AES 123rd Convention Program

October 5 – 8, 2007

Jacob Javits Convention Center, New York, NY

Special Event
LIVE SOUND SYMPOSIUM: SURROUND LIVE V
Delivering the Experience
Thursday, October 4, 9:00 am – 4:00 pm
Broad Street Ballroom
41 Broad Street, New York, NY 10004

Preconvention Special Event; additional fee applies

Chair: **Frederick J. Ampel**, Technology Visions, Overland Park, KS, USA

Panelists: *Kurt Graffy*
Fred Aldous
Randy Conrod
Jim Hilson
Michael Nunan
Michael Pappas
Tom Sahara
beyerdynamic, Neural Audio, and others

Once again the extremely popular Surround Live event returns to AES's 123rd Convention in New York City. Created and Produced by Frederick Ampel of Technology Visions with major support from the Sports Video Group, this marks the event's fifth consecutive workshop exclusively provided to the AES.

Using a specially calibrated Klein & Hummel Studio monitor system configured for expanded 5.1 playback, along with a DigiCo console, acoustic matrix processing from TiMax, and professional DVD playback hardware, Surround Live is a stunning audio experience for all who attend. Major sponsorship for the event is provided by The Sports Video Group, Sennheiser USA, beyerdynamic, Neural Audio, DigiCo, and Harris Corporation.

This year the Sports Video Group will sponsor and provide a special panel of experts, including Ron Scalise of ESPN, Bob Seidel and Bruce Goldfeder of CBS, and Jim Starzynski of NBC, for an in-depth discussion "Surround Operational Issues—What Happens When It Leaves the Truck?" moderated by Ken Kerschbaumer of Sports Video Group.

The program is scheduled to include demonstrations and presentations from:

- Kurt Graffy of ARUP Acoustics, San Francisco, CA, USA—Keynote Speaker
- Fred Aldous, Fox Sports
- Randy Conrod, Harris Corporation
- Jim Hilson, Dolby Laboratories
- Michael Nunan, CTV Television
- Michael Pappas KUVO Radio
- Tom Sahara Turner Networks
- beyerdynamic, Neural Audio, and others

Program:

8:15 am – Registration and Continental Breakfast
9:00 am – Event Introduction – Frederick Ampel
9:10 am – Andrew Goldberg – K&H System Overview
9:20 am – The Why and How of Surround – Kurt Graffy Arup
9:50 am – Coffee Break
10:00 am – Neural Audio Overview
10:10 am – TiMax Overview and Demonstration
10:20 am – Fred Aldous – Fox Sports
10:55 am – Jim Hilson – Dolby Labs
11:40 am – Mike Pappas – KUVO Radio
12:25 pm – Lunch
1:00 pm – Tom Sahara – Turner Networks
1:40 pm – Sports Video Group Panel – Ken Kirschenbaumer
2:45 pm – Afternoon Break
3:00 pm – beyerdynamic – Headzone Overview
3:10 pm – Mike Nunan – CTV Specialty Television, Canada
4:00 pm – Q&A; Closing Remarks

PLEASE NOTE: PROGRAM SUBJECT TO CHANGE PRIOR TO THE EVENT. FINAL PROGRAM WILL DEPEND ON PRESENTER AVAILABILITY AND SCHEDULES. SPACE IS LIMITED TO THE FIRST 200 WHO REGISTER.

Standards Committee Meeting **Thursday, October 4**
1:30 pm **Room 1E02**

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

Session P1 **Friday, October 5**
9:00 am – 12:00 noon **Room 1E07**

PERCEPTION, PART 1

Chair: **William Martens**, McGill University, Montreal, Quebec, Canada

9:00 am

P1-1 **Room Reflections Misunderstood?**—*Siegfried Linkwitz*, Linkwitz Lab, Corte Madera, CA, USA

In a domestic living space a 2-channel monopolar and a dipolar loudspeaker system are compared for perceived differences in their reproduction of acoustic events. Both sound surprisingly similar and that is further enhanced by extending dipole behavior to frequencies above 1.4 kHz. The increased bandwidth of reflections is signifi-

cant for spatial impression. Measured steady-state frequency response and measured reflection patterns differ for the two systems, while perceived sound reproduction is nearly identical in terms of timbre, phantom image placement, and sound stage width. The perceived depth in the recording is greater for the dipole loudspeaker. Auditory pattern recognition and precedence effects appear to explain these observations. Implications upon the design of loudspeakers, room treatment, and room equalization are discussed.
Convention Paper 7162

9:30 am

P1-2 Aspects of Reverberation Echo Density—*Patty Huang, Jonathan Abel*, Stanford University, Stanford, CA, USA

Echo density, and particularly its time evolution at the reverberation impulse response onset, is thought to be an important factor in the perceived time domain texture of reverberation. In this paper the psychoacoustics of reverberation echo density is explored using reverberation impulse responses synthesized via a Poisson process to have a variety of static and evolving echo densities. In addition, a recently proposed echo density measure called the normalized echo density, or NED, is explored, and related via a simple expression to echo density specified in echoes per second using echo patterns with static echo densities. A continuum of perceived time-domain texture was noted, from “sputtery” around 100 echoes per second to “smooth” above about 20,000 echoes per second, at which point it was perceptually identical to Gaussian noise. The character of the reverberation impulse response onset was explored for various rates of echo density increase, and ranged from “sputtery” for long mixing times to “instantly smooth” for short mixing times.
Convention Paper 7163

10:00 am

P1-3 Localization in Spatial Audio—From Wave Field Synthesis to 22.2—*Judith Liebetrau,*[1] *Thomas Sporer,*[1] *Thomas Korn*,[1] *Kristina Kunze,*[2] *Christoph Man,*[2] *Daniel Marquard,*[2] *Timo Matheja,*[2] *Stephan Mauer,*[2] *Thomas Mayenfels,*[2] *Robert Möller,*[2] *Michael-Andreas Schnabel,*[2] *Benjamin Slobbe,*[2] *Andreas Überschär*[2]
[1]Fraunhofer IDMT, Ilmenau, Germany
[2]Technical University of Ilmenau, Ilmenau, Germany

Spatial audio reproduction used to concentrate on systems with a low number of loudspeakers arranged in the horizontal plane. Wave Field Synthesis (WFS) and NHK's 22.2 two systems promise better localization and envelopment. Comparisons of 22.2 with 5.1 concerning spatial attributes on one hand, and evaluation of spatial properties of WFS on the other hand have been published in the past, but different methods have been used. In this paper a listening test method is presented that is tailored on the evaluation of localization of 3-D audio formats at different listener positions. Two experiments have been conducted. In the first experiment the localization precision of 22.2 reproduction was evaluated. In a second experiment the localization precision in the horizontal plane as a function of spatial sampling was studied.
Convention Paper 7164

10:30 am

P1-4 Thresholds for Discriminating Upward from Downward Trajectories for Smooth Virtual Source Motion within a Sagittal Plane—*David H. Benson, William L. Martens, Gary P. Scavone*, McGill University, Montreal, Quebec, Canada

In virtual auditory display, sound source motion is typically cued through dynamic variations in two types of localization cues: the inter-aural time delay (ITD) and binaural spectral cues. Generally, both types of cues contribute to the perception of sound source motion. For certain spatial trajectories, however, namely those lying on the surfaces of cones of confusion, ITD cues are absent, and motion must be inferred solely on the basis of spectral variation. This paper tests the effectiveness of these spectral cues in eliciting motion percepts. A virtual sound source was synthesized that traversed sections of a cone of confusion on a particular sagittal plane. The spatial extent of the source’s trajectory was systematically varied to probe directional discrimination thresholds.
Convention Paper 7165

11:00 am

P1-5 Headphone Transparification: A Novel Method for Investigating the Externalization of Binaural Sounds—*Alastair Moore,*[1] *Anthony Tew*,[1] *Rozenn Nicol*[2]
[1]University of York, York, UK
[2]France Telecom R&D, Lannion, France

The only way to be certain that binaurally rendered sounds are properly externalized is to compare them to real sound sources in a discrimination experiment. However, the presence of the headphones required for the binaural rendering interfere with the real sound source. A novel technique is presented that uses small compensating signals applied to the headphones at the same time as the real source is active, such that the signals reaching the ears are the same as if the headphones were not present.
Convention Paper 7166

11:30 am

P1-6 On the Sound Color Properties of Wavefield Synthesis and Stereo—*Helmut Wittek*,[1,2] *Francis Rumsey*,[2] *Günther Theile*[3]
[1]Schoeps Mikrofone GmbH, Karlsruhe, Germany
[2]University of Surrey, Guildford, Surrey, UK
[3]Institut für Rundfunktechnik, Munich, Germany

The sound color reproduction properties of wavefield synthesis are analyzed by listening tests and compared with that of stereophony. A novel technique, "OPSI," designed to avoid spatial aliasing is presented and analyzed in theory and practice. Both stereophonic phantom

sources as well as OPSI sources were perceived to be less colored than was predicted by coloration predictors based on the spectral alterations of the ear signals. This leads to the hypothesis that a decoloration process exists for stereophonic reproduction as proposed in the "association model" of Theile.
Convention Paper 7167

Session P2 **Friday, October 5**
9:00 am – 11:00 am **Room 1E16**

SIGNAL PROCESSING, PART 1

Chair: **Duane Wise**, Wholegrain Digital Systems, LLC, Boulder, CO, USA

9:00 am

P2-1 Suppression of Musical Noise Artifacts in Audio Noise Reduction by Adaptive 2-D Filtering—*Alexey Lukin,*[1] *Jeremy Todd*[2]
[1]Moscow State University, Moscow, Russia
[2]iZotope, Inc., Cambridge, MA, USA

Spectral attenuation algorithms for audio noise reduction often generate annoying musical noise artifacts. Most existing methods for suppression of musical noise employ a combination of instantaneous and time-smoothed spectral estimates for calculation of spectral gains. In this paper a 2-D approach to the filtering of a time-frequency spectrum is proposed, based on a recently developed non-local means image denoising algorithm. The proposed algorithm demonstrates efficient reduction of musical noise without creating "noise echoes" inherent in time-smoothing methods.
Convention Paper 7168

9:30 am

P2-2 Perceptually Motivated Gain Filter Smoothing for Noise Suppression—*Alexis Favrot, Christof Faller*, Illusonic LLC, Chavannes, Switzerland

Stationary noise suppression is widely used, mostly for reducing noise in speech signals or for audio restoration. Most noise suppression algorithms are based on spectral modification, i.e., a real-valued gain filter is applied to short-time spectra of the speech signal to reduce noise. The more noise is to be removed, the more likely are artifacts due to aliasing effects and time variance of the gain filter. A perceptually motivated systematic time and frequency smoothing of the gain filter is proposed to improve quality, considering the frequency resolution of the auditory system and masking. Comparison with a number of previous methods indicates that the proposed noise suppressor performs as good as the best other method, while computational complexity is much lower.
Convention Paper 7169

10:00 am

P2-3 A Novel Automatic Noise Removal Technique for Audio and Speech Signals—
Harinarayanan E.V.,[1] *Deepen Sinha*,[2] *Shamail Saeed*,[1] *Anibal Ferreira*[2,3]
[1]ATC Labs, Noida, India
[2]ATC Labs, Chatham, NJ, USA
[3]University of Porto, Porto, Portugal

This paper introduces new ideas on wideband stationary/nonstationary noise removal for audio signals. Current noise reduction techniques have generally proven to be effective, yet these typically exhibit certain undesirable characteristics. Distortion and/or alteration of the audio characteristics of primary audio sound is a common problem. Also user intervention in identifying the noise profile is sometimes necessary. The proposed technique is centered on the classical Kalman filtering technique for noise removal but uses a novel architecture whereby advanced signal processing techniques are used to identify and preserve the richness of the audio spectrum. The paper also includes conceptual and derivative results on parameter estimation, a description of multi-parameter Signal Activity Detector (SAD), and our new-found improved results.
Convention Paper 7170

10:30 am

P2-4 The Concept, Design, and Implementation of a General Dynamic Parametric Equalizer—
Duane Wise, Wholegrain Digital Systems, LLC, Boulder, CO, USA

The classic operations of dynamics processing and parametric equalization control two separate domains of an audio signal. The operational nature of the two processors give insight to a manner in which they may be combined into a single processor. This integrated processor can perform as the equivalent of a standalone dynamics processor or parametric equalizer, but can also modify the boost and/or cut of an equalizer stage over time following a dynamics curve. The design of a digital version of this concept is discussed herein, along with implementation issues and proposals for their resolutions.
Convention Paper 7171

Workshop 1 **Friday, October 5**
9:00 am – 12:00 noon **Room 1E08**

RECORDING LARGE ENSEMBLES IN MULTICHANNEL

Chair: **Martha DeFrancisco**, McGill University, Montreal, Quebec, Canada

Panelists: *Chuck Ainlay*, Independent Producer/Engineer
Michael Bishop, Telarc
Akira Fukada, NHK
Lawrence Manchester, Bricks & Mortar Music
Everett Porter, Polyhymnia

Born in the European salons of the 18th century aristocracy, the symphony orchestra still plays an important role in the musical life of our times. Capturing the sounds of these and other large ensembles has never ceased to be a challenge to recording engineers and producers worldwide.

Multichannel recording adds a new dimension of realism and enhanced means of expressivity to large-ensemble recordings. This workshop explores the ways in which leading recording engineers deal with the age-old question of how to transport a huge body of sound from a hall into the listeners' private spaces.

Workshop 2 **Friday, October 5**
9:00 am – 11:00 am **Room 1E12/13**

TRENDS OF STORAGE TECHNOLOGIES FOR AUDIO RECORDING AND MASTERING

Chair: **Kimio Hamasaki**, NHK Science & Technical Research Laboratories, Tokyo, Japan

Panelists: *Jeff Levison*, Producer/Consultant, USA
Takehiro Moriya, NTT Communication Science
Labs, Atsugi, Japan
Masaki Shimmachi, Fostex Company
Junich Yoshio, Pioneer Corporation, Japan

Various technologies in terms of storage are currently used for audio recording and mastering. Necessary data rate is also increasing due to the complexity of audio production—such as multichannel sound production. This workshop reviews the current situation of technologies regarding the storage for audio recording and mastering and discuss the future of storage. Panels from both media industries and production will discuss the requirement for the next-generation storage, and next-generation audio recording and mastering systems.

Live Sound Seminar 1 **Friday, October 5**
9:00 am – 12:00 noon **Room 1E09**

CHURCH AND HOUSE OF WORSHIP AUDIO AND ACOUSTICS: CONSIDERATIONS SPECIFIC TO CHURCHES AND HOWS BEFORE, DURING AND AFTER THE DESIGN AND INSTALLATION PHASES

Chair: **Tom Young**, Electroacoustic Design Services, Oxford, CT, USA

Presenters: *Chip Sams*, William Sams Acoustics Inc., Orlando, FL, USA
Neil Thompson Shade, Acoustical Design Collaborative, Ltd., Ruxton, MD, USA
Dale Shirk, Shirk Audio & Acoustics, Terre Hill, PA, USA
Bill Thrasher, Thrasher Design Group, Inc., Kennesaw, GA, USA

One of the industry's largest and most rapidly expanding markets, the House of Worship sector, has been a boon to both professional audio services and to the HOW's spreading of their message. Issues ranging from budget to design and install, service to training and operation will be examined from the perspectives of the client, consultant, contractor, and operator.

Standards Committee Meeting **Friday, October 5**
9:00 am **Room 1E02**

Standards Committee Meeting SC-05-05 Grounding and EMC Practices.

Tutorial 1 **Friday, October 5**
9:30 am – 11:30 am **Room 1E15**

AUDIO ON THE PERSONAL COMPUTER

Presenter: **Elliot Omiya**, Microsoft Corp., Redmond, WA, USA

Confused by WAV, PCM, AAC, WMA, MP3, and other unfathomable TLAs (three letter acronyms)? Ever wondered how audio works in a Windows or Mac PC? This tutorial will cover the past, present, and future of PC audio, including what innards of the PC make audio happen, how the audio industry has contributed to personal computer audio, the mechanisms by which audio streams through a PC, an explanation of various audio formats on the PC, and finally a look at the future of PC audio. No prior knowledge is assumed but detailed questions will be entertained.

Broadcast Session 1 **Friday, October 5**
10:00 am –12:00 noon **Room 1E10**

CONSIDERATIONS FOR FACILITY DESIGN

Chair: **Paul McLane**, Radio World

Presenters: *Dan Braverman*, Radio Systems
Vince Fiola, Studio Technology
Christian Holbrook, WireCAD
Dave Prentice, VCA
John Storyk, Walters Storyk Design Group
Bice Wilson, Meridian Design

There are many details to consider when designing and building a facility. We will discuss: radio, television, recording studio, production, wiring, acoustics, ergonomics, system integration, computer-aided design, and budgeting. The presenters will discuss facilities of various sizes and uses.

Technical Committee Meeting **Friday, October 5**
10:00 am **Room 1E05**

Technical Committee Meeting on Hearing and Hearing Loss Prevention.

Training Session **Friday, October 5**
11:00 am – 1:00 pm **Room 1E04**

BOARD ROOM AUDIO INSTALLATIONS TECHNICAL ISSUE PANEL DISCUSSION

Presenters: **Steven Emspak, Moderator,** Shen, Milsom, & Wilke
Mark Bertrand, Tannoy N. America
Robert Moreau, ClockAudio
Felix Robinson, SPL Integrated Solutions
Craig Richardson, Polycom

Moderating the session will be Steven Emspak, Partner at Shen, Milsom, & Wilke, one of the world's leading authorities and consultants in the A/V world. Focus will be on the technical obstacles and solutions available to properly design and configure a modern boardroom for audio intelligibility and video conferencing. The panel discussion will be followed by a personal networking session sponsored by Tannoy and Polycom.

Technical Committee Meeting **Friday, October 5**
11:00 am **Room 1E05**

Technical Committee Meeting on Archiving, Restoration, and Digital Libraries.

Standards Committee Meeting **Friday, October 5**
11:00 am **Room 1E02**

Standards Committee Meeting SC-05-02 Audio Connectors.

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Friday, October 5, 12:00 noon – 1:30 pm
Room 1E12/13

Opening Remarks:
- Executive Director Roger Furness
- President Wieslaw Woszczyk
- Convention Chair Jim Anderson

Program:
- AES Awards Presentation
- Introduction of Keynote Speaker
- Keynote Address by Daniel Levitin, author of *This Is Your Brain on Music*, Laboratory for Musical Perception, Cognition, and Expertise, McGill University, Montreal, Quebec, Canada

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:

CITATIONS: David Bialik, Francisco Miranda Kirchner
BOARD OF GOVERNORS AWARD: John Strawn
FELLOWSHIP AWARD: Graham Blyth, Bob Ludwig, Neil Muncy, Phil Ramone, Josef Zikovsky
SILVER MEDAL AWARD: John Meyer

Keynote Speaker
This year's Keynote Speaker is Dan Levitin. Levitin's research in musical perception has raised many questions: By the age of 5 we are all musical experts. How is the brain able to do this? How does music affect our emotions? How does memory work? How do people with autism think and why, with ability in math, is there no musical aptitude? Tens of thousands of years ago, why did our ancestors first pick up instruments and begin to play? Does the brain experience a live performance differently from a recorded one? What is the true impact of powerful music on the human brain?

Levitin feels that music triggers the reward centers in our brains and that we are hardwired for music. Is music more fundamental to our species than language? Levitin will explore these and other questions.

Special Event
FREE HEARING SCREENINGS
CO-SPONSORED BY THE AES
AND HOUSE EAR INSTITUTE

Friday, October 5	12:00 noon–6:00 pm
Saturday, October 6	10:00 am–6:00 pm
Sunday, October 7	10:00 am–6:00 pm

Exhibit Hall

Attendees are invited to take advantage of a free hearing screening co-sponsored by the AES and House Ear Institute. Four people can be screened simultaneously in the mobile audiological screening unit located on the exhibit floor. A daily sign-up sheet at the unit will allow individuals to reserve a screening time for that day. This hearing screening service has been developed in response to a growing interest in hearing conservation and to heighten awareness of the need for hearing protection and the safe management of sound. For more information and the location of the hearing screenings, please refer to the *Exhibitor Directory* and posted signs.

Standards Committee Meeting **Friday, October 5**
1:00 pm **Room 1E02**

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

Session P3 **Friday, October 5**
1:30 pm – 5:30 pm **Room 1E07**

PERCEPTION, PART 2

Chair: **Poppy Crum**, Johns Hopkins University School of Medicine, Baltimore, MD, USA

1:30 pm

P3-1 Short-Term Memory for Musical Intervals: Cognitive Differences for Consonant and Dissonant Pure-Tone Dyads—*Susan Rogers, Daniel Levitin*, McGill University, Montreal, Quebec, Canada

To explore the origins of sensory and musical consonance/dissonance, 16 participants performed a short-term memory task by listening to sequentially presented dyads. Each dyad was presented twice; during each trial participants judged whether a dyad was novel or familiar. Nonmusicians showed greater recognition of musically dissonant than musically consonant dyads. Musicians recognized all dyads more accurately than predicted. Neither group used sensory distinctiveness as a recognition cue, suggesting that the frequency ratio, rather than the frequency difference between two tones, underlies memory for musical intervals. Participants recognized dyads well beyond the generally understood auditory short-term memory limit of 30 seconds, despite the inability to encode stimuli for long-term storage.
Convention Paper 7172

2:00 pm

P3-2 Multiple Regression Modeling of the Emotional Content of Film and Music—*Rob Parke, Elaine Chew, Chris Kyriakakis*, University of Southern California, Los Angeles, CA, USA

Our research seeks to model the effect of music on the perceived emotional content of film media. We used participants' ratings of the emotional content of film-alone, music-alone, and film-music pairings for a collection of emotionally neutral film clips and emotionally provocative music segments. Mapping the results onto a three-dimensional emotion space, we observed a strong relationship between the ratings of the film- and music-alone clips, and those of the film-music pairs. Previously, we modeled the ratings in each dimension independently. We now develop models, using stepwise regression, to describe the film-music ratings using quadratic terms and based on all dimensions simultaneously. We demonstrate that while linear-terms are sufficient for single emotion dimensional models, regression models that consider multi-

ple emotion dimensions yield better results.
Convention Paper 7173

2:30 pm

P3-3 Measurements and Perception of Nonlinear Distortion—Comparing Numbers and Sound Quality—*Alex Voishvillo*, JBL Professional, Northridge, CA, USA

The discrepancy between traditional measures of nonlinear distortion and its perception is commonly recognized. THD, two-tone and multitone intermodulation and coherence function provide certain objective information about nonlinear properties of a DUT, but they do not use any psychoacoustical principles responsible for distortion perception. Two approaches to building psychoacoustically-relevant measurement methods are discussed: one is based on simulation of the hearing system's response similar to the methods used for assessment of codec's sound quality. The other approach is based on several ideas such as distinguishing low-level versus high-level nonlinearities, low-order versus high-order nonlinearities, and spectral content of distortion signals that occur below the spectrum of an undistorted signal versus one that overlaps the signal's spectrum or occurs above it. Several auralization examples substantiating this approach are demonstrated
Convention Paper 7174

3:00 pm

P3-4 Influence of Loudness Level on the Overall Quality of Transmitted Speech—*Nicolas Côté*,[1,2] *Valérie Gautier-Turbin*,[1] *Sebastian Möller*[2]
[1]France Télécom R&D, Lannion, France
[2]Berlin University of Technology, Berlin, Germany

This paper consists of a study on the influence of the loudness on the perceived quality of transmitted speech. This quality is based on judgments of particular quality features, one of which is loudness. In order to determine the influence of loudness on perceived speech quality, we designed a two-step auditory experiment. We varied the speech level of selected speech samples and degraded them by coding and packet-loss. Results show that loudness has an effect on the overall speech quality, but that effect depends on the other impairments involved in the transmission path, and especially on the bandwidth of the transmitted speech. We tried to predict the auditory judgments with two quality prediction models. The signal-based WB-PESQ model, which normalizes the speech signals to a constant speech level, does not succeed in predicting the speech quality for speech signals with only impairments due to a non-optimum speech level. However, the parametric E-model, which includes a measure of the listening level, provides a good estimation of the speech quality.
Convention Paper 7175

3:30 pm

P3-5 On the Use of Graphic Scales in Modern Listening Tests—*Slawomir Zielinski, Peter Brooks, Francis Rumsey*, University of Surrey, Guildford, Surrey, UK

This paper provides a basis for discussion of the perception and use of graphic scales in modern listening tests. According to the literature, the distances between the adjacent verbal descriptors used in typical graphic scales are often perceptually unequal. This implies that the scales are perceptually nonlinear and the ITU-R Quality Scale is shown to be particularly nonlinear in this respect. In order to quantify the degree of violation of linearity in listening tests, the evaluative use of graphic scales was studied in three listening tests. Contrary to expectation, the results showed that the listeners use the scales almost linearly. This may indicate that the listeners ignore the meaning of the descriptors and use the scales without reference to the labels.
Convention Paper 7176

4:00 pm

P3-6 A Model-Based Technique for the Perceptual Optimization of Multimodal Musical Performances—*Daniel Valente, Jonas Braasch*, Rensselaer Polytechnic Institute, Troy, NY, USA

As multichannel audio and visual processing becomes more accessible to the general public, musicians are beginning to experiment with performances where players are in two or more remote locations. These co-located or telepresence performances challenge the conventions and basic rules of traditional musical experience. While they allow for collaboration with musicians and audiences in remote locations, the current limitations of technology restricts the communication between musicians. In addition, a telepresence performance introduces optical distortion that can result in impaired auditory communication, resulting in the need to study certain auditory-visual interactions. One such interaction is the relationship between a musician and a virtual visual environment. How does the attendant visual environment affect the perceived presence of a musician? An experiment was conducted to determine the magnitude of this effect. Two pre-recorded musical performances were presented through virtual display in a number of acoustically diverse environments under different relative background lighting conditions. Participants in this study were asked to balance the level of the direct-to-reverberant ratio, and reverberant level until the virtual musician's acoustic environment is congruent with that of the visual representation. One can expect auditory-visual interactions in the perception of a musician in varying virtual environments. Through a multivariate parameter optimization, the results from this paper will be used to develop a parametric model that will control the current auditory rendering system, Virtual Microphone Control (ViMiC), in order to create a more perceptually accurate auditory visual environment for performance.
Convention Paper 7177

4:30 pm

P3-7 Subjective and Objective Rating of Intelligibility of Speech Recordings—
Bradford Gover, John Bradley, National Research Council, Ottawa, Ontario, Canada

Recordings of test speech and an STIPA modulated noise stimulus were made with several microphone systems placed in various locations in a range of controlled test spaces. The intelligibility of the test speech recordings was determined by a subjective listening test, revealing the extent of differences among the recording systems and locations. Also, STIPA was determined for each physical arrangement and compared with the intelligibility test scores. The results indicate that STIPA was poorly correlated with the subjective responses, and not very useful for rating the microphone system performance. A computer program was written to determine STIPA in accordance with IEC 60268-16. The result was found to be highly sensitive to the method of determining the modulation transfer function at each modulation frequency, yielding the most accurate result when normalizing by the premeasured properties of the specific stimulus used.
Convention Paper 7178

5:30 pm

P3-8 Potential Biases in MUSHRA Listening Tests—*Slawomir Zielinski, Philip Hardisty, Christopher Hummersone, Francis Rumsey*, University of Surrey, Guildford, Surrey, UK

The method described in the ITU-R BS.1534-1 standard, commonly known as MUSHRA (MUltiple Stimulus with Hidden Reference and Anchors), is widely used for the evaluation of systems exhibiting intermediate quality levels, in particular low-bit rate codecs. This paper demonstrates that this method, despite its popularity, is not immune to biases. In two different experiments designed to investigate potential biases in the MUSHRA test, systematic discrepancies in the results were observed with a magnitude up to 20 percent. The data indicates that these discrepancies could be attributed to the stimulus spacing and range equalizing biases.
Convention Paper 7179

Session P4 Friday, October 5
1:30 pm – 4:30 pm Room 1E16

SIGNAL PROCESSING, PART 2

Chair: **Alan Seefeldt**, Dolby Laboratories, San Francisco, CA, USA

1:30 pm

P4-1 Loudness Domain Signal Processing—*Alan Seefeldt*, Dolby Laboratories, San Francisco, CA, USA

Loudness Domain Signal Processing (LDSP) is a new framework within which many useful audio processing tasks may be achieved with high quality results. The LDSP framework presented here involves first transforming audio into a perceptual representation utilizing a psychoacoustic model of loudness perception. This model maps the nonlinear variation in loudness perception with signal frequency and level into a domain where loudness perception across frequency and time is represented on a uniform scale. As such, this domain is ideal for performing various loudness modification tasks such as volume control, automatic leveling, etc. These modifications may be performed in a modular and sequential manner, and the resulting modified perceptual representation is then inverted through the psychoacoustic loudness model to produce the final processed audio.
Convention Paper 7180

2:00 pm

P4-2 Design of a Flexible Crossfade/Level Controller Algorithm for Portable Media Platforms—*Danny Jochelson*,[1] *Stephen Fedigan*,[2] *Jason Kridner,*[3] *Jeff Hayes*[3]
[1]Texas Instruments, Inc., Dallas, TX, USA
[2]General Dynamics SATCOM, Richardson, TX, USA
[3]Texas Instruments, Inc., Stafford, TX, USA

The addition of a growing number of multimedia capabilities on mobile devices necessitate rendering multiple streams simultaneously, fueling the need for intelligent mixing of these streams to achieve proper balance and address the tradeoff between dynamic range and saturation. Additionally, the crossfading of subsequent streams can greatly enhance the user experience on portable media devices. This paper describes the architecture, features, and design challenges for a real-time, intelligent mixer with crossfade capabilities for portable audio platforms. This algorithm shows promise in addressing many audio system challenges on portable devices through a highly flexible and configurable design while maintaining low processing requirements.
Convention Paper 7181

2:30 pm

P4-3 Audio Delivery Specification—*Thomas Lund*, TC Electronic A/S, Risskov, Denmark

From the quasi-peak meter in broadcast to sample by sample assessment in music production, normalization of digital audio has traditionally been based on a peak level measure. The paper demonstrates how low dynamic range material under such conditions generally comes out the loudest, and how the recent ITU-R BS.1770 standard offers a coherent alternative to peak level fixation. Taking the ITU-R recommendations into account, novel ways of visualizing short-term loudness and loudness history are presented; and applications for compatible statistical descriptors portraying an entire music track or broadcast program are discussed.
Convention Paper 7182

3:00 pm

P4-4 Multi-Core Signal Processing Architecture for Audio Applications—*Brent Karley, Sergio Liberman, Simon Gallimore*, Freescale Semiconductor, Inc., Austin, TX, USA

As already seen in the embedded computing industry and other consumer markets, the trend in audio signal processing architectures is toward multi-core designs. This trend is expected to con-

tinue given the need to support higher performance applications that are becoming more prevalent in both the consumer and professional audio industries. This paper describes a multi-core audio architectures being promoted to the audio industry and details the various architectural hardware, software, and system level trade-offs. The proper application of multi-core architectures is addressed for both consumer and professional audio applications and a comparison of single core, multi-core, and multi-chip designs is provided based on the authors' experience in the design, development, and application of signal processors.
Convention Paper 7183

3:30 pm

P4-5 Rapid Prototyping and Implementing Audio Algorithms on DSPs Using Model-Based Design and Automatic Code Generation— *Arvind Ananthan*, The MathWorks, Natick, MA, USA

This paper explores the increasingly popular model-based design concept to design audio algorithms within a graphical design environment, Simulink, and automatically generate processor specific code to implement it on target DSP in a short time without any manual coding. The final fixed-point processors targeted in this paper will be Analog Devices Blackfin processor and Texas Instruments C6416 DSP. The concept of model-based design introduced here will be explained primarily using an acoustic noise cancellation system (using an LMS algorithm) as an example. However, the same approach can be applied to other audio and signal processing algorithms; other examples that will be shown during the lecture will include a 3-Band a parametric equalizer, reverberation model, flanging, voice pitch shifting, and other audio effects. The design process starting from a floating point model to easily converting it to a fixed-point model is clearly demonstrated in this paper. The model is then implemented on C6416 DSK board and Blackfin 537 EZ-Kit board using the automatically generated code. Finally, the paper also explains how to profile the generated code and optimize it using C-intrinsics (C-callable assembly libraries).
Convention Paper 7184

4:00 pm

P4-6 Filter Reconstruction and Program Material Characteristics Mitigating Word Length Loss in Digital Signal Processing-Based Compensation Curves Used for Playback of Analog Recordings—*Robert S. Robinson*, Channel D Corporation, Trenton, NJ, USA

Renewed consumer interest in pre-digital recordings, such as vinyl records, has spurred efforts to implement playback emphasis compensation in the digital domain. This facilitates realizing tighter design objectives with less effort than required with practical analog circuitry. A common assumption regarding a drawback to this approach, namely bass resolution loss (word length truncation) of up to approximately seven bits during digital de-emphasis of recorded program material, ignores the reconstructive properties of compensation filtering and the characteristics of typical program material. An analysis of the problem is presented, as well as examples showing a typical resolution loss of zero to one bits. The worst case resolution loss, which is unlikely to be encountered with music, is approximately three bits.
Convention Paper 7185

Workshop 3 — Friday, October 5
1:30 pm – 3:00 pm — Room 1E11

EVALUATION OF SURROUND MAIN MIKINGS FOR CLASSICAL ORCHESTRA

Chair: **Mick Sawaguchi**, Pioneer Corporation

Panelists: *Akira Fukada*, NHK
Hideo Irimajiri, Mainichi Broadcasting Corporation
Toru Kamekawa, Tokyo National University of Fine Arts and Music, Tokyo, Japan
Masayuki Mimura, Yomiuri Telecasting Corporation
Hideaki Nishida, Asahi Broadcasting Corporation

There are many different setups for surround main microphones for classical music and orchestra. But it is very difficult to research and study them practically and academically under identical conditions and judge their performance. Consequently the AES Japan Surround Study Project has been organized and put into practice after one and a half years of preparation. It was organized around 10 broadcasters and 2 universities; 12 manufacturers supported by HBF provided financial support. There were 15 different combinations of main and ambience microphone setups that were recorded on 96 channels independently in Pro Tools HD at 24 bit / 96-kHz. The musical examples were performed by the Osaka Philharmonic Orchestra on September 24–-27, 2006.

