AES 135th Convention Program

October 17 – 20, 2013

Jacob Javits Convention Center, New York, NY, USA

At recent AES conventions, authors have had the option of submitting complete 4- to 10-page manuscripts for peer-review by subject-matter experts. The following papers have been recognized as co-winners of the AES 135th Convention Peer-Reviewed Paper Award.

Proposal of Optical Wave Microphone and Physical Mechanism of Sound Detection
—*Yoshito Sonoda, Toshiyuki Nakamiya*,
Tokai University, Kumamoto, Japan
Convention Paper 8924

To be presented on Thursday, October 17, in Session 1 —Transducers—Part 1: Microphones

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Level-Normalization of Feature Films Using Loudness vs Speech—*Esben Skovenborg, Thomas Lund*, TC Electronic A/S, Risskov, Denmark
Convention Paper 8983

To be presented on Saturday, October 19, in Session 13 —Applications in Audio—Part 2

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

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

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

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

(a) The paper was accepted for presentation at the AES 135th 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 Co-Winners of the 135th AES Convention Student Paper Award are:

Measuring Speech Intelligibility in Noisy Environments Reproduced with Parametric Spatial Audio—*Teemu Koski,[1] Ville Sivonen,[2] Ville Pulkki[1]*
[1]Aalto University, Espoo, Finland
[2]Cochlear Nordic AB, Vantaa, Finland
Convention Paper 8952

To be presented on Thursday, Oct. 17, in Session P7 —Spatial Audio—Part 1

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A Perceptual Evaluation of Room Effect Methods for Multichannel Spatial Audio—*David Romblom,[1,2,3] Richard King,[1,2,3] Catherine Guastavino[1,2,3]*
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
[3]In collaboration with Sennheiser Technology and Innovation, San Francisco, CA, USA
Convention Paper 9006

To be presented on Sunday, Oct. 20, in Session P16 —Spatial Audio—Part 2

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Session P1 — Thursday, Oct. 17
9:00 am – 11:00 am — Room 1E07

TRANSDUCERS—PART 1: MICROPHONES

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

9:00 am

P1-1 Portable Spherical Microphone for Super Hi-Vision 22.2 Multichannel Audio—*Kazuho Ono,[1] Toshiyuki Nishiguchi,[2] Kentaro Matsui,[2] Kimio Hamasaki[2]*
[1]NHK Engineering System, Inc., Setagaya-ku, Tokyo, Japan
[2]NHK Science & Technology Research Laboratories, Setagaya, Tokyo, Japan

NHK has been developing a portable microphone for the simultaneous recording of 22.2ch multichannel audio. The microphone is 45 cm in diameter and has acoustic baffles that partition the sphere into angular segments, in each of which an omnidirectional microphone element is mounted. Owing to the effect of the baffles, each segment works as a narrow angle directivity and a constant beam width in higher frequencies above 6 kHz. The directivity becomes wider as

frequency decreases and that it becomes almost omnidirectional below 500 Hz. The authors also developed a signal processing method that improves the directivity below 800 Hz.
Convention Paper 8922

9:30 am

P1-2 Sound Field Visualization Using Optical Wave Microphone Coupled with Computerized Tomography—*Toshiyuki Nakamiya*,[1] *Fumiaki Mitsugi*,[2] *Yoichiro Iwasaki*,[1] *Tomoaki Ikegami*,[2] *Ryoichi Tsuda*,[1] *Yoshito Sonoda*[1]
[1]Tokai University, Kumamoto, Japan
[2]Kumamoto University, Kumamoto, Japan

The novel method, which we call the "Optical Wave Microphone (OWM)" technique, is based on a Fraunhofer diffraction effect between a sound wave and a laser beam. The light diffraction technique is an effective sensing method to detect the sound and is flexible for practical uses as it involves only a simple optical lens system. OWM is also very useful to detect the sound wave without disturbing the sound field. This new method can realize high accuracy measurement of slight density change of atmosphere. Moreover, OWM can be used for sound field visualization by computerized tomography (CT) because the ultra-small modulation by the sound field is integrated along the laser beam path.
Convention Paper 8923

10:00 am

P1-3 Proposal of Optical Wave Microphone and Physical Mechanism of Sound Detection—
Yoshito Sonoda, Toshiyuki Nakamiya, Tokai University, Kumamoto, Japan

An optical wave microphone with no diaphragm, which uses wave optics and a laser beam to detect sounds, can measure sounds without disturbing the sound field. The theoretical equation for this measurement can be derived from the optical diffraction integration equation coupled to the optical phase modulation theory, but the physical interpretation or meaning of this phenomenon is not clear from the mathematical calculation process alone. In this paper the physical meaning in relation to wave-optical processes is considered. Furthermore, the spatial sampling theorem is applied to the interaction between a laser beam with a small radius and a sound wave with a long wavelength, showing that the wavenumber resolution is lost in this case, and the spatial position of the maximum intensity peak of the optical diffraction pattern generated by a sound wave is independent of the sound frequency. This property can be used to detect complex tones composed of different frequencies with a single photo-detector. Finally, the method is compared with the conventional Raman-Nath diffraction phenomena relating to ultrasonic waves.
Convention Paper 8924

10:30 am

P1-4 Numerical Simulation of Microphone Wind Noise, Part 2: Internal Flow—*Juha Backman*, Nokia Corporation, Espoo, Finland

This paper discusses the use of the computational fluid dynamics (CFD) for computational analysis of microphone wind noise. The previous part of this work showed that an external flow produces a pressure difference on the external boundary, and this pressure causes flow in the microphone internal structures, mainly between the protective grid and the diaphragm. The examples presented in this work describe the effect of microphone grille structure and microphone diaphragm properties on the wind noise sensitivity related to the behavior of this kind of internal flows.
Convention Paper 8925

Session P2 Thursday, Oct. 17
9:00 am – 12:00 noon Room 1E09

SIGNAL PROCESSING—PART 1

Chair: **Jaeyong Cho**, Samsung Electronics, Suwon, Korea

9:00 am

P2-1 Linear Phase Implementation in Loudspeaker Systems: Measurements, Processing Methods, and Application Benefits—
Rémi Vaucher, NEXO, Plailly, France

The aim of this paper is to present a new generation of EQ. It provides a way to ensure phase compatibility from 20 Hz to 20 kHz over a range of different speaker cabinets. This method is based on a mix of FIR filters and IIR filters. The use of FIR filters allows a tuning of the phase independently from magnitude and allows an acoustic linear phase above 500 Hz. All targets used to compute FIR coefficient are based upon extensive measurement and subjective listening tests. A template has been set to normalize the crossover frequencies in the low range, enabling compatibility of every sub-bass with the main cabinets.
Convention Paper 8926

9:30 am

P2-2 Applications of Inverse Filtering to the Optimization of Professional Loudspeaker Systems—*Daniele Ponteggia*,[1] *Mario Di Cola*[2]
[1]Studio Ponteggia, Terni (TR), Italy
[2]Audio Labs Systems, Casoli (CH), Italy

The application of FIR filter technology to implement Inverse Filtering into Professional Loudspeakers Systems nowadays is easier and more affordable because of the latest development of DSP technology and also because of the existence of a new DSP platform dedicated to the end user. This paper presents an analysis, based on real world examples, of a possible methodology that can be used in order to synthesize an appropriate Inverse Filter both to process a single driver, from a Time Domain perspective in a multi-way system, and to process the output pass-band of a multi-way system for phase linearization. The analysis and discussion of results for some applications will be shown through real world test and measurements.
Convention Paper 8927

10:00 am

P2-3 Live Event Performer Tracking for Digital Console Automation Using Industry-Standard Wireless Microphone Systems—
Adam J. Hill,[1] *Kristian "Kit" Lane*,[1] *Adam P. Rosenthal*,[2] *Gary Gand*[2]
[1]University of Derby, Derby, Derbyshire, UK
[2]Gand Concert Sound, Elk Grove Village, IL, USA

The ever-increasing popularity of digital consoles for audio and lighting at live events provides a greatly expanded set of possibilities regarding automation. This research works toward a solution for performer tracking using wireless microphone signals that operates within the existing infrastructure at professional events. Principles of navigation technology such as received signal strength (RSS), time difference of arrival (TDOA), angle of arrival (AOA), and frequency difference of arrival (FDOA) are investigated to determine their suitability and practicality for use in such applications. Analysis of potential systems indicates that performer tracking is feasible over the width and depth of a stage using only two antennas with a suitable configuration, but limitations of current technology restrict the practicality of such a system.
Convention Paper 8928

10:30 am

P2-4 Real-Time Simulation of a Family of Fractional-Order Low-Pass Filters—*Thomas Hélie*,
IRCAM-CNRS-UPMC, Paris, France

This paper presents a family of low-pass filters, the attenuation of which can be continuously adjusted from 0 decibel per octave (filter is a unit gain) to -6 decibels per octave (standard one-pole filter). This continuum is produced through a filter of fractional-order between 0 (unit gain) and 1 (one-pole filter). Such a filter proves to be a (continuous) infinite combination of one-pole filters. Efficient approximations are proposed from which simulations in the time-domain are built.
Convention Paper 8929

11:00 am

P2-5 A Computationally Efficient Behavioral Model of the Nonlinear Devices—*Jaeyong Cho*,[1]
Hanki Kim,[1] *Seungkwan Yu*,[1] *Haekwang Park*,[1] *Youngoo Yang*[2]
[1]Samsung Electronics DMC R&D Center, Suwon, Korea
[2]Sungkyunkwan University, Suwon, Korea

This paper presents a new computationally efficient behavioral model to reproduce the output signal of the nonlinear devices for the real-time systems. The proposed model is designed using the memory gain structure and verified for its accuracy and computational complexity compared to other nonlinear models. The model parameters are extracted from a vacuum tube amplifier, Heathkit's W-5M, using the exponentially-swept sinusoidal signal. The experimental results show that the proposed model has 27% of the computational load against the generalized Hammerstein model and maintains similar modeling accuracy.
Convention Paper 8930

11:30 am

P2-6 High-Precision Score-Based Audio Indexing Using Hierarchical Dynamic Time Warping—
Xiang Zhou,[1] *Fangyu Ke*,[2] *Cheng Shu*,[2] *Gang Ren*,[2] *Mark F. Bocko*[2]
[1]Bose Corporation, Framingham, MA, USA
[2]University of Rochester, Rochester, NY, USA

We propose a novel audio signal processing algorithm of high-precision score-based audio indexing that accurately maps a music score with its corresponding audio. Specifically we improve the time precision of existing score-audio alignment algorithms to find the accurate positions of audio onsets and offsets. We achieve higher time precision by (1) improving the resolution of alignment sequences, and (2) admitting a hierarchy of spectrographic analysis results as audio alignment features. The performance of our proposed algorithm is testified by comparing the segmentation results with manually composed reference datasets. Our proposed algorithm achieves robust alignment results and enhanced segmentation accuracy and thus is suitable for audio engineering applications such as automatic music production and human-media interactions.
Convention Paper 8931

Workshop 1 **Thursday, Oct. 17**
9:00 am – 10:30 am **Room 1E11**

APPLICATIONS OF 3-D AUDIO IN AUTOMOTIVE

Chair: **Alan Trevena**, Jaguar Land Rover, Gaydon, UK

Panelists: *Andreas Silzle*, Fraunhofer
Jean-Marc Jot, DTS, Inc., Los Gatos, CA, USA
Gilbert Soulodre, Camden Labs, Ottawa, ON, Canada
Bert Van Daele, Auro Technologies NV, Mol, Belgium

While there are a number of technologies aimed at improving the spatial rendering of recorded sounds in automobiles, few offer the advantages and challenges as 3D surround. This workshop will explore theoretical applications; system configurations as well as limitations of 3D surround applications in automotive. Questions such as what is the reference experience, and how is a system evaluated will be addressed.

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

Tutorial 1 **Thursday, Oct. 17**
9:00 am – 10:30 am **Room 1E13**

FXPERTISE: COMPRESSION

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

Compressors were invented to control dynamic range. The next day, engineers started doing so much more—increasing loudness, improving intelligibility, adding distortion, extracting ambience, and, most importantly, reshaping timbre. This diversity of signal processing possibilities is realized only indirectly, by choosing the right compressor for the job and coaxing the parameters of ➥

ratio, threshold, attack, and release into place. Learn when to reach for compression, know a good starting place for compressor settings, and advance your understanding of what to listen for and which way to tweak.

Tutorial 2 Thursday, Oct. 17
9:00 am – 10:30 am Room 1E14

ACOUSTIC ENHANCEMENT SYSTEMS —THE BASICS

Presenter: **Ben Kok**, SCENA acoustic consultants, Uden, The Netherlands

In the last decades multiple systems for electronic acoustic enhancement have been introduced. Some have disappeared over time but others appear to be settled firmly. The claim for these systems is that the acoustics of a venue can be changed at the press of a button, at cost significantly lower than variable acoustics by structural means. Also, in situations where the natural acoustics did not work out properly, these systems are used to correct the flaws, again at lower costs than the structural alternatives. The question, of course, is do these systems really work and if so, what are the differences between these systems? And what would be the structural alternatives?

This tutorial will identify what acoustic properties can or should be influenced by an acoustic enhancement system. In relation to this, working principles and philosophies of some of the most popular systems are analyzed and similarities and differences are identified and related to specific acoustic situations.

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

Broadcast/Streaming Media Session 1
Thursday, Oct. 17 9:00 am – 10:30 am
Room 1E08

TELEVISION LOUDNESS AND METADATA

Chair: **Fred Willard**, Univision, Washington, DC, USA

Presenters: *J. Todd Baker*, DTS
Arne Borsum, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Florian Camerer, ORF - Austrian TV, Vienna, Austria; EBU - European Broadcasting Union
Tim Carroll, Linear Acoustic Inc., Lancaster, PA, USA
Michael Kahsnitz, RTW, Pulheim, Germany
Robert Orban, Orban, San Leandro, CA, USA

Television broadcasters and Multichannel Video Program Distributors (MVPDs) are required to put in place procedures, software, and hardware to “effectively control program-to-interstitial loudness … and loudness management at the boundaries of programs and interstitial content.” Objective data must be supplied to the FCC to support compliance with the legislation as well as timely resolution of listener complaints. Similar rules have been developed in the UK and other parts of the world.

Members of our panel of experts have worked tirelessly to either create loudness control recommendations that have become the law or to bring those recommendations to implementation at the companies they represent. This session will cover the FCC’s Report and Order on the CALM Act, the development of the ATS’s A/85 Recommended Practice that is now part of the U.S. legislation and both domestic and European technical developments by major media distributors and P/LOUD.

Live Sound Seminar 1 Thursday, Oct. 17
9:00 am – 11:00 am Room 1E12

AC POWER AND GROUNDING

Chair: **Bruce C. Olson**, Olson Sound Design, Brooklyn Park, MN, USA; Ahnert Feistel Media Group, Berlin, Germany

Panelist: *Bill Whitlock*, Jensen Transformers

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

Product Design Session 1 Thursday, Oct. 17
9:00 am – 10:30 am Room 1E10

BEST PRACTICES IN AUDIO SOFTWARE DEVELOPMENT

Presenter: **Pascal Brunet**, Setem Technologies, Newbury, MA, USA

This presentation reviews best practices accumulated through 25 years of software development experience. We first present the classical development “V” cycle: requirement specifications, prototyping, design (general and specific), coding and tests, validation. We then focus on each individual step: what should be included in good specifications; importance and good usage of prototyping; design methods; coding guidelines; testing methods; independent validation and beta testing. We finish with miscellaneous topics: estimation methods and risk assessment, project management, team work, source code control.

Thursday, Oct. 17 9:00 am Room 1E04
Technical Committee Meeting: Audio for Games

Thursday, Oct. 17 10:00 am Room 1E04
Technical Committee Meeting: Perception and Subjective Evaluation of Audio Signals

Workshop 2 Thursday, Oct. 17
10:30 am – 12:00 noon Room 1E13

FX DESIGN PANEL: COMPRESSION

Chair: **Alex Case**, University of Massachusetts—Lowell, Lowell, MA, USA

Panelists: *David Derr*, Empirical Labs, Parsippany, NJ, USA
Dave Hill, Dave Hill Designs, Crane Song
Colin McDowell, McDSP Filterbank

Meet the designers whose talents and philosophies are reflected in the products they create, driving sound quality, ease of use, reliability, price, and all the other attributes that motivate us to patch, click, and tweak their effects processors.

Workshop 3 **Thursday, Oct. 17**
10:30 am – 12:30 pm **Room 1E14**

ACOUSTIC ENHANCEMENTS SYSTEMS —IMPLEMENTATIONS

Chair: **Ben Kok**, SCENA acoustic consultants, Uden, The Netherlands

Panelists: *Steve Barbar*, Lares Associates
Diemer de Vries
Peter Mapp, Peter Mapp Associates, Colchester, Essex, UK
Thomas Sporer, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany; Ilmenau University of Technology, Ilmenau, Germany
Takayuki Watanabe, Yamaha Corp., Hamamatsu, Shizuoka, Japan
Wieslaw Woszczyk, McGill University, Montreal, Quebec, Canada

Acoustic enhancement systems offer the possibility to change the acoustics of a venue by electronic means. How this is achieved varies by the working principle and philosophy of the system implemented. In this workshop various researchers, consultants, and suppliers active in the field of enhancement systems will discuss working principles and implementations.

This workshop is in close relation with the tutorial on acoustic enhancement systems; people not yet too familiar with the applications and working principles of these systems are recommended to attend the tutorial before attending the workshop.

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

Tutorial 3 **Thursday, Oct. 17**
10:30 am – 11:15 am **Room 1E11**

3-D AUDIO—EXPERIENCE THE SOUND OF THE FUTURE

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

The experience of listening to 3-D audio is an amazing adventure! 3-D is available in large cinema environments now but will it arrive in home applications or on mobile entertainment anytime? Furthermore, how will content be created and edited, and which tools can enable this? Is it a large investment of money and time to enhance the skills of the engineers? These questions will be discussed and many 3-D audio examples of different genres like music, game and film will be shown at the dedicated listening session at NYU on Saturday. See the section on Workshops with Height, page 153.

Broadcast/Streaming Media Session 2
Thursday, Oct. 17 **10:30 am – 12:00 noon**
Room 1E08

AUDIO FOR 4K TV

Chair: **Jonathan Abrams**, Nutmeg Post, New York, NY, USA

Presenters: *Robert Bleidt*, Fraunhofer USA Digital Media Technologies, San Jose, CA, USA
Tim Carroll, Linear Acoustic Inc., Lancaster, PA, USA
Dave Casey, DTS
Poppy Crum, Dolby
Robert Orban, Orban, San Leandro, CA, USA
Robert Reams, Psyx Research
Jim Starzynski, NBC Universal, New York, NY, USA

4K Television is the future. Video will be improved but what is happening to the audio? How will audio enhance the video experience? This panel will discuss television's future sound.

Session P3 **Thursday, Oct. 17**
11:00 am – 12:00 noon **Room 1E07**

AUDIO EDUCATION

Chair: **Tim Ryan**, Webster University, St. Louis, MO, USA

11:00 am

P3-1 Music to Our Ears: The Effect of Background Music in Higher Education Learning Environments—*Adam J. Hill*, University of Derby, Derby, Derbyshire, UK

Learning and teaching practice in higher education has embraced various forms of technology over recent years directed at enhancing the learning experience. Background music is well-known to benefit learning in elementary schools but has been largely ignored in higher education. There is evidence that background music is particularly beneficial for students with previous musical training, which is important for educators of audio engineering or similar courses linked closely with music. This work aims to determine if there are merits to background music in higher education and to point toward future work required to give definitive proof.
Convention Paper 8932

11:30 am

P3-2 Recording History in Audio Education— *Jeffrey Ratterman*, Front Range Community College, Fort Collins, CO, USA

This research will discuss the state of history education for the audio recording field. As the audio industry evolves, it is becoming more apparent that its history, for purposes of teaching, is rather unorganized. Also, on the whole, students learning the practice of audio recording are not being well educated in its history. As a result, students studying to become audio experts are not gaining awareness of a fundamental aspect of the field. This report surveys experiences and viewpoints of professionals, highlights existing recording history in audio education, and explores methods for audio history pedagogy. In addition, a suggested framework of historical periods of audio is presented and a concise list of resources is suggested.
Convention Paper 8933

Workshop 4 **Thursday, Oct. 17**
11:00 am – 1:00 pm **Room 1E12**

MICROPHONE SPECIFICATIONS—BELIEVE IT OR NOT

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

Panelists: *Juergen Breitlow*, Neumann, Berlin, Germany
Jackie Green, Audio-Technica U.S., Inc., Stow, OH, USA
Bill Whitlock, Jensen Transformers, Inc., Chatsworth, CA, USA; Whitlock Consulting, Oxnard, CA, USA
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 the 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 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.

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

Thursday, Oct. 17 **11:00 am** **Room 1E04**
Technical Committee Meeting: Archiving, Restoration, and Digital Libraries

Tutorial 4 **Thursday, Oct. 17**
11:15 am – 12:00 noon **Room 1E11**

AURO 3-D DISCOVERING THE CEILING FOR STEREO AND SURROUND

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

The 3-D sound (especially Auro 3-D sound) dominates our production studios more and more. At the same time the listening and work habits among professionals and music lovers are also changing. The use of the acoustic environment of the sound sources plus that use of the rooms acoustic, alters the aesthetics of the recording. The room and its features are more consciously heard and perceived. The hall is not only used as a means of "beauty," but wins a role as an important component of communication in the sound perception.

Is the possibility of a ceiling reflection reproduction an enrichment for the stereo and surround?

One Auro 3-D microphone setup provides various mixes: stereo; surround; Auro 3-D.

The influence of the ceiling reflection—signals used in stereo and surround mixes and the sound image of the sound sources will be shown and discussed.

Game Audio Session 1 **Thursday, Oct. 17**
11:15 am – 12:30 pm **Room 1E10**

PLANES, TRAINS, AND AUTOMOBILES: CREATING AND IMPLEMENTING VEHICLE SOUNDS FOR GAMES

Presenter: **Mike Caviezel**, Microsoft Game Studios, Redmond, WA, USA

This session will discuss some of the basic vehicle audio design concepts commonly found in games today. We'll talk about system design, recording and sound design methodology, and various implementation techniques and tricks for making vehicles sound great in games.

Workshop 5 **Thursday, Oct. 17**
12:00 noon – 12:45 pm **Room 1E11**

HEIGHT CHANNELS: THEORY, PRACTICE, AND "EARS-ON" EXPERIENCE

Chair: **David Bowles**, Swineshead Productions LLC, Berkeley, CA, USA

Panelists: *Paul Geluso*, New York University, New York, NY, USA
Agnieszka Roginska, New York University, New York, NY, USA

Over the past century sound recording and reproduction has dealt with an increasing number of audio channels to simulate spatial dimensions via capturing horizontal axes in stereo and surround-sound. The next step in immersive audio is to move into the vertical dimension through height channel recording and reproduction. The members of this panel will discuss different recording techniques to capture height channels and whether this audio information can be integrated into conventional stereo and surround-sound recordings. Vital to this dialog is a clearer understanding via psychoacoustics of our hearing perception outside the horizontal planes: how this perception influences engineers' choices in microphone technique and speaker placement. Last, post-production methods to create 3-D sonic imagery for recordings not originating in 3-D will be discussed. This workshop will be divided into two parts: a technical panel discussion at Javits Center, followed by playback sessions at the James Dolan Studios at New York University.

Thursday, Oct. 17 **12:00 noon** **Room 1E04**
Technical Committee Meeting: Audio for Telecommunications

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Thursday, October 17, 1:00 pm – 2:00 pm
Room 1E15/16

Opening Remarks:
- Executive Director Bob Moses
- President Frank Wells
- Convention Chair Jim Anderson

Program:
- AES Awards Presentation
- Introduction of Keynote Speaker
- Keynote Address by Josh McDermott

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:

GOLD MEDAL AWARD
- **Floyd Toole**
- **Rudolph Van Gelder**

SILVER MEDAL AWARD
- **Laurence Fincham**

BOARD OF GOVERNORS AWARD
- **William Crabtree**
- **Michael Fleming**
- **Janos Gyori**
- **Michael Kelly**
- **Jim McTigue**
- **Jan Abildgaard Pedersen**
- **Valerie Tyler**
- **Umberto Zanghieri**

FELLOWSHIP AWARD
- **Theresa Leonard**
- **Joel Lewitz**
- **Tim Shuttleworth**

CITATION
- **Bozena Kostek**

HONORARY MEMBERSHIP AWARD
- **Ronald E. Uhlig**

Keynote Speaker

Josh McDermott is a perceptual scientist studying sound, hearing, and music in the Department of Brain and Cognitive Sciences at MIT. His research addresses human and machine audition using tools from experimental psychology, engineering, and neuroscience. He is particularly interested in using the gap between human and machine competence to both better understand biological hearing and design better algorithms for analyzing sound. McDermott obtained a BA in Brain and Cognitive Science from Harvard, an MPhil in Computational Neuroscience from University College London, a PhD in Brain and Cognitive Science from MIT, and postdoctoral training in psychoacoustics at the University of Minnesota and in computational neuroscience at NYU. He is the recipient of a Marshall Scholarship, a National Defense Science and Engineering fellowship, and a James S. McDonnell Foundation Scholar Award. He is currently an Assistant Professor in the Department of Brain and Cognitive Sciences at MIT. the title of his keynote is, "Understanding Audition via Sound Synthesis."

Humans infer many important things about the world from the sound pressure waveforms that enter the ears. In doing so we solve a number of difficult and intriguing computational problems. We recognize sound sources despite large variability in the waveforms they produce, extract behaviorally relevant attributes that are not explicit in the input to the ear, and do so even when sound sources are embedded in dense mixtures with other sounds. This talk will describe recent progress in understanding these remarkable auditory abilities. The work stems from the premise that a theory of the perception of some property should enable the synthesis of signals that appear to have that property. Sound synthesis can thus be used to test theories of perception and to explore representations of sound. I will describe several examples of this approach.

Thursday, Oct. 17 2:00 pm Room 1E04
Technical Committee Meeting: Microphones and Applications

Thursday, Oct. 17 2:00 pm Room 1E02
Standards Committee Meeting: SC-02-02 Working Group on Digital Audio Input/Output Interfacing

Broadcast/Streaming Media Session 3
Thursday, Oct. 17 2:15 pm – 3:45 pm
Room 1E08

LISTENER FATIGUE AND RETENTION

Chair: **Richard Burden**, Richard W. Burden Associates, Canoga Park, CA, USA

Panelists: *Frank Foti*, Telos, New York, NY, USA
Greg Ogonowski, Orban, San Leandro, CA, USA
Sean Olive, Harman International, Northridge, CA, USA
Robert Reams, Psyx Research
Elliot Scheiner

This panel will discuss listener fatigue and its impact on listener retention. While listener fatigue is an issue of interest to broadcasters, it is also an issue of interest to telecommunications service providers, consumer electronics manufacturers, music producers, and others. Fatigued listeners to a broadcast program may tune out, while fatigued listeners to a cell phone conversation may switch to another carrier, and fatigued listeners to a portable media player may purchase another company's product. The experts on this panel will discuss their research and experiences with listener fatigue and its impact on listener retention.

Student/Career Development Event
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Thursday, October 17, 2:15 pm – 3:30 pm
Room 1E14

Moderator: **Colin Pfund**

Participants: *Andrea Pepper*
Simon-Claudius Wystrach
Magdalena Plewa
Kyle P. Snyder
Ezequiel Morfi
Marija Kovacina

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

All students and educators are invited to participate in this meeting. Election results and Recording Competition and Design Competition Awards will be given at the Student Delegate Assembly Meeting–Part 2 on Sunday, October 20.

Session P4 Thursday, Oct. 17
2:30 pm – 4:30 pm Room 1E07

ROOM ACOUSTICS

Chair: **Ben Kok**, SCENA acoustic consultants, Uden, The Netherlands

2:30 pm

P4-1 Investigating Auditory Room Size Perception with Autophonic Stimuli—*Manuj Yadav, Densil A. Cabrera, Luis Miranda, William L. Martens, Doheon Lee, Ralph Collins*, University of Sydney, Sydney, NSW, Australia

Although looking at a room gives a visual indicator of its "size," auditory stimuli alone can also provide an appreciation of room size. This paper investigates such aurally perceived room size by allowing listeners to hear the sound of their own voice in real-time through two modes: natural conduction and auralization. The auralization process involved convolution of the talking-listener's voice with an oral-binaural room impulse response (OBRIR; some from actual rooms, and others manipulated), which was output through head-worn ear-loudspeakers, and thus augmented natural conduction with simulated room reflections. This method allowed talking-listeners to rate room size without additional information about the rooms. The subjective ratings were analyzed against relevant physical acoustic measures derived from OBRIRs. The results indicate an overall strong effect of reverberation time on the room size judgments, expressed as a power function, although energy measures were also important in some cases.
Convention Paper 8934

3:00 pm

P4-2 Digitally Steered Columns: Comparison of Different Products by Measurement and Simulation—*Stefan Feistel*,[1] *Anselm Goertz*[2]
[1]AFMG Technologies GmbH, Berlin, Germany
[2]Institut für Akustik und Audiotechnik (IFAA), Herzogenrath, Germany

Digitally steered loudspeaker columns have become the predominant means to achieve satisfying speech intelligibility in acoustically challenging spaces. This work compares the performance of several commercially available array loudspeakers in a medium-size, reverberant church. Speech intelligibility as well as other acoustic quantities are compared on the basis of extensive measurements and computer simulations. The results show that formally different loudspeaker products provide very similar transmission quality. Also, measurement and modeling results match accurately within the uncertainty limits.
Convention Paper 8935

3:30 pm

P4-3 A Concentric Compact Spherical Microphone and Loudspeaker Array for Acoustical Measurements—*Luis Miranda, Densil A. Cabrera, Ken Stewart*, University of Sydney, Sydney, NSW, Australia

Several commonly used descriptors of acoustical conditions in auditoria (ISO 3382-1) utilize omnidirectional transducers for their measurements, disregarding the directional properties of the source and the direction of arrival of reflections. This situation is further complicated when the source and the receiver are collocated as would be the case for the acoustical characterization of stages as experienced by musicians. A potential solution to this problem could be a concentric compact microphone and loudspeaker array, capable of synthesizing source and receiver spatial patterns. The construction of a concentric microphone and loudspeaker spherical array is presented in this paper. Such a transducer could be used to analyze the acoustic characteristics of stages for singers, while preserving the directional characteristics of the source, acquiring spatial information of reflections and preserving the spatial relationship between source and receiver. Finally, its theoretical response and optimal frequency range are explored.
Convention Paper 8936

4:00 pm

P4-4 Adapting Loudspeaker Array Radiation to the Venue Using Numerical Optimization of FIR Filters—*Stefan Feistel*,[1] *Mario Sempf*,[1] *Kilian Köhler*,[2] *Holger Schmalle*[1]
[1]AFMG Technologies GmbH, Berlin, Germany
[2]IBS Audio, Berlin, Germany

Over the last two decades loudspeaker arrays have been employed increasingly for sound reinforcement. Their high output power and focusing ability facilitate extensive control capabilities as well as extraordinary performance. Based on acoustic simulation, numerical optimization of the array configuration, particularly of FIR filters, adds a new level of flexibility. Radiation characteristics can be established that are not available for conventionally tuned sound systems. It is shown that substantial improvements in sound field uniformity and output SPL can be achieved. Different real-world case studies are presented based on systematic measurements and simulations. Important practical implementation aspects are discussed such as the spatial resolution of driven sources, the number of FIR coefficients, and the quality of loudspeaker data.
Convention Paper 8937

Session P5 **Thursday, Oct. 17**
2:30 pm - 5:30 pm **Room 1E09**

SIGNAL PROCESSING—PART 2

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

2:30 pm

P5-1 Evaluation of Dynamics Processors' Effects Using Signal Statistics—*Tim Shuttleworth*, Oceanside, CA, USA

Existing methods of evaluating the action of dynamics processors, i.e., limiters, compressors, expanders, and gates do not provide results that have a direct correlation with the perceived and actual effect on the signals dynamics; aspects such as crest factor, dynamic range, and subjective acceptability of the processed signal or degree of optimization of the use of the transmission medium. A method is described that uses statistical analysis of the pre- and post-processed signal

to allow the processor's action to be characterized in a manner that correlates to the perceived effects and actual modification of signal dynamics. A number of signal statistical and user definable characteristics are introduced and, in addition to well-known statistical techniques, form the basis for this evaluation method.
Convention Paper 8938

3:00 pm

P5-2 A New Ultra Low Delay Audio Communication Coder—*Brijesh Singh Tiwari,*[1] *Midathala Harish,*[1] *Deepen Sinha*[2]
[1]ATC Labs, Noida, India
[2]ATC Labs, Newark, NJ, USA

We propose a new full bandwidth audio codec that has algorithmic delay requirement as low as 0.67 ms to a maximum of 2.7 ms. Low delay is a critical requirement in real many time applications such as networked music performances, wireless speakers and microphones, and Bluetooth devices. The proposed Ultra Low Delay Audio Communication Codec (ULDACC) is a perceptual transform codec utilizing very small transform windows the shape of which is optimized to compensate for the lack of frequency resolution. Specially adapted psychoacoustic model and intra-frame coding techniques are employed to achieve transparent audio quality for bit rates approaching 128 kbps/channel at the algorithmic delay of about 1 ms.
Convention Paper 8939

3:30 pm

P5-3 Cascaded Long Term Prediction of Polyphonic Signals for Low Power Decoders
—*Tejaswi Nanjundaswamy, Kenneth Rose,* University of California, Santa Barbara, Santa Barbara, CA, USA

An optimized cascade of long term prediction filters, each corresponding to an individual periodic component of the polyphonic audio signal, was shown in our recent work to be highly effective as an inter-frame prediction tool for low delay audio compression. The earlier paradigm involved backward adaptive parameter estimation, and hence significantly higher decoder complexity, which is unsuitable for applications that pose a stringent power constraint on the decoder. This paper overcomes this limitation via extension to include forward adaptive parameter estimation, in two modes that trade complexity for side information: (i) a subset of parameters is sent as side information and the remaining is backward adaptively estimated; (ii) all parameters are sent as side information. We further exploit inter-frame parameter dependencies to minimize the side information rate. Objective and subjective evaluation results clearly demonstrate substantial gains and effective control of the tradeoff between rate-distortion performance and decoder complexity.
Convention Paper 8940

4:00 pm

P5-4 Voice Coding with Opus—*Koen Vos,*[1] *Karsten Vandborg Sørensen,*[1] *Søren Skak Jensen,*[2] *Jean-Marc Valin*[3]
[1]vocTone, San Francisco, CA, USA
[2]GN Netcom A/S, Ballerup, Denmark
[3]Mozilla Corporation, Mountain View, CA, USA

In this paper we describe the voice mode of the Opus speech and audio codec. As only the decoder is standardized, the details in this paper will help anyone who wants to modify the encoder or gain a better understanding of the codec. We go through the main components that constitute the voice part of the codec, provide an overview, give insights, and discuss the design decisions made during the development. Tests have shown that Opus quality is comparable to or better than several state-of-the-art voice codecs, while covering a much broader application area than competing codecs.
Convention Paper 8941

4:30 pm

P5-5 High-Quality, Low-Delay Music Coding in the Opus Codec—*Jean-Marc Valin,*[1] *Gregory Maxwell,*[1] *Timothy B. Terriberry,*[1] *Koen Vos*[2]
[1]Mozilla Corporation, Mountain View, CA, USA
[2]vocTone, San Francisco, CA, USA

The IETF recently standardized the Opus codec as RFC6716. Opus targets a wide range of real-time Internet applications by combining a linear prediction coder with a transform coder. We describe the transform coder, with particular attention to the psychoacoustic knowledge built into the format. The result out-performs existing audio codecs that don't operate under real-time constraints.
Convention Paper 8942

Workshop 6 — Thursday, Oct. 17
2:30 pm – 4:30 pm — Room 1E10

DESIGN AND USAGE OF ANCHORS IN LISTENING TESTS FOR AUDIO QUALITY EVALUATION

Chair: **Frederik Nagel**, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany; International Audio Laboratories, Erlangen, Germany

Panelists: *Schuyler Quackenbush*, Audio Research Labs, Scotch Plains
Francis Rumsey, Logophon Ltd., Oxfordshire, UK
Thomas Sporer, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany; Ilmenau University of Technology, Ilmenau, Germany
Nick Zacharov, DELTA SenseLab, Iisalmi, Finland

Listening tests for the evaluation of audio or speech quality (like MUSHRA or P800) employ anchors in order to stabilize rating scales and facilitate comparison among testing sites. As coding technology progressed, traditional anchors (including band-limited or noisy signals) are no longer related to artifacts occurring in state-of-the art audio codecs. Expert listeners probably have an internal reference for basic audio quality that may allow them to evaluate audio quality even without any anchors in case of mono and two–channel stereo material. However, especially for newly emerging technologies (including multichannel audio) anchors could have a huge impact as many listeners are still building their internal reference scale.

