification of audio data that will accommodate third party plugins and work with multi-dimensional feature/descriptor space as input.
Convention Paper 9075

Monday, April 28 — **17:00** — **Room Cannes**

Technical Committee Meeting on Microphones and Applications

Student Events and Career Development
EVE AUDIO ACOUSTIC LAB
Monday, April 28, 18:00 – 20:00
Off site

Our visit to EVE Audio Acoustic Lab consists of a tour to an anechoic chamber where we will experience a measurement and listening session of a speaker in the chamber and then in a room without acoustic treatment for comparison testing; touring an echo chamber, a room to evaluate the acoustic power response of a speaker; and have a sound demonstration of various EVE Audio monitors in a stereo and 5.1 surround setup.

Special Event
ORGAN CONCERT
Monday, April 28, 19:00 – 21:00
St. Matthias Church
Goltzstraße 29 (am Winterfeldtplatz)
10781 Berlin

By kind permission of the church and its organist, Ulrich Gembaczka, Francis Rumsey will be giving an organ recital during the 136th Convention in Berlin. Held on the splendid instrument at St. Matthias Catholic Church, widely regarded as one of the biggest and most impressive-sounding in the city, the recital will include Alain's famous "Litanies" and the rousing "March on a Theme of Handel" by Guilmant. These are complemented by works from Buxtehude, Couperin, Bach, Mendelssohn, and Reveyron.

The current organ at St. Matthias was originally built in 1958 by the firm of Romanus Seifert & Son, consisting of four manuals and pedals. It is distributed spatially and the considerable reverberation of the space gives it an impressive but challenging acoustic context. From 1972–74 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 2008–09 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.dieorgelseite.de/specials/stmatthias/stmatthias_e.htm, and http://st-matthias-berlin.de/musik/die-st-matthias-orgel.html

Session P11 — **Tuesday, April 29**
09:00 – 13:00 — **Room Paris**

SPATIAL AUDIO

Chair: **Clemens Par**, Swiss Audec, Morges, Switzerland

09:00

P11-1 Control of Frame Loudspeaker Array for 3-D Television— *Akio Ando, Masafumi Fujii*, University of Toyama, Toyama, Japan

To obtain a stable sound localization on the TV display, the use of a loudspeaker array set on the frame of the display may be a solution. However, the frequency response and the shape of the wave front reproduced by the array sometimes deteriorate. This is because the wave field synthesis with Rayleigh integrals may not be effective in the absence of a secondary source on the display. In this study we use the Rayleigh I integral to calculate input signals of the loudspeakers and introduce weighting coefficients for the signals to alleviate the deterioration. Error functions are defined to scale such deterioration and minimized by the simulated annealing. As the result, the frequency response and the wave surface were improved regardless of the virtual source position.
Convention Paper 9076

09:30

P11-2 Ambidio: Sound Stage Width Extension for Internal Laptop Loudspeakers—*Tsai-yi Wu,*[1] *Agnieszka Roginska,*[1] *Ralph Glasgal*[2]
[1]New York University, New York, NY, USA
[2]Ambiophonics Institute, Rockleigh, NJ, USA

This paper introduces a sound stage width extension method for internal loudspeakers. Ambidio is a real-time application that enhances a stereo sound file playing on a laptop in order to provide a more immersive experience over built-in laptop loudspeakers. The method, based on Ambiophonics principles, is relatively robust to a listener's head position and requires no measured/synthesized HRTFs. The key novelty of the approach is the pre/post-processing algorithm that dynamically tracks the image spread and modifies it to fit the hardware setting in real-time. Two detailed evaluations are provided to assess the robustness of the proposed method. Experimental results show that the average perceived stage width of Ambidio is 176° using internal speakers, while keeping a relatively flat frequency response and a higher user preference rating.
Convention Paper 9077

10:00

P11-3 On Spatial-Aliasing-Free Sound Field Reproduction Using Infinite Line Source Arrays—*Frank Schultz, Till Rettberg, Sascha Spors*, University of Rostock, Rostock, Germany

Concert sound reinforcement systems aim at the reproduction of homogeneous sound fields over extended audiences for the whole audio bandwidth. For the last two decades this has been mostly approached by using so called line source arrays due to their superior abilities of producing homogeneous sound fields. Design and setup criteria for line source arrays were derived as Wavefront Sculpture Technology in literature. This paper introduces a viewpoint on the problem at hand by utilizing a signal processing model for sound field synthesis. It will be shown that the optimal radiation of a line source array can be considered as a special case of spatial-aliasing-free synthesis of a wave front that propagates perpendicular to the array. For high frequencies the so called waveguide operates as a spatial low-pass filter and therefore attenuates energy that otherwise would lead to spatial aliasing artifacts.
Convention Paper 9078