In this workshop each individual setup will be played back. Participants will have the opportunity for feedback in a listening test environment, and the data will be collected for subjective evaluation.

This workshop will be presented twice during the convention.

Broadcast Session 2 — Friday, October 5
1:30 pm – 2:30 pm — Room 1E10

BROADCAST TUTORIAL: BUILDING A FACILITY

Presenters: **Allan Black**, WNYC Radio
Edward Haber, WNYC Radio
Steve Shultis, WNYC Radio
James Williamson, WNYC Radio

WNYC New York Public Radio is building out a state-of-the-art broadcast studio and office facility in lower Manhattan, scheduled to go on the air in March 2008. The facility encompasses about 70,000 sq. ft. of upper-floor space for a staff of 200 and includes a technical plant of just under 40 studio spaces plus a 3,500 sq. ft. street-level performance/broadcast venue. This presentation will be in two sections: the first will speak about the program and workflow planning for how we arrived at this studio architecture; and the second will be a close-up look at one aspect of the technical integration of which we were challenged: the design and integration of our SAS Audio broadcast mixing platform with our large-format API music recording console for our two music broadcast control rooms.

Tutorial 2 **Friday, October 5**
1:30 pm – 3:30 pm **Room 1E09**

TINNITUS. JUST ANOTHER BUZZ WORD?

Presenter: **Neil Cherian**
Michael Santucci, Sensaphonics Hearing Conservation, Chicago, IL, USA

Tinnitus is a common yet poorly understood disorder where sounds are perceived in the absence of an external source (phantom). Significant sound exposure with or without hearing loss is the single most common risk factor. Tinnitus can be debilitating, affecting quality of life or even one's ability to function. Given the potential harm of sound in the development of tinnitus, more aggressive and proactive attitudes must be taken. In-ear monitoring strategies further necessitate meaningful conversations regarding hearing awareness, hearing protection, safe standards for listening, and appropriate safeguards for products.

This tutorial introduces the concept of tinnitus, the pertinent anatomy and physiology, the audiologic parameters of tinnitus, current research, guidelines for identifying high risk behaviors, and how to determine that you have a problem.

Master Class 1 **Friday, October 5**
1:30 pm – 3:00 pm **Room 1E08**

HENRY OTT

Presenter: **Henry Ott**, Henry Ott Consultants, Livingston, NJ, USA

Audio Interconnections—Dispelling the Myths

High signal-to-noise ratio is an important goal for most audio systems. AC power connections, however, unavoidably create ground voltage differences, magnetic fields, and electric fields. The class will cover cabling, grounding, balancing, and shielding issues in audio interconnections for electromagnetic compatibility and maximum signal integrity. Design ideas will be given to insure the ability of a device, product, or system to operate properly in its intended electromagnetic environment, without degradation and without being a source of interference.

Henry W. Ott is President and Principal Consultant of Henry Ott Consultants, an Electromagnetic Compatibility (EMC) training and consulting organization. Prior to starting his own consulting company, he was with AT&T Bell Laboratories for thirty years, where he was a Distinguished Member of the Technical Staff. Ott is the author of *Noise Reduction Techniques in Electronic Systems*, is a Life Fellow of the Institute of Electrical and Electronic Engineers (IEEE), and an Honorary Life Member of the IEEE EMC Society. In addition, he is a former Distinguished Lecturer of the EMC Society and continues to lectures widely on the subject of EMC.

Live Sound Seminar 2 **Friday, October 5**
1:30 pm – 3:30 pm **Room 1E15**

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

Chair: **Chris Evans**, Benedum Center

Panelists: *Nathan Allers*, Syntonic Design Group, Milan, NY, USA
Lucas "Rico" Corrubia, Masque Sound
Paul Garrity, Auerbach Assoc.
Dan Gerhard, Freelance Engineer
Dominic Sack, Sound Assoc.
Christopher Sloan, Mixer on "Curtains"

The best intentions of the sound designer don't always fit in with the venue's design 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.

Student Event/Career Development
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Friday, October 5, 1:30 pm – 2:30 pm
Room 1E06

Chair: **Josh Tidsbury**

Vice Chair: **Jose Leonardo Pupo**

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. 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 America Regions, announce the finalists in the recording competition categories, hand out the judges' sheets to the nonfinalists, and announce any upcoming events of the convention. Students and student sections will be given the opportunity to introduce themselves and their past/upcoming activities. In addition, candidates for the SDA election will be invited to the stage to give a brief speech outlining their platform.

All students and educators are invited and encouraged to participate in this meeting. Also at this time there will be the opportunity to sign up for the mentoring sessions, a popular activity with limited space for participation. Election results and Recording Competition and Poster Awards will be given at the Student Delegate Assembly Meeting–2 on Monday, October 8 at 12:30 pm.

Exhibitor Seminar **Friday, October 5**
1:30 pm – 2:30 pm **Room 1E04**

RENKUS HEINZ

Presenters: **Wolfgang Ahnert, Stefan Feistel**

Tools for Measuring and Optimizing Loudspeaker Systems

The Ahnert/Feistel Media Group will present EASERA SysTune, a new software tool for quick and accurate system tuning using live program material while an audience is present. The presenters will discuss actual applications and share their extensive experience with measuring and tuning complex loudspeaker systems for live sound reinforcement.

Technical Committee Meeting **Friday, October 5**
1:30 pm **Room 1E05**

Technical Committee Meeting on Semantic Audio Analysis.

Session P5 **Friday, October 5**
2:00 pm - 3:30 pm **Foyer 1E**

POSTERS: ACOUSTIC MODELING

2:00 pm

P5-1 Modeling of Nonlinearities in Electrodynamic Loudspeakers—*Delphine Bard, Göran Sandberg*, Lund University, Lund, Sweden

This paper proposes a model of the nonlinearities in an electrodynamic loudspeaker based on Volterra series decomposition and taking into account the thermal effects affecting the electrical parameters when temperature increases. This model will be used to predict nonlinearities taking place in a loudspeaker and their evolution as the loudspeaker is used for a long time and/or at high power rates and its temperature increases. A temperature increase of the voice coil will cause its series resistance value to increase, therefore reducing the current flowing in the loudspeaker. This phenomenon is known as power compression.
Convention Paper 7186

2:00 pm

P5-2 Listening Tests of the Localization Performance of Stereodipole and Ambisonic Systems—*Andrea Capra*,[1,2] *Simone Fontana*,[1,3] *Fons Adriaensen*,[1] *Angelo Farina*,[1,2] Yves Grenier[3]
[1]LAE Group, Parma, Italy
[2]University of Parma, Parma, Italy
[3]Ecole Nationale Supérieure des Télécommunications, Paris, France

In order to find a possible correlation of objective parameters and subjective descriptors of the acoustics of theaters, auditoria or music halls, and perform meaningful listening tests, we need to find a reliable 3-D audio system that should give the correct perception of the distances, a good localization all around the listener, and a natural sense of realism. For this purpose a Stereo Dipole system and an Ambisonic system were installed in a listening room at La Casa Della Musica (Parma, Italy). Listening tests were carried out for evaluating the localization performances of the two systems.
Convention Paper 7187

2:00 pm

P5-3 Round Robin Comparison of HRTF Simulation Results: Preliminary Results—*Raphaël Greff*,[1] *Brian F. G. Katz*[2]
[1]A-Volute, Douai, France
[2]LIMSI—CNRS, Orsay, France

Variability in experimental measurement techniques of the HRTF is a concern that numerical calculation methods can hope to avoid. Numerical techniques such as the Boundary Element Method (BEM) allow for the calculation of the HRTF over the full audio spectrum from a geometrical model. While numerical calculations are not prone to the same errors as physical measurements, other problems appear that cause variations: geometry acquisition and modeling of real shapes as meshes can be performed in different ways. An on-going international round-robin study, “Club Fritz,” gathers HRTF data measured from different laboratories on a unique dummy head. This paper presents preliminary results of numerical simulation based on an acquired geometrical model of this artificial head.
Convention Paper 7188

2:00 pm

P5-4 Simulation of Complex and Large Rooms Using a Digital Waveguide Mesh—*Jose Lopez*,[1] *Jose Escolano*,[2] *Basilio Pueo*[3]
[1]Technical University of Valencia, Valencia, Spain
[2]University of Jaen, Jaen, Spain
[3]University of Alicante, Alicante, Spain

The Digital Waveguide Mesh (DWM) method for room acoustic simulation has been introduced in the last years to solve sound propagation problems numerically. However, the huge computer power needed in the modeling of large rooms and the complexity to incorporate realistic boundary conditions has delayed their general use, being restricted to the validation of theoretical concepts using simple and small rooms. This paper presents a complete DWM implementation that includes a serious treatment of boundary conditions, and it is able to cope with different materials in very large rooms up to reasonable frequencies. A simulation of a large building modeled with a high degree of precision has been carried out, and the obtained results are presented and analyzed in detail.
Convention Paper 7189

2:00 pm

P5-5 The Flexible Bass Absorber—*Niels W. Adelman-Larsen*,[1] *Eric Thompson,*[2] *Anders C. Gade*[2]
[1]Flex Acoustics, Lyngby, Denmark
[2]Technical University of Denmark, Lyngby, Denmark

Multipurpose concert halls face a dilemma. They host different performance types that require significantly different acoustic conditions in order to provide the best sound quality to the performers, sound engineers, and the audience. Pop and rock music contains high levels of bass sound but still require a high definition for good sound quality. The mid- and high-frequency absorption is easily regulated, but adjusting the low-frequency absorption has typically been too expensive or requires too much space to be practical for multipurpose halls. A practical solution to this dilemma has been developed. Measurements were made on a variable and mobile low-frequency absorber. The paper presents the results of prototype sound absorption measurements as well as elements of the design.
Convention Paper 7190

2:00 pm

P5-6 The Relation between Active Radiating Factor and Frequency Responses of Loudspeaker Line Arrays – Part 2—*Yong Shen, Kang An, Dayi Ou*, Nanjing University, Nanjing, China

Active Radiating Factor (ARF) is an important parameter for evaluating the similarity between a real loudspeaker line array and the ideal continuous line source. Our previous paper dealt with the relation between ARF of the loudspeaker line array and the Differential chart of its Frequency Responses in two distances (FRD). In this paper an improved way to estimate ARF of the loudspeaker line array by measuring on-axis frequency responses is introduced. Some further problems are discussed and experiment results are analyzed. The results may give some help to loudspeaker array designers.
Convention Paper 7191
[Paper not presened, but Convention Paper 7191 is available for purchase.]

2:00 pm

P5-7 Time Varying Behavior of the Loudspeaker Suspension—*Bo Rohde Pedersen*,[1] *Finn Agerkvist*[2]
[1]Aalborg University, Esbjerg, Denmark
[2]Technical University of Denmark, Lyngby, Denmark

The suspension part of the electrodynamic loudspeaker is often modeled as a simple linear spring with viscous damping. However, the dynamic behavior of the suspension is much more complicated than predicted by such a simple model. At higher levels the compliance becomes nonlinear and often changes during high excitation at high levels. This paper investigates how the compliance of the suspension depends on the excitation, i.e., level and frequency content. The measurements are compared with other known measurement methods of the suspension.
Convention Paper 7192

2:00 pm

P5-8 Diffusers with Extended Frequency Range—*Konstantinos Dadiotis, Jamie Angus, Trevor Cox*, University of Salford, Salford, Greater Manchester, UK

Schroeder diffusers are unable to diffuse sound when all their wells radiate in phase, a phenomenon known as flat plate effect. This phenomenon appears at multiple frequencies of pf0, where p is the integer that generates the well depths and f0 the design frequency. A solution is to send the flat plate frequencies above the bandwidth of interest. For QRDs and PRDs to achieve this goal, impractically long sequences are needed. This paper presents power residue diffusers, of small length in comparison to their prime generator, as solutions to the problem. Their characteristics are investigated and their performance when applied to Schroeder diffusers is explored while modulation is used to cope with periodicity. The results confirm the expectations.
Convention Paper 7193

2:00 pm

P5-9 Waveguide Mesh Reverberator with Internal Decay and Diffusion Structures—*Jonathan Abel, Patty Huang, Julius Smith III*, Stanford University, Stanford, CA, USA

Loss and diffusion elements are proposed for a digital waveguide mesh reverberator. The elements described are placed in the interior of the waveguide mesh and may be viewed as modeling objects within the acoustical space. Filters at internal scattering junctions provide frequency-dependent losses and control over decay rate. One proposed design method attenuates signals according to a desired reverberation time, taking into account the local density of loss junctions. Groups of one or several adjacent scattering junctions are altered to break up propagating wavefronts, thereby increasing diffusion. A configuration that includes these internal elements offers more flexibility in tailoring the reverberant impulse response than the common waveguide mesh construction where loss and diffusion elements are uniformly arranged solely at the boundaries. Finally, such interior decay and diffusion elements are ideally suited for use with closed waveguide structures having no boundaries, such as spherical or toroidal meshes, or meshes formed by connecting the edges or surfaces of two or more meshes.
Convention Paper 7194

Special Event
GEOFF EMERICK/SGT. PEPPER
Friday, October 5, 2:00 pm – 3:30 pm
Room 1E12/13

Marking the 40th Anniversary of the release of *Sgt. Pepper's Lonely Hearts Club Band*, Geoff Emerick, the Beatles engineer on the original recording was commissioned by the BBC to re-record the entire album on the original vintage equipment using contemporary musicians for a unique TV program.

Celebrating its own 60th Anniversary, the APRS is proud to present for a select AES audience, this unique project featuring recorded performances by young UK and US artists including the Kaiser Chiefs, The Fray, Travis, Razorlight, the Stereophonics, the Magic Numbers, and a few more—and one older Canadian, Bryan Adams.

These vibrant, fresh talents recorded the original arrangements and orchestrations of the Sgt. Pepper album using the original microphones, desks, and hard-learned techniques directed and mixed in mono by the Beatles own engineering maestro, Geoff Emerick.

Hear how it was done, how it should be done, and how many of the new artists want to do it in the future. Geoff will be available to answer a few questions about the recording of each track and, of course, more general questions regarding the recording processes and the innovative contribution he and other Abbey Road wizards made to the best ever album.

APRS, The Association of Professional Recording Services, promotes the highest standards of professionalism and quality within the audio industry. Its members are recording studios, postproduction houses, mastering, replication, pressing, and duplicating facilities, and providers of education and training, as well as record producers, audio engineers, manufacturers, suppliers, and consultants. Its primary aim is to develop and maintain excellence at all levels within the UK's audio industry.

➥

Workshop 4 **Friday, October 5**
2:30 pm – 5:00 pm **Room 1E08**

SURROUND 5.1 MIXING DO'S AND DON'TS

Chair: **George Massenburg**

Panelists: *Frank Filipetti*, Right Track Recording, New York, NY, USA
Crispin Murray
Ronald Prent, Galaxy Studios, Mol, Belgium
Jeff Wolpert

Even with the increasing confusion over the future of surround music mixes there is an increasing demand for efficiently produced surround products for film and television, as well as for music-only release. At the same time, it is economically tempting for film and television producers to accept surround mixing simulacrums ("faux 5.1" processing such as Unwrap or Logic 7) even when multichannel originals are available for remix.

Methods and techniques will be presented in this workshop to demonstrate how good, modern surround mixes are being made by successful professional practitioners. The workshop will cover subjects such as:

- Different approaches to "spreading out" multichannel sources
- Strategies for different media
- Use of the center channel
- Bass management
- Reverb, delay, and other effects
- Monitoring

Broadcast Session 3 **Friday, October 5**
2:30 pm – 4:30 pm **Room 1E10**

INNOVATIONS IN DIGITAL RADIO

Chair: **David Bialik**

Panelists: *Robert Bleidt*, Fraunhofer
David Day, Day Sequerra
Eric Hoehn, XM Radio
David Layer, NAB
Skip Pizzi, Microsoft/Radio World/NRSC
Daniel Schwab, NPR
Geir Skaaden, Neural Audio
Dave Wilson, CEA

Having matured from theory to reality, digital radio is an integral component of today's multifaceted communications industry. This broadcast event is a discussion of the development, implementation, and adaptation of digital radio technology in the United States. It will cover terrestrial, satellite,surround, multicasting, broadcaster adoption, transmission, reception, and consumer reaction. This is a chance to become familiar with the newest radio technology that is being successfully adopted in the U.S.

Technical Committee Meeting **Friday, October 5**
2:30 pm **Room 1E05**

Technical Committee Meeting on Acoustics and Sound Reinforcement.

Tutorial 3 **Friday, October 5**
3:00 pm – 5:00 pm **Room 1E11**

7.1 MOVIE MIXES FOR BLU-RAY AND HD DVD: WORKFLOW FROM MIX TO ENCODE

Presenters: **Ronny Katz**, DTS, Inc.
Jeff Levison, DTS Consultant/Levison Audio

With the advent of Blu-ray and HD DVD discs on the market, consumers can enjoy up to 8 channels of surround audio. For example, there can be more than the usual two surround channels, or additional loudspeakers for height information can be added. Theatrical movie presentation is just beginning to evolve to take advantage of these new loudspeaker locations, and few re-recording stages are fully implemented to mix in these new configurations. To meet the demand for 7.1 audio for the new optical disc formats, new approaches to original production and remixing techniques will be needed. Jeff Levison will walk you through a variety of strategies for both music and film. Musical approaches from in-studio productions to live recordings will be presented.

As movies are released in these new formats, remixing 5.1 releases to create new 7.1 versions is inevitable. A film mix from 5.1 stems will be used to evaluate decisions for making a new 7.1 mix and the final version compared to the original 5.1 mix. Using this 7.1 master, Ronny Katz will review the workflow of audio encoding for next-generation formats, including PCM audio preparation, surround channel layouts, interactive secondary/sub audio, a bit budgeting overview, and quality control. The tutorial will include a demonstration of audio quality control and discuss relevant topics such as batch encoding, bit-rate analysis for lossless audio, and bit-stream editing.

Student Event/Career Development
CAREER WORKSHOP
Friday, October 5, 3:00 pm – 5:00 pm
Room 1E06

Moderator: **Keith Hatschek**

Panelists: *Paul Antonell*, The Clubhouse Studio Inc.
Nancy Matter, Moonlight Mastering
Bob Power, Producer
Rich Tozzoli, Author and Engineer

This interactive workshop has been programmed based on student member input. Topics and questions were submitted by various student sections, polling students for the most in-demand topics. The final chosen topics are focused on education and career development within the audio industry and a panel selected to best address the chosen topics. An online discussion based on this talk will continue on the forums at aes-sda.org, the official student website of the AES.

Standards Committee Meeting **Friday, October 5**
3:00 pm **Room 1E02**

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

Workshop 5 **Friday, October 5**
3:30 pm – 6:30 pm **Room 1E09**

AUDIO QUALITY EVALUATION, DESCRIPTIVE ANALYSIS 2: INDIVIDUAL VOCABULARY DEVELOPMENT

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

Panelists: *Sylvain Choisel*, Bang and Olufsen a/s, Struer, Denmark
William Martens, McGill University, Montreal, Quebec, Canada

As knowledge about the listener experience of audio applications is fundamental in research and product design within the audio field, methods that can be used for evaluation of perceived sound quality are essential to explore. In order to capture and quantify listener experience, different methodological approaches are utilized. One of the approaches involves development of individual vocabulary of test subjects. This considers a collection of techniques that can be used for evaluating the detailed perceptual characteristics of products or systems through listening tests. This workshop aims to provide guidance to the researcher and experimenter regarding the nature of descriptive analysis by means of individual vocabulary development techniques and their application in audio.

Training Session **Friday, October 5**
3:30 pm – 4:30 pm **Room 1E04**

NEXT-GENERATION VIDEO GAME PRODUCTION

Presenters: **Steve Horowitz, Moderator**, Nickonline/The Code International/MPA
Chris Burke, Bong & Dern
Alistair Hirst, Omni Interactive Audio
Gene Semel, Sony

Next-Generation game consoles provide some hefty firepower when it comes to audio production. From orchestral recordings to surround sound mixing the sky seems to be the limit, or is it? This panel produced by Manhattan Producer's Alliance executive Steve Horowitz, will take a look at some of the significant advances in composing, mixing, scoring, and mastering for next generation consoles. Attendees will get a practical top-down view of how game production is developing and what the advancements mean for composers, sound designers, and producers working in the field.

Technical Committee Meeting **Friday, October 5**
3:30 pm **Room 1E05**

Technical Committee Meeting on High Resolution Audio.

Workshop 6 **Friday, October 5**
4:00 pm – 7:00 pm **Room 1E12/13**

HOW TO CHOOSE, PURCHASE, RESTORE, AND UPGRADE A MULTITRACK ANALOG TAPE MACHINE

Chair: **Steve Puntolillo**, Sonicraft A2DX Lab

Panelists: *John Chester*, Consultant, High Bridge, NJ, USA
Dominick Costanzo, Sony Music Studios, New York, NY, USA
John French, JRF Magnetics, Greendell, NJ, USA
John Klett, Tech Mecca, Carmel, NY, USA
Mike Spitz, ATR Services & ATR Magnetics, York, PA, USA

If you are considering adding an analog multitrack tape recorder to your facility, you might be wondering which one to buy, how much to pay, and what is required to get it running and keep it running. Or, if you are already using an analog multitrack, you might want to determine if it is performing at its peak and if you can improve its performance over stock configuration.

This workshop includes a presentation, panel discussion, and question-and-answer period designed to help you answer all of these questions and more. The moderator and panel of industry experts will share in-depth information based on their wealth of hands-on experience with analog tape machine selection, acquisition, restoration, maintenance, customization, and optimization over a wide variety of makes and models.

Live Sound Seminar 3 **Friday, October 5**
4:00 pm – 6:00 pm **Room 1E15**

AN INTERVIEW WITH ABE JACOB: A FOUNDING FATHER'S REMEMBRANCES AND PERSPECTIVE, WITH JOHN KILGORE

Moderator: **John Kilgore**

A Founding Father's Remembrances and Perspective

Moderator John Kilgore, a legendary Broadway sound man in his own right, will talk with the "Godfather of Theatre Sound Design." Abe Jacob began his career mixing live sound for Jimi Hendrix and the Mamas and the Papas. The rock 'n roll phase of his career is highlighted by designing the Monterey Pop Festival sound system. Moving on to Broadway, Jacob worked on such long-run hits as *Jesus Christ Superstar, A Chorus Line, Pippin, Chicago, Cats*, and *Evita*. A not-to-be-missed, behind-the-scenes look at the formative years of contemporary live sound design.

Broadcast Session 4 **Friday, October 5**
4:30 pm – 6:30 pm **Room 1E10**

FACILITY WIRING AND DISTRIBUTION SYSTEMS

Chair: **Eric Hoehn**, XM Radio

Panelists: *Dan Braverman*, Radio Systems - Studio Hub
Steve Gray, Cobra-Net
Howard Mullinack, Sierra Automated Systems and Engineering Corporation (SAS)
Marty Sacks, Axia Audio

There are many considerations when planning a facility. One of the most important is how you plan to wire and route the audio. From hard-wiring to multi-input-out routing systems, this session will discuss various techniques and technologies to accomplish the task.

Special Event
TECnology HALL OF FAME
Friday, October 6, 5:00 pm – 6:00 pm
Room 1E16

The Mix Foundation for Excellence in Audio will present it's Fourth Annual TECnology Hall of Fame induction ceremony at Javits, hosted by the AES. The Foundation, presenters of the TEC Awards, established the TECnology Hall of Fame in 2004 to honor and recognize audio products and innovations that have made a significant contribution to the advancement of audio technology. ➥

For this year's inductees, see www.mixfoundation.org/ hof/ techof.html.

The ceremony will be conducted by George Petersen, executive editor Mix and director the TECnology Hall of Fame. It is being sponsored by the AMP Alliance at Microsoft, Disc Makers and the College of Recording Arts and Sciences.

Student Event/Career Development
RECORDING COMPETITION—
STEREO CLASSICAL/POP
Friday, October 5, 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. Student members can submit stereo and surround recordings in the categories classical, jazz, folk/world music, and pop/rock. Meritorious awards will be presented at the closing Student Delegate Assembly Meeting on Monday.

Judges include: Martha De Francisco, David Frost, Richard King, Kimio Hamasaki, Ronald Prent, Andreas Mayo, Jim Anderson, Peter Cook, Tim Martyn, Rebecca Pulliam, Bob Katz, Michael Nunan, and Ronald Sadoff.

Sponsors for this event include: AEA, Yamaha, Genelec, Harmon/JBL, Sony Creative Software, Neumann, Shure, Lavry Engineering, Schoeps, Millennia Music and Media Systems, PCM, and Sennheiser.

Training Session **Friday, October 5**
5:00 pm – 6:00 pm **Room 1E04**

SEISMIC CHANGES: THE LANDSCAPE OF FILM AND TV MUSIC

Presenters: **Joe Carroll, Moderator**, Manhattan Producers Alliance
Chris Mangum, Film Composer
Robert Marshall, Source Connect
Chris Stone, Audio Impressions

Technological evolution coupled with economic factors are transforming how music for television and film is created and used. At the vanguard are organizations like the Manhattan Producers Alliance, building a collaborative network of composers from diverse areas of the business. Companies like Source Connect and eSession, make it possible for composers and producers to collaborate over unlimited distances. Makers of the newest sampling technology and virtual instruments like Audio Impressions allow us to more accurately simulate live performance. Music libraries like Pump Audio and Omni Music make vast amounts of independently created music easily available to film and TV producers and Apple computer's innovative music software is blurring the distinction between music listener and music maker. Joe Carroll, founder of the Manhattan Producers Alliance will moderate a lively discussion on navigating this new landscape with guest speakers.

Standards Committee Meeting **Friday, October 5**
5:00 pm **Room 1E02**

Standards Committee Meeting SC-04-01 Acoustics and Sound Source Modeling.

Tutorial 4 **Friday, October 5**
5:30 pm – 7:30 pm **Room 1E07**

SNARE STRATEGIES FOR STAGE AND STUDIO

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

Perhaps no other instrument in rock and roll offers the recording engineer so much opportunity—for success or failure. With a range of sonic capabilities from whispering brushes to cannon fire, the snare drum challenges engineers to know their craft. From our ears to our gear, the snare pushes the limits. Alex Case distills the challenge: from musical acoustics to psychoacoustics, from microphone techniques to signal processing strategies. Please join this discussion as we bring some order to any session involving drums.

Overcome the chaos of studio basics, reduce the stress of the live gig, and take maximum advantage of all the tools available to you at final mixdown.

Historical Program
A CELEBRATION OF HISTORY-MAKING NY STUDIOS
Friday, October 5, 5:30 pm – 7:30 pm
Room 1E08

Moderator: **Dan Daley**, Industry Journalist and Contributor to WIRED, Fast Company and History Channel Magazine

Panelists: *Jimmy Douglass*, engineer
Lou Gonzales, Quad Founder
Laurel Gonzalez-Kerlew, studio manager
Eddie Korvin, Blue Rock founder
Estelle Lazarus, studio manager
Jan Rathbun Horowitz, engineer
Joel Schenuneman, Manhattan Center Chief Engineer
John Storyk, studio architect

Quad Studios and Manhattan Center are both celebrating 30th anniversaries this year. Blue Rock, was a highly regarded studio for over 16 years, racking up hits for The Kinks, Joe Jackson, Bob Dylan, and many other timeless artists. Quad credits range from Madonna and Coldplay to Aerosmith, Tupac, and LL Cool J. Originally built by Oscar Hammerstein in 1906, Manhattan Center continues to serve as one of New York's largest and most active studios. Studio owners, artists, engineers, producers, and surprise guests will celebrate the glory days and the continuing legacy of these classic NY Studios.

Technical Committee Meeting **Friday, October 5**
5:30 pm **Room 1E05**

Technical Committee Meeting on Loudspeakers and Headphones.

Workshop 7 **Friday, October 5**
6:00 pm – 7:30 pm **Room 1E15**

VIDEO TO AUDIO TIMING MEASUREMENT AND CORRECTION IN TODAY'S SYSTEMS

Chair: **Randy Conrod**, Harris Corporation, Toronto, Ontario, Canada

Panelists: *Ken Hunold,* Dolby Labs
Tom McDonough, AZCAR
Tom Sahara,Turner Sports

Sorting out timing issues between video and audio in today's systems is difficult to manage. This workshop addresses HD and SD video and the conversions necessary in today's hybrid environment and the plethora of audio that is supposed to match up with the video: analog, AES, embedded in HD or SD, coded (compressed), and up/down mix technologies. The workshop will consist of a power point presentation and demonstration of Harris's "V2A" test signal and measurement and correction method.

Special Event
MIXER PARTY
Friday, October 5, 6:00 pm – 8:00 pm
Exhibition Hall

A mixer party will be held on Friday evening to enable convention attendees to meet in a social atmosphere after the opening day's activities to catch up with friends and colleagues from the industry. There will be a cash bar and snacks.

Special Event
ORGAN RECITAL BY GRAHAM BLYTH
Friday, October 5, 8:00 pm – 9:30 pm
Brick Presbyterian Church
62 E. 92nd St., New York

Graham Blythe's traditional organ concert will be given at the Brick Presbyterian Church, which has been serving New York City since 1767.

Chapel Organ: In November 1994 Guilbault-Thérien, Inc. of St-Hyacinthe, Québec, was commissioned to build a new organ for the chapel. By the end of May 1996 the organ had been completely assembled, voiced, and tuned. This two manual and pedal instrument has nineteen stops and twenty-six ranks. The organ's overall design is strongly influenced by the French choir organs (orgues de choeur) and smaller grand orgues of the second part of the 19th century, especially those instruments built by the famous Parisian firm of Aristide Cavaillé-Coll. Although this instrument is not an exact replica of such instruments, one can easily define this new organ as built à la manière of Cavaillé-Coll with adaptations incorporating late 20th century organ-building practices. The organ case is of solid American black walnut and is stained to match the chapel's existing walnut woodwork. The case design closely resembles the orgue de salon built by Cavaillé-Coll for the composer Charles-François Gounod. Hand carved ornaments and moldings are patterned after the chapel's architectural details. The console is patterned after those by Cavaillé-Coll. The suspended action keyboards have mammoth tusk ivory naturals and ebony sharps; the pedalboard is of maple with ebony caps for the sharps. All pipe scalings, materials, construction, and on-site voicing reflect the Cavaillé-Coll tradition with due consideration given to the chapel's size and acoustic. The organ is equipped with a state-of-the-art, touch-screen controlled combination and sequencer system designed for this organ by Syncordia International, Inc. of St.-Lambert, Québec.

Sanctuary Organ – Casavant Frères op. 3837: From May through August 2005, the new Casavant organ was installed and voiced in the Sanctuary of Brick Church. This instrument of 4 manuals and 118 ranks (6,288 pipes) is one of New York City's landmark organs. It has few peers in North America as it accurately reproduces the sounds of the great 19th century French organs, especially those built by Aristide Cavaillé-Coll. This organ is a gift of an anonymous donor in honor of former Senior Pastor, Dr. Herbert B. Anderson and his wife, Mary Lou S. Anderson.

The program features an all British first half (Parry, Stanford, Matthias) with the complete 5th Symphony by Widor in the second half.

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. 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.

Graham Blyth's Organ Recital is sponsored by the Soundcraft Studer Group.