We will discuss:

• How should anchors be created in order to exhibit typical artifacts of modern audio codecs?

• Is it possible to create anchors with an expected quality for a variety of different input material?

• Which features should anchors have that are used in multichannel testing?

• How important are anchors at all, compared to internal anchors and references of experts?

• What effect has modification of anchors to the results?

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

Workshop 7 **Thursday, Oct. 17**
2:30 pm – 4:00 pm **Room 1E13**

TOOLS AND WORKFLOW FOR THE CREATION OF IMMERSIVE CONTENT

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

Panelists: *Fred Maher*, DTS
Jurgen Scharpf, Dolby
Wilfried Van Baelen, Auro Technologies, Mol, Belgium
Brian Vessa, Sony Pictures

In this workshop the requirements for new tools and workflows to create three-dimensional, immersive content are discussed. The panel members present their own solutions and discuss the creative possibilities as well as the requirements for compatibility between different systems and deliverables.

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

Game Audio Session 2 **Thursday, Oct. 17**
2:30 pm – 4:30 pm **Room 1E11**

DIABLO III—POST MORTEM

Presenters: **Russell Brower**, Blizzard Entertainment
Derek Duke, Blizzard Entertainment
Joseph Lawrence, Blizzard Entertainment

Look behind the curtain as the Audio Team behind Diablo III shows us the world of game audio development from multiple perspectives—the audio director, sound designer, and composer. Discover the tips, tricks, and techniques of a major AAA title's audio design process from conception to completion in this postmortem.

Live Sound Seminar 2 **Thursday, Oct. 17**
2:30 pm – 4:30 pm **Room 1E12**

AUDIO NETWORK AND TRANSPORT

Chair: **Jim Risgin**, On Stage Audio, Wood Dale, IL, USA

Panelists: *Mark Dittmar*, Firehouse Productions
Phil Reynolds, System Tech, The Killers
Rob Silfvast, Avid

As audio and control over network become more predominate in today's live sound environment managing the network becomes more challenging. This panel will discuss the problems, challenges, and solutions required that are associated with sharing the bandwith between audio and control as well as the unique challenges created by all the different manufacturers and protocols. Our discussion will rely heavily upon questions and comments from the audience as your experiences, pitfalls and questions are central to these common challenges today.

Session P6 **Thursday, Oct. 17**
3:00 pm – 4:30 pm **1E Foyer**

POSTERS: SPATIAL AUDIO

3:00 pm

P6-1 Improvement of 3-D Sound Systems by Vertical Loudspeaker Arrays—*akira saji, Keita Tanno, Jie Huang*, University of Aizu, Aizuwakamatsu City, Japan

Recently we proposed a 3-D sound system using a horizontal arrangement of loudspeakers by combining the effect of HRTF and the amplitude panning method. In that system, loudspeakers are set at the height of subject's ear level and its sweet-spot is limited by the height of loudspeakers. When the listener's ear level is different from the loudspeakers, it will cause difficulty of sound localization or breakdown of sound localization. However, it is difficult to adjust properly both the height of loudspeakers and subject's ear level every time. In this paper we aimed to improve the robustness of the 3-D sound system using vertical loudspeaker arrays. As a result of experiments, we prove that the loudspeaker arrays can improve the robustness of the 3-D sound system.
Convention Paper 8944

3:00 pm

P6-2 An Integrated Algorithm for Optimized Playback of Higher Order Ambisonics— *Robert E. Davis, D. Fraser Clark*, University of the West of Scotland, Paisley, Scotland, UK

An algorithm is presented that gives improved playback performance of higher order ambisonic material on practical loudspeaker arrays. The optimizations are based on sound field reproduction theories with additional parameters to account for the compensation of loudspeaker and listener positioning constraints and numbers of listeners. Automatic calculation of loudspeaker distances is also achieved based on room dimensions and a gain calibration routine is incorporated. Results are given to quantify the resulting algorithm performance, informal listening tests were carried out, and aspects of implementation are discussed.
Convention Paper 8945

3:00 pm

P6-3 I Hear NY3D: Ambisonic Capture and Reproduction of an Urban Sound Environment—*Braxton Boren, Areti Andreopoulou, Michael Musick, Hariharan Mohanraj, Agnieszka Roginska*, New York University, New York, NY, USA

This paper describes "I Hear NY3D," a project for capturing and reproducing 3D soundfields in New York City. First order Ambisonic recordings of various locations in Manhattan have taken place, to be used for both aesthetic and informational purposes. The collected data allows for the creation of high quality, fully immersive auditory soundscapes that can be played back at any periphonic speaker array configuration through real time matrixing. Binaural renderings of the same data are also available for more portable applications.
Convention Paper 8946

3:00 pm

P6-4 I Hear NY3D: An Ambisonic Installation Reproducing NYC Soundscapes—*Michael Musick, Areti Andreopoulou, Braxton Boren, Hariharan Mohanraj, Agnieszka Roginska*, New York University, New York, NY, USA

This paper describes the development of a reproduction installation for the "I Hear NY3D" project. This project's aim is the capture and reproduction of immersive soundfields around Manhattan. A means of creating an engaging reproduction of these soundfields through the medium of an installation will also be discussed. The goal for this installation is an engaging, immersive experience that allows participants to create connections to the soundscapes and observe relationships between the soundscapes. This required the consideration of how to best capture and reproduce these recordings, the presentation of simultaneous multiple soundscapes, and a means of interaction with the material.
Convention Paper 8947

3:00 pm

P6-5 Auralization of Measured Room Impulse Responses Considering Head Movements— *Anthony Parks, Jonas Braasch, Samuel W. Clapp*, Rensselaer Polytechnic Institute, Troy, NY, USA

The purpose of this paper is to describe a novel method for auralizing measured room impulse responses over headphones using impulse responses recorded from a 16-channel spherical microphone array decoded to eight virtual loudspeakers mixed-down binaurally using nonindividualized HRTFs. The novelty of this method lies not in the ambisonic binaural-mixdown process, but rather, the use of head pose estimation code from the Kinect API sent to a Max/MSP patch using Open Sound Control messages. This provides a fast, reliable alternative to auralizations over headphones that allow for head movements without the need for head-related transfer function interpolation by performing a rotation on the spherical harmonic that corresponds to the listener's head rotation.
Convention Paper 8948

3:00 pm

P6-6 Reduced Representations of HRTF Datasets: A Discriminant Analysis Approach—*Areti Andreopoulou, Agnieszka Roginska, Juan Pablo Bello*, New York University, New York, NY, USA

This paper discusses reduced representations of HRTF datasets, fully descriptive of one's personalized properties. The data reduction is achieved through elimination of the least discriminative binaural-filter pairs from a set. For this purpose Linear Discriminant Analysis (LDA) was applied on the Music and Audio Research Laboratory (MARL) database of repeated HRTF measurements, which resulted in 67% data reduction. The effectiveness of the sparse HRTF mapping is assessed by way of the performance of a database matching system, followed by a subjective evaluation study. The results indicate that participants have demonstrated strong preference towards the selected HRTF sets, in contrast to the generic KEMAR set and the least similar selections from the repository.
Convention Paper 8949

3:00 pm

P6-7 Investigation of HRTF Sets Using Content with Limited Spatial Resolution—*Johann-Markus Batke*,[1] *Stefan Abeling*,[1] *Stefan Balke*,[2] *Gerald Enzner*[3]
[1]Audio & Acoustics, Technicolor Research & Innovation, Hannover, Germany
[2]Leibniz Universität Hannover, Hannover, Germany
[3]Ruhr-Universität Bochum, Bochum, Germany

Headphone rendering of sound fields represented by Higher Order Ambisonics (HOA) is greatly facilitated by the binaural synthesis of virtual loudspeakers. Individualized head-related transfer function (HRTF) sets corresponding to the spatial positions of the virtual loudspeakers are used in conjunction with head-tracking to achieve the externalization of the sound event. We investigate the localization accuracy for HOA representations of limited spatial resolution using individualized and generic HRTF sets.
Convention Paper 8950

Thursday, Oct. 17 3:00 pm Room 1E04
Technical Committee Meeting: Loudspeakers and Headphones

Student/Career Development Event
EDUCATION FORUM PANEL
Thursday, October 17, 3:30 pm – 4:45 pm
Room 1E14

Moderator: **John Krivit**, Chair, AES Education Committee

Panelists: *Wesley Bulla*, Belmont University, Nashville, TN, USA
Bill Crabtree, Middle Tennessee State University, Murfreesboro, TN, USA
Michael Fleming
George Massenburg, McGill University, Montreal,Quebec, Canada
Konrad Strauss

Audio Preservation & Education

Three topics of importance to audio educators will be briefly presented:
• Konrad Strauss & George Massenburg (Audio Preservation and Education)
• Wesley Bulla (ABET Accreditation and a Possible Role for the AES)

• Bill Crabtree & Michael Fleming (Synopsis of the 50th AES Conference on Audio Education)

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

Broadcast/Streaming Media Session 4
Thursday, October 17 3:45 pm – 5:15 pm
Room 1E08

LOUDNESS CONTROL FOR RADIO AND INTERNET STREAMING

Moderator: **David Bialik**, CBS, New York, NY, USA

Panelists: *Robert Bleidt*, FraunhoferUSA
Florian Camerer, ORF, Vienna, Austria
Frank Foti, Telos, New York, NY, USA
John Kean, NPR
Robert Orban, Orban, San Leandro, CA, USA

Is the "Loudness War" in radio over? Has it moved over to internet streaming? With content being injected from multiple sources, levels are varying. How can we control level without disrupting the audience? Some countries are introducing regulation—is it needed?

Thursday, Oct. 17 4:00 pm Room 1E04
Technical Committee Meeting: Automotive Audio

Session P7 Thursday, Oct. 17
4:30 pm – 7:00 pm Room 1E07

SPATIAL AUDIO—PART 1

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

4:30 pm

P7-1 Reproducing Real-Life Listening Situations in the Laboratory for Testing Hearing Aids— *Pauli Minnaar, Signe Frølund Albeck, Christian Stender Simonsen, Boris Søndersted, Sebastian Alex Dalgas Oakley, Jesper Bennedbæk*, Oticon A/S, Smørum, Denmark

The main purpose of the current study was to demonstrate how a Virtual Sound Environment (VSE), consisting of 29 loudspeakers, can be used in the development of hearing aids. A listening test was done by recording everyday sound scenes with a spherical microphone array that has 32 microphone capsules. The playback in the VSE was implemented by convolving the recordings with inverse filters, which were derived by directly inverting a matrix of 928 measured transfer functions. While listening to 5 sound scenes, 10 hearing-impaired listeners could switch between hearing aid settings in real time, by interacting with a touch screen in a MUSHRA-like test. The setup proves to be very valuable for ensuring that hearing aid settings work well in real-world situations.
Convention Paper 8951

5:00 pm

P7-2 Measuring Speech Intelligibility in Noisy Environments Reproduced with Parametric Spatial Audio—*Teemu Koski*,[1] *Ville Sivonen*,[2] *Ville Pulkki*[1]
[1]Aalto University, Espoo, Finland
[2]Cochlear Nordic AB, Vantaa, Finland

This work introduces a method for speech intelligibility testing in reproduced sound scenes. The proposed method uses background sound scenes augmented by target speech sources and reproduced over a multichannel loudspeaker setup with time-frequency domain parametric spatial audio techniques. Subjective listening tests were performed to validate the proposed method: speech recognition thresholds (SRT) in noise were measured in a reference sound scene and in a room where the reference was reproduced by a loudspeaker setup. The listening tests showed that for normally-hearing test subjects the method provides nearly indifferent speech intelligibility compared to the real-life reference when using a nine-loudspeaker reproduction setup in anechoic conditions (<0.3 dB error in SRT). Due to the flexible technical requirements, the method is potentially applicable to clinical environments.
Convention Paper 8952

5:30 pm

P7-3 On the Influence of Headphones on Localization of Loudspeaker Sources— *Darius Satongar*,[1] *Chris Pike*,[2] *Yiu W. Lam*,[1] *Anthony I. Tew*[3]
[1]University of Salford, Salford, Greater Manchester, UK
[2]BBC Research and Development, Salford, Greater Manchester, UK
[3]University of York, York, UK

When validating systems that use headphones to synthesize virtual sound sources, a direct comparison between virtual and real sources is sometimes needed. This paper presents objective and subjective measurements of the influence of headphones on external loudspeaker sources. Objective measurements of the effect of a number of headphone models are given and analyzed using an auditory filter bank and binaural cue extraction. Objective results highlight that all of the headphones had an effect on localization cues. A subjective localization test was undertaken using one of the best performing headphones from the measurements. It was found that the presence of the headphones caused a small increase in localization error but also that the process of judging source location was different, highlighting a possible increase in the complexity of the localization task.
Convention Paper 8953

6:00 pm

P7-4 Binaural Reproduction of 22.2 Multichannel Sound with Loudspeaker Array Frame— *Kentaro Matsui, Akio Ando*, NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

NHK has proposed a 22.2 multichannel sound system to be an audio format for future TV broadcasting. The system consists of 22 loudspeakers and 2 low frequency effect loudspeakers for reproducing three-dimensional spatial

sound. To allow 22.2 multichannel sound to be reproduced in homes, various reproduction methods that use fewer loudspeakers have been investigated. This paper describes binaural reproduction of 22.2 multichannel sound with a loudspeaker array frame integrated into a flat panel display. The processing for binaural reproduction is done in the frequency domain. Methods of designing inverse filters for binaural processing with expanded multiple control points are proposed to enlarge the listening area outside the sweet spot.
Convention Paper 8954

6:30 pm

P7-5 An Offline Binaural Converting Algorithm for 3D Audio Contents: A Comparative Approach to the Implementation Using Channels and Objects—*Romain Boonen*, SAE Institute Brussels, Brussels, Belgium

This paper describes and compares two offline binaural converting algorithms based on HRTFs (Head-Related Transfer Functions) for 3D audio contents. Recognizing the widespread use of headphones by the typical modern audio content consumer, two strategies to binaurally translate the 3D mixes are explored in order to give a convincing 3D aural experience "on the go." Aiming for the best output quality possible and avoiding the compromises inherent to real-time processing, the paper compares the channel- and the object-based models, notably looking respectively into the spectral analysis of channels for usage of HRTFs at intermediate positions between the virtual speakers and the dynamic convolution of the HRTFs with the objects according to their position coordinates in time.
Convention Paper 8955

Workshop 8 — Thursday, Oct. 17
4:30 pm – 7:00 pm — Room 1E11

DIGITAL ROOM CORRECTION—DOES IT REALLY WORK?

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

Ass't Chair: **Michael Chafee**, Michael Chafee Enterprises, Sarasota, FL, USA

Panelists: *Ulrich Brüggemann*, AudioVero, Herzebrock, Germany
Will Eggleston, Genelec Inc., Natick, MA, USA
Curt Hoyt, 3-D Audio Consultant, Huntington Beach, CA, USA; Trinnov Audio USA Operations
Floyd Toole, Acoustical consultant to Harman, ex. Harman VP Acoustical Engineering, Oak Park, CA, USA

The practice of digital room and loudspeaker correction (DRC) is an especially fruitful beneficiary of Moore's law and increased skills among DSP programmers. DRC is a hot topic of interest for recording, mixing and mastering engineers, and studio designers. The workshop will explore the principles of DRC with three panelists and an expert guest.

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

Live Sound Seminar 3 — Thursday, Oct. 17
4:30 pm – 6:30 pm — Room 1E12

SOUND SYSTEM OPTIMIZATION

Chair: **Bob McCarthy**, Meyer Sound Labs

Panelists: *Jamie Anderson*, Optimization program developer, system tuner, educator
Josh Evans, system designer, tuner, and educator
John Sandrett, sound co owner and system designer
Tom Young, system designer, and tuner
Geoff Zink, system designer

Sound system tuning is a multi-step process that begins long before the pink noise can be heard and generally goes until the keyboard is pried from our hands. What are the steps and procedures taken to ensure a successful tuning? How do we convince clients to give us the time and resources to do this vital work? How do we prioritize our limited resources when time is short? (Like always) What can be done ahead of time?

Panel members are all very experienced with the process of system optimization, albeit from a variety of perspectives. Please join us and add your voice to this discussion of system optimization.

Network Audio Session 1 — Thursday, Oct. 17
4:30 pm – 6:00 pm — Room 1E13

ONE NETWORK TO RULE THEM ALL

Chair: **Kevin Gross**, AVA Networks, Boulder, CO, USA

Panelists: *Mattias Allevik*, Video Corporation of America, New York, NY, USA
Dave Revel, Technical Multimedia Design, Inc. Burbank, CA, USA

Networked audio distribution is now less frequently accomplished as a separate infrastructure. The promise of running audio on the same network as other facility services and applications is now coming to fruition. This workshop will discuss the motivation for combining services, the challenges in doing so, and requirements this approach puts on audio networking technologies.

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

Tutorial 5 — Thursday, Oct. 17
4:45 pm – 6:45 pm — Room 1E10

AN OVERVIEW OF AUDIO SYSTEM GROUNDING AND SIGNAL INTERFACING

Presenter: **Bill Whitlock**, Jensen Transformers, Inc., Chatsworth, CA, USA; Whitlock Consulting, Oxnard, CA, USA

Equipment makers like to pretend the problems don't exist, but this tutorial replaces hype and myth with insight and knowledge, revealing the true causes of system noise and "ground loops." Unbalanced interfaces are exquisitely vulnerable to noise due to an intrinsic prob- ➥

lem. Although balanced interfaces are theoretically noise-free, they're widely misunderstood by equipment designers, which often results in inadequate noise rejection in real-world systems. Because of a widespread design error, some equipment has a built-in noise problem. Simple, no-test-equipment troubleshooting methods can pinpoint the exact location and cause of system noise. Ground isolators in the signal path solve the fundamental noise coupling problems. Optimum interfaces between unbalanced and balanced connections, RF interference, and power-line treatments are also discussed. Some widely-used "cures" are both illegal and deadly.

Workshop 9 Thursday, Oct. 17
5:00 pm – 6:30 pm Room 1E14

CAN WE MEASURE EMOTIONS?

Chair: **Judith Liebetrau**, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany; Ilmenau University of Technology, Ilmenau, Germany

Panelists: *Frederik Nagel*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany; International Audio Laboratories, Erlangen, Germany
Mark Sandler, Queen Mary University of London, London, UK
Chia-Jung Tsay

Music evokes and carries emotions. Music emotion recognition (MER), as a part of music information retrieval (MIR), examines the question: Which parts of music evoke what emotions and how can they be automatically classified? Classification systems need to be trained in terms of feature selection and prediction. As training data, musical pieces of which the average emotional impact is known have to be used. Due to the subjectivity of emotions, the generation of such ground truth data poses several challenges. In this workshop obstacles in measuring and automatically predicting emotions evoked by music will be displayed.

Among others, the workshop will address the following topics: What is an emotion and how can it be defined? Is there a difference between felt and perceived emotion? Can the emotional impact of a musical piece be subjectively measured? Can the emotional impact of a musical piece be predicted?

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

Special Event
PRODUCING ACROSS GENERATIONS: NEW CHALLENGES, NEW SOLUTIONS—MAKING RECORDS FOR NEXT TO NOTHING IN THE 21ST CENTURY
Thursday, October 17, 5:00 pm – 7:00 pm
Room 1E15/16

Moderator: **Nick Sansano**

Panelists: *Frank Filipetti*
Jesse Lauter
Carter Matschullat
Bob Power
Hank Shocklee
Craig Street

Budgets are small, retail is dying, studios are closing, fed up audiences are taking music at will … yet devoted music professionals continue to make records for a living. How are they doing it? How are they getting paid? What type of contracts are they commanding? In a world where the "record" has become an artists' business card, how will the producer and mixer derive participatory income? Are studio professionals being left out of the so-called 360 deals? Let's get a quality bunch of young rising producers and a handful of seasoned vets in a room and finally open the discussion about empowerment and controlling our own destiny.

Thursday, Oct. 17 5:00 pm Room 1E04
Technical Committee Meeting: Electro Magnetic Compatibility

Thursday, Oct. 17 5:00 pm Room 1E02
Standards Committee Meeting: SC-02-08 Working Group on Audio-File Transfer and Exchange

Session EB1 Thursday, Oct. 17
5:30 pm – 7:00 pm Room 1E09

AUDIO PROCESSING

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

5:30 pm

EB1-1 Modeling the Korg35 Lowpass and Highpass Filters—*Will Pirkle*, University of Miami, Coral Gables, FL, USA

The Korg35 Filter is a voltage controlled Sallen-Key topology capable of producing both lowpass (LPF) and highpass (HPF) filter responses. It is known for its ability to self-oscillate as well as its saturated or distorted output as the damping factor of the filter approaches zero where self-oscillation occurs. Both LPF and HPF are second order but the highpass version features a 6 dB/octave roll off instead of the usual 12 dB/octave giving it a distinctive sound with more bass response. The analog Sallen-Key topology uses a delay-less positive feedback loop to implement the Q control of the filter. The saturation circuit is also inside this loop. Rather than use a typical biquad or state variable filter, we choose to use Virtual Analog (VA) filters [1] as building blocks to replicate the analog filter topology directly from its signal flow graph, including the delay-less loop as well as the saturation circuit. Both loaded and unloaded (lossy) versions of the Sallen-Key topology were designed and implemented in C++ with the point of self-oscillation exactly matching both analog transfer functions. Sample code and extra documentation are available at www.willpirkle.com.
Engineering Brief 103

5:45 pm

EB1-2 Modeling the Harmonic Exciter—*Priyanka Shekar, Julius O. Smith, III*, Stanford University, Stanford, CA, USA

A harmonic exciter is an audio effects signal processor applied to enhance the brightness and

clarity of a sound, particularly used for vocals. This is achieved by inducing a measured amount of high-frequency distortion. In this paper an exciter is digitally modeled and implemented as a standalone application (or plugin) using the FAUST (Functional AUdio STream) programming language for real-time audio. The model is based on the Aural Exciter by Aphex Ltd., an analog hardware unit. Technical specifications of the Aural Exciter are drawn from the original 1979 patent. The digital model performs as expected, recreating an "intage" style audio effect.
Engineering Brief 104

6:00 pm

EB1-3 Fourier Transforms, Audio Engineering, and the Quantum Nature of Reality—*Scott Hawley*, Belmont University, Nashville, TN, USA

A short interdisciplinary, educational survey is presented illustrating ways in which audio spectral analysis and quantum physics are intimately related. A basic conceptual understanding of Fourier transforms and their applications in audio engineering is sufficient to grasp aspects of quantum wave packets, the Heisenberg Uncertainty Principle, and more. Similarly, concepts from quantum mechanics can inform the understanding of audio effects such as aliasing, convolutions and wavelet transforms. The presenter is a computational physicist who authored a computer audio analysis suite for audio engineering students, noting several interdisciplinary connections in the process.
Engineering Brief 105

6:15 pm

EB1-4 Experiments with Dither in Level-Calibrated Floating Point Audio Processing—*Douglas Rollow*, Sennheiser Technology and Innovation, San Francisco, CA, USA

The use of dither to decorrelate quantization error in fixed point signal processing systems is a well-established practice in professional audio. Floating point computation, however, is quite common due to the ease of use and ubiquity of high performance platforms, among other reasons. Dither is (anecdotally) less frequently found in floating point audio systems, until the final mapping to fixed point representation, but quantization error occurs in the rounding operation during intermediate calculations. Widrow and others have provided detailed treatment of the quantization error in floating point audio calculations, and in the present work experiments using dither during the internal rounding operation in a floating point unit are compared to the external addition of noise when the signal levels are known to be calibrated from the original analog source.
Engineering Brief 106

6:30 pm

EB1-5 Virtual Development of Audio Systems—Application of CAE Methods—*Alfred Svobodnik*, Konzept-X, Karlsruhe, Austria

This paper will give an overview of state-of-the-art CAE methods for virtual product development of audio systems, especially focusing on automotive audio and small transducers. Matrix based CAE methods will be discussed to be used for multiphysical modeling of transducers, acoustic enclosures (e.g., doors or rear shelves of automobiles, or acoustic systems for phones) and listening spaces, especially automotive car cabins. Ultimately a process model allowing the simulation of complete systems, representing a fully virtualized product development environment, will be shown.
Engineering Brief 107

6:45 pm

EB1-6 A New DSP Tool for Drum Leakage Suppression—*Elias Kokkinis, Alexandros Tsilfidis, Thanos Kostis, Kostas Karamitas*, accusonus, Patras Innovation Hub, Patras, Greece

Microphone leakage is a problem that sound engineers face every day. Leakage complicates audio editing, processing, and mixing, and it is a well known problem in drum recordings. To this day, sound engineers have only a limited amount of options available in order to address this problem. These mostly consist of simple and empirical methods. A novel technology that addresses the problem of microphone leakage in multichannel drum recordings is presented here. In addition we discuss the problem definition as deduced from the specific properties of drum recordings, as well as the resulting signal processing framework.
Engineering Brief 108

Broadcast/Streaming Media Session 5
Thursday, Oct. 17 5:30 pm – 7:00 pm
Room 1E08

IS IT TIME TO RETIRE THE MP3 PROTOCOL FOR STREAMING

Chair: **Ray Archie**, CBS, New York, NY, USA; Music is My First Language, New York, NY, USA

Panelists: *Karlheinz Brandenburg*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany; Ilmenau University of Technology, Ilmenau, Germany
John Kean, NPR
Jan Nordmann, Fraunhofer USA, San Jose, CA, USA
Greg Ogonowski, Orban, San Leandro, CA, USA
Greg Shay, The Telos Alliance, Cleveland, OH, USA

It has been over 25 years since the MP3 codec was introduced to the audio community. With lossy audio encoding, such as an MP3, there is a not-so-fine balance between audio quality and file size. With the ever increasing availability of bandwidth, file size has diminished as a consideration for audio streaming and codec related loss in audio quality is much more apparent.

This panel will be an in-depth discussion about this phenomenon. We will also discuss challenges related to introducing new codecs into the space. ➥

Network Audio Session 2 **Thursday, Oct. 17**
6:00 pm – 7:00 pm **Room1E13**

A PRIMER ON FUNDAMENTAL CONCEPTS OF MEDIA NETWORKING

Presenter: **Landon Gentry**, Audinate

This session will cover the OSI model and how data travels through network layers (a "networking stack"): Layers 1, 2, 3 and 4; Cables, MAC Addresses, IP Addresses, and networking protocols. An overview of some networking standards and standards organizations, including the IEEE and the IETF. An introduction to IP data networking . . . it is how everything is already wired together. Identify some of the advantages and limitations of IP data networks with respect to real-time media. A brief discussion of IP networking standards and protocols that can be leveraged for media networking.

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

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Thursday, October 17, 7:15 pm – 8:30 pm
Room 1E15/16

Lecturer: **George Massenburg**

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 135th AES Convention is George Massenburg. Massenburg has participated (individually and collaboratively) in over four hundred records over the past 45 years. His work includes recordings of Earth, Wind & Fire, Linda Ronstadt, Little Feat, Lyle Lovett, Aaron Neville, Mary Chapin Carpenter, Natasha Bedingfield, Herbie Hancock, Arlo Guthrie, Billy Joel, the Dixie Chicks and many more. His studio work has gained him international recognition and four Grammys (including the Grammy for Technical Achievement in 1998, at the time making him one of only seventeen individuals to receive that honor) as well as numerous Mix Magazine TEC Awards. In 1988 he also won the Academy of Country Music Record of the Year Award. George has designed, built, and managed several recording studios, and has contributed to the acoustical and architectural design of many other studios, including George Lucas' Skywalker Sound. In 1982, he founded George Massenburg Labs, a pioneering audio electronics company that has released an extensive range of innovative recording technologies, all based on his original designs. In 1999, he and a partner founded Massenburg Design Works, making high-resolution digital processors and plugins. George was awarded an honorary Doctorate of Music by Boston's Berklee College of Music. Regularly published in professional journals and trade magazines worldwide, he received the Gold Medal from the Audio Engineering Society in 2008. In 2013 he was assigned patent # 8,510,361 in the US (as well as other countries) for his Variable Exponent Averaging detector and Dynamic Range Controller. He is a member of the National Recording Preservation Board of the Library of Congress and an advisor to the Committee for Library Information Resources. George serves as the Chief Technical Officer of META (the Music Engineering Technical Alliance), a strategic union of music producers and engineers dedicated to the highest standards of audio and delivery of music. Currently, he is an Associate Professor of Sound Recording at the Schulich School of Music at McGill University in Montreal, Quebec, Canada; a Visiting Lecturer at the Berklee College of Music in Boston and Valencia, Spain; and the University of Memphis in Memphis, TN. He and his wife, Cookie Rankin, and dog, Charlie, live in Montreal, Quebec. The title of his lecture is, "4-44 and Me: Stagnation to Transformation: The Real Future of Music."

There are so many advances in technology and science these days, we are quickly overwhelmed with data. From batteries to the brain, new possibilities abound. It's difficult to know what information is meaningful and how to digest it, not to mention how to apply it in our own best interests. Often it seems we are immersed in irrelevant noise, hesitant to accommodate change.

It used to be extremely expensive to record anything of technical quality. Analog tape machines, ¼-inch, then ½-inch, then 1-inch, and 2-inch reels of tape, large-frame analog consoles, microphones, peripheral processing, reverb, relatively good studio acoustics were not affordable to most artists. Also, the expertise (producers, engineers, mixers, mastering engineers) needed to get the best sounds possible out of the air onto the tape—these all cost a lot of money. The technical complexity of recorded music production is magnitudes cheaper now. But for the cognitively complex tasks that traditional producers, arrangers, A&R men and women, and engineers brought to clarify and to enhance artistry in composition and performance the picture is different. It still takes a lot of practice to be good at complex tasks. As Malcolm Gladwell points out, "Talent is important, but achievement is talent plus preparation." In cognitively demanding fields, there are no naturals, and the making of a quality recording is a process demanding cognitive skills to process musical ideas and then comparing them objectively to retained experience. These areas of cognition/thinking move from and between levels of complexity simultaneously, and seemingly without reason or even awareness–these are tasks for the "right brain." Unfortunately, along with the disruption of the traditional music industry came the conclusion that there's little importance in these cognitive skills, and the "10,000 hours" that might be required to refine them is time wasted.

Every week the antiquated record industry trumpets its sales figures and the even more ancient media industry repeats them. But despite the best attempts to discredit the new emerging industry, the supposedly "impossible" is happening all around us as many "unsigned" artists top the sales charts of the digital music stores and sell millions of units of music.

Never before in history has there been an opportunity as we now have before us. And, as Steve Jobs demonstrated, people will pay if you give them a high-quality offering. "Good enough" is no longer good enough. The job is to transform ourselves. Making music requires nimbleness, out-of-the-box thinking, resourcefulness, risk-taking, courage, and skill. And always taking a new approach. Take Neil Young. Now, despite criticism from some in the professional audio sector, he's proposing a music download-service, player, and audio format, whose aims are "to confront the compressed audio inferiority that MP3s offer," "to present songs as they first sound during studio recording sessions," Dylan says it's clear why Neil Young has not tumbled into musical dotage: "An artist like Neil always has the upper hand," he says. "It's the pop world that has to make adjustments. All the conventions of the pop world are only temporary and carry no weight. It's basically two different things that have nothing to do with each other."

There's no going back. There's no road map for those hoping to understand possible future roles in music as a profession–it's still evolving. But there is hope. Music is a part of all cultures around the world. It takes on different forms and is constantly changing–developing in new directions. These fundamental facts are the best proof of the importance of music to mankind. Never before in history has there been an opportunity as we now have before us.