10:30

P11-4 2-D to 3-D Upmixing Based on Perceptual Band Allocation (PBA)—*Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Listening tests were carried out to evaluate the performance of a 2-D to 3-D ambience upmixing technique based on "Perceptual Band Allocation (PBA)," which is a novel vertical image extension method. Five-channel recordings were made with a 3-channel frontal microphone array and a 4-channel ambience array in a concert hall. The 4-channel ambience signals were low- and high-pass filtered at three different crossover frequencies: 0.5 k, 1 k, and 4 kHz. For 2-D to 3-D upmixing, the low-passed signals were routed to the corresponding lower-layer loudspeakers while the high-passed ones to the upper-layer loudspeakers configured in a 9-channel Auro3D-inspired setup. Results suggested that the proposed method produced a similar or greater magnitude of perceived 3-D listener envelopment compared to an original 9-channel ambience recording as well as the original 5-channel recording, depending on the crossover frequency.
Convention Paper 9079

11:00

P11-5 Customization of Head-Related Impulse Response via Two-Dimension Common Factor Decomposition and Sampled Measurements—*Zhixin Wang, Cheung Fat Chan*, City University of Hong Kong, Kowloon, Hong Kong

A method based on subject-dependent impulse response extraction is proposed for the customization of head-related impulse response. In the training step, a two-dimension common factor decomposition algorithm is applied to train a set of direction-dependent impulse responses that are common for all subjects. A subject-dependent impulse response is extracted simultaneously for each subject to capture the subject-dependent information. In the customization step, the subject-dependent impulse response of a target subject is extracted from several head-related impulse response measurements of the subject. The extracted subject-dependent impulse response is then convolved with the trained direction-dependent impulses to construct all head-related impulse responses for the target subject. It is shown that with head-related impulse responses measured at a few directions for a target subject, head-related impulse responses at all trained directions can be customized with fairly low distortion.
Convention Paper 9080
[Paper not presented but is available for purchase]

Convention Paper 9081 was withdrawn.

11:30

P11-6 A Flexible System Architecture for Collaborative Sound Engineering in Object-Based Audio Environments—*Gabriel Gatzsche, Christoph Sladeczek*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Object-based sound reproduction, on the one hand, allows sound engineers to interact with sound objects, not only during production but also in the reproduction venue. On the other hand object-based systems are quite complex. Multicore audio processors are used to render complex sound scenes consisting of hundreds of audio objects to be reproduced using a large number of loudspeaker channels. This results in the need for applications optimally adapted to the user. Working tasks need to be parallelized. This paper outlines a software architecture that helps to incorporate the multitude of audio processing components of an object-based spatial audio environment into a unified system. The architecture allows multiple sound engineers to access, monitor, control, and/or change these system components parameters collaboratively using wireless mobile devices.
Convention Paper 9082

12:00

P11-7 Effect of Microphone Number and Positioning on the Average of Frequency Responses in Cinema Calibration—*Giulio Cengarle, Toni Mateos*, Dolby Laboratories, Barcelona, Spain

When measuring the response of a loudspeaker by averaging multiple points in a room, the results typically vary according to the number of microphones employed and their positions. We present an interpretation of the average procedure that shows that averaging converges to a compromise response over the relevant listening area, at a rate inverse to the square root of the number of microphones employed. We then provide real-world examples by performing measurements in a dubbing stage and a cinema theater, and analyzing the variations of average frequency responses over a large set of different microphone number and positioning. Results confirm the predicted scaling of the deviations and quantify their magnitude in typical rooms. The data provided helped to establish the point of diminishing returns in number of microphones.
Convention Paper 9083

Workshop 14 — Tuesday, April 29
09:00 – 11:30 — Estrel Hall B

MYTH BUSTING MICROPHONE SPECIFICATIONS

Chair: **Eddy B. Brixen**, EBB-consult, Smørum, Denmark; DPA Microphones A/S, Allerød, Denmark

Panelists: *Jürgen Breitlow*, Georg Neumann, Berlin, Germany
David Josephson, Josephson Engineering, Inc., Santa Cruz, CA, USA
Martin Schneider, Georg Neuman Berlin, Berlin, Germany
Helmut Wittek, SCHOEPS GmbH, Karlsruhe, Germany
Joerg Wuttke, Joerg Wuttke Consultancy, Pfinztal, Germany

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 its looks.) Nevertheless, there is essential and very useful information to be found in the microphone specifications. This workshop will present the most important microphone specs and provide the attendees with up-to-date information on how these data are obtained and understood. Each member of the panel—all related to industry top brands—will present one item from the spec sheet. The workshop takes a critical look on how specs are presented to the user, what to look for and what to expect. If you have questions regarding microphones, this is definitely the place to ask them! The workshop is organized by the AES Technical Committee on Microphones and Applications.