Session P6 **Saturday, October 6**
9:00 am – 12:00 noon **Room 1E07**

PERCEPTION, PART 3

Chair: **Brent Edwards**, Starkey Hearing Research Center, Berkeley, CA, USA

9:00 am

P6-1 Deriving Physical Predictors for Auditory Attribute Ratings Made in Response to Multichannel Music Reproductions—
Sungyoung Kim, William Martens, McGill University, Montreal, Quebec, Canada

A group of eight students engaged in a Tonmeister training program were presented with multichannel loudspeaker reproductions of a set of solo piano performances and were asked to complete two attribute rating sessions that were well separated in time. Five of the eight listeners produced highly consistent ratings after a six-month period during which they received further Tonmeister training. Physical predictors for the obtained attribute ratings were developed from the analysis of binaural recordings of the piano reproductions in order to support comparison between these stimuli and other stimuli, and thereby to establish a basis for independent variation in the attributes to serve both creative artistic goals and further scientific exploration of such multichannel music reproductions.
Convention Paper 7195

9:30 am

P6-2 Interaction between Loudspeakers and Room Acoustics Influences Loudspeaker

Preferences in Multichannel Audio Reproduction—*Sean Olive*,[1] William Martens[2]
[1]Harman International Industries, Inc., Northridge, CA, USA
[2]McGill University, Montreal, Quebec, Canada

The physical interaction between loudspeakers and the acoustics of the room in which they are positioned has been well established; however, the influence on listener preferences for loudspeakers that results from such variation in room acoustics has received little experimental verification. If listeners adapt to listening room acoustics relatively quickly, then room acoustic variation should not significantly influence loudspeaker preferences. In the current paper two groups of listeners were given differential exposure to listening room acoustics via a binaural room scanning (BRS) measurement and playback system. Although no significant difference in loudspeaker preference was found between these two groups of listeners, the room acoustic variation to which they were exposed did significantly influence loudspeaker preferences.
Convention Paper 7196

10:00 am

P6-3 Evaluating Off-Center Sound Degradation in Surround Loudspeaker Setups for Various Multichannel Microphone Techniques—*Nils Peters,*[1] *Stephen McAdams*,[1] *Jonas Braasch*[2]
[1]McGill University, Montreal, Quebec, Canada
[2]Rensselaer Polytechnic Institute, Troy, NY, USA

Many listening tests have been undertaken to estimate listeners' preferences for different multichannel recording techniques. Usually these tests focus on the sweet spot, the spatial area where the listener maintains optimal perception of virtual sound sources, thereby neglecting to consider off-center listening positions. The purpose of the present paper is to determine how different microphone configurations affect the size of the sweet spot. A perceptual method is chosen in which listening impressions achieved by three different multichannel recording techniques for several off-center positions are compared with the listening impression at the sweet spot. Results of this listening experiment are presented and interpreted.
Convention Paper 7197

10:30 am

P6-4 The Effects of Latency on Live Sound Monitoring—*Michael Lester,*[1] *Jon Boley*[2]
[1]Purdue University, West Lafayette, IN, USA
[2]Shure Incorporated, Niles, IL, USA

A subjective listening test was conducted to determine how objectionable various amounts of latency are for performers in live monitoring scenarios. Several popular instruments were used and the results of tests with wedge monitors are compared to those with in-ear monitors. It is shown that the audibility of latency is dependent on both the type of instrument and monitoring environment. This experiment shows that the acceptable amount of latency can range from 42 ms to possibly less than 1.4 ms under certain conditions. The differences in latency perception for each instrument are discussed. It is also shown that more latency is generally acceptable for wedge monitoring setups than for in-ear monitors.
Convention Paper 7198

11:00 am

P6-5 A Perforated Desk Surface to Diminish Coloration in Desktop Audio-Production Environments—*Karl Gentner*,[1] *Jonas Braasch,*[2] *Paul Calamia*[2]
[1]BRC Acoustics & Technology, Seattle, WA, USA
[2]Rensselaer Polytechnic Institute, Troy, NY, USA

In audio-production rooms, a common source of harmful reflections is the mixing console or desk surface itself. A perforated material is proposed as an alternative desk surface to reduce coloration by achieving acoustical transparency. A variety of desk surfaces and perforation schemes were tested within common room conditions. The resulting psychoacoustic study indicates that the fully-perforated desk provides lower coloration than that of the solid desk in every condition. A partially-perforated desk shows a similar decrease in coloration, specifically when the perforated area is determined by the Fresnel zones dictated by the source and receiver positions.
Convention Paper 7199

11:30 am

P6-6 Perceptually Modeled Effects of Interchannel Crosstalk in Multichannel Microphone Technique—*Hyun-Kook Lee*,[1] *Russell Mason,*[2] *Francis Rumsey*[2]
[1]LG electronics, Seoul, Korea
[2]University of Surrey, Guildford, Surrey, UK

One of the most noticeable perceptual effects of interchannel crosstalk in multichannel microphone techniques is an increase in perceived source width. The relationship between the perceived source-width-increasing effect and its physical causes was analyzed using an IACC-based objective measurement model. A description of the measurement model is presented, and the measured data obtained from stimuli created with crosstalk and those without crosstalk are analyzed visually. In particular, frequency and envelope dependencies of the measured results and their relationship with the perceptual effect are discussed. The relationship between the delay time of the crosstalk signal and the effect of different frequency content on the perceived source width is also discussed in this paper.
Convention Paper 7200

Session P7 — Saturday, October 6
9:00 am – 11:30 am — Room 1E16

SIGNAL PROCESSING, PART 3

Chair: **Dana Massie**, Audience, Mountain, View, CA, USA

9:00 am

P7-1 Sigma-Delta Modulators Without Feedback

Around the Quantizer?—*Stanley Lipshitz, John Vanderkooy, Bernhard Bodmann*, University of Waterloo, Waterloo, Ontario, Canada

We use a result due to Craven and Gerzon—the "Integer Noise Shaping Theorem"—to show that the internal system dynamics of the class of sigma-delta modulators (or equivalently noise shapers) with integer-coefficient FIR error-feedback filters can be completely understood from the action of simple, linear pre- and de-emphasis filters surrounding a (possibly nonsubtractively dithered) quantizer. In this mathematically equivalent model, there is no longer any feedback around the quantizer. The major stumbling block, which has previously prevented a complete dynamical analysis of all such systems of order higher than one, is thus removed. The class of integer noise shapers includes, but is not restricted to, the important family of "Pascal" shapers, having all their zeros at dc.
Convention Paper 7201

9:30 am

P7-2 The Effect of Different Metrics on the Performance of "Stack" Algorithms for Look-Ahead Sigma Delta Modulators— *Peter Websdell, Jamie Angus***,** University of Salford, Salford, Greater Manchester, UK

Look-ahead Sigma-Delta modulators look forward k samples before deciding to output a "one" or a "zero." The Viterbi algorithm is then used to search the trellis of the exponential number of possibilities that such a procedure generates. This paper describes alternative tree based algorithms. Tree based algorithms are simpler to implement because they do not require back-tracking to determine the correct output value. They can also be made more efficient using "Stack" algorithms. Both the tree algorithm and the more computationally efficient "Stack" algorithms are described. In particular, the effects of different error metrics on the performance of the "Stack" algorithm are described and the average number of moves required per bit discussed. The performance of the "Stack" algorithm is shown to be better than previously thought.
Convention Paper 7202

10:00 am

P7-3 Evaluation of Time-Frequency Analysis Methods and Their Practical Applications— *Pascal Brunet, Zachary Rimkunas, Steve Temme*, Listen, Inc., Boston, MA, USA

Time-Frequency analysis has been in use for more than 20 years and many different time-frequency distributions have been developed. Four in particular, Short Time Fourier Transform (STFT), Cumulative Spectral Decay (CSD), Wavelet, and Wigner-Ville have gained popularity and firmly established themselves as useful measurement tools. This paper compares these four popular transforms, explains their trade-offs, and discusses how to apply them to analyzing audio devices. Practical examples of loudspeaker impulse responses, loose particles, and rub & buzz defects are given as well as demonstration of their application to common problems with digital/analog audio devices such as Bluetooth headsets, MP3 players, and VoIP telephones.
Convention Paper 7203

10:30 am

P7-4 Time-Frequency Characterization of Loudspeaker Responses Using Wavelet Analysis—*Daniele Ponteggia*,[1] *Mario Di Cola*[2]
[1]Audiomatica, Florence, Italy
[2]Audio Labs Systems, Milan, Italy

An electroacoustic transducer can be characterized by measuring its impulse response (IR). Usually the collected IR is then transformed by means of the Fourier Transform to get the complex frequency response. IR and complex frequency response form a pair of equivalent views of the same phenomena. An alternative joint time-frequency view of the system response can be achieved using wavelet transform and a color-map display. This work illustrates the implementation of the wavelet transform into a commercial measurement software and presents some practical results on different kinds of electroacoustic systems.
Convention Paper 7204

11:00 am

P7-5 Equalization of Loudspeaker Resonances Using Second-Order Filters Based on Spatially Distributed Impulse Response Measurements—*Jakob Dyreby, Sylvain Choisel*, Bang & Olufsen A/S, Struer, Denmark

A new approach for identifying and equalizing resonances in loudspeakers is presented. The method optimizes the placement of poles and zeros in a second-order filter by minimization of the frequency-dependent decay. Each resonance may be equalized by the obtained second-order filter. Furthermore, the use of spectral decay gives opportunity for optimizing on multiple measurements simultaneously making it possible to take multiple spatial directions into account. The proposed procedure is compared to direct inversion and minimum-phase equalization. It makes it possible to equalize precisely the artifacts responsible for ringing, while being largely unaffected by other phenomena such as diffractions, reflections, and noise.
Convention Paper 7205

Workshop 8 Saturday, October 6
9:00 am – 12:00 noon Room 1E11

SURROUND WORKSHOP PLAYBACK SESSION

Chair: **Martha DeFrancisco**, McGill University, Montreal, Quebec, Canada

Panelists: *Chuck Ainlay*, Independent Producer/Engineer
Akira Fukada, NHK
Lawrence Manchester, Bricks & Mortar Music
Everett Porter, Polyhymnia

Experience immersive Surround Sound as Martha DeFrancisco, Akira Fukada, Everett Porter, Chuck Ainlay and Lawrence Manchester—panelists of Workshop 1, Recording Large Ensembles in Multichannel—demonstrate their recordings.

Broadcast Session 5 **Saturday, October 6**
9:00 am – 10:30 am **Room 1E08**

THE ART OF SOUND EFFECTS

Presenters: **Tom Curley**
Butch D'Ambrosio
Gary Kiffell
David Shinn
Marc Wiener
Sue Zizza

Sound effects: footsteps, doors opening and closing, a bump in the night. These are the sounds that can take the flat one-dimensional world of audio, television, and film and turn them into realistic three-dimensional environments. From the early days of radio to the sophisticated modern day High Def Surround Sound of contemporary film; sound effects have been the final color on the director's palatte. Join Gary Kiffel of the David Letterman Show and SFX and Foley Artists Sue Zizza and David Shinn as they explore the creating, recording, and use of various primary sound effect elements. Both manual and recorded effects will be discussed.

Tutorial 5 **Saturday, October 6**
9:00 am – 11:00 am **Room 1E15**

ARCHITECTURAL ACOUSTICS, PART 1

Presenter: **Tony Hoover**, Mckay Conant Hoover, Inc., Westlake Village, CA, USA

This tutorial emphasizes architectural acoustics consulting, which is the art of applying the science of sound in a timely, practical, and easily-understood manner. Three distinct areas are covered: sound isolation (airborne and structure-borne), HVAC noise control, and surface treatments (absorption, reflection, and diffusion). The format has been used for decades in meetings and seminars, and for over 40 semesters at Berklee College of Music.

The objective is to provide an overview of fundamentals and terminology, bases for design decisions and evaluations, and confidence to navigate the oceans of information, misinformation, and propaganda about "acoustical" materials, products, and practices.

Live Sound Seminar 4 **Saturday, October 6**
9:00 am – 11:00 am **Room 1E09**

SOUND SYSTEM ALIGNMENT AND ACOUSTIC MEASUREMENT—UNDERSTANDING AND USING THE TOOLS AVAILABLE TO ACHIEVE BETTER SOUND

Chair: **Sam Berkow**, SIA Acoustics, New York, NY, USA

Panelists: *Mark Dennis*, MGM.Mirage/Cirque du Soleil
"Decibel" Dave Dennison, DBDave Audio/Meyer Sound Labs
Ted Leamy, JBL Professional
Bob McCarthy, Alignment and Design, Inc.
Robert Scovill, Digidesign
Steve Sockey, SIA Acoustics

There are numerous tools available to help "optimize" sound system performance. This panel will review the tools leading sound designers and operators use to help them achieve high quality results. The focus will be on practical applications and users' experiences in a real room. Discussion points will include how measurements can help us understand what we hear and how time alignment works on different scales, from the very small to the very large.

Student Event/Career Development
"STEPS TO THE FUTURE"—ONE ON ONE MENTORING SESSION, PART 1
Saturday, October 6, 9:00 am – 12:00 noon
Room 1E06

Students are invited to sign-up for an individual meeting with a distinguished mentor from the audio industry. The opportunity to sign up will be given at the end of the opening SDA meeting. Any remaining open spots will be posted in the student area. All students are encouraged to participate in this exciting and rewarding opportunity for individual discussion with industry mentors.

Technical Committee Meeting **Saturday, October 6**
9:00 am **Room 1E05**

Technical Committee Meeting on Microphones and Applications.

Standards Committee Meeting **Saturday, October 6**
9:00 am **Room 1E02**

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

Session P8 **Saturday, October 6**
9:30 am – 11:00 am **Foyer 1E**

POSTERS: APPLICATIONS IN AUDIO

9:30 am

P8-1 Pump Up the Volume: Enhancing Music Phone Audio Quality and Power Using Supercapacitors for Power Management— *Pierre Mars*, CAP-XX (Australia) Pty. Ltd., Sydney, NSW, Australia

As multimedia and music phones grow in popularity, consumers want an iPod-quality, uninterrupted audio experience without the buzzing and clicks associated with wireless transmission. This paper describes the problems delivering high power and high quality audio in music-enabled mobile phones and how a supercapacitor can overcome them. Typically, the audio amplifier power supply input in a mobile phone is connected directly to Vbattery. This paper compares audio performance between the typical setup and connecting the audio amp supply to a supercapacitor charged to 5V through a current limited boost converter.
Convention Paper 7206

9:30 am

P8-2 Digital Audio Processing on a Tiny Scale: Hardware and Software for Personal Devices—*Peter Eastty*, Oxford Digital Limited, Oxfordshire, UK

The design of an audio signal processor, graphical programming environment, DSP software, and parameter adjustment tool is described with reference to the hardware and software requirements of the audio sweetening function in personal devices, particularly cell phones. Special care is taken in the hardware design to ensure low operating power, small size (4mm*4mm package), or 0.5 to 1 sq. mm area depending on geometry, stereo analog, and digital I/O and high performance. The parameter adjustment tool allows real time control of the DSP so that processing may be customized to the actual properties of the audio sources and the acoustic properties of the enclosure and loudspeakers. A live demonstration of the programming and parameter adjustment of the processor will be given as part of the presentation of the paper.
Convention Paper 7207

9:30 am

P8-3 Enhancing End-User Capabilities in High Speed Audio Networks—*Nyasha Chigwamba, Richard Foss*, Rhodes University, Grahamstown, South Africa

Firewire is a digital network technology that can be used to interconnect professional audio equipment, PCs, and electronic devices. The Plural Node Architecture splits connection management of firewire audio devices between two nodes namely, an enabler and a transporter. The Audio Engineering Society's SC-02-12-G Task Group has produced an Open Generic Transporter guideline document that describes a generic interface between the enabler and transporter. A client-server implementation above the Plural Node Architecture allows connection management of firewire audio devices via TCP/IP. This paper describes enhancements made to connection management applications as a result of additional capabilities revealed by the Open Generic Transporter document.
Convention Paper 7208

9:30 am

P8-4 Sharing Acoustic Spaces over Telepresence Using Virtual Microphone Control—*Jonas Braasch,*[1] *Daniel L. Valente*,[1] *Nils Peters*[2]
[1]Rensselaer Polytechnic Institute, Troy, NY, USA
[2]McGill University, Montreal, Quebec, Canada

This paper describes a system that is used to project musicians in two or more co-located venues into a shared virtual acoustic space. The sound of the musicians is captured using spot microphones. Afterward, it is projected at the remote end using spatialization software based on virtual microphone control (ViMiC) and an array of loudspeakers. In order to simulate the same virtual room at all co-located sites, the ViMiC systems communicate using the Open-Sound Control protocol to exchange room parameters and the room coordinates of the musicians.
Convention Paper 7209

9:30 am

P8-5 A Tutorial: Fiber Optic Cables and Connectors for Pro-Audio—*Ronald Ajemian*, Owl Fiber Optics, Flushing, NY, USA

There have been many technological breakthroughs in the area of fiber optic technology that have allowed an easier transition to migrate into the professional audio arena. Since the current rise of copper prices in the worldwide markets, there has been an increase of usage in fiber optic based equipment, cables, and connectors deployed for pro-audio and video. This prompted the writing of this tutorial to bring the professional audio community up to date with some old and new fiber optic cables and connectors now being deployed in pro-audio. This tutorial will help audio professionals understand the jargon and to better understand fiber optic technology now being deployed in pro-audio.
Convention Paper 7210

9:30 am

P8-6 The Most Appropriate Method of Producing TV Program Audio Focusing on the Audience—*Hisayuki Ohmata*,[1] *Akira Fukada*,[2] *Hiroshi Kouchi*[3]
[1]NHK Science & Technical Research Laboratories, Tokyo, Japan
[2]NHK Broadcasting Center, Tokyo, Japan
[3]NHK Kofu Station, Kofu, Yamanashi, Japan

When audiences watch TV programs, they often perceive a difference in audio levels. This is a real annoyance for them, and it is caused by differences in program audio. In order to have equal audio levels, it is necessary to produce audio under the same conditions for all programs. To solve this problem, we propose a method to produce TV program audio. We make clear the manner in which different monitoring levels influence mixing balance at various mixing stages. This paper also describes management of audio levels for programs with different digital broadcasting head rooms.
Convention Paper 7211

9:30 am

P8-7 Beyond Splicing: Technical Ear Training Methods Derived from Digital Audio Editing Techniques—*Jason Corey*, University of Michigan, Ann Arbor, MI, USA

The process of digital audio editing, especially with classical or acoustic music using a source-destination method, offers an excellent opportunity for ear training. Music editing involves making transparent connections or splices between takes of a piece of music and often requires specifying precise edit locations by ear. The paper outlines how aspects of digital editing can be used systematically as an ear training method, even out of the context of an editing session. It describes a software tool that uses specific techniques from audio editing to cre- ➥

ate an effective ear training method that offers benefits that transfer beyond audio editing.
Convention Paper 7212

9:30 am

P8-8 New Trends in Sound Reinforcement Systems Based on Digital Technology—
Piotr Kozlowski,[1] *Pawel Dziechcinski*,[1] *Wojciech Grzadziel*[2]
[1]Wroclaw University of Technology, Wroclaw, Poland;
[2]Pracownia Akustyczna, Acoustic Design Team, Wroclaw, Poland

This paper presents new aspects of modern sound reinforcement system's designing that came into view because of the prevalence of digital technology. The basic structure of modern digital electro acoustical systems is explained using as an example the one installed at Wroclaw Opera House. This paper focuses on some aspects connected to digital transmission of audio signals, proper audience area sound coverage, achieving smooth frequency response, getting directive propagation at low frequencies, and controlling the system. Some measurement and tests about the topics presented in the paper have been done during the tuning of the system at Wroclaw Opera House. Achieved results prove that it is possible to acquire these targets.
Convention Paper 7213

9:30 am

P8-9 Using Audio Classifiers as a Mechanism for Content-Based Song Similarity—*Benjamin Fields, Michael Casey*, Goldsmiths College, University of London, London, UK

As collections of digital music become larger and more widespread, there is a growing need for assistance in a user's navigation and interaction with a collection and with the individual members of that collection. Examining pairwise song relationships and similarities, based upon content derived features, provides a useful tool to do so. This paper looks into a means of extending a song classification algorithm to provide song to song similarity information. In order to evaluate the effectiveness of this method, the similarity data is used to group the songs into k-means clusters. These clusters are then compared against the original genre sorting algorithm.
Convention Paper 7267

Workshop 9 **Saturday, October 6**
9:30 am – 11:00 am **Room 1E10**

MASTERING FOR NEW MEDIA

Cochairs: **Gavin Lurssen**, Lurssen Mastering, Hollywood, CA, USA
Joe Palmaccio, The Place . . . For Mastering, Nashville, TN, USA

Panelists: *Chris Athens*, Mastering Engineer, Sterling Sound, New York, NY, USA
Derek Jones, Music Producer, Megatrax, Los Angeles, CA, USA
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA

One common denominator in all commercial music is that it is mastered. This has created a visceral expectation to the untrained ear.

As the industry changes, mastering engineers have become more in demand to fulfill this expectation specifically for new media outlets. Music libraries, production houses, and artists are competing with major record labels vying for alternate revenue streams capitalizing on music for new media. Labels and artists are mastering TV tracks and libraries are now sending their catalogs to mastering facilities.

This development along with new technology has created some changes in the way mastering studios can service clients, one example being the delivery of raw files vs. finished and edited CD's.

This panel will ask, "What remains the same and what has changed?"

Training Session **Saturday, October 6**
10:00 am – 11:00 am **Room 1E04**

SONY AUDIO-FOR-GAMES REPORT

Presenters: *Dan Bardino*, SCEE
Dave Murrant, SCEA, Sony Computer Entertainment America
Dave Ranyard, SCEE, Sony Computer Entertainment Europe
Gene Semel, SCEA

The big picture perspective of interactive entertainment/video game audio as told from the perspective of Sony Computer Entertainment's (America and Europe) leading game sound designers. The session will cover topics such as:

- The manpower resources required to create and implement audio in an interactive product;
- The tools required to perform this work from a traditional and proprietary standpoint;
- How video game audio requires a "new breed" of audio professional. Today's interactive audio professional must have a broad knowledge of not only traditional signal flow, engineering and recording, and content creation but requires continous education on the rapidly changing technologies that directly effect their day to day workloads;
- The Studios! The facilities required to raise the audio quality bar for consumers and next generation interactive products—pictures and discussion of SCEA's new world class studio build out (nominated for Studio Design at this year's TEC Awards);
- Ancillary topics: Field recording requirements have raised; Foley recording is almost now standard, etc.
- Demonstrations: SingStar product overview—Highly successful Karaoke video game; SCEE developer Tools specific to the SingStar will give the audience a view of technology that has a cross-discipline application (video and music).

Technical Committee Meeting **Saturday, October 6**
10:00 am **Room 1E05**

Technical Committee Meeting on Audio for Games.

Workshop 10 **Saturday, October 6**
11:00 am – 1:00 pm **Room 1E15**

FROM SAC TO SAOC—RECENT DEVELOPMENTS IN PARAMETRIC CODING OF SPATIAL AUDIO

Chair: **Jürgen Herre**, Fraunhofer IIS, Erlangen, Germany

Panelists: *Jeroen Breebaart*, Philips Research, Eindhoven, The Netherlands
Christof Faller, EPFL, Lausanne, Switzerland
Barbara Resch, Coding Technologies, Stockholm, Sweden

One of the most remarkable innovations in low bit rate audio coding during the recent years was the arrival of Spatial Audio Coding (SAC) technology. Exploiting the human perception of spatial sound, these coding schemes are capable of transmitting high quality surround sound using bit rates that have been used for carrying traditional two-channel stereo audio so far. Following the recent finalization of the MPEG Surround specification, a next technology generation is envisaged for standardization within ISO/MPEG allowing bit-rate-efficient and backward compatible coding of several sound objects. On the receiving side, such a Spatial Audio Object Coding (SAOC) system renders the objects interactively into a sound scene on a reproduction setup of choice. The workshop reviews the principles and current status of spatial audio coding schemes and discusses their evolution toward spatial audio object coding with particular focus on the ongoing ISO/MPEG audio standardization activities in this field. The results of the first round of technology selection within MPEG will be discussed.

Broadcast Session 6 **Saturday, October 6**
11:00 am – 1:00 pm **Room 1E10**

LOUDNESS WORKSHOP

Chair: **Emil Torrick**, CBS, retired

Panelists: *Frank Foti*, Omnia
Thomas Lund, TC Electronics
Jeffrey Riedmiller, Dolby Laboratories
Gilbert Soulodre, Communications Research Centre

New challenges and opportunities await broadcast engineers concerned about optimum sound quality in this contemporary age of multichannel sound and digital broadcasting. The earliest studies in the measurement of loudness levels were directed to telephony issues, with the publication in 1933 of the equal-loudness contours of Fletcher and Munson, and the Bell Labs tests of more than a half-million listeners at the 1938 New York Worlds Fair demonstrating that age and gender are also important factors in hearing response. A quarter of a century later, broadcasters began to take notice of the often-conflicting requirements of controlling both modulation and loudness levels. These are still concerns today as new technologies are being adopted. This session will explore the current state of the art in the measurement and control of loudness levels and look ahead to the next generation of techniques that may be available to audio broadcasters.

Live Sound Seminar 5 **Saturday, October 6**
11:00 am – 1:00 pm **Room 1E09**

SOUND REINFORCEMENT FOR JAZZ AND OTHER ACOUSTIC MUSIC—IT ISN'T ROCK AND ROLL!

Chair: **Jim Brown**, Audio Systems Group

Panelists: *Robert Auld*
Rick Chinn
Jim van Bergen, AudioArt Sound

Jazz and other types of acoustical instrument-only performances often suffer from poor sound reinforcement. Amplification is often wildly excessive, failing to capture the dynamics of the music. Music timbre is distorted by poor microphone techniques and poor mixing. Ad hoc sound systems are often simpler than they should be, so musical balance varies widely over the audience. This live sound event begins with a discussion of what makes jazz and other acoustic music different from other forms of reinforced music. Attention is given to mixing and microphone techniques, with both photographs and recorded examples of good practices and those that should be avoided. The concept of “zero-based mixing” is discussed.

Special Event
PLATINUM PRODUCERS
Saturday, October 6, 10:00 am – 11:30 am
Room 1E12/13

Moderator: **Paul Verna**

Panelists: *Ed Cherney*
Jack Joseph Puig
Hal Willner

The Platinum Producers Panel will feature such impeccably credentialed record makers as: Ed Cherney (Clapton, Bonnie Raitt, Jackson Browne, The Rolling Stones, Bob Dylan); Hal Willner (Lou Reed, Deborah Harry, Lucinda Williams); Al Kooper (Lynyrd Skynyrd, Joe Ely, B.B. King, The Tubes); Jack Joseph Puig (Green Day, John Mayer, Black Crows, Verve Pipe) and a coterie of equally accomplished producers. They will explore the creative, technical, business, and career-management challenges producers face in today's rapidly evolving recording industry. The panel will be moderated by pro audio author and journalist Paul Verna, a veteran of *Billboard* and *Mix* magazines and co-author of *The Encyclopedia of Record Producers*.

Technical Committee Meeting **Saturday, October 6**
11:00 am **Room 1E05**

Technical Committee Meeting on Audio for Telecommunications.

Standards Committee Meeting **Saturday, October 6**
11:00 am **Room 1E02**

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

Special Event
LUNCHTIME KEYNOTE:
CHARLES J. LIMB AND PATRICK J. DONNELLY
Saturday, October 6, 11:30 am – 12:30 pm
Room 1E08

Introduction by Poppy Crum

Beethovens Deafness

The compositions of Ludwig van Beethoven are widely regarded as seminal masterpieces that changed not only the course of musical history, but also human history more broadly. The tremendous impact of his musical output is even more striking in the context of the composer's well-known hearing loss. Despite recent advances in medical understanding of hearing loss, however, the cause of Beethoven's hearing loss remains a mystery. This session will review Beethoven's musical compositions from the perspective of a musician with progressive hearing loss, and in the context of medical diagnoses based on modern otologic principles. Recent data, including studies of Beethoven's hair, will also be discussed as part of this investigative study of one of the great mysteries of classical music.

Dr. Charles J. Limb is a neurotologist and skull base surgeon in the Department of Otolaryngology-Head and Neck Surgery at Johns Hopkins Hospital in Baltimore, Maryland. He is also a faculty member at the Peabody Conservatory of Music. He received his undergraduate degree from Harvard University, medical degree from Yale University, and surgical training from Johns Hopkins Hospital. He treats a wide range of otologic conditions and specializes in hearing restoration using cochlear implants, as well as in the treatment of musicians with hearing disorders. His research focuses on the neural mechanisms responsible for music perception.

Patrick J. Donnelly has been a lifelong scholar of the life and music of Beethoven. He received his undergraduate degrees in Musicology and Computer Science from Washington University in St. Louis and is completing graduate studies in Musicology and Computer Music Research at the Peabody Conservatory of Music in Baltimore. He is currently a researcher at the Sound and Music Perception Laboratory at Johns Hopkins Hospital, where his work focuses on algorithmic musical analysis and cochlear implant-mediated perception of music.

Student Event/Career Development
CAREER/JOB FAIR
Saturday, October 6, 11:30 am – 1:30 pm
Room 1E Foyer

The Career/Job Fair will feature several companies from the exhibit floor. All attendees of the convention, students and professionals alike, are welcome to come talk with representatives from the companies and find out more about job and internship opportunities in the audio industry. Bring your resume!

Confirmed participating companies include Fraunhofer IIS, Rane Corporation, AEA, Røde Microphones, THAT Corporation, Eventide, Inc., and Dolby Laboratories.

Exhibitor Seminar **Saturday, October 6**
11:30 am – 12:30 pm **Room 1E04**

THAT CORPORATION

Presenters: **Gary Hebert, Bob Moses, Les Tyler**

Analog Design Secrets Your Mother Never Told You

As the performance of modern digital audio technology surpasses the range of human hearing, the weak links in the signal chain are often found in the analog domain. For example, the 120-dB dynamic range achievable by a modern A/D converter is sacrificed when it is fed a noisy signal from a poorly balanced input with insufficient common mode rejection. Thus, in order to achieve the best possible performance at the lowest possible cost, it is important that designers employ good analog engineering practices. This seminar will provide tips and techniques to achieve optimal performance in the analog signal path. Topics to be covered will range from inputs (balanced line and microphone level) to outputs (balanced to single-ended) and will include analog signal processing (VCAs, RMS detection, and Analog Engines). The presenters are analog IC design experts from THAT Corporation, the world's only semiconductor foundry specializing exclusively in pro audio ICs.