Session P8 **Friday, Oct. 18**
9:00 am – 12:00 noon **Room 1E07**

RECORDING AND PRODUCTION

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

9:00 am

P8-1 Music Consumption Behavior of Generation Y and the Reinvention of the Recording Industry—*Barry Marshall*, The New England Institute of Art, Brookline, MA, USA

This paper will give an overview of the last 15 years of the recording industry's problems with piracy and decreasing sales, while reporting on research into the music consumption behavior of a group of audio students in both the United States and in eight European countries. Audio students have a unique perspective on the issues surrounding the recording industry's problems since the advent of Napster and the later generations of illegal file sharing. Their insights into issues like the importance of access to music, the quality of the listening experience, and the ethical quandary of participating in copyright infringement, may help point to a direction for the future of the recording industry.
Convention Paper 8956

9:30 am

P8-2 (Re)releasing the Beatles—*Brett Barry*, Syracuse University, Syracuse, NY, USA

This paper presents a comparative analysis of various Beatles releases, including original 1960s vinyl, early compact discs, and present-day digital downloads through services like iTunes. Original research will be providedusing source material and interviews with persons directly involved in recording and releasing Beatles albums, examining variations in dynamic range, spectral distribution, psychoacoustics, and track anomalies. Considerations are given to mastering and remastering a catalog of classics.
Convention Paper 8957

10:00 am

P8-3 Maximum Averaged and Peak Levels of Vocal Sound Pressure—*Braxton Boren, Agnieszka Roginska, Brian Gill*, New York University, New York, NY, USA

This work describes research on the maximum sound pressure level achievable by the spoken and sung human voice. Trained actors and singers were measured for peak and averaged SPLs at an on-axis distance of 1 m at three different subjective dynamic levels and also for two different vocal techniques ("back" and "mask" voices). The "back" sung voice was found to achieve a consistently lower SPL than the "mask" voice at a corresponding dynamic level. Some singers were able to achieve high averaged levels with both spoken and sung voice, while others produced much higher levels singing than speaking. A few of the vocalists were able to produce averaged levels above 90 dB_A, the highest found in the existing literature.
Convention Paper 8958

10:30 am

P8-4 Listener Adaptation in the Control Room: The Effect of Varying Acoustics on Listener Familiarization—*Richard King,*[1,2] *Brett Leonard,*[1,2] *Scott Levine,*[1] *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

The area of auditory adaptation is of central importance to a recording engineer operating in unfamiliar or less-than-ideal acoustic conditions. This study prompts expert listeners to perform a controlled level-balancing task while exposed to three different acoustic conditions. The length of exposure is varied to test the role of adaptation on such a task. Results show that there is a significant difference in the variance of participants' results when exposed to one condition for a longer period of time. In particular, subjects seem to most easily adapt to reflective acoustic conditions.
Convention Paper 8959

11:00 am

P8-5 Spectral Characteristics of Popular Commercial Recordings 1950-2010—*Pedro Duarte Pestana,*[1,2] *Zheng Ma,*[3] *Joshua D. Reiss,*[3] *Alvaro Barbosa,*[1] *Dawn A. A. Black*[3]
[1]Catholic University of Oporto - CITAR, Oporto, Portugal
[2]Lusíada Universityof Portugal (ILID); Centro de Estatística e Aplicacoes
[3]Queen Mary University of London, London, UK

In this work the long-term spectral contours of a large dataset of popular commercial recordings were analyzed. The aim was to analyze overall trends, as well as yearly and genre-specific ones. A novel method for averaging spectral distributions is proposed that yields results that are prone to comparison. With it, we found out that there is a consistent leaning toward a target equalization curve that stems from practices in the music industry but also to some extent mimics natural, acoustic spectra of ensembles.
Convention Paper 8960

11:30 am

P8-6 A Knowledge-Engineered Autonomous Mixing System—*Brecht De Man, Joshua D. Reiss*, Queen Mary University of London, London, UK

In this paper a knowledge-engineered mixing engine is introduced that uses semantic mixing rules and bases mixing decisions on instrument tags as well as elementary, low-level signal features. Mixing rules are derived from practical mixing engineering textbooks. The performance of the system is compared to existing automatic mixing tools as well as human engineers by means of a listening test, and future directions are established.
Convention Paper 8961

Session P9 Friday, Oct. 18
9:00 am – 11:30 am Room 1E09

APPLICATIONS IN AUDIO—PART 1

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

9:00 am

P9-1 Audio Device Representation, Control, and Monitoring Using SNMP—*Andrew Eales*,[1,2] *Richard Foss*[2]
[1]Wellington Institute of Technology, Wellington, New Zealand
[2]Rhodes University, Grahamstown, South Africa

The Simple Network Management Protocol (SNMP) is widely used to configure and monitor networked devices. The architecture of complex audio devices can be elegantly represented using SNMP tables. Carefully considered table indexing schemes support a logical device model that can be accessed using standard SNMP commands. This paper examines the use of SNMP tables to represent the architecture of audio devices. A representational scheme that uses table indexes to provide direct-access to context-sensitive SNMP data objects is presented. The monitoring of parameter values and the implementation of connection management using SNMP are also discussed.
Convention Paper 8962

9:30 am

P9-2 IP Audio in the Real-World; Pitfalls and Practical Solutions Encountered and Implemented when Rolling Out the Redundant Streaming Approach to IP Audio—*Kevin Campbell*, WorldCast Systems /APT, Belfast, N Ireland; Miami, Florida

This paper will review the development of IP audio links for audio delivery and chiefly look at the possibility of harnessing the flexibility and cost-effectiveness of the public internet for professional audio delivery. We will discuss first the benefits of IP audio when measured against traditional synchronous audio delivery and also the typical problems associated with delivering real-time broadcast audio across packetized networks, specifically in the context of unmanaged IP networks. The paper contains an examination of some techniques employed to overcome these issues with an in-depth look at the redundant packet streaming approach.
Convention Paper 8963

10:00 am

P9-3 Implementation of AES-64 Connection Management for Ethernet Audio/Video Bridging Devices—*James Dibley, Richard Foss*, Rhodes University, Grahamstown, Eastern Cape, South Africa

AES-64 is a standard for the discovery, enumeration, connection management, and control of multimedia network devices. This paper describes the implementation of an AES-64 protocol stack and control application on devices that support the IEEE Ethernet Audio/Video Bridging standards for streaming multimedia, enabling connection management of network audio streams.
Convention Paper 8964

10:30 am

P9-4 Simultaneous Acquisition of a Massive Number of Audio Channels through Optical Means—*Gabriel Pablo Nava, Yutaka Kamamoto, Takashi G. Sato, Yoshifumi Shiraki, Noboru Harada, Takehiro Moriya,* NTT Communicatin Science Labs, Atsugi-shi, Kanagawa-ken, Japan

Sensing sound fields at multiple locations often may become considerably time consuming and expensive when large wired sensor arrays are involved. Although several techniques have been developed to reduce the number of necessary sensors, less work has been reported on efficient techniques to acquire the data from all the sensors. This paper introduces an optical system, based on the concept of *visible light communication*, which allows the simultaneous acquisition of audio signals from a massive number of channels via arrays of light emitting diodes (LEDs) and a high speed camera. Similar approaches use LEDs to express the sound pressure of steady state fields as a scaled luminous intensity. The proposed sensor units, in contrast, transmit optically the actual digital audio signal sampled by the microphone in real time. Experiments to illustrate two examples of typical applications are presented: a remote acoustic imaging sensor array and a spot beamforming based on the *compressive sampling* theory. Implementation issues are also addressed to discuss the potential scalability of the system.
Convention Paper 8965

11:00 am

P9-5 Blind Microphone Analysis and Stable Tone Phase Analysis for Audio Tampering Detection—*Luca Cuccovillo*,[1] *Sebastian Mann*,[1] *Patrick Aichroth*,[1] *Marco Tagliasacchi*,[2] *Christian Dittmar*[1]
[1]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[2]Politecnico di Milano, Milan, Italy

In this paper we present an audio tampering detection method based on the combination of blind microphone analysis and phase analysis of stable tones, e.g., the electrical network frequency

(ENF). The proposed algorithm uses phase analysis to detect segments that might have been tampered. Afterwards, the segments are further analyzed using a feature vector able to discriminate among different microphone types. Using this combined approach, it is possible to achieve a significantly lower false-positive rate and higher reliability as compared to standalone phase analysis.
Convention Paper 8966

Workshop 10 Friday, Oct. 18
9:00 am – 10:30 am Room 1E10

NATIONAL RECORDING PRESERVATION PLAN: BEST PRACTICES FOR CREATING AND PRESERVING BORN-DIGITAL AUDIO FILES

Chair: **Konrad Strauss**, Indiana University, Bloomington, IN, USA

Panelists: *Chris Lacinak*, AVPreserve, New York, NY, USA
George Massenburg, Schulich School of Music, McGill University
Charles Van Winkle, Adobe

In December of 2012 the Library of Congress released the National Recording Preservation Plan. The result of nearly 10 years of work by the Library and the National Recording Preservation Board, the Plan outlines a series of recommendations for implementing a national recorded sound preservation plan. This workshop will explore recommendations 2.7: Best Practices for Creating and Preserving Born-Digital Audio Files, and 2.6: Tools to Support Preservation throughout the Content Life Cycle; and will focus on best practices for the creation of born-digital recordings and strategies for short-term backup and long-term preservation.

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

Tutorial 6 Friday, Oct. 18
9:00 am – 10:30 am Room 1E13

FXPERTISE: EQUALIZATION

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

Equalization might be the most intuitive of effects. We've had tone controls since we were kids, after all. Advanced applications of equalization are born from a deep understanding of EQ parameters and technologies, plus broad knowledge of the spectral properties and signatures of the most common pop and rock instruments. In this tutorial, Alex Case shares his approach for applying EQ and strategies for its use: fixing frequency problems, fitting the spectral pieces together, enhancing flattering features, and more.

Broadcast/Streaming Media Session 6
Friday, Oct. 18 9:00 am – 10:30 am
Room 1E08

AUDIO FOR MOBILE TELEVISION

Moderator: **Joe Giardina**

Panelists: *J. Todd Baker*, DTS
Tim Carroll, Linear Acoustic Inc., Lancaster, PA, USA
Greg Ogonowski, Orban, San Leandro, CA, USA
Robert Reams, Psyx Research
Geir Skaaden, DTS, Inc.
Jim Starzynski, NBC Universal, New York, NY, USA

A panel discussion highlighting the various challenges facing Mobile TV audio transmissions. Focus will be on dialog intelligibility, signal routing and issues and applications unique to Mobile TV audio broadcasts.

Games Audio Session 3 Friday, Oct. 18
9:00 am – 11:00 am Room 1E11

SCORING "TOMB RAIDER": THE MUSIC OF THE GAME

Presenter: **Alex Wilmer**, Crystal Dynamics

"Tomb Raider's" score has been critically acclaimed as being uniquely immersive and at a level of quality on par with film. It is a truly scored experience that has raised the bar for the industry. To achieve this, new techniques in almost every part of the music's production needed to be developed. This event will focus on the process of scoring "Tomb Raider." Every aspect will be covered from the music's creative direction, composition, implementation, and the technology behind it.

Live Sound Seminar 4 Friday, Oct. 18
9:00 am – 11:00 am Room 1E12

DESIGNING FOR BROADWAY THEATER

Chair: **Tom Morse**, Morse Sound Design, New York, NY, USA

Panelists: *Kai Harada*, Harada Sound Design, New York, NY, USA
Abe Jacob
Peter Fitzgerald
Joshua Reid

We will focus first on a brief history of how sound on Broadway began, then on what makes Broadway Sound Design unique in the audio industry. That will include what is required of a sound designer on Broadway, working with the producer, director, other designers, and what paperwork is needed in order to bid, build, and install the show. A Broadway theater is a four-wall rental meaning the production is placed in an empty shell. All equipment is brought in and installed for what may be a week or could turn into 15 years. This requires a great deal of planning and forethought because the show evolves and can change drastically from first rehearsal (when the paperwork is already over due) to opening night.

Student/Career Development Event
STUDENT DESIGN COMPETITION
Friday, October 18, 9:00 am – 10:30 am
Room 1E14

Moderator: **Colin Pfund**, University of Hartford, West Hartford, CT, USA

The three graduate level and three undergraduate level finalists of the AES Student Design Competition will present and defend their designs in front of a panel of expert ➥

judges. This is an opportunity for aspiring student hardware and software engineers to have their projects reviewed by the best minds in the business. 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.

Judges are: *Dave Amels*, AnaMod Analog Modeling/Bomb Factory Digital; *Bill Whitlock*, Jensen Transformers; *Dave Hill*, Crane Song/Dave Hill Designs; *Scott Dorsey*, Stinger-Ghaffarian Technologies (formerly at Raytheon Technical Services).

Friday, Oct. 18 9:00 am Room 1E04
Technical Committee Meeting: Audio Forensics

Knowledge Center
Friday, October 18, 10:00 am – 11:30 am
Booth 2738

LECTROSONICS SHOWCASE WITH KARL WINKLER

Presenter: **Karl Winkler**, Lectrosonics, Rio Rancho, NM, USA

Karl Winkler, Director of Business Development at Lectrosonics shows off all the latest and greatest from that company.

Knowledge Center
Friday, October 18, 10:00 am – 11:00 am
Room 1E06

PMC "MASTERS OF AUDIO":
"BASSY" BOB BROCKMAN

Bassy Bob (P. Diddy/Santana/Christina Aquilera/Sting/the Fugees) gives an insight into his 3-D mixing approach using a combination of EQ, compression, and placement.

Workshop 11 Friday, Oct. 18
10:30 am – 12:30 pm Room 1E08

AUDIO SOURCE SEPARATION

Chair: **Gautham Mysore**, Adobe Research, San Francisco, CA, USA

Panelists: *Nicholas Bryan*, Stanford University, Stanford, CA, USA
Derry FitzGerald, Cork Institute of Technology, Cork, Ireland
Elias Kokkinis, Accusonus
Pierre Leveau, Audionamix, Paris, France

Audio source separation algorithms aim to take a recording of a mixture of sound sources as an input and provide the separated sources as outputs. Algorithmically, this is a very challenging problem. However, some recent technological advances have made this possible for multiple real world scenarios such as denoising in the presence of complex noises, pitch correcting certain notes while preserving others, processing only the vocals of a song while preserving the background music, extracting dialogue from old films to provide a higher quality soundtrack, removing microphone leakage from multichannel drum recordings, upmixing mono to stereo with panning of sound sources, and more generally, music remixing. Some of these technologies are available in products (Adobe Audition CC, Melodyne, ISSE, ADX Trax, Drumatom). Others are used by specialized sound engineers and are offered as a service (Audionamix, Derry Fitzgerald). This panel is comprised of some of the inventors of these technologies, who will discuss the ideas and their practical use.

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

Workshop 12 Friday, Oct. 18
10:30 am – 12:00 noon Room 1E13

FX DESIGN PANEL: EQUALIZATION

Chair: **Francis Rumsey**, Logophon Ltd., Oxfordshire, UK

Panelists: *Nir Averbuch*, Sound Radix Ltd., Israel
George Massenburg, Schulich School of Music, McGill University, Montreal, Quebec, Canada
Saul Walker, New York University, NY, USA

Meet the designers whose talents and philosophies are reflected in the products they create, driving sound quality, ease of use, reliability, price, and all the other attributes that motivate us to patch, click, and tweak their effects processors.

Tutorial 7 Friday, Oct. 18
10:30 am – 12:00 noon Room 1E14

AUTOMATIC SPEAKER RECOGNITION, VOICE BIOMETRICS, AND FORENSIC SPEAKER COMPARISON: PUTTING IT ALL TOGETHER

Presenters: **Catalin Grigoras**, University of Colorado at Denver, Denver, CO, USA
Jeff Smith, University of Colorado Denver, Denver, CO, USA

Automatic speaker recognition systems allow for an assessment of the similarities between two voices given a population of speakers in order to inform investigations or to grant biometric access to secure material. However, no internationally accepted standard for voice comparison procedure is in place for its use in forensics.

This tutorial will present the latest developments in automatic speaker recognition and considerations for its application in forensic speaker comparison. A methodology made up of two parts will be presented that relies on both automatic speaker recognition and semi-automatic vowel formant analysis. This two-pronged method allows the analyst to cross-validate the assessment of a speaker's identity as a likelihood ratio. Additional discussion will include real cases from the past and very recent events.

Tutorial for all levels from those interested in an introduction to those with advanced knowledge in speech signal processing.

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

Student/Career Development Event
STUDENT DESIGN EXHIBITION
Friday, October 18, 10:30 am – 12:00 noon
1E Foyer

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

Tutorial 8 **Friday, Oct. 18**
10:45 am – 11:45 am **Room 1E10**

NATIONAL RECORDING PRESERVATION PLAN: AUDIO SYSTEM PERFORMANCE TESTING

Presenters: **Chris Lacinak**, AVPreserve, New York, NY, USA
Ian Dennis, Prism Sound
Rob Friederich, Library of Congress

Recommendation 2.4 from the Library of Congress National Recording Preservation Plan published this year states that we need improved tools and metrics for system performance testing. This session will discuss work coming out of the Federal Agencies Digitization Guidelines Initiative on performance testing of audio digitization systems tasked with preservation and archiving. Specifically this session will discuss the issues of Interstitial Errors, a term used to describe momentary artifacts caused by failure in a DAW's writing of data to a storage medium, and Analog-to-Digital Converter performance testing. The latter of these was just accepted as a formal standards project by SC-02-01 at the last AES and addresses both development of a test method and performance criteria for converters used in preservation systems.

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

Live Sound Seminar 5 **Friday, Oct. 18**
11:00 am – 1:00 pm **Room 1E12**

DEALING WITH NOISE POLLUTION IN THEATERS

Chair: **Tom Clark**, Acme Design

Panelists: *Damian Doria*
Scott Lehrer
Tom Morse

As video projection, moving lights, and automated scenery have become common in Broadway productions, the noise they each create has become a problem for the sound design team to overcome without losing the subtle use of sound reinforcement and sound effects. This is a discussion among designers about ways to deal with this increasing problem.

Special Event
FRIDAY KEYNOTE:
THE CURRENT AND FUTURE DIRECTION OF THE RECORDING PROCESS FROM AN ARTIST, ENGINEER, AND PRODUCER'S PERSPECTIVE
Friday, October 18, 11:00 am – 12:00 noon
Room 1E15/16

Presenter: **Jimmy Jam**

Five-time GRAMMY Award winner Jimmy Jam is a renowned songwriter, record producer, musician, entrepreneur, and half of the most influential and successful writing/producing duo in modern music history. Since forming their company Flyte Tyme Productions in 1982, Jam and partner Terry Lewis have collaborated with such diverse and legendary artists as Janet Jackson, Mary J. Blige, Gwen Stefani, Robert Palmer, Mariah Carey, Boyz II Men, Rod Stewart, Yolanda Adams, Sting, Heather Headley, Usher, Celine Dion, Kanye West, Chaka Khan, Trey Songz, and Michael Jackson, among others. Jimmy and Terry have written and/or produced over 100 albums and singles that have reached gold, platinum, multi-platinum, or diamond status, including 26 No. 1 R&B and 16 No. 1 pop hits, giving the pair more Billboard No. 1's than any other duo in chart history. Jimmy Jam's Lunchtime Keynote address will focus on the current and future direction of the recording process from various perspectives. As a songwriter, artist, engineer, and producer, Jimmy is uniquely qualified to give a bird's-eye view of how each of these "personalities" interact and contribute to the overall final product, and along the way, how technology has evolved and what it has meant to his craft. In Jimmy's words, "Of course it all starts with a great song, but then, it's important to consider how and what technology should be used to capture that creativity. It's that intersection between the technology and creativity that I have always looked at every day throughout my career. Ultimately, it's my job as a artist/producer to have those two elements meet and not crash. And that's when you're using the available technology to capture the artist in their purest form."

Project Studio Expo
Friday, October 18, 11:00 am – 12:00 pm
Stage

IT WON'T SOUND RIGHT IF YOU DON'T HEAR IT RIGHT: STUDIO ACOUSTICS, MONITORING & CRITICAL LISTENING

Presenters: **Hugh Robjohns**, Technical Editor, Sound on Sound, Cambridge, UK
Paul White

The monitoring environment acoustics and the monitoring loudspeakers are critical links in every music production chain. Any weaknesses impact negatively not only on the overall quality of mixes but also on the confidence and ability of the user to assess and process audio material efficiently and effectively. This workshop examines the theoretical requirements and practical optimization of high-quality monitoring systems for home and project studios, drawing on the author's experiences in the "Studio SOS" series published in Sound On Sound magazine. It will also explore choosing new monitoring loudspeakers, optimizing control room acoustics, and honing critical listening skills.

Knowledge Center
Friday, October 18, 11:00 am – 12:00 noon
Room 1E06

PMC "MASTERS OF AUDIO": A JOE FERLA RETROSPECTIVE WITH SPARS

Presenter: **Joe Ferla**, Joe Ferla, Stamford, NY, USA

Joe Ferla, a five-time Grammy Award recipient and renowned engineer to some of the best musicians in the ➥

industry, will be presenting tracks for listening from his extensive discography. After each track is played he will answer any questions regarding the track such as how he recorded the artist, which mics he used on what instrument, etc. Songs from artists such as Roberta Flack (that started his career off), David Sanborn, John Scofield, Eliane Elias, Dave Douglas, Christian McBride, Charlie Hunter, and others will be featured in this unique listening experience.

Knowledge Center
Friday, October 18, 11:00 am – 12:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Knowledge Center
Friday, October 18, 11:00 am – 12:30 pm
Room 1E03

IZOTOPE PRESENTS: TIPS FROM A PRO: LIVE! AUDIO WOES IN MUSIC AND POST PRODUCTION

Presenters: **Jared Bartlett**
Chris Shaw
Jonathan Wyner, Berklee College of Music, Boston, MA, USA; M Works Mastering

In a discussion lead by Jonathan Wyner (Mastering Engineer, M-Works Mastering), Chris Shaw (Engineer and Producer: Bob Dylan, Leonard Cohen), and Jared Bartlett (Post Production Mixer, Clean Cuts) share their stories of audio gone wrong and how they saved the take. Hear about their audio production processes and see how their perspectives, from an array of audio job titles, can be applied to your audio workflow.

Friday, Oct. 18 11:00 am Room 1E04
Technical Committee Meeting: Human Factors in Audio Systems

Friday, Oct. 18 11:00 am Room 1E02
Standards Committee Meeting: SC-04-08 Working Group on Measurement and Equalization of Sound Systems in Rooms

Sound for Picture 1 Friday, Oct. 18
11:30 am – 1:00 pm Room 1E11

CREATIVE DIMENSION OF IMMERSIVE SOUND —SOUND IN 3-D

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

Panelists: *Marti Humphrey CAS*, The Dub Stage, Burbank, CA, USA
Branko Neskov, Loudness Films, Lisbon, Portugal

Audio for Cinema has always struggled to replicate the motion shown on the screen, a fact that became more apparent with 3-D films. Several methodologies for "immersive sound" are currently under evaluation by the industry, with theater owners and film companies both advising that they will not tolerate a format war, with a common format a commercial requirement.

The two major methods of creating immersive sound and audio motion are referred to as "object-based" and "channel-based." Each has its strengths and limitations for retrofit into the current cinema market. With few sound mixers experienced in either of these techniques, we're pleased to welcome two of the pioneers, one with experience at Auro3D and the other in Atmos, in a discussion of their experiences and comments on working with the two systems.

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

Knowledge Center
Friday, October 18, 11:30 am – 1:00 pm
Booth 2738

RADIO ACTIVE DESIGNS: ASK AN RF EXPERT!

Presenter: **James Stoffo**, Radio Active Designs, Key West, FL, USA

Now's your chance to ask frequency coordinator par excellance James Stoffo anything you want about wireless frequencies, dealing with the white space crunch, your wireless setup or Radio Active Designs's suite of products! He'll be at the booth during the indicated times, or you can tweet to @gothamsound #RADanswers all weekend!

If you've turned on a TV in the last 30 years, you've probably heard James Stoffo's work. His credits include 14 consecutive Superbowls, the Olympics, the World Cup, multiple NBA All Star Games, NBA Finals, and Rose Bowl games. James is also one of the principals of Radio Active Designs, a new manufacturer of innovative wireless communication devices meant to alleviate spectral congestion while providing excellent audio quality, reliability, and RF performance.

Tutorial 9 Friday, Oct. 18
11:45 am – 1:15 pm Room 1E10

NATIONAL RECORDING PRESERVATION PLAN: PRESERVATION PLANNING

Presenters: **Chris Lacinak**, AVPreserve, New York, NY, USA
David Ackerman, Harvard University, Cambridge, MA, USA
Matthew Barton, Library of Congress
Mark Hood, Indiana University, Bloomington, IN, USA

Recommendation 2.2 from the Library of Congress National Recording Preservation Plan published this year states the need to devise means to assist organizations in conducting comprehensive appraisals of audio collections with the goal of establishing priorities for preservation. This session will discuss recent developments toward these ends, focusing on efforts coming out of Indiana University, Library of Congress, AudioVisual Preservation Solutions, and Harvard University. Software, including open-source freely available applications, and associated workflows will be demonstrated and discussed. Case studies will be used to provide concrete examples of project logistics, details and outcomes, providing tangible, practical information to organizations in need of obtaining the information necessary to perform prioritization, planning, budgeting, and more around the preservation and access of their audio collections.

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

Game Audio Session 4 — Friday, Oct. 18
11:45 am – 12:45 am — Room 1E

LOUDNESS IN INTERACTIVE SOUND ROUNDUP

Presenter: **Garry Taylor**, Sony Computer Entertainment Europe, Cambridge, Cambridgeshire, UK

Over the years there has been much talk of reigning in the loudness problem in the games industry. It's not just talk anymore. Listen to those who has made progress in this field and learn how to apply their efforts to your title. Recently, Sony's Audio Standards Working Group (ASWG) released loudness recommendations for their first party titles. Garry Taylor, Audio Director at Sony Computer Entertainment, looks at the work of the ASWG, the data they collected, and how that data influenced their recommendations. He looks at their first loudness paper and how their titles are measured and tested at Quality Assurance.

Session EB2 — Friday, Oct. 18
12:00 noon – 12:45 pm — Room 1E07

RECORDING AND PRODUCTION

Chair: **Scott Levine**, Skywalker Sound, San Francisco, CA, USA; The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

12:00 noon

EB2-1 Controlling Drum Bleed with Laser Vibrometry—*Andrew Greenwood*, *Sebastian Chafe*, Sennheiser Technology and Innovation, San Francisco, CA, USA

Using multiple microphones to capture the sound of multiple drums on a drum kit is common practice. As well, the bleed captured by such a setup is a common problem for sound engineers. Gating is often used in an attempt to manage drum and instrument bleed into individual drum channels. However, the overlap in amplitude and frequency content of different drums makes gating based solely on the microphone audio difficult and unwanted triggering of the gate is a common problem. By measuring the physical vibration of the drum head using simple laser vibrometry and using this signal to run the sidechain of a gate, the dynamic range of the gate's signal follower is increased and false triggering is easier to avoid. This allows for more precise control over each drum channel's tone and dynamics
Engineering Brief 109

12:15 pm

EB2-2 The Urban Mix Engineer—*Paul "Willie Green" Womack*, Willie Green Music, Brooklyn, NY, USA

Although hip-hop is over 40 years old and influences the sound of everyone from Stevie Wonder to Taylor Swift, the term "hip-hop engineer" often evokes visions of kids with cracked software and distorted records. This presentation explores arguments often referenced in this ongoing debate of skill, including mixing "real vs. synthetic" instruments, the loop based nature of hip-hop, and mixing samples. It also illustrates a time when engineers in other genres once overcame similar criticisms. Providing an honest, first-hand look at what's involved in mixing urban records, and the hurdles that exist in and out of the studio, this presentation sheds light on the importance of the mixing engineer, in any genre, as a vital part of the arrangement process.
Engineering Brief 110

12:30 pm

EB2-3 Subjective Comparison of Surround Microphone Recording Techniques Presented With and Without Video—*Luiz Fernando Kruszielski*, Globo TV Network, Rio de Janeiro, Brazil; Tokyo University of the Arts, Tokyo, Japan

The comparison of different setups of microphones for surround recordings of music is a topic that has a large interest in the audio community. Although it is known that image has a strong effect on sound perception, particularly in spatial aspects, very little research had been done aiming the surround sound recordings with accompanied video. To compare possible influences in this perception, a test was created using five different surround recording techniques that was done simultaneously at a Brazilian carnival parade. The subjects were presented to the sound with and without video and asked to rate four different aspects: localization, deepness, immersion, and preference. The results show that there is a difference in perception depending on the presence or absence of video.
Engineering Brief 111

Workshop 13 — Friday, Oct. 18
12:00 noon – 1:00 pm — Room 1E13

MICROPHONE AND RECORDING TECHNIQUES FOR THE MUSIC ENSEMBLES OF THE UNITED STATES MILITARY ACADEMY

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

Panelist: *Joseph Skinner*, United States Army, West Point Band, West Point, NY, USA

The United States Military Academy is home to the oldest active duty military band. Our mission is to provide ➥

world-class music to educate, train, and inspire the Corps of Cadets and to serve as ambassadors of the United States Military Academy to the local, national, and international communities. This workshop will discuss advanced microphone and recording techniques (stereo and multi-track) used to capture the different elements of the West Point Band including the Concert Band, Jazz Knights, and Field Music group in a recording/studio or live sound reinforcement setting. The recording of other USMA musical elements including the Cadet Glee Club and Cadet Pipes and Drums will also be discussed.

Project Studio Expo
Friday, October 18, 12:00 noon – 1:00 pm
Stage

TOTAL TRACKING: GET IT RIGHT AT SOURCE

Presenters: **Bill Gibson**, Hal Leonard Performing Arts Publishing Group - Seattle, WA, USA; Berklee College of Music Online
Paul White

Everyone wants to record and produce music that's successful! Although the secret to releasing a hit is multifold, the entire creative and technical process is doomed without two indispensable assets: well-recorded tracks and finely crafted musical components. This session focuses closely on methods for miking, processing, and capturing excellent sounds that require little or no manipulation during mixdown. Covered at the same time are considerations and techniques for establishing a solid musical foundation, resulting in a recording that nearly mixes itself because the mix ingredients are crafted to compliment each other while building power through intelligent combinations. Record better music more quickly and easily by getting it right at the source!

Friday, Oct. 18 12:00 noon Room 1E04
Technical Committee Meeting: Recording Technology and Practices

Broadcast/Streaming Media Session 7
Friday, Oct. 18 12:30 pm – 2:00 pm
Room 1E08

BROADCASTING DURING A DISASTER

Chair: **Glynn Walden**, CBS Radio, Philadelphia, PA, USA

Panelists: *Rob Bertrand*, CBS
Howard Price, ABC/Disney
Tom Ray, Tom Ray Broadcasting Consulting
Richard Ross, WADO/Univision

No power!
Water rising!
Roads out!
No phones!

Broadcasters have always been "First Informers." Not only do they convey information to the audience, the station has to maintain the broadcast throughout the disaster. Superstorm Sandy challenged many broadcasters. This panel will discuss how they dealt with non-ideal situations to keep the broadcast on during the storm, the aftermath, and the recovery.

Knowledge Center
Friday, October 18, 12:30 pm – 2:00 pm
Room 1E06

PMC "MASTERS OF AUDIO": JIMMY DOUGLASS

Jimmy Douglass (also known as The Senator) is an American four-time Grammy winning recording engineer and record producer, whose prolific career has spanned more than four decades. JD will present some of his latest projects like Justin Timberlake's 20/20 Experience, as well as some of his favorite past time productions. JD will take ample time to answer your questions!!

Special Event
FROM THE MOTOR CITY TO BROADWAY: MAKING "MOTOWN THE MUSICAL" CAST ALBUM
Friday, October 18, 12:45 pm – 2:15 pm
Room 1E15/16

Moderator: **Harry Weinger**, Universal Music Enterprises (UMe), New York, NY, USA

Panelists: *Frank Filipetti*
Jawan Jackson
Ethan Popp, Special Guest Music Productions, LLC, New York, NY, USA

Tracing the path taken by pop-R&B classics known the world over to the Broadway stage and the modern-day recording studio—and how cast albums get made with no time and no do-overs.

A panel and Q&A with album producer and mixer Frank Filipetti, a multi-Grammy Award winner, and co-producer Ethan Popp, the show's Tony-nominated musical supervisor.

Historical Program/Special Event
A TRIBUTE TO RAY DOLBY
Friday, October 18, 1:00 pm – 1:45 pm
Room 1E14

Presenter: **Ioan Allen**, Dolby Laboratories Inc., San Francisco, CA, USA

Ray Dolby died last month at the age of 80. In this special presentation, Ioan Allen will not only cover a few of the highlights of Ray's distinguished career but will also spend more time reminiscing on what it was like to work for over four decades with this iconic figure.

Project Studio Expo
Friday, October 18, 1:00 pm – 2:00 pm
Stage

MIXING SECRETS: PRODUCTION TRICKS TO USE WITH ANY DAW

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

Affordable DAW software now provides all the processing tools you need to create commercially competitive music mixes within a home, college, or project studio. As such, the overriding concern for budget-conscious engineers these days should be to develop effective habits with regard to studio monitoring, mix balancing, and quality control. Important techniques in each of these three areas are often neglected in small-scale productions, leading to mixes that don't stack up against professional releases, or that collapse on some mass-market listening systems. In this seminar Sound On Sound magazine's "Mix Rescue" columnist Mike Senior will draw on his experience of thousands of project-studio mixes to highlight the most frequently overlooked studio tricks. In the process he'll demonstrate how these methods can powerfully upgrade your sonics without breaking the bank, no matter which DAW you're using.

Knowledge Center
Friday, October 18, 1:00 pm – 2:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Friday, Oct. 18 1:00 pm Room 1E04
Technical Committee Meeting: Network Audio Systems

Friday, Oct. 18 1:00 pm Room 1E02
Standards Committee Meeting: SC-05-02 Working Group on Audio Connectors

Special Event
LUNCHTIME KEYNOTE:
ON THE TRANSMIGRATION OF SOULS
Friday, October 18, 1:15 pm – 2:15 pm
Room 1E11

Presenter: **Michael Bishop**

"On the Transmigration of Souls," is a multi-Grammy winning work for orchestra, chorus, children's choir, and prerecorded tape is a composition by composer John Adams. It was commissioned by the New York Philharmonic and Lincoln Center's Great Performers and Mr. Adams received the 2003 Pulitzer Prize in music for the piece. Its premiere recording received the 2005 Grammy Award for Best Classical Album, Best Orchestral Performance, and Best Classical Contemporary Composition and the 2009 Grammy Award for Best Surround Sound Album. Surround Recording Engineer, Michael Bishop will discuss the surround production process and play the work in its entirety.

Knowledge Center
Friday, October 18, 1:30 pm – 2:30 pm
Booth 2738

WISYCOM SHOWCASE WITH MASSIMO POLO

Presenter: **Massimo Polo**, Wisycom, Vicenza, Italy

Imagine a world without 25 MHz block restrictions. That's what the Wisycom family of products envisions, with high quality diversity wireless systems built to operate over an impressively wide bandwidth.

Session P10 Friday, Oct. 18
2:00 pm – 2:30 pm Room 1E07

AMPLIFIERS

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

[Convention Paper 8967 not presented]

2:00 pm

P10-2 Supply-Feedback Fully-Digital Class D Audio Amplifier Featuring 100 dBA+ SNR and 0.5 W to 1 W Selectable Output Power—*Rossella Bassoli, Federico Guanziroli, Carlo Crippa, Germano Nicollini***,** ST-Ericsson, Monza Brianza, Italy

This paper presents a real-time power supply noise correction technique in a fully-digital class D audio amplifier. The power supply is scaled and applied to a 12-bits Nyquist ADC to modify the amplitude of the Pulse-Width-Modulator reference carrier. An improved supply extrapolation algorithm results to a power supply rejection from one to two orders of magnitude higher than reported implementations. Class D sensitivity to clock jitter is presented. SNR higher than 100 dBA have been measured in the presence of both power supply ripple and clock jitter. The PWM and output stage are integrated in the same chip in a 0.13 µm digital CMOS technology, whereas an external ADC has been used to demonstrate the validity of the supply-feedback algorithm.
Convention Paper 8968

[Convention Paper 8969 not presented]

Workshop 14 Friday, Oct. 18
2:00 pm – 3:30 pm Room 1E14

THE TIMELINE NEVER LIES: AUDIO ENGINEERS AIDING FORENSIC INVESTIGATORS IN CASES OF SUSPECTED MUSIC PIRACY

Chair: **Martha de Francisco**, McGill University, Montreal, Quebec, Canada

Panelists: *Cynthia Arato*, Shapiro, Arato & Isserles, New York, NY, USA
Joe Bennett, Bath Spa University, UK
Thomas J. Owen, Owen Forensic Services

Technological evolution in the music world has lead to a heightened threat of illegal copying of music performances. Based upon their professional practice audio engineers are being consulted by repertoire owners and their lawyers in cases of suspected music piracy. The expert witnesses develop systematic approaches to obtaining proof of authenticity of the music material based on their professional expertise in analytical listening of the recorded sound and in studio production tools and techniques as well as on their ability to evaluate minute details of the music performance. Their findings may lead the courts to uncover dubious practices of falsification and subsequent release of copyrighted music files. ➥

A musicologist, a recording engineer/producer, a forensic audio specialist, and a litigation attorney present and discuss aspects of forensic audio and intellectual property.