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

Workshop 15 — Tuesday, April 29
09:00 – 10:30 — Estrel Hall C1

AUDIO LOUDNESS FOR AUDIOVISUAL ARCHIVES

Presenter: **Stefano S. Cavaglieri**, Fonoteca Nazionale Svizzera, Lugano, Switzerland

Explaining audio loudness in a few words is not obvious. This workshop is aimed to address some basic questions, such as “What is the difference between loudness and level?” “Does loudness affect our listening habits?” “Does it affect archiving?” “And what about dissemination?”—approaching the topic from a technical perspective, with improved listening quality as a target. After watching an introductory short Loudness War video, we will define a proper setup for the digitization/transfer process in the analog and in the digital domain, we will have a closer look at existing international standards, the issue of metering, and the importance of the working environment.

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

Tuesday, April 29 — 09:00 — Room Cannes

Technical Committee Meeting on Loudspeakers and Headphones

Tuesday, April 29 — 09:00 — Room Nizza

Standards Committee Meeting AESSC Plenary

Student Events and Career Development
EDUCATION FORUM
Tuesday, April 29, 09:30 – 11:00
Estrel Hall A

Moderators: **Marija Kovacina**, SDA Chair for European and International Regions
Andrea Pepper, SDA Chair for North and Latin American Regions
Simon-Claudius Wystrach, SDA Vice Chair for European and Intenational Regions
Steven Van Dyne, SDA Vice Chair for North and Latin American Regions

Student Roundtable

Come share your unique experience as a student of audio. Bring your thoughts and perspectives to an open discussion to be moderated by the AES Student Delegate Assembly Officers who want to encourage this dialog. How are you learning about audio? What is unique about your program and its facilities? How do co-curricular activities like the ones sponsored by AES and other organizations contribute to your experience? Explore strategies for making connections with the professional world and discuss the curriculums and philosophies of your programs. Students, faculty, alumni, industry professionals, and anyone interested in commenting on the state of audio education are welcome to participate.

Session EB3 **Tuesday, April 29**
10:00 – 11:30 **Foyer**

ENGINEERING BRIEFS—POSTERS: PART 2

10:00

EB3-1 An Approach to Bass Enhancement in Portable Computers Employing Smart Virtual Bass Synthesis Algorithms—*Piotr Hoffmann, Tomasz Sanner, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

The aim of this paper is to present a novel approach to the Virtual Bass Synthesis (VBS) algorithms applied to portable computers. The developed algorithms are related to intelligent, rule-based setting of synthesis parameters according to music genre of an audio excerpt and to the type of a portable device in use. To find optimum synthesis parameters of the VBS algorithms, subjective listening tests based on a parametric procedure were performed. The classification of music genres is automatically executed employing MPEG 7 parameters and the Principal Component Analysis method applied to reduce information redundancy. The VBS algorithm performs the synthesis based on a nonlinear device (NLD) or phase vocoder (PV) depending on the content of an audio file excerpt. A soft computing (fuzzy logic) algorithm is employed to set optimum synthesis parameters depending on a given song.
Engineering Brief 144

10:00

EB3-2 Energy Based Traffic Density Estimation Using Embedded Audio Processing Unit—*György Nagy, Rene Rodigast, Danilo Hollosi*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

In this paper we present preliminary results of an audio-based traffic density estimation application, developed within the EU-FP7 project EAR-IT [1]. The algorithm exploits that the energy of environmental noise, generated by vehicles, is related to the prevalent traffic conditions. Noise analysis and derived restrictions were made to improve the solution, which was implemented on an embedded platform. This approach follows the current trends—distributed and local processing—and directly targets the requirements for smart cities and wireless sensor networks. Using traffic monitoring wireless sensors, provided by the testbed SmartSantander [2], development setup was established to support the audio related algorithm deployment, testing, and assessment.
Engineering Brief 145

10:00

EB3-3 Objective Evaluation Method for the Perceived Quality of Car Horns—*Taejin Shin, Sang-Kwon Lee*, Inha University, Inchon, Korea