Session P9 **Saturday, October 6**
12:30 pm – 5:00 pm **Room 1E07**

AUDIO CODING

Chair: **James Johnston**, Microsoft Corporation, Redmond, WA, USA

12:30 pm

P9-1 Network Music Performance with Ultra-Low-Delay Audio Coding under Unreliable Network Conditions—*Ulrich Kraemer,*[1] *Jens Hirschfeld,*[1] *Gerald Schuller,*[1] *Stefan Wabnik,*[1] *Alexander Carôt,*[2] *Christian Werner*[2]
[1]Fraunhofer IDMT, Ilmenau, Germany
[2]University of Lübeck, Lübeck, Germany

A key issue for successfully interconnecting musicians in real-time over the Internet is minimizing the end-to-end signal delay for transmission and coding. The variance of transmission delay ("jitter") occasionally causes some packets to arrive too late for playback. To avoid this problem previous approaches are working with rather large receive buffers while accepting larger delay. In this paper we will present a novel solution that keeps buffer sizes and delay minimal. On the network layer we are using a highly optimized audio framework called "Soundjack" and on the coding layer we are working with an ultra low-delay codec for high-quality audio. We analyze and evaluate a modified transmission and coding scheme for the Fraunhofer Ultra-Low-Delay (ULD) audio coder, which is designed to be more resilient to lost and late arriving data packets.
Convention Paper 7214

1:00 pm

P9-2 A Very Low Bit-Rate Protection Layer to Increase the Robustness of the AMR-WB+ Codec against Bit Errors—*Philippe Gournay*, University of Sherbrooke, Sherbrooke, Quebec, Canada

Audio codecs face various channel impairments when used in challenging applications such as digital radio. The standard AMR-WB+ audio codec includes a concealment procedure to handle lost frames. It is also inherently robust to bit

errors, although some bits within any given frame are more sensitive than others. Motivated by this observation, the present paper makes two contributions. First, a detailed study of the sensitivity of individual bits in AMR-WB+ frames is provided. All the bits in a frame are then divided into three sensitivity classes so that efficient unequal error protection (UEP) schemes can be designed. Then, a very low bit rate protection layer to increase the robustness of the codec against bit errors is proposed and assessed using the results of subjective audio quality tests. Remarkably, in contrast to the standard codec, where some errors have a very discernable effect, the protection layer ensures that the decoded audio is free of major channel artifacts even at a significant 0.5 percent bit error rate.
Convention Paper 7215

1:30 pm

P9-3 Trellis Based Approach for Joint Optimization of Window Switching Decisions and Bit Resource Allocation—*Vinay Melkote, Kenneth Rose*, University of California at Santa Barbara, Santa Barbara, CA, USA

The fact that audio compression for streaming or storage is usually performed offline alleviates traditional constraints on encoding delay. We propose a rate-distortion optimized approach, within the MPEG Advanced Audio Coding framework, to trade delay for optimal window switching and resource allocation across frames. A trellis is constructed where stages correspond to audio frames, nodes represent window choices, and branches implement transition constraints .A suitable cost comprising bit consumption and psychoacoustic distortion, is optimized via multiple passes through the trellis until the desired bit-rate is achieved. The procedure offers optimal window switching as well as better bit distribution than conventional bit-reservoir schemes that are restricted to “borrow” bits from past frames. Objective and subjective tests show considerable performance gains.
Convention Paper 7216

2:00 pm

P9-4 Transcoding of Dynamic Range Control Coefficients and Other Metadata into MPEG-4 HE AAC—*Wolfgang Schildbach,*[1] *Kurt Krauss*,[1] *Jonas Rödén*[2]
[1]Coding Technologies, Nuremberg, Germany
[2]Coding Technologies, Stockholm, Sweden

With the introduction of HE-AAC (also known as aacPlus) into several new broadcasting systems, the topic of how to best encode new and transcode pre-existing metadata such as dynamic range control (DRC) data, program reference level and downmix coefficients into HE-AAC has gained renewed interest. This paper will discuss the means of carrying metadata within HE-AAC and derived standards like DVB, and present studies on how to convert metadata persistent in different formats into HE-AAC. Listening tests are employed to validate the results.
Convention Paper 7217

2:30 pm

P9-5 Advanced Audio for Advanced IPTV Services—*Roland Vlaicu, Oren Williams*, Dolby Laboratories, San Francisco, CA, USA

Television service providers have significant new requirements for audio delivery in next-generation broadcast systems such as high-definition television and IPTV. These include the capability to deliver soundtracks from mono to 5.1 channels and beyond with greater efficiency than current systems. Compatibility with existing consumer home cinema systems must also be maintained. A new audio delivery system, Enhanced AC-3, has been developed to meet these requirements, and has been standardized in DVB, . . ., as well as in ATSC. Also, Enhanced AC-3 is being included in widely used middleware solutions and paired with RTP considerations. This paper describes how operators can manage multichannel assets on linear broadcast turn-around and video-on-demand services in order to provide a competitive IPTV offering.
Convention Paper 7218

3:00 pm

P9-6 A Study of the MPEG Surround Quality versus Bit-Rate Curve—*Jonas Rödén*,[1] *Jeroen Breebaart*,[2] *Johannes Hilpert*,[3] *Heiko Purnhagen*,[1] *Erik Schuijers,*[4] *Jeroen Kippens*,[4] *Karsten Linzmeier,*[3] *Andreas Hölzer*[3]
[1]Coding Technologies, Stockholm, Sweden
[2]Philips Research Laboratories, Eindhoven, The Netherlands
[3]Fraunhofer Institute for Integrated Circuits, Erlangen, Germany
[4]Philips Applied Technologies, Eindhoven, The Netherlands

MPEG Surround provides unsurpassed multichannel audio compression efficiency by extending a mono or stereo audio coder with additional side information. This compression method has two important advantages. The first is its backward compatibility, which is important when MPEG Surround is employed to upgrade an existing service. Second, the amount of side information can be varied over a wide range to enable high-quality multichannel audio compression at extremely low bit rates up to perceptual transparency at higher bit rates. The present paper provides a study of the performance of MPEG Surround, highlighting the various tradeoffs that are available when using MPEG Surround. Furthermore a quality versus bit rate curve describing the MPEG Surround performance will be presented.
Convention Paper 7219

3:30 pm

P9-7 Quality Impact of Diotic Versus Monaural Hearing on Processed Speech—*Arnault Nagle,*[1] *Catherine Quinquis,*[1] *Aurélien Sollaud,*[1] *Anne Battistello*,[1] *Dirk Slock*[2]
[1]France Telecom Research and Development, Lannion, France
[2]Institut Eurecom, Antipolis Cedex, France

In VoIP audio conferencing, hearing is done ➥

over handsets or headphones, so through one or two ears. In order to keep the same loudness perception between the two modes, a listener can only tune the sound level. The goal of this paper is to show that monaural or diotic hearing has a quality impact on speech processed by VoIP coders. It can increase or decrease the differences in perceived quality between tested coders and even change their ranking according to the sound level. This impact on the ranking of the coders will be explained thanks to the normal equal-loudness-level contours over headphones and the specifics of some coders. It is important to be aware of the impact of the hearing system and its associated sound level.
Convention Paper 7220

4:00 pm

P9-8 A Novel Audio Post-Processing Toolkit for the Enhancement of Audio Signals Coded at Low Bit Rates—*Raghuram Annadana*,[1] *Harinarayanan E.V.*,[1] *Deepen Sinha*,[2] *Anibal Ferreira*[2,3]
[1]ATC Labs, Noida, India
[2]ATC Labs, Chatham, NJ, USA
[3]University of Porto, Porto, Portugal

Low bit rate audio coding often results in the loss of a number of key audio attributes such as audio bandwidth and stereo separation. Additionally, there is also typically a loss in the level of details and intelligibility and/or warmth in the signal. Due to the proliferation, e.g., on Internet, of low bit rate audio coded using a variety of coding schemes and bit rates over which the listener has no control, it is becoming increasingly attractive to incorporate processing tools in the player that can ensure a consistent listener experience. We describe a novel post-processing toolkit which incorporates tools for (i) stereo enhancement, (ii) blind bandwidth extension, (iii) automatic noise removal and audio enhancement, and, (iv) blind 2-to-5 channel upmixing. Algorithmic details, listening results, and audio demonstrations will be presented.
Convention Paper 7221

4:30 pm

P9-9 Subjective Evaluation of Immersive Sound Field Rendition System and Recent Enhancements—*Chandresh Dubey,*[1] *Raghuram Annadana*,[1] *Deepen Sinha*,[2] *Anibal Ferreira*[2,3]
[1]ATC Labs, Noida, India
[2]ATC Labs, Chatham, NJ, USA
[3]University of Porto, Porto, Portugal

Consumer audio applications such as satellite radio broadcasts, multichannel audio streaming, and playback systems coupled with the need to meet stringent bandwidth requirements are eliciting newer challenges in parametric multichannel audio coding schemes. This paper describes the continuation of our research concerning the Immersive Soundfield Rendition (ISR) system. In particular we present detailed subjective result data benchmarking the ISR system in comparison to MPEG Surround and also characterizing the audio quality level at different sub-modes of the system. We also describe enhancements to various algorithmic components in particular the blind 2-to-5 channel upmixing algorithm and describe a novel scheme for providing enhanced stereo downmix at the receiver for improved decoding by conventional matrix decoding systems.
Convention Paper 7222

Session P10 Saturday, October 6
12:30 pm – 3:30 pm Room 1E16

AUTOMOTIVE AUDIO AND AMPLIFIERS

Chair: **Richard Stroud**, Stroud Audio, Kokomo, IN, USA

12:30 pm

P10-1 Improved Stereo Imaging in Automobiles—*Michael Smithers*, Dolby Laboratories, Sydney, NSW, Australia

A significant challenge in the automobile listening environment is the predominance of off-axis listening positions. This leads to audible artifacts including comb filtering and indeterminate stereo imaging; both in traditional stereo and more recent multichannel loudspeaker configurations. This paper discusses the problem of off-axis listening as well as methods to improve stereo imaging in a symmetric manner using all-pass FIR and IIR filters. This paper also discusses a more efficient IIR filter design that achieves similar performance to previous filter designs. Use of these filters results in stable, virtual sources in front of off-axis listeners.
Convention Paper 7223

1:00 pm

P10-2 A Listening Test System for Automotive Audio—Part 3: Comparison of Attribute Ratings Made in a Vehicle with Those Made Using an Auralization System—*Patrick Hegarty, Sylvain Choisel, Søren Bech*, Bang & Olufsen a/s, Struer, Denmark

A system has been developed to allow listening tests of car audio sound systems to be conducted over headphones. The system employs dynamic binaural technology to capture and reproduce elements of an in-car soundfield. An experiment, a follow-up to a previous work, to validate the system is described. Seven trained listeners were asked to rate a range of stimuli in a car as well as over headphones for 15 elicited attributes. Analysis of variance was used to compare ratings from the two hardware setups. Results show the ratings for spatial attributes to be preserved while differences exist for some timbral and temporal attributes.
Convention Paper 7224

1:30 pm

P10-3 A Listening Test System for Automotive Audio—Part 4: Comparison of Attribute Ratings Made by Expert and Non-Expert Listeners—*Sylvain Choisel,*[1] *Patrick Hegarty*,[1]

Flemming Christensen,[2] *Benjamin Pedersen,*[2] *Wolfgang Ellermeier,*[2] *Jody Ghani,*[2] *Wookeun Song*[2]
[1]Bang & Olufsen a/s, Struer, Denmark
[2]Aalborg University, Aalborg, Denmark

A series of experiments was conducted in order to validate an experimental procedure to perform listening tests on car audio systems in a simulation of the car environment in a laboratory, using binaural synthesis with head-tracking. Seven experts and 40 non-expert listeners rated a range of stimuli for 15 sound-quality attributes developed by the experts. This paper presents a comparison between the attribute ratings from the two groups of participants. Overall preference of the non-experts was also measured using direct ratings as well as indirect scaling based on paired comparisons. The results of both methods are compared.
Convention Paper 7225

2:00 pm

P10-4 The Application of Direct Digital Feedback for Amplifier System Control—*Craig Bell, David Jones, Robert Watts*, Zetex Semiconductors, Oldham, UK

An effective feedback topology is clearly a beneficial requirement for a well performing digital amplifier. The ability to cancel corrupting influences such as power supply ripple and unmatched components is necessary for good sonic performance. Additional benefits derive from the fact that the feedback information is processed in the digital domain. Current delivered into the loudspeaker load can be inferred. The amplifier acts as a voltage source, the value of which is derived from the recorded source material. The current delivered into the loudspeaker is also clearly influenced by the load impedance, which varies with frequency and other factors. This paper describes the ability of the system to measure current and derive loudspeaker impedance and the actual delivered power and goes on to illustrate the applications in real systems.
Convention Paper 7226

2:30 pm

P10-5 Generation of Variable Frequency Digital PWM—*Pallab Midya*, Freescale Semiconductor Inc., Lake Zurich, IL, USA

Digital audio amplifiers convert digital PCM to digital PWM to be amplified by a power stage. This paper introduces a method to generate a quantized duty ratio digital PWM with a switching frequency over a 20 percent range to mitigate EMI issues. The method is able to compensate for the variation in switching frequency such that the SNR in the audio band is comparable to fixed frequency PWM. To obtain good rejection of the noise introduced by the variation of the PWM frequency higher order noise shapers are used. This paper describes in detail the algorithm for a fourth order noise shaper. Using this method dynamic range in excess of 120 dB unweighted over a 20 kHz bandwidth is achieved.
Convention Paper 7227

3:00 pm

P10-6 Recursive Natural Sampling for Digital PWM—*Pallab Midya, Bill Roeckner, Theresa Paulo*, Freescale Semiconductor Inc., Lake Zurich, IL, USA

This paper presents a highly accurate and computationally efficient method for digital-domain computation of naturally sampled digital pulse width modulation (PWM) signals. This method is used in a switching digital audio amplifier. The method is scalable for performance versus calculation complexity. Using a second order version of the algorithm with no iteration, intermodulation linearity of better than 113 dB is obtained with a full scale input at 19 kHz and 20 kHz. Matlab simulation and measured results from a digital amplifier implemented with this algorithm are presented. Overall system performance is not limited by the accuracy of the natural sampling method.
Convention Paper 7228

Master Class 2 **Saturday, October 6**
12:30 pm – 2:00 pm **Room 1E11**

KEITH O. JOHNSON

Presenter: **Keith O. Johnson**, Reference Recordings, Pacifica, CA, USA

The Art and Science of Making and Playing Great Recordings—The High Resolution Experience

A listener's emotional involvement with a recording and its reproduction depends on intangible factors such as imaging, transparency, articulation, and delineation. A brief overview shows how these features pose design and performance issues with halls, microphones, electronics, processes, and loudspeakers. Tests that fit each component are described and perceptual hearing models are introduced to interpret results. Topics, presented from a multichannel perspective, include how instruments, halls, and visual cues create a live concert experience and how microphone choices, placements, and mix-in strategies can constrict or expand a soundscape. Other topics include time corrected stereo accenting and eigensonic corrected loudspeakers for supplemental hall driving as well as reduction of loudspeaker presence and front biased imaging from playback systems.

Keith Johnson has spent over thirty years developing a reputation for innovative thinking, technical achievement, and musicianship. He is one of the recording industry's visionaries. His intensive investigation of electronic behavior and acoustic perception led to his development (with digital engineer Michael Pflaumer) of the revolutionary High Definition Compatible Digital encoding process. For more than twenty years Johnson has served as Technical Director, Recording Engineer, and partner in Reference Recordings. His 100-plus recordings for the label have long been considered the standard for high fidelity, and include three GRAMMY award-winners and eight additional GRAMMY nominations.

Student Event/Career Development
RESUME REVIEW
Saturday, October 6, 12:30 pm – 2:00 pm
Room 1E06

Reviewer: **John Strawn**

This session is aimed at job candidates in electrical engineering and computer science who want a private, no-cost, no-obligation confidential review of their resume. You can expect feedback such as: what is missing from the resume; what you should omit from the resume; how to strengthen your explanation of your talents and skills. Recent graduates, juniors, seniors, and graduate students who are now seeking, or will soon be seeking, a full-time employment position in the audio and music industries in hardware or software engineering will especially benefit from participating, but others with more industry experience are also invited. You will meet one-on-one with someone from a company in the audio and music industries with experience in hiring for R&D positions. Bring a paper copy of your resume and be prepared to take notes. Signup sheets for a specific appointment will be located near the Student Delegate Assembly meeting room.

Exhibitor Seminar **Saturday, October 6**
12:30 pm – 2:30 pm **Room 1E04**

RENKUS HEINZ

Presenters: **Wolfgang Ahnert, Stefan Feistel**

Tools for Measuring and Optimizing Loudspeaker Systems

The Ahnert/Feistel Media Group will present EASERA SysTune, a new software tool for quick and accurate system tuning using live program material while an audience is present. The presenters will discuss actual applications and share their extensive experience with measuring and tuning complex loudspeaker systems for live sound reinforcement.

Workshop 11 **Saturday, October 6**
1:00 pm – 3:00 pm **Room 1E08**

SURROUND CONTRIBUTION FOR RADIO AND TV

Chair: **Jon McClintock**, APT

Panelists: *Tim Caroll*, Linear Acoustic
Kimio Hamasaki, NHK, Tokyo, Japan
Heinz Peter Reykers, WDR, Cologne, Germany
Geir Skaaden, Neural Audio

This workshop will cover technology over various types of networks for surround sound contribution for radio and tv (not emission systems).

Many systems operate over E1 or T1 synchronous networks. Managed private IP-networks are also now beginning to be used. The Internet is slowly coming with higher bit rates and may in the future also be used for surround contribution of professional audio formats.

Panelists will discuss the pros and cons of various solutions and present practical implementations as case studies.

Workshop 12 **Saturday, October 6**
1:00 pm – 3:00 pm **Room 1E15**

ADAPTIVE MUSIC FOR GAMES: INTERACTIVE XMF IN-DEPTH

Chair: **Chris Grigg**, IASIG

The Interactive eXtensible Music Format (iXMF) is the first nonproprietary standard for interactive audio content, recently produced by the Interactive Audio SIG (IASIG). Technical lead Chris Grigg explains iXMF technology and its benefits for audio artists, game designers, and game engine designers. iXMF addresses the difficulty and expense of interactive audio by providing (1) a nonproprietary, cross-platform file format, and (2) an abstract model for the runtime audio engine. The format bundles audio assets with dynamic behavior rules. The model defines how the rules are interpreted in real time to play those assets. Because the model is based on a handful of simple low-level primitives, it is both very portable and very general. Scenario testing indicates any known interactive audio behavior style is possible.

Broadcast Session 7 **Saturday, October 6**
1:00 pm – 2:00 pm **Room 1E10**

BROADCAST TUTORIAL: UNDERSTANDING RF FROM MICROPHONES TO ANTENNAS

Presenters: **Doug Irwin**, WKTU
Chris Scherer, Radio Magazine

Just What Is RF, Anyway?

An introduction to RF and a primer on its use. Emphasis on the uses of RF with respect to AM/FM radio broadcast technology; optimizing the use of RF systems such as wireless microphone and IFB systems; and finally mitigation of common problems associated with RF, such as interference.

Special Event
19TH ANNUAL GRAMMY RECORDING SOUNDTABLE
Saturday, October 6, 1:00 pm – 3:00 pm
Room 1E12/13

Moderator: **Zoe Thrall**

Panelists: *Tony Bongiovi*
Bob Clearmountain
Jason Corsaro
Neil Dorfsman
James Farber
Nile Rodgers

The 19th Annual GRAMMY Recording SoundTable is presented by the National Academy of Recording Arts & Sciences Inc. (NARAS) and hosted by AES.

Good Times: The Enduring Legacy of NY's Power Station Studios

Founded in 1977, New York's legendary Power Station Studios can claim the birth of landmark recordings by Aerosmith, the B-52s, Chic, Dire Straits, Duran Duran, Madonna, Iggy Pop, Roxy Music, Bruce Springsteen, John Lennon, and The Power Station, among many, many others. Moderated by Studio at the Palms Director Zoe Thrall, this special event will feature the Power Station Studio's lynchpin engineers and producers—and their music—including Bob Clearmountain, Jason Corsaro, Neil Dorfsman, James Farber and Nile Rodgers, along with studio owner and designer Tony Bongiovi, delving into just what it was that created the scene, the sound—and especially, the music—that continues to inspire us today.

Technical Committee Meeting **Saturday, October 6**
1:30 pm **Room 1E05**

Technical Committee Meeting on Human Factors in Audio Systems.

Session P11 **Saturday, October 6**
2:00 pm – 3:30 pm **Foyer 1E**

POSTERS: PERCEPTION

2:00 pm

P11-1 Impact of Equalizing Ear Canal Transfer Function on Out-of-Head Sound Localization
—*Masataka Yoshida*,[1] *Akihiro Kudo*,[2] *Haruhide Hokari,*[1] *Shoji Shimada*[1]
[1]Nagaoka University of Technology, Nagaoka, Niigata, Japan
[2]Tomakomai National College of Technology, Tomakomai, Hokkaido, Japan;

Several papers have pointed out that the frequency characteristics of the ear canal transfer functions (ECTFs) depend on headphone type, ear placement position of headphones, and subject's ear canal shape/volume. However, the effect of these factors on creating out-of-head sound localization has not been sufficiently clarified. The purpose of this paper is to clarify this effect. Sound localization tests using several types of headphones are performed in three conditions: listener's (individualized) ECTFs, HATS's (non-individualized) ECTFs, and omitted ECTFs. The results show that employing the individualized ECTFs generally yields accurate localization, while omitting the use of ECTFs increase the horizontal average localization error in accordance with the type of headphone employed.
Convention Paper 7229

2:00 pm

P11-2 A Method for Estimating the Direction of Sound Image Localization for Designing a Virtual Sound Image Localization Control System—*Yoshiki Ohta, Kensaku Obata*,
Pioneer Corporation, Tsurugashima-city, Saitama, Japan

We developed a method of estimating the direction of sound image localization. Our method is based on the sound pressure distribution in the vicinity of a listener. In the experiment, band noises that only differ in phase were produced from two loudspeakers. We determined what relation existed between the subjective direction of the sound image localization and the objective sound pressure distribution in the vicinity of the listener. We found that an azimuth of localization can be expressed as a linear combination of sound pressure levels in the vicinity of the listener. Our method can be used to estimate azimuths with a high degree of accuracy and to associate phase differences with azimuths. Therefore, it can be used to design a system for controlling virtual sound image localization.
Convention Paper 7230

2:00 pm

P11-3 A Preliminary Experimental Study on Perception of Movement of a Focused Sound Using a 16-Channel Loudspeaker Array—
Daiki Sato,[1] *Teruki Oto*,[2] *Kaoru Ashihara*,[3] *Ryuzo Horiguchi*,[1,2] *Shogo Kiryu*[1]
[1]Musashi Institute of Technology, Setagaya-ku, Tokyo, Japan
[2]Kenwood Corporation, Tokyo, Japan
[3]Advanced Industrial Science and Technology, Tsukuba, Japan

We have been developing a sound field effecter by using a loudspeaker array. In order to design a practical system, psychoacoustic experiments for recognition of sound fields are required. In this paper perception of a sound focus is investigated using a 16-channel loudspeaker array. Listening experiments were conducted in an anechoic room and a listening room. The movement of 25 cm in horizontal direction and the movement of 100 cm in the direction from the loudspeaker array toward the subject could be recognized in both rooms, but that in the vertical could not be perceived in both rooms.
Convention Paper 7231

2:00 pm

P11-4 Perceptual Categories of Artificial Reverberation for Headphone Reproduction of Music—*Atsushi Marui*, Tokyo National University of Fine Arts and Music, Tokyo, Japan

In the studies of artificial reverberations, the focus is usually on recreating the natural reverberation that can be heard in the real environment. However, little attention was paid to the evaluation of useful ranges in application of the artificial reverberation in music production. The focus of this paper is to discuss and evaluate three artificial reverberation algorithms intended for headphone reproduction of music, and to propose iso-usefulness contour on those algorithms for several different types of musical sounds.
Convention Paper 7232

2:00 pm

P11-5 Correspondence Relationship between Physical Factors and Psychological Impressions of Microphone Arrays for Orchestra Recording—*Toru Kamekawa,*[1]
Atsushi Marui,[1] *Hideo Irimajiri*[2]
[1]Tokyo National University of Fine Arts and Music, Tokyo, Japan
[2]Mainichi Broadcasting Corporation, Kita-ku, Osaka, Japan

Microphone technique for the surround sound recording of an orchestra is discussed. Eight types of well known microphone arrays recorded in a concert hall were compared in subjective listening tests on seven attributes such as spaciousness, powerfulness, and localization using a method inspired by MUSHRA (MUltiple Stimuli with Hidden Reference and Anchor). The result of the experiment shows similarity and dissimilarity between each microphone array. It is estimated that directivity of a microphone and distance between ➥

each microphone are related to the character of the microphone array, and these similarities are changed by music character. The relations of the physical factors of each array were also compared, such as SC (Spectral Centroid), LFC (Lateral Fraction Coefficient), and IACC (Inter Aural Cross-correlation Coefficient) from the impulse response of each array or recordings by a dummy head. The correlation of these physical factors and the attribute scores show that the contribution of these physical factors depends on music.
Convention Paper 7233

2:00 pm

P11-6 Assessment of the Quality of Digital Audio Reproduction Devices by Panels of Listeners of Different Professional Profiles—*Piotr Odya*,[1] *Marek Pluta*,[2] *Szymon Piotrowski*[1]
[1]AGH University of Science and Technology, Krakow, Poland
[2]Academy of Music in Cracow, Krakow, Poland

A series of experiments has been conducted, where different panels of listeners assessed the quality of some selected digital audio reproduction devices. The quality of the devices covered a very wide range from budget MP3 players through to a professional high resolution digital-to-analog conversion system. The main goal of this research was to investigate whether panels of listeners of different professional profiles are able to give different evaluations of the sound quality. Some interesting results have been obtained.
Convention Paper 7234

Broadcast Session 8 — Saturday, October 6
2:00 pm – 4:00 pm — Room 1E10

AUDIO FOR HDTV: THE LIP SYNC ISSUE

Chair: **Brad Dick**, Broadcast Engineering Magazine

Panelists: *Randy Conrod*, Harris Broadcast Communications Division
Ken Hunnold, Dolby Laboratories
Chris Smith, Pixel Instruments

One of the big issues of HDTV presentation is trying to sync the audio with the picture. This is a problem caused by high definition video's extensive processing requirements: from pre- and postproduction, to broadcast, to playback on your TV—resulting in audio being delivered a split second ahead of the video.

Live Sound Seminar 6 — Saturday, October 6
2:00 pm – 4:00 pm — Room 1E09

LARGE SCALE MULTICHANNEL, MULTISYSTEM WIRELESS DESIGN, DEPLOYMENT, AND OPERATION

Chair: **James Stoffo**

Panelists: *Peter Erskine*, BEST Audio
Stephen Mendelsohn, ABC TV
Gary Stocker, Masque Sound
Ed Wieczorek

Today's mega-productions require hundreds of channels of wireless microphones, in-ear monitors, intercom, and interruptible foldback (IFB). This panel will discuss and outline the strategy, tactics, and practices of implementing these large multiple wireless systems in RF challenging environments, from the first phone call for the production through the event itself.

Student Event/Career Development
RECORDING COMPETITION
STEREO/JAZZ/FOLK
Saturday, October 6, 2:00 pm – 4:00 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. Student members can submit stereo and surround recordings in the categories classical, jazz, folk/world music, and pop/rock. Meritorious awards will be presented at the closing Student Delegate Assembly Meeting on Monday.

Judges include: Martha De Francisco, David Frost, Richard King, Kimio Hamasaki, Ronald Prent, Andreas Mayo, Jim Anderson, Peter Cook, Tim Martyn, Rebecca Pulliam, Bob Katz, Michael Nunan, and Ronald Sadoff.

Sponsors for this event include: AEA, Yamaha, Genelec, Harmon/JBL, Sony Creative Software, Neumann, Shure, Lavry Engineering, Schoeps, Millennia Music and Media Systems, PCM, and Sennheiser.

Standards Committee Meeting — Saturday, October 6
2:00 pm — Room 1E02

Standards Committee Meeting AESSC Plenary 1.

Exhibitor Seminar — Saturday, October 6
2:30 pm – 3:30 pm — Room 1E04

AUDIOMATICA

Presenters: **Mauro Bigi, Daniele Ponteggia**

CLIO 8 System Overview and Applications

Audiomatica presents the new CLIO 8 System and demonstrates its new applications. Of great interest: the firewire hardware interface; the security features of CLIO 8 software; the new Quality Control functionalities; and the new software analysis tools.

Technical Committee Meeting — Saturday, October 6
2:30 pm — Room 1E05

Technical Committee Meeting on Studio Practices and Production.

Tutorial 6 — Saturday, October 6
3:00 pm – 5:00 pm — Room 1E15

ARCHITECTURAL ACOUSTICS, PART 2

Presenter: **Tony Hoover**, Mckay Conant Hoover, Inc., Westlake Village, CA, USA

This tutorial emphasizes architectural acoustics consulting, which is the art of applying the science of sound in a timely, practical, and easily-understood manner. Three distinct areas are covered: sound isolation (airborne and structurborne), HVAC noise control, and surface treatments (absorption, reflection, and diffusion). The format has been used for decades in meetings and seminars, and for over 40 semesters at Berklee College of Music.

The objective is to provide an overview of fundamentals and terminology, bases for design decisions and evaluations, and confidence to navigate the oceans of information, misinformation, and propaganda about "acoustical" materials, products, and practices.

Tutorial 7 **Saturday, October 6**
3:00 pm – 5:00 pm **Room 1E08**

ADAPTIVE MUSIC FOR GAMES: INTRODUCTION TO NONLINEAR MUSIC

Presenters: **Simon Amarasingham**, dsonic
Brian Gomez, Alchemic Productions

Linear music in a nonlinear world? How can a completely interactive and unpredictable environment like a video game reproduce continuous musical content that adapts to gameplay? This tutorial will provide an insiders' view to the creation and implementation of adaptive music. Discussed will be various approaches for creating adaptive music; dealing with the game environment, what the game designer wants, object-oriented thinking vs. multitrack thinking, and more. While existing tools for adaptive music are barely past their infancy, a description of the available game audio tools will be presented. Following the tutorial will be a demonstration of adaptive music dSonic created for Unreal Tournament 2004 using ISACT—an under-the-hood view of a real world example of constructing adaptive music.

Special Event
LOW DELAY CODEC OPTIMIZED NETWORK MUSIC PERFORMANCE
Saturday, October 6, 3:00 pm – 6:00 pm
New York University, 35 West 4th St. Room 303

Panelists: *Alexander Carôt*, University of Lübeck, International School of New Media
Jeremy Cooperstock
Ulrich Kraemer
Robert Rowe
Gerald Schuller

This demonstration will show musicians playing together remotely, using the Internet as a connection between them. In our demonstration session we will establish a connection between New York University and McGill University and play jazz music as if the musicians were located in the same room. We will show a system that can even be used over home DSL or cable Internet connections.

Having musicians play together over the Internet presents several challenges in terms of low delay audio capture and transmission, time synchronization, and bandwidth requirements. One of the main problems for this kind of low delay musical interaction is the network jitter, which leads to audio data packets that arrive too late for playback on the receiving side. Previous approaches used uncompressed audio for transmission. Our new approach is to use compressed audio, with a packet loss resilient version of our Ultra Low Delay audio coder, to accommodate lower bit rate connections and to reduce the network jitter by reducing the load on the network connection. For the network connection the Soundjack software will be used, which can be described as a low latency UDP streaming application.

This Special Event is free to participants and buses will be provided to and from the venue. Seating is limited, however, and tickets will be required. They may be obtained at the AES Tours Desk.

Session P12 **Saturday, October 6**
3:30 pm – 5:30 pm **Room 1E16**

AUDIO CONTENT MANAGEMENT

Chair: **Rob Maher**, Montana State University, Bozeman, MT, USA

3:30 pm

P12-1 Music Structure Segmentation Using the Azimugram in Conjunction with Principal Component Analysis—*Dan Barry, Mikel Gainza, Eugene Coyle*, Dublin Institute of Technology, Dublin, Ireland

A novel method to segment stereo music recordings into formal musical structures such as verses and choruses is presented. The method performs dimensional reduction on a time-azimuth representation of audio, which results in a set of time activation sequences, each of which corresponds to a repeating structural segment. This is based on the assumption that each segment type such as verse or chorus has a unique energy distribution across the stereo field. It can be shown that these unique energy distributions along with their time activation sequences are the latent principal components of the time-azimuth representation. It can be shown that each time activation sequence represents a structural segment such as a verse or chorus.
Convention Paper 7235

4:00 pm

P12-2 Using the Semantic Web for Enhanced Audio Experiences—*Yves Raimond, Mark Sandler*, Queen Mary, University of London, London, UK

In this paper we give a quick overview of some key Semantic Web technologies, allowing us to overcome the limitations of the current web of documents to create a machine-processable web of data, where information is accessible by automated means. We then detail a framework for dealing with audio-related information on the Semantic Web: the Music Ontology. We describe some examples of how this ontology has been used to link together heterogeneous data sets, dealing with editorial, cultural or acoustic data. Finally, we explain a methodology to embed such knowledge into audio applications (from digital jukeboxes and digital archives to audio editors and sequencers), along with concrete examples and implementations.
Convention Paper 7236

4:30 pm

P12-3 Content Management Using Native XML and XML-Enabled Database Systems in Conjunction with XML Metadata Exchange Standards—*Nicolas Sincaglia*, DePaul University, Chicago, IL, USA

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The digital entertainment industry has developed communication standards to support the distribution of digital content using XML technology. Recipients of these data communications are challenged when transforming and storing the hierarchical XML data structures into more traditional relational database structures for content management purposes. Native XML and XML-enabled database systems provide possible solutions to many of these challenges. This paper will consider several data modeling design options and evaluate the suitability of these alternatives for content data management.
Convention Paper 7237

5:00 pm

P12-4 Music Information Retrieval in Broadcasting: Some Visual Applications—*Andrew Mason, Michael Evans, Alia Sheikh*, British Broadcasting Corporation Research, Tadworth, Surrey, UK

The academic research field of music information retrieval is expanding as rapidly as the MP3 collection of a stereotypical teenager. This could be no coincidence: the benefit of an automated genre classifier increases when the music collection contains several thousand tracks. Of course, there are other applications of music information retrieval. Here we highlight a few that make use of a simple, visual, representation of an audio signal, based on three easy-to-calculate audio features. The applications range from simple navigation around consumer recordings of broadcasts, to a music video production planning tool, to a short term "Listen Again" eye-catching display.
Convention Paper 7238

Master Class 3 **Saturday, October 6**
3:30 pm – 5:00 pm **Room 1E12/13**

MICHAEL BRAUER

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

Mixing—From Rough Mix To "Yogi"

Michael Brauer is an accomplished and eclectic mixer with mix credits that include Luther Vandross, Aretha Franklin, The Rolling Stones, Aimee Mann, Coldplay, Ben Folds, and John Mayer's critically acclaimed CD *Continuum*. Michael will talk about some key aspects of mixing that cannot be purchased in a box, whether hardware or software—the human aspects. How do you prepare yourself to mix a song that you are hearing for the first time? How do you balance the power of sitting at the console with the need to remain humble when discussing the song and mix with the artist? How do you work not only with the tracks, but also with the needs and personalities of the band, manager, and A&R? How do you make the leap from a mix that just documents the song, to one that brings out its spirit and message? Michael will also talk tech about his multi-stage/multi-buss mix technique and also show how to adapt it to mixing "ITB" with Pro Tools. He might even explain the term "Yogi."

Technical Committee Meeting **Saturday, October 6**
3:30 pm **Room 1E05**

Technical Committee Meeting on Audio Recording and Mastering Systems.

Standards Committee Meeting **Saturday, October 6**
3:30 pm **Room 1E02**

Standards Committee Meeting SC-03-02 Transfer Technologies.

Broadcast Session 9 **Saturday, October 6**
4:00 pm – 6:00 pm **Room 1E10**

LISTENER FATIGUE AND LISTENER RETENTION

Chair: **David Wilson**, CEA

Presenters: *Frank Foti*, Omnia
Thomas Lund, TC Electronics
David Reaves
Ellyn Sheffield

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.