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

Game Audio Session 5 **Friday, Oct. 18**
2:00 pm – 4:00 pm **Room 1E10**

GAME AUDIO: A PRIMER AND EDUCATIONAL RESOURCES

Moderator: **Stephen Harwood Jr.**, Education Working Group Chair, IASIG, New York, NY, USA

Presenters: *Andrew Aversa*, Drexel University
Leonard J. Paul, School of Video Game Audio
Jean-Luc Sinclair, NYU, Steinhardt School of Music Technology, New York, USA
Michael Sweet, Berklee College of Music

Game-curious? Interested in the video game industry but unsure of what exactly it is that we do here? Video game production values are improving rapidly, creating increased demand for top-notch, experienced audio professionals, but many composers, sound designers, and producers looking to bring their expertise from the world of film and TV into the video game industry are uncertain about what it is they'll be getting themselves into. Fortunately, the field of game audio education is developing rapidly—more schools are offering related courses each year. Following a presentation of the differences between audio production for games and for film and television, this session will feature a discussion of best practices and suggestions for how to learn what it takes to succeed as an audio professional working in the game space. Come prepared to inquire, be inspired, and take notes.

Project Studio Expo
Friday, October 18, 2:00 pm – 3:00 pm
Stage

MODULAR SYNTHESIZERS: CREATIVE USES IN THE STUDIO

Presenter: **Gino Robair**

Modular synthesizers continue to grow in popularity, but not just with instrumentalists. Increases in sound quality and product reliability over the last 15 years, along with a jump in processing power thanks to analog/digital hybridization in circuit designs, have resulted in an increased use of modular systems in a production capacity while tracking and mixing. In this workshop, Electronic Musician magazine's Technical Editor, Gino Robair will demonstrate how producers and engineers take advantage of voltage controllable hardware systems in the modern studio utilizing DAWs, MIDI, and specialized tools such as MOTU Volta and Expert Sleepers Silent Way.

Knowledge Center
Friday, October 18, 2:00 pm – 3:00 pm
Room 1E06

PMC "MASTERS OF AUDIO": RONALD PRENT/ DARCY PROPER, "THE LORI LIEBERMANN SURROUND SESSIONS"

Award winning engineers Ronald and Darcy will take you through the process of mixing and mastering the first ever surround recordings of singer/songwriter Lorie Lieberman's new album: *Bricks against the Glass*, recently released on the Pure Audio Blu Ray audio only format in high resolution. As the writer of the world-renowned song "Killing Me Softly," which has been made famous by Roberta Flack and more recently by the Fugees, the new album shows the immense songwriting talent and musicianship and includes the world's first ever 5.1 mix of "Killing Me Softly." The surround album was produced by PMC's own Maurice Patist.

Friday, Oct. 18 **2:00 pm** **Room 1E04**
Technical Committee Meeting: Coding of Audio Signals

Session P11 **Friday, Oct. 18**
2:15 pm – 4:45 pm **Room 1E09**

PERCEPTION—PART 1

Chair: **Jason Corey**, University of Michigan, Ann Arbor, MI, USA

2:15 pm

P11-1 On the Perceptual Advantage of Stereo Subwoofer Systems in Live Sound Reinforcement—*Adam J. Hill*,[1] *Malcolm O. J. Hawksford*[2]
[1]University of Derby, Derby, Derbyshire, UK
[2]University of Essex, Colchester, Essex, UK

Recent research into low-frequency sound-source localization confirms the lowest localizable frequency is a function of room dimensions, source/listener location, and reverberant characteristics of the space. Larger spaces therefore facilitate accurate low-frequency localization and should gain benefit from broadband multichannel live-sound reproduction compared to the current trend of deriving an auxiliary mono signal for the subwoofers. This study explores whether the monophonic approach is a significant limit to perceptual quality and if stereo subwoofer systems can create a superior soundscape. The investigation combines binaural measurements and a series of listening tests to compare mono and stereo subwoofer systems when used within a typical left/right configuration.
Convention Paper 8970

2:45 pm

P11-2 Auditory Adaptation to Loudspeakers and Listening Room Acoustics—*Cleopatra Pike, Tim Brookes, Russell Mason*, University of Surrey, Guildford, Surrey, UK

Timbral qualities of loudspeakers and rooms are often compared in listening tests involving short listening periods. Outside the laboratory, listening occurs over a longer time course. In a study by Olive et al. (1995) smaller timbral differences between loudspeakers and between rooms were reported when comparisons were made over shorter versus longer time periods. This is a form of timbral adaptation, a decrease in sensitivity to timbre over time. The current study confirms this adaptation and establishes that it is not due to response bias but may be due to timbral memory, specific mechanisms compensating for

transmission channel acoustics, or attentional factors. Modifications to listening tests may be required where tests need to be representative of listening outside of the laboratory.
Convention Paper 8971

3:15 pm

P11-3 Perception Testing: Spatial Acuity—*P. Nigel Brown*, Ex'pression College for Digital Arts, Emeryville, CA, USA

There is a lack of readily accessible data in the public domain detailing individual spatial aural acuity. Introducing new tests of aural perception, this document specifies testing methodologies and apparatus, with example test results and analyses. Tests are presented to measure the resolution of a subject's perception and their ability to localize a sound source. The basic tests are designed to measure minimum discernible change across a 180° horizontal soundfield. More complex tests are conducted over two or three axes for pantophonic or periphonic analysis. Example results are shown from tests including unilateral and bilateral hearing aid users and profoundly monaural subjects. Examples are provided of the applicability of the findings to sound art, healthcare, and other disciplines.
Convention Paper 8972

3:45 pm

P11-4 Evaluation of Loudness Meters Using Parameterization of Fader Movements—
Jon Allan, Jan Berg, Luleå University of Technology, Piteå, Sweden

The EBU recommendation R 128 regarding loudness normalization is now generally accepted and countries in Europe are adopting the new recommendation. There is now a need to know more about how and when to use the different meter modes, Momentary and Short term, proposed in R 128, as well as to understand how different implementations of R 128 in audio level meters affect the engineers' actions. A method is tentatively proposed for evaluating the performance of audio level meters in live broadcasts. The method was used to evaluate different meter implementations, three of them conforming to the recommendation from EBU, R 128. In an experiment, engineers adjusted audio levels in a simulated live broadcast show and the resulting fader movements were recorded. The movements were parameterized into "Fader movement," "Adjustment time," "Overshoot," etc. Results show that the proposed parameters produced significant differences caused by the meters and that the experience of the engineer operating the fader is a significant factor.
Convention Paper 8973

4:15 pm

P11-5 Validation of the Binaural Room Scanning Method for Cinema Audio Research—
Linda A. Gedemer,[1,2] *Todd Welti*[2]
[1]University of Salford, Salford, UK
[2]Harman International, Northridge, CA, USA

Binaural Room Scanning (BRS) is a method of capturing a binaural representation of a room using a dummy head with binaural microphones in the ears and later reproducing it over a pair of calibrated headphones. In this method multiple measurements are made at differing head angles that are stored separately as data files. A playback system employing headphones and a headtracker recreates the original environment for the listener, so that as they turn their head, the rendered audio during playback matches the listeners' current head angle. This paper reports the results of a validation test of a custom BRS system that was developed for research and evaluation of different loudspeakers and different listening spaces. To validate the performance of the BRS system, listening evaluations of different in-room equalizations of a 5.1 loudspeaker system were made both in situ and via the BRS system. This was repeated using three different loudspeaker systems in three different sized listening rooms.
Convention Paper 8974

Live Sound Seminar 6 — **Friday, Oct. 18**
2:30 pm – 4:30 pm — **Room 1E12**

WIRELESS MICROPHONES AND PERFORMERS: MIC PLACEMENT AND HANDLING FOR MULTIPLE ACTORS

Presenters: **Mary McGregor**, Freelance Local 1, NYC
Stephanie Vetter, Freelance Local 1, NYC

Fitting actors with wireless microphone elements and transmitters has become a detailed art form. From ensuring the actor is comfortable and the electronics are safe and secure, to getting the proper sound with minimal detrimental audio effects all while maintaining the visual illusion, one of the most widely recognized artisans in this field provide hands on demonstrations of basic technique along with some time tested "tricks of the trade."

Network Audio Session 3 — **Friday, Oct. 18**
2:30 pm – 3:30 pm — **Room 1E13**

THE ROLE OF STANDARDS IN AUDIO NETWORKING

Chair: **Mark Yonge**, AES Standards Secretary, UK

Panelists: *Jeff Berryman*, Bosch Communications, Ithaca, NY, USA
Kevin Gross, AVA Networks, Boulder, CO, USA
Andreas Hildebrand, ALC NetworX, Munich, Germany
Lee Minich, Lab X Technologies, Rochester, NY, USA

A number of standards organizations and industry associations have been active in promoting standards relating to audio networks, such as EBU, IEC, and not least AES with recent standards AES64, AES67, and project X-210. Networks themselves are standardized under the auspices of bodies such as the IEEE and IETF. This session will describe the landscape of standards bodies and their areas of interest in audio networking and will examine the questions:

- Are standards important?
- How does all this standard activity impact the real world of audio networks?
- How do these standards benefit the marketplace, end users and the technology suppliers to this market?

• Is development of and adherence to standards better for suppliers and end users than letting the manufacturers' proprietary solutions compete for market dominance?

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

Sound for Picture 2 Friday, Oct. 18
2:30 pm – 4:00 pm Room 1E11

CINEMA SOUND STANDARDS COLLAPSE LEAVING TURMOIL—AN OVERVIEW OF THE STATE OF THE ART

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

Panelists: *Glenn A. Leembruggen*, Acoustics Directions Pty Ltd., Summer Hill, NSW, Australia
David Murphy, Krix Loudspeakers, Australia

Dr. Floyd Toole first documented in his book *Sound Reproduction: Loudspeakers and Rooms* in 2008 the failure of the Standards process in producing quality sound in movie theaters. The work was expanded on in experiments done by a group led by Philip Newell in Europe, and this work was cited by Brian McCarty in order to get both the SMPTE and AES to begin work on scientific, comprehensive new Standards for cinema and eventually home audio reproduction.

This workshop reviews the flawed Standards and presents new experiments that further define the areas of work that will need to be undertaken for new Standards to be written.

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

Special Event
DTV AUDIO GROUP FORUM: AUDIO PRODUCTION AND DISTRIBUTION IN AN EVOLVING TELEVISION DELIVERY LANDSCAPE
Friday, October 18, 2:30 pm – 5:30 pm
Room 1E08

Presenters: **Tim Carroll**, Linear Acoustic Inc., Lancaster, PA, USA
Kevin Cleary, ESPN
Craig Cuttner, HBO
Michael Englehaupt, KQED
Stacey Foster, Saturday Night Live
Richard M. Friedel, Fox Networks Engineering & Op - Los Angeles, CA, USA
Ken Hahn, Sync Sound Inc. / Digital Cinema, LLC, New York, NY, USA
Bruce Jacobs, Twin Cities Public Television
Lawrence Manchester, Late Night with Jimmy Fallon, New York, NY, USA
Sean Richardson, Starz Entertainment, Englewood, CO, USA
Thomas Sahara, Turner Sports, Atlanta, GA, USA
Steven Silva, FOX Network Operations & Engineering, Los Angeles, CA, USA
Jim Starzynski, NBC Universal, New York, NY, USA

The forum is intended to explore the opportunities and challenges presented by advanced encoding schemes and to debate whether ubiquitous mobile and over-the-top content delivery demands a retrenchment to more limited audio or could lead to further audio advances. The discussion will also address the long-term implications of mobile data's inevitable annexation of available broadcast spectrum and the resulting impact on wireless production, and will once again revisit the challenges of producing multichannel music for television.

"The transition from traditional broadcasting to a largely stream-based model opens up a lot of possibilities but potentially adds to confusion as different entities pursue a range of formats and encoding solutions. The demand for more sophisticated interactive and object-oriented services on next-generation streaming appliances, and the transition to streaming of highly sophisticated cinema formats at the very high end, are directly at odds with the common perception that television audio now needs to be dumbed down for mobile and desktop streaming. This disconnect between competing visions creates a strategic dilemma for content producers who are looking for universal delivery standards and workflow practices across a range of delivery platforms." — Roger Charlesworth, Executive Director, DTV Audio Group

Discussion topics will include:

A Paler Shade of White: The impending disaster of shrinking white spaces.

Objects Are Closer than They Appear: Production and distribution workflow implications of object-oriented-audio.

How many Channels Is Your Cloud? Competing visions of television audio for mobile and over-the-top streaming.

Television Versus Music: Round Two: Revisiting the joys of multichannel music and the struggle for stereo compatibility.

Special Event
PLATINUM ENGINEERS
Friday, October 18, 2:30 pm – 4:30 pm
Room 1E15/16

Moderator: **Justin Colletti**, SonicScoop, Brooklyn, NY, USA; Trust Me, I'm a Scientist

Panelists: *Chris Coady*
Patrick Dillet
Tom Elmhirst
Manny Marroquin

Engineers of a particularly creative breed, these multifaceted audio gurus reflect a singular studio fluency that has inspired and produced some of today's most sonically expressive, adventurous, and influential recordings. Typically recording, mixing, and co-producing entire albums, these craftsmen often collaborate with artists whose distinct POVs come across not only in the songwriting and playing, but also in the sound of their records. Though they may program, play and/or produce on their projects, these panelists are engineers first, with the skillset to truly play the studio as an instrument. Participants will discuss the creative recording and mixing techniques they've developed, playing samples of their work to illustrate some of the most successful collaborations.

Friday, Oct. 18 2:30 pm Room 1E02
Standards Committee Meeting: SC-05-05 Working Group on Grounding and EMC Practices

Session P12 Friday, Oct. 18
3:00 pm – 4:30 pm 1E Foyer

POSTERS: SIGNAL PROCESSING

3:00 pm

P12-1 Temporal Synchronization for Audio

Watermarking Using Reference Patterns in the Time-Frequency Domain—*Tobias Bliem, Juliane Borsum, Giovanni Del Galdo, Stefan Krägeloh*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Temporal synchronization is an important part of any audio watermarking system that involves an analog audio signal transmission. We propose a synchronization method based on the insertion of two-dimensional reference patterns in the time-frequency domain. The synchronization patterns consist of a combination of orthogonal sequences and are continuously embedded along with the transmitted data, so that the information capacity of the watermark is not affected. We investigate the relation between synchronization robustness and payload robustness and show that the length of the synchronization pattern can be used to tune a trade-off between synchronization robustness and the probability of false positive watermark decodings. Interpreting the two-dimensional binary patterns as one-dimensional N-ary sequences, we derive a bond for the autocorrelation properties of these sequences to facilitate an exhaustive search for good patterns.
Convention Paper 8975

3:00 pm

P12-2 Sound Source Separation Using Interaural Intensity Difference in Real Environments—*Chan Jun Chun, Hong Kook Kim*, Gwangju Institute of Science and Tech (GIST), Gwangju, Korea

In this paper, a sound source separation method is proposed by using the interaural intensity difference (IID) of stereo audio signal recorded in real environments. First, in order to improve the channel separability, a minimum variance distortionless response (MVDR) beamformer is employed to increase the intensity difference between stereo channels. Then, IID between stereo channels processed by the beamformer is computed and applied to sound source separation. The performance of the proposed sound source separation method is evaluated on the stereo audio source separation evaluation campaign (SASSEC) measures. It is shown from the evaluation that the proposed method outperforms a sound source separation method without applying a beamformer.
Convention Paper 8976

3:00 pm

P12-3 Reverberation and Dereverberation Effect on Byzantine Chants—*Alexandros Tsilfidis*,[1] *Charalampos Papadakos*,[1] *Elias Kokkinis*,[1] *Georgios Chryssochoidis*,[2] *Dimitrios Delviniotis*,[2] *Georgios Kouroupetroglou*,[2] *John Mourjopoulos*[1]
[1]University of Patras, Patras, Greece
[2]National and Kapodistrian University of Athens, Athens, Greece

Byzantine music is typically monophonic and is characterized by (i) prolonged music phrases and (ii) Byzantine scales that often contain intervals smaller than the Western semitone. As happens with most religious music genres, reverberation is a key element of Byzantine music. Byzantine churches/cathedrals are usually characterized by particularly diffuse fields and very long Reverberation Time (RT) values. In the first part of this work, the perceptual effect of long reverberation on Byzantine music excerpts is investigated. Then, a case where Byzantine music is recorded in non-ideal acoustic conditions is considered. In such scenarios, a sound engineer might require to add artificial reverb on the recordings. Here it is suggested that the step of adding extra reverberation can be preceded by a dereverberation processing to suppress the originally recorded non ideal reverberation. Therefore, in the second part of the paper a subjective test is presented that evaluates the above sound engineering scenario.
Convention Paper 8977

3:00 pm

P12-4 Cepstrum-Based Preprocessing for Howling Detection in Speech Applications—*Renhua Peng*,[1,2] *Jian Li*,[1,2] *Chengshi Zheng*,[1,2] *Xiaoliang Chen*,[1,2] *Xiaodong Li*[1,2]
[1]Chinese Academy of Sciences, Beijing, China
[2]Chinese Academy of Sciences, Shanghai, China

Conventional howling detection algorithms exhibit dramatic performance degradations in the presence of harmonic components of speech that have the similar properties with the howling components. To solve this problem, this paper proposes a cepstrum preprocessing-based howling detection algorithm. First, the impact of howling components on cepstral coefficients is studied in both theory and simulation. Second, according to the theoretical results, the cepstrum pre-processing-based howling detection algorithm is proposed. The Receiver Operating Characteristic (ROC) simulation results indicate that the proposed algorithm can increase the detection probability at the same false alarm rate. Objective measurements, such as Speech Distortion (SD) and Maximum Stable Gain (MSG), further confirm the validity of the proposed algorithm.
Convention Paper 8978

3:00 pm

P12-5 Delayless Method to Suppress Transient Noise Using Speech Properties and Spectral Coherence—*Chengshi Zheng*,[1,2] *Xiaoliang Chen*,[1,2] *Shiwei Wang*,[1,2] *Renhua Peng*,[1,2] *Xiaodong Li*[1,2]
[1]Chinese Academy of Sciences, Beijing, China
[2]Chinese Academy of Sciences, Shanghai, China

This paper proposes a novel delayless transient noise reduction method that is based on speech properties and spectral coherence. The proposed method has three stages. First, the transient noise components are detected in each subband by using energy-normalized variance. Second, we apply the harmonic property of the voiced speech and the continuity of the speech signal to reduce speech distortion in voiced speech segments. Third, we define a new spectral coherence to distinguish the unvoiced speech from the transient noise to avoid suppressing the unvoiced speech. Compared with those existing methods, the proposed method is computationally efficient and casual. Experimental results show that the proposed algorithm can ➥

effectively suppress transient noise up to 30 dB without introducing audible speech distortion.
Convention Paper 8979

3:00 pm

P12-6 Artificial Stereo Extension Based on Hidden Markov Model for the Incorporation of Non-Stationary Energy Trajectory—*Nam In Park,*[1] *Kwang Myung Jeon,*[1] *Seung Ho Choi,*[2] *Hong Kook Kim*[1]

[1]Gwangju Institute of Science and Technology (GIST), Gwangju, Korea
[2]Seoul National University of Science and Technology, Seoul, Korea

In this paper an artificial stereo extension method is proposed to provide stereophonic sound from mono sound. While frame-independent artificial stereo extension methods, such as Gaussian mixture model (GMM)-based extension, do not consider the correlation of energies of previous frames, the proposed stereo extension method employs a minimum mean-squared error estimator based on a hidden Markov model (HMM) for the incorporation of non-stationary energy trajectory. The performance of the proposed stereo extension method is evaluated by a multiple stimuli with a hidden reference and anchor (MUSHRA) test. It is shown from the statistical analysis of the MUSHRA test results that the stereo signals extended by the proposed stereo extension method have significantly better quality than those of a GMM-based stereo extension method.
Convention Paper 8980

3:00 pm

P12-7 Simulation of an Analog Circuit of a Wah Pedal: A Port-Hamiltonian Approach—*Antoine Falaize-Skrzek, Thomas Hélie,* IRCAM-CNRS-UPMC, Paris, France

Several methods are available to simulate electronic circuits. However, for nonlinear circuits, the stability guarantee is not straightforward. In this paper the approach of the so-called "Port-Hamiltonian Systems" (PHS) is considered. This framework naturally preserves the energetic behavior of elementary components and the power exchanges between them. This guarantees the passivity of the (source-free part of the) circuit.
Convention Paper 8981

3:00 pm

P12-8 Improvement in Parametric High-Band Audio Coding by Controlling Temporal Envelope with Phase Parameter—*Kijun Kim,*[1] *Kihyun Choo,*[2] *Eunmi Oh,*[2] *Hochong Park*[1]

[1]Kwangwoon University, Seoul, Korea
[2]Samsung Electronics Co., Ltd., Suwon, Korea

This study proposes a method to improve temporal envelope control in parametric high-band audio coding. Conventional parametric high-band coders may have difficulties with controlling fine high-band temporal envelope, which can cause the deterioration in sound quality for certain audio signals. In this study a novel method is designed to control temporal envelope using spectral phase as an additional parameter. The objective and the subjective evaluations suggest that the proposed method should improve the quality of sound with severely degraded temporal envelope by the conventional method.
Convention Paper 8982

Tutorial 10 **Friday, Oct. 18**
3:00 pm – 5:00 pm **Room 1E07**

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

Presenter: **John Dawson**

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.

Student/Career Development Event
STUDENT RECORDING CRITIQUES
Friday, October 18, 3:00 pm – 4:00 pm
Room 1E06

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

Students! Bring your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on CD, DVD, memory-stick, or hard disk, as clearly labeled 44.1 KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers!

Project Studio Expo
Friday, October 18, 3:00 pm – 4:00 pm
Stage

MASTER YOUR TRACKS: DIY RESULTS TO COMPETE WITH THE PROS

Presenter: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

Mastering is the final step prior to duplication, and as such, represents the last opportunity to make any final tweaks to a piece of music for highest sonic quality—and maximum transportability among playback systems. Traditionally, musicians have used professional mastering engineers in order to take advantage of their experience and ears, but in today's tight economy—and with the advent of tools that allow for "do-it-yourself" mastering—many musicians are choosing to do their own mastering. This workshop describes the pitfalls and advantages of "project mastering" as well as the main mistakes to avoid but primarily emphasizes practical techniques that can bring out the very best in a piece of music. It also covers the process of album assembly and how to make sure the music in a collection or album provides a smooth, cohesive listening experience.

Knowledge Center
Friday, October 18, 3:00 pm – 4:00 pm
Booth 2738

MULTITRACK RECORDING IN THE FIELD WITH JON TATOOLES OF SOUND DEVICES

Presenters: **Jon Tatooles**, Sound Devices
Pat McConnell, Sound Devices

Join our favorite Sound Devices duo, Jon Tatooles and Pat McConnell as they discuss the rise of unscripted dialogue sound production and the increasing demand for iso track as well as mix track deliverables. Let these field savvy mixers fill you with the tools and techniques for perfect delivery, every time.

Knowledge Center
Friday, October 18, 3:00 pm – 4:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Friday, Oct. 18 3:00 pm Room 1E04
Technical Committee Meeting: Hearing and Hearing Loss Prevention

Network Audio Session 4 Friday, Oct. 18
3:30 pm – 5:00 pm Room 1E13

COMMAND AND CONTROL PROTOCOLS, TARGET APPLICATION USE CASES

Chair: **Tim Shuttleworth**, Renkus Heinz, Oceanside, CA, USA

Panelists: *Jeff Berryman*, Bosch Communications, Ithaca, NY, USA
Andrew Eales, Wellington Institute of Technology, Wellington, New Zealand
Richard Foss, Rhodes University, Grahamstown, Eastern Cape, South Africa
Jeff Koftinoff, Meyer Sound Canada, Vernon, BC, Canada

With the increasing utilization of data networks for the command and control of audio devices a number of protocols have been defined and promoted. These competing protocol initiatives, while providing methods suited to their target applications, have created confusion among potential adopters as to which protocol best fits their needs. In addition, the question is being asked, Why do we need so many "standard" protocols? At least four different industry organizations have involved themselves in some form of standardized protocol effort. AES is currently pursuing standardization of two such protocols, AES64 and X-210 (aka OCA). IEC has IEC62379, while IEEE is defining AVDECC (IEE1722.1) and ESTA offers ACN and there's OSC from opensoundcontrol.org. This workshop addresses what differentiates these protocols by examining their target use applications.

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

Friday, Oct. 18 3:30 pm Room 1E02
Standards Committee Meeting: SC-02-01 Working Group on Digital Audio Measurement

Broadcast/Streaming Media Session 8
Friday, Oct. 18 3:45 pm – 5:15 pm
Room 1E14

CONTENT DELIVERY AND THE MOBILE INITIATIVE

Chair: **Neil Glassman**, WhizBangPowWow, Jersey City, NJ, USA

Panelists: *Karlheinz Brandenburg*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany; Ilmenau University of Technology, Ilmenau, Germany
John Kean, NPR
Ray Lau, Ramp
Leigh Newsome, Targetspot, New York, NY, USA
Jan Nordmann, Fraunhofer USA, San Jose, CA, USA
Greg Ogonowski, Orban, San Leandro, CA, USA

Consumer use of mobile devices for entertainment and information is exploding. Smart phones and tablets are used for both primary programming and "second screen" applications. These devices are increasingly being integrated into the "connected car," where legacy receivers are no longer the only built-in listening option. Applying the term "streaming" to a broad range of delivery platforms, this panel will look at the established and nascent technical advancements that have enabled content providers to reach the expanding mobile audience. We'll also explore whether audio and data technologies are changing consumer preferences or merely keeping up with them. Panelists will also pull out their crystal balls to predict the future technologies that will help help some of the platforms grow their listener base and turn other platforms dark.

Special Event
BRIDGING THE GAP BETWEEN CREATIVITY & TECHNOLOGY: WORKING WITH COMPOSERS ON FILM AND MEDIA PROJECTS
Friday, October 18, 4:00 pm – 5:15 pm
Room 1E03

Moderator: **Frank Ferrucci**, Manhattan Producers Alliance VP, Composer/Producer

This seminar gives a behind the scenes look into the technological challenges composers and engineers face when collaborating on film, television, and other visual media projects. The presentation addresses some less ➥

obvious but no less important ways that Music Engineers and Film Mixers can work best with composers and how technology can be used to help this collaboration be as seamless as possible.

Student/Career Development Event
RECORDING COMPETITION—PART 1
Friday, October 18, 4:00 pm – 6:00 pm
Room 1E11

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

4:00 pm Traditional Studio Recording

Judges: Jim Anderson, Jim Kaiser, Andres Mayo

5:00 pm Modern Studio Recording

Judges: Richard King, Jonathan Wyner, Ronald Prent

Project Studio Expo
Friday, October 18, 4:00 pm – 5:00 pm
Stage

YOU ASK, WE ANSWER

Presenters: **Hugh Robjohns**, Technical Editor, Sound on Sound, Cambridge, UK
Mike Senior, Sound On Sound, Munich, Germany; Cambridge Music Technology
Paul White

Open Q&A session on recording and mixing techniques.

Friday, Oct. 18 4:00 pm Room 1E04
Technical Committee Meeting: Semantic Audio Analysis

Live Sound Seminar 7 Friday, Oct. 18
4:30 pm – 6:30 pm Room 1E12

DESIGN FOR HOUSES OF WORSHIP AND INSTALLED SOUND

Chair: **Bill Thrasher, Sr.**, Thrasher Desig Group, Inc., Kennesaw, GA, USA

One of the professional audio industry's largest and persistently expanding markets, the House of Worship sector has matured into a highly sophisticated, demanding, and incredibly diverse collection.

This panel will discuss issues ranging from budget to design and install, from service and support to operational training, from the perspectives of management, manufacturers, consultants/designers, contractors, vendors, users, operators, and the listeners.

Knowledge Center
Friday, October 18, 4:30 pm – 5:30 pm
Room 1E06

PMC "MASTERS OF AUDIO": ULRIKE SCHWARZ

Presenter: **Ulrike Schwarz**, Bayerischer Rundfunk, Munich, Germany

Last year, Latvian conductor Mariss Jansons fulfilled his heart's desire and performed all of Beethoven's symphonies together with the Bavarian Radio Symphony Orchestra in one of the world's most beautiful concert halls—the Suntory Hall in Tokyo. Ulrike Schwarz was the recording and mixing engineer for this immense project. A total of 67 stringed instruments made the trip (34 violins, 13 violas, 11 cellos, and 9 basses), plus 23 brass instruments, and a full complement of wind. Nor is this just "any" orchestra. In 2008, Gramophone placed the Bavarian Radio Symphony Orchestra sixth in the world, ahead of numerous (and far) more fancied competitors.

Knowledge Center
Friday, October 18, 4:30 pm – 6:00 pm
Booth 2738

RF MICS, SPECTRUM, AND THE FUTURE WITH KARL WINKLER OF LECTROSONICS

Presenter: **Karl Winkler**, Lectrosonics, Rio Rancho, NM, USA

Straight from Rio Rancho, Karl Winkler has been staying up well past his bedtime preparing to tell you about the future of RF. Are the terms "spectrum" and "FCC Regulation" keeping you up at night? Do you worry about your wireless future? Then let Karl be your crystal ball and do not miss this presentation!

Session EB3 Friday, Oct. 18
5:00 pm – 6:30 pm 1E Foyer

E-BRIEF POSTERS—PART 1

5:00 pm

EB3-1 A Special Room for 3D Audio and Ultra High Definition Video for Quality Assessment of Future TV—*Matthieu Parmentier*, francetélévisions, Paris, France

francetélévisions, the French public broadcaster, is involved in collaborative projects that aim to embrace the future of television. Ultra High Definition video in conjunction with 3-D sound are today explored within the range of content production, techniques, costs, and quality of experience for consumer applications. With its new-dedicated room, the innovations and developments department of francetélévisions has built a necessary tool to drive its strategy for facing the upcoming challenges.
Engineering Brief 112

5:00 pm

EB3-2 Real-Time Head-Related Impulse Response Filtering with Distance Control—*Julian Villegas, Michael Cohen*, University of Aizu, Aizu Wakamatsu-shi, Fukashima-ken, Japan

We present a new software application based on

a recently collected HRIR database comprising measurements at different distances. The new application, programmed in Pure-data, is capable of directionalizing sound objects at any azimuth, at elevations between -40 degrees and 90 degrees, and at distances 20-160 cm. This truly 3D spatialization is done by pre-calculating the minimum-phase version of the HRIRs and computing the interpolation of a maximum of four HRIR measurements, depending upon the virtual location. In the same way, interaural time differences are computed and applied to the convolved signal. For demanding real-time constraints, the number of taps used for the convolution can be adjusted, up to a maximum of 1024.
Engineering Brief 113

5:00 pm

EB3-3 Results on Automated Tuning of a Voice Quality Enhancement System Using Objective Quality Measures—*Daniele Giacobello, Joshua Atkins, Jason Wung, Raghavendra Prabhu*, Beats Electronics, LLC, Santa Monica, CA, USA

In this work we present a formal procedure for automating the tuning of the various parameters comprising a voice quality enhancer. First, we formalize the problem of tuning as a large-scale nonlinear programming problem. Second, we evaluate the performance of perceptual objective quality measures as optimization criteria for our tuning problem. We then perform a subjective quality assessment to compare the output of a voice enhancer obtained with parameters calculated with these different criteria and also with those obtained through a conventional approach of tuning by expert listening. The results show that using this automated methodology performs well in finding reasonable solutions for the tuning problem, potentially saving time and resources over manual evaluation and tuning.
Engineering Brief 114

5:00 pm

EB3-4 Influence of Loudspeaker Systems on Acquisition of Head-Related Transfer Functions—*Shouichi Takane, Koji Abe, Kanji Watanabe, Sojun Sato*, Akita Prefecture University, Yurihonjo, Akita, Japan

Head-Related Transfer Function (HRTF) is defined by the ratio of sound pressure at the center of the head without listener and the one at his/her ear. Frequency characteristics of a sound source ought to be cancelled in its acquisition based on this definition, but they are not when the sound sources are spatially distributed such as conventional multiple-driver loudspeaker systems. In this e-brief such influence was investigated by using the HRTFs acquired with various types of loudspeaker systems. As a result, it was found that the HRTFs acquired with four types of loudspeakers roughly agreed when the distance from the sound source is 1.5 m and farther.
Engineering Brief 115

5:00 pm

EB3-5 Application of Audio Engineering and Psychoacoustic Principles to Audible Medical Alarms—*Christopher Bennett*,[1,2] *Colby N. Leider*,[1] *Richard McNeer*[1]
[1]University of Miami, Coral Gables, FL, USA
[2]Oygo Sound LLC, Miami, FL, USA

Audible medical alarms standards have recently undergone extensive review by regulatory and safety organizations due to reported ineffectiveness of alarms and the role of "alarm fatigue" in contributing to morbidity and mortality among patients. Many of the problems associated with alarm fatigue stem from an improper application of psychoacoustic and audio engineering principles and naive design of auditory streams that lead to poor segregation, confusion among clinicians, and ultimately fatigue. The audio engineer has a clear role in defining solutions to problems arising in hospital units, some of which have previously been addressed in sound production, sound design, and auditory scene analysis. The roles of sonification, psychoacoustics, and sound perception are discussed as they apply to audible medical alarms.
Engineering Brief 116

5:00 pm

EB3-6 Revisiting the Space—Applying 5.1 Surround Sound—*Mike Godwin*, University of Western Ontario, London, ON, Canada

Origins of this project grew from requests from Faculty and Performers wishing there was a way to better experience the live acoustic again while listening to our archival recordings. As such, my objective was to research an approach to record one of our early music choirs performing in an ambient venue utilizing 5.1 surround techniques. Then through listening tests, obtain subject preferences for the stereo vs. 5.1 versions with comment categories for each. For this project the goal was to use the simplest possible setup as we are most often in a live concert environment, and where setup time is also a consideration. Initially I did recordings testing both cardioid and omni microphones to decide on the best patterns, and placements.
Engineering Brief 117

Tutorial 11 — Friday, Oct. 18
5:00 pm – 6:30 pm — Room 1E09

OPTIMIZED CROSSTALK CANCELLATION FOR UNCOLORED 3-D AUDIO FROM LOUDSPEAKERS: RECENT ADVANCES AND APPLICATIONS (A MASTER CLASS EVENT)

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

This Master Class starts with a brief review of the three main methods for 3-D sound reproduction over loudspeakers: (1) Wave Field Synthesis, (2) Ambisonics, and (3) Binaural audio through loudspeakers (BAL). The main focus of the talk is on recent advances with the third method. I will show that crosstalk cancellation (XTC) allows BAL to deliver to the listener the necessary cues for real 3-D audio but that it inherently imposes an intolerably high spectral coloration on the audio. I will describe recent breakthroughs, which allow producing optimized XTC filters that impose no spectral coloration and then discuss the two other problems that have ➥

retarded the commercialization of XTC: the fixed and single sweet spot problems. I will show how the first problem is solved through advanced head tracking; and the second problem is solved using head tracking and phased array speakers, allowing the delivery of high-spatial-fidelity 3-D audio to multiple moving listeners in real listening rooms. Following the talk, there will be a demo with playback of recorded music and natural sounds.