This paper presents an objective evaluation method for the perceived quality of car horn sound based on a psychoacoustic metric and a subjective test. A new psychoacoustic metric called the "spectrum decay (SD) slope" was developed to evaluate a luxury timbre in the sound quality of the horn sound. Eight synthetic sounds with a variety of SDS slope are designed. The synthetic sounds are subjectively evaluated by 41 subjects. A sound quality index for car horn sound is developed based on the correlation between the SD slope and subjective rating for synthetic sounds. The sound quality index is applied to the estimation of the sound quality of horn sounds of ten passenger cars measured inside the cars. The measured horn sounds are also evaluated subjectively by the same 41 subjects. The correlation between the estimated subjective rating and the subjective rating evaluated by the subjects is sufficient ($R= 0.93$, $p <0.01$) for the validation of the sound quality index.
Engineering Brief 146

10:00

EB3-4 Fitting the Mobile Device Characteristics to the User's Hearing Preferences—*Kuba Lopatka, Piotr Suchomski, Andrzej Ciarkowski, Piotr Odya, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

A method for fitting the mobile computer audio characteristics to the user's hearing preferences is proposed. The process consists of two stages: calibration and dynamics processing. During the calibration phase the user performs a loudness scaling test giving their response regarding the perceived loudness. The dynamics processing made on above basis sets the loudness to the most comfortable level. The processing accounts both for the user's hearing preferences (or possible deficiencies) and for the playback characteristics of the device. The solution is implemented as a standalone PC calibration application and as an APO object installed in the system's audio driver.
Engineering Brief 147

10:00

EB3-5 Recreating Robb: The Sound of the World's First Electronic Organ—*Michael Murphy, Max Cotter*, Ryerson University, Toronto, Ontario, Canada

This paper follows up from a 2013 AES presentation reporting on the recreation of the sound of the Robb Wave Organ, the first successful electronic organ, prototyped in 1927. The organ employed primitive "PCM-like" sampling techniques in its tone wheel construction. Our work has led to the compilation of recordings of the last known organ into a MIDI-operable cross-platform sample library for public use, bringing the instrument back to life and out of obscurity. The presentation will feature playable sample sets as well as audio comparisons between the Wave organ and the pipe organs it attempted to reproduce. Audiences will be encouraged to interact with the instrument while gaining a sense of its history and contribution to early sample-based synthesis.
Engineering Brief 148

10:00

EB3-6 Reproduction of the Radiation Pattern from a Practical Source by an Acoustic Array and the Equivalent Source Method—*Wan-Ho Cho*, Korea Research Institute of Standards & Science, Yuseonggu, Daejeon, Korea

Complicated radiated patterns and strengths from actual source can be approximately described by the expansion of spherical harmonics or, in other words, ideal sources, in various orders. If the signals are superposed to meet the requirement for generating a specially designed radiation pattern of ideal sources in various orders, an arbitrary radiation pattern simulating the actual source of interest can be reproduced by this designed filter. The method based on the equivalent source method is proposed to design a source array to reproduce not only frequency response but also the spatial response to simulate the sound field, and the suggested method is applied to reproduce the radiation pattern of musical instruments with spherically distributed loudspeaker array.
Engineering Brief 149

Tutorial 17 **Tuesday, April 29**
11:00 – 12:30 **Estrel Hall C1**

AUDIO NETWORKS 1.01: WHAT AUDIO PEOPLE NEED TO KNOW ABOUT NETWORKS

Presenters: **John Grant**, Nine Tiles
Mark Yonge

Networked audio in a professional environment is not the same as delivering MP3 music over the Internet. The differences are mainly concerned with assuring professional performance criteria despite a much larger quantity of audio data. When, for example, would you communicate using a network in preference to a traditional point-to-point connection, like AES3, AES10 (MADI), or analogue?

This tutorial is aimed at audio engineers who may have used networks for administration but are now looking to understand the details of networking professional audio. This will be a useful primer for the AES67 workshops starting at 13:30. It will explore usefulness of the network layer model, time delays (latency) in specific layers, and overall end-to-end latency in audio applications.

Student Events and Career Development
STUDENT DELEGATE ASSEMBLY MEETING—PART 2
Tuesday, April 29, 11:00 – 12:30
Estrel Hall 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. Plans for future student activities at local, regional, and international levels will be summarized.

Workshop 16 **Tuesday, April 28**
12:45 – 14:15 **Estrel Hall A**

LOUDNESS WARS: GIVE PEAKS A CHANCE

Presenters: **Thomas Lund**, TC Electronic A/S, Risskov, Denmark
Florian Camerer, ORF-Austrian TV
and chairman EBU group PLOUD

Music production, distribution, and consumption has been caught in a vicious spiral rendering two decades of our music heritage irreversibly damaged. Today, new tracks and remastered ones typically sound worse than what could even be expected from compact cassette. As a pro society, do we just sit by and let that happen on our watch?