Tutorial 8 **Saturday, October 6**
4:00 pm – 6:00 pm **Room 1E11**

AUDIO EAR TRAINING—AN ESSENTIAL RESOURCE

Presenter: **David Moulton**, David Moulton Labs., Groton, MA, USA

Audio ear training is a study sequence that enables listeners to effectively recognize and describe the audio spectrum, relative amplitudes, time intervals, and signal processing effects without external objective measurement. This tutorial will present a typical array of drills, interspersed with explanations of the various perceptual problems and processes that occur when using these techniques.

Live Sound Seminar 7 **Saturday, October 6**
4:00 pm – 5:00 pm **Room 1E09**

WHITESPACE STATUS REPORT: THE CURRENT OUTLOOK FOR LICENSE-FREE BROADBAND "WHITESPACE" DEVICES IN THE UHF TV SPECTRUM AND HOW WE AS THE ENTERTAINMENT INDUSTRY ARE PREPARING

Panelists: *Mark Brunner*, Shure
David Donovan, MSTV
Ed Reihl, Shure

As whitespace proponents and the entertainment industry vie for the post-DTV transition UHF spectrum, and the FCC considers all sides relative to its congressional mandate, key industry individuals at the frontline of this issue will present an up to the minute report on legislation, device testing, FCC decisions, and industry preparations.

Student Event/Career Development
"STEPS TO THE FUTURE"—ONE ON ONE MENTORING SESSION, PART 2
Saturday, October 6, 4:00 pm – 6:00 pm
Room 1E06

Students are invited to sign-up for an individual meeting with a distinguished mentor from the audio industry. The opportunity to sign up will be given at the end of the opening SDA meeting. Any remaining open spots will be posted in the student area. All students are encouraged to participate in this exciting and rewarding opportunity for individual discussion with industry mentors.

Exhibitor Seminar **Saturday, October 6**
4:00 pm – 5:00 pm **Room 1E04**

KLIPPEL

Presenter: **Wolfgang Klippel**

Vibration and Radiation Analysis of Loudspeaker Cones

A new scanning technique for measuring the vibration and geometry of loudspeaker parts is presented. This data is the basis for numerical simulations using BEM and FEA. A decomposition technique reveals radial and circular modes and the vibration component generating the acoustical output. New ways for loudspeaker diagnostics and cone optimization are illustrated on woofers, tweeters, headphones, and micro-speakers.

Technical Committee Meeting **Saturday, October 6**
4:30 pm **Room 1E05**

Technical Committee Meeting on Automotive Audio.

Workshop 13 **Saturday, October 6**
5:00 pm – 7:00 pm **Room 1E09**

INTERACTIVE AUDIO AND HUMAN PERCEPTION—CAN WE CONNECT KNOWLEDGE WITH PRACTICE?

Chair: **Renato Pellegrini**, sonic emotion ag, Obergltt (Zurich), Switzerland

Panelists: *Durand Begault*, NASA Ames Research Center, Mountainview, CA, USA
Jyri Huopaniemi, Nokia, Finnland
John-Marc Jot, Creative Advanced Technology Center, Ottawa, Ontario, Canada
William Martens, McGill University ,Montreal, Quebec, Canada

The term "Interactive audio" is used here in its most general sense to mean real-time modification of an audio signal in which attributes of a reproduced sound source (such as the source's apparent position, timbral character, etc.) can be controlled by a user's actions (using interaction interfaces such as trackers, game-controllers, joysticks, etc.). Interactive audio is a growing field in today's audio environments. The complexity of interactive environments in computer games and simulations continues to grow. Understanding the perceptual effects of this increase in complexity is becoming a greater challenge. This workshop investigates these perceptual effects by exploring the design of interactive spaces and by highlighting what is already known from established techniques for generating virtual environments. Most elaborate systems can be found in games and simulators, but also telecommunication products and even music listening and production make use of interactive systems such as head-trackers, game-controllers, joysticks, etc. This workshop is jointly elaborates psychoacoustic evidence with current product knowledge from the different market fields.

Tutorial 9 **Saturday, October 6**
5:00 pm – 6:30 pm **Room 1E07**

PREPARING YOUR MIXES FOR MASTERINGS

Presenter: **Adam Ayan**, Gateway Mastering & DVD, Portland, ME, USA

Mastering is often described as the final step in the creative process of making a record and the first step in manufacturing. These days, it's common to receive mixes for a project from multiple engineers, producers, and studios—in multiple formats. The job of the mastering engineer is to bring the mixes to their fullest musical potential and combine all of these sources into a cohesive-sounding album. To do that, it's vital that all source mixes are clearly labeled and that all source media and mastering notes are well organized. This tutorial will cover all facets of preparing mixes for mastering, from technical and organizational aspects to final creative and musical mix decisions.

Master Class 4 **Saturday, October 6**
5:00 pm – 6:30 pm **Room 1E08**

BRUNO PUTZEYS

Presenter: **Bruno Putzeys**, Hypex Electronics, Rotselaar, Belgium

Hard Earned Design Secrets—Reconciling the Objectivist and Subjectivist Points of View

Topics to be discussed:
- *Revealed preference*: Aligning philosophical preference with actual behavior. Your listening habits tell you whether you're after transparent reproduction or emotionally engaging sound—- whether you like it or not.
- *Op amp theory refresher*: Understanding how they work, not what they're made of. An intermediate level of abstraction between the "three pin triangle" and the transistor implementation simplifies analysis and understanding.
- *The op amp's two dirty little secrets*: PSRR and input common-mode distortion—opportunities for improving amplifier performance by discrete implementation.
- *Minimalist design, or not*—When adding more parts really helps.
- *When does negative feedback sound bad?*
- *Intelligent design in audio*: Recipes for success and disaster in co-developing and subcontracting audio electronics.
- *Assessing EMI behavior in class D amplifiers*.
- *Requirements for switch-mode power supplies for audio*.
- *That digital sound*: Deviating from theory is the problem, not the solution: an exposé of digital filtering and its relation to "the digital sound."
- *Asynchronous sample rate conversion*: The fine print: it's only 99% digital.
- *Digital loudspeaker EQ and cross-over*: Just anoth-

er tool in the box. Do's and don'ts of designing DSP filters for loudspeakers.

Bruno Putzeys is one of the world's foremost inventors working in digital audio. A *cum laude* graduate of the National Technical School for Radio and Film, he worked for ten years at the Philips Applied Technologies where he developed digital- and analog-controlled class D amplifiers, noise shapers and modulation methods, and invented the "UcD" class D circuit. In 2005 he left Philips to divide his time between Grimm Audio and Hypex Electronics. Current activities include designing ultra-high-performance discrete AD/DA converters and analog signal processing circuits, DSP algorithms, class D power amplifiers, and switch-mode power supplies.

Historical Program
LEGENDS OF NASHVILLE SOUND/
HISTORY OF NASHVILLE'S RECORDING STUDIOS
Saturday, October 6, 5:00 pm – 7:00 pm
Room 1E15

Moderators: **Jim Kaiser**
Bil VornDick

Panelists: *Wesley Bulla*, Mike Curb College of Entertainment & Music Business
Cosette Collier, Middle Tennessee State University
Michael Janas, Mike Curb College of Entertainment & Music Business

The development of a recording industry in Nashville began in the 1940s with facilities in local radio stations like WSM and later with Castle Recording Studios in 1946. With the dominance of live programming from the Grand Ole Opry in country radio and the move by major labels such as Victor, Columbia, and Decca to Nashville, there was a real need for more professional recording studios.

In 1954 Owen and Harold Bradley moved their recording studio to 16th Avenue South to become the first business on what would become known as "Music Row." This studio, dubbed "the Quonset Hut," was originally designed for recording songs for film but quickly became part of traditional country recording history with timeless hits by Patsy Cline.

RCA Studio B, built in 1957, is the oldest recording studio in Nashville. Designed on a napkin by Chet Atkins and his engineer Bill Miltenburg, it was home to a string of hits from 1957 to 1977 with artists including Perry Como, Al Hirt, the Everly Brothers, Bobby Goldsboro, the Monkees, Jerry Reed, Eddy Arnold, Willie Nelson, Charlie Pride, Dolly Parton, Roy Orbison, and Elvis.

Within these studio walls, engineers, musicians, producers, and artists created what will be forever known as "The Nashville Sound." This is their story, recounted by the people involved, including renowned recording engineer Bill Porter, producer Fred Foster, "A Team" musicians Harold Bradley, Bob Moore, Boots Randolph, Lloyd Green, Charlie McCoy, and others. In April, 2006, the AES Nashville Section sponsored a "Legends in the Round" roundtable discussion at the Country Music Hall of Fame with these illustrious participants, followed by an historic recreation at RCA Studio B of their recording of the hits "Crazy" (Patsy Cline), "Please Help Me I'm Falling" (Hank Locklin), "Yackety Sax" (Boots Randolph), and "Today I Started Loving You Again" (Charlie McCoy). Videotape from these two events will be a major part of the presentation, along with how these two recording studios have been restored to their original condition for use as both museums and hands-on educational opportunities for music industry students.

Standards Committee Meeting **Saturday, October 6**
5:00 pm **Room 1E02**

Standards Committee Meeting SC-03-04 Storage and Handling of Media.

Exhibitor Seminar **Saturday, October 6**
5:00 pm – 6:00 pm **Room 1E04**

FLEX ACOUSTICS

Presenter: **Niels Werner Adelman-Larsen**

Rock Acoustics—The Flexible Bass Absorber

Flex Acoustics made what seems to be the first ever scientific investigation into the recommended room acoustics for rock and pop concerts. The results showed the necessity of ad0ditional bass absorption and led to the invention of a flexible bass absorber. The investigation and the AqFlex absorber will be presented.

Technical Committee Meeting **Saturday, October 6**
5:30 pm **Room 1E05**

Technical Committee Meeting on Electro-Magnetic Compatibility.

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL
AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Saturday, October 6, 7:00 pm – 8:30 pm
Room 1E12/13

Lecturer: **Leo Beranek**

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 123rd AES Convention is Leo Beranek. Beranek received his Bachelor of Arts degree from Cornell College (Iowa) and his Doctor of Science from Harvard University. During World War II he headed the Electro-Acoustic Laboratory at Harvard. He served as Associate Professor of Communications Engineering at MIT from 1947 to 1958 and Technical Director of its Acoustics Laboratory. From 1952 to 1971 he was President of Bolt Beranek and Newman, one of the world's largest acousical consulting firms. A lifelong interest in music led him to specialize in concert hall and opera house acoustics. Following trips to over 100 of the world's leading halls and interviews of over a hundred conductors and music critics, he wrote three books on concert and opera halls, the most recent completely revised edition is *Concert Halls and Opera Houses: Music, Acoustics, and Architecture* (Springer-Verlag 2004).

Recently Beranek has been Acoustical Consultant for four concert halls, one opera house, and two drama theaters in Tokyo and has been consultant on many other concert halls, including the Tanglewood Music Shed in Western Massachusetts, the Aula Magna in Caracas, and the Meyerhoff Hall in Baltimore. He has recieved numerous awards, including Gold Medals from the

Acoustical Society of America and the Audio Engineering Society, Honorary Membership in the American Institute of Architects, the U.S. President's National Medal of Science in 2003 and the Per Bruel Gold Medal of the A.S.M.E in 2004. The title of his lecture is "Listening to Music in Spaces."

Listening to music performances in approximately 200 venues, consulting with architects on a number of them and assembling measured acoustical data on a hundred of them, have given the speaker a broad base for understanding the relative importance of the various acoustical parameters that either are being measured or are proposed for the evaluation of acoustics of spaces for music. The acoustical parameters treated are: Apparent Source Width (ASW), Listener Envelopment (LEV), Lateral Fraction (LF), Interaural Cross-Correlation Coefficient (IACC), Reverberation Time (RT30), Early Decay Time (EDT), Initial-Time-Delay Gap (ITDG), Strength (G) (and various substrengths including early and late relative levels, and early and late lateral relative levels), Perceived Bass, Texture, Just Noticeable Differences, and Instrumentation. The extents to which these parameters are being met in halls of different architectural designs are presented. Finally, the relation of these parameters to living-room listening venues will be touched on.

Beranek's presentation will be followed by a reception hosted by the AES Technical Council.

Session P13 — **Sunday, October 7**
8:30 am – 12:30 pm — **Room 1E07**

ACOUSTIC MODELING, PART 1

Chair: **Kurt Graffy**, Arup, San Francisco, CA, USA

8:30 am

P13-1 Addressing the Discrepancy Between Measured and Modeled Impulse Responses for Small Rooms —*Zhixin Chen, Robert Maher*, Montana State University, Bozeman, MT, USA

Simple computer modeling of impulse responses for small rectangular rooms is typically based on the image source method, which results in an impulse response with very high time resolution. Image source method is easy to implement, but simulated impulse responses are often a poor match to measured impulse responses because descriptions of sources, receivers, and room surfaces are often too idealized to match real measurement conditions. In this paper a more elaborate room impulse response computer modeling technique is developed by incorporating measured polar responses of loudspeaker, measured polar responses of microphone, and measured reflection coefficients of room surfaces into basic image source method. Results show that compared with basic image source method, the modeled room impulse response using this method is a better match to the measured room impulse response, as predicted by standard acoustical theories and principles.
Convention Paper 7239

9:00 am

P13-2 Comparison of Simulated and Measured HRTFs: FDTD Simulation Using MRI Head Data—*Parham Mokhtari, Hironori Takemoto, Ryouichi Nishimura, Hiroaki Kato*, NICT/ATR, Kyoto, Japan

This paper presents a comparison of computer-simulated versus acoustically measured, front-hemisphere head related transfer functions (HRTFs) of two human subjects. Simulations were carried out with a 3-D finite difference time domain (FDTD) method, using magnetic resonance imaging (MRI) data of each subject's head. A spectral distortion measure was used to quantify the similarity between pairs of HRTFs. Despite various causes of mismatch including a different head-to-source distance, the simulation results agreed considerably with the acoustic measurements, particularly in the major peaks and notches of the front ipsilateral HRTFs. Averaged over 133 source locations and both ears, mean spectral distortions for the two subjects were 4.7 dB and 3.8 dB respectively.
Convention Paper 7240

9:30 am

P13-3 Scattering Uniformity Measurements and First Reflection Analysis in a Large Nonanechoic Environment—*Lorenzo Rizzi*,[1] *Angelo Farina*,[2] *Paolo Galaverna*,[3] *Paolo Martignon,*[1] *Andrea Rosati,*[1] *Lorenzo Conti*[1]
[1]LAE – Laboratorio di Acustica ed Elettroacustica, Parma, Italy
[2]Università di Parma, Parma, Italy
[3]Genesis Acoustic Workshop, Parma, Italy

A new campaign of experiments was run on the floor of a large room to obtain a long enough anechoic time window. This permitted us to study the first reflection from the panels themselves and their diffusion uniformity. The results are discussed, comparing them with past measurements and with the ones from a simplified set-up with a smaller geometry. Some key matters to measurement are discussed; they were proposed in a recent comment letter posted to the specific AES-4id document committee on its reaffirmation. An analysis of the single reflection and reflectivity data was undertaken to investigate the behavior of a perforated panel and the measurement set-up overall potential.
Convention Paper 7241

10:00 am

P13-4 A Note On the Implementation of Directive Sources in Discrete Time-Domain Dispersive Meshes for Room Acoustic Simulation—*José Escolano*,[1] *José J. López*,[2] *Basilio Pueo*,[3] *Maximo Cobos*[2]
[1]University of Jaén, Jaén, Spain
[2]Technical University of Valencia, Valencia, Spain
[3]University of Alicante, Alicante, Spain

The use of wave methods to simulate room impulse responses provides the most accurate solutions. Recently, a method to incorporate directive sources in discrete-time methods, such as finite differences and digital waveguide mesh has been proposed. It is based in the proper combination of monopoles in order to achieve the desired directive pattern in far field conditions. However, this method is used without taking into account the inherent dispersion in ➥

most of these discrete-time paradigms. This paper analyzes how influent is the dispersion in order to get the proper directivity through different study cases.
Convention Paper 7242

10:30 am

P13-5 Rendering of Virtual Sound Sources with Arbitrary Directivity in Higher Order Ambisonics—*Jens Ahrens, Sascha Spors*, Technical University of Berlin, Berlin, Germany

Higher order Ambisonics (HOA) is a spatial audio reproduction technique aiming at physically synthesizing a desired sound field. It is based on the expansion of sound fields into orthogonal basis functions (spatial harmonics). In this paper we present an approach to the two-dimensional reproduction of virtual sound sources at arbitrary positions having arbitrary directivities. The approach is based on the description of the directional properties of a source by a set of circular harmonics. Consequences of truncation of the circular harmonics expansion and spatial sampling as occurring in typical installations of HOA systems due to the employment of a finite number of loudspeakers are discussed. We illustrate our descriptions with simulated reproduction results.
Convention Paper 7243

11:00 am

P13-6 The Ill-Conditioning Problem in Sound Field Reconstruction—*Filippo Fazi, Philip Nelson*, University of Southampton, Southampton, UK

A method for the analysis and reconstruction of a three dimensional sound field using an array of microphones and an array of loudspeakers is presented. The criterion used to process the microphone signals and obtain the loudspeaker signals is based on the minimization of the least-square error between the reconstructed and the original sound field. This approach requires the formulation of an inverse problem that can lead to unstable solutions due to the ill-conditioning of the propagation matrix. The concepts of generalized Fourier transform and singular value decomposition are introduced and applied to the solution of the inverse problem in order to obtain stable solutions and to provide a clear understanding of the regularization method.
Convention Paper 7244

11:30 am

P13-7 Analysis of Edge Boundary Conditions on Multiactuator Panels—*Basilio Pueo,*[1] *José Escolano,*[2] *José J. López,*[3] *Sergio Bleda*[1]
[1]University of Alicante, Alicante, Spain
[2]University of Jaén, Jaén, Spain
[3]Technical University of Valencia, Valencia, Spain

Distributed mode loudspeakers consist of a flat panel of a light and stiff material to which a mechanical exciter is attached, creating bending waves that are then radiated as sound fields. It can be used to build arrays for wave field synthesis reproduction by using multiple exciters in a single vibrating surface. The exciter interaction with the panel, the panel material, and the panel contour clamp conditions are some of the critical points that need to be evaluated and improved. In this paper we address the edge boundary conditions influence the quality of the emitted wave field. The measures of the wave fields have been interpreted in the wavenumber domain, where the source radiation is decomposed into plane waves for arbitrary angles of incidence. Results show how the wave field is degraded when the boundary conditions are modified.
Convention Paper 7245

12:00 noon

P13-8 Acoustics in Rock and Pop Music Halls—*Niels W. Adelman-Larsen,*[1] *Eric Thompson,*[2] *Anders C. Gade*[2]
[1]Flex Acoustics, Lyngby, Denmark
[2]Technical University of Denmark, Lyngby, Denmark

The existing body of literature regarding the acoustic design of concert halls has focused almost exclusively on classical music, although there are many more performances of rhythmic music, including rock and pop. Objective measurements were made of the acoustics of twenty rock music venues in Denmark and a questionnaire was used in a subjective assessment of those venues with professional rock musicians and sound engineers. Correlations between the objective and subjective results lead, among others, to a recommendation for reverberation time as a function of hall volume. Since the bass frequency sounds are typically highly amplified, they play an important role in the subjective ratings and the 63-Hz-band must be included in objective measurements and recommendations.
Convention Paper 7246

Session P14 — **Sunday, October 7**
9:00 am – 12:00 noon — **Room 1E16**

SIGNAL PROCESSING APPLIED TO MUSIC

Chair: **John Strawn**, S Systems, Larkspur, CA. USA

9:00 am

P14-1 Interactive Beat Tracking for Assisted Annotation of Percussive Music—*Michael Evans*, British Broadcasting Corporation, Tadworth, Surrey, UK

A practical, interactive beat-tracking algorithm for percussive music is described. Regularly-spaced note onsets are determined by energy-based analysis and users can then explore candidate beat periods and phases as the overall rhythm pattern develops throughout the track. This assisted approach can allow more flexible rhythmic analysis than purely automatic algorithms. An open-source software package based on the algorithm has been developed, along with several practical applications to allow more effective annotation, segmentation, and analysis of music.
Convention Paper 7247

9:30 am

P14-2 Identification of Partials in Polyphonic Mixtures Based on Temporal Envelope Similarity—*David Gunawan, D. Sen*, The University of New South Wales, Sydney, NSW, Australia

In musical instrument sound source separation, the temporal envelopes of the partials are correlated due to the physical constraints of the instruments. With this assumption, separation algorithms then exploit the similarities between the partial envelopes in order to group partials into sources. In this paper we quantitatively investigate the partial envelope similarities of a large database of instrument samples and develop weighting functions in order to model the similarities. These model partials then provide a reference to identify similar partials of the same source. The partial identification algorithm is evaluated in the separation of polyphonic mixtures and is shown to successfully discriminate between partials from different sources.
Convention Paper 7248

10:00 am

P14-3 Structural Decomposition of Recorded Vocal Performances and it's Application to Intelligent Audio Editing—*György Fazekas, Mark Sandler*, Queen Mary University of London, London, UK

In an intelligent editing environment, the semantic music structure can be used as beneficial assistance during the postproduction process. In this paper we propose a new approach to extract both low and high level hierarchical structure from vocal tracks of multitrack master recordings. Contrary to most segmentation methods for polyphonic audio, we utilize extra information available when analyzing a single audio track. A sequence of symbols is derived using a hierarchical decomposition method involving onset detection, pitch tracking, and timbre modeling to capture phonetic similarity. Results show that the applied model well captures similarity of short voice segments.
Convention Paper 7249

10:30 am

P14-4 Vibrato Experiments with Bassoon Sounds by Means of the Digital Pulse Forming Synthesis and Analysis Framework—
Michael Oehler,[1] *Christoph Reuter*[2]
[1]Institute for Music and Drama, Hanover, Germany
[2]University of Cologne, Cologne, Germany

The perceived naturalness of real and synthesized bassoon vibrato sounds is investigated in a listening test. The stimuli were generated by means of a currently developed synthesis and analysis framework for wind instrument sounds, based on the pulse forming theory. The framework allows controlling amplitude and frequency parameters at many different stages during the sound production process. Applying an ANOVA and Tukey HSD test it could be shown that timbre modulation (a combined pulse width and cycle duration modulation) is an important factor for the perceived naturalness of bassoon vibrato sounds. Obtained results may be useful for sound synthesis as well as in the field of timbre research.
Convention Paper 7250

11:00 am

P14-5 A High Level Musical Score Alignment Technique Based on Fuzzy Logic and DTW—*Bruno Gagnon, Roch Lefebvre, Charles-Antoine Brunet*, University of Sherbrooke, Sherbrooke, Quebec, Canada

This paper presents a method to align musical notes extracted from an audio signal with the notes of the musical score being played. Building on conventional alignment systems using Dynamic Time Warping (DTW), the proposed method uses fuzzy logic to create the similarity matrix used by DTW. Like a musician following a score, the fuzzy logic system uses high level information as its inputs, such as note identity, note duration, and local rhythm. Using high level information instead of frame by frame information reduces substantially the size of the DTW similarity matrix and thus reduces significantly the complexity to find the best path for alignment. Finally, the proposed method can automatically track where a musician starts and stops playing in a musical score.
Convention Paper 7251

11:30 am

P14-6 Audio Synthesis and Visualization with Flash CS3 and ActionScript 3.0—*Jordan Kolasinski*, New York University, New York, NY, USA

This paper explains the methods and techniques used to build a fully functional audio synthesizer and FFT-based audio visualizer within the newest version of Flash CS3. Audio synthesis and visualization have not been possible to achieve in previous versions of Flash, but two new elements of ActionScript 3.0—the Byte Array and Compute Spectrum function—make it possible even though it is not included in Flash's codebase. Since Flash is present on 99 percent of the world's computers, this opens many new opportunities for audio on the web.
Convention Paper 7252

Workshop 14 — Sunday, October 7
9:00 am – 11:00 am — Room 1E12/13

USER-CENTERED DESIGN OF CONTROLLERS FOR PRO AUDIO

Chair: **William Martens**, McGill University, Montreal, Quebec, Canada

Panelists: *Durand Begault*, NASA Ames Research Center, Mountainview, CA, USA
Jeremy Cooperstock, McGill University, Montreal, Quebec, Canada
Bob Ludwig, Gateway Mastering Studios, Portland, ME, USA
George Massenburg, GML, TN, USA

The design of controllers and/or work surfaces for music recording and mixing will be examined from the experienced user's perspective. This examination motivated by the belief that there may be better ways to organize controllers, ways that prioritize access to controls in a manner that is based upon what experienced users know about how they do their work. The list of workshop panelists, which includes representatives from live sound, postproduction, and mastering, will have a brainstorming session about new control layouts using tools designed to stimulate "thinking outside of the box." Also planned are breakout sessions, within which small groups will be lead to focus on particular applications and/or problems, with results that hopefully contribute some new and useful ideas.

Workshop 15 **Sunday, October 7**
9:00 am – 12:00 noon **Room 1E15**

FORENSIC AUDIO, VIDEO, AND VOICE IDENTIFICATION IN THE DIGITAL AGE

Chair: **Tom Owen**, Owl Investigations, Colonia, NJ, USA

Panelists: *Stuart Allen*, International Media Services, Plainfield, NJ, USA
Durand Begault, Charles Salter and Associates, San Francisco, CA, USA
Catalin Grigoras, Consultant, Romania
Garrett Husveth, Latent Technology Group, Denville, NJ, USA
Richard Sanders, University of Colorado, Denver, CO, USA
Greg Stutchman, Stutchman Forensic Laboratory, Napa, CA, USA

Digital recording has been embraced by both law enforcement and the criminals they seek to apprehend. The corporate side of business has embraced digital audio and video recording to the point where there is no longer any expectation of privacy. With the advent of security digital systems, webcams, phone cameras, and audio/video everywhere in one's daily life, it's no wonder the Forensic Examiner is called on to render an opinion to questions regarding the authenticity of recordings, editing of recordings, "Who said what" and "Does the evidence actually reflect the event as it actually occurred"?

Today's examiner must have an arsenal of tools at his/her disposal to be able to answer these questions to a reasonable degree of scientific certainty. This tutorial will discuss and provide tools of the trade, the realistic expectation of analysis (also known as The CSI Effect) famous cases (The Night Whisperer, John Gotti Jr.), and will include eight panel members from across the world who are internationally known for their expertise in these areas.

This seminar will also include before and after examples of forensic enhancement, digital editing analysis, encase data recovery of "erased files," digital voice identification, and many other facets of this area.

Also discussed will be employment opportunities in the Forensic Field.

Workshop 16 **Sunday, October 7**
9:00 am – 11:00 am **Room 1E10**

FIR OR IIR? THAT IS THE QUESTION!

Chair: **Tim Nind**, Harman/Becker Automotive Systems, Bridgend, UK

Panelists: *Nilo Casimiro*, Dirac
Earl Geddes
Tomlinson Holman, Audyssey Laboratories, Inc., Los Angeles, CA, USA
Stanley Lipshitz, University of Waterloo, Waterloo, Ontario, Canada
Ryan Mihelich, Harman
Sean Olive, Harman International Industries, Inc., Northridge, CA, USA
Jan Pederson, Lyngdorf Audio, Skive, Denmark

There is much debate about the best type of filters to use for equalization of loudspeakers in rooms and cars and a number of products marketed as providing an "ideal" result. This workshop seeks to explore the pros and cons of equalization strategies using IIR or FIR filters in such an application. It presents the engineering facts of both types, looks at the evidence to substantiate claims that absolute and relative phase is audible, and presents case studies for real world FIR and IIR implementations.

Broadcast Session 10 **Sunday, October 7**
9:00 am – 11:00 am **Room 1E08**

THE MACY'S THANKSGIVING DAY PARADE —AN AUDIO PRIMER

Moderator: **Joel Spector**, Music Playback Mixer, NBC, retired

Presenters: *Milton Delugg*, Freelance Music Director
Ed Greene, Freelance Senior Audio Mixer
Dick Maitland, Freelance Sound Effects Mixer
Bob Palladino, Commercial Integration Mixer, NBC

What's a parade without a little music, a few marching bands, some famous performers, and the ever-present clowns? They abound at the Macy*s Parade, a tradition reaching back to 1924! Today, NBC broadcasts the three-hour live event in HDTV and surround sound. We'll learn how a team of audio technicians based at three locations capture and transmit the excitement of "America's Parade" to millions each year. Technical expertise, musical savvy, back-up plans, and a little history will be discussed by Senior Audio Mixer Ed Greene, Music Director Milton Delugg, Music Playback Engineer Joel Spector, Sound Effects Engineer Dick Maitland, and Commercial Integration Mixer Bob Palladino. Portions of the 2006 broadcast will be shown at the session.

Live Sound Seminar 8 **Sunday, October 7**
9:00 am – 11:00 am **Room 1E09**

LIVIN' IN A MATERIAL WORLD: SOUND PRODUCTION FOR CORPORATE EVENTS

Chair: **Mac Kerr**

Panelists: *Bruce Cameron*, Sound Divas
Brad Mulligan, Milligan & Humphry Consulting Inc.
Bob Rendon, PRG Audio
Mike Walshe

A discussion of the special issues involved with corporate presentations, from the perspective of the production, the soundman, and the equipment supplier.

Standards Committee Meeting **Sunday, October 7**
9:00 am **Room 1E02**

Standards Committee Meeting SC-03-06 Digital Library and Archive Systems.

Historical Program
SURROUND SOUND: THE BEGINNING, 1925–1940
Sunday, October 7, 9:30 am – 11:30 am
Room 1E11

Presenter: **Robert Auld**

Robert Auld, Audio Engineer/Sound Designer and principal at Auldworks, NY-based recording facility will present a comprehensive assessment of the origin and early development of this still evolving medium. From the initial encouragement of famed conductor Leopold Stokowski to early work at Bell Labs and Disney Studios, the genesis of Surround Sound will be explored in a comprehensive multi-media presentation, which will include the first electrical recording of a symphony orchestra and surround sound excerpts from *Fantasia*.

Technical Committee Meeting **Sunday, October 7**
10:00 am **Room 1E05**

Technical Committee Meeting on Multichannel and Binaural Audio Technologies.

Exhibitor Seminar **Sunday, October 7**
10:30 am – 12:30 pm **Room 1E04**

RENKUS-HEINZ, INC.

Presenter: **Ralph Heinz**

Digital Audio Networking in Live Sound Applications

Industry experts will join Ralph Heinz, Senior Vice President of Renkus-Heinz, to explore the practical advantages of networked audio in live sound and the new possibilities that networking open up. Topics will include digital audio networking technologies, RHAON ("rayon"), the Renkus-Heinz Audio Operations Network, and other networking products. Audience Q & A.

Broadcast Session 11 **Sunday, October 7**
11:00 am – 1:00 pm **Room 1E10**

AUDIO FOR HDTV: DIALNORM

Chair: **Andy Butler**

Panelists: *Mike Babbitt*, Dolby Laboratories
Tim Carroll, Linear Acoustics
Robert Seidel, CBS

What is the magic number? Is it 27; is it 24; or maybe it's 31?

Dialnorm was designed to help consumers tailor their audio decoding to a variety of listening environments, but how does that work if content providers don't agree on how to set this vital parameter. Several major networks have picked different values for distribution to their affiliates, and there is certainly no consistency at the station level. Is it a hopeless muddle or simply another opportunity for lively debate? Bring your questions and join the discussion.

Master Class 5 **Sunday, October 7**
11:00 am – 12:30 pm **Room 1E12/13**

OLIVER ARCHUT/KLAUS HEYNE

Presenters: **Oliver Archut**, TAB-Funkenwerk, Gaylord, KS, USA
Klaus Heyne, German Masterworks, Portland, OR, USA

Vintage Microphone Mystique—
How Sex Appeal Trumps Specs

The current phenomenon of frequently using fifty-year old microphones in critical recording applications, from pop vocals to classical symphony orchestras, is unprecedented and seemingly illogical in a world of rapidly evolving recording technologies. Oliver and Klaus will examine why, subjectively and objectively, recording microphones seem to occupy this uniquely antiquarian role. They will also discuss how and why many of today's manufacturers yield to the demand and desire of the recording community for microphones from yesteryear by issuing products that emulate the past.

Oliver Archut possesses an encyclopedic knowledge of the technology of German vintage vacuum tube audio. He was a lead production engineer at Telefunken GmbH and later at TAB. This was a time when giants such as Telefunken and Siemens were literally throwing out the past, filling dumpsters with thousands of pages of engineering details, metallurgical formulae, schematics, and even complete working machinery as German industry transitioned to miniaturization and solid-state. Oliver began an avid vocation of rescuing and collecting whatever he could of the then obsolete, but today precious, technology—the essential knowledge that the great German audio alchemists had created for turning materials such as copper, glass, and nickel alloy into the sound of gold.

Klaus Heyne is one of the world's top authorities on the restoration and modification of classic vacuum tube microphones. His restorations have been used by many of pop music's most recognized voices. Klauss will join Oliver for a lively discussion and moderated Q&A.

