AES 129th Convention Program

November 4 – 7, 2010

Moscone Center, San Francisco, CA, USA

LIVE SOUND SYMPOSIUM: SURROUND LIVE 8
Absolutely Surrounded
Wednesday, November 3, 9:00 am – 5:00 pm
Room 130

Preconvention Special Event; additional fee applies

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

NOTE: Program subject to change based on availability of personnel.

8:30 am - 9:00 am – Registration

9:00 am - 12:00 noon – Formal Presentations

12:00 noon - 1:00 pm – Lunch

1:00 pm - 4:30 pm – Showcase Presentations

4:30 pm - 5:00 pm – Q&A and Wrap-Up

MORNING SCHEDULE 9:00 AM – 12 NOON

Keynote: **Russ Berger**, RBDG- Dallas, TX

The Winter Olympics: **Michael Nunan**, CTV Canada

Hockey Night in Canada: **Michael Nunan**, CTV Canada

Surround for Radio: **Kyle Wesloh**, American Public Media

Randall Smith, Post Haste Sound, Mixer for Rocky Horror Picture Show and other programming in 7.1 for Blu-ray

AFTERNOON SCHEDULE 1:00 PM – 5:00 PM

Technology Showcase Special Presentations:

- beyerdynamic— Headzone Demo 5.1 headphone monitoring system: **Paul Froula**, beyerdynamic/American Music & Sound
- Upmix-Downmix—The Current State of the Art
- Six Current Approaches—TC Electronic, Audionamix, Dolby, DTS, Inc., Isostem, Penteo

Open Format Q&A

This Event is Sponsored Exclusively by DTS Inc. with Technical Support from: American Music & Sound, which is providing the Allen & Heath iLive-R72 with the iDR-32 32x16 stage box and Dynaudio Acoustic Air Monitor System in 7.1 configuration provided by TC Electronic.

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DTS Inc. will be sponsoring a special raffle for a Sony PS3 Game System during the event – All paid ticketholders are eligible. The Winner will have their prize shipped to them after the event.

All attendees will receive a free special DTS demo disc for attending.

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.

Session P1 **Thursday, Nov. 4**
9:30 am – 12:30 pm **Room 220**

TRANSDUCERS AND PROCESSING FOR LIVE SOUND

Chair: **Scott Norcross**, Communication Research Centre, Ottawa, Ontario, Canada

9:30 am

P1-1 A Performance Ranking of Seven Different Types of Loudspeaker Line Arrays—*D. B. (Don) Keele, Jr.*, DBK Associates and Labs, Bloomington, IN, USA

Seven types of loudspeaker line arrays were ranked considering eight performance parameters including (1) beamwidth uniformity, (2) directivity uniformity, (3) sound field uniformity, (4) side lobe suppression, (5) uniformity of polar response, (6) smoothness of off-axis frequency response, (7) sound pressure rolloff versus distance, and (8) near-far polar pattern uniformity. Line arrays analyzed include: (1) un-shaded straight-line array, (2) Hann-shaded straight-line array, (3) "J"-line array, (4) spiral- or progressive-line array, (5) un-shaded circular-arc array, (6) CBT circular-arc array, and (7) a CBT delay-curved straight-line array. All arrays were analyzed assuming no extra drive signal processing other than frequency-independent shading. A weighted performance analysis yielded the following ranking from best to worse 6, 7, 5, 4, 3, 2, 1, with the CBT Legendre-shaded circular-arc array on top and the un-shaded straight-line array on the bottom.
Convention Paper 8155

10:00 am

P1-2 A Reliable Procedure for Polarity Measurements on Line Arrays—*Gregor Schmidle, Markus Becker*, NTi Audio AG, Schaan, Liechtenstein

The performance of a line array strongly depends on the correct installation of its loudspeakers. For instance, a single loudspeaker with incorrect polarity may clearly compromise the sound level and directivity of the whole system. The identification of such errors, however, can be very time consuming. Therefore, it is desirable to have a fast, yet reliable procedure to finding such array elements. This paper presents a step-by-step method to check the integrity of a line array and to find the cause in case of a polarity problem. Besides the theoretical background, a successful practical case is described.
Convention Paper 8156

10:30 am

P1-3 Calculating Time Delays of Multiple Active Sources in Live Sound—*Alice Clifford, Josh Reiss*, Queen Mary, University of London, London, UK

Delays caused by differences in distance between sources and microphones cause many problems in live audio, most notably comb filtering. This paper presents a new method that is able to calculate the relative time delays of multiple active sources to multiple microphones where previous methods are unable to. The calculated time delays can be used to compensate for delays that cause comb filtering and can also be used in source separation methods that utilize delays. The proposed method is shown to be able to calculate delays in configurations where other methods fail and is also able to give an estimate of sources physical positions. The results show that multiple delays can be accurately calculated when multiple sources are active and that noise can affect the accuracy of the method.
Convention Paper 8157

11:00 am

P1-4 Coherent Superposition of Acoustic Sources as a Function of Environmental Parameters—*Stefan Feistel*,[1] *Rainer Feistel*[2]
[1]Ahnert Feistel Media Group, Berlin, Germany
[2]Institut für Ostseeforschung, Warnemuende, Germany

Sound reinforcement systems and loudspeaker arrays consist of numerous, spatially distributed sources. The signal alignment of these components is crucial to provide even level coverage and consistent spectral distribution throughout the audience areas. One usually assumes that sources located close to each other sum coherently in contrast to sources spaced far apart which sum energetically at the receiver. However, in reality this assumption seldom holds since environmental conditions, such as fluctuations of temperature and air flow and their spatial correlation, determine the actual level of coherence. We derive the theoretical framework for this important effect based on stochastic theory. We quantify the resulting coherence as a function of the different environmental parameters. Results are verified using numerical models and measurement.
Convention Paper 8158

11:30 am

P1-5 Optimizing the Controls of Homogeneous Loudspeaker Array—*Michael Terrell, Mark Sandler*, Queen Mary University of London, London, UK

An optimization technique is demonstrated that can be used to control the frequency response at multiple receiver locations to a homogenous loudspeaker array. Each loudspeaker element in the array is identical but can be controlled individually using vertical and horizontal angle, delay, broadband gain, a 7-band parametric equalizer, and a 5th order all pass filter. The arrays examined contain 8 loudspeakers, and the controls are optimized using a genetic algorithm. This is demonstrated for a 2-D case study based on the requirement that the magnitude response at each location is the same.
Convention Paper 8159

12:00 noon

P1-6 Perceptual Dimensions of Stage-Floor Vibration Experienced During a Musical Performance— *Clemeth L. Abercrombie*,[1] *Jonas Braasch*[2]
[1]Artec Consultants Inc., New York, NY, USA
[2]Rensselaer Polytechnic Institute, Troy, NY, USA