Game Audio Session 6 Friday, Oct. 18
5:00 pm – 6:30 pm Room 1E10

IN THE TRENCHES

Moderator: **Scott Selfon**, Microsoft, Redmond, WA, USA

Presenters: *Russell Brower*, Blizzard Entertainment
Jason Kanter, Avalanche Studios
D. Chadd Portwine, Vicarious Visions
Alex Wilmer, Crystal Dynamics

The guys doing the work know the most. Let's hear what they have to say about what bugs them, makes them smile, makes them drink. Tool sets both commercial and proprietary are how we get the job done. What works, what needs improvement? Who do these people rely upon for tech help, production info, direction, physical therapy? What goes on behind closed doors?

Historical Event
THE ART OF RECORDING THE BIG BAND, REVISITED
Friday, October 18, 5:00 pm – 6:30 pm
Room 1E13

Presenter: **Robert Auld**, Auldworks, New York, NY, USA

The jazz big band was born in the 1920s, came of age in the 1930s, enjoyed its greatest popularity in the 1940s, and went into popular decline in the 1950s. In the 1960s the big band enjoyed a comeback of sorts but was displaced from the front pages by The Beatles and other things. In the 1970s it looked like the big band would either expire, or be transformed out of recognition. And yet, it persists; people still play in big bands, still dance to them, still record them. It has proved a most durable ensemble.

In his presentation Robert Auld will survey the history of jazz big bands, both from a musical and a technical, recording point of view. He will also show how he recorded a modern big band in 2011, using techniques typical of recording sessions in the "golden age" of stereo in the late 1950s. The presentation will include photos and recorded examples.

Special Event
INSIDE ABBEY ROAD 1967—
PHOTOS FROM THE SGT. PEPPER SESSIONS
Friday, October 18, 5:00 pm – 6:30 pm
Room 1E15/16

Moderator: **Allan Kozinn**, NY Times, New York, NY, USA

Panelists: *Henry Grossman*
Brian Kehew, CurveBender Publishing, Los Angeles, CA, USA

Allan Kozinn, noted Beatles expert and reviewer for the *NY Times* will moderate this panel, which shows a behind-the-scenes look at EMI/Abbey Road studios during the making of the landmark "Sgt. Pepper's Lonely Hearts Club Band." Famed Beatles photographer Henry Grossman visited the sessions where he took several hundred photos, many of which are still largely unseen. Henry will show photos and share memories of that creative era. Brian Kehew (co-author of the acclaimed *Recording the Beatles* book) will illustrate key technical aspects found in Grossman's photos. (Henry Grossman is also the author of *Kaleidoscope Eyes: A Day in the Life of Sgt. Pepper* and *Places I Remember: My Time with The Beatles*, considered two of the greatest collections of Beatles photography to date.)

Project Studio Expo
Friday, October 18, 5:00 pm – 6:00 pm
Stage

JIMMY JAM: Q&A WITH RENOWNED PRODUCER

Moderator: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

Presenter: **Jimmy Jam**, Flyte Tyme Productions

Friday, Oct. 18 5:00 pm Room 1E04
Technical Committee Meeting: Signal Processing

Broadcast/Streaming Media Session 9
Friday, Oct. 18 5:30 pm – 7:00 pm
Room 1E14

MODERN AUDIO TRANSPORTATION TECHNIQUES FOR REMOTE BROADCASTS

Chair: **Herb Squire**

Panelists: *Chris Crump*, Comrex
Chris Nelson, NPR
Greg Shay, The Telos Alliance, Cleveland, OH, USA
Chris Tobin, CCS-IPcodecs, Newark, NJ, USA

Evolving technology has made great strides in audio transport versatility, connectivity, availability, and reliability. Whether wired or wireless, this discussion will provide real-time remote program solution options for broadcasters trying to make ends meet.

Product Design Session 2 Friday, Oct. 18
5:30 pm – 7:00 pm Room 1E07

HIGH-ORDER HARMONIC DISTORTION MEASUREMENT OF AMPLIFIERS AND ITS IMPACT ON FIDELITY

Presenter: **Dan Foley**, Audio Precision, Worcester, MA, USA
Roger Gibboni, Rogers High Fidelity, Warwick, NY, USA

The electronics side of the audio industry has standardized on THD and THD+N as the main means of characterizing distortion, especially for amplifiers. However in 1942, RCA engineers who wrote the *Radiotron Handbook* proposed a weighted THD metric that weighted the energy of high-order harmonics to a much greater degree than low-order harmonics. Listening tests back then did show a correlation of amplifiers with very little high-order harmonic distortion being more acceptable compared to other designs with greater high-order distortion even though THD differed slightly. This presentation will focus on current measurement methods that can be used to separate high-order and low-order distortion.

Workshop 15 **Friday, Oct. 18**
6:30 pm – 8:00 pm **Room 1E11**

**SEX, LIES AND SURROUNDTAPES—
WHAT HAPPENED TO ALL THE FUN IN THE WORLD?**

Chair: **Florian Camerer**, Austrian TV

Panelists: *Ronald Prent*, Wisseloord Studios
Bosse Ternstrom, Swedish Radio

Surround Sound is again rising—literally, as height channels appear and immerse the listener in three dimensions. But what about good ol' 5.1? Have we mastered it to the point where we are in desperate need for new challenges? Two nostalgic veterans of 5.1, Bosse Ternstrom from Swedish Radio and Florian Camerer from Austrian TV, will play their "20 golden 5.1 hits" and muse about the sex appeal of the tracks as well as slay some potential dragons and lies (LFE! :-))... Come by for an evening listening session to heighten your senses—and also to enjoy some of the great mixes of Maestro George Massenburg!

Special Event
ORGAN RECITAL BY GRAHAM BLYTH
Friday, October 18, 8:30 pm – 9:30 pm
Church of Saint Mary the Virgin
145 West 46th Street, NY

Graham Blyth's traditional organ recital will be given at the Church of Saint Mary the Virgin, founded in 1868. The recital includes works by Marchand, Bach, and Franck

The organ at the Church of Saint Mary the Virgin, Aeolian-Skinner Opus 891, was installed in January 1933, and the dedicatory recital was played by Professor Palmer Christian of the University of Michigan on January 11th of that year. The organ, designed by organbuilder G. Donald Harrison, is widely considered a forerunner of the "American Classic" school. The program notes for the dedicatory recital include the following: "The new organ . . . is something of an innovation, if one may call a return to ancient principles an innovation, a return to the principles of the 'classic' organ, the organ of the Thomas-Kirche and of the older French and German builders." The new organ, though incomplete, certainly attracted quite a stir, with numerous articles being devoted to it, and all appreciative of its place as a revolutionary instrument.

One of the main goals in American Classic organ building was for a satisfactory organ ensemble sound, rather than an emphasis on solo voices, and an ability to play organ music of all styles and periods. The organ, when first installed and today, however, certainly bears a strong French influence, and has always been appreciated for its ability to render French organ music appropriately. The organ was revised, with substantial changes in a number of divisions, by G. Donald Harrison in 1942 and renamed Opus 891-A. Nevertheless, the organ was still not complete. A number of small changes occurred over the next 40 years. Finally, in 1988, the completion of the organ was begun by Mann & Trupiano Organbuilders of Brooklyn. A number of ranks were added, including the planned Bombarde division, and the organ was brought very close to Harrison's original design, although the intended case has never been built.

The most recent enhancements include a 16-foot Bourdon in the pedal, and a floating chorus trumpet. This trumpet stop has a particularly interesting history—it was installed in 1956 by G. Donald Harrison for the organ at Saint Thomas Church Fifth Avenue (and used in Marcel Dupré's famous recording on that instrument). Not long after that, however, it was discarded during revisions of their organ. At the instigation of McNeil Robinson (then organist at Saint Mary's), it found its way to St. Mary's, where it was stored in the basement for 38 years. Lawrence Trupiano, curator of organs, restored and installed the trumpet for use in the organ. It seems appropriate that this trumpet, one of the last reed stops designed by Harrison, has found a new home in one of his first instruments with the Aeolian-Skinner Company. Great organ works may be heard on a weekly basis at Solemn Mass and in the church's recital series.

Graham Blyth was born in 1948, began playing the piano aged 4 and received his early musical training as a Junior Exhibitioner at Trinity College of Music in London, England. Subsequently, at Bristol University, he took up conducting, performing Bach's St. Matthew Passion before he was 21. He holds diplomas in Organ Performance from the Royal College of Organists, The Royal College of Music and Trinity College of Music. In the late 1980s he renewed his studies with Sulemita Aronowsky for piano and Robert Munns for organ. He gives numerous concerts each year, principally as organist and pianist, but also as a conductor and harpsichord player. He made his international debut with an organ recital at St. Thomas Church, New York in 1993 and since then has played in San Francisco (Grace Cathedral), Los Angeles (Cathedral of Our Lady of Los Angeles), Amsterdam, Copenhagen, Munich (Liebfrauen Dom), Paris (Madeleine and St. Etienne du Mont), and Berlin, among other cities. He has lived in Wantage, Oxfordshire, since 1984 where he is currently Artistic Director of Wantage Chamber Concerts and Director of the Wantage Festival of Arts.

He divides his time between being a designer of professional audio equipment (he is a co-founder and Technical Director of Soundcraft) and organ related activities. In 2006 he was elected a Fellow of the Royal Society of Arts in recognition of his work in product design relating to the performing arts.

Special Event
**STORIES FOR THE EARS:
LIVE AUDIO DRAMA AND NARRATION**
Friday, October 18, 8:30 pm – 10:00 pm
Presented at The Paley Center for Media
(Doors open at 8:00 pm – show begins at 8:30 pm)
Limited seating, tickets required.

Fantasy, Fiction, and Fun!

The HEAR Now Festival and SueMedia Productions in conjunction with the Audio Engineering Society (AES) presents an evening of live audio/radio drama along with narrative readings celebrating the art of sonic storytelling.

Hosted by Bob Kaliban (CBS Mystery Theater) featuring performances by Audie Award winning and Golden Voice narrators Jim Dale, Katherine Kellgren, Robin Miles, and Barbara Rosenblat, and the award winning NY-based audio drama troupe VoiceScapes Audio Theater.

The sponsors of this event are CCS-IPCodecs, SueMedia Productions, Hear Now Festival, and the Audio Engineering Society.

Session P13 **Saturday, Oct. 19**
9:00 am – 11:30 am **Room 1E07**

APPLICATIONS IN AUDIO—PART 2

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

9:00 am

P13-1 Level-Normalization of Feature Films Using

Loudness vs Speech—*Esben Skovenborg, Thomas Lund*, TC Electronic A/S, Risskov, Denmark

We present an empirical study of the differences between level-normalization of feature films using the two dominant methods: loudness normalization and speech ("dialog") normalization. The sound of 35 recent "blockbuster" DVDs were analyzed using both methods. The difference in normalization level was up to 14 dB, on average 5.5 dB. For all films the loudness method provided the lowest normalization level and hence the greatest headroom. Comparison of automatic speech measurement to manual measurement of *dialog anchors* shows a typical difference of 4.5 dB, with the automatic measurement producing the highest level. Employing the speech-classifier to process rather than measure the films, a listening test suggested that the automatic measure is positively biased because it sometimes fails to distinguish between "normal speech" and speech combined with "action" sounds. Finally, the DialNorm values encoded in the AC-3 streams on DVDs were compared to both the automatically and the manually measured speech levels and found to match neither one well.
Convention Paper 8983

9:30 am

P13-2 Sound Identification from MPEG-Encoded Audio Files—*Joseph G. Studniarz, Robert C. Maher*, Montana State University, Bozeman, MT, USA

Numerous methods have been proposed for searching and analyzing long-term audio recordings for specific sound sources. It is increasingly common that audio recordings are archived using perceptual compression, such as MPEG-1 Layer 3 (MP3). Rather than performing sound identification upon the reconstructed time waveform after decoding, we operate on the undecoded MP3 audio data as a way to improve processing speed and efficiency. The compressed audio format is only partially processed using the initial bitstream unpacking of a standard decoder, but then the sound identification is performed directly using the frequency spectrum represented by each MP3 data frame. Practical uses are demonstrated for identifying anthropogenic sounds within a natural soundscape recording.
Convention Paper 8984

10:00 am

P13-3 Pilot Workload and Speech Analysis: A Preliminary Investigation—*Rachel M. Bittner*,[1] *Durand R. Begault*,[2] *Bonny R. Christopher*[3]
[1]New York University, New York, NY, USA
[2]Human Systems Integration Division, NASA Ames Research Center, Moffett Field, CA, USA
[3]San Jose State University Research Foundation, NASA Ames Research Center, Moffett Field, CA, USA

Prior research has questioned the effectiveness of speech analysis to measure a talker's stress, workload, truthfulness, or emotional state. However, the question remains regarding the utility of speech analysis for restricted vocabularies such as those used in aviation communications. A part-task experiment was conducted in which participants performed Air Traffic Control read-backs in different workload environments. Participant's subjective workload and the speech qualities of fundamental frequency (F_0) and articulation rate were evaluated. A significant increase in subjective workload rating was found for high workload segments. F_0 was found to be significantly higher during high workload while articulation rates were found to be significantly slower. No correlation was found to exist between subjective workload and F_0 or articulation rate.
Convention Paper 8985

10:30 am

P13-4 Gain Stage Management in Classic Guitar Amplifier Circuits—*Bryan Martin*, McGill University, Montreal, QC, Canada

The guitar amplifier became a common tool in musical creation during the second half of the 20th Century. This paper attempts to detail some of the internal mechanisms by which the tones are created and their dependent interactions. Two early amplifier designs are examined to determine the circuit relationships and design decisions that came to define the sound of the electric guitar.
Convention Paper 8986

11:00 am

P13-5 Audio Pre-Equalization Models for Building Structural Sound Transmission Suppression —*Cheng Shu,*[1] *Fangyu Ke,*[1] *Xiang Zhou,*[2] *Gang Ren*,[1] *Mark F. Bocko*[1]
[1]University of Rochester, Rochester, NY, USA
[2]Bose Corporation, Framingham, MA, USA

We propose a novel audio pre-equalization model that utilizes the transmission characteristics of building structures to reduce the interference reaching adjacent neighbors while maintaining the audio quality for the target listener. The audio transmission profiles are obtained by field acoustical measurements in several typical types of building structures. We also measure the spectrum of audio to adapt the pre-equalization model to a specific audio segment. We apply a computational auditory model to (1) monitor the perceptual audio quality for the target listener and (2) access the interference caused to adjacent neighbors. The system performance is then evaluated using subjective rating experiments.
Convention Paper 8987
[Paper was not presented but is available for download from the E-Library]

Tutorial 12 **Saturday, Oct. 19**
9:00 am – 10:30 am **Room 1E13**

FXPERTISE: DISTORTION

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

Distortion can be good, or bad. With the right touch, it can lift a track up out of a crowded arrangement and add excitement to a performance. Yet too much distortion renders the track too messy, too murky to be enjoyed. Accidental distortion is a certain sign that the production is unprofessional. Amps, stomp boxes, tubes, transformers, tape machines, the plug-ins that emulate them, and the plug-ins that create wholly new forms of distortion all offer a rich palette of possibilities. Audio engineers must choose the right tool for the job and then tailor the distortion to the music. This advanced tutorial takes a close look at distortion, detailing the technical goings-on when things break-up, and defining the production potential of this always-tempting effect.

Game Audio Session 7 **Saturday, Oct. 19**
9:00 am – 10:00 am **Room 1E10**

CODE MONKEY: MAPPING AUDIO INTO A 3-D GAME WORLD

Presenter **Michael Kelly**, DTS, Inc., London, UK

In a game, audio lives not in isolation but often as part of a rich and complex 3-D world. This code monkey session gives an overview of the links between the 3-D game world and the world of audio DSP, with particular emphasis on the representation of sound within a 3-D world. The tutorial is aimed at audio engineers who are looking to brush up on a little 3-D math and helper SDKs as well as those who are new to this field. Attendees can expect to hear coverage of converting from Carestian to spherical coordinates, matrix transformation, vector-base amplitude panning, distance modeling, programming examples with XAudio2, and how all this fits together.

Live Sound Seminar 8 **Saturday, Oct. 19**
9:00 am – 11:00 am **Room 1E12**

DESIGN MEETS REALITY: THE A2'S AND PRODUCTION SOUND MIXER'S CHALLENGES, OBSTACLES, AND RESPONSIBILITIES FOR LOADING IN AND IMPLEMENTING THE SOUND DESIGNER'S CONCEPT

Moderator: **Christopher Evans**, Benedum Center, Pittsburgh, PA, USA

Panelists: *Collie Bustin,* IRES-Partners, LLC, New York, NY, USA
Paul Garrity, Auerbach Pollock Friedlander, New York, NY, USA
Scott Lehrer, Scott Lehrer Sound Design, Ltd., New York, NY, USA
Augie Propersi, NYC City Center
Dominic Sack, Sound Associates, Inc.
Christopher Sloan, Production Engineer, The Book of Mormon

The best intentions of the sound designer don't always fit in with the venue's interior or infrastructure, other departments' needs, or other changes as a production is loaded in and set up for the first time. How the designer's designated representative on site addresses these issues is critical to keeping the overall vision of the sound design and production aesthetics intact while keeping an eye on the budget and schedule.

Product Design Session 3 **Saturday, Oct. 19**
9:00 am – 11:00 am **Room 1E09**

TELEPHONY: AN INTRODUCTION TO THE ACOUSTICS OF PERSONAL TELECOMMUNICATIONS DEVICES

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

The basic concepts of telephonometry and electroacoustics of telephones and personal telecommunications devices are introduced. Techniques for assessing analog, digital, cellular, VoIP, USB, and other telephone devices are presented. Objective evaluation of the performance of handsets, headsets, speakerphones, and hands-free devices is discussed and interfacing to these devices is explained. Selection, calibration, and use of microphones, ear simulators, mouth simulators, and test fixtures are described. The send, receive, sidetone, and echo transmission paths are defined. The use of real speech test signals and pulsed noise for distortion is illustrated using examples from IEEE 269. The concept of Loudness Rating, its history, and standardized methods for its calculation are reviewed. Methods specified in ITU-T, IEEE, TIA, ETSI, and the 3GPP standards are explained.

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

Sound for Picture 3 **Saturday, Oct. 19**
9:00 am – 10:30 am **Room 1E11**

DIALOG EDITING AND MIXING FOR FILM (SOUND FOR PICTURES MASTER CLASS)

Presenters: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia
Fred Rosenberg

Film soundtracks contain three elements—dialog, music, and sound effects. Dialog is the heart of the process, with "telling the story" the primary goal of the dialog. With multiple sources of dialog available, the assessment and planning of the dialog and subsequent mixing is a critical element in the process. This Master Class with one of Hollywood's leading professionals puts the process under the microscope.

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

Student/Career Development Event
SPARS SPEED COUNSELING WITH EXPERTS —MENTORING ANSWERS FOR YOUR CAREER
Saturday, October 19, 9:00 am – 11:00 am
Room 1E14

Moderator: **Kirk Imamura**

Mentors: Studio Production:
Mark Rubel
David Kahne
Craig Schumacher
Chris Mara
Glenn Lorbecki,
Drew Waters
Barry Rudolph
Pat McMakin
Kevin Killen
Todd Whitelock
Sound for Picture:
Leslie Mona-Mathus
Eric Johnson

Jamie Baker
Bill Higley
Jun Mizumachi
Rick Senechal
Gaming:
Tom Salta
Gina Zdanowicz
Scott Selfon
Randy Coppinger
Live Sound/Live Recording:
Rick Camp
Jeri Palumbo
Erik Zobler
Theatrical Sound:
Peter Hylenski
Nevin Steinberg
Kai Harada
Simon Matthews

This event is specially suited for students, recent graduates, young professionals, and those interested in career advice. Hosted by SPARS in cooperation with the AES Education Committee and G.A.N.G., career-related Q&A sessions will be offered to participants in a speed group mentoring format. A dozen students will interact with 4–5 working professionals in specific audio engineering fields or categories every 20 minutes. Audio engineering fields/categories include gaming, live sound/live recording, audio manufacturer, mastering, sound for picture, and studio production. Listed mentors are subject to change.

Saturday, Oct. 19 9:00 am Room 1E04
Technical Committee Meeting: Transmission and Broadcasting

Broadcast/Streaming Media Session 10
Saturday, Oct. 19 9:30 am – 11:00 am
Room 1E08

TECHNOLOGY AND STORYTELLING: HOW CAN WE BEST USE THE TOOLS AVAILABLE TO TELL OUR STORIES?

Presenters: **Butch D'Ambrosio**, Manual SFX
Robert Fass, Voice Talent
Bill Rogers, Voice Talent
David Shinn, SueMedia Productions, Carle Place, NY, USA
Sue Zizza, SueMedia Productions, Carle Place, NY, USA

This session will showcase three examples of how the choices we make around technology and the way we use it affect the storytelling process for all entertainment media. With on-site demonstrations by Sue Zizza and David Shinn of SueMedia Productions.

(1) *Microphones and the Voice in Storytelling*. Whether producing an audiobook or narration for a film or game, you want your talent to sound right for the story. This session will begin by looking at how we select microphones for voice talent. Two voice actors will demonstrate how working with different microphones affect their performance abilities.

(2) *Sound Effects: Studio vs. On Location Recordings*. Sound Effects enhance the storytelling process by helping to create location, specific action, emotion, and more. Do you have to create every sound effect needed for your project, or can you work with a combination of already recorded elements, alongside studio produced sound effects (foley), or on-location effects, and what are some tips and tricks to recording sound design elements?

(3) *Digital Editing and Mixing*. How can you better manage multiple voice, sound effect, and music elements into "stems," or sub-mixes for better control over final mixing as well as integrating plug-ins for mastering.

Knowledge Center
Saturday, October 19, 10:00 am – 11:30 am
Booth 2738

MULTITRACK RECORDING IN THE FIELD WITH JON TATOOLES OF SOUND DEVICES

Presenters: **Jon Tatooles**, Sound Devices
Pat McConnell, Sound Devices

Join our favorite Sound Devices duo, Jon Tatooles and Pat McConnell as they discuss the rise of unscripted dialogue sound production and the increasing demand for iso track as well as mix track deliverables. Let these field savvy mixers fill you with the tools and techniques for perfect delivery, every time.

Knowledge Center
Saturday, October 19, 10:00 am – 11:30 am
Room 1E06

PMC "MASTERS OF AUDIO": JIM ANDERSON, " 'SIXTEEN SUNSETS' JAZZ IN SURROUND WITH JANE IRA BLOOM"

Presenter: **Jim Anderson**, New York University, New York, NY, USA

Award winning soprano saxophonist Jane Ira Bloom has always had a special feeling for ballads. The album *Sixteen Sunsets* features nine tunes from the American songbook classics including: Gershwin's "I Loves You Porgy," Kern's "The Way You Look Tonight," Arlen's "Out of This World," Weill's "My Ship," Jimmy Van Heusen's "Darn That Dream," and Billie Holiday and Mal Waldron's "Left Alone," among others.

The album was recorded in 5.1 high-resolution Surround Sound at New York's famed Avatar Studio B by renowned engineer Jim Anderson who also co-produced and pushed the envelope of how a jazz quartet could sound using 5.1 recording techniques. JIB felt that "Surround" was a perfect match for the soprano sax because the sound doesn't emanate directly from the bell of the horn, it radiates out in all directions from the instrument in a more diffuse way. The saxophone was literally surrounded by a satellite array of mics for the sessions, and JIB's playing style was very well suited to the technique since she is always moving when playing.

Knowledge Center
Saturday, October 19, 10:00 am – 11:00 am
Room 1E03

THE MUSICAL IPAD

Presenters: **Vincent Leonard**, Invinceable Entertainment, Glen Mills, PA USA
Thomas Rudolph

Thousands of music apps—designed to assist you with every aspect of your life as a musician, hobbyist, student, or educator—are available for the iPad. Rudolph and Leonard guide you step by step through the most popular and productive musical apps for the iPad, demonstrating how to apply them in your musical life.

Saturday, Oct. 19 10:00 am Room 1E04
Technical Committee Meeting: High Resolution Audio

Game Audio Session 8 Saturday, Oct. 19
10:15 am – 11:15 am Room 1E10

AUDIO SHORTS

Presenters: **D. Chadd Portwine**, Vicarious Visions
Stephen Harwood Jr., Education Working Group Chair, IASIG, New York, NY, USA
Jason Kanter, Avalanche Studios, New York, NY, USA

This hour long session will be split into three twenty-minute segments. Each segment will go in depth into a subject that is near and dear to the presenter. Audio Shorts is designed to pack in as much usable information in as short of period of time as possible. It's like the Reader's Digest of game audio tutorials. You won't want to miss this one.

Shorty #1:
Follow the Sound of My Voice: A Localization Retrospective—Follow a VO line as it travels through the localization process for Skylanders: Swap Force. We see how a movie screenplay written in English becomes one-hundred and fifty thousand .wav files in more than ten languages. Screenshots from lip-sync, special-effects, surround sound, and game-mix projects will be viewed and discussed.

Shorty #2:
In-DAW Prototyping: WYSIWYG Approval and Delivery—Armed with video capture of gameplay, a content creator can develop sounds and music, as well as their in-game behavior, all without leaving the comfort of their favorite DAW. The workflow demonstrated will provide maximum protection against costly communication breakdowns, e.g., false-positive client approval and errant implementation.

Shorty #3: My Favorite Plugin!

Workshop 16 Saturday, Oct. 19
10:30 am – 12:00 noon Room 1E13

FX DESIGN PANEL: DISTORTION

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

Panelists: *Ken Bogdanowicz*, SoundToys
Marc Gallo, Studio Devil Virtual Tube Amplification, New York, NY, USA
Aaron Wishnick, iZotope, Somerville, MA, USA

Meet the designers whose talents and philosophies are reflected in the products they create, driving sound quality, ease of use, reliability, price, and all the other attributes that motivate us to patch, click, and tweak their effects processors.

Sound for Picture 4 Saturday, Oct. 19
10:30 am – 12:00 noon Room 1E11

MUSIC PRODUCTION FOR FILM
(SOUND FOR PICTURES MASTER CLASS)

Presenters: **Brian McCarty**, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia
Simon Franglen, Class1 Media, Los Angeles, CA, USA; London, UK
Chris Hajian

Film soundtracks contain three elements: dialog, music, and sound effects. The creation of a music soundtrack is far more complex than previously, now encompassing "temp music" for preview screenings, synthesizer-enhanced orchestra tracks, and other special techniques. This Master Class with one of Hollywood's leading professionals puts the process under the microscope.

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

Saturday, Oct. 19 10:30 am Room 1E02
Standards Committee Meeting: SC-04-03 Working Group on Loudspeaker Modeling and Measurement

Live Sound Seminar 9 Saturday, Oct. 19
11:00 am – 12:30 pm Room 1E12

ASSURING HIGH QUALITY SPEECH INTELLIGIBILITY FOR SPORTS EVENTS IN STADIUMS

Chair: **Renato Cipriano**, Walters-Storyk Design Group Brazil

Panelists: *Sergio Molho*, Walters-Storyk Design Group, Highland, NY, USA
John Storyk, Walters-Storyk Design Group, Highland, NY, USA

Establishing high quality speech intelligibility for sports events in stadiums requires a somewhat different mindset than that required for an optimum concert sound. However, stadiums frequently host both these types of events, and systems must be adaptable to both. Two key issues to consider are *Speech Intelligibility* (more critical than the frequency response of the system for sports events) and *Uniform Sound Coverage*, which is critical to meeting FIFA rules and regulations. For the past two+ years, WSDG Brazil has been working on three major stadium projects *simultaneously*, in preparation for the 2016 Olympics. A number of venerable older Brazilian arenas are currently undergoing substantial upgrades. Work on the Mineirão Stadium (1965) in Belo Horizonte was completed last year. It has already hosted Paul McCartney and will host the 2014 World Cup. Renovations on Independencia (1950), Brazil's largest stadium, also in Belo Horizonte are approaching completion. Renovations on the Maracanã Stadium (1950) in Rio de Janeiro will be completed later this year. WSDG Brazil is tasked with designing the acoustics and the complete audio and video systems for all three stadiums. This presentation will cover: analysis of requirements involved in the design process of stadium sound systems— including frequency response, target STL, and STI values, coverage (SPL distribution) zoning, architectural and structural integration, redundancy, etc. Also to be covered are: overall electroacoustical simulations and auralization for large arenas. Additionally, the presentation will address: sound system design for security (evacuation/public address announcements) including broadcast requirements and zoning/distribution issues that require special attention and drive design decisions.

Product Design Session 4 Saturday, Oct. 19
11:00 am – 12:30 pm Room 1E09

LOUDSPEAKER NONLINEAR IDENTIFICATION

Presenter: **Pascal Brunet**, Setem Technologies, Newbury, MA, USA

This presentation reviews recent developments in the domain of loudspeaker nonlinear identification and explores new possibilities to improve modeling that is a better match to the loudspeaker response. First we present the loudspeaker operation principles and the major causes of distortion, then we explore the successive modeling approaches that have been investigated in the last decades. Finally we provide new directions of research in the frequency domain and propose two techniques based on state-space for modeling of loudspeakers that can effectively be used in identification process.

Knowledge Center
Saturday, October 19, 11:00 am – 12:00 pm
Room 1E03

ABLETON GROOVES

Presenter: **Josh Bess**

Ableton Grooves empowers you to create realistic-sounding drum grooves using Ableton Live and the Ableton Grooves Drum Racks, specifically created by certified Ableton Live trainer and presenter Josh Bess. The concepts demonstrated by Bess become stepping-stones to a new way of thinking and creating while introducing diverse groove styles.

Knowledge Center
Saturday, October 19, 11:00 am – 12:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Saturday, Oct. 19 11:00 am Room 1E04
Technical Committee Meeting: Spatial Audio

Session EB4 Saturday, Oct. 19
11:30 am – 1:15 pm Room 1E07

APPLICATIONS IN AUDIO

Chair: **David Romblom**, McGill University, Montreal, Quebec, Canada; Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada

11:30 am

EB4-1 SyncAV—Workflow Tool for File-Based Video

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Shootings—*Andreas Fitza*, University of Applied Science Mainz, Mainz, Germany

The Sync-AV workflow tool eases the sorting and synchronization of video and audio footage without the need for expensive special hardware. It supports pre-production, shooting and post-production. It consists of these elements: a script-information and metadata-gathering web app that's connected to a server database; a local import client that manages the footage ingest and sorts the files together; the client also takes care of the synchronization of the video that contains audio and separately recorded audio files and it renames the files and implements the metadata; and the client uploads this synchronized preview files to our server so they can be shown at our web app. This e-Brief shows the current development and some specific solutions of Sync-AV.
Engineering Brief 118

11:45 am

EB4-2 Inconsistencies in the Practical Design and Measurement of Sound Systems in Reverberant Spaces Requiring a Minimum STI Standard—*David McNutt*, The McNutt Group, Chicago, IL, USA; Columbia College Chicago, Chicago, IL, USA

Minimum Speech Transmission Index measurement is now a requirement for Emergency Communication Systems as set forth in NFPA 72 2013 code. Professional audio design engineers have the greatest effect on potential intelligibility through their choice of the type, number, and distribution of loudspeakers and the power at which they are driven. Design Engineers often model sound systems for STI using EASE. Using this STI modeling approach can lead to varying results especially in reverberant sound fields. This brief discusses the conflicting results of three design/build projects in highly reverberant spaces in the Federal Plaza in Chicago.
Engineering Brief 119

12:00 noon

EB4-3 The Advantages of Using Active Crossovers in High-End Wireless Speakers—*David Jones*, CSR Limited - Manchester, UK

With the availability of standardized wireless interfaces and high performance codecs, wireless speakers can be designed that suit the consumer demands of compactness and ease of use. This paper will examine the performance benefits of using active crossovers and digital equalization in an amplification subsystem based on a high performance digital input switching amplifier. Measurements of distortion and damping factors will be compared in an example signal chain and the influence these parameters have on the perceived audio quality of the speaker system will be discussed.
Engineering Brief 85 [Previously presented at the 134th Convention in Rome]

12:15 pm

EB4-4 Low Latency Replacement of ISDN and 4-Wire for Remote Broadcasts—*Anthony Faust*, Atlantic Post Production, Toronto, ON, Canada

Integrated Services Digital Network (ISDN) lines are being replaced by other forms of Internet Protocol (IP) connectivity for high-quality remote broadcasts. In particular, the use of bonding diversity (diversity) over multiple public Internet networks for remote broadcasts has been proven in challenging environments with excellent results. This high-performance of this approach makes it likely to become the standard for remote broadcasts.
Engineering Brief 120

12:30 pm

EB4-5 Design and Construction of the Stringer: A Polyphonic Signal Switcher for 13-pin DIN M—*Michael Palumbo*,[1] *Pouya Hamidi*,[2] *Richard King*,[2,3] *Donald Pavlasek*[2]
[1]Concordia University, Montreal, Quebec, Canada
[2]McGill University, Montreal, Quebec, Canada
[3]The Centre for Interdisciplinary Research in Music Media and Technology, Montreal, Quebec, Canada

The Stringer is a polyphonic signal switcher for use with 13-pin DIN MIDI guitar pickups. Used as an intermediary between a guitar and a synthesizer pedal, the purpose of the device is to isolate a single string for monophony, such as bass note accompaniment. A height-and-depth-adjustable fulcrum bar supports the performer's feet, and brings them closer to the foot switches, allowing for smoother and faster string switching. The current model can isolate strings 6, 5, and 4; mute all strings; and can run in bypass mode to pass all string signals through for standard operation. The circuit is powered by a 9V DC external adapter, and housed in a custom aluminum chassis.
Engineering Brief 121

12:45 pm

EB4-6 Design of a Sound Reinforcement System for Koerner Hall—*Jeffery Bamford*, Engineering Harmonics Inc., Toronto, ON, Canada

Built over three years, the 1,135-seat Koerner Hall is the jewel of the new TELUS Centre for Performance and Learning at the Royal Conservatory of Music in Toronto, Canada. Since its opening in September 2009, Koerner Hall's beautiful design, flexible performance characteristics and superb acoustics have been praised by critics and performers alike. The hall achieved the highest possible acoustic rating—N1 —rendering it ideal for the finest acoustical performances of classical music, jazz, and world music. The incorporation of variable acoustics makes it equally well suited to amplified music, lectures, and film presentations. This Engineering Brief will review the process and design of the sound reinforcement system. It features an innovative and almost invisible "voice-stick" to maximize intelligibility, rather than sound reinforcement. The system must provide coverage for the audience as to performers on and around the stage in an extremely intimate venue. Testing the design with a computer and mock-up will also be discussed.
Engineering Brief 122

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1:00 pm

EB4-7 Consistently Stable Loudspeaker Measurements Using a Tetrahedral Enclosure—*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. A system is proposed that offers a unique and stable test environment giving an opportunity to standardize and compare results between measurement sites. It works by having an enclosure shape that eliminates standing-waves and having acoustic foam to eliminate any remaining high frequencies. It then rigidly defines the measurement geometry together with interchangeable sub baffles, ensuring rapid and accurate change over and repeatable measurements. So that with several in use in the design, production and customer chain results will be comparable unit to unit throughout the world to an unprecedented degree.
Engineering Brief 123

Student/Career Development Event
EDUCATION AND CAREER/JOB FAIR
Saturday, October 19, 11:00 am – 12:30 pm
1E Foyer

The combined AES 135th 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 135th 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.