Live Sound Seminar 9 **Sunday, October 7**
11:00 am – 1:00 pm **Room 1E09**

MUSICALS—FROM BROADWAY TO LAS VEGAS:
THE SIMILARITIES, DIFFERENCES,
AND CHALLENGES OF MOVING BETWEEN THE TWO

Chair: **Nevin Steinberg**, Acme Sound Partners

Panelists: *Jonathan Deans*
Sten Severson, Acme Sound Partners
TBA

Analyzing the issues and logistics of transplanting a musical production originating on Broadway or London's West End to Las Vegas and making it a "musical spectacular."

Student Event/Career Development
STUDIO SLAVE TO AUDIO PROFESSIONAL:
WORKSHOP ON INTERNSHIPS AND JOBS
Sunday, October 7, 11:00 am – 1:00 pm
Room 1E08

Moderator: **Gary Gottlieb**, Webster University, Webster Groves, MO, USA

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Panelists: *Matt Allen*, Blevins Audio, Nashville, TN, USA
Matt Brown, Yonas Media, San Francisco, CA, USA
John Krivit, New England Institute of Art, Brookline, MA, USA
Richard McIlvery, University of Southern California, Los Angeles, CA, USA

The gravest concern and deepest fear of many students pertains to their first professional position. Whether it is an internship or an entry level job, many students express nervousness regarding this leap from the relative calm of education to the vast unknown of the professional world. In this workshop a group of educators and students or graduates who have recently interned will discuss their views on the traits and characteristics most sought after by potential employers, and will share tips for success at the entry level of the audio profession. A significant portion of this workshop will be devoted to a question and answer period for students.

Technical Committee Meeting Sunday, October 7
11:00 am Room 1E05

Technical Committee Meeting on Coding of Audio Signals.

Standards Committee Meeting Sunday, October 7
11:00 am Room 1E02

Standards Committee Meeting SC-03-07 Audio Metadata.

Special Event
LUNCHTIME KEYNOTE:
BARRY BLESSER
Sunday, October 7, 11:30 am – 12:30 pm
Room 1E08

Introduction by Alex Case

The Art of Space: Audio Engineers as Aural Architects

Regardless of their technical focus, audio and sound engineers also function as aural architects who shape a listener's experience of the environment. Space and sound are inexorably linked because space changes our experience of sound, and sound changes our experience of space.

Aural architecture has its own language, which includes at least five dimensions of spatiality: aesthetic, symbolic, musical, navigational, and social. The language of spatiality exists in parallel with the language of physical acoustic and perceptual psychology. In addition, the experience of aural space need not be consistent with visual space.

While engineers try to answer questions using technical methods and tools, those answers depend on first having identified the relevant questions and assumptions. For example, although subjective preferences and perceptual experiences of spatial acoustics can sometimes be measured using scientific methods, the life style of sensory subcultures determines the cognitive and perceptual strategy of listeners, which is neither stable nor consistent. In any given cultural context, some spatiality dimensions may dominate others. This lack of consistency forces a sound engineer to also be a sonic artist and an aural architect.

As a fellow of the Audio Engineering Society, *Dr. Blesser* has been an active contributor for over 40 years. During that time, he served as its president, organized the first digital audio conference, and received the Silver, Bronze, Board of Governor, and publications awards. He has been a reviewer for the *Journal* for many decades, and he currently serves as its consulting technical editor. Blesser was one of the pioneers of the digital audio revolution. He invented the first commercial digital audio reverberation system, the EMT-250; he helped start Lexicon; he was the principle architect of Orban's Audicy digital audio workstation; he has published numerous papers on many aspects of digital audio; and he holds more than a dozen patents. Blesser received his S.B., S.M. and Ph. D. degrees in electrical engineering from M.I.T. in 1964, 1965, and 1969 respectively. After 9 years as an Associate Professor of Electrical Engineering and Computer Science at M.I.T., he has spent the last 30 years as a technical and management consultant. Last year, MIT Press published his book, *Spaces Speak, Are You Listening? Experiencing Aural Architecture*.

Student Event/Career Development
EDUCATION FAIR
Sunday, October 7, 11:30 am – 1:30 pm
Room 1E Foyer

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.

Workshop 17 Sunday, October 7
12:00 noon – 1:30 pm Room 1E11

EVALUATION OF SURROUND MAIN MIKINGS FOR CLASSICAL ORCHESTRA

Chair: **Mick Sawaguchi**, Pioneer Corporation

Panelists: *Akira Fukada*, NHK
Hideo Irimajiri, Mainichi Broadcasting Corporation
Toru Kamekawa, Tokyo National University of Fine Arts and Music, Tokyo, Japan
Masayuki Mimura, Yomiuri Telecasting Corporation
Hideaki Nishida, Asahi Broadcasting Corporation

There are many different setups for surround main microphones for classical music and orchestra. But it is very difficult to research and study them practically and academically under identical conditions and judge their performance. Consequently the AES Japan Surround Study Project has been organized and put into practice after one and a half years of preparation. It was organized around 10 broadcasters and 2 universities; 12 manufacturers supported by HBF provided financial support. There were 15 different combinations of main and ambience microphone setups that were recorded on 96 channels independently in Pro Tools HD at 24 bit / 96-kHz. The musical examples were performed by the Osaka Philharmonic Orchestra on September 24–-27, 2006.

In this workshop each individual setup will be played back. Participants will have the opportunity for feedback in a listening test environment, and the data will be collected for subjective evaluation.

This workshop will be presented twice during the convention.

Tutorial 10 **Sunday, October 7**
12:30 pm – 2:30 pm **Room 1E15**

AUDIO FOR GAMES: GAME DEVELOPMENT

Presenter: **Marc Schaefgen**, Midway Home Entertainment, San Diego, CA, USA

Audio for games has come along way from the simplistic bloops and bleeps in synch to virtual game of ping pong. The latest AAA titles have production values that rival Hollywood movies. Do you ever wonder how they do that? This tutorial takes you through the development cycle of the modern video game, not just content creation or production work but the start to finish of the whole process. You will see parallels to film/tv/animation style productions but focus on the uniqueness of game development and how a well-integrated audio team does their thing.

Session P15 **Sunday, October 7**
1:00 pm – 5:00 pm **Room 1E07**

ACOUSTIC MODELING, PART 2

Chair: **Geoff Martin**, Bang & Olufsen a/s, Struer, Denmark

1:00 pm

P15-1 Improvement of One-Dimensional Loudspeaker Models—*Juha Backman*, Nokia Corporation, Espoo, Finland, and Helsinki University of Technology, Espoo, Finland

Simple one-dimensional waveguide models of loudspeaker enclosures describe well enclosures with simple interior geometry, but their accuracy is limited if used with more complex internal structures. The paper compares the results from one-dimensional models to FEM models for some simplified enclosure geometries found in typical designs. Based on these results it is apparent that one-dimensional models need to be refined to take some three-dimensional aspects of the sound field in close proximity of drivers into account. Approximations matched to FEM solutions are presented for enclosure impedance as seen by the driver and for the end correction of ports, taking both edge rounding and distance to the back wall into account.
Convention Paper 7253

1:30 pm

P15-2 Simulating the Directivity Behavior of Loudspeakers with Crossover Filters—*Stefan Feistel,*[1] *Wolfgang Ahnert*,[1] *Charles Hughes*,[2] *Bruce Olson*[3]
[1]Ahnert Feistel Media Group, Berlin, Germany
[2]Excelsior Audio Design & Services, LLC, Gastonia, NC, USA
[3]Olson Sound Design, Brooklyn Park, MN, USA

In previous publications the description of loudspeakers was introduced based on high-resolution data, comprising most importantly of complex directivity data for individual drivers as well as of crossover filters. In this paper it is presented how this concept can be exploited to predict the directivity balloon of multi-way loudspeakers depending on the chosen crossover filters. Simple filter settings such as gain and delay and more complex IIR filters are utilized for loudspeaker measurements and simulations, results are compared and discussed. In addition advice is given how measurements should be made particularly regarding active and passive loudspeaker systems.
Convention Paper 7254

2:00 pm

P15-3 Intrinsic Membrane Friction and Onset of Chaos in an Electrodynamic Loudspeaker— *Danijel Djurek*,[1] *Ivan Djurek,*[2] *Antonio Petosic*[2]
[1]Alessandro Volta Applied Ceramics (AVAC), Zagreb, Croatia
[2]University of Zagreb, Zagreb, Croatia

Chaotic state observed in an electrodynamic loudspeaker results from a nonlinear equation of motion and is driven by an harmonic restoring term being assisted by intrinsic membrane friction. This friction is not the smooth function of displacements but the sum of local hysteretic surface fluctuations, which give rise to its high differentiability in displacements, being responsible for onset of Feigenbaum bifurcation cascades and chaos. When an external small perturbation of low differentiability is added to the friction, another type of chaotic state appears, and this state involves period-3 window evidenced for the first time in these experiments.
Convention Paper 7255

2:30 pm

P15-4 Damping of an Electrodynamic Loudspeaker by Air Viscosity and Turbulence—*Ivan Djurek,*[1] *Antonio Petosic*,[1] *Danijel Djurek*[2]
[1]University of Zagreb, Zagreb, Croatia
[2]Alessandro Volta Applied Ceramics (AVAC), Zagreb, Croatia

Damping of an electrodynamic loudspeaker has been studied with respect to air turbulence and viscosity. Both quantities were evaluated as a difference of damping friction measured in air and in an evacuated space. The viscous friction dominates for small driving currents (< 10 mA) and is masked by turbulence for currents extending up to 100 mA. Turbulence contribution was evaluated as a difference of air damping friction at 1.0 and 0.1 bars, and it was studied for selected driving frequencies. Hot wire anemometry has been adopted to meet requirements of convection study from the loudspeaker, and obtained spectra were compared to measured turbulence friction, in order to trace the perturbation of emitted signal by turbulent motion.
Convention Paper 7256

3:00 pm

P15-5 Energetic Sound Field Analysis of Stereo and Multichannel Loudspeaker Reproduction— *Juha Merimaa*, Creative Advanced Technology Center, Scotts Valley, CA, USA

Energetic sound field analysis has been previ-

ously applied to encoding the spatial properties of multichannel signals. This paper contributes to the understanding of how stereo or multichannel loudspeaker signals transform into energetic sound field quantities. Expressions for the active intensity, energy density, and energetic diffuseness estimate are derived as a function of signal magnitudes, cross-correlations, and loudspeaker directions. It is shown that the active intensity vector can be expressed in terms of the Gerzon velocity and energy vectors, and its direction can be related to the tangent law of amplitude panning. Furthermore, several cases are identified where the energetic analysis data may not adequately represent the spatial properties of the original signals.
Convention Paper 7257

3:30 pm

P15-6 A New Methodology for the Acoustic Design of Compression Driver Phase-Plugs with Concentric Annular Slots—*Mark Dodd*,[1,2] *Jack Oclee-Brown*[2,3]
[1]Celestion International Ltd., Ipswich, UK
[2]GP Acoustics (UK) Ltd., Maidstone, UK
[3]University of Southampton, UK

In compression drivers a large membrane is coupled to a small horn throat resulting in high efficiency. For this efficiency to be maintained to high frequencies the volume of the resulting cavity, between horn and membrane, must be kept small. Early workers devised a phase-plug to fill most of the cavity volume and connect membrane to horn throat with concentric annular channels of equal length to avoid destructive interference . Later work, representing the cavity as a flat disc, describes a method for calculating the positions and areas of these annular channels where they exit the cavity, giving least modal excitation, thus avoiding undesirable response irregularities. In this paper the result of applying both the equal path-length and modal approaches to a phase-plug with concentric annular channels coupled to a cavity shaped as a flat disc is further explored. The assumption that the cavity may be represented as a flat disc is investigated by comparing its behavior to that of an axially vibrating rigid spherical cap radiating into a curved cavity. It is demonstrated that channel arrangements derived for a flat disc are not optimum for use in a typical compression driver with a curved cavity. A new methodology for calculating the channel positions and areas giving least modal excitation is described. The impact of the new approach will be illustrated with a practical design.
Convention Paper 7258

4:00 pm

P15-7 A Computational Model for Optimizing Microphone Placement on Headset Mounted Arrays—*Philip Gillett, Marty Johnson, Jamie Carneal*, Virginia Tech Vibration and Acoustics Laboratories, Blacksburg, VA, USA

Microphone arrays mounted on headsets provide a platform for performing transparent hearing, source localization, focused listening, and enhanced communications while passively protecting the hearing of the wearer. However it is not trivial to determine the microphone positions that optimize these capabilities, as no analytical solution exists to model acoustical diffraction around both the human and headset. As an alternative to an iterative experimental approach for optimization, an equivalent source model of the human torso, head, and headset is developed. Results show that the model closely matches the microphone responses measured from a headset placed on a Kemar mannequin in an anechoic environment.
Convention Paper 7259

4:30 pm

P15-8 A Simple Simulation of Acoustic Radiation from a Vibrating Object—*Cynthia Bruyns Maxwell*, University of California at Berkeley, Berkeley, CA, USA

The goal of this paper is to explore the role that fluid coupling plays on the vibration of an object, and to investigate how one can model such effects. We want to determine whether the effects of coupling to the medium surrounding a vibrating object are significant enough to warrant including them into our current instrument modeling software. For example, we wish to examine how the resonant frequencies of an object change due to the presence of a surrounding medium. We also want to examine the different methods of modeling acoustic radiation in interior and exterior domains. Using a simple 2-D beam as an example, this investigation shows that coupling with dense fluids, such as water, dramatically changes the resonant frequencies of the system. We also show that using a simple finite element model and modal analysis, we can simulate the acoustic radiation profile and determine a realistic sound pressure level at arbitrary points in the domain in real-time.
Convention Paper 7260

Session P16 **Sunday, October 7**
1:00 pm – 4:00 pm **Room 1E16**

SIGNAL PROCESSING FOR ROOM CORRECTION

Chair: **Rhonda Wilson**, Meridian Audio, UK

1:00 pm

P16-1 Sampling the Energy in a 3-D Sound Field—*Jan Abildgaard Pedersen*, Lyngdorf Audio, Skive, Denmark

The energy in the 3-D sound field in a room holds crucial information needed when designing a room correction system. This paper shows how measured sound pressure in at least 4 randomly selected positions scattered across the entire listening room is a robust estimate of the energy in the 3-D sound field. The reproducibility was investigated for a different number of random positions, which lead to an assessment of the robustness of a room correction system based on different number of random microphone positions.
Convention Paper 7261

1:30 pm

P16-2 Multi-Source Room Equalization: Reducing Room Resonances—*John Vanderkooy*, University of Waterloo, Waterloo, Ontario, Canada

Room equalization traditionally has been implemented as a single correction filter applied to all the channels in the audio system. Having more sources reproducing the same monophonic low-frequency signal in a room has the benefit of not exciting certain room modes, but it does not remove other strong room resonances. This paper explores the concept of using some of the loudspeakers as sources, while others are effectively sinks of acoustic energy, so that as acoustic signals cross the listening area, they flow preferentially from sources to sinks. This approach resists the buildup of room resonances, so that modal peaks and antimodal dips are reduced in level, leaving a more uniform low-frequency response. Impulse responses in several real rooms were measured with a number of loudspeaker positions and a small collection of observer positions. These were used to study the effect of source and sink assignment, and the derivation of an appropriate signal delay and response to optimize the room behaviour. Particular studies are made of a common 5.0 loudspeaker setup, and some stereo configurations with two or more standard subwoofers. A measurable room parameter is defined that quantifies the deleterious effects of low-frequency room resonances, supported by a specific room equalization philosophy. Results are encouraging but not striking. Signal modification needs to be considered.
Convention Paper 7262

2:00 pm

P16-3 A Low Complexity Perceptually Tuned Room Correction System—*James Johnston, Serge Smirnov*, Microsoft Corporation, Redmond, WA, USA

In many listening situations using loudspeakers, the actualities of room arrangements and the acoustics of the listening space combine to create a situation where the audio signal is unsatisfactorily rendered from the listener's position. This is often true not only for computer-monitor situations, but also for home theater or surround-sound situations in which some loudspeakers may be too close to too far from the listener, in which some loudspeakers (center, surrounds) may be different than the main loudspeakers, or in which room peculiarities introduce problems in imaging or timbre coloration. In this paper we explain a room-correction algorithm that restores imaging characteristics, equalizes the first-attack frequency response of the loudspeakers, and substantially improves the listeners' experience by using relatively simple render-side DSP in combination with a sophisticated room analysis engine that is expressly designed to capture room characteristics that are important for stereo imaging and timbre correction.
Convention Paper 7263

2:30 pm

P16-4 Variable-Octave Complex Smoothing—*Sunil Bharitkar*, Audyssey Labs, Los Angeles, CA, USA, and University of Southern California, Los Angeles, CA, USA

In this paper we present a technique for processing room responses using a variable-octave complex-domain (viz., time-domain) smoother. Traditional techniques for room response processing, for equalization and other applications such as auralization, have focused on a constant-octave (e.g., 1/3 octave) and with magnitude domain smoothing of these room responses. However, recent research has shown that room responses need to be processed with a high resolution especially in the low-frequency region to characterize the discrete room modal structure as these are distinctly audible. Coupled this with the need for reducing the computational requirements associated with filters obtained from undesirable over-fitting the high-frequency part of the room response with such a high-Q complex-domain smoother, and knowledge of the fact that the auditory filters have wider bandwidth (viz., lower resolution) in the high-frequency part of the human hearing, the present paper proposes a variable-octave complex-domain smoothing. Thus this paper incorporates, simultaneously, the high low-frequency resolution requirement as well as the requirement of relatively lower-resolution fitting of the room response in the high-frequency part through a perceptually motivated approach.
Convention Paper 7264

3:00 pm

P16-5 Multichannel Inverse Filtering With Minimal-Phase Regularization—*Scott Norcross*,[1] *Martin Bouchard*[2]
[1]Communications Research Centre, Ottawa, Ontario, Canada
[2]University of Ottawa, Ottawa, Ontario, Canada

Inverse filtering methods are used in numerous audio applications such as loudspeaker and room correction. Regularization is commonly used to limit the amount of the original response that the inverse filter attempts to correct in an effort to reduce audible artifacts. It has been shown that the amount and type of regularization used in the inversion process must be carefully chosen so that it does not add additional artifacts that can degrade the audio signal. A method of designing a target function based on the regularization magnitude was introduced by the authors, where a minimal-phase target function could be used to reduce any pre-response caused by the regularization. In the current paper a multichannel inverse filtering scheme is introduced and explored where the phase of the regularization itself can be chosen to reduce the audibility of the added regularization. In the single-channel case, this approach is shown to be equivalent to the technique that was previously introduced by the authors.
Convention Paper 7265

3:30 pm

P16-6 An In-flight Low-Latency Acoustic Feedback Cancellation Algorithm—*Nermin Osmanovic*,[1] *Victor E. Clarke*,[2] *Erich Velandia*[2]

[1]Consultant, Seattle, WA, USA
[2]Gables Engineering, Coral Gables, FL, USA

Acoustic feedback is a common problem in high gain systems; it is very unpredictable and unpleasant to the ear. Cockpit communication systems on aircraft may suffer from acoustic feedback between a pilot's boomset microphone and high gain cockpit loudspeaker. The acoustic feedback tone can compromise flight safety by temporarily blocking communication between the pilot and ground control. This paper presents the design of an in-flight low latency (<6 ms) digital audio processing system that automatically detects and removes acoustic feedback tones from the microphone to loudspeaker audio path. We present information about the acoustic feedback cancellation algorithm including the calculation of feedback existence probability, as implemented in an aircraft cockpit communication system.
Convention Paper 7266

Broadcast Session 12 **Sunday, October 7**
1:00 pm – 2:00 pm **Room 1E10**

BROADCAST TUTORIAL: HOW TO WORK WITH TELECOM TO GET THE JOB DONE

Presenters: **Bruce Berensen**, XM Radio
Angela DePascale, Global Digital Datacom Services Inc.

How do you order the right circuit? What do you do to get problems resolved?

Master Class 6 **Sunday, October 7**
1:00 pm – 2:30 pm **Room 1E08**

KEN HAHN

Presenter: **Ken Hahn**, Sync Sound, New York, NY, USA

Advanced Ideas and Techniques in High-Definition Surround Music Postproduction for Film and Television

Ken Hahn is co-owner of Sync Sound®, Inc., a New York City audio postproduction facility, which he founded with partner Bill Marino in 1984. He has more than 30 years of experience mixing and editing audio and has received five Emmy Awards, three Cinema Audio Society's Outstanding Achievement in Sound Mixing Awards, *Mix Magazine*'s Tec Award for Best Audio Post Production Mixer, and has engineered several Grammy Award-winning albums. Recent credits include Britney Spears, The Dixie Chicks, Janet Jackson, Garth Brooks, Gypsy Kings, Billy Joel, Reba McEntire, The Eagles, Billy Joel, Carly Simon, Cyndi Lauper, Roger Daltrey, Tony Bennett, and Michael Jackson.

Please join him for a program that will cover such topics as: How to achieve 5.1/stereo/mono compatible mixes; the reasons why everyone should work at a fixed monitoring level; how to achieve a mix that both you and your client are satisfied with; audio postproduction "tools of the trade"; dealing with delivery specs and how they dictate workflow; is louder really better? Along with many other topics…

Special Event
PLATINUM MASTERING
Sunday, October 7, 1:00 pm – 2:30 pm
Room 1E12/13

Moderator: **Bob Ludwig**, Grammy award winning president of Gateway Mastering & DVD, Portland, ME, USA

Panelists: *Brad Blackwood*, Euphonic Mastering, Arlington, TN, USA
Greg Calbi, Sterling Sound, NY, USA
Gavin Lurssen, Lurssen Mastering, Hollywood, CA, USA
Paul Stubblebine, Paul Stubblebiine Mastering, San Francisco, CA, USA
Tim Young, Metropolis Mastering, London, UK

Level Wars and Other Mastering Issues in the Context of the New Record-Business Paradigm

Multi platinum and Grammy-winning mastering engineer Bob Ludwig of Gateway Mastering & DVD (Rolling Stones, Bruce Springsteen, The Police, Nirvana, Rush), will moderate an all star panel featuring Paul Stubblebine (Grateful Dead, Sly Stone, Journey, Santana); Greg Calbi of Sterling Sound (Norah Jones, Bowie, John Lennon, Paul Simon's *Graceland*); Europe's Tim Young (The Clash, The Smiths, Bjork, Van Morrison); Gavin Lurssen, Lurssen Mastering (Lucinda Williams, Matchbox 20, Alison Krauss and the *Knocked Up* soundtrack); and Brad Blackwood of Euphonic Mastering (Fuel, Sister Hazel, Lucero, Nine Days).

Mastering hot levels has been with us since the LP and 45 singles. It has gotten out of hand for CDs and downloads. We will discuss the issue from traditional and unorthodox perspectives and cover other topics that impact on the art and business of mastering. In addition to the opportunity to trade ideas, tips, and war stories with our peers, we look forward to a lively Q&A with the audience.

Student Event/Career Development
INSIDE THE JOB INTERVIEW
Sunday, October 7, 1:00 pm – 2:30 pm
Room 1E06

Moderator: **John Strawn**, S Systems, Inc., Larkspur, CA, USA

Panelists: *Tony Agnello*, Eventide, Little Ferry, NJ, USA
Mark Brunner, Shure, Niles, IL, USA
Mark Gilbert, Shure, Niles, IL, USA

This panel discussion is aimed at job candidates in electrical engineering and computer science who want to learn more about the job interview process. Recent graduates, juniors, seniors, and graduate students who are now seeking, or will soon be seeking, a full-time employment position in the audio and music industries in hardware or software engineering should benefit from attending. The panel consists of several in-house staff involved in recruiting for engineering research/development positions at audio/music companies. Two companies are large and well-established; smaller companies are also represented. The panel is moderated by a recruiter, who will also offer one recruiter's perspective. Each panel member will briefly describe their company's typical interviewing procedure including what questions to expect. Questions from the floor are encouraged. Although the focus is on the interview itself, other questions relating to job search and career development are welcome. If you've never had a job interview in industry, or only had a few, this is meant for you.

Exhibitor Seminar **Sunday, October 7**
1:00 pm – 2:00 pm **Room 1E04**

MINNETONKA

Presenters: **Steve Clarke, John Schur, Jim Weber**

AudioTools WorkFlow Architecture

Managing content efficiently and effectively is difficult but essential in every media organization. The challenge centers on managing digital audio assets through an increasingly complex pipeline of editing, processing, versioning, collaboration, distribution, and archiving. The AudioTools workflow architecture addresses this complex set of problems to create greater efficiencies and productivity.

Technical Committee Meeting **Sunday, October 7**
1:00 pm **Room 1E05**

Meeting of the Advisory Group on Regulations.

Standards Committee Meeting **Sunday, October 7**
1:00 pm **Room 1E02**

Standards Committee Meeting SC-03-12 Forensic Audio.

Broadcast Session 13 **Sunday, October 7**
2:00 pm – 4:00 pm **Room 1E10**

AUDIO PROCESSING FOR HD (RADIO AND TV)

Chair: **Glynn Walden**, CBS

Presenters: *Tim Carroll*, Linear Acoustics
Mike Dorrough, Dorrough Electronics
Frank Foti, Omnia
Thomas Lund, TC Electronics
Steve Lyman, Dolby Laboratories
David Reaves, Translantech

The broadcasting industry, like all other media, is in a rapid transition to digital with television facing an eminent turn-off of analog and radio having a longer period before turning off its analog audio.

Both radio and television origination and production are well into the transition with audio processing questions being left open of how to deal with the greater dynamic range in their transmission systems. The additional dynamic range has exaggerated the issues of greater artistic freedom vs. annoying changes in loudness. This panel will address the issues of processing in a digital broadcast world.

Live Sound Seminar 10 **Sunday, October 7**
2:00 pm – 4:00 pm **Room 1E09**

PA FOR TV: SOUND SYSTEM DESIGN AND MIXING FOR A LIVE AUDIENCE WHILE SATISFYING PRODUCTION DEMANDS

Chair: **Bob Rendon**, PRG Audio

Panelists: *Daryl Bornstein*
Bill Daly, Wheel of Fortune, Jeopardy
Dan Gerhard, Comedy Central, HBO (TONY Awards)
Jimmy Hores, Rachel Ray, Tony Danza
Mikael Stewart, ATK Audiotek

Many televised events are live and/or have a live audience. The sound system designer and FOH engineer must balance what the live audience is hearing with the needs of the production to get a clean recording. These designers and engineers will talk about how they have achieved that balance.

Historical Program
SURROUND SOUND: QUADRAPHONIC SOUND IN THE 1970S—RECORDING CLASSICAL MUSIC
Sunday, October 7, 2:00 pm – 4:00 pm
Room 1E11

Moderator: **Robert Auld**

Panelist: *Andrew Kazdin*, Former Producer Columbia Masterworks
Charles Repka, Former Engineer Vanguard Records
Max Wilcox, Former Producer RCA Red Seal

Starting in 1969, the record industry tried to introduce “the next big thing”—four-channel or “quadraphonic” sound for the home. The result was mass confusion as two tape formats and no less than five disc formats competed in the marketplace. In the middle of all this, producers and engineers did what they have always done—made the best recordings they knew how to make for present and future playback systems.

Many of the veterans of the “quad wars” are still with us. This presentation will feature a distinguished panel of producers and engineers who will play excerpts from quad recordings of classical music that they helped make, and discuss the rise and fall of quadraphonic sound.

Technical Committee Meeting **Sunday, October 7**
2:00 pm **Room 1E05**

Technical Committee Meeting on Perception and Subjective Evaluation of Audio.

Session P17 **Sunday, October 7**
2:30 pm – 4:00 pm **Foyer 1E**

POSTERS: SIGNAL PROCESSING APPLIED TO MUSIC

2:30 pm

P17-1 Toward Textual Annotation of Rhythmic Style in Electronic Dance Music—*Kurt Jacobson, Matthew Davies, Mark Sandler*, Queen Mary University of London, London, UK

Music information retrieval encompasses a complex and diverse set of problems. Some recent work has focused on automatic textual annotation of audio data, paralleling work in image retrieval. Here we take a narrower approach to the automatic textual annotation of music signals and focus on rhythmic style. Training data for rhythmic styles are derived from simple, precisely labeled drum loops intended for content creation. These loops are already textually annotated with the rhythmic style they represent. The training loops are then compared against a database of music content to apply textual annotations of rhythmic style to unheard music signals. Three distinct methods of rhythmic analysis are explored. These methods are tested on a small collection of electronic dance music resulting in a labeling accuracy of 73 percent.
Convention Paper 7268

➥

2:30 pm

P17-2 Key-Independent Classification of Harmonic Change in Musical Audio—*Ernest Li, Juan Pablo Bello*, New York University, New York, NY, USA

We introduce a novel method for describing the harmonic development of a musical signal by using only low-level audio features. Our approach uses Euclidean and phase distances in a tonal centroid space. Both measurements are taken between successive chroma partitions of a harmonically segmented signal, for each of three harmonic circles representing fifths, major thirds, and minor thirds. The resulting feature vector can be used to quantify a string of successive chord changes according to changes in chord quality and movement of the chordal root. We demonstrate that our feature set can provide both unique classification and accurate identification of harmonic changes, while resisting variations in orchestration and key.
Convention Paper 7269

2:30 pm

P17-3 Automatic Bar Line Segmentation—*Mikel Gainza, Dan Barry, Eugene Coyle*, Dublin Institute of Technology, Dublin, Ireland

A method that segments the audio according to the position of the bar lines is presented. The method detects musical bars that frequently repeat in different parts of a musical piece by using an audio similarity matrix. The position of each bar line is predicted by using prior information about the position of previous bar lines as well as the estimated bar length. The bar line segmentation method does not depend on the presence of percussive instruments to calculate the bar length. In addition, the alignment of the bars allows moderate tempo deviations
Convention Paper 7270

2:30 pm

P17-4 The Analysis and Determination of the Tuning System in Audio Musical Signals—
Peyman Heydarian, Lewis Jones, Allan Seago, London Metropolitan University, London, UK

The tuning system is an essential aspect of a musical piece. It specifies the scale intervals and contributes to the emotions of a song. There is a direct relationship between the musical mode and the tuning of a piece for modal musical traditions. In a broader sense it represents the different genres. In this paper algorithms based on spectral and chroma averages are developed to construct patterns from audio musical files. Then a similarity measure like the Manhattan distance or the cross-correlation determines the similarity of a piece to each tuning class. The tuning system provides valuable information about a piece and is worth incorporating into the metadata of a musical file.
Convention Paper 7271

Workshop 18 **Sunday, October 7**
2:30 pm – 4:30 pm **Room 1E15**

AUDIO FOR GAMES: DIALOG RECORDING, WORKFLOW, AND ASSET MANAGEMENT

Chair: **Steve Martz**, THX

Panelist: *Jory Prum*, studio.jory.org, Fairfax, CA, USA

It is not unheard-of to have 9,000 lines of dialog (or other audio assets) in a game. Understanding how to prepare, organize, and store these assets highlights a principle difference between game audio and other media. This workshop takes a step-by-step look at the monumental task of the recording, editing, and organization process, as well as strategies to keep it under control. Methods will be presented for script management, session set-up, session flow, actor interfacing, and asset organization and protection, as well as a comparison to similar techniques in other types of media production.

Tutorial 11 **Sunday, October 7**
2:30 pm – 5:00 pm **Room 1E08**

WHAT IS THE MATRIX?

Presenters: **Geoff Martin**, Bang and Olufson a/s, Struer, Denmark
Helmut Wittek, Schoeps Mikrofone, Karlsruhe, Germany

This tutorial will be an introduction to the use of matrixing in microphone and mixing techniques for stereo and multichannel. The basics of microphone polar patterns will be explained, followed by the fundamentals of techniques such as "textbook" M/S and soundfield recording. Included in the tutorial will be a discussion of how to use matrixing to "re-mix" an existing recording, to modify microphone configurations in postproduction, and to manipulate spatial characteristics of stereo mixes. In addition, information on the exciting possibilities in the fast-paced world of karaoke soundtrack production will be presented.

Exhibitor Seminar **Sunday, October 7**
2:30 pm – 4:30 pm **Room 1E04**

RENKUS-HEINZ, INC.

Presenter: **Jim Mobley**

IONYX Digitally Steerable Arrays

ICONYX systems meet architectural and acoustical challenges everywhere from houses of worship to cruise ships, casino lounges and corporate meetings. This seminar presents the latest developments in ICONYX technology, integration into RHAON, the Renkus-Heinz audio operations network, and Beamware 3 for 3-D array optimization.

Standards Committee Meeting **Sunday, October 7**
2:30 pm **Room 1E02**

Standards Committee Meeting SC-02-12 Audio Applications of IEEE 1394.