The human ability to distinguish differences in tactile signals generated by a musical instrument and experienced on typical stage-floor constructions is explored using an audio-tactile display (headphones and calibrated motion platform). Audio and vibration signals generated by a contrabass are combined with mechanical impedance measurements of five stage floors to create stimuli. Test participants are asked to report differences between tactile signals given a fixed audio environment. Multidimensional scaling is used to identify perceptual dimensions in subjective responses. Results show that stage vibration exceeds the threshold of perception, with acceleration up to 0.04 ms-2 Wk-peak. Sensation level dominates perceived differences between tactile signals measured for different stage-floor constructions, while audio-tactile time delays have negligible influence.
Convention Paper 8160

Session P2 **Thursday, Nov. 4**
9:30 am – 12:30 pm **Room 236**

SPEECH PROCESSING

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

9:30 am

P2-1 Language Scrambling for In-Game Voice-Chat Applications—*Nicolas Tsingos, Charles Robinson*, Dolby Laboratories, San Francisco, CA, USA

We propose a solution to enable speech-driven alien language synthesis in an in-game voice-chat system. Our technique selectively replaces the users' input speech with a corresponding alien language output synthesized on-the-fly. It is optimized for a client-server architecture and uses a concatenative synthesis framework. To limit memory requirements, as well as preserve forwarding capabilities on the server, the concatenative synthesis is performed in the coded domain. For gaming applications, our approach can be used to selectively scramble the speech of opposing team members in order to provide compelling in-game voice feedback without exposing their strategy. The system has been implemented with multiple alien races in a virtual environment with effective, entertaining results.
Convention Paper 8161

10:00 am

P2-2 Speech Referenced Limiting: Controlling the Loudness of a Signal with Reference to its Speech Loudness—*Michael Fisher,*[1,2] *Nicky Chong-White,*[1,2] *Harvey Dillon*[1]
[1]The HEARing CRC, Melbourne, Victoria, Australia
[2]National Acoustic Laboratories, Sydney, NSW, Australia

A novel method of sound amplitude limiting for signals conveying speech is presented. The method uses the frequency-specific levels of the speech conveyed by the signal to generate a set of time-varying speech reference levels. It limits the level of sounds conveyed by the signal to these speech reference levels. The method is called speech referenced limiting (SRL). It provides minimal limiting of speech while providing greater control over the loudness of non-speech sounds compared to conventional (fixed threshold) limiters. It is appropriate for use in applications where speech is the primary signal of interest such as telephones, computers, amplified hearing protectors, and hearing aids. The effect of SRL on speech and non-speech sounds is presented.
Convention Paper 8162

10:30 am

P2-3 Individually Adjustable Signal Processing Algorithms for Improved Speech Intelligibility with Cochlear Implants—*Isabell Kiral-Kornek,*[1] *Bernd Edler,*[1] *Jörn Ostermann,*[1] *Andreas Büchner*[2]
[1]Leibniz Universität Hannover, Hannover, Germany
[2]Hörzentrum Hannover, Hannover Medical School, Germany

Thanks to the development of cochlear implants (CIs) the treatment of certain hearing impairments and even deafness has become possible. However, up to now individual adjustments of the speech processing within a cochlear implant are limited to an unequal amplification of different frequency bands. As the perception of speech among patients differs beyond just loudness, improvements concerning individually adjustable signal processing are to be made. A novel approach is being presented that aims to increase the intelligibility for patients by extending common speech recognition tests to allow for an optimized speech processing tailored to the patients' needs.
Convention Paper 8163

11:00 am

P2-4 Objective Evaluation of Wideband Speech Codecs for Voice Communication over Bluetooth—*Gary Spittle,*[1] *Jacek Spiewla,*[2] *Walter Kargus,*[2] *Walter Zuluaga,*[2] *Xuejing Sun*[2]
[1]Cambridge Silicon Radio (CSR), Cambridge, Cambs., UK
[2]Cambridge Silicon Radio (CSR), Detroit, MI, USA

Bluetooth devices that stream audio are becoming increasingly popular. The user expectation has increased and as a result the requirements on wireless audio devices using Bluetooth has produced a number of challenges. This paper discusses the impact on the wideband speech due to various forms of adverse connection conditions. These are measured in terms of speech quality and intelligibility. A detailed understanding of various forms of degradation allows proper solutions to be provided.
Convention Paper 8164

11:30 am

P2-5 Speech Synthesis Controlled by Eye Gazing—*Andrzej Czyzewski, Kuba Lopatka, Bartosz Kunka, Rafal Rybacki, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

A method of communication based on eye gaze controlling is presented. Investigations of using gaze tracking have been carried out in various context applications. The solution proposed in the paper could be referred to as "talking by eyes" providing an innovative approach in the domain of speech synthesis. The application proposed is dedicated to disabled people, especially to persons in a so-called locked-in syndrome who cannot talk and move any part of their body. The paper describes a methodology of determining the fixation point on a computer screen. Then it presents an algorithm of concatenative speech synthesis used in the solution engineered. An analysis of working with the system is provided. Conclusions focusing on system characteristics are included.
Convention Paper 8165

12:00 noon

P2-6 Voice Samples Recording and Speech Quality Assessment for Forensic and Automatic Speaker Identification—*Andrey Barinov*, Speech Technology Center Ltd., Saint Petersburg, Russia

The task of speaker recognition or speaker identification becomes very important in our digital world. Most of the law enforcement organizations use either automatic or manual speaker identification tools for investigation processes. In any case, before carrying out the identification analysis, they usually need to record a voice sample from the suspect either for one to one comparison or to fill in the database. In this paper we describe the parameters of speech signal that are important for ➥

speaker identification performance, we propose the approaches of quality assessment, and provide the practical recommendations of taking the high quality voice sample, acceptable for speaker identification. The materials of this paper might be useful for both soft/hardware developers and forensic practitioners.
Convention Paper 8166

Tutorial 1 **Thursday, November 4**
9:30 am – 10:30 am **Room 132**

iPHONE SOUND DESIGN—LESSONS LEARNED

Presenter: **Jeff Essex**, Audiosyncrasy, Albany, CA, USA

This one-hour session will be a comprehensive review of factors to be considered when creating audio for the iPhone, as well as strategies and tools for getting the best audio performance. Topics to be covered include:

- Audio architecture overview
- Using the s/w and h/w channels (compressed music and .caf SFX)
- Designing for the speaker as well as headphones
- Frequency characteristics compared: iPod touch, iPad, iPhone 3x, iPhone 4
- Handy tools: TTW, SoundConverter, real time analyzers
- Case study: iRingPro, and using GarageBand to create ringtones
- 3rd party audio middleware: fmod
- Product demos

Workshop 1 **Thursday, November 4**
9:30 am – 11:15 am **Room 206**

APPLICATIONS OF TIME-FREQUENCY PROCESSING IN SPATIAL AUDIO

Chair: **Ville Pulkki**, Aalto University School of Science and Technology, Helsinki, Finland

Panelists: *Christof Faller*, Illusonic LLC, Lausanne, Switzerland
Aki Harma, Philips Research Laboratories
Jean-Marc Jot, DTS Inc.