Project Studio Expo
Saturday, October 19, 11:00 am – 12:00 noon
Stage

KEEPING THE HUMAN ELEMENT IN THE DIGITAL AGE: WAYS TO KEEP MUSIC SOUNDING ALIVE AND INTERESTING

Presenter: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

It's vital to keep the "art" in "state of the art." This workshop starts off by examining how the brain responds to music and the ways music was made in the 50s and 60s compared to how music is being made today—what we've gained and what we've lost. The discussion then segues into practical ways to retain a human quality in both live performance and studio recordings, including ways to make modeling software sound more "organic," how to use techniques like snapping to the grid and pitch correction in ways that don't compromise the music's humanity, superior methods of quantization, using control surfaces to turn mixes into performances (not just static changes in level), and more. If you want to show technology who's the boss, this workshop is a must.

Tutorial 13 **Saturday, Oct. 19**
11:15 am – 12:45 pm **Room 1E14**

A HOLISTIC APPROACH TO CROSSOVER SYSTEMS AND EQUALIZATION FOR LOUDSPEAKERS (A MASTER CLASS EVENT)

Presenter: **Malcolm O. J. Hawksford**, University of Essex, Colchester, Essex, UK

Loudspeaker systems employ crossover filters and equalization to optimize their performance in the presence of electroacoustic transducers limitations and associated loudspeaker enclosures. This Master Class will discuss both analog and digital techniques and include examples of constant-voltage, all-pass, and constant-delay crossover alignments together with the constraints imposed by the choice of signal processing. The meaning of "minimum-phase" will be described including its linkage to causality and digital equalization strategies presented that emphasize the importance of loudspeaker impulse response decomposition into minimum-phase and excess-phase transfer functions. The session will include demonstrations on minimum-phase response derivation from a magnitude-frequency response and on the audibility of pure phase distortion to justify the use of the Linkwitz-Riley 4th-order class of analog crossover alignment.

Workshop 17 **Saturday, Oct. 19**
11:30 am – 1:00 pm **Room 1E08**

HOW ARE WE LEARNING MASTERING: TEACHING MASTERING—THE NEXT WAVE

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

Panelists: *Scott Hull*, Masterdisk, New York, NY, USA
Mike Wells, Mike Wells Mastering, Los Angeles, CA, USA

Traditionally mastering has been learned by apprenticing. Now with the proliferation of educational resources and the evolution of affordable high quality in-the-box processing, more people are practicing mastering in more places than ever before. Teaching a young engineer to become a top flight mastering engineer can be challenging.

Have you wondered: What does "Experienced Mastering Engineer" mean? What's the secret of mastering? In this workshop seasoned mastering engineers and educators discuss how the craft is being taught and learned and how the next generation of mastering engineers will learn from their contemporaries. Topics will include what time tested practices remain essential and what is new in the discipline of mastering.

Attendees of this workshop will walk away with a clearer understanding of what it takes to thrive in today's mastering market, how to assess internship/mentorship over "going solo" early in a mastering career, and how to grow/build your mastering skills in today's market.

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

Game Audio Session 9 **Saturday, Oct. 19**
11:30 am – 1:00 pm **Room 1E10**

AUDIO ON WEB—OVERVIEW AND APPLICATION

Presenter: **Jan Linden**, Google, Mountain View, CA, USA
Jory K. Prum, studio.jory.org, Fairfax, CA, USA

In the stampede to replace proprietary web browser plug-ins with a patchwork of open standards collectively known as HTML5, audio was once the largest gap in capability. In the past 18 months, however, great strides have been made to close this gap: browser support is nearly ubiquitous (with only one major hold-out), the standards body is marching toward completion of the first publication of the Web Audio API, and progress is being made in drafting and implementing the Web MIDI API, too. Developers are clearly excited, as interesting and advanced uses of the technology have been plentiful. This session will take a look at where we've come since last year's AES, discussing browser and codec support, shining a spotlight on a number of examples from developers across the globe (Infinite Jukebox, Step Daddy, BBC Radiophonic Workshop, Chrome Racer), take a look at how easy it is to work with the Web Audio API to implement sound within a web browser, and explore a few of the many libraries developers have created to make implementation even easier (Gibberish, Tuna, component.fm).

Special Event
PLATINUM PRODUCERS
Saturday, October 19, 11:30 am – 1:30 pm
Room 1E15/16

Moderator: **David Weiss**

Panelists: *Jeff Jones*
Dano "ROBOPOP" Omelio
Dave Tozer

The musical continuum, and its role in music production, comes into focus at this year's Platinum Producers Panel. How does an understanding of music's past, present, and future serve the producer in their quest to fully realize the artist's vision? We'll go deep with this elite panel of Jeff Jones (Eric Clapton, Wynton Marsalis, Norah Jones), ROBOPOP (Gym Class Heroes, Maroon 5, Lana Del Ray), and Dave Tozer (John Legend, Kanye West, Justin Timberlake), and moderated by David Weiss (Founder/Editor of SonicScoop). Their collective experience spans decades and has produced hit singles and albums in rock, R&B, hip-hop, pop, jazz, and beyond. From their application of classic techniques to late-breaking revelations, this trio of hit makers will provide inside information on tracking, mixing, mastering, and getting the very best out of artists in the studio.

Knowledge Center
Saturday, October 19, 11:30 am – 1:00 pm
Room 1E06

PMC "MASTERS OF AUDIO": MICHAEL BRAUER

Presenter: **Michael Brauer**

In the Mix with Q&A

Multi-Grammy winning engineer Michael Brauer is a New York-based mix engineer whose credits encompass a wide range of genres and include The Rolling Stones, Bob Dylan, Paul McCartney, Coldplay, John Mayer, Ash, My Morning Jacket, Ben Folds, Dream Theater, The New Radicals, Change, Fountains of Wayne, David Poe, Wilco, Alpha Rev, and Ron Sexsmith.

Workshops with Height at NYU
3-D AUDIO ENVIRONMENTS AT NYU STEINHARDT
Saturday, October 19, 12:00 noon – 5:00 pm

This event has a limited capacity. Tickets will be allocated on a first-come first-served basis. To participate, please sign up at the AES Tours Desk.

Chair: **Paul Geluso**

Presenters: *Malgorzata Albinska-Frank*
Tom Ammerman
Tom Beyer
David Bowles
Jonathan Hong
Lasse Nipkow
Agnieszka Roginska
Bert Van Daele
Gregor Zielinsky

The Music Technology program at NYU Steinhardt will host a multi-presentation 3D audio listening session. Visitors will experience music, film sound, live concert, and environmental sound recordings produced specifically for 3D listening environments. There will be 10 concurrent demo sessions at the NYU Steinhardt studios. In the James L. Dolan Music Recording Studio control room, recordings made by the faculty and students of NYU Music Technology and McGill University will be presented in 3D on a 10.2 speaker system. In the studio live room, music and film/video sound tracks created specifically for AURO 3D will be presented in 9.1 and discussed by the creators of the content. In the NYU Sound Research Lab, environmental sound captured at many locations around NYC using Ambisonic recording techniques as part of the Center for Urban Science and Progress (CUSP) research project will be presented on a 16.2 speaker system. In addition, a Telematic music performance featuring musicians at NYU and at a remote site will be taking place live. Viewers will be immersed in the sound from the remote location mixed with the sound of the on-site live musicians.

Demo Sessions:

- *Recording with Height*, David Bowles and Paul Geluso
- *Immersive Urban Soundscapes—I Hear NY3D*, Agnieszka Roginska
- *Immersive Distributive Performance*, Tom Beyer
- *The New Creative Dimension of Immersive Sound*, Bert Van Daele
- *Properties of Auro 3D Signals*, Lasse Nipkow
- *Making of: 3D Audio at the David Bowie Worldwide Exhibition*, Gregor Zielinsky
- *Bernstein in 3D*, Gregor Zielinsky
- *Alps in Auro 3D*, Malgorzata Albinska-Frank
- *Experience the Sound of the Future*, Tom Ammerman
- *Mixing with Space Builder—3D Reverberation Tool*, Jonathan Hong

Tutorial 14 **Saturday, Oct. 19**
12:00 noon – 1:00 pm **Room 1E13**

WHY FACILITIES NEED TECH MANAGERS AND HOW TO BE ONE!

Presenter: **Eric Wenocur**, Lab Tech Systems, Silver Spring, MD, USA

This tutorial describes the role of a Technical Manager and discusses the need for Technical Management at facilities of any size—but especially at small shops where people wear many hats. Good and bad experiences from the field will help to emphasize why this matters and approaches for handling Tech Manager duties will be suggested. The perspective is from someone who designs and builds facilities, interacts with managers and operators, and sees what happens when this role is left to chance!

Project Studio Expo
Saturday, October 19, 12:00 noon – 1:00 pm
Stage

IT WON'T SOUND RIGHT IF YOU DON'T HEAR IT RIGHT: STUDIO ACOUSTICS, MONITORING AND CRITICAL LISTENING

Presenters: **Hugh Robjohns**, Technical Editor, Sound on Sound, Cambridge, UK
Paul White

The monitoring environment acoustics and the monitoring loudspeakers are critical links in every music production chain. Any weaknesses impact negatively not only on the overall quality of mixes, but also on the confidence and ability of the user to assess and process audio material efficiently and effectively. This workshop examines the theoretical requirements and practical optimization of high-quality monitoring systems for home and project studios, drawing on the author's experiences in the "Studio SOS" series published in Sound On Sound magazine. It will also explore choosing new monitoring loudspeakers, optimizing control room acoustics, and honing critical listening skills.

Saturday, Oct. 19 **12:00 noon** **Room 1E02**
Standards Committee Meeting: SC-04-04 Working Group on Microphone Measurement and Characterization

Historical Event
RESTORING PEGGY LEE'S CAPITOL RECORDS ALBUM "JUMP FOR JOY"
Saturday, October 19, 12:15 pm – 1:15 pm
Room 1E11

Presenter: **Alan Silverman**, Arf! Mastering, New York, NY, USA; NYU/Steinhardt Dept. of Music Technology

"Jump for Joy," featuring Peggy Lee and arranged by Nelson Riddle, was one of the first records released by Capitol as a stereo LP. The year was 1959, the year the label first made stereo LPs available to the public, but this seminal album was never released in stereo on CD, only in mono. An assignment to master the original stereo mixes for digital release led to the discovery of a 54-year old audio mystery. Had something gone awry at the original stereo mix date? This special event uses photos and high-resolution transfers of original session material to detail a surprising finding and the steps that were taken to reach back in time to restore the album for today's audience as it was intended to heard.

Workshop 18 **Saturday, Oct. 19**
12:30 pm – 1:30 pm **Room 1E09**

THE CLOUD-CONNECTED FUTURE OF MEDIA CREATION

Chair: **Jay LeBoeuf**, iZotope, San Francisco, CA, USA

Panelists: *Tristan Jehan*, The Echo Nest, Somerville, MA, USA
Chris Kantrowitz, Gobbler, Hollywood, CA, USA
Charles Van Winkle, Adobe

Thanks to a world full of mobile devices, innovative algorithms, and cloud computing, we are seeing a massive democratization of the media creation and production process. Innovative mobile applications, web services, and cloud-based audio sharing sites are turning virtually everyone into a content producer. In this session leading commercial technologists will debate the future. We will discuss quantitative and qualitative trends and the obstacles to media creation in the cloud.

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

Knowledge Center
Saturday, October 19, 12:30 pm – 2:30 pm
Booth 2738

GOTHAM SOUND PRESENTS MARTIN KELLY: MASTERING MULTITRACK AUDIO USING NETWORK BASED SOLUTIONS ON SPIKE TV'S INK MASTER

Presenter: **Martin Kelly**, Pine Bush, NY, USA

Join Martin Kelly for a fascinating presentation on custom solutions for the technical challenges he faced as the audio supervisor on Spike TV's "Ink Master."

Competition reality television shows have especially complex audio requirements. Even in the best of circumstances, the sound department must provide multiple mixes in real time, all with pristine audio, with no retakes. On "Ink Master," the audio department faces the additional dilemma of noisy tattoo guns and a massive number of microphone channels for contestants, judges, and the human canvases.

Sound Supervisor Martin Kelly discusses the details of the design and implementation of "Ink Master's" production sound mixing and recording systems. Dante network technology was used to connect and integrate mixers and recorders from multiple manufacturers including Yamaha, Lectrosonics, JoeCo, Sound Devices, and Boom Recorder.

After the discussion, a custom touch screen audio control surface created to allow an operator to route the 40 inputs to up to 8 camera mixes on the fly will be demonstrated followed by a Q&A session.

Saturday, Oct. 19 **12:30 pm** **Room 1E04**
Technical Committee Meeting: Sound for Digital Cinema and Television

Student/Career Development Event
STUDENT RECORDING CRITIQUES
Saturday, October 19, 1:00 pm – 2:15 pm
Room 1E06

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

Students! Bring your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on CD, DVD, memory-stick, or hard disk, as clearly labeled 44.1 KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers!

Project Studio Expo
Saturday, October 19, 1:00 pm – 2:00 pm
Stage

TAKE YOUR STUDIO ON STAGE: LIVE PERFORMANCE WITH LAPTOPS, LOOPING PEDALS & OTHER STUDIO TECH

Presenter: **Craig Anderton**, Harmony Central / Electronic Musician, Santa Fe, NM, USA

For many musicians, as well as DJs and electronic acts, a 21st century live performance requires much more than just a mixer and a bunch of amps. This workshop takes a practical look at how to use technology on stage without being overwhelmed by it, ways to insure a smooth performance, and includes invaluable information on the "care and feeding" of laptops to insure optimum performance—and uninterrupted performances. Other topics include using controllers for a more vibrant live performance, performing with Ableton Live and dedicated control surfaces, improvisation with looping pedals and DAW software, and the evolution of DJ controller/ laptop combinations into tools for a musical, complex new art form.

Knowledge Center
Saturday, October 19, 1:00 pm – 2:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Knowledge Center
Saturday, October 19, 1:00 pm – 2:00 pm
Room 1E03

IZOTOPE TIPS FROM A PRO: LIVE! SOUND DESIGN FOR TRAILER MUSIC

Presenter: **Anthony Baldino**, Composer, Sound Designer, Sencit Music

Chances are, you've heard the work of Anthony Baldino in some of this summer's blockbuster movie trailers (*Star Trek: Into Darkness, Zero Dark Thirty*) or in your favorite video games (Tom Clancy: Splinter Cell Blacklist). Touching on topics such as sample sourcing, emerging technologies, and effects processing, this presentation aims to prompt even more creativity and ingenuity in sound design for a variety of applications, from sound sourcing to composition techniques.

Tutorial 15 **Saturday, Oct. 19**
1:30 pm – 3:00 pm **Room 1E14**

THE ART OF DRUM PROGRAMMING

Presenter: **Justin Paterson**, London College of Music, University of West London, London, UK

Drum programming has often faced boundaries in terms of how effectively it could address the complexities of certain genres. This tutorial will explore and push some of these boundaries as implemented in contemporary professional practice, showing contrasting techniques used in the creation of both human emulation and the unashamedly synthetic. Alongside this, many of the studio production techniques often used to enhance such work will be discussed, ranging from dynamic processing to intricate automation. The session will include numerous live demonstrations covering a range of approaches. Although introducing all key concepts from scratch, its range and hybridization should provide inspiration even for experienced practitioners.

Broadcast/Streaming Media Session 11
Saturday, Oct. 19 **1:30 pm – 3:00 pm**
Room 1E08

MAINTENANCE, REPAIR, AND TROUBLESHOOTING

Chair: **John Bisset**, Telos Alliance

Panelists: *Michael Azzarello*, CBS
Bill Sacks, Orban / Optimod Refurbishing, Hollywood, MD, USA
Kimberly Sacks, Optimod Refurbishing, Hollywood, MD, USA

Much of today's audio equipment may be categorized as "consumer, throw-away" gear, or so complex that factory assistance is required for a board or module swap. The art of Maintenance, Repair, and Troubleshooting is actually as important as ever, even as the areas of focus may be changing. This session brings together some of the sharpest troubleshooters in the audio business. They'll share their secrets to finding problems, fixing them, and working to ensure they don't happen again. We'll delve into troubleshooting on the systems level, module level, and the component level, and explain some guiding principles that top engineers share.

Special Event
IRONS IN THE FIRE: CAREER AND BUSINESS DEVELOPMENT MENTORING WITH THE MANHATTAN PRODUCERS ALLIANCE
Saturday, October 19, 1:30 pm – 2:45 pm
Room 1E13

Moderator: **Joe Carroll**

Presenters: *John Blair*
Chuck Callahan
Steve Horowitz
John Kiehl
Andrew Schwartz

Bring your energy, enthusiasm, business ideas, and questions. At this event the focus is on YOU!

Succeeding in music today is, more than ever, challenging. Members of the Manhattan Producers Alliance will give a brief talk about developing your brand and your business and functioning as a creative talent in an ever-changing music business. Take this unique opportunity to meet some ManhatPro members and spend some time learning some tips and tricks for business development. You'll participate in our open discussions, discuss your personal career goals one on one, and get a chance to meet some ManhatPro members.

Sound for Picture 5 Saturday, Oct. 19
2:00 pm – 3:30 pm Room 1E11

SOUND DESIGN FOR FILM
(SOUND FOR PICTURES MASTER CLASS)

Presenters: **Michael Barry**
Brian McCarty, Coral Sea Studios Pty. Ltd., Clifton Beach, QLD, Australia
Eugene Gearty
Skip Lievsay

Film soundtracks contain three elements: dialog, music, and sound effects. Sound effects, which used to be an afterthought, are now constructed by sound designers, often working from the start of production. This Master Class with Hollywood's leading professionals puts the process under the microscope.

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

Project Studio Expo
Saturday, October 19, 2:00 pm – 3:00 pm
Stage

HOW TO CREATE, PRODUCE, AND DISTRIBUTE YOUR MUSIC COMPLETELY IN THE CLOUD

Presenter: **Andrea Pejrolo**, Berklee College of Music, Boston, MA, USA

Learn how to create and produce your music through real-time and off-line long distance collaborations using exclusively the cloud and your laptop or tablet. Through practical examples and scenarios you will learn how to:

• Set up your production environment for long distance collaborations and recording

• Choose the right hardware and software for collaborating in the cloud

• Choose the best strategies and social tools to find trustworthy creative musicians for your sessions

• Set in place the right workflow for efficient and productive recording sessions in the cloud

• Take advantage of creative talent around the world to add an original touch to your music production

Knowledge Center
Saturday, October 19, 2:00 pm – 3:00 pm
Room 1E03

INTELLIGENT COMPRESSION IN THE ANALOG DOMAIN

Presenter: **Michael Deming**, CharterOak Acoustic Devices

Utilizing a unique control circuit, a totally new approach to AGC, and entirely discrete electronics, the CharterOak SCL-1 provides completely artifact free compression. The device achieves this through waveform differentiation and integration. The SCL-1 employs a rectifier circuit that has a parabolic average charge curve. The intent of the circuit is to provide fast releases of rhythmic and staccato peaks, and longer releases of legato notes, within the user established range of dynamic compression, which release to a continually changing average level or sustained music, which is determined by the parabolic charge curve of the storage capacitors.

Game Audio Session 10 Saturday, Oct. 19
2:15 pm – 3:15 pm Room 1E10

GAME AUDIO BREAKTHROUGHS FOR HTML5 AND MOBILE

Presenters: **Garrett Nantz**, Luxurious Animals

The common practice is that once game development is almost complete, sound design just gets added. We will show you a better way to use sound design at the beginning of a project as a ideation tool to inform the design and development of games.

Using the award-winning Lux Ahoy www.luxahoy.com game, we will take a behind the scenes look at the process to create audio experiences for HTML5 and Android. Topics covered will include audio workflows, tricks and techniques to combat platform and browser sound issues, creating memorable sound effects, binaural 3-D sound, audio loop creation, music sourcing, and coding libraries.

Saturday, Oct. 19 2:15 pm Room 1E04
Historical Committee Meeting

Session P14 Saturday, Oct. 19
2:30 pm – 6:00 pm Room 1E07

TRANSDUCERS—PART 2: HEADPHONES AND LOUDSPEAKERS

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

2:30 pm

P14-1 Application of Matrix Analysis to Identification of Mechanical and Acoustical Parameters of Compression Drivers—*Alexander Voishvillo*, JBL/Harman Professional, Northridge, CA, USA

In previous work of the author, special measurement methods were used to obtain the transfer matrices of compression drivers. This data was coupled with the results of the FEA simulations of horns. It made it possible to simulate the frequency amplitude and directivity responses of horn drivers without building actual physical horns. In this work, a different set of measure-

ments is used to obtain the transfer matrix of a vibrating diaphragm. This approach results in a more detailed and flexible method to analyze and design compression drivers. Other parameters used in the identification process are the electrical parameters of the motor and the acoustical parameters of compression chamber and phasing plug. The method was used in design and optimization of the new JBL dual-diaphragm compression driver to be used in a new JBL line array system.
Convention Paper 8988

3:00 pm

P14-2 Application of Static and Dynamic Magnetic Finite Elements Analysis to Design and Optimization of Transducers Moving Coil Motors—*Alexander Voishvillo, Felix Kochendörfer*, JBL/Harman Professional, Northridge, CA USA

Transducer motors are a potential source of nonlinear distortion. There are several nonlinear mechanisms that generate nonlinear distortion in motors. Typical loudspeaker nonlinear models include the dependence of the *Bl*-product and the voice coil inductance L_{VC} on the voice coil position and current. These effects cause nonlinearity in the driving force, electrodynamic damping, and generate nonlinear flux modulation and reluctance force. In reality, the voice coil inductance and resistive losses depend also on frequency. To take these effects into account the so-called LR-2 impedance model is used. The L_2 and R_2 elements are nonlinear functions of the voice coil position and current. In this work detailed analysis of a nonlinear model incorporating these elements is performed. The developed approach is illustrated by the FEA-based design and optimization of a new JBL ultra-linear transducer to be used in a new line array system.
Convention Paper 8989

3:30 pm

P14-3 End-of-Line Test Concepts to Achieve and Maintain Yield and Quality in High Volume Loudspeaker Production—*Gregor Schmidle*, NTi Audio AG, Schaan, Liechtenstein

Managing high volume, multiple line, and location loudspeaker production is a challenging task that requires interdisciplinary skills. This paper offers concepts for designing and maintaining end-of-line test systems that help to achieve and maintain consistent yield and quality. Topics covered include acoustic and electric test parameter selection, mechanical test jig design, limit finding strategies, fault-tolerant workflow creation, test system calibration and environmental influence handling as well as utilizing statistics and statistic process control.
Convention Paper 8990

4:00 pm

P14-4 Advances in Impedance Measurement of Loudspeakers and Headphones—
Steve Temme,[1] *Tony Scott*[2]
[1]Listen, Inc., Boston, MA, USA
[2]Octave Labs, LLC, Eastchester, NY, USA

Impedance measurement is often the sole electrical measurement in a battery of QC tests on loudspeakers and headphones. Two test methods are commonly used—single channel and dual channel. Dual Channel measurement offers greater accuracy as both the voltage across the speaker (or headphone) and the reference resistor are measured to calculate the impedance. Single Channel measurement methods are more commonly used on the production line because it only requires one channel of a stereo soundcard, which leaves the other free for simultaneous acoustic tests. They are less accurate, however, due to the test methods making assumptions of constant voltage or constant current. In this paper we discuss a novel electrical circuit that offers similar impedance measurement accuracy compared to complex dual channel measurement methods but using just one channel. This is expected to become popular for high throughput production line measurements where only one channel is available as the second channel of the typical soundcard is being used for simultaneous acoustic tests.
Convention Paper 8991

4:30 pm

P14-5 Auralization of Signal Distortion in Audio Systems —Part 1: Generic Modeling—
Wolfgang Klippel, Klippel GmbH, Dresden, Germany

Auralization techniques are developed for generating a virtual output signal of an audio system where the different kinds of signal distortion are separately enhanced or attenuated to evaluate the impact on sound quality by systematic listening or perceptive modeling. The generation of linear, regular nonlinear and irregular nonlinear distortion components is discussed to select suitable models and measurements for the auralization of each component. New methods are presented for the auralization of irregular distortion generated by defects (e.g., rub & buzz) where no physical models are available. The auralization of signal distortion is a powerful tool for defining the target performance of an audio product in marketing, developing products at optimal performance-cost ratio and for ensuring sufficient quality in manufacturing.
Convention Paper 8992

5:00 pm

P14-6 Free Plus Diffuse Sound Field Target Earphone Response Derived from Classical Room Acoustics Theory—*Christopher Struck*, CJS Labs, San Francisco, CA, USA

The typical standardized free or diffuse field reference or target earphone responses in general represent boundary conditions rather than a realistic listening situation. Therefore a model using classical room acoustics is introduced to derive a more realistic target earphone response in a direct plus diffuse sound field. The insertion gain concept as applied to earphone response measurements using an ear simulator equipped test manikin is detailed in order to appropriately apply the model output to a typical earphone design. Data for multiple sound sources, multiple ➥

rooms, and variants of the direct 0° on-axis free field response are shown. Limits of the method are discussed and the results are compared to the well-known free and diffuse field responses.
Convention Paper 8993

5:30 pm

P14-7 Listener Preference for In-Room Loudspeaker and Headphone Target Responses—*Sean Olive, Todd Welti, Elisabeth McMullin*, Harman International, Northridge, CA USA

Based on preference, listeners adjusted the relative bass and treble levels of three music programs reproduced through a high quality stereo loudspeaker system equalized to a flat in-room target response. The same task was repeated using a high quality circumaural headphone equalized to match the flat in-room loudspeaker response as measured at the eardrum reference point (DRP). The results show that listeners on average preferred an in-room loudspeaker target response that had 2 dB more bass and treble compared to the preferred headphone target response. There were significant variations in the preferred bass and treble levels due to differences in individual taste and listener training.
Convention Paper 8994

Live Sound Seminar 10 **Saturday, Oct. 19**
2:30 pm – 4:30 pm **Room 1E12**

PRODUCTION WIRELESS SYSTEMS: AN EXAMINATION OF ANTENNAS, COAX, FILTERS, AND OTHER TIPS AND TRICKS FROM THE EXPERTS

Chair: **James Stoffo**, Independent Frequency Coordinator

Panelists: *Brooks Schroeder*, Frequency Coordination Group, Orlando, FL, USA
Vinny Siniscal, Firehouse Productions, Red Hook, NY, USA
Ed Weizcerak, Freelance, New York, NY, USA

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

Special Event
GRAMMY SOUNDTABLES
Saturday, October 19, 2:30 pm – 4:00 pm
Room 1E15/16

Presenters: **Jim Boyer**
Peter Chaikin
Jill Dell'Abate
Mark Ethier
Frank Filipetti
Jimmy Jam
Leslie Ann Jones
Bob Ludwig
Rob Mathes
BJ Ramone
Elliot Scheiner
Al Schmitt

What Would Ramone Do?

This educational and inspirational career retrospective will delve into the music, creativity, and vision of legendary 14-time GRAMMY Award winning producer/engineer/technologist Phil Ramone. From Marilyn Monroe's performance/rendition of "Happy Birthday" for JFK, Getz/Gilberto's "Girl From Ipanema," Billy Joel's "Just The Way You Are," Paul Simon's "50 Ways to Leave Your Lover," Frank Sinatra's *Duets* album and live concerts in Italy with Luciano Pavarotti, to overseeing groundbreaking sound evolutions for the GRAMMY Awards Telecast, Phil Ramone's career spanned more than 50 years of artistic and technical innovation. For this retrospective, we'll go behind the scenes with colleagues, footage, and friends for an analysis of the wisdom and knowledge behind his achievements. This session is guaranteed to be insightful and thought-provoking.

Knowledge Center
Saturday, October 19, 2:30 pm – 3:30 pm
Room 1E06

PMC "MASTERS OF AUDIO": A JOE FERLA RETROSPECTIVE WITH SPARS

Presenter: **Joe Ferla**, Joe Ferla, Stamford, NY, USA

Joe Ferla, a five-time Grammy Award recipient and renowned engineer to some of the best musicians in the industry, will be presenting tracks for listening from his extensive discography. After each track is played he will answer any questions regarding the track such as how he recorded the artist, which mics he used on what instrument, etc. Songs from artists such as Roberta Flack (that started his career off), David Sanborn, John Scofield, Eliane Elias, Dave Douglas, Christian McBride, Charlie Hunter and others will be featured in this unique listening experience.

Session P15 **Saturday, Oct. 19**
3:00 pm – 4:30 pm **1E Foyer**

POSTERS: APPLICATIONS IN AUDIO—PART I

3:00 pm

P15-1 An Audio Game App Using Interactive Movement Sonification for Targeted Posture Control—*Daniel Avissar, Colby N. Leider, Christopher Bennett, Robert Gailey*, University of Miami, Coral Gables, FL, USA

Interactive movement sonification has been gaining validity as a technique for biofeedback and auditory data mining in research and development for gaming, sports, and physiotherapy. Naturally, the harvesting of kinematic data over recent years has been a function of an increased availability of more portable, high-precision sensory technologies, such as smart phones, and dynamic real time programming environments, such as *Max/MSP*. Whereas the overlap of motor skill coordination and acoustic events has been a staple to musical pedagogy, musicians and music engineers have been surprisingly less involved than biomechanical, electrical, and computer engineers in research efforts in these fields. Thus, this paper proposes a prototype for an accessible virtual gaming interface that uses

music and pitch training as positive reinforcement in the accomplishment of target postures.
Convention Paper 8995

3:00 pm

P15-2 Evaluation of the SMPTE X-Curve Based on a Survey of Re-Recording Mixers—*Linda A. Gedemer*, University of Salford, Salford, UK; Haman International, Northridge, CA, USA

Cinema calibration methods, which include targeted equalization curves for both dub stages and cinemas, are currently used to ensure an accurate translation of a film's sound track from dub stage to cinema. In recent years, there has been an effort to re-examine how cinemas and dub-stages are calibrated with respect to preferred or standardized room response curves. Most notable is the work currently underway reviewing the SMPTE standard ST202:2010 "For Motion-Pictures – Dubbing Stages (Mixing Rooms), Screening Rooms and Indoor Theaters – B-Chain Electroacoustic Response." There are both scientific and anecdotal reasons to question the effectiveness of the SMPTE standard in its current form. A survey of re-recording mixers was undertaken in an effort to better understand the efficaciousness of the SMPTE standard from the users' point of view.
Convention Paper 8996

3:00 pm

P15-3 An Objective Comparison of Stereo Recording Techniques through the Use of Subjective Listener Preference Ratings—*Wei Lim*, University of Michigan, Ann Arbor, MI, USA

Stereo microphone techniques offer audio engineers the ability to capture a soundscape that approximates how one might hear realistically. To illustrate the differences between six common stereo microphone techniques, namely XY, Blumlein, ORTF, NOS, AB, and Faulkner, twelve study participants were asked to rate recordings of a Yamaha Disklavier piano. I examined the interrating correlation between subjects to find a preferential trend toward near-coincidental techniques. Further evaluation showed that there was a preference for clarity over spatial content in a recording. Subjects did not find that wider microphone placements provided for more spacious-sounding recordings. Using this information, this paper also discusses the need to re-evaluate how microphone techniques are typically categorized by distance between microphones.
Convention Paper 8997

3:00 pm

P15-4 Tampering Detection of Digital Recordings Using Electric Network Frequency and Phase Angle—*Jidong Chai*,[1] *Yuming Liu*,[2] *Zhiyong Yuan*,[3] *Richard W. Conners*,[4] *Yilu Liu*[1]
[1]University of Tennessee, Knoxville, TN, USA
[2]Electrical Power Research Institute, Chongqing Electric Power Corp., Chongqing, China
[3]China Southern Power Grid, Guangzhou, China
[4]Virginia Polytechnic Institute and State University, Blacksburg, VA, USA

In the field of forensic authentication of digital audio recordings, the ENF (electric network frequency) Criterion is one of the possible tools and has shown promising results. An important task for forensic authentication is to determine whether the recordings are tampered or not. Previous work performs tampering detection by looking for the discontinuity in either the extracted ENF or phase angle from digital recordings. However, using only frequency or phase angle to detect tampering may not be sufficient. In this paper both frequency and phase angle with a corresponding reference database are used to do tampering detection of digital recordings, which result in more reliable detection. This paper briefly introduces the Frequency Monitoring Network (FNET) at UTK and its frequency and phase angle reference database. A Short-Time Fourier transform (STFT) is employed to estimate the ENF and phase angle embedded in audio files. A procedure of using the ENF criterion to detect tampering, ranging from signal preprocessing, ENF and phase angle estimation, frequency database matching to tampering detection, is proposed. Results show that utilizing frequency and phase angle jointly can improve the reliability of tampering detection in authentication of digital recordings.
Convention Paper 8998

3:00 pm

P15-5 Portable Speech Encryption Based Anti-Tapping Device—*C. R. Suthikshn Kumar*, Defence Institute of Advanced Technology (DIAT), Girinagar, Pune, India

Tapping telephones nowadays is a major concern. There is a need for a portable device that can be attached to a mobile phone that can prevent tapping. Users want to encrypt their voice during conversation, mainly for privacy. The encrypted conversation can prevent tapping of the mobile calls as the network operator may tap the calls for various reasons. In this paper we propose a portable device that can be attached to the mobile phone/landline phone that serves as an anti-tapping device. The device encrypts the speech and decrypts the encrypted speech in real time. The main idea is that speech is unintelligible when encrypted.
Convention Paper 8999

3:00 pm

P15-6 Personalized Audio Systems—A Bayesian Approach—*Jens Brehm Nielsen*,[1,2] *Bjørn Sand Jensen*,[1] *Toke Jansen Hansen*,[1] *Jan Larsen*[1]
[1]Technical University of Denmark, Kongens Lyngby, Denmark
[2]Widex A/S, Lynge, Denmark

Modern audio systems are typically equipped with several user-adjustable parameters unfamiliar to most listeners. To obtain the best possible system setting, the listener is forced into non-trivial multi-parameter optimization with respect to the listener's own objective and preference. To address this, the present paper presents a general interactive framework for robust personalization of such audio systems. The framework builds on Bayesian Gaussian process regression in which the *belief* about the user's *objective* ➥

function is updated sequentially. The parameter setting to be evaluated in a given trial is carefully selected by sequential experimental design based on the belief. A Gaussian process model is proposed that incorporates assumed correlation among particular parameters, which provides better modeling capabilities compared to a standard model. A five-band constant-Q equalizer is considered for demonstration purposes, in which the equalizer parameters are optimized for each individual using the proposed framework. Twelve test subjects obtain a personalized setting with the framework, and these settings are significantly preferred to those obtained with random experimentation.
Convention Paper 9000

Workshop 19 — Saturday, Oct. 19
3:00 pm – 5:00 pm — Room 1E13

"HELP! I HAVE A TAPE RECORDER!"— RESTORATION AND REBUILDING ANALOG TAPE MACHINES

Moderator: **Noah Simon**, New York University, NY, NY, USA

Panelists: *John French*
Bob Shuster, Shuster Sound, Smithtown, NY, USA
Dan Zellman

A new generation of engineers, musicians, and audiophiles are discovering how the analog recorders from the "good old days" are helping them get a better sound or get that "analog sound" into their recordings. At the same time at the other end, archivists, preservationists, remastering engineers, and high end audiophiles need to know what's involved in taking care of these machines. This workshop will discuss the various options for these folks when they look for purchasing, maintaining, restoring, and using these recorders. During the workshop discussion, we hope to show examples of tape recorder repairs and restoration and have a running Q&A session.