Special Event
PLATINUM ENGINEERS
Sunday, October 7, 3:00 pm – 4:30 pm
Room 1E12/13

Moderator: **Paul Verna**

Panelists: *Bob Clearmountain*
Young Guru
John Horesco
Kevin Killen
Tony Maserati
Steve Rosenthal
Nick Sansano
Ralph Sutton

The AES Platinum Engineers panel will feature a mix of veteran and up-and-coming engineers and mixers who have worked with some of the top artists in the business. Confirmed panelists include Bob Clearmountain, Kevin Killen, Tony Maserati, Ralph Sutton, Nick Sansano, John Horesco, Steve Rosenthal, and Young Guru, all of whom will share their insights on the recording process and their tips on how to navigate the challenges of today's recording industry. The panel will be moderated by pro audio author and journalist Paul Verna, a veteran of *Billboard* and *Mix* magazines and co-author of *The Encyclopedia of Record Producers*.

Student Event/Career Development
AUDIO EDUCATION FORUM
Sunday, October 7, 3:00 pm – 5:00 pm
Room 1E06

Moderator: **Mark Parsons**, Parsons Audio & Center for Audio Studies, Wellesley, MA, USA

Panelists: *Jim Anderson*, NYU Clive Davis Department of Recorded Music, New York, NY, USA
Eddy B. Brixen, EBB-consult, Smorum, Denmark
Alex Case, University of Massachusetts, Lowell, MA, USA
Paul Foeckler, Digidesign, Daly City, CA, USA
Leslie Ann Jones, Skywalker Sound (Lucasfilm), CA, USA; SPARS President
Dave Moulton, Sausalito Audio Works, Groton, MA, USA

The Job in Your Future:
What Does it Take to Be Employable?

How—and how well—is audio education preparing prospective professionals for the marketplace? This session will encompass:

- What does it take to be employable?
- What skills need more emphasis? Entrepreneurship, business savvy, etc.
- Future-proofing: what skills can be taught that will be applicable for years to come?
- How to get the most for your education dollar?
- What mistakes students should avoid?

Moderator: Mark Parsons: Parsons Audio & Center for Audio Studies (Wellesley, MA).

Panelists: Jim Anderson, Chairman of the NYU Clive Davis Department of Recorded Music, and a recording engineer whose work has won many Grammy Awards; Eddy B. Brixen: leading audio consultant with Denmark's EEB; Alex Case, U. of Mass. Professor; Paul Foeckler: Digidesign's Vice President of Sales & Marketing and a principal in Digidesign's education program; Dave Moulton: Sausalito Audio Works, a Grammy-nominated engineer, producer, and a committed educator.

Technical Committee Meeting — Sunday, October 7
3:00 pm — Room 1E05

Technical Committee Meeting on Network Audio Systems.

Session P18 — Sunday, October 7
4:00 pm – 6:00 pm — Room 1E16

AUDIO FORENSICS

Chair: **Durand R. Begault**, Charles M. Salter Associates, San Francisco, CA, USA

4:00 pm

P18-1 Experiment in Computational Voice Elimination Using Formant Analysis—*Durand R. Begault*, Charles M. Salter Associates, San Francisco, CA, USA

This paper explores the use of a computational approach to the elimination of a known from an unknown voice exemplar in a forensic voice elimination protocol. A subset of voice exemplars from 11 talkers, taken from the TIMIT database, were analyzed using a formant tracking program. Intra- versus inter-speaker mean formant frequencies are analyzed and compared.
Convention Paper 7272

4:30 pm

P18-2 Applications of ENF Analysis Method in Forensic Authentication of Digital Audio and Video Recordings—*Catalin Grigoras*, Forensic Examiner, Bucharest, Romania

This paper reports on the electric network frequency (ENF) method as a means of assessing the integrity of digital audio/video evidence analysis. A brief description is given to different ENF types and phenomena that determine ENF variations, analysis methods, stability over different geographical locations on continental Europe, interlaboratory validation tests, uncertainty of measurement, real case investigations, different compression algorithm effects on ENF values and possible problems to be encountered during forensic examinations. By applying the ENF Method in forensic audio/video analysis, one can determine whether and where a digital recording has been edited, establish whether it was made at the time claimed, and identify the time and date of the registering operation.
Convention Paper 7273

5:00 pm

P18-3 Quantifying the Speaking Voice: Generating a Speaker Code as a Means of Speaker Identification Using a Simple Code-Matching Technique—*Peter S. Popolo*,[1,2] *Richard W. Sanders*,[1,3] *Ingo R. Titze*[2,3]
[1]National Center for Voice and Speech, Denver, CO, USA
[2]University of Iowa, Iowa City, IA, USA
[3]University of Colorado at Denver, Denver, CO, USA

This paper looks at a methodology of quantifying the speaking voice, by which temporal and spectral features of the voice are extracted and processed to create a numeric code that identifies speakers, so those speakers can be searched in a database much like fingerprints. The parameters studied include: (1) average fundamental frequency (F0) of the speech signal over time, (2) standard deviation of the F0, (3) the slope and (4) sign of the FO contour, (5) the average energy, (6) the standard deviation of the energy, (7) the spectral energy contained from 50 Hz to 1,000 Hz, (8) the spectral energy from 1,000 Hz to 5,000 Hz, (9) the Alpha Ratio, (10) the average speaking rate, and (11) the total duration of the spoken sentence.
Convention Paper 7274

5:30 pm

P18-4 Further Investigations into the ENF Criterion for Forensic Authentication—*Eddy Brixen*, EBB-consult, Smørum, Denmark

In forensic audio one important task is the authentication of audio recordings. In the field of digital audio and digital media one single complete methodology has not been demonstrated yet. However, the ENF (Electric Network Frequency) Criterion has shown promising results and should be regarded as a major tool in that respect. By tracing the electric network frequency in the recorded signal a unique timestamp is provided. This paper analyses a number of situations with the purpose to provide further information for the assessment of this methodology. The topics are: ways to establish reference data, spectral contents of the electromagnetic fields, low bit rate codecs' treatment of low level hum components, and tracing ENF harmonic components.
Convention Paper 7275

Student Event/Career Development
RECORDING COMPETITION
SURROUND
Sunday, October 7, 4:00 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. Student members can submit stereo and surround recordings in the categories classical, jazz, folk/world music, and pop/rock. Meritorious awards will be presented at the closing Student Delegate Assembly Meeting on Monday.

Judges include: Martha De Francisco, David Frost, Richard King, Kimio Hamasaki, Ronald Prent, Andreas Mayo, Jim Anderson, Peter Cook, Tim Martyn, Rebecca Pulliam, Bob Katz, Michael Nunan, and Ronald Sadoff.

Sponsors for this event include: AEA, Yamaha, Genelec, Harmon/JBL, Sony Creative Software, Neumann, Shure, Lavry Engineering, Schoeps, Millennia Music and Media Systems, PCM, and Sennheiser.

Technical Committee Meeting **Sunday, October 7**
4:00 pm **Room 1E05**

Technical Committee Meeting on Transmission and Broadcasting.

Workshop 19 **Sunday, October 7**
4:30 pm – 7:30 pm **Room 1E09**

UNEVEN BASS REPRODUCTION IN AUTOMOBILES

Chair: **Tom Nousaine**, Listening Technology

Panelists: *Natan Budiono*, Sr. Acoustical Engineer, Panasonic
David Carlstrom, Audio Specialist, Chrysler
David Clark, Director of Research, Alpine-America
Kenneth Deetz, Design Engineer, Delphi
Daniel Field, Chief Engineer, Harman/Becker Automotive Systems
Robert Klacza, Jeep Speaker and Audio Engineer, Chrysler
Richard Stroud, Stroud Audio Inc., Kokomo, IN, USA

In the evaluation of 750 OEM Autosound systems Tom Nousaine, Listening Technology, has found that a majority of systems suffer from spectral uniformity problems at low frequencies. This simply means that bass sounds are uneven. For example on a program with an acoustic bass solo, some notes almost disappear while others may seem unusually loud. It is commonly felt that this is a simple equalization issue. Thus, it is puzzling that this problem continues to exist when modern car electronics have significant active sound control capabilities. The panel will discuss the issue, causes, and possible solutions.

Workshop 20 **Sunday, October 7**
4:30 pm – 6:30 pm **Room 1E15**

GAME AUDIO FOR BROADBAND PHONES

Chair: **Peter Drescher**, Danger, Inc.

Broadband data connections for cell phones will change everything. This session examines how massive portable storage and speedy data transfer rates will affect interactive audio for mobile devices in general, and mobile game soundtracks in particular. The author presents an overview of the current state of the art, then discusses new cell phone features that are about to come to market. He concludes by making predictions about the future of mobile game audio, based on experience, and supported by analogies drawn from the development of PC/console games and the Internet.

Broadcast Session 14 **Sunday, October 7**
4:30 pm – 6:30 pm **Room 1E08**

AUDIO PLAYBACK AND AUTOMATION FOR RADIO

Chair: **Skip Pizzi**, Microsoft Corp.

Presenters: *Don Backus*, ENCO Systems
Neil Glassman, Broadcast Electronics
Paul Weiland, DAVID Systems

Most radio stations now use some form of PC-based control and/or audio origination, but this audio technology niche continues to evolve. Hear about the latest developments and what's coming next from some of the leading purveyors in the radio automation space at this informative session.

Tutorial 12 **Sunday, October 7**
4:30 pm – 7:00 pm **Room 1E10**

HIGH DEFINITION PERCEPTION—PSYCHOACOUSTICS, PHYSIOLOGY, AND THE NUMBERS RACE

Presenter: **Poppy Crum**, Johns Hopkins University School of Medicine, Baltimore, MD, USA

Improvements in digital technology have led to higher fidelity recordings by some definitions, but they have also generated much controversy. Sampling rates and bit depths well beyond the limits of conventionally accepted human perception can be used, but are they necessary? This tutorial addresses psychophysical and physiological research of hearing beyond the known limits of audible frequency and dynamic range. In addition to studies of human perception and physiology, we will look at species such as frogs, bats, and hummingbirds known for their development of unique sensitivity to properties of sounds that are crucial in their navigation and communication. Nonstandard mechanisms of hearing transduction will be reviewed across species as an ecological approach to hearing development is considered in the ongoing high-definition debate.

Historical Program
CHAOS BEGETS ORDER:
LIVE SOUND BECOMES AN INDUSTRY
Sunday, October 7, 4:30 pm – 7:30 pm
Room 1E12/13

Moderators: **John Chester**, formerly Chief Sound Engineer Fillmore East
Roy Clair, Clair Brothers Audio
Dinky Dawson, Dawson Sound
Bill Hanley, Hanley Sound

Once upon a time music wasn't loud (or even electric!). There were no large touring sound systems. Then came the 1960s, and sound systems struggled to keep up. Systems were touring—but they weren't designed for the road. Portable mixers existed, but not portable consoles. Multi-kilowatt amplifiers were unheard of. Somehow, lots of good music was still made. Sound engineers created, improvised, and invented—and laid the foundations for today's live sound industry.

Join us for stories and rare photos from people who were on the road (and in the workshop) in the 1960s and 70s. We'll present music history from the sound engineer's perspective.

Tutorial 13 **Sunday, October 7**
5:00 pm – 6:30 pm **Room 1E07**

CONSUMER AUDIO NETWORKING

Presenter: **Steven Harris**
Chris Lacinak

This is a tutorial about the current situation in consumer home audio networking.

Topics include:
- Trends in physical media versus download of music
- Home networking environment, wired and wireless options
- Playback use cases including server and network player interaction
- Digital Rights Management
- Standards for interoperability and control
- Demonstration

The demonstration will consist of a small home network and some examples of commercially available PC servers and networked audio playback devices.

Exhibitor Seminar **Sunday, October 7**
5:00 pm – 6:00 pm **Room 1E04**

CAKEWALK

Presenter: **Allen Sides**

Recording Mary J. Blige—High-Bandwidth Tracking Powered Through Multi-Core Computing

Two-time Grammy-winning engineer/producer Allen Sides recently had a history-making recording date with Mary J. Blige performing with a full orchestra. In this groundbreaking session he recorded 48 tracks simultaneously at 24-bit/192 kHz using one Intel-based PC running SONAR. Allen will be interviewed about the session, followed by brief Q&A. [Sponsored by Cakewalk & Intel]

Technical Committee Meeting **Sunday, October 7**
5:00 pm **Room 1E05**

Technical Committee Meeting on Signal Processing.

Technical Committee Meeting **Sunday, October 7**
6:00 pm **Room 1E16**

Technical Committee Meeting on Audio Forensics.

Student Event/Career Development
DESIGN COMPETITION
Monday, October 8, 8:30 am – 11:30 am
Room 1E06

The design competition is a competition for audio projects developed by students at any university or recording school challenging students with an opportunity to showcase their technical skills. This is not for recording projects or theoretical papers, but rather design concepts and prototypes. Designs will be judged by a panel of industry experts in design and manufacturing. Multiple prizes will be awarded. This event is sponsored by Universal Audio.

Judges: Scott Dorsey, Robert Maher, Christopher Struck

Session P19 **Monday, October 8**
9:00 am – 12:00 noon **Room 1E07**

SIGNAL PROCESSING FOR 3-D AUDIO, PART 1

Chair: **Jean-Marc Jot**, Creative Advanced Technology Center, Scotts Valley, CA, USA

9:00 am

P19-1 Spatial Audio Scene Coding in a Universal Two-Channel 3-D Stereo Format—*Jean-Marc Jot, Arvindh Krishnaswami, Jean Laroche, Juha Merimaa, Mike Goodwin*, Creative Advanced Technology Center, Scotts Valley, CA, USA

We describe a frequency-domain method for phase-amplitude matrix decoding and up-mixing of two-channel stereo recordings, based on spatial analysis of 2-D or 3-D directional and ambient cues in the recording and re-synthesis of these cues for consistent reproduction over

any headphone or loudspeaker playback system. The decoder is compatible with existing two-channel phase-amplitude stereo formats; however, unlike existing time-domain decoders, it preserves source separation and allows accurate reproduction of ambiance and reverberation cues. The two-channel spatial encoding/decoding scheme is extended to incorporate 3-D elevation, without relying on HRTF cues. Applications include data-efficient storage or transmission of multichannel soundtracks and computationally-efficient interactive audio spatialization in a backward-compatible stereo encoding format.
Convention Paper 7276

9:30 am

P19-2 Binaural 3-D Audio Rendering Based on Spatial Audio Scene Coding—*Michael Goodwin, Jean-Marc Jot*, Creative Advanced Technology Center, Scotts Valley, CA, USA

In standard virtualization of stereo or multichannel recordings for headphone reproduction, channel-dependent interaural relationships based on head-related transfer functions are imposed on each input channel in the binaural mix. In this paper we describe a new binaural reproduction paradigm based on frequency-domain spatial analysis-synthesis. The input content is analyzed for channel-independent positional information on a time-frequency basis, and the binaural signal is generated by applying appropriate HRTF cues to each time-frequency component, resulting in a high spatial resolution that overcomes a fundamental limitation of channel-centric virtualization methods. The spatial analysis and synthesis algorithms are discussed in detail and a variety of applications are described.
Convention Paper 7277

10:00 am

P19-3 Real-Time Spatial Representation of Moving Sound Sources—*Christos Tsakostas*,[1] *Andreas Floros*[2]
[1]Holistiks Engineering Systems, Athens, Greece
[2]Ionian University, Corfu, Greece

The simulation of moving sound sources represents a fundamental issue for efficiently representing virtual worlds and acoustic environments but it is limited by the Head Related Transfer Function resolution measurement, usually overcome by interpolation techniques. In this paper a novel time-varying binaural convolution / filtering algorithm is presented that, based on a frequency morphing mechanism that takes into account both physical and psychoacoustic criteria, can efficiently simulate a moving sound source. It is shown that the proposed algorithm overcomes the excessive calculation load problems usually raised by legacy moving sound source spatial representation techniques, while high-quality 3-D sound spatial quality is achieved in both terms of objective and subjective criteria.
Convention Paper 7279

10:30 am

P19-4 The Use of Cephalometric Features for Headmodels in Spatial Audio Processing—*Sunil Bharitkar*,[1,2] *Pall Gislason*[1]
[1]Audyssey Labs, Los Angeles, CA, USA
[2]University of Southern California, Los Angeles, CA, USA

In two-channel or stereo applications, such as for televisions, automotive infotainment, and hi-fi systems, the loudspeakers are typically placed substantially close to each other. The sound field generated from such a setup creates an image that is perceived as monophonic while lacking sufficient spatial "presence." Due to this limitation, a stereo expansion technique may be utilized to widen the soundstage to give the perception to listener(s) that sound is originated from a wider angle (e.g., +/– 30 degrees relative to the median plane) using head-related-transfer functions (HRTF's). In this paper we propose extensions to the headmodel (viz., the ipsilateral and contralateral headshadow functions) based on analysis of the diffraction of sound around head cephalometric features, such as the nose, whose dimensions are of the order to cause variations in the headshadow responses in the high-frequency region. Modeling these variations is important for accurate rendering of a spatialized sound-field for 3-D audio applications. Specifically, this paper presents refinements to the existing spherical head-models for spatial audio applications.
Convention Paper 7280

11:00 am

P19-5 MDCT Domain Analysis and Synthesis of Reverberation for Parametric Stereo Audio—*K. Suresh, T. V. Sreenivas*, Indian Institute of Science, Bangalore, India

We propose a parametric stereo coding analysis and synthesis directly in the MDCT domain using an analysis by synthesis parameter estimation. The stereo signal is represented by an equalized sum signal and spatialization parameters. Equalized sum signal and the spatialization parameters are obtained by sub-band analysis in the MDCT domain. The de-correlated signal required for the stereo synthesis is also generated in the MDCT domain. Subjective evaluation test using MUSHRA shows that the synthesized stereo signal is perceptually satisfactory and comparable to the state of the art parametric coders.
Convention Paper 7281

11:30 am

P19-6 Correlation-Based Ambience Extraction from Stereo Recordings—*Juha Merimaa, Michael M. Goodwin, Jean-Marc Jot*, Creative Advanced Technology Center, Scotts Valley, CA, USA

One of the key components in current multichannel upmixing techniques is identification and extraction of ambience from original stereo recordings. This paper describes correlation-based ambience extraction within a time-frequency analysis-synthesis framework. Two new estimators for the time- and frequency-dependent amount of ambience in the input channels are analytically derived. These estimators are discussed in relationship to two other algorithms from the literature and evaluated with simula-

tions. It is also shown that the time constant used in a recursive correlation computation is an important factor in determining the performance of the algorithms. Short-time correlation estimates are typically biased such that the amount of ambience is underestimated.
Convention Paper 7282

Broadcast Session 15 **Monday, October 8**
9:00 am – 11:00 am **Room 1E08**

INNOVATIONS IN SPORTS BROADCASTING

Presenter: **Ken Hunold**, Dolby Laboratories

Panelists: *Bob Dixon*, Project Manager, SoundDesign, NBC Olympics
Jim Hilson, Dolby Laboratories, Inc., San Francisco, CA, USA

Many sporting events are now being produced in High Definition television. Most of those are also being produced in Surround Sound. What is the perspective of the people behind the consoles creating these audio programs? How do they make multichannel audio work within their production schedules? How does audio enhance the armchair athlete's viewing experience in the home? What other innovations do they have up their sleeves? Mixers and others intimately involved with Sports Broadcasting will offer their view on how multiple channels of audio affect their day-to-day work.

Tutorial 14 **Monday, October 8**
9:00 am – 11:00 am **Room 1E15**

PLANNING AND EXECUTING AUDIO PRESERVATION WORKFLOWS

Presenter: **Dave Ackerman**

This tutorial will address workflow planning for audio preservation activities, including audio transfers, quality assurance and control, metadata documentation, and repository ingest. The first presentation will address the fundamental concepts and principles of developing a preservation-oriented workflow with a focus on quality assessment and control. The second presentation will demonstrate the practical application of these concepts and principles as developed and documented over the past 15 months in the NEH-funded collaboration between Indiana University and Harvard University known as the Sound Directions project. This project studied multiple workflow options for the audio preservation community. One of the outputs from this project is a suite of open source tools for executing audio preservation work-flows. This presentation will offer a walkthrough of the Sound Directions Toolkit showing demonstrative examples of how it can be applied to automate routine, repetitive and mundane tasks, and create more consistent collection output.

Tutorial 15 **Monday, October 8**
9:00 am - 11:00 am **Room 1E10**

LOUDNESS METERING TECHNIQUES AND STANDARDS FOR BROADCASTING

Chair: **Steve Lyman**, Dolby Laboratories, San Francisco, CA, USA

Panelists: *Thomas Lundh*, TC Electronic A/S, Denmark
Andrew Mason, BBC R&D
Gilbert Soulodre, Communications Research Centre, Ottawa, Ontario, Canada

This workshop on loudness metering techniques for broadcasting will describe the basics of new standards from ITU-R and give advice on how the actual meters should be used in practice.

ITU-R has recently issued two standards, BS.1770 and BS.1771. Both the European Broadcasting Union and the World Broadcasting Union have encouraged members to study these new standards and give advice on suitable measuring equipment.

ITU Working Party WP6J gathers broadcasters and manufacturers of equipment to further study and improve the use of loudness meters. For digital contribution and emission, levels need to be more controlled and monitored today. Between programs and within the same program the levelshift can be more than 10 dB.

Tutorial 16 **Monday, October 8**
9:00 am – 11:00 am **Room 1E12/13**

TURNTABLE TECHNIQUE: THE ART OF THE DJ

Presenter: **Stephen Webber**, Berklee College of Music, Boston, MA, USA

The last 30 years have seen a revolution in DJ skills, transforming the once endangered turntable into a musical instrument and revered cultural icon. Originally an expression of hip-hop culture, turntablism has swept the globe and made inroads into jazz, art rock, metal, electronica, advertising, and even symphony hall. Approaching a vinyl platter and a mixer's faders, pots and switches in the same way as one approaches piano keys or trumpet valves is a concept that can transform an audio engineer's relationship to their tools. This tutorial will present attendees with a brief history of the origins of turntable technique (including short video performances by influential practitioners), an overview of the concepts and specific skills involved, and strategies for appropriating the aesthetics of turntablism into other areas of music production and engineering.

Live Sound Seminar 11 **Monday, October 8**
9:00 am – 11:00 am **Room 1E09**

A FAILURE TO COMMUNICATE: WHEN PRODUCTION REQUIREMENTS EXCEED THE BASIC 2- 4-CHANNEL INTERCOM SYSTEM (OR HOW TO EXPAND YOUR WIMPY INTERCOM)

Chair: **Peter Erskine**, Independent

Panelists: *Dave Brand*, Intracom Systems LLC
Larry Estrin, BEST Audio
Mac Kerr
Michael Mason, CP Communications

Intercom requirements for even small to moderate size productions are exceeding the capabilities of basic two to four bus power supplies and base stations. This panel of intercom veterans will explain the various options and techniques for expanding the number of buses, deploying distant user stations, interfacing with other intercom systems, telephone connectivity, and troubleshooting without necessarily moving to large scale matrix frames.

Training Session **Monday, October 8**
9:00 am – 6:00 pm **Room 1E04**

NATIONAL SEMICONDUCTOR TRAINING

Presenters: *Mark Brasfield*, Principal Applications Engineer
John DeCelles, Principal Applications Engineer
Kevin Hoskins, Staff Applications Engineer

A new series of National Semiconductor seminars at the Audio Engineering Society Convention (AES) will help you solve your toughest design challenges. Learn about approaches to achieving the highest technical specifications and sonic quality required by the most discriminating audio professionals.

AGENDA:

9:00 am – 10:15 am Signal Path Circuit Performance

10:30 am – 1:00 pm High-Performance Power Amplifiers (includes lunch)

1:00 pm –- 2:30 pm Signal Path Circuit Performance II

3:00 pm – 4:30 pm High-Performance Power Amplifiers II

ADMISSION: BY ON-LINE REGISTRATION ONLY

Standards Committee Meeting **Monday, October 8**
9:00 am **Room 1E02**

Standards Committee Meeting AESSC Plenary 2.

Session P20 **Monday, October 8**
9:30 am – 12:00 noon **Room 1E16**

APPLICATIONS IN AUDIO, PART 1

Chair: **Michael Kelly**, Sony Computer Entertainment Europe, London, UK

9:30 am

P20-1 A Study of Hearing Damage Caused by Personal MP3 Players—*Adriano Farina*, Liceo Ginnasio statale G.D. Romagnosi, Parma, Italy

This paper aims to assess the actual in-hear sound pressure level during use of mp3 players. The method is based on standard EN 50332 (100 dB as maximum SPL), IEC 60959 (HATS), and IEC 60711 (ear simulators), as explained in the January 2007 issue of the Bruel and Kjaer Magazine (page 13). In this study a number of MP3 players were tested, employing a dummy head and a software for spectrum analysis. The measurements were aimed to assess the hearing damage risk for youngsters who employ an MP3 player for several hours/day. The students of an Italian high school (15 to 18 years old) were asked to supply their personal devices for testing, leaving untouched the gain from the last usage. The results show that the risk of hearing damage is real for many of the devices tested, which revealed to be capable of reproducing average sound pressure levels well above the risk threshold.
Convention Paper 7283

10:00 am

P20-2 Electret Receiver for In-Ear Earphone— *Shu-Ru Lin, Dar-Ming Chiang, I-Chen Lee, Yan-Ren Chen*, Industrial Technology Research Institute (ITRI), Chutung, Hsinchu, Taiwan

This paper presents an electret receiver developed for in-ear earphones. The electret diaphragm is fabricated by a nano-porous fluoropolymer and charged by the corona method at room temperature. The electret diaphragm is driven to vibrate as a piston motion and sound by the electrostatic force while the audio signal is applied. The influence factors, such as electrostatic charge quantities of electret diaphragm and distance between the electrode plate and diaphragm, are investigated to promote the output sound pressure level of the in-ear earphone. An enclosure with resonators is also designed to improve the efficient performance of the in-ear earphone. Consequently, the output sound pressure inside the 2cc coupler can be lifted to exceed 105 dB at 1 kHz with the driving voltage of sound signal Vpp=±3V and remarkably enlarge the output sound pressure level response at low frequency.
Convention Paper 7284

10:30 am

P20-3 New Generation Artificial Larynx—*Andrzej Czyzewski*,[1,2] *Piotr Odya*,[1] *Bozena Kostek*,[1,2] *Piotr Szczuko*[1]
[1]Gdansk University of Technology, Gdansk, Poland
[2]Excellence Center, PROKSIM, Warsaw, Poland

The aim of the presented paper is to show a new generation of devices for laryngectomy patients. The artificial larynx has many disadvantages. The major problem is a background noise caused by the device. There are two different approaches to solve this task. The first one focuses on the artificial larynx. The artificial larynx engineered was equipped with a digital processor and an amplifier. Two algorithms, namely spectral subtraction algorithm and the comb filter, were proposed for noise reduction. The second approach employs PDA to generate speech. A speech synthesis is performed, allowing for playing back any sentence, therefore any text can be entered by a user and played through PDA speaker.
Convention Paper 7285

11:00 am

P20-4 A Graphical Method for Studying Spectra Containing Harmonics and Other Patterns— *Palmyra Catravas*, Union College, Schenectady, NY, USA

A technique for identifying and characterizing patterns in spectra is described. Multiple harmonic series, odd and even harmonics, and missing modes produce identifiable signatures. Motion enhances visual recognition of systematic effects. The technique is adapted for use with more complicated, inharmonic spectral patterns.
Convention Paper 7286

11:30 am

P20-5 Immersive Auditory Environments for Teaching and Learning—*Elizabeth Parvin*, New York University, New York, NY, USA

3-D audio simulations allow for the creation of immersive auditory environments for enhanced and alternative interactive learning. Several supporting teaching and learning philosophies are presented. Experimental research and literature on spatial cognition and sound perception provide further backing. Museums, schools, research and training facilities, as well as online educational websites all significantly can benefit from its use. Design dependence on project purpose, content, and audience is explored. An example installation is discussed.
Convention Paper 7287

Session P21 Monday, October 8
10:00 am – 11:30 am Foyer 1E

POSTERS: SIGNAL PROCESSING, PART 1

10:00 am

P21-1 Dynamic Bit-Rate Adaptation for Speech and Audio—*Nicolle H. van Schijndel*,[1] *Laetitia Gros*,[2] *Steven van de Par*[1]
[1]Philips Research, Eindhoven, The Netherlands
[2]France Telecom R&D, Lannion, France

Many audio and speech transmission applications have to deal with highly time-varying channel capacities, making dynamic adaptation to bit rate an important issue. This paper investigates such adaptation using a coder that is driven by rate-distortion optimization mechanisms, always coding the full signal bandwidth. For perceptual evaluation, the continuous quality evaluation methodology was used, which has specifically been designed for dynamic quality testing. Results show latency and smoothing effects in the judged audio quality, but no quality penalty for the switching between quality levels; the overall quality using adaptation is comparable to using the average available bit rate. Thus, dynamic bit-rate adaptation has a clear benefit as compared to always using the lowest guaranteed available rate.
Convention Paper 7288

10:00 am

P21-2 A 216 kHz 124 dB Single Die Stereo Delta Sigma Audio Analog-to-Digital Converter—*YuQing Yang, Terry Scully, Jacob Abraham*, Texas Instruments, Inc., Austin, TX, USA

A 216 kHz single die stereo delta sigma ADC is designed for high precision audio applications. A single loop, fifth-order, thirty-three level delta sigma analog modulator with positive and negative feedforward path is implemented. An interpolated multilevel quantizer with unevenly weighted quantization levels replaces a conventional 5-bit flash type quantizer in this design. These new techniques suppress the signal dependent energy inside the delta sigma loop and reduce internal channel noise coupling. Integrated with an on-chip bandgap reference circuit, DEM (dynamic element matching) circuit and a linear phase, FIR decimation filter, the ADC achieves 124 dB dynamic range (A-weighted), –110 dB THD+N over a 20 kHz bandwidth. Inter-channel isolation is 130 dB. Power consumption is approximately 330 mW.
Convention Paper 7289

10:00 am

P21-3 Encoding Bandpass Signals Using Level Crossings: A Model-Based Approach—*Ramdas Kumaresan, Nitesh Panchal*, University of Rhode Island, Kingston, RI, USA

A new approach to representing a time-limited, and essentially bandlimited signal x(t), by a set of discrete frequency/time values is proposed. The set of discrete frequencies is the set of frequency locations at which (real and imaginary parts of) the Fourier transform of x(t) cross certain levels and the set of discrete time values corresponds to the traditional level crossings of x(t). The proposed representation is based on a simple bandpass signal model called a Sum-of-Sincs (SOS) model, that exploits our knowledge of the bandwidth/timewidth of x(t). Given the discrete fequency/time locations, we can reconstruct the x(t) by solving a least-squares problem. Using this approach, we propose an analysis/synthesis algorithm to decompose and represent composite signals like speech.
Convention Paper 7290

10:00 am

P21-4 Theory of Short-Time Generalized Harmonic Analysis (SGHA) and its Fundamental Characteristics—*Teruo Muraoka, Takahiro Miura, Daisuke Ochiai, Tohru Ifukube*, University of Tokyo, Tokyo, Japan

Current digital signal processing was utilized practically by rapid progress of processing hardware brought by IC technology and processing algorithms such as FFT and digital filtering. In short, they are for modifying any digitalized signals and classified into following two methods: (1) digital filtering [parametric processing] and (2) analysis-synthesis [non-parametric processing]. Both methods commonly have a weak point when detecting and removing any locally existing frequency components without any side effects. This difficulty will be removed by applying inharmonic frequency analysis. Its fundamental principle was proven by N. Wiener in his publication of "Generalized Harmonic Analysis (GHA)" in 1930. Its application to practical signal processing was achieved by Dr. Y. Hirata in 1994, and the method corresponds to GHA's short time and sequential processing, therefore let us call it Short-Time Generalized Harmonic Analysis (SGHA). The authors have been engaged in research of its fundamental characteristics and application to noise reduction and reported the results at previous AES conventions. This time, SGHA's fundamental theory will be explained together with its characteristics.
Convention Paper 7291

➥

10:00 am

P21-5 Quality Improvement Using a Sinusoidal Model in HE-AAC—*Jung Geun Kim,*[1] *Dong-Il Hyun,*[1] *Dae Hee Youn*,[1] *Young Cheol Park*[2]
[1]Yonsei University, Seoul, Korea
[2]Yonsei University, Wonju-City, Korea

This paper identifies a phenomenon that a signal is distorted because noise floor is generated when restoring a tone in HE-AAC, which does not exist in the original input signal. To solve this matter, it suggests how to restore only the original tonal components in decoding by adding a sinusoidal model to the HE-AAC encoder. In this process, the sinusoidal model is used to analyze a tone and to move it to the place where noise floor is reduced. The lower the bit-rate is, the lower the frequency where the restoration by SBR (Spectral Band Replication) is started becomes; and in the lower frequency, the distortion phenomenon by noise inflow can be sensed easily. Thus, the effect of improvement in the suggested method is greater, and it is beneficial that no additional information or operation in the decoding process is needed.
Convention Paper 7292

10:00 am

P21-6 Special Hearing Aid for Stuttering People— *Piotr Odya*,[1] *Andrzej Czyzewski*[1,2]
[1]Gdansk University of Technology, Gdansk, Poland
[2]Excellence Center, PROKSIM, Warsaw, Poland

Owing to recent progress in digital signal processor developments it has been possible to build a subminiature device combining speech and a hearing aid. Despite its small dimensions, the device can execute quite complex algorithms and can be easily reprogrammed. The paper puts an emphasis on issues related to the design and implementation of algorithms applicable to both speech and hearing aids. Frequency shifting or delaying the audio signal are often used for speech fluency improvement. The basic frequency altering algorithm is similar to the sound compression algorithm used in some special hearing aids. Therefore, the experimental device presented in this paper provides a universal hearing and speech aid that may be used by hearing or speech impaired persons or by persons suffering from both problems, simultaneously.
Convention Paper 7293

10:00 am

P21-7 An Improved Low Complexity AMR-WB+ Encoder Using Neural Networks for Mode Selection—*Jérémie Lecomte, Roch Lefebvre, Guy Richard*, Université de Sherbrooke, Sherbrooke, Quebec, Canada

This paper presents an alternative mode selector based on neural networks to improve the low-complexity AMR-WB+ standard audio coder especially at low bit rates. The AMR-WB+ audio coder is a multimode coder using both time-domain and frequency-domain modes. In low complexity operation, the standard encoder determines the coding mode on a frame-by-frame basis by essentially applying thresholding to parameters extracted from the input signal and using a logic that favors time-domain modes. The mode selector proposed in this paper reduces this bias and achieves a mode decision, which is closer to the full complexity encoder. This results in measurable quality improvements in both objective and subjective assessments.
Convention Paper 7294

Workshop 21 **Monday, October 8**
11:00 am – 1:00 pm **Room 1E10**

IS YOUR JOB KILLING YOU?
(OR JUST MAKING YOU DEAF)

Chair: **Sarah Jones**, Editor, Mix magazine

Panelists: *DeeDee Acquisto*, Musicares
Dave Hampton, Engineer/Producer
Kathy Peck, Executive Director and Co-Founder of H.E.A.R. (Hearing Education and Awareness for Rockers)
Craig Schumacher, Studio Owner/Producer/Founder, TapeOpCon
Andy Vermiglio, Audiologist and Senior Research Associate, House Ear Institute

Audio engineers are in this field because they have great passion for their work. However that same passion, combined with long hours in stressful working environments can contribute to serious health problems ranging from hearing loss to RSI, stress disorders and even heart problems, if they're not proactive about taking care of themselves.