The time-frequency resolution of human hearing has been taken into account for a long time in perceptual audio codecs. Recently, the spatial resolution of humans has been exploited in time-frequency processing as well. This has already led to some commercial applications. This workshop covers the capabilities and incapabilities of human spatial hearing and the audio techniques that exploit these features. Typically the techniques are based on the estimation of directional information for each auditory frequency channels, which information is then used in further processing. The application areas discussed in the workshop include audio coding, microphone techniques, upmixing, directional microphones, and studio effects.

Broadcast/Media Streaming Session 1
Thursday, November 4 **9:30 am – 11:00 am**
Room 133

BROADCAST FACILITY DESIGN: ATTENDING TO THE DETAILS

Chair: **John Storyk**, Walters-Storyk Design Group

Presenters: *Keith Hanadel*, HLW Broadcast Facility Design
Bill Jarett, Food Network
Jim Servies, LA ESPN
Bob Skye

Architect/Acoustician John Storyk, co-principal Walters-Storyk Design Group, will chair a blue-ribbon panel focused on the innumerable details that must be addressed when designing or upgrading a broadcast production/post-production facility. Panelists include: Food Network VP of Engineering Bill Jarett; leading SF-based acoustician Bob Skye; Keith Hanadel, architect/ project manager, HLW Broadcast Facility Design; and a TV program producer/ facility user. Among the topics to be covered are: facilitating work flow via intelligent systems design; determining and achieving exact acoustic requirements; a variety of real world facility design specific issues; and, the end-users' perspective—working the room.

Live Sound Seminar 1 **Thursday, November 4**
9:30 am – 11:15 am **Room 131**

THE GREENING OF LIVE AUDIO FOR MEDIUM AND SMALL OPERATORS

Chair: **Tony Tissot**, 4dB Sound

Presenters: *Bill Gelow*, VP Engineering, Bosch/Electrovoice
Pat Quilter, Senior Amplifier Designer, QSC Audio Products
David Scheirman, Tour Sound Director, JBL Professional
Jeff Touzeau, Author of *The Green Musician's Guide*, President of Hummingbird Media, Inc.
Noah Waldron, Capsicum Pro Audio & Visual

How can smaller live audio providers embrace practices that are environmentally sound? What are the latest Earth-friendly methods and the most conscientious equipment choices? What options exist for lessened environmental impact through reduced power draw, decreased transportation, and labor costs?

Workshop 2 **Thursday, November 4**
9:45 am – 11:45 am **Room 120**

STANDARDS FOR MULTICHANNEL AUDIO DISTRIBUTION

Chair: **Veronique Larcher**, Sennheiser Research

Panelists: *Peter Jax*, Peter Jax, Technicolor, Research, & Innovation
Rozenn Nicol, France Télécom R&D
Nils Peters, CNMAT, ICSI, UC Berkeley
Jack Vad, San Francisco Symphony
Wilfried van Baelen, Galaxy Studios

The case of 7.1 audio distribution seems to be well covered with Blu-ray disks and Dolby TrueHD or DTS-HD formats. But how should this audio content be streamed? What mechanisms are in place to play it back on mobile platforms? Or to broadcast it to stadiums? To the home? The benefits offered by more channels and specifically by surround sound with height are gaining traction in the car industry. What benefits exactly? Will the channel inflation ever stop? The video industry has made progress toward ubiquitous high-definition and 3-D formats. What are their constraints to combine their video content to our multichannel audio? This workshop will gather practitioners from these application fields and collect everyone's hopes and constraints towards the next multichannel audio distribution standard.

Thursday, November 4 10:00 am Room 113
Technical Committee Meeting on Audio for Games

Thursday, November 4 11:00 am Room 113
Technical Committee Meeting on Archiving, Restoration, and Digital Libraries

Master Class 1 Thursday, November 4
11:30 am – 1:00 pm Room 206

PERCEPTION / EVALUATION OF AUDIO

Presenter: **David Griesinger**, Cambridge, MA, USA

This master class focuses on performance acoustics in venues large and small. We will demonstrate a new neural mechanism that can detect the direct sound (DS) as separate from reverberation. The DS is vital to creating and holding the undivided attention of a listener. We will show how the perception of DS in a strong reverberant field depends on frequency, the direct to reverberant ratio, and the time delay between DS and the reverberant energy. The implications for listening rooms and hall design will be discussed. Some conclusions: listening rooms benefit from directional loudspeakers, small concert halls should not have a shoe-box shape, early lateral reflections are not necessarily beneficial, and electronic enhancement of late reverberation may be vital in small halls.

Tutorial 2 Thursday, November 4
11:30 am – 1:00 pm Room 130

EQUALIZATION—ARE YOU GETTING THE MOST OUT OF THIS HUMBLE EFFECT?

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

Track by track, mix by mix, we reach for equalization constantly. Easy at first, EQ becomes more intuitive when you have a deep understanding of the parameters, types, and technologies used—plus deep knowledge of the spectral content of the most common pop and rock instruments. Alex Case offers a routine for applying EQ and strategies for its use: fixing problems, enhancing features, fitting the spectral pieces together, and more.

Broadcast/Media Streaming Session 2
Thursday, November 4 11:30 am –1:00 pm
Room 133

INNOVATIONS IN DIGITAL TV

Chair: **Jerry Whitaker**, ATSC

Panalists: *Tim Carroll*, Linear Acoustic
Sterling Davis, Cox Communications
David Layer, NAB
Geir Skaaden, DTS Inc.
David Wilson, CEA

With the transition to digital television in North America well behind us, the various elements of the broadcast-to-consumer chain continue to look for ways to improve the service. Recent developments include technologies such as non-real-time delivery of program material and Internet-enabled television sets. Cooperative efforts are underway to develop new services while at the same time preserving the legacy services enjoyed by millions of consumers. This session will examine work currently underway to advance digital television to the next level, including concepts, options, and possible timelines.

Product Design Session 1 Thursday, November 4
11:30 am – 1:00 pm Room 132

IS YOUR EQUIPMENT DESIGN A NOISE PROBLEM WAITING TO HAPPEN?