Florian and Thomas are at the helm of two European initiatives to reverse this spiral: EBU R128 and EU legislation to prevent early hearing loss from listening to personal music players. That combination will soon make even the most ignorant A&R manager realize that it's futile to master music louder than -16 LUFS.

Tutorial 18 **Tuesday, April 29**
11:30 – 13:30 **Estrel Hall B**

TEACHING SAMPLE AND OBJECT BASED AUDIO DSP VIA MATLAB

Presenter: **Jamie A. S. Angus**, University of Salford, Salford, Greater Manchester, UK

This tutorial describes and demonstrates how one can teach sample based audio signal processing to undergraduates who are meeting Audio DSP for the first time. Unlike many MATLAB-based courses, this tutorial demonstrates how you can use it to teach both sample by sample based processing that is theoretically capable of handling an infinite data stream as well as an object focused approach that is both modular, extendable, and programmable in non-object orientated languages. It will also discuss how this might be done in languages other than MATLAB. The presentation will be interactive and will have a Q&A session.

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

Tutorial 19 **Tuesday, April 29**
12:45 – 14:15 **Room Paris**

THE PERCEPTION AND MEASUREMENT OF HEADPHONE SOUND QUALITY: DO LISTENERS AGREE ON WHAT MAKES A HEADPHONE SOUND GOOD?

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

Consumers are experiencing more of their audio content through headphones connected to mobile devices. Last year, headphone sales were $8.2 billion worldwide and continue to grow. Premium headphones (>$100) now account for 90% of the annual revenue growth. Market research indicates sound quality is a driving factor in headphone purchases with brand and fashion also being important factors among younger consumers. Yet, ironically the science behind what makes a headphone sound good and how to measure it is poorly understood. This, combined with the lack of perceptually meaningful headphone standards may explain why purchasing a headphone today is like playing Russian roulette with your ears. The magic bullet to achieving more consistent headphone sound quality is science.

The results of research conducted to better understand the relationship between consumers' perceived sound quality and acoustic performance show that when the influence of brand, fashion, and celebrity endorsement is removed from headphone tests, listeners generally agree on what makes headphones sound good, and their preferences can be correlated with acoustic measurements. Come and hear what we've learned.

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

Workshop 17 **Tuesday, April 29**
13:30 – 15:00 **Estrel Hall C1**

AES 67 AUDIO NETWORKING

Chair: **Greg Shay**, The Telos Alliance, Cleveland, OH, USA

Panelists: *Kevin Gross*
Andreas Hildebrand
Gints Linis

The AES67 standard was published in September 2013. This standard provides comprehensive interoperability recommendations for pro- ➥

fessional audio over IP networks in the areas of synchronization, media clock identification, network transport, encoding and streaming, session description, and connection management.

The convenience and installation-cost benefits of IT-style networking have been recognised for many years, and basic networking technology has also increased in speed and capacity. The most recent generation of media networks use a diversity of proprietary and standard protocols. Despite a common basis in Internet Protocol, the systems do not interoperate. In 2010 a project was set up in cooperation with the EBU to specify a set of existing IP networking parameters to ensure a more general interoperability of professional audio in all applications, including synchronization with film and television pictures. This workshop will explore the choices that were made in AES67 and the underlying logic.

Tuesday, April 29 14:00 Room Cannes

Technical Committee Meeting on Recording Technology and Practices

Session P12 Tuesday, April 29
14:30 – 16:30 Room Paris

APPLICATIONS IN AUDIO

Chair: **Dylan Menzies**, De Montfort University, Leicester, UK

14:30

P12-1 A Delayed Parallel Filter Structure with an FIR Part Having Improved Numerical Properties—*Balázs Bank*,[1] *Julius O. Smith, III*[2]
[1]Budapest University of Technology and Economics, Budapest, Hungary;
[2]Stanford University, Stanford, CA, USA

In real-world applications high-order IIR filters are often converted to series or parallel second-order sections to decrease the negative effects of coefficient truncation and round-off noise. While series biquads are more common, the parallel structure is gaining more interest due to the possibility of full code parallelization. In addition, it is relatively simple to design a filter directly in a parallel form, which can be efficiently utilized for logarithmic frequency resolution filtering often required in audio. If the numerator order of the original transfer function is higher than that of the denominator, a parallel FIR part arises in addition to the second-order IIR sections. Unfortunately, in this case the gain of the sections and that of the FIR filter can be significantly higher than that of the final transfer function, which requires the downscaling of the filter coefficients to avoid overload. This leads to a significant loss of useful bit-depth. This paper analyzes problem and suggests delaying the IIR part so that there is no overlap between the responses of the FIR part and the second-order sections.
Convention Paper 9084