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

Tutorial 16 — Saturday, Oct. 19
3:00 pm – 4:30 pm — Room 1E14

TURNTABLE TECHNIQUE: THE ART OF THE DJ

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

Mastering faders, pots, and switches as musical instruments? Turntable technique is where the audio engineer and the artist become one. Stephen Webber is the Director of Music Technology Innovation at Berklee College of Music in Valencia Spain, the composer or the "Stylus Symphony," and the author of *Turntable Technique: The Art of the DJ*. His presentation, which has been featured around the world, challenges traditional notions of music and technology.

Project Studio Expo
Saturday, October 19, 3:00 pm – 4:00 pm
Stage

LOUDNESS, LEVELS, AND METERING

Presenter: **Hugh Robjohns**, Technical Editor, Sound on Sound, Cambridge, UK

This seminar will cover the development and history of audio metering and discuss why traditional analog instruments are obsolete in the digital age. It will then cover digital metering and the associated problems, and contrast the concepts and practices of peak and loudness normalization. That will lead on to the aims of the ITU-R BS1770 loudness standard, its practical implementation, and then examples of how it has been implemented by a number of manufacturers and how it works in practice. There will be audio/visual examples throughout.

Knowledge Center
Saturday, October 19, 3:00 pm – 4:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Knowledge Center
Saturday, October 19, 3:00 pm – 6:00 pm
Booth 2738

WISYCOM SHOWCASE WITH MASSIMO POLO

Presenter: **Massimo Polo**, Wisycom, Vicenza, Italy

Imagine a world without 25 MHz block restrictions. That's what the Wisycom family of products envisions, with high quality diversity wireless systems built to operate over an impressively wide bandwidth.

Broadcast/Streaming Media Session 12
Saturday, Oct. 19 — 3:15 pm – 4:45 pm
Room 1E08

HTML5 AND STREAMING

Chair: **Valerie Tyler**, College of San Mateo, San Mateo, CA, USA

Panelists: *Jan Linden, Google*, Mountain View, CA, USA
Greg Ogonowski, Orban, San Leandro, CA, USA
Charles Van Winkle, Adobe Systems Incorporated, Minneapolis, MN, USA

HTML5 is a language for structuring and presenting content for the World Wide Web, a core technology of the Internet. It is the fifth revision of the HTML standard. HTML5 has many features built into the code. One feature is the media player and how it handles media being downloaded or streamed. This session will look into the technical considerations for media to be played back as well as the user interfaces.

Game Audio Session 11 **Saturday, Oct. 19**
3:15 pm – 4:45 pm **Room 1E10**

LEARNING FROM THE FUTURE

Presenters: **Scott Selfon**, Microsoft, Redmond, WA, USA
Garry Taylor, Sony Computer Entertainment Europe, Cambridge, Cambridgeshire, UK

With the "next generation" of game consoles soon to be this generation, what have we learned from games already in development? Is it really just "more of everything" or are other trends emerging as the defining factors for game audio production, implementation, and integration? In this panel we will discuss patterns and practices that are changing, accelerating, or declining for the titles of the next year and the next decade.

Product Design Session 5 **Saturday, Oct. 19**
3:15 pm – 4:45 pm **Room 1E09**

THE POWER OF THE BRAND

Presenter: **Adrian Weidmann**, StoreStream Metrics

This session will define "Brand" and explore its power and importance for the commercial success of your product development and/or service—be it a microphone, audio processing software, or recording studio. Developing, defining, and maintaining your brand and its message may the most important "product" you ever develop. This session will explore the power of Brand and outline seven key components to define your brand. The "Brand as Publisher" concept will be introduced, defined, and examples will be presented that can be used to create a meaningful dialog between your customers and your brand across available customer touchpoints—human, mobile, social media, web, and print. Understanding the power of this customer dialog can provide innovative insights for your product design team as well as propel your Brand.

Sound for Picture 6 **Saturday, Oct. 19**
3:30 pm – 5:00 pm **Room 1E11**

WORLD-CLASS CINEMA SOUND MIXERS DISCUSS THEIR CRAFT

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

Panelists: *Marti Humphrey*, The Dub Stage, Burbank, CA, USA
Chris Jacobson, The Dub Stage, Burbank, CA, USA
Branko Neskov

In what is fast becoming one of the most popular events in the "sound for picture" track, we again put together a panel of four of the top sound mixers for film and television.

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

Knowledge Center
Saturday, October 19, 3:30 pm – 5:30 pm
Room 1E06

PMC "MASTERS OF AUDIO": YOUNG GURU

Presenter: **Young Guru**, Roc Nation, Brooklyn, NY, USA

The Era of the Engineer

Revered as "The Sound of New York," Young Guru (Jay Z, Alicia Keys, Rihanna, Beyonce) possesses over a decade of experience in sound engineering and production for the acclaimed Roc-A-Fella Records and Def Jam Recordings. Through his lecture and demo series, #eraoftheengineer, Guru examines the recent emergence of a new generation of do-it-yourself engineers, analyzing and demonstrating what it means for the culture at large. There will be ample time for Q&A so this is your chance to ask Young Guru that question you always wanted to ask!!

Saturday, Oct. 19 **3:30 pm** **Room 1E02**
Standards Committee Meeting: SC-02-12 Working Group on Audio Applications of Networks

Special Event
MUSIC AND AUDIO FOR THE SMALLER SCREEN
Saturday, October 19, 4:00 pm – 5:00 pm
Room 1E03

Moderator: **Jerome Rossen**

Presenters *Christopher Kaufman*
John Kiehl
Steve Horowitz
Richard Warp

What are the important issues to take into account when you're composing, compiling, and refining your masterpiece for the small screen? What should you prioritize during preproduction? How does the smaller screen affect your creative decision making? How can you mix for success? What do you need to know if you're creating for iOS, Android and the Web? Join members of the Manhattan Producers Alliance as they conduct this panel addressing how to make the best possible audio for the "smaller screen."

Project Studio Expo
Saturday, October 19, 4:00 pm – 5:00 pm
Stage

ASK THE EDITORS

Presenters: **Kevin W. Becka**, Conservatory of Recording Arts and Sciences/Mix Magazine, Gilbert, AZ, USA
Strother Bullins, Pro Audio Review Magazine, North Carolina
Larry Crane, Tape Op Magazine, Portland, OR, USA; Jackpot! Recording Studio
Mike Metlay, RECORDING Magazine, Boulder, CO, USA
Hugh Robjohns, Technical Editor, Sound on Sound, Cambridge, UK
Frank Wells, Pro Sound News, Murfreesboro, TN, USA; Pro Audio Review, Music Festival Business
Paul White

Open Q&A session—ask the magazine guys about product reviews or technique articles. Tell us what you like. Tell us what you don't like! Or you can just ask about recording: after all, we are all practitioners too. ➥

Saturday, Oct. 19 4:00 pm Room 1E04
Technical Committee Meeting: Fiber Optics for Audio

Live Sound Seminar 11 Saturday, Oct. 19
4:30 pm – 7:00 pm Room 1E12

TVBDS, GEO-LOCATION DATABASES, AND UPCOMING SPECTRUM AUCTIONS: AN IN-DEPTH LOOK AND THEIR IMPACT ON WIRELESS MICROPHONE OPERATIONS

Chair: **Henry Cohen**, CP Communications

Panelists: *Joe Ciaudelli*, Sennheiser Electronic Corporation, Old Lyme, CT, USA
Ira Keltz, Federal Communications Commission
Michael Marcus, Marcus Spectrum Solutions, Cabin John, MD, USA
David Pawlik, Skadden, Arps, Slate, Meagher & Flom, Washington, DC
Edgar Reihl, Shure, Incorporated, Niles, IL, USA
Peter Stanforth, Spectrum Bridge
James Stoffo, Independent Frequency Coordinator

Television band devices (TVBD) and geo-location databases directing TVBD operations are a reality, and the first certified fixed TVBDs are in service. The 600 MHz auction may likely occur in 2014 with a vacate date within the next six to eight years. Operating wireless microphones, IEMs, intercoms, and cueing in this new environment requires understanding how the databases work, the rules governing both licensed and unlicensed wireless production equipment, and what spectrum is currently available and will be available in the future. This panel brings together a diverse group of individuals intimately involved from the beginning with TVBDs, databases, spectrum auctions, and the new FCC rules as well as seasoned veterans of medium- to large-scale wireless microphone deployments to discuss how the databases operate, how to use the database for registering TV channel usage, and best procedures and practices to insure minimal problems.

Special Event
BRUCE SWEDIEN: I HAVE NO SECRETS
Saturday, October 19, 4:30 pm – 6:00 pm
Room 1E15/16

This Special Event showcases the mindset of one of music's most-important engineers—ever! Interviewed by author Bill Gibson, Bruce Swedien generously shares the depth of his technical and artistic insights, inspiring greatness in the musical application of technology in recording and production. In an industry propelled by the excessive use of plug-ins, automatic tuning, and processing, Swedien reveals a different approach—his approach, which achieves massive sonic power through the mastery of musical and technical fundamentals and the insightful understanding of the role of microphones, the acoustical environment, effects processors, and the all-important emotional component in the recording process.

Bring your questions. Don't miss a chance to learn from an audio industry master—a legend, an icon, and a friend to engineers around the world. Bruce Swedien has always been generous with his knowledge—he has no secrets! There will be space in the program for you to ask questions.

A five-time Grammy winner—and thirteen-time Grammy nominee—Swedien's impact on popular music is undeniable! His approach to recording music has proven to be a game changer, with engineers at all levels referencing his work as a definitive sonic standard. From recording and mixing Michael Jackson's albums (Off the Wall, Thriller, Bad, Dangerous, Invincible, and HIStory) to many of Quincy Jones' hits (The Dude, Back on the Block, Q's Jook Joint, and more) to the music of greats such as Count Basie, Duke Ellington, the Brothers Johnson, and Natalie Cole, Bruce Swedien has always operated at the very highest level of excellence and expertise in the recording industry.

Session EB5 Saturday, Oct. 19
5:00 pm – 6:30 pm 1E Foyer

E-BRIEF POSTERS—PART 2

5:00 pm

EB5-1 Digital Model of the Passive James/Baxandall Tonestack—*Christopher Bennett*,[1,2] *Jonathon Toft-Nielsen*,[3] *Connor McCullough*[1]
[1]University of Miami, Coral Gables, FL, USA
[2]Oygo Sound LLC, Miami, FL, USA
[3]Intelligent Hearing Systems, Miami, FL, USA

E. J. James described a two-knob tone control in 1949 with easily selectable boost/cut depths as well as cutoff frequencies. This design was later popularized by P. J. Baxandall to provide negative feedback in active circuits, and was subsequently popular in many high-end amplifiers. Here, the authors analyzed the circuit in the s-domain, preserving parametric control of bass and treble potentiometer values. Poles and zeros were found using Ferrari's solution to a quartic equation, followed by bilinear transformation to the z-domain, and finally lumping into second-order sections to produce a computationally efficient and faithful emulation of this classic tonestack.
Engineering Brief 124

5:00 pm

EB5-2 Automatic Analog Preamp Gain Control Using Digital Command—*Nicolas Sturmel*,[1] *Fusheng Yu*[2]
[1]Digigram S.A., Montbonnot Saint Martin, France
[2]ENSEEIHT–INP, Toulouse, France

Automatic Gain Control (AGC) is a common tool for field recording, but it usually requires specific hardware such as voltage controlled amplifiers. In this paper, we address the problem of designing an AGC when none of this hardware is present, using an ubiquitous digitally controlled high end analog preamp. To do this, we have to overcome two problems: fixed gain steps and variable delay of the gain command. In order to propose an efficient solution, we will first study the effects of each of those two problems. Finally, a very simple digitally controlled automatic gain, but of high quality will be proposed, using only 10MIPS of processing power from our high end USB sound card.
Engineering Brief 125

5:00 pm

EB5-3 Testing Watermark Robustness against

Application of Audio Restoration Algorithms **—***Bozena Kostek, Janusz Cichowski, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

The purpose of this study was to test to what extent watermarks embedded in distorted audio signals are immune to audio restoration algorithm performing. Several restoration routines such as noise reduction, spectrum expansion, clipping or clicks reduction were applied in the online website system. The online service was extended with some copyright protection mechanisms proposed by the authors. They contain low-level music features embedded as watermarks using the non-blind approach. After applying restoration algorithms, the watermark is extracted from the audio track. It was shown in experiments, that a watermark "attacked" by the restoration procedures may still be detected. However in some cases it is possible to retrieve only a binary information about the watermark presence in the audio carrier.
Engineering Brief 126

5:00 pm

EB5-4 PsychoMasker: An iOS Application for the Visualization of PsychoAcoustic Principles—*Andrew Ayers,*[1] *Robert Rehrig,*[1] *Christopher Bennett,*[1,2] *Colby N. Leider*[1]
[1]University of Miami, Coral Gables, FL, USA
[2]Oygo Sound LLC, Miami, FL, USA

The concept of masking in psychoacoustics has invaded the daily lives of almost every audio listener since the initial release of the MPEG-1 standard. With the ubiquity of the MP3, the consumption of perceptually coded audio is impossible to avoid. While many people understand the concept of perceptual coding, it can be difficult to visualize what is actually happening to the information in the audio files. PsychoMasker is an App that provides real-time visualization of the psychoacoustic principles used in MPEG encoding to anyone with an iPad. The PsychoMasker App shows the user how the encoding process affects any song in the user's iTunes library step-by-step.
Engineering Brief 127

5:00 pm

EB5-5 Using MIDI Control Surfaces with MATLAB Programs and Simulink—*Charlie DeVane*, MathWorks, Natick, MA, USA

MATLAB and Simulink are widely used in the design of software and hardware for audio products. MATLAB programs and Simulink models can simulate signal processing algorithms, control logic, and other aspects of the system design in real time, yielding substantial improvements in designer productivity and product quality. When exploring a new algorithm or product concept, designers often need to simultaneously tune multiple system parameters while simulating. This can become cumbersome using GUIs, but MIDI control surfaces provide a natural, intuitive interface, further enhancing the designer's work flow. In some work flows, such as rapid prototyping, MIDI control surfaces can eliminate the need to create a GUI. Using numerous examples, including a simple reverberator, this brief shows how to use MIDI control surfaces to interactively control running MATLAB programs and Simulink models.
Engineering Brief 128

5:00 pm

EB5-6 MIDI to CV Conversion Using a Livid BrainV2 and I2C Protocol—*Mark Gill*, University of Miami, Coral Gables, FL, USA

Bob Moog developed Control voltage (CV) in the 1960s. His introduction of CV called for an oscillator's pitch to vary at the rate of 1 volt per octave. This scheme is widely used today in analog circuits, and can be mimicked digitally. The CV output is determined by converting various MIDI messages including USB MIDI, and physical controls including analog potentiometer inputs and momentary on, momentary off buttons. The Livid BrainV2 handles all the MIDI inputs and directs them to a digital to analog converter to create the CV signal controlled by I2C communication. This paper documents the hardware used to create the converter, the mathematical considerations for conversion, and techniques used to overcome the limited ability of 8-bit MIDI messages to be portrayed as an analog signal. Sound clips are available at my website markolgill.com
Engineering Brief 129

Workshop 20 **Saturday, Oct. 19**
5:00 pm – 7:30 pm **Room 1E13**

WHAT'S RIGHT AND WHAT'S WRONG WITH TODAY'S MOTION PICTURE SOUND?

Chair: **John F. Allen**, High Performance Stereo, Newton, MA, USA

Panelists: *Mark Collins*, Marcus Theatres, Milwaukee, WI, USA
Douglas Greenfield, Dolby Labs, Burbank, CA, USA
Brian A. Vessa, Sony Pictures Entertainment, Culver City, CA, USA; Chair SMPTE 25 CSS standards committee

Do you think movies are too loud? Do you admire their sound quality? Why do so many complain about motion picture sound? The answers may come as a surprise. To fully understand the complexities involved, one must separately explore both the way movies are made and they way they are played.

This workshop consists of a panel of experts that actually work in both creating and presenting motion pictures. Their candid presentations will begin by exploring the often inaccurate way sound system measurements are interpreted. Complicating matters, the resulting equalization errors are different for different parts of the audio spectrum. Theater sound system mis-calibration errors cannot only diminish the sound quality but can cause significant unintended playback level increases as well. This presentation will not only describe these problems but will offer solutions as well.

These and other issues are the focus of the recent standards work. The obstacles presented when sometimes working at the limits of technology will be described by a senior studio sound engineer who is also the chairman of the largest SMPTE committee assigned to motion picture sound. ➥

Movies mixed all over the world must be created with such consistency that they can all be played in a theater without the need to adjust a fader or an equalizer. Perhaps no part of the audio production industry is closer to achieving this goal than motion pictures. This demands hours of work and many long days often diplomatically supporting movie makers and assisting them in building the final product they are striving to create. One of our panelists in a leader in this rather exclusive field.

After years in the making of a film, it all comes down to theatrical presentation. Building and maintaining hundreds and even thousands of screens is an art in itself, often executed with mixed results. One of exhibitions most accomplished technical directors will detail the day to day challenges one faces in such a role.

Workshop 21 **Saturday, Oct. 19**
5:00 pm – 7:00 pm **Room 1E10**

LIES, DAMN LIES, AND AUDIO GEAR SPECS

Chair: **Ethan Winer**, RealTraps, New Milford, CT, USA

Panelists: *Scott Dorsey*, Williamsburg, VA, USA
David Moran, Boston Audio Society, Boston, MA, USA
Mike Rivers, Gypsy Studio, Falls Church, VA, USA

The fidelity of audio devices is easily measured, yet vendors and magazine reviewers often omit important details. For example, a loudspeaker review will state the size of the woofer but not the low frequency cut-off. Or the cut-off frequency is given but without stating how many dB down or the rate at which the response rolls off below that frequency. Or it will state distortion for the power amps in a powered monitor but not the distortion of the speakers themselves, which of course is what really matters. This workshop, therefore, defines a list of standards that manufacturers and reviewers should follow when describing the fidelity of audio products. It will also explain why measurements are a better way to assess fidelity than listening alone.

Workshop 22 **Saturday, Oct. 19**
5:00 pm – 7:00 pm **Room 1E14**

LOUDNESS WARS: LEAVE THOSE PEAKS ALONE

Chair: **Thomas Lund**, TC Electronic A/S, Risskov, Denmark

Panelists: *John Atkinson*
Florian Camerer, ORF, Vienna, Austria
Bob Ludwig, Gateway Mastering Studios, Inc., Portland, ME, USA
George Massenburg, Schulich School of Music, McGill University, Montreal, Quebec, Canada
Susan Rogers

Music production, distribution, and consumption has been caught in a vicious spiral rendering two decades of our music heritage damaged. Because of irreversible dynamics processing and data reduction from production onwards, new tracks and remastered ones typically sound worse than what could even be expected from compact cassette. However, with Apple, WiMP, and Spotify now engaged in a competition on quality, and FM radio in Europe adopting EBU R128 loudness normalization, limbo-practice is finally losing its grip on distribution.

The panel uses terms “Peak to Loudness Ratio” (PLR) and “Headroom” to analyze recorded music fidelity over the past 50 years from four different angles: physiological, production, distribution, and consumption. In the new realm, it’s futile to master music louder than –16 LKFS.

Broadcast/Streaming Media Session 13
Saturday, Oct. 19 **5:00 pm – 6:30 pm**
Room 1E08

FACILITY DESIGN

Chair: **Sergio Molho**, Walters-Storyk Design Group, Highland, NY, USA

Panelist: *Jim Servies*, ESPN
John Storyk, Walters-Storyk Design Group, Highland, NY, USA

Part 1: A Ground Up Design – ESPN, Bridgeport, CT

Part 2: Corrective Measures – QTV Doha, Qatar

The wisest course of action to insure optimal acoustics for broadcast facilities is to begin at the design stage. ESPN’s new production complex in Bridgeport, CT, represents an ideal example of the value of bringing acousticians in at the earliest possible opportunity. The panel will illustrate the critical issues to be addressed and the many advantages of acoustician participation at the design phase of a facility design. In a contrasting scenario, the panel will discuss QTV in Doha, Qatar. Last year after construction was completed on this state-of-the-art broadcast production complex, the three primary permanent sets designed for the new complex required sophisticated (and undetectable) acoustic treatments to alleviate excessive reverberation and related sound reflection/absorption issues. A commitment to a mid-December broadcast première presented acousticians with an inflexible sixty-day window to accomplish and evaluate, critical acoustic measurements and simulation tests, present recommendations, and complete the installation. This panel will provide insights into the evaluation and recommendation process, including a description of programs and tools, supplier outreach, installation issues, and client coordination concerns.

Historical Event
THE 35MM ALBUM MASTER FAD
Saturday, October 19, 5:00 pm – 7:00 pm
Room 1E09

Presenter: **Thomas Fine**, Sole Proprietor of Private Studio, Brewster, NY, USA

In the late 1950s and early 1960s, a new market emerged for ultra-high fidelity recordings. Once cutting and playback of the stereo LP were brought up to high quality levels, buyers of this new super-realistic format wanted ever more “absolute” sound quality. The notion emerged, first with Everest Records, a small independent record label in Queens, to use 35mm magnetic film as the recording and mastering medium. 35mm had distinct advantages over tape formulations and machines of that time —lower noise floor, less wow and flutter, higher absolute levels before saturation, almost no crosstalk or print-through, etc. Everest Records made a splash with the first 35mm LP masters not connected to motion-picture soundtracks but quickly faltered as a business. The unique set of recording equipment and the Everest studio remained intact and was used to make commercially successful 35mm records for Mercury, Command, Cameo-Parkway, and Project 3. The fad faded by the mid-60s as tape machines and tape formulations improved, and the high cost of working with 35mm magnetic film became unsustainable. The original Everest equipment survived to be used in the Mercury Living Presence remasters for CD. Just recently, the original Everest 35mm recordings have been reissued in new high-resolution digital remasters. This presentation will trace the history of 35mm magnetic recording, the brief but high-profile fad of 35mm-based LPs, and the after-life of those original recordings. We will also look at the unique set of hardware used to make the vast majority of the 35mm LPs. The presentation will be augmented with plenty of audio examples from the original recordings.

Student/Career Development Event
RECORDING COMPETITION—PART 2
Saturday, October 19, 5:00 pm – 7:00 pm
Room 1E11

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

5:00 pm Traditional Acoustic Recording

Judges: Martha de Francisco, David Bowles, Ulrike Schwarz

6:00 pm Sound for Visual Media

Judges: Brian McCarty, Skip Lievsay, Bob Bronow

Saturday, Oct. 19 5:00 pm Room 1E02
Standards Committee Meeting: SC-07-01 Working Group on Audio Metadata

Student/Career Development Event
AES/SPARS ROCKIN' STUDENT PARTY
Saturday, October 19, 9:30 pm – 12:00 midnight
Avatar Recording Studios
441 West 53rd Street, NY, NY 10019

Audio Students! Rub elbows with industry mentors and mucky mucks at the AES Student Party to be held at the legendary Avatar Studios. A special SPARS Legacy Award will be presented to Howard Schwartz.

Many artists, producers, and engineers have passed through the doors of this studio. This facility served as Madonna's home when she recorded "Like a Virgin" and played host to luminaries including Bruce Springsteen, The Rolling Stones, David Bowie, and Roxy Music. More recently, artists such as Paul McCartney, Paul Simon, John Mayer, Trey Anastasio, Muse, Josh Groban, Donald Fagen, Norah Jones, Diana Krall, My Morning Jacket, Kings of Leon, and Bruno Mars recorded their projects at Avatar.

Present your Full Access Student Badge for Admission

Session P16 Sunday, Oct. 20
9:00 am – 12:00 noon Room 1E07

SPATIAL AUDIO—PART 2

Chair: **Jean-Marc Jot**, DTS, Inc., Calabasas, CA, USA

9:00 am

P16-1 Defining the Un-Aliased Region for Focused Sources—*Robert Oldfield, Ian Drumm,* University of Salford, Salford, Greater Manchester, UK

Sound field synthesis reproduction techniques such as wave field synthesis can accurately reproduce wave fronts of arbitrary curvature, including sources with the wave fronts of a source in front of the array. The wave fronts are accurate up until the spatial aliasing frequency, above which there are no longer enough secondary sources (loudspeakers) to reproduce the wave front accurately, resulting in spatial aliasing contribution manifesting as additional wave fronts propagating in directions other than intended. These contributions cause temporal, spectral, and spatial errors in the reproduced wave front. Focused sources (sources in front of the loudspeaker array) have a unique attribute in this sense in that there is a clearly defined region around the virtual source position that exhibits no spatial aliasing contributions even at an extremely high frequency. This paper presents a method for the full characterization of this un-aliased region using both a ray-based propagation model and a time domain approach.
Convention Paper 9001

9:30 am

P16-2 Using Ambisonics to Reconstruct Measured Soundfields—*Samuel W. Clapp,*[1] *Anne E. Guthrie,*[1,2] *Jonas Braasch,*[1] *Ning Xiang*[1]
[1]Rensselaer Polytechnic Institute, Troy, NY, USA
[2]Arup Acoustics, New York, NY, USA

Spherical microphone arrays can measure a soundfield's spherical harmonic components, subject to certain bandwidth constraints depending on the array radius and the number and placement of the array's sensors. Ambisonics is designed to reconstruct the spherical harmonic components of a soundfield via a loudspeaker array and also faces certain limitations on its accuracy. This paper looks at how to reconcile these sometimes conflicting limitations to produce the optimum solution for decoding. In addition, binaural modeling is used as a method of evaluating the proposed decoding method and the accuracy with which it can reproduce a measured soundfield.
Convention Paper 9002

10:00 am

P16-3 Subjective Evaluation of Multichannel Sound with Surround-Height Channels—*Sungyoung Kim,*[1] *Doyuen Ko,*[2,3] *Aparna Nagendra,*[1] *Wieslaw Woszczyk*[3]
[1]Rochester Institute of Technology, Rochester, NY, USA
[2]Belmont University, Nashville, TN, USA
[3]McGill University, Montreal, Quebec, Canada

In this paper we report results from an investiga- ➥

tion of listener perception of surround-height channels added to standard multichannel stereophonic reproduction. An ITU-R horizontal loudspeaker configuration was augmented by the addition of surround-height loudspeakers in order to reproduce concert hall ambience from above the listener. Concert hall impulse responses (IRs) were measured at three heights using an innovative microphone array designed to capture surround-height ambience. IRs were then convolved with anechoic music recordings in order to produce seven-channel surround sound stimuli. Listening tests were conducted in order to determine the perceived quality of surround-height channels as affected by three loudspeaker positions and three IR heights. Fifteen trained listeners compared each reproduction condition and ranked them based on their degree of appropriateness. Results indicate that surround-height loudspeaker position has a greater influence on perceived sound quality than IR height. Listeners considered the naturalness, spaciousness, envelopment, immersiveness, and dimension of the reproduced sound field when making judgments of surround-height channel quality.
Convention Paper 9003

10:30 am

P16-4 A Perceptual Evaluation of Recording, Rendering, and Reproduction Techniques for Multichannel Spatial Audio—*David Romblom,*[1,2,3] *Richard King,*[1,2,3] *Catherine Guastavino*[1,2,3]
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
[3]In collaboration with Sennheiser Technology and Innovation, San Francisco, CA, USA

The objective of this project is to perceptually evaluate the relative merits of two different spatial audio recording and rendering techniques within the context of two different multichannel reproduction systems. The two recordings and rendering techniques are “natural,” using main microphone arrays, and “virtual,” using spot microphones, panning, and simulated acoustic delay. The two reproduction systems are the 3/2 system (5.1 surround) and a 12/2 system, where the frontal L/C/R triplet is replaced by a 12-loudspeaker linear array. The perceptual attributes of multichannel spatial audio have been established by previous authors. In this study magnitude ratings of selected spatial audio attributes are presented for the above treatments and results are discussed.
Convention Paper 9004

11:00 am

P16-5 The Optimization of Wave Field Synthesis for Real-Time Sound Sources Rendered in Non-Anechoic Environments—*Ian Drumm, Robert Oldfield*, University of Salford, Salford, Greater Manchester, UK

Presented here is a technique that employs audio capture and adaptive recursive filter design to render in real time dynamic, interactive, and content rich soundscapes within non-anechoic environments. Typically implementations of wave field synthesis utilize convolution to mitigate for the amplitude errors associated with the application of linear loudspeaker arrays. Although recursive filtering approaches have been suggested before, this paper aims to build on the work by presenting an approach that exploits Quasi Newton adaptive filter design to construct components of the filtering chain that help compensate for both the particular system configuration and mediating environment. Early results utilizing in-house developed software running on a 112-channel wave field synthesis system show the potential to improve the quality of real-time 3-D sound rendering in less than ideal contexts.
Convention Paper 9005

11:30 am

P16-6 A Perceptual Evaluation of Room Effect Methods for Multichannel Spatial Audio—
David Romblom,[1,2,3] *Richard King,*[1,2,3] *Catherine Guastavino*[1,2,3]
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music Media and Technology (CIRMMT), Montreal, Quebec, Canada
[3]In collaboration with Sennheiser Technology and Innovation, San Francisco, CA, USA

The room effect is an important aspect of sound recording technique and is typically captured separately from the direct sound. The perceptual attributes of multichannel spatial audio have been established by previous authors, while the psychoacoustic underpinnings of room perception are known to varying degrees. The Hamasaki Square, in combination with a delay plan and an aesthetic disposition to “natural” recordings, is an approach practiced by some sound recording engineers. This study compares the Hamasaki Square to an alternative room effect and to dry approaches in terms of a number of multichannel spatial audio attributes. A concurrent experiment investigated the same spatial audio attributes with regard to the microphone and reproduction approach. As such, the current study uses a 12/2 system based upon 3/2 (5.1 surround) where the frontal L/C/R triplet has been replaced by a linear wavefront reconstruction array.
Convention Paper 9006

Workshop 23 — Sunday, Oct. 20
9:00 am – 10:30 am — Room 1E12

CLASSIC SPEECH INTELLIGIBILITY AT THE ROYAL DANISH THEATRE

Chair: **Jan Voetmann**, Voetmann-Akustik, Frederiksberg C, Denmark

Panelists: *Eddy B. Brixen*, EBB-consult/DPA Microphones, Smørum, Denmark
Karsten Wolstad, Royal Danish Theatre Dramahouse; Danish National School of Performing Arts, Copenhagen, Denmark

Speech intelligibility tests are crucial for evaluating the acoustic quality of performing theaters, cinemas, and auditoriums. But also for PA systems in churches, railway stations, etc. The new grand auditorium of the Royal Danish Theatre in Copenhagen has shown some interesting differences in the subjective perceived speech quality at different seats. In order to analyze the situation a "classical" speech intelligibility test, reading nonsense words buried in standard sentences, was conducted alongside objective STI measurements.

The workshop will present the result of this unique and rare test situation, and discussions with the audience will hopefully lead to important improvements before the final project is launched.

This research project is supported by the Danish Sound Innovation Network ("Danish Sound"). Danish Sound's primary services are innovation projects, matchmaking, networking, knowledge dissemination and globalization activities.

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

Workshop 24 **Sunday, Oct. 20**
9:00 am – 10:30 am **Room 1E11**

DSD VS DXD: EXTREME DSD AND PCM RESOLUTIONS COMPARED

Chair: **Dominique Brulhart**, Merging Technologies

Panelists: *Morten Lindberg*, 2L (Lindberg Lyd AS), Oslo, Norway
John Newton, Soundmirror, Inc., Jamaica Plain, MA, USA

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, John Newton from Soundmirror and Morten Lindberg from 2L, to present some of their recordings and discuss about their experience making productions in DSD and DXD. Recent recordings in 11.2 MHz DSD, DSD, and DXD will be presented and recording, editing, mixing, and mastering techniques and considerations using DSD and DXD will be discussed and compared.

Workshop 25 **Sunday, Oct. 20**
9:00 am – 10:30 am **Room 1E14**

AUDIO @ THE NEAR SPEED OF LIGHT WITH FIBER OPTICS

Chair: **Ronald Ajemian**, Owl Fiber Optics, Flushing, NY, USA

Panelists: *Marc Brunke*, Optocore GmbH, Grafelfing, Germany
Steve Lampen, Belden, San Francisco, CA, USA
Fred Morgenstern, Neutrik USA
Warren Osse, Vistacom, Inc., Allentown, PA, USA

As the growth of audio/video integrates and increases, so does the demand for more fiber optic technology. Sending, receiving, and storing a good quality audio/video data file is crucial in many areas of our industry. The workshop panel will address the audience to educate, inform, and update by having a question and answer format. Everyone who attends this workshop will walk away with more knowledge to be better prepared on existing and future use of fiber optic technology as it pertains to professional audio/video.

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

Tutorial 17 **Sunday, Oct. 20**
9:00 am – 10:30 am **Room 1E13**

FXPERTISE: REVERB

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

Reverberation in the recording studio comes from a variety of technologies and achieves a great range of results. Echo chambers, plates, and springs still have their place in contemporary music production even as digital reverb algorithms dominate. This tutorial reviews the technologies behind studio reverb units, shares a broad range of measurement data, and offers organization and insight into the creative, musical applications of reverb. Audio engineers reach for reverb effects to create space and ambience, to be sure. Reverb is also employed to influence timbre, create textures, invoke scene changes, manipulate masking, and synthesize new sounds entirely.

Tutorial 18 **Sunday, Oct. 20**
9:00 am – 11:00 am **Room 1E08**

SMALL, LOUD-SPEAKERS: TAKING PHYSICS TO THE LIMIT

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

This tutorial plays on the challenge of engineering small loudspeakers that can also be loud, have high efficiency, and low distortion. While the basic construction and principle of the moving coil loudspeaker has remained unchanged for a century, our understanding of its operation and limitations has increased very considerably. Modern signal processing techniques, coupled with careful design of the magnet, voice coil, and suspension can be employed to create small drive units that have an optimum trade-off between the crucial design factors. An understanding of the perceptual effects that arise from these design trade-offs helps to bring about loudspeakers that not only measure well but sound good. This is particularly important as compact devices such as mobile phones demand more and more quality from tiny loudspeakers in minimal enclosures.

Tutorial 19 **Sunday, Oct. 20**
9:00 am – 10:00 am **Room 1E09**

NATIONAL RECORDING PRESERVATION PLAN: AUDIO PRESERVATION/ARCHIVING: NEW TECHNOLOGIES AND ACCESS ISSUES

Presenters: **Matt Barton**, Library of Congress
Brad McCoy, Library of Congress
Gerald Seligman, National Recording Preservation Foundation

This presentation gives an overview of the Library of ➥

Congress National Recording Preservation Plan (NRPP) published in December 2012, with a focus on the portions that particularly include and involve AES. Brad McCoy will focus on Sections 1.7 (New Technologies for Audio Preservation: Encourage scientific and technical research leading to the development of new technologies to recover, reformat, and preserve audio recording media) and 1.8 (Documentation of Legacy Technologies: Research, collect, document, and preserve information on legacy record practices and technologies). Matthew Barton will discuss recommendations 3.1, 3.2, and 3.3, which deal with public and educational access, using a range of radio broadcasts, interviews, field recordings, and the recording formats and sonic issues particular to each one. The proposed national discography and national directory of sound recording collections will be discussed in this context, as will cataloging issues. This presentation will include audio examples from the Library of Congress collections to illustrate points.