Presenter: **Bill Whitlock**

A design goal for all audio equipment is freedom from hum and buzz. But AC power normally creates a system environment of ground voltage differences. While a balanced interface is the first line of defense against this noise source, the balanced interface itself is very poorly understood by most engineers. This leads them to design balanced input circuits that perform impressively in the lab but have very poor noise rejection in real-world systems. To make matters worse, internal equipment grounding schemes are often thoughtlessly designed. Two common results are noise coupled via cable shield connections (the "Pin 1" problem) and the AC power cord (so-called "sensitive" equipment). These and other design pitfalls, and how to avoid them, are the focus of this class.

Game Audio Session 1 Thursday, November 4
12:00 noon – 1:00 pm Room 120

CODE MONKEY PART 1: WHAT GAME AUDIO CONTENT PROVIDERS NEED TO KNOW ABOUT C++ PROGRAMMING

Presenter: **Peter "pdx" Drescher**, Sound Designer, Twittering Machine

It's like Jane Goodall and the chimps—learn the ways of programmers and they will let you into their group. Being able to speak "the language" is helpful not only in communicating your ideas to them, but also in tracking down implementation bugs and understanding what "exactly" your programmer has been saying all these years. The author has written an application in C++ / Objective-C for Mac OSX that plays FMOD Designer's Interactive Music examples and will use it to illustrate basic programming concepts. The source code and Xcode project will be available for download.

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Thursday, November 4, 1:00 pm – 2:30 pm
Room 134

Opening Remarks:
- Executive Director Roger Furness
- President Diemer de Vries
- Convention Cochairs Jim McTigue, Valerie Tyler

Program:
- AES Awards Presentation
- Introduction of Keynote Speaker
- Keynote Address by Bob Margouleff

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:

PUBLICATIONS AWARD

- **Stefan Feistel,**
- **Ambrose Thompson**
- **Wolfgang Ahnert**

BOARD OF GOVERNORS AWARD

- **Eddy B. Brixen**
- **Joel Vieira De Brito**
- **Josh Reiss**
- **Agnieszka Roginska**
- **John Strawn**
- **Alan Trevena**
- **Valerie Tyler**

FELLOWSHIP AWARD

- **Steven Green**
- **Francisco Miranda Kirchner**

DISTINGUISHED SERVICE MEDAL

- **Ron Streicher**

Keynote Speaker

Veteran engineer/producer and studio owner Bob Margouleff will present the Keynote Address at AES 129th Convention. Co-winner (with Malcolm Cecil) of the 1974 Best Engineered Album Non-Classical Grammy for Stevie Wonder's *Innervisions*, Margouleff has worked with such major artists as: Devo, Marvin Gaye, Martha Reeves, The Isley Brothers, and of course, Stevie Wonder. He has co-owned Mi Casa Multimedia Studios in Hollywood with partner Brant Biles since the 1990s. The studio specializes in 5.1 soundtrack restoration for feature film DVD and Blu-ray release. Recent credits include: *The Sound of Music*, all the *James Bond* and *Lord of the Rings* films.

Margouleff's keynote address is entitled: "What the Hell Happened?"

Margouleff plans to examine the influence of fast-paced technological developments on creativity—both for better and for worse—and the creative artist's needs in a technological world. "Yes, we've had a digital revolution in music and film, shifting power away from a few big companies and toward greater access," says Margouleff. "But we must not forget the importance of collaboration among talented artists, engineers, and business people who built an entertainment industry that changed the world."

Session P3 Thursday, Nov. 4
2:30 pm – 6:30 pm Room 236

ACOUSTICAL MEASUREMENTS

Chair: **John Vanderkooy**, University of Waterloo, Waterloo, Ontario, Canada

2:30 pm

P3-1 Methods for Extending Room Impulse Responses Beyond Their Noise Floor—*Nicholas J. Bryan, Jonathan S. Abel*, Stanford University, Stanford, CA, USA

Two methods of extending measured room impulse responses below their noise floor and beyond their measured duration are presented. Both methods extract frequency-dependent reverberation energy decay rates, equalization levels, and noise floor levels, and subsequently extrapolate the reverberation decay toward silence. The first method crossfades impulse response frequency bands with a late-field response synthesized from Guassian noise. The second method imposes the desired decay rates on the original impulse response bands. Both methods maintain an identical impulse response prior to the noise floor arrival in each band and seamlessly transition to a natural sounding decay after the noise floor arrival.
Convention Paper 8167

3:00 pm

P3-2 On the Use of Ultrasound Transducer Arrays to Account for Time-Variance on Room Acoustics Measurements—*Joel Paulo*,[1,2] *José Bento Coelho*[2]
[1]ISEL- Instituto Superior de Engenharia de Lisboa, Lisbon, Portugal
[2]CAPS – Instituto Superior Técnico, TU Lisbon, Lisbon, Portugal

In real room acoustical measurements, the assumption of time-invariant system is usually not verified. A measurement technique was set-up with the purposes of monitoring the acoustical media, searching for time variance phenomena, and for low SNR situations. A probe test signal in the ultrasonic band is sent to the room by using a parametric loudspeaker array, with high polar pattern directivity, simultaneous with the test signal frames. The relevant parameters to establish time-variance and associated thresholds are then estimated from the acquired ultrasonic sound. The valid test signal frames, which pass the thresholds test, are labeled with a weighting factor depending on its significance. Otherwise, the frames are rejected, not entering on the averaging process. Results are presented and discussed herein.
Convention Paper 8168

3:30 pm

P3-3 Impulse Response Measurements in the Presence of Clock Drift—*Nicholas J. Bryan, Miriam A. Kolar, Jonathan S. Abel*, Stanford University, Stanford, CA, USA

There are many impulse response measurement scenarios in which the playback and recording devices maintain separate unsynchronized digital clocks resulting in clock drift. Clock drift is problematic for impulse response measurement techniques involving convolution, including sinusoidal sweeps and pseudo-random noise sequences. We present analysis of both a drifting record clock and playback clock, with a focus on swept sinusoids. When using a sinusoidal sweep without accounting for clock drift, the resulting impulse response is seen to be convolved with an allpass filter having the same frequency trajectory form as the input swept sinusoid with a duration proportional to the input sweep length. Two methods are proposed for estimating the clock drift and compensating for its effects in producing an impulse response measurement. Both methods are shown to effectively eliminate any clock effects in producing room impulse response measurements.
Convention Paper 8169

4:00 pm

P3-4 Quasi-Anechoic Loudspeaker Measurement Using Notch Equalization for Impulse Shortening—*Richard Stroud*, Stroud Audio Inc., Kokomo, IN, USA

The length of the impulse response of a typical piston driver is largely determined by the characteristic second-order high-pass response of the driver. This time response makes anechoic (i.e., gated) measurement difficult in non-anechoic environments, as reflections must be suppressed to returns of 30 ms. or more. This paper outlines a quasi-anechoic frequency and phase response modification technique using a tuned notch, or band-cut, equalization that shortens the impulse response and allows correct full-range loudspeaker measurement in moderately sized non-anechoic rooms.
Convention Paper 8170