15:00

P12-2 A Loudness-Based Adaptive Equalization Technique for Subjectively Improved Sound Reproduction—*Konstantinos Drossos, Andreas Floros, Nikolaos-Grigorios Kanellopoulos*, Ionian University, Corfu, Greece

Sound equalization is a common approach for objectively or subjectively defining the reproduction level at specific frequency bands. It is also well-known that the human auditory system demonstrates an inner process of sound-weighting. Due to this, the perceived loudness changes with the frequency and the user-defined sound reproduction gain, resulting into a deviation of the intended and the perceived equalization scheme as the sound level changes. In this work we introduce a novel equalization approach that takes into account the above perceptual loudness effect in order to achieve subjectively constant equalization. A series of listening tests shows that the proposed equalization technique is an efficient and listener-preferred alternative for both professional and home audio reproduction applications.
Convention Paper 9085

15:30

P12-3 New Sound and Visual System of the State Parliament of North-Rhine Westphalia in Duesseldorf, Germany—*Enno Finder, Wolfgang Ahnert*, ADA Acoustics & Media Consultants GmbH, Berlin, Germany

In 2012 the Assembly Hall of the State Parliament of North-Rhine Westphalia, located in Düsseldorf, Germany, was renovated extensively. In this context the floor structure has been changed to improve the situation for handicapped delegates and guests. The new sound design had to solve two tasks, namely excellent sound coverage from the lectern and president's position and sound localization for speeches from a delegate desk by using a kind of a new delta-stereophonic network. The paper will explain all design issues including new microphones on the lectern. The issue to fit the new speaker systems and the new microphone types into the architectural environment is discussed as well the start of renovation of the visual systems.
Convention Paper 9086

16:00

P12-4 Latest Improvements for Spatial Sound Reinforcement: Configuration's Automation, Remote Control Using Mobile Devices, and Object Based Room Simulation—*Javier Frutos-Bonilla, Gabriel Gatzsche, Rene Rodigast*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Fraunhofer IDMT presented in 2005 a sound reinforcement system based on the precedence effect that could recreate the natural spatial impression of static and dynamic sources on the stage regardless of the position of the listeners. Based on the experience learned over the last years, this paper presents three different developments for this system. These improvements are related to the automatic determination of the configuration parameters, the multiuser adjustment of fine parameters on the tribune using portable devices, and a new room simulation approach that takes into consideration both source and listener's dependencies.
Convention Paper 9087

Session P13 Tuesday, April 29
15:00 – 16:30 Foyer

POSTERS: APPLICATIONS IN AUDIO / EDUCATION / FORENSICS

15:00

P13-1 Diffused System of Noise Measurement, Concept, and Implementation—*Bartlomiej Kruk, Michal Luczynski, Adrian Pralat*, Wroclaw University of Technology, Wroclaw, Poland

The main purpose of this paper is to explore the possibilities of improving the process of noise measurement. It is a known fact that performing simultaneous noise measurements in multiple locations requires the involvement of many individuals in order to operate equipment and ensure that the results are valid. The goal is to improve the measurement process by utilizing modern technology in a way allowing for the data to be collected in a controlled

way and submitted to a central location. It allows for elimination of data preprocessing process and facilitates the acquisition and analysis process in measurements requiring data acquisition in multiple locations, reducing the human labor requirements and financial cost of measurements.
Convention Paper 9088

15:00

P13-2 Improving the Performance of an In-Home Acoustic Monitoring System by Integrating a Vocal Effort Classification Algorithm—*Emanuele Principi, Roberto Bonfigli, Stefano Squartini, Francesco Piazza*, Universitá Politecnica della Marche, Ancona (AN), Italy

The research interest in technologies for supporting people in their own homes is constantly increasing. In this context this paper proposes a speech-interfaced system for recognizing home automation commands and distress calls. The robustness of the system is increased by employing Power Normalized Cepstral Coefficients as features and by using an adaptive algorithm to reduce known sources of interference. In addition, the mismatch introduced by vocal effort variability is reduced employing a vocal effort classifier and multiple acoustic models. The performance has been evaluated on ITAAL, a recently proposed corpus of home automation commands and distress calls in Italian. The results confirm that the adopted solutions are effective to be employed in a distorted acoustic scenario.
Convention Paper 9089

15:00

P13-3 Eyes-Free Interaction for Personal Media Devices—*Thomas Svedström,*[1] *Aki Härmä*[2]
[1]Aalto University, Espoo, Finland
[2]Philips Research, Eindhoven, The Netherlands