Game Audio Session 12 — Sunday, Oct. 20
9:00 am – 11:00 am — Room 1E10

PROFESSIONAL GAME AUDIO—OPPORTUNITIES IN THE MOBILE SPACE

Moderator: **Stephen Harwood Jr.**, Education Working Group Chair, IASIG, New York, USA

Presenters: *Andrew Aversa*, Impact Soundworks
Steve Horowitz, The Code International Inc., San Francisco, CA, USA
Jory K. Prum, studio.jory.org
Michael Sweet, Berklee College of Music
Gina Zdanowicz, Serial Lab Studios, NJ, USA

In addition to sound design, composition, and production supervision, game audio requires skill sets that are rarely encountered elsewhere, including interactive audio programming and implementation. This broad array of work types provides for an equally broad range of career opportunities. Whatever your background and area of specialized expertise might be, there is room for you in this rapidly growing industry. In this session a panel of accomplished industry veterans will discuss how to begin and develop a successful career in game audio with a focus on the new opportunities available in the booming mobile gaming and web apps marketplace. Audience members will take away a comprehensive understanding of the many opportunities available to audio professionals in the video game industry, as well as valuable suggestions and insights into how to land that first gig.

Tutorial 20 — Sunday, Oct. 20
10:00 am – 11:00 am — Room 1E09

NATIONAL RECORDING PRESERVATION PLAN: AUDIO PRESERVATION/ARCHIVING: DOCUMENTATION OF PRESERVATION TECHNIQUES FOR ARCHIVAL FORMATS

Presenters: **Mark Hood**, Indiana University, Bloomington, IN, USA
Brad McCoy, Library of Congress
Marcos Sueiro Bal, WNYC / New York Public Radio, New York, NY, USA; Masterdisk, New York, NY, USA

The transmission of playback techniques to younger practitioners has become increasingly difficult in the often isolated archive environments. This is addressed in Recommendation 1.8 of the NRPP: “Documentation of Legacy Technologies: Research, collect, document, and preserve information on legacy recording practices and technologies,” which specifically recommends to “initiate a program to videotape interviews and demonstrations by senior audio engineers.” The Association of Recorded Sound Collections (ARSC) Technical Committee will introduce its efforts in this regard, initiated through support from the National Recording Preservation Plan and others.

Knowledge Center
Sunday, October 20, 10:00 am – 11:00 am
Room 1E06

PMC “MASTERS OF AUDIO”: CHRIS TABRON

Presenter: **Chris Tabron**

Beyonce “SOS, the Reggae Mix”

Having worked with the likes of Beyonce, Robin Thicke, Santigold, John Legend, Shiny Toy Guns, and Mike Posner Chris will present the latest single he mixed for Beyonce, as well as a cross selection of his versatile work while explaining his approach.

Knowledge Center
Sunday, October 20, 10:00 am – 11:00 am
Booth 2738

RF MICS, SPECTRUM, AND THE FUTURE WITH KARL WINKLER OF LECTROSONICS

Presenter: **Karl Winkler**, Lectrosonics, Rio Rancho, NM, USA

Straight from Rio Rancho, Karl Winkler has been staying up well past his bedtime preparing to tell you about the future of RF. Are the terms “spectrum” and “FCC Regulation” keeping you up at night? Do you worry about your wireless future? Then let Karl be your crystal ball and do not miss this presentation!

Systems Sound Symposium 1
Sunday, October 20, 10:15 am – 11:00 am
Stage

LOUD AND CLEAR—HOW EXPERTISE IN INTELLIGIBILITY MEASUREMENT CAN BUILD YOUR AV INTEGRATION BUSINESS

Increasing adoption of Mass Notification and Emergency Communications (MNEC) standards will trigger a surge in demand for audio expertise as architects address requirements for greater intelligibility in voice evacuation systems.

Session P17 — Sunday, Oct. 20
10:30 am – 12:00 noon — 1E Foyer

POSTERS: APPLICATIONS IN AUDIO—PART 2

10:30 am

P17-1 Source of ENF in Battery-Powered Digital Recordings—*Jidong Chai*,[1] *Fan Liu*,[2] *Zhiyong Yuan*,[3] *Richard W. Conners*,[4] *Yilu Liu*[1]
[1]University of Tennessee, Knoxville, TN, USA
[2]Chongqing University, Chongqing, China
[3]China Southern Power Grid, Guangzhou, China
[4]Virginia Polytechnic Institute and State University, Blacksburg, VA, USA

Forensic audio authenticity has developed remarkably over the last few years due to advances in technology of digital recording processing. The ENF (Electric Network Frequency) Criterion is one of the possible tools and has shown very promising results in forensic authentication of digital recordings. However, currently there are very few experiments and papers on studying the source of ENF signals existing in digital recordings. In addition, it is unclear whether or not there are detectable ENF traces in battery-powered digital audio recordings. In this paper the study of ENF source in battery-powered digital recordings is presented, and it shows that ENF in these recordings may not be mainly caused by low frequency electromagnetic field induction but by low frequency audible hum. This paper includes a number of experiments to explore the possible sources of ENF in battery-powered digital recordings. In these experiments, the electric and magnetic field strength in different locations is measured and the results of corresponding ENF extraction are analyzed. Understanding this underlying phenomenon is critical to verify the validity of ENF techniques.
Convention Paper 9007

10:30 am

P17-2 The Audio Performance Comparison and Method of Designing Switching Amplifiers Using GaN FET —*Jaecheol Lee, Haejong Kim, Keeyeong Cho, Haekwang Park,* Samsung Electronics DMC R&D Center, Suwon, Korea

This paper addresses physical characteristics of FET materials, the method of designing switching amplifiers using GaN FET, and the audio performance comparison of silicon and GaN FET. The physical characteristics of GaN FET are excellent, but there is a technical limitation to apply to consumer electronics. Depletion mode GaN FET is used in the proposed system. Its characteristic is better than Enhance mode. But it has the characteristic of normally turn on. To solve this problem, a cascaded GaN switch block is used. It is a combination of depletion mode GaN and enhanced mode Si. The proposed method has more of an outstanding audio performance than the switching amplifier used in silicon.
Convention Paper 9008

10:30 am

P17-3 Audio Effect Classification Based on Auditory Perceptual Attributes—*Thomas Wilmering, György Fazekas, Mark B. Sandler*, Queen Mary University of London, London, UK

While the classification of audio effects has several applications in music production, the heterogeneity of possible taxonomies, as well as the many viable points of view for organizing effects, present research problems that are not easily solved. Creating extensible Semantic Web ontologies provide a possible solution to this problem. This paper presents the results of a listening test that facilitates the creation of a classification system based on auditory perceptual attributes that are affected by the application of audio effects. The obtained results act as a basis for a classification system to be integrated in a Semantic Web Ontology covering the domain of audio effects in the context of music production.
Convention Paper 9009

10:30 am

P17-4 Development of Volume Balance Adjustment Device for Voices and Background Sounds within Programs for Elderly People—
Tomoyasu Komori,[1,2] *Atsushi Imai*,[3] *Nobumasa Seiyama*,[3] *Reiko Takou*,[3] *Tohru Takagi*,[3] *Yasuhiro Oikawa*[2]
[1]NHK Engineering System, Inc., Setagaya-ku, Tokyo, Japan
[2]Waseda University, Shinjuku-ku, Tokyo, Japan
[3]NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan

Elderly people sometimes feel that the background sounds (music and sound effects) of broadcast programs are annoying. In response, we have developed a device that can adjust the mixing balance of program sounds suitable for elderly people, on the receiver side. The device suppresses uncorrelated components in the stereo background sound in speech segments (intervals in which narration and dialog are mixed with background sounds), and suppresses background sounds only without deterioration by gain control alone in non-speech segments. By subjective evaluations, we have verified that the proposed method can suppress the background sounds of programs by an equivalent of 6 dB, and viewing experiments with elderly people have shown that program sounds have become easier to understand.
Convention Paper 9010

10:30 am

P17-5 Acoustical Measurement Software Housed on Mobile Operating Systems Test—*Felipe Tavera*, Walters Storyk Design Group, Highland, NY, USA

A measurement test is devised to provide comparative results between a dedicated type I Sound Level Pressure Meter and a PDA and mobile application with proprietary additional components. The test pretends to analyze and compare results considering only frequency response, linearity over selected dynamic range, and transducer's directivity under controlled on-site conditions. This, under the purpose of examining the accuracy of the non-dedicated hardware to perform acoustic measurements.
Convention Paper 9011

10:30 am

P17-6 Evaluating iBall—An Intuitive Interface and Assistive Audio Mixing Algorithm for Live Football Events—*Henry Bourne, Joshua D. Reiss*, Queen Mary University of London, London, UK

Mixing the on-pitch audio for a live football event is a mentally challenging task requiring the experience of a skilled operator to capture all the important audio events. iBall is an intuitive interface coupled with an assistive mixing algorithm ➥

that aids the operator in achieving a comprehensive mix. This paper presents the results of subjective and empirical evaluation of the system. Using multiple stimulus comparison, event counting, fader tracking, and cross-correlation of mixes using different systems, this paper shows that lesser skilled operators can produce more reliable, more dynamic, and more consistent mixes using iBall than when mixing using the traditional fader-based approach, reducing the level of skill required to create broadcast quality mixes.
Convention Paper 9012

10:30 am

P17-7 A Definition of XML File Format and an Editor Appliction for Korean Traditional Music Notation System—*Keunwoo Choi, Yong Ju Lee, Yongju Lee, Kyeongok Kang*, Electronics and Telecommunications Research Institute (ETRI), Daejeon, Korea

In this paper a computer-based system for representing Jeongganbo, the Korean traditional music notation system, is introduced. The system consists of an XML Document Type Definition, an editor application, and a converter into MusicXML. All information of Jeongganbo, including notes, directions, playing techniques, and lyrics, are encoded into XML using the grammar of the proposed Document Type Definition. In addition, users can create and edit Jeongganbo XML files using the proposed editor and export them as a MusicXML file. As a result, users can represent, edit, and share the musical content of Korean traditional music in the digital domain, as well as analyze score-based content for information retrieval.
Convention Paper 9013

10:30 am

P17-8 The Structure of Noise Power Spectral Density-Driven Adaptive Post-Filtering Algorithm—*Jie Wang,*[1] *Chengshi Zheng,*[2,3] *Chunliang Zhang,*[1] *Yueyan Sun*[1]
[1]Guangzhou University, Guangzhou, China
[2]Chinese Academy of Sciences, Beijing, China
[3]Chinese Academy of Sciences, Shanghai, China

Conventional post-filtering (CPF) algorithms often use a fixed filter bandwidth to estimate the auto-spectra and the cross-spectrum. This paper first studies the drawback of the CPF algorithms under the stochastic model and discusses the ways to improve the performances of the CPF algorithms. To improve noise reduction without introducing audible speech distortion, we propose a novel spectral estimator, which is based on the structure of the noise power spectral density (NPSD). The proposed spectral estimator is applied to improve the performance of the CPF. Experimental results verify that the proposed algorithm is better than the CPF algorithms in terms of the segmental signal-to-noise-ratio improvement and the noise reduction, especially the noise reduction, is about 6 dB higher than the CPF.
Convention Paper 8943

Workshop 26 — Sunday, Oct. 20
10:30 am – 12:00 noon — Room 1E13

FX DESIGN PANEL: REVERB

Chair: **Josh Reiss**, Queen Mary University of London, London, UK

Panelists: *Michael Carnes*, Exponential Audio, Cottonwood Heights, UT, USA
Casey Dowdell, Bricasti

Meet the designers whose talents and philosophies are reflected in the products they create, driving sound quality, ease of use, reliability, price, and all the other attributes that motivate us to patch, click, and tweak their effects processors.

Sound for Picture 7 — Sunday, Oct. 20
10:30 am – 12:30 pm — Room 1E11

SOUND FOR “A DEADLIEST CATCH”—REALITY IS HARD WORK

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

Panelists: *Bob Bronow*, Max Post, Burbank, CA; Audio Cocktail
Josh Earl, Original Productions, Burbank, CA, USA
Sound Crew from “Deadliest Catch”

Television has seen the development of a new category of TV program—the "reality" show. While many of these shows are TV fluff, one of the first was set around the dangerous profession of crab fishing in the Bering Sea. This hit rated show is one of the most difficult and challenging productions for not only the fisherman but for the capture and mixing of the soundtrack. We present two of the key Emmy-winning sound professionals in this workshop.

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

Workshop 27 — Sunday, Oct. 20
11:00 am – 12:30 pm — Room 1E10

DSP STUDIO MONITORING

Chair: **Dave Malekpour**, Professional Audio Design Inc., Pembroke, MA, USA

Panelists: *Michael Blackmer*, Professional Audio Design
David Kotch, Criterion Acoustics
Andrew Munro, Munroe Acoustics, Dynaudio Acoustics
Carl Nappa, Extreme Institute by Nelly
Paul Stewart, Genelec USA

Monitoring systems have evolved over the last decade to include DSP systems for both room correction and system voicing. We will explore how this affects the listening environment for control rooms, mastering, and critical listening environments. We will examine room measurements, and correction curves employed with subsequent results. The panelists will discuss their experiences and show real world examples of how this has worked and not worked in applications.

We will look at how this impacts the users listening environment for accuracy and sonic quality. We will also explore how it affects studio design and how it is implemented by manufacturers to get the results they want from the speaker design.

Tutorial 21 **Sunday, Oct. 20**
11:00 am – 12:00 noon **Room 1E09**

NATIONAL RECORDING PRESERVATION PLAN: AUDIO PRESERVATION/ARCHIVING: DISC PLAYBACK STYLUS SELECTION

Presenter: **Marcos Sueiro Bal**, WNYC / New York Public Radio, New York, NY, USA; Masterdisk, New York, NY, USA

Draft document AES-16id-2010 on stylus selection states that stylus selection "will always be subjective," but there may be mountains of data waiting to help us with our selection. As a proof of concept this tutorial will attempt to establish whether there is a correlation between playback stylus size and year of recording using data from a recent project transferring over1500 lacquer disc sides from radio station WNYC. This kind of research aligns well with the NRPP's recommendation to develop "scientific and technical research leading to the development of new technologies to recover, reformat, and preserve audio recording media."

Live Sound Seminar 12 **Sunday, Oct. 20**
11:00 am – 1:00 pm **Room 1E12**

AN INTERVIEW WITH DAVE NATALE

Presenters **Keith Clark**, ProSoundWeb
Dave Natale, Audio Resource Group, Inc., Lancaster, PA, USA

Dave Natale is a veteran of over 30 years mixing front of house for the biggest names in concert touring, having spent much of that time working for Clair Brothers Audio (now Clair Global). Dave will discuss his career, knowledge gained along the way, and what all FOH mixers should know and strive for. A Q&A will follow the interview. *Keith Clark* is the Editor of ProSoundWeb and has been involved in the pro audio publishing field for more than 20 years.

Network Audio Session 5 **Sunday, Oct. 20**
11:00 am – 12:30 pm **Room 1E08**

X192 / AES67: HOW THE NEW NETWORKED AUDIO INTEROPERABILITY STANDARD WAS DESIGNED

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

Panelists: *Kevin Gross*, AVA Networks, Boulder, CO, USA
Stefan Heinzmann
Andreas Hildebrand, ALC NetworX, Munich, Germany
Gints Linis, University of Latvia, Riga, Latvia

It is said, to really understand a solution, you must clearly understand the problems it is solving. The nature of a technical specification like AES67 is that it is the end result of much discussion and deliberation. However, many of the intentions, the tradeoffs that were made, and an understanding of what problems were being solved, are not fully contained in the resulting document.

This panel will present the background of a number of the decisions that were made and embodied into AES67. It will describe the problems that were targeted to be solved, as best as they were understood. What were some of the difficult tradeoffs?

Networked audio will be new for some users, while some of the roots of the networked audio experience of the members of X192 go back 20 years. Given a proverbial clean slate by the AES, come listen to the reasons why the choices in AES67 were made.

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

Product Design Session 6 **Sunday, Oct. 20**
11:00 am – 12:00 noon **Room 1E14**

RAZORBILL: A HIGH-RESOLUTION, HIGH-CAPACITY, LOW-POWER AUTONOMOUS ACOUSTIC RECORDER

Presenter: **Peter Marchetto**, Cornell University, Ithaca, NY, USA

High-fidelity, battery powered acoustic recorders are getting ever smaller and more ubiquitous. Most commercially available devices focus on increasing bit depth and sampling frequency, while de-emphasizing low power consumption. However, there is an increasing need for very long duration continuous recordings for biological and environmental studies. To fill this gap, we have developed and tested a new autonomous recorder capable of recording at a 16-bit depth and a sampling frequency of 48 kHz for over a month on only three alkaline batteries. This unit has a dynamic range of over 135 dB re 20 µPa when combined with a standard electret microphone, with a THD+N figure of <1.5 %. Data is stored on standard Secure Digital (SD) cards for easy offloading. The newest version under development will increase the recording capacity to a 24-bit depth and 96kHz sampling frequency, with the aim of maintaining the month-long duration.

Special Event
PLATINUM MASTERING: THE STATE OF MASTERING – 2013
Sunday, October 20, 11:00 am – 1:00 pm
Room 1E15/16

Moderator: **Bob Ludwig**, Gateway Mastering Studios, Inc., Portland, ME, USA

Panelists: *Greg Calbi*, Sterling Sound, New York, NY, USA
Darcy Proper, Wisseloord Studios, Hilversum, The Netherlands
Douglas Sax, The Mastering Lab, Ojai, CA, USA
Tim Young, Metropolis Mastering, London, UK

Ten years ago top mastering studios generally mastered and created final production masters for only the Compact Disc. Now we commonly create production masters for CDs, Downloads, files for streaming, special "Mastered for iTunes" downloads, and high resolution files for vinyl disc cutting, HDtracks, and Pure Audio Blu-ray masters.

Our Platinum Panelists will talk about the ramifications of State-of-Mastering in 2013 and what the future may hold. We will include some special sound demonstrations.

Knowledge Center
Sunday, October 20, 11:00 am – 12:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA ➡

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Sunday, Oct. 20 11:00 am Room 1E04
Technical Committee Meeting: Acoustics and Sound Reinforcement

Systems Sound Symposium 2
Sunday, October 20, 11:15 am – 12:30 pm
Stage

BIG IN NYC—HOW THE METRO AREA'S PREMIER AV INSTALLATIONS INFLUENCE THE INDUSTRY

As the center of the media universe, New York is home to early adopters of high-end audio systems. How do these top-level sound systems in clubs, performing arts venues, retail outlets, restaurants, stadiums, houses of worship, and boardrooms represent the cutting-edge of systems implementation?

Knowledge Center
Sunday, October 20, 11:30 am – 12:30 pm
Room 1E06

PMC "MASTERS OF AUDIO": A JOE FERLA RETROSPECTIVE WITH SPARS

Presenter: **Joe Ferla**, Joe Ferla, Stamford, NY, USA

Joe Ferla, a five-time Grammy Award recipient and renowned engineer to some of the best musicians in the industry, will be presenting tracks for listening from his extensive discography. After each track is played he will answer any questions regarding the track such as how he recorded the artist, which mics he used on what instrument, etc. Songs from artists such as Roberta Flack (that started his career off), David Sanborn, John Scofield, Eliane Elias, Dave Douglas, Christian McBride, Charlie Hunter, and others will be featured in this unique listening experience.

Tutorial 22 Sunday, Oct. 20
12:00 noon – 1:00 pm Room 1E09

THE VINYL FRONTIER

Moderator: **Chris Mara**, Welcome to 1979, Nashville, TN, USA

Panelists: *Alex DeTurk*, Masterdisk, New York, NY, USA
Paul Gold, Salt Mastering, New York, NY, USA
Cameron Henry, Welcome to 1979, Nashville, TN, USA
Scott Hull, Masterdisk, New York, NY, USA

There is little doubt that vinyl records are in the midst of a resurgence. First hand knowledge of the vinyl mastering process is in high demand, yet short supply. This session is dedicated to discuss the truths of vinyl mastering, lacquer cutting, and vinyl production while also dispelling myths.

Student/Career Development Event
STUDENT DELEGATE ASSEMBLY MEETING—PART 2
Sunday, October 20, 12:00 pm – 1:30 pm
Room 1E13

Moderator: **Colin Pfund**

Participants: *Marija Kovacina*
Andrea Pepper
Simon-Claudius Wystrach
Magdalena Plewa
Kyle P. Snyder

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

Workshop 28 Sunday, Oct. 20
12:30 pm – 1:30 pm Room 1E11

PRACTICAL TECHNIQUES FOR RECORDING AMBIENCE IN SURROUND

Chair: **Helmut Wittek**, SCHOEPS Mikrofone GmbH

Panelist: *Mike Williams*, Sounds of Scotland

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 who have to choose an ambience setup in practice.

Special Event
LUNCHTIME KEYNOTE:
STUDIO OF THE FUTURE: 2020–2050
Sunday, October 20, 12:30 pm – 1:30 pm
Room 1E07

Presenter: **John La Grou**, Millennia Music & Media Systems

A brief look at the evolution of audio electronics, a theory of innovation, and a sweeping vision for the next forty years of audio production technology. Informed by the growth theories of Moore, Cray, and Kurzweil, we project the next forty years of professional audio products, production techniques, and delivery formats.

Knowledge Center
Sunday, October 20, 12:30 pm – 4:00 pm
Booth 2738

RADIO ACTIVE DESIGNS: ASK AN RF EXPERT!

Presenter: **James Stoffo**, Radio Active Designs, Key West, FL, USA

Now's your chance to ask frequency coordinator par excellance James Stoffo anything you want about wireless frequencies, dealing with the white space crunch, your wireless setup or Radio Active Designs' suite of products! He'll be at the booth during the indicated times, or you can tweet to @gothamsound #RADanswers all weekend!

If you've turned on a TV in the last 30 years, you've probably heard James Stoffo's work. His credits include 14 consecutive Superbowls, the Olympics, the World Cup, multiple NBA All Star Games, NBA Finals, and Rose Bowl games. James is also one of the principals of Radio Active Designs, a new manufacturer of innovative wireless communication devices meant to alleviate spectral congestion while providing excellent audio quality, reliability, and RF performance.

Knowledge Center
Sunday, October 20, 12:45 pm – 2:00 pm
Room 1E06

PMC "MASTERS OF AUDIO": ISTANDARD PRODUCERS

iStandard Producers Teams up with PMC Audio and Platinum Producer !llmind (Producer for Kanye West, 2Chainz, Eminem, 50 Cent, and More) to bring you the BLAP Celebrity Beat Cypher. Often done at iStandard "Beat Camp" at SAE in various cities, powered by PMC Audio, BLAP is a round robin type producer showcase where your favorite Hip Hop producers play music from their catalog as well as new beats exclusive to the audience. Also confirmed are multi-Grammy Award Winning / multi-platinum Producer Rockwilder (producer for Jay-z, 50 Cent, Pink, Missy, Redman, and More) and Jimi Kendrix, multi-platinum producer for 50 Cent, Ja Rule, Jay-Z, Tupac, and More.

Workshop 29 **Sunday, Oct. 20**
1:00 pm – 3:00 pm **Room 1E08**

MIKING FOR PA

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

Panelists: *Giacomo De Caterini*, Casal Bauer/ Accademia di Santa Cecilia, Rome, Italy
Henrik Kjelin, Complete Vocal Institute, Copenhagen, Denmark
Cathrine Sadolin, Complete Vocal Institute, Copenhagen, Denmark
Nevin Steinberg, Nevin Steinberg Sound Design, New York, NY, USA

Miking for PA is a very important task. Providing amplification to the spoken voice or the acoustical music instrument requires good knowledge about the sound source, about the PA-system, about the monitoring system—and about the microphones. This workshop takes you through some of the important issues and decisions when selecting the microphone with regards to peak level capacity, sensitivity, directivity, frequency response, sensitivity, etc. Getting balance, getting definition, getting the right timbre or "sound"—and still avoiding acoustical feedback, that's the thing. Recognized engineers and sound designers will generously share their experiences from their work on the stages. Warning: Some of the attendees may pick up ideas that will change their habits forever…

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

Historical Event
A CONTRIBUTION TO THE HISTORY OF FIELD TAPE RECORDING, 1939–1940
Sunday, October 20, 1:00 pm – 2:00 pm
Room 1E14

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

Digital Transfer of The Armando Leça Folk Music Collection

Nadja Wallaszkovits will discuss the restoration, transfer, and digitization of a unique collection of folk music recorded during 1939–1940 in the rural areas and mountain villages of Portugal by the folklorist Armando Leça, in collaboration with National Radio (Emissora Nacional). Her presentation will open with a short introduction of this important field research project, which resulted in the first known collection of recordings documenting rural musical practices from nearly all the regions of Portugal. Thereafter, she will present a historical overview of early magnetic tape developments and the birth of audio tape recorder technology, focusing on the characteristics of the individual tape machine used by Armando Leça. The problems of carrier handling, restoration, and transfer of these valuable original tapes will be discussed, along with the judicial use of signal enhancement during the playback process.

Systems Sound Symposium 3
Sunday, October 20, 1:00 pm — 2:15 pm
Stage

AV/IT CONVERGENCE—THE PRACTICALITIES OF NETWORKED AUDIO IN PERMANENT INSTALLATIONS

The basics on digital audio networking in applications large and small. How are things shaping up in the real world?

Knowledge Center
Sunday, October 20, 1:00 pm – 2:30 pm
T-1 Exhib. Fl.

SOUNDCRAFT MWP SI TRAINING

Presenter: **Tom Der**, Soundcraft USA

Join Soundcraft for hands-on training with the popular Si Expression and Si Performer series of digital audio consoles. Learn how to operate these consoles from factory expert and product specialist Tom Der, in a 90 minute training session right on the show floor. As well as surface operation, system configuration, and facility integration will be discussed and demonstrated in detail; learn how to multitrack record via the latest in network technology such as Dante, MADI, and USB Firewire and how ➥

you can finally solve the master-slave issue between two consoles! Everyone is welcome and you can also experience the 53-ft demonstration trailer, kitted out with the latest technologies from Harman!

Special Event
ERA OF THE ENGINEER
Sunday, October 20, 1:30 pm – 3:00 pm
Room 1E15/16

Presenter: **Young Guru**

Revered as "The Sound of New York," Young Guru possesses over a decade of experience in sound engineering and production for the acclaimed Roc-A-Fella Records and Def Jam Recordings. Through his lecture and demo series, #eraoftheengineer, Guru examines the recent emergence of a new generation of do-it-yourself engineers, analyzing and demonstrating what it means for the culture at large.

Session P18 **Sunday, Oct. 20**
2:00 pm – 4:00 pm **Room 1E07**

PERCEPTION—PART 2

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

2:00 pm

P18-1 Negative Formant Space, "O Superman," and Meaning—*S. Alexander Reed*, Ithaca College, New York, NY, USA

This in-progress exploration considers both some relationships between sounding and silent formants in music and the compositional idea of spectral aggregates. Using poststructuralist lenses and also interpretive spectrographic techniques informed by music theorist Robert Cogan, it offers a reading of Laurie Anderson's 1982 hit "O Superman" that connects the aforementioned concerns of timbre with interpretive processes of musical meaning. In doing so, it contributes to the expanding musicological considerations of timbre beyond its physical, psychoacoustic, and orchestrational aspects.
Convention Paper 9014

2:30 pm

P18-2 The Effects of Interaural Level Differences Caused by Interference between Lead and Lag on Summing Localization—*M. Torben Pastore, Jonas Braasch*, Rensselaer Polytechnic Institute, Troy, NY, USA

Traditionally, the perception of an auditory event in the summing localization range is shown as a linear progression from a location between a coherent lead and lag to the lead location as the delay between them increases from 0-ms to approximately 1-ms. This experiment tested the effects of interference between temporally overlapping lead and lag stimuli on summing localization. We found that the perceived lateralization of the auditory event oscillates with the period of the center frequency of the stimulus, unlike what the traditional linear model would predict. Analysis shows that this is caused by interaural level differences due to interference between a coherent lead and lag.
Convention Paper 9015

3:00 pm

P18-3 Paired Comparison as a Method for Measuring Emotions—*Judith Liebetrau,*[1,2] *Johannes Nowak,*[1,2] *Thomas Sporer,*[1,2] *Matthias Krause,*[1] *Martin Rekitt,*[1] *Sebastian Schneider*[1]
[1]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[2]Ilmenau University of Technology, Ilmenau, Germany

Due to the growing complexity and functionality of multimedia systems, quality evaluation becomes a cross-disciplinary task, taking technology-centric assessment, as well as human factors into account. Undoubtedly, emotions induced during perception, have a reasonably high influence on the experienced quality. Therefore the assessment of users' affective state is of great interest for development and improvement of multimedia systems. In this work problems of common assessment methods as well as newly applied methods in emotional research will be displayed. Direct comparison of stimuli as a method intended for faster and easier assessment of emotions is investigated and compared to previous work. The results of the investigation showed that paired comparison seems inadequate to assess multidimensional items/problems, which often occur in multi-media applications.
Convention Paper 9016

3:30 pm

P18-4 Media Content Emphasis Using Audio Effect Contrasts: Building Quantitative Models from Subjective Evaluations—*Xuchen Yang, Zhe Wen, Gang Ren, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

In this paper we study media content emphasis patterns of audio effects and construct their quantitative models using subjective evaluation experiments. The media content emphasis patterns are produced by contrasts between effect-sections and non-effect sections, which change the focus of audience attention. We investigate media emphasis patterns of typical audio effects including equalization, reverberation, dynamic range control, and chorus. We compile audio test samples by applying different settings of audio effects and their permutations. Then we construct quantitative models based on the audience rating of the "subjective significance" of test audio segments. Statistical experiment design and analysis techniques are employed to establish the statistical significance of our proposed models.
Convention Paper 9017

Knowledge Center
Sunday, October 20, 2:00 pm – 3:00 pm
Room 1E06

PMC "MASTERS OF AUDIO": PMC @ THE MOVIES

As the leading provider of high resolution monitors in the composer and movie scoring market, PMC will present the original 5.1 music tracks from movies like Turbo, Pacific Rim, Game of Thrones, Wreck it Ralph, X-Men, and many more! Imagine what a movie would sound like without the great music!

Workshop 30 **Sunday, Oct. 20**
2:30 pm – 4:00 pm **Room 1E14**

MASTERING OUR FUTURE MUSIC

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

Panelists: *Mandy Parnell*, Black Saloon Studios, London, UK
Michael Romanowski, Michael Romanowski Mastering, San Francisco, CA, USA
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, 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 regard to high resolution audio, secure digital content, and the inclusion of album artwork and interactive features. Each of these contemporary initiatives has an impact on the way the audio is mastered and finalized.

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

Workshop 31 **Sunday, Oct. 20**
2:30 pm – 4:00 pm **Room 1E13**

BEAM STEERING LOUDSPEAKERS AND LINE ARRAYS

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

Panelists: *Stefan Feistel*, AFMG
Ralph Heinz, Renkus Heinz
Philippe Robineau, Tannoy, Coatbridge, Scotland, UK
Evert Start, Duran Audio, Zaltbommel, The Netherlands
Ambrose Thompson, Martin Audio

Beam Steered Line Arrays have been commercially available for more than 15 years. Although originally intended for and restricted to speech applications, in the last few years, full range music systems have also started to enter the market. This workshop will discuss the technology behind the systems, their application, and potential limitations. The panel members all have a wide experience of the steered arrays and so are able to cover all aspects of their design and application. The workshop will include a number of case histories and aims to get anyone not familiar with the technology up to speed as well as providing experienced users with some answers to long standing questions.

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

Tutorial 23 **Sunday, Oct. 20**
2:30 pm – 4:00 pm **Room 1E09**

WFMU'S ADVENTURES IN 24/7 ARCHIVING: USING ON-DEMAND PROGRAMS AND CROWD-SOURCED METADATA TO GROW A RADIO AUDIENCE

Presenters: **Tom Miller**, New York University, New York, NY, USA
Ken Freedman, WFMU, Jersey City, NJ, USA

WFMU, the most renowned freeform radio station in America, has archived itself 24 hours a day for ten years. Unlike any other radio station, WFMU keeps all its audio archives available to the public at all times. This is made possible by direct licenses from over 2000 record labels and artists. While many radio stations face the challenge of digitizing their physical libraries, WFMU has never attempted to do that, opting instead to build an internal digital library as well as the Free Music Archive, an interactive library of high-quality, legal audio downloads. Ken Freedman, WFMU Station Manager/Program Director, and sound/media anthropologist Tom Miller discuss these innovations, issues, and technical challenges including crowd-sourced metadata, trans-coding projects, and coordinating various streaming and archiving formats.

Live Sound Seminar 13 **Sunday, Oct. 20**
2:30 pm – 4:30 pm **Room 1E12**

AUDIO FOR CORPORATE PRESENTATIONS

Chair: **Michael (Bink) Knowles**, Freelance Engineer, Oakland, CA, USA

Panelists: *Paul Bevan*
Bruce Cameron, House to Half Inc., Carmel, NY, USA
Lee Kalish, Positive Feedback, Kingston, NY, USA

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

Product Design Session 7 **Sunday, Oct. 20**
2:30 pm – 4:30 pm **Room 1E10**

IS YOUR EQUIPMENT DESIGN A NOISE PROBLEM WAITING TO HAPPEN?

Presenter: **Bill Whitlock**, Jensen Transformers, Inc., Chatsworth, CA, USA; Whitlock Consulting, Oxnard, CA, USA

A design goal for all audio equipment is freedom from hum and buzz. But AC power normally creates a system environment of ground voltage differences. While a balanced interface is the first line of defense against this noise source, the balanced interface itself is very poorly understood by most engineers ... and practical aspects of its design are rarely taught in engineering schools. This leads engineers to design balanced input circuits that perform impressively in the lab but exhibit poor noise ➡

rejection in real-world systems. To make matters worse, internal equipment grounding schemes are often thoughtlessly designed. Two common results are noise coupled via cable shield connections, known as the “pin 1 problem,” and by the AC power cord (so-called “sensitive” equipment). These and other design pitfalls, and how to avoid them, are the focus of this class.

Systems Sound Symposium 4
Sunday, October 20, 2:45 pm — 4:00 pm
Stage

BEYOND BACKGROUND MUSIC—DESIGNING SONICALLY DRIVEN SPACES

Once an afterthought, audio and acoustics are taking a more significant role in project design.

Student/Career Development Event
STUDENT RECORDING CRITIQUES
Sunday, October 20, 3:00 pm – 4:00 pm
Room 1E06

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

Students! Bring your stereo or surround projects to these non-competitive listening sessions and a panel will give you valuable feedback and comments on your work! Students should sign-up for time slots at the first SDA meeting, on a first come, first served basis. Bring your stereo or 5.1 work on CD, DVD, memory-stick, or hard disc, as clearly labeled 44.1 KHz WAVE or AIFF files. Finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. The Student Recording Critiques are generously sponsored by PMC, and you get to hear your work on some amazing loudspeakers!