4:30 pm

P3-5 Estimating Room Impulse Responses from Recorded Balloon Pops—*Jonathan S. Abel, Nicholas J. Bryan, Patty P. Huang, Miriam A. Kolar, Bissera V. Pentcheva*, Stanford University, Stanford, CA, USA

Balloon pops are convenient for probing the acoustics of a space, as they generate relatively uniform radiation patterns and consistent “N-wave” waveforms. However, the N-wave spectrum contains nulls that impart an undesired comb-filter-like quality when the recorded balloon pop is convolved with audio. Here, a method for converting recorded balloon pops into full bandwidth impulse responses is presented. Rather than directly processing the balloon pop recording, an impulse response is synthesized according to the echo density and frequency band energies estimated in running windows over the balloon pop. Informal listening tests show good perceptual agreement between measured room impulse responses using a loudspeaker source and a swept sine technique and those derived from recorded balloon pops.
Convention Paper 8171

5:00 pm

P3-6 Complex Modulation Transfer Function and its Applications in Transducer and Room Acoustics Measurements—*Juha Backman*, Nokia Corporation, Espoo, Finland

Modulation transfer function in audio applications describes well the clarity of sound, but conventional definitions and measurement methods are not easily applicable to transducer measurements, low-frequency acoustics, or capturing effects of narrow-band phenomena. A revised definition of modulation transfer function, taking into account the magnitude and phase of modulation transfer for each carrier and modulator frequency combination is presented. This function is derived from the complex frequency response by analyzing the response at the carrier frequency and at the modulation sidebands. Also the distortion of modulation envelope arising from the asymmetry especially in the phase transfer properties is discussed. Examples of the use of the complex modulation transfer function are presented for simple filters, anechoic response measurements of loudspeakers, and for loudspeakers in rooms.
Convention Paper 8172

5:30 pm

P3-7 Practical Implementation of Perceptual Rub & Buzz Distortion and iExperimental Results—*Steve Temme, Pascal Brunet, Brian Fallon*, Listen, Inc., Boston, MA, USA

In a previous paper [1], we demonstrated how an auditory perceptual model based on an ITU standard can be used to detect audible Rub & Buzz defects in loudspeakers using a single tone stimulus. In this paper we demonstrate a practical implementation using a stepped sine sweep stimulus and present detailed experimental results on loudspeakers including comparison to human listeners and other perceptual methods.
Convention Paper 8173

6:00 pm

P3-8 Measurement of Turbulent Air Noise Distortion in Loudspeaker Systems—*Wolfgang Klippel, Robert Werner*, Klippel GmbH, Dresden, Germany

Air leaks in the dust cap and cabinets of loudspeakers generate turbulent noise that highly impairs the perceived sound quality as rub and buzz and other loudspeaker defects do. However, traditional measurement techniques often fail in the detection of air leaks because the noise has a large spectral bandwidth but a low power density and similar spectral properties as ambient noise generated in a production environment. The paper models the generation process of turbulent air noise and develops a novel measurement technique based on asynchronous demodulation and envelope averaging. The technique accumulates the total energy of the leak noise radiated during the measurement interval and increases the sensitivity by more than 20 dB for measurement times larger than 1s. The paper also presents the results of the practical evaluation and discusses the application to end-of-line testing.
Convention Paper 8174

Session P4 Thursday, Nov. 4
2:30 pm – 5:00 pm Room 220

LOUDNESS AND DYNAMICS

Chair: **Brett Crockett**, Dolby Laboratories, San Francisco, CA, USA

2:30 pm

P4-1 The Loudness War: Background, Speculation, and Recommendations—*Earl Vickers*, STMicroelectronics, Inc., Santa Clara, Ca, USA

There is growing concern that the quality of commercially distributed music is deteriorating as a result of mixing and mastering practices used in

the so-called "loudness war." Due to the belief that "louder is better," dynamics compression is used to squeeze more and more loudness into the recordings. This paper reviews the history of the loudness war and explores some of its possible consequences, including aesthetic concerns and listening fatigue. Next, the loudness war is analyzed in terms of game theory. Evidence is presented to question the assumption that loudness is significantly correlated to listener preference and sales rankings. The paper concludes with practical recom- mendations for de-escalating the loudness war.
Convention Paper 8175

3:00 pm

P4-2 Subjective Evaluation of Gating Methods for Use with the ITU-R BS.1770 Loudness Algorithm—*Scott Norcross, Michel Lavoie*, Communications Research Centre, Ottawa, Ontario, Canada

Loudness measurements using ITU-R Recommendation BS.1770 can be biased downward relative to the perceived loudness level when periods of silence and/or low level signals are present in the program being measured. To address this, it has been proposed that some form of gating be added to the loudness algorithm. To evaluate various gating methods, a formal subjective test was conducted to measure the subjective loudness of broadcast material. The results of the subjective test were used to assess the performance of the gating technique proposed by the EBU P/LOUD expert group on loudness. The study further explored the effect of gating threshold and analysis window size on the accuracy of the objective measurement. While the use of gating did improve the accuracy of the loudness algorithm no single combination could be found that satisfied all scenarios.
Convention Paper 8176

3:30 pm

P4-3 Comparing Continuous Subjective Loudness Responses and Computational Models of Loudness for Temporally Varying Sounds—
Sam Ferguson,[1] *Densil Cabrera*,[2] *Emery Schubert*[1]
[1]University of New South Wales, Sydney, NSW, Australia
[2]The University of Sydney, Sydney, NSW, Australia

There are many ways in which loudness can be objectively estimated, including simple weighted models based on physical sound level, as well as complex and computationally intensive models that incorporate many psychoacoustical factors. These complex models have been generated from principles and data derived from listening experiments using highly controlled, usually brief, artificial stimuli; whereas the simple models tend to have a real world emphasis in their derivation and validation. Loudness research has recently also focused on estimating time-varying loudness, as temporal aspects can have a strong effect on loudness. In this paper continuous subjective loudness responses are compared to time-series outputs of loudness models. We use two types of stimuli: a sequence of sine tones and a sequence of band-limited noise bursts. The stimuli were analyzed using a variety of loudness models, including those of Glasberg and Moore, Chalupper and Fastl, and Moore, Glasberg and Baer. Continuous subjective responses were obtained from 24 university students, who rated loudness continuously in time over the period of the experiment, while using an interactive interface.
Convention Paper 8177