The use of visual user interfaces in smartphones and other personal media devices (PMD) leads to decreased situational awareness, for example, in city traffic. It is proposed in the paper that many menu navigation functions in PMDs can be replaced by an eyes-free auditory interface and an input device based on acoustic recognition of tactile gestures. We demonstrate, using a novel experimental setup, that the use of the proposed auditory interface reduces the reaction times to external events in comparison to a visual UI. In addition, while the task completion times in menu navigation are somewhat increased in the auditory interface the subjects were able to complete the given interaction tasks correctly within a reasonable time.
Convention Paper 9090

15:00

P13-4 Supporting TV Sound in the UK – A New Role for Education?—*Patrick Quinn*, Glasgow Caledonian University, Glasgow, Scotland, UK

The demands placed on staff working in TV sound have changed and grown over the last few decades particularly for those in a senior role. Based on observations and interviews with senior staff this paper gives an overview of the challenges for those working in TV sound in the UK and suggests an enhanced role for Higher and Further Education to support the industry.
Convention Paper 9091

15:00

P13-5 Cross Level Peer Tutoring to Support Students Learning Audio Programming—*David Moore, Steven Walters*, Glasgow Caledonian University, Glasgow, Scotland, UK

Computer programming supports learning of key concepts in audio and music technology education, including digital audio processing and sound synthesis. However, programming is a subject that can pose a challenge–particularly for students whose primary focus of study is not pure computer science. This paper examines cross level peer tutoring as a method for supporting audio students learning programming as part of an audio processing module. It will examine the viability of this scheme as a method for enhancing student self-efficacy and achievement. The paper will explore the benefits and issues from the point of view of the tutees as well as the tutors through both quantitative and qualitative research.
Convention Paper 9092

15:00

P13-6 From Faraday to Fourier: Teaching Audio Technology Fundamentals Using Loudspeaker Design—*Scott Beveridge*, Glasgow Caledonian University, Glasgow, Scotland, UK

This paper presents a novel method of teaching basic audio principles. We describe a loudspeaker design activity that encompasses a large number of core learning outcomes. These include the basics of sound and hearing, digital audio, the audio signal path, and electroacoustics. Following the constructivist learning paradigm, the task encourages students to actively develop their own understanding. The task also promotes deep learning strategies in addition to providing a fun and engaging practical learning experience. From an instructor's perspective the activity presents a unified, structured, cost-effective method of presenting course content.
Convention Paper 9093

15:00

P13-7 Efficient Cross-Codec Framing Grid Analysis for Audio Tampering Detection—*Daniel Gärtner,*[1] *Christian Dittmar,*[1] *Patrick Aichroth,*[1] *Luca Cuccovillo,*[1] *Sebastian Mann,*[1] *Gerald Schuller*[2]
[1]Fraunhofer IDMT, Ilmenau, Germany
[2]Ilmenau University of Technology, IImenau, Germany

In this paper we present an audio tampering detection method based on the analysis of discontinuities in the framing grid, caused either by manipulations within the same recording or across recordings even with codec changes. The approach extends state of the art methods for MP3 framing grid detection with respect to efficiency and robustness, and multi-codec support, adding mp3PRO, AAC, and HE-AAC schemes. An evaluation has been carried out using a publicly available dataset. A high performance is reported on both detecting tampering and codecs showing the usefulness of the approach in audio forensics.
Convention Paper 9094

Session EB4 Tuesday, April 29
15:00 – 16:30 Estrel Hall B

ENGINEERING BRIEFS—PAPERS: PART 2

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

15:00

EB4-1 Creating Dynamic Psychoacoustic Maps of Hearing Threats for Outdoor Concerts Employing Supercomputing Grid—*Jozef Kotus, Maciej Szczodrak, Karolina Marciniuk, Andrzej Czyzewski, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

The auditory effects caused by the outdoor concert are discussed in this paper. The analysis is based on the computation results obtained by means of supercomputing PL-Grid infrastructure and specific computational algorithms developed by the authors. The software consists of the outdoor sound propagation module and psychoacoustical noise dosimeter. The simulation was performed by means of real music recordings and the following outdoor propagation conditions were taken into account: speaker directivity, ground effect, building reflection, distance attenuation, and sound absorption by the atmosphere. On the basis of the proposed methodology the dynamic (one minute time resolution) psychoacoustic maps of hearing threats for considered area were created expressed by TTS (Temporary Threshold SHIT) values in critical bands. Moreover, the results also include maps of sound level and noise dose values.
Engineering Brief 150

15:15

EB4-2 APE: Audio Perceptual Evaluation Toolbox for MATLAB—*Brecht De Man, Joshua D. Reiss*, Queen Mary University of London, London, UK