The biggest health issue facing engineers is hearing damage, often a taboo issue in the SPL-heavy recording world. It's a vicious cycle: Those who depend on their hearing to do their job put it at risk by doing that job, day after day. Our ears our livelihood; why aren't we protecting them?

And, it's not just our ears that are at risk. We've had to adapt quickly to a world that seems to spin faster than ever, making us work harder to survive. Communication technology makes it easier than ever to stay in top of things, but harder to get away from work. And industry changes have made everyone's jobs more demanding than ever. In the face of budget cuts, more and more audio engineers are "sole proprietors"- not just tracking and mixing, but archiving, bookkeeping, promoting their work. On the production side, technology advances at lightning speed, making it a constant battle to stay on the cutting edge. And what about making time for our personal lives? The fast pace can take a physical toll.

That said, there are proven steps audio pros can take to preserve and improve their health and well-being—and maximize their career potential. This panel of industry experts (including seasoned veterans who have been through it all and can teach us from experience) will discuss hazards that contribute to noise-induced hearing loss, and examine ways audio engineers can protect their hearing while still doing their job. They will tackle problems inherent in this "workaholic" job culture, offering time-tested solutions for surviving long hours in the studio and life on the road, keeping up with the blinding pace of technology, and thriving in expanded job roles—all while balancing work lives and "real" lives.

Live Sound Seminar 12 — **Monday, October 8**
11:00 am – 1:00 pm — **Room 1E09**

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

Panelists: *John Cooper*
Mary McGregor

Fitting actors with wireless microphone elements and wireless transmitters has become a detailed art form. From ensuring the actor is comfortable and the electronics is safe and secure, to getting the proper sound with minimal detrimental audio effects all while maintaining the visual illusion, two 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."

Special Event
LUNCHTIME KEYNOTE:
JOHN CHOWNING
Monday, October 8, 11:30 am – 12:30 pm
Room 1E08

Introduction by Agnieszka Roginska

FM Synthesis: 40 Years in Art and Industry

It was in 1957, 50 years ago, that Max Mathews at Bell Telephone Laboratories wrote the first sound synthesis program, Music I that he developed and released as Music IV in 1963. Running on mainframe computers at large institutions, the production of music was slow and costly. My naïve discovery in 1967 of frequency modulation synthesis—computationally efficient and having few but perceptually salient parameters—led to a rapid increase in music synthesized by computers, first by software synthesis, then by real-time hardware synthesis ten years later. In 1983, Yamaha's DX7 coupled with the development of MIDI and computer music "hit the streets" resulting in the widespread use of computers in music that continues to this day.

There were many elegant technical and aesthetic contributions to the development of FM synthesis, by a number of people, many unknown to the public. The presentation will include sound-synchronous animations that demonstrate this development ranging from the first experiments from 40 years ago, the breakthrough in 1971, to my most recent composition, "Voices."

John M. Chowning was born in Salem, New Jersey in 1934. Following military service and studies at Wittenberg University, he studied composition in Paris for three years with Nadia Boulanger. In 1964, with the help of Max Mathews of Bell Telephone Laboratories and David Poole of Stanford, he set up a computer music program using the computer system of Stanford University's Artificial Intelligence Laboratory. Beginning the same year he began the research leading to the first generalized sound localization algorithm implemented in a quad format in 1966. He received the doctorate in composition from Stanford University in 1966, where he studied with Leland Smith. The following year he discovered the frequency modulation synthesis (FM) algorithm that led to the most successful synthesis engine in the history of electronic instruments.

Chowning was elected to the American Academy of Arts and Sciences in 1988. He was awarded the Honorary Doctor of Music by Wittenberg University in 1990. The French Ministre de la Culture awarded him the Diplôme d'Officier dans l'Ordre des Arts et Lettres in 1995 and he was given the Doctorat Honoris Causa in 2002 by the Université de la Méditerranée. He taught computer-sound synthesis and composition at Stanford University's Department of Music and was founder and director of the Center for Computer Research in Music and Acoustics (CCRMA), one of the leading centers for computer music and related research.

Workshop 22 — **Monday, October 8**
12:00 noon – 6:00 pm — **Room 1E11**

5.1 MIXING TECHNIQUES EXPLAINED AND OBSERVED

Panelists: **Frank Filipetti**, Right Track Recording, New York, NY, USA
George Massenburg
Ronald Prent, Galaxy Studios, Mol, Belgium
Jeff Wolpert, Desert Fish Inc.

5.1 Mixing Techniques Explained and Observed is a hands-on demonstration of mixing with audience participating interactivity. The mixes are presented with the original multitrack sessions, so they can be broken down into the individual elements, and mixing techniques can be experienced intuitively.

Special Event
PLATINUM ROAD WARRIORS
Monday, October 8, 12:00 pm – 1:30 pm
Room 1E12/13

Moderator: **Clive Young**

Panelists: *Howard Page*
Robert Scovill
Brian Speiser

The Platinum Road Warriors, moderated by Clive Young, *Pro Sound News* Sr. Editor and author of *CRANK IT UP: Live Sound Secrets of the Top Tour Engineers*, will offer insights from a freewheeling panel covering the latest trends, techniques, and tools that shape modern sound reinforcement.

Participating FOH engineers include: Robert Scovill, who has accrued over 3,000 live event mixing credits during the course of a 27-year career, where he has mixed the likes of Tom Petty & The Heartbreakers, Matchbox Twenty, Prince, Rush, and Def Leppard, among many others. He is a technical consultant and product endorser for such manufacturers as: Neumann, Audio Technica, Electrovoice, Servo Drive, and Alesis. He also serves as Market Manager for Live Sound Products for Digidesign. Scovill also operates his own Eldon's Boy Productions Inc. and MusiCanvas Recording Studio.

Brian Speiser began touring 10 years ago at the age of 20 when he began working for Long Island-based SK Sound. For the past eight years, he's worked as the front of house engineer for They Might Be Giants, and has mixed several live releases for the band as well as a live Direct TV concert filmed for Disney Records. Additionally, he has "done a short jail sentence as Ryan Adams' front of house engineer" and has handled FOH duties for The Indigo Girls on their tours around the world for the past two years.

Howard Page is director of engineering for Showco, a division of Clair Brothers Audio. Having spent decades ➥

in the live pro audio business, Page has mixed tours for acts as varied as Van Halen and Sade, but his biggest impact arguably may be his conception and development of Showco's Showconsole, one of the first digitally controlled analog live mixing desks—the sixth console he has designed and built over his career. A native of Australia, Page spent 17 years as a partner in Jans Concert Productions, before coming to America in 1989, where he has worked for Showco ever since.

These and other TBA FOH engineers will share war stories and field questions from the audience.

Tutorial 17 **Monday, October 8**
12:30 pm – 4:00 pm **Room 1E15**

REPRODUCER ALIGNMENT AND CALIBRATION FOR OPTIMAL PLAYBACK

Presenter: **Sean Davies**
Eric Jacobs
Mike Spitz

This tutorial is intended to provide a practical overview and insight into the art and science of proper setup and calibration of turntables and analog tape reproducers. In a world where audio is born nearly digital, we now have entire generations of audio engineers who have never learned the proper setup for analog reproducers. Given the enormous amount of cultural heritage that exists only in disc and tape format, it is imperative that the skills necessary to retrieve the original signal from these carriers not be allowed to pass into extinction. This session will be divided into two segments. The first will deal with tone arm calibration, stylus selection, compensation for off center discs, etc. The second segment will discuss the considerations for choosing a playback head, its alignment and the calibration of the tape machines level and equalization electronics. Ideally, reproduction equipment and video cameras will be on hand to allow exhibition of detailed techniques during the presentations.

Tutorial 18 **Monday, October 8**
12:30 pm – 2:30 pm **Room 1E08**

"HERE COMES THE SUN"—ARE YOU LISTENING?

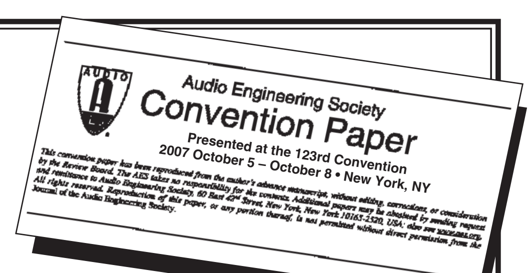
Presenter: **William Moylan,** University of Massachusetts, Lowell, MA, USA

Arguably the most important skill in audio work, listening is perhaps the most difficult to understand and master. Hearing requires learning what to listen for (the dimensions of live and reproduced sound), then developing skill in the process of listening (the technique of discovering, experiencing and understanding these many dimensions and details of sound). Listening in audio often requires almost simultaneous attention to a multitude of levels of perspective, critical and analytical listening information, and more. Through listening and analysis of "Here Comes the Sun" and other Beatles productions, this tutorial illustrates key ways to enhance your listening skills and to learn from those who create the albums we love.

Student Event/Career Development
STUDENT DELEGATE ASSEMBLY MEETING —PART 2
Monday, October 8, 12:30 pm – 1:30 pm
Room 1E06

The closing meeting of the SDA will host the election of a new vice chair. Votes will be cast by a designated representative from each recognized AES student section or academic institution in the North/Latin America Regions present at the meeting. Judges' comments and awards will be presented for the Recording and Design Competitions. Plans for future student activities at local, regional, and international levels will be summarized and discussed among those present at the meeting

123rd Convention Papers and CD-ROM



Convention Papers of many of the presentations given at the 123rd Convention and a CD-ROM containing the 123rd Convention Papers are available from the Audio Engineering Society. Prices follow:

123rd CONVENTION PAPERS (single copy)
Member: $ 5. (USA)
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123rd CD-ROM (152 papers)
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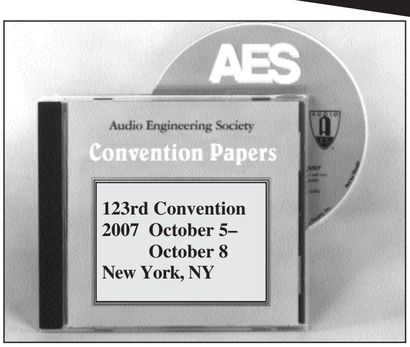

Session P22 **Monday, October 8**
1:00 pm – 4:30 pm **Room 1E07**

SIGNAL PROCESSING FOR 3-D AUDIO, PART 2

Chair: **Søren Bech**, Bang & Olufsen a/s, Struer, Denmark

1:00 pm

P22-1 Real-Time Auralization Employing a Not-Linear, Not-Time-Invariant Convolver— *Angelo Farina*,[1] *Adriano Farina*[2]
[1]University of Parma, Parma, Italy
[2]Liceo Ginnasio Statale G. D. Romagnosi, Parma, Italy

The paper reports the first results of listening tests performed with a new software tool, capable of not-linear convolution (employing the Diagonal Volterra Kernel approach) and of time-variation (employing efficient morphing among a number of kernels). The listening tests were done in a special listening room, employing a menu-driven playback system, capable of presenting blindly sound samples recorded from real-world devices and samples simulated employing the new software tool, and, for comparison, samples obtained by traditional linear, time-invariant convolution. The listener answers a questionnaire for each sound sample, being able to switch them back and forth for better comparing. The results show that this new device-emulation tool provides much better results than already-existing convolution plug-ins (which only emulate the linear, time-invariant behavior), requiring little computational load and causing short latency and prompt reaction to user's action.
Convention Paper 7295

1:30 pm

P22-2 Real-Time Panning Convolution Reverberation —*Rebecca Stewart, Mark Sandler*, Queen Mary University of London, London, UK

Convolution reverberation is an excellent method for generating high-quality artificial reverberation that accurately portrays a specific space, but it can only represent the static listener and source positions of the measured impulse response being convolved. In this paper multiple measured impulse responses along with interpolated impulse responses between measured locations are convolved with dry input audio to create the illusion of a moving source. The computational cost is decreased by using a hybrid approach to reverberation that recreates the early reflections through convolution with a truncated impulse response, while the late reverberation is simulated with a feedback delay network.
Convention Paper 7296

2:00 pm

P22-3 Ambisonic Panning—*Martin Neukom*, Zurich University of the Arts, Zurich, Switzerland

Ambisonics is a surround-system for encoding and rendering a 3-D sound field. Sound is encoded and stored in multichannel sound files and is decoded for playback. In this paper a panning function equivalent to the result of ambisonic encoding and so-called in-phase decoding is presented. In this function the order of ambisonic resolution is just a variable that can be an arbitrary positive number not restricted to integers and that can be changed during playback. The equivalence is shown, limitations and advantages of the technique are mentioned, and real time applications are described.
Convention Paper 7297

2:30 pm

P22-4 Adaptive Karhunen-Lòeve Transform for Multichannel Audio—*Yu Jiao, Slawomir Zielinski, Francis Rumsey*, University of Surrey, Guildford, Surrey, UK

In previous works, the authors proposed the hierarchical bandwidth limitation technique based on Karhunen-Lòeve Transform (KLT) to reduce the bandwidth for multichannel audio transmission. The subjective results proved that this technique could be used to reduce the overall bandwidth without significant audio quality degradation. Further study found that the transform matrix varied considerably over time for many recordings. In this paper the KLT matrix was calculated based on short-term signals and updated adaptively over time. The perceptual effects of the adaptive KLT process were studied using a series of listening tests. The results showed that adaptive KLT resulted in better spatial quality than nonadaptive KLT but introduced some other artifacts.
Convention Paper 7298

3:00 pm

P22-5 Extension of an Analytic Secondary Source Selection Criterion for Wave Field Synthesis —*Sascha Spors*, Berlin University of Technology, Berlin, Germany

Wave field synthesis (WFS) is a spatial sound reproduction technique that facilitates a high number of loudspeakers (secondary sources) to create a virtual auditory scene for a large listening area. It requires a sensible selection of the loudspeakers that are active for the reproduction of a particular virtual source. For virtual point sources and plane waves suitable intuitively derived selection criteria are used in practical implementations. However, for more complex virtual source models and loudspeaker array contours the selection might not be straightforward. In a previous publication the author proposed secondary source selection criterion on the basis of the sound intensity vector. This contribution will extend this criterion to data-based rendering and focused sources and will discuss truncation effects.
Convention Paper 7299

3:30 pm

P22-6 Adaptive Wave Field Synthesis for Sound Field Reproduction: Theory, Experiments, and Future Perspectives—*Philippe-Aubert Gauthier, Alain Berry*, Université de Sherbrooke, Sherbrooke, Quebec, Canada

Wave field synthesis is a sound field reproduction technology that assumes that the ➥

reproduction environment is anechoic. A real reproduction space thus reduces the objective accuracy of wave field synthesis. Adaptive wave field synthesis is defined as a combination of wave field synthesis and active compensation. With adaptive wave field synthesis the reproduction errors are minimized along with the departure penalty from the wave field synthesis solution. Analysis based on the singular value decomposition connects wave field synthesis, active compensation, and Ambisonics. The decomposition allows the practical implementation of adaptive wave field synthesis based on independent radiation mode control. Results of experiments in different rooms support the theoretical propositions and show the efficiency of adaptive wave field synthesis for sound field reproduction.
Convention Paper 7300
Winner of the Student Paper Award

4:00 pm

P22-7 360° Localization via 4.x RACE Processing—
Ralph Glasgal, Ambiophonics Institute, Rockleigh, NJ, USA

Recursive Ambiophonic Crosstalk Elimination (RACE), implemented as a VST plug-in, convolved from an impulse response, or purchased as part of a TacT Audio or other home audiophile product, properly reproduces all the ITD and ILD data sequestered in most standard two or multichannel media. Ambiophonics is so named because it is intended to be the replacement for 75 year old stereophonics and 5.1 in the home, car, or monitoring studio, but not in theaters. The response curves show that RACE produces a loudspeaker binaural sound field with no audible colorations, much like Ambisonics or Wavefield Synthesis. RACE can do this starting with most standard CD/LP/DVD two, four or five-channel media, or even better, 2 or 4 channel recordings made with an Ambiophone, using one or two pairs of closely spaced loudspeakers. The RACE stage can easily span up to 170° for two channel orchestral recordings or 360° for movie/electronic-music surround sources. RACE is not sensitive to head rotation and listeners can nod, recline, stand up, lean sideways, move forward and back, or sit one behind the other. As in 5.1, off center listeners can easily localize the center dialog even though no center speaker is ever needed.
Convention Paper 7301

Session P23 Monday, October 8
1:00 pm – 3:30 pm Room 1E16

APPLICATIONS IN AUDIO, PART 2

Chair: **Juha Backman**, Nokia Mobile Phones, Espooo, Finland

1:00 pm

P23-1 Loudspeaker Systems for Flat Television Sets— *Herwig Behrends, Werner Bradinal, Christoph Heinsberger*, NXP Semiconductors, Hamburg, Germany

The rapidly increasing sales of liquid crystal- and plasma display television sets lead to new challenges to the sound processing inside the TV-sets. Flat cabinets do not sufficiently accommodate room for loudspeakers that are able to reproduce frequencies below 100 to 200 Hz without distortions and with a reasonable sound pressure level. Cost reduction forces the set makers to use cheap and small loudspeakers, which are in no way comparable to the loudspeakers used in cathode ray tube televisions. In this paper we will describe the trends and the requirements of the market and discuss different approaches and a practical implementation of a new algorithm, which tackle these problems.
Convention Paper 7302

1:30 pm

P23-2 Loudspeakers for Flexible Displays—
Takehiro Sugimoto,[1] *Kazuho Ono*,[1] *Kohichi Kurozumi*,[2] *Akio Ando*,[1] *Akira Hara*,[3] *Yuichi Morita*,[3] *Akito Miura*[3]
[1]NHK Science & Technical Research Laboratories, Setagaya-ku, Tokyo, Japan
[2]NHK Engineering Services, Setagaya-ku, Tokyo, Japan
[3]Foster Electric Co., Ltd., Akishima, Tokyo, Japan

Flexible displays that can be rolled up would allow users to enjoy programs wherever they are. NHK Science & Technical Research laboratories have been developing flexible displays for mobile television. The loudspeaker for such televisions must have the same features as the displays; they must be thin, lightweight, and flexible. We created two types of loudspeakers; one was made of polyvinylidene fluoride and the other used electro-dynamic actuators. Their characteristics were demonstrated to be suitable for mobile use and promising for flexible displays.
Convention Paper 7303

2:00 pm

P23-3 Software-Based Live Sound Measurements, Part 2—*Wolfgang Ahnert, Stefan Feistel, Alexandru Radu Miron, Enno Finder*, Ahnert Feistel Media Group, Berlin, Germany

In previous publications the authors introduced the software-based measuring system EASERA to be used for measurements with prerecorded music and speech signals. This second part investigates the use of excitation signals supplied from an independent external source in real-time. Using a newly developed program module live-sound recordings or speech and music signals from a microphone input and from the mixing console can be utilized to obtain impulse response data for further evaluation. New noise suppression methods are presented that allow these impulse responses to be acquired in full-length even in occupied venues. As case studies, room acoustic measurements based on live sound supply are discussed for a concert hall and a large cathedral. Required measuring conditions and limitations are derived as a result.
Convention Paper 7304

2:30 pm

P23-4 A System for Remote Control of the Height of Suspended Microphones—*Douglas McKinnie*, Middle Tennessee State University, Murfreesboro, TN, USA

An electrically driven pulley system allowing remote control of the height of cable-suspended microphones is described. It can be assembled from inexpensive and readily available component parts. A reverse block-and tackle system is used to allow many meters of cable to be drawn into a 1.2 meter long space, allowing the cable to remain connected and the microphone to remain in use during movement. An advantage of this system is that single microphones, stereo pairs, or microphone arrays can be remotely positioned "by ear" during rehearsal, soundcheck, or warmup.
Convention Paper 7305

3:00 pm

P23-5 Music at Your Fingertips: An Electrotactile Fader—*Jörn Loviscach*, Hochschule Bremen (University of Applied Sciences), Bremen, Germany

Tactile sensations can be invoked by applying short high-voltage low-current electrical pulses to the skin. This phenomenon has been researched extensively to support visually or hearing impaired persons. However, it can also be applied to operate audio production tools in eyes-free mode and without acoustical interferences. The electrotactile fader presented in this paper is used to indicate markers or to "display" a track's short-time spectrum using five electrodes mounted on the lever. As opposed to mechanical solutions, which may for instance involve the fader's motor, the electrotactile display neither causes acoustic noise nor reduces the fader's input precision due to vibration.
Convention Paper 7306

Broadcast Session 16 — Monday, October 8
1:00 pm – 2:00 pm — Room 1E10

BROADCAST TUTORIAL:
BUILDING A RADIO FACILITY
IN THE DEVELOPING WORLD

Presenter: **Dan Braverman**

"You can't find a 10/32 rack screw in Mchinji province in Central Malawi to save your life! I know—I spent a whole afternoon trying!"

As founder of the African Broadcast Alliance, speaker Daniel Braverman recounts his personal experiences building radio stations in the Developing Countries of Central Africa and why 30 years of building stations in the US was only an apprenticeship for the rigors of working in Africa. "It's the most challenging, frustrating, and fulfilling broadcast assignment you'll ever take on."

"Oh, and a 4.5 mm screw will substitute for a 10/32 in a pinch." He'll share that tip and a few more along with great pictures, advice, and recommendations on how anyone can get involved with building stations that serve a higher goal.

Session P24 — Monday, October 8
1:30 pm – 3:00 pm — Foyer 1E

POSTERS: SIGNAL PROCESSING, PART 2

1:30 pm

P24-1 Concept and Components of a Sound Field Effector Using a Loudspeaker Array—*Teruki Oto,*[1] *Tomoaki Tanno,*[2] *Jiang Hua,*[2] *Risa Tamaura,*[2] *Syogo Kiryu,*[2] *Toru Kamekawa*[3]
[1]Kenwood Corporation, Tokyo, Japan
[2]Musashi Institute of Technology, Tokyo, Japan
[3]Tokyo National University of Fine Arts and Music, Tokyo, Japan

Most effectors used for electrical music instruments provide some temporal changes to sounds. If effectors aimed at spatial expressions had been developed, artists could have a new performance. We propose a Sound Field Effector using a loudspeaker array. Various sound fields such as a focus can be controlled in real time by sound engineering and/or artists. The Sound Field Effector is mainly divided to software parts and hardware parts. A 16-ch. system was developed as a prototype. The system can change sound fields within 1 msec. A focal pattern produced with the system was measured in an anechoic room.
Convention Paper 7307

1:30 pm

P24-2 A Novel Mapping with Natural Transition from Linear to Logarithmic Scaling—*Joerg Panzer*, R&D Team, Salgen, Germany

The area hyperbolic function ArSinh has the interesting property of performing a linear mapping at arguments close to zero and a quasi-logarithmic mapping for large arguments. Further, it works also with a negative abscissa and at the zero-point. The transition from the linear to the logarithmic range is monotonic, so is the transition to the negative range. This paper demonstrates the use of the ArSinh-function in a range of application examples, such as zooming into the display of transfer-functions, sampling of curves with high density at a specific point, and a coarse resolution elsewhere. The paper also reviews the linear and logarithmic mapping and discusses the properties of the new ArSinh-mapping.
Convention Paper 7308

1:30 pm

P24-3 Real Time Implementation of an Innovative Digital Audio Equalizer—*Stefania Cecchi,*[1] *Paolo Peretti,*[1] *Lorenzo Palestini,*[1] *Francesco Piazza,*[1] *Ferruccio Bettarelli,*[2] *Ariano Lattanzi*[2]
[1]Università Politecnica Delle Marche, Ancona, Italy
[2]Leaff Engineering, Porto Potenza Picena (MC), Italy

Fixed frequency response audio equalization has well-known problems due to algorithms computational complexity and to the filters design techniques. This paper describes the design and the real time implementation of an ➥

M-band linear phase digital audio equalizer. Beginning from multirate systems and filterbanks, an innovative uniform and nonuniform bands audio equalizer is derived. The idea of this work arises from different approaches employed in filterbanks to avoid aliasing in the case of adaptive filtering in each band. The effectiveness of the real time implementation is shown comparing it with a frequency domain equalizer. The solution presented here has several advantages in terms of low computational complexity, low delay, and uniform frequency response avoiding ripple between adjacent bands.
Convention Paper 7309

1:30 pm

P24-4 Wideband Beamforming Method Using Two-Dimensional Digital Filter—*Koji Kushida,*[1] *Yasushi Shimizu,*[1] *Kiyoshi Nishikawa*[2]
[1]Yamaha Corporation, Japan
[2]Kanazawa University, Kanazawa, Japan

This paper presents a method for designing a DSP-controlled directional array loudspeaker with constant directivity and specified sidelobe level over the wideband frequency by means of the two-dimensional (2-D) Fourier series approximation. The band of the constant directivity can be extended in the lower frequency band by using the nonphysical area in the 2-D frequency plane, where the target amplitude response of the 2-D filter is set to design the 2-D FIR filter. We discuss that the beamwidth of the array loudspeaker can be narrowed in the lower frequency band with a modification of the original algorithm by K. Nishikawa, et al.
Convention Paper 7310

1:30 pm

P24-5 Linear Phase Mixed FIR/IIR Crossover Networks: Design and Real-Time Implementation—*Lorenzo Palestini,*[1] *Paolo Peretti,*[1] *Stefania Cecchi,*[1] *Francesco Piazza,*[1] *Ariano Lattanzi,*[2] *Ferruccio Bettarelli*[2]
[1]Università Politecnica Delle Marche, Ancona, Italy
[2]Leaff Engineering, Porto Potenza Picena (MC), Italy

Crossover networks are crucial components of audio reproduction systems and therefore they have received great attention in literature. In this paper the design and implementation of a digital crossover will be presented. A mixed FIR/IIR solution has been explored in order to exploit the respective strengths of FIR and IIR realizations, aiming at designing a low delay, low complexity, easily extendible, approximately linear phase crossover network. A software real-time implementation for the NU-Tech platform of the proposed system will be shown. Practical tests have been carried out to evaluate the performance of the proposed approach.
Convention Paper 7311

1:30 pm

P24-6 Convolutive Blind Source Separation of Speech Signals in the Low Frequency Bands—*Maria Jafari, Mark Plumbley*, Queen Mary University of London, London, UK

Sub-band methods are often used to address the problem of convolutive blind speech separation, as they offer the computational advantage of approximating convolutions by multiplications. The computational load, however, often remains quite high, because separation is performed on several subbands. In this paper we exploit the well known fact that the high frequency content of speech signals typically conveys little information, since most of the speech power is found in frequencies up to 4 kHz, and consider separation only in frequency bands below a certain threshold. We investigate the effect of changing the threshold, and find that separation performed only in the low frequencies can lead to the recovered signals being similar in quality to those extracted from all frequencies.
Convention Paper 7312

1:30 pm

P24-7 A Highly Directive 2-Capsule Based Microphone—*Christof Faller*, Illusonic LLC, Chavannes, Switzerland

While microphone technology has reached a high level of performance in terms of signal-to-noise ratio and linearity, directivity of commonly used first order microphones is limited. Higher order gradient based microphones can achieve higher directivity but suffer from signal-to-noise ratio issues. The usefulness of beamforming techniques with multiple capsules is limited due to high cost (a high number of capsules is required for high directivity) and high frequency variant directional response. A highly directive 2-capsule-based microphone is proposed, using two cardioid capsules. Time-frequency processing is applied to the corresponding two signals. A highly directive directional response is achieved that is time invariant and frequency invariant over a large frequency range.
Convention Paper 7313

Live Sound Seminar 13 **Monday, October 8**
1:30 pm – 3:30 pm **Room 1E09**

CONCERT SOUND SYSTEM DESIGN, SETUP, OPERATION, AND THE CREATIVE USE OF DIGITAL MIXING TECHNOLOGY

Chair: **Robert Scovill**

Panelists: *Forrest Grosz*
Eddie Mapp
David Morgan
Scott Regsdale
Barry Sanders
Blake Suib

The rapid acceptance of digital routing and mixing technologies in live concert sound reinforcement has opened the door to numerous new workflows and solutions. An entire generation of mixers, both veterans and newcomers are currently reshaping the way live sound is perceived and executed. Enjoy a lively discussion by users of these new technologies and gain insight into how digital technology is reshaping sound reinforcement for the foreseeable future. Topics will include digital console user interfaces, automation concepts, remote control, multi-track archiving, virtual sound check and media delivery among others.

Tutorial 19 **Monday, October 8**
2:00 pm – 4:00 pm **Room 1E12/13**

COMPRESS FOR SUCCESS—MASTER THE MOST MISUNDERSTOOD FX DEVICE

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

Dynamic range compression confounds many recording engineers, from rookies to veterans. As an audio effect it can be difficult to hear and even more difficult to describe. As a tool its controls can be counterintuitive and its meters and flashing lights uninformative. This tutorial organizes the broad range of effects created by audio compressors, as audio engineers use it to reduce/control dynamic range, increase perceived loudness, improve intelligibility and articulation, reshape the amplitude envelope, add creative doses of distortion, and extract ambience cues, breaths, squeaks, and rattles. Attendees will learn when to reach for compression, know a good starting place for compression parameters (ratio, threshold, attack, and release), and advance their understanding of what to listen for and which way to tweak.

Workshop 23 **Monday, October 8**
2:30 pm – 5:00 pm **Room 1E08**

CONCERT HALLS AND THEIR ACOUSTICS

Chair: **John Allen**

Panelists: *Chris Blair*
Barry Blesser
Leo L. Beranek
Larry Kirkegaard
Alan Valentine

In the years following the pioneering work in acoustics by Wallace Sabine and the more recent ground breaking contributions of Leo Beranek, how well have the known principles of good acoustics been applied to the concert halls being built today? Hear from some of the leaders in the field, including Beranek. Topics of the panel will include why we respond to the acoustic environment we are in, why good acoustics are so important for the musicians as well as the audience, case studies of how poor concert halls have been improved, and finally hear about America’s newest concert hall and why it may be one of the best ever built.

Broadcast Session 17 **Monday, October 8**
2:30 pm – 4:30 pm **Room 1E10**

INTERNET STREAMING— AUDIO QUALITY, MEASUREMENT, & MONITORING

Chair: **David Bialik**

Panelists: *Brian Carroll*, Akamai
David Day, Day Sequerra
Frank Foti, Omnia
Thomas E. Mintner, NTI Americas Inc.
Skip Pizzi, Microsoft Corp.
John Rosso, ABC Radio – Citadel Interactive

Internet Streaming has become a provider of audio and video content to the public. Now that the public has recognized the medium, the provider needs to deliver the content with a quality comparable to other mediums. Audio monitoring is becoming important, and a need to quantify the performance is important so that the streamer can deliver product of a standard quality.

Live Sound Seminar 14 **Monday, October 8**
3:30 pm – 6:00 pm **Room 1E09**

TUTORIAL: RADIO FREQUENCY INTERFERENCE AND AUDIO SYSTEMS

Chair: **Jim Brown**, Audio Systems Group

This tutorial begins by identifying and discussing the fundamental mechanisms that couple RF into audio systems and allow it to be detected. Attention is then given to design techniques for equipment and systems that avoid these problems, and methods of fixing problems with existing equipment and systems that have been poorly designed or built.