4:00 pm

P4-4 Measuring Dynamics: Comparing and Contrasting Algorithms for the Computation of Dynamic Range—*Jon Boley*,[1] *Michael Lester*,[2] *Christopher Danner*[3]
[1]LSB Audio LLC, Lafayette, IN, USA
[2]Shure Incorporated, Niles, IL, USA
[3]University of Miami, Coral Gables, FL, USA

There is a consensus among many in the audio industry that recorded music has grown increasingly compressed over the past few decades. Some industry professionals are concerned that this compression often results in poor audio quality with little dynamic range. Although some algorithms have been proposed for calculating dynamic range, we have not been able to find any studies suggesting that any of these metrics accurately represent any perceptual dimension of the measured sound. In this paper we review the various proposed algorithms and compare their results with the results of a listening test. We show that none of the tested metrics accurately predict the perceived dynamic range of a musical track, but we identify some potential directions for future work.
Convention Paper 8178

4:30 pm

P4-5 Dynamic Range Control for Audio Signals Using Fourth-Order Level Estimation—*Qing Yang, John Harris*, University of Florida, Gainesville, FL, USA

The human auditory system has been shown to be more sensitive to transient signals than stationary signals given the same energy. Conventional second-order measurements based on energy or root-mean-squared value cannot adequately characterize the auditory perception of non-stationary audio signals. A fourth-order dynamic range control (DRC) algorithm is proposed in this paper. The perceptual quality and dynamic range reduction effectiveness are evaluated for both second-order and fourth-order DRC algorithms. Evaluation results show that our proposed fourth-order DRC algorithm offers better balance of perceptual quality and dynamic range reduction than the conventional second-order approach.
Convention Paper 8179

Master Class 2 Thursday, November 4
2:30 pm – 4:30 pm Room 206

HIGH RESOLUTION COMPUTER AUDIO

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

Computers, televisions, and mobile devices are functionally merging and integrating to become easy to set up fun to run systems. Right now, server-like versions can play high-resolution multichannel files from workstations, function as a control center, create loudspeaker crossovers and equalization, as well as perform room correction using all loudspeakers. Data management and processing create these advanced features but systems like these can present issues with processing activity, sample rate conversion, jitter, noise propagation, digital conversion, and interfaces. One encounters discussion about perceptual differences from technical changes that should not affect accuracy nor produce large differences in timing spectra. The class will briefly introduce systems and components, then show potential behavioral artifacts within system parts and describe test methods that might reveal explanations. Then we'll explore process monitoring, buffers, quick locking low jitter clocks, floating conversion environments, jitter from op amps and a test methodology that targets process related intrinsic behavioral problems. Cluster, subtraction and DSP overload tests will be described along with projected human perceptual and mental load that might have audibility or hinder playback involvement. This presentation of an overall background knowledge should encourage studies that are more detailed and it should be helpful with creation and development of very high quality systems.

Tutorial 3 **Thursday, November 4**
2:30 pm – 4:00 pm **Room 132**

HEADPHONES, HEADSETS, AND EARPHONES: ELECTROACOUSTIC DESIGN AND VERIFICATION

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

This presentation reviews the basic electroacoustic concepts of gain, sensitivity, sound fields, signals, linear, and non-linear systems for ear-worn devices. The Insertion Gain concept is introduced. The orthotelephonic response is described as a target for both the free and diffuse fields. Equivalent volume and acoustic impedance are defined. Ear simulators and test manikins appropriate for Circum-, Supra-, and Intra-aural earphones are presented. The salient portions of the IEC 60268-4 standard are reviewed and examples are given of the basic measurements: Frequency Response, Distortion, Impedance. The basic concepts of Noise Canceling devices are also presented.

Workshop 3 **Thursday, Nov. 4**
2:30 pm – 4:15 pm **Room 130**

THE FUTURE IS TACTILE: HOW DOES WHOLE BODY VIBRATION AFFECT PERCEPTION OF BINAURAL AUDIO OVER HEADPHONES?

Chair: **Todd Welti**, Harman Internationnal

Panelists: *Clemeth L. Abercrombie*, Artec Consultents Inc.
M. Ercan Altinsoy, Technical University of Dresden
Sungyoung Kim, Yamaha Corporation
Sean Olive, Harman International

Audio playback at moderate to high levels over headphones alone does not recreate tactile sensations of the recorded or simulated environment. When striving for an accurate and compelling experience, inclusion of tactile stimuli can make the playback system more physically accurate; however including accurate tactile stimulation is not trivial. For example, how do you reproduce tactile stimuli accurately, and does it significantly enhance the experience? More specifically, what is the effect of tactile stimuli on perceived spectral response and loudness? What is the effect of timing asynchrony between aural and tactile channels? How do tactile and aural stimuli interact perceptually? Some of the psychoacoustic issues for haptic feedback systems may also be relevant to binaural playback.

Game Audio Session 2 **Thursday, November 4**
2:30 pm – 4:30 pm **Room 120**

ROCK ON! – THE ROCK BAND NETWORK DEMYSTIFIED

Presenter: **Jeff Marshall**, Executive Producer, Rock Band Network, Harmonix

With the popularity of music games like Rock Band there comes opportunity for musicians, engineers, and producers to provide musical content. The Rock Band Network was created to broaden the spectrum of available content and allow your music to be experienced by a unique and untapped market. This in-depth how-to session will give you plenty to chew on, straight from the horse's mouth!

Student Event/Career Development
EDUCATION FORUM PANEL
Thursday, November 4, 2:30 pm – 4:00 pm
Room 131

Moderator: **Alex U. Case**, University of Massachusetts Lowell, Lowell, MA, USA

Panelists: *Nathan Breitling*, Art Institute of CA-San Francisco, San Francisco, CA, USA
Paula Froehle, Tribeca Flashpoint Media Arts Academy, Chicago, IL, USA
Ryan Kleeman, Art Institute of CA-San Francisco, San Francisco, CA, USA
John Murray, Tribeca Flashpoint Media Arts Academy, Chicago, IL, USA
Agnieszka Roginska, New York University, New York, NY, USA
John Storyk, Walters-Storyk Design Group, New York, NY, USA

Creating the Out-of-Class Experience

Audio education has always pushed beyond the scope of the traditional lecture classroom experience. Recent innovations in pedagogy and investments in facilities point to a future of audio curricula rich with cutting edge research, intense hands-on experience, and deep interaction with other disciplines. Educators and students are invited to join the discussion with our panelists as they present their benchmark work in creating a compelling out-of-class experience.

"Production In Action: An Innovative Educational Approach to Collaboration with the Industry" will be led by Paula Froehle, Academic Dean, Tribeca Flashpoint Media Arts Academy, Chicago, and John Murray, Recording Arts, Tribeca Flashpoint Media Arts Academy, Chicago.