We present a toolbox for multi-stimulus perceptual evaluation of audio samples. Different from MUSHRA (typical for evaluating audio codecs), the audio samples under test are represented by sliders on a single axis, encouraging careful rating, relative to adjacent samples, where both the reference and anchor are optional. Intended as a more flexible, versatile test design environment, subjects can rate the same samples on different scales simultaneously, with separate comment boxes for each sample, an arbitrary rating scale, and various randomization options. Other tools include a pairwise evaluation tool and a loudness equalization stage. We discuss some notable experiences and considerations based on various studies where these tools were used. We have found this test design to be highly effective when perceptually evaluating qualities pertaining to music and audio production.
Engineering Brief 151

15:30

EB4-3 Principals of a Tunable Diaphragmatic Bass Absorber—*Philippe Jeansonne*, McGill University, Montreal, QC, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, QC, Canada

Once propagated in air, low frequency energy can be difficult to attenuate without physically prominent bass absorbers. The tunable diaphragmatic bass absorber fulfills this task in a discrete way. Mounted on an aluminum frame, a tunable membrane is separated by a thin air-gap to a layer of acoustic fibergalss. The membrane's excitation is restricted by the layer of acoustic fiberglass resulting in attenuation of a desired range of low frequencies. The proposed use for this new design is to attenuate a particular LF room mode or its harmonic.
Engineering Brief 152

15:45

EB4-4 A Motorized Telescope Mount as A Computer-Controlled Rotational Platform for Dummy Head Measurements—*Matthew Shotton, Chris Pike, Frank Melchior*, BBC Research and Development, Salford, UK

This paper covers the construction and validation of an affordable and accurate two degree-of-freedom rotational mount for making HRTF (head-related transfer function) and BRIR (binaural room impulse response) measurements using a dummy head microphone. We review the design requirements for a rotational mount in the context of measurements for binaural rendering, with reference to perceptual factors. In order to achieve a low-cost solution, we evaluate the suitability of a motorized telescope mount. Issues considered during design of the system are discussed. The use of affordable electronics to convert the mount into a general-purpose computer-controlled rotational platform is presented, as well as objective measurements to validate performance. Finally the limitations of this system are discussed and further use cases proposed.
Engineering Brief 153

16:00

EB4-5 Measurements and Visualization of Sound Intensity around the Human Head Using Acoustic Vector Sensor—*Jozef Kotus, Magdalena Plewa, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

Measurements and visualization of sound intensity around a human head are presented in this paper. The sound intensity field was obtained by means of a Cartesian robot that was applied to precise positioning of the acoustic vector sensor. Measurements were performed in the free field using a head and torso simulator and a configuration of either one, two, or four loudspeakers. The acoustic vector sensor was positioned around the head with 5 cm step. Sound intensity was measured in 277 points. During every step the three orthogonal sound intensity components were calculated. Tonal signals for frequencies: 250, 500, 1000, 2000, 4000, and 8000 Hz were applied. Obtained results were used to prepare visualizations of sound intensity distribution around the human head.
Engineering Brief 154

16:15

EB4-6 Spatial Audio Objects Recording Using Wireless Acoustic Sensor Networks—*Tomasz Zernicki, Piotr Szczechowiak, Lukasz Januszkiewicz, Marcin Chryszczanowicz*, Zylia sp. z.o.o., Poznan, Poland

This paper presents the development of a prototype system, which would be able to capture spatial audio scene using a wireless acoustic sensor network (WASN). Sound recording is performed in real-time by microphones embedded in each sensor node. Proposed approach is focused on processing of spatial audio objects instead of multichannel audio representation. It gives the flexibility of sound processing, mastering, and reproduction on different sound systems. This paper discusses key aspects and technologies used to build a prototype system, which are related to time synchronization of captured sounds, wireless protocols, sound source separation, and 3-D audio compression.
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Workshop 18 **Tuesday, April 29**
15:00 – 16:30 **Estrel Hall C1**

USING AES67 NETWORKING

Chair: **Mark Yonge**

Panelists: *Kevin Gross*, AVA Networks
Stefan Heinzmann, Consultant
Andreas Hildebrand, ALC Networx GmbH
Gints Linis, University of Latvia
Greg Shay, Telos Alliance

The AES67 standard provides comprehensive interoperability recommendations for professional audio over IP networks in the areas of synchronization, media clock identification, network transport, encoding and streaming, session description, and connection management.

This workshop will discuss the practical issues that will arise when AES67 is deployed in small- and large-scale installations, including physical media, switches and routers, and timing and latency.