"Studios and Labs for Future Music Technologists" will be led by Agnieszka Roginska and Paul Geluso, Music Technology, New York University, and John Storyk, Architect/Acoustician/ Co-Principal, Walters-Storyk Design Group, New York, NY, USA. ➥

"Summative Assessment in the Studio: Building a Curriculum That Is More than the Sum of its Parts" will be led by Nathan Breitling, D.M.A., Director, Audio Production, AiCA-San Francisco and Ryan Kleeman, MFA, Faculty, Audio Production, AiCA-San Francisco.

Session P5 **Thursday, Nov. 4**
3:00 pm - 4:30 pm **Room 226**

POSTERS: EMERGING APPLICATIONS

3:00 pm

P5-1 A Robust Audio Feature Extraction Algorithm for Music Identification—*Jiajun Wang*,[1] *Marie-Luce Bourguet*[2]
[1]Beijing University of Posts and Telecommunications, Beijing, China
[2]Queen Mary University of London, London, UK

In this paper we describe a novel audio feature extraction method that can effectively improve the performance of music identification under noisy circumstances. It is based on a dual box approach that extracts from the sound spectrogram point clusters with significant energy variation. This approach was tested in a song finder application that can identify music from samples recorded by microphone in the presence of dominant noise. A series of experiments show that under noisy circumstances, our system outperforms current state-of-the-art music identification algorithms and provides very good precision, scalability, and query efficiency.
Convention Paper 8180

3:00 pm

P5-2 The Low Complexity MP3-Multichannel Audio Decoding System—*Hyun Wook Kim, Han Gil Moon*, Samsung Electronics, Suwon, Korea

In this paper a low complexity MP3 multichannel audio system is proposed. Utilizing the proposed decoding system, the advanced multichannel MP3 decoder can play high quality multichannel audio as well as the legacy stereo audio with low processing power. The system mainly consists of two parts, one of which is an MP3 decoding part and the other one a parametric multichannel decoding part. The transform domain convolution-synthesis method is equipped to replace the PQMF module in the MP3 decoding part and several small point DFT modules instead of the large point DFT module used in the multichannel decoding part. This combination can reduce computing power dramatically without any loss of decoded audio signal.
Convention Paper 8181

3:00 pm

P5-3 The hArtes CarLab: A New Approach to Advanced Algorithms Development for Automotive Audio—*Stefania Cecchi,*[1] *Andrea Primavera,*[1] *Francesco Piazza*,[1] *Ferruccio Bettarelli,*[2] *Emanuele Ciavattini*,[2] *Romolo Toppi*,[3] *Jose Gabriel De Figueiredo Coutinho,*[4] *Wayne Luk*,[4] *Christian Pilato,*[5] *Fabrizio Ferrandi*,[5] *Vlad M. Sima,*[6] *Koen Bertels*[6]
[1]Università Politecnica delle Marche, Ancona, Italy
[2]Leaff Engineering, Ancona, Italy
[3]FAITAL S.p.a., Milano, Italy
[4]Imperial College London, London, UK
[5]Politecnico di Milano, Milano, Italy
[6]Delft University of Technology, Delft, The Netherlands

In the last decade automotive audio has been gaining great attention by the scientific and industrial community. In this context, a new approach to test and develop advanced audio algorithms for a heterogeneous embedded platform has been proposed within the European hArtes project. A real audio laboratory installed in a real car (hArtes CarLab) has been developed employing professional audio equipment. The algorithms can be tested and validate on a PC exploiting each application as a plug-in of a real time framework. Then a set of tools (hArtes Toolchain) can be used to generate code for the embedded platform starting from plug-in implementation. An overview of the entire system is here presented, showing its effectiveness.
Convention Paper 8182

Convention Paper 8183 has been withdrawn

3:00 pm

P5-4 Real-Time Speech Visualization System for Speech Training and Diagnosis—*Yuichi Ueda, Tadashi Sakata, Akira Watanabe*, Kumamoto University, Kumamoto-shi, Japan

We have been interested in visualizing speech information to observe speech phenomena, analyze speech signals, and substitute the hearing disorders or the speech disorders. In order to realize such speech visualization, we have developed a software tool, Speech-ART, and utilized it in investigating speech. Although the functional advantages of system have been effective in offline operation, the use of a speech training tool or real-time observation of speech sound has been restricted. Consequently, we have increased efficiency in analyzing speech parameters and displaying speech image, and then developed a real-time speech visualizing system. In this paper we describe the background of speech visualization, the characteristics of our system, and the applications of the system in the future.
Convention Paper 8184

3:00 pm

P5-5 Underdetermined Binaural 3-D Sound Localization of Simultaneous Active Sources—*Martin Rothbucher, David Kronmüller, Hao Shen, Klaus Diepold*, Technische Universität München, Munich, Germany

Mobile robotic platforms are equipped with multimodal human-like sensing, e.g., haptic, vision, and audition, in order to collect data from the environment. Recently, robotic binaural hearing approaches based on Head-Related Transfer Functions (HRTFs) have become a promising technique to localize sounds in a three-dimensional environment with only two microphones. Usually, HRTF-based sound localization approaches are restricted to one sound source. To cope with this difficulty, Blind Source Separation (BSS) algorithms were utilized to separate the sound sources before applying HRTF localization. However,

those approaches usually are computationally expensive and restricted to sparse and statistically independent signals for the underdetermined case. In this paper we present underdetermined sound localization that utilizes a super-positioned HRTF database. Our algorithm is capable of localizing sparse, as well as broadband signals, whereas the signals are not statistically independent.
Convention Paper 8185
Paper presented by Tim Habigt, TU Munchen, Munich, Germany

3:00 pm

P5-6 Wireless Multisensor Monitoring of the Florida Everglades: A Pilot Project—*Colby Leider*,[1] *Doug Mann*,[2] *Dan Dickinson*[1]
[1]University of Miami, FL, USA
[2]Peavey Electronics Corporation, Meridian, MS, USA

Prior work (e.g., Calahan 1984; Havstad and Herrick 2003) describes the need for long-term ecological monitoring of environmental data such as surface temperature and water quality. Newer studies by Maher, Gregoire, and Chen (2005) and Maher (2009, 2010) motivate the value in similarly documenting natural sound environments in U.S. national parks on the order of a year. Building on these ideas we describe a new system capable of combined remote audio and environmental monitoring on the order of multiple years that is currently being tested in the Florida Everglades.
Convention Paper 8186

Broadcast/Media Streaming Session 3
Thursday, November 4 3:00 pm – 4:30 pm
Room 133

LIP SYNC ISSUE

Chair: **Jonathan S. Abrams**, CBNT, Chief Technical Engineer, Nutmeg Post

Panelists: *Paul Briscoe*, Manager, Strategic Engineering, Harris Broadcast Communications Division
Dan Desmet, Flanders Scientific, Inc.
Pat Waddell, Chair of ATSC TSG/S6
Dave Wilson, Sr. Director, Technology and Standards, Consumer Electronics Association

Lip sync remains a complex problem, with several causes and few solutions. From production through transmission and reception there are many points where lip sync can either be properly corrected or made even worse. This session's panel will discuss several key issues. Where do the latency issues exist? How can the latency be measured? What correction techniques exist for controlled environments? How does video display design affect lip sync? Who is responsible for implementing the mechanisms that ensure lip sync is maintained when the signal reaches your television? Join us as our panel addresses these questions and more.

Historical Event
HISTORICAL FILM SOUND AT DOLBY
Thursday, November 4, 3:00 pm – 5:00 pm
Dolby Theatre, San Francisco, CA

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

Stereo Optical Film Soundtracks—76 Years

This two-hour event takes place at the renowned Dolby Laboratories Presentation Studio—a theater, projection room, and recording studio located at Dolby's headquarters built to provide a nearly-perfect listening and viewing environment for film, recorded sound, and live presentations. See and hear actual historical film footage demonstrating the evolution of 35mm stereo optical film soundtracks, from the first experimental recordings made by Alan Blumlein in 1934 through to today's digital soundtracks. The steady progress in technical performance will be described by Ioan Allen, Oscar recipient, AES Fellow, and recipient of the AES Silver Medal award. Mr. Allen spearheaded the introduction of many breakthrough film audio formats that have revolutionized the film sound experience.

Tickets: A limited number of $10 tickets will be available exclusively to registered convention attendees at the tours booth at Moscone. The marked bus will begin loading at 2:30 pm for the short ride to Dolby.

Thursday, November 4 3:00 pm Room 113
Technical Committee Meeting on Audio for Telecommunications

Thursday, November 4 3:00 pm Room 232
Standards Committee Meeting SC-02-02 Digital Input/Output Interfacing

Thursday, November 4 4:00 pm Room 113
Technical Committee Meeting on Audio Forensics

Tutorial 4 Thursday, November 4
4:30 pm – 6:30 pm Room 132

DO-IT-YOURSELF SEMANTIC AUDIO

Presenter: **Jörn Loviscach**, University of Applied Sciences Bielefeld, Bielefeld, Germany

How do you tell music files from speech files? How do you find similar-sounding tracks in an unlabeled collection? Content-based music information retrieval (MIR) and similar applications require advanced algorithms that often overburden non-expert developers. However, many building blocks are available for free and in ready-to-use packaging to significantly ease software development, for instance of similarity search methods, or to serve as components for ad-hoc solutions, for instance in forensics or linguistics. This tutorial looks into a variety of software that helps with the extraction of audio features and/or executes machine learning algorithms. Focusing on solutions that require little to no programming in the classical sense, the tutorial's major part consists in live demos of hand-picked routes to roll one's own semantic audio application.

Tutorial 5 Thursday, November 4
4:30 pm – 6:30 pm Room 130

IMMERSAV—"INFINITE-CHANNEL" SURROUND SOUND WITH HD VIDEO—A NEW ENTERTAINMENT FORMAT

Presenter: **Robert B. Schulein**, Asius Technologies, Longmont, CO, USA

In previous tutorials, AES125 and 126 in San Francisco and Munich, the essential elements of binaural hearing, recording, and playback were presented from a historical, current practice, and future trends perspective. One future trend presented the entertainment potential derivable from the synergy of binaural audio and high definition video. Unlike traditional audio recordings, often experienced with ones eyes closed, visual image cues directly related to an audio recording are readily observed to heighten the perceived spatial accuracy of the audio experience. In recognition of the abundance of entertainment audio and video being experienced with earphones connected to personal media players and computers, a scenario can be made for producing entertainment in this fashion. What can result is a "you are there," infinite-multichannel surround audio format with high definition video. The focus of this tutorial is to present the elements of creating such productions from an artistic, and technical perspective. Of particular importance, are the considerations given to the acoustic space, the music and the musician. A range of production examples will be presented supported by a variety of headphone and high definition video playback systems.

Tutorial 6 Thursday, November 4
4:30 pm – 6:30 pm Room 131

MANAGING TINNITUS AS A WORKING AUDIO PROFESSIONAL

Presenters: **Neil Cherian**, Cleveland Clinic, Cleveland, OH, USA
Michael Santucci, Sensaphonics Hearing Conservation, Chicago, IL, USA

Tinnitus is a common yet poorly understood disorder where sound is perceived in the absence of an external source. Significant sound exposure, with or without hearing loss, is the most common risk factor. Tinnitus can be debilitating and can impair quality of life. Anxiety, depression, and sleep disorders are potential consequences. Most importantly for those in the audio industry, it can significantly impair auditory perception.

This tutorial will focus on methods in managing tinnitus in the life of an audio professional. Background information will be provided regarding the basic concept of tinnitus, pertinent anatomy and physiology, audiologic parameters of tinnitus, and an overview of current research. Suggestions for identifying and mitigating high risk behaviors will be covered. Elements of medical and audiologic evaluations of tinnitus will also reviewed.

Broadcast/Media Streaming Session 4
Thursday, November 4 5:00 pm – 6:00 pm
Room 133

CASE STUDY OF PUNGANET: UNITING RADIO STATIONS ACROSS A COUNTRY

Presenters: **Kirk Harnack**, Telos
Igor Zukina, AVC-Group

The traditional, one-to-many audio distribution model is ineffective. Indeed, it's only half a solution when affiliates each have contributions for all or parts of the organization. Consider that a network of related radio stations has resources that are likely geographically widespread. A Central Router Management System provides a management cloud from which individual affiliates may choose published resources and may publish their own talent and programming assets. PungaNet is an ingenious suite of mostly off-the-shelf, IP-connected technologies that enables many simultaneous network topologies, each on a scheduled or ad hoc basis, for distributing and sharing audio, control, metadata, and broadcast business processes over standard IP infrastructure.

Game Audio Session 3 Thursday, November 4
5:00 pm – 6:30 pm Room 206

THE WIDE WONDERFUL WORLD OF 5.1 ORCHESTRAL RECORDINGS

Presenters: **Richard Dekkard**, Director, Orphic Media LLC
Tim Gedemer, Owner/Supervising Sound Editor, Source Sound Inc

When recording an orchestra was a simple affair using two or three microphones, the performance, the choice, and placement of said microphones, and the quality of the recording medium were all that factored into the result. These days, orchestral recording takes almost as many forms as pop recording. Spot mics, multichannel arrays, postproduction, and editing are all used in the production process. In this panel, experts in both game and film audio will be discussing the means by which producers and engineers arrive at their final goals for different formats and deal with the challenges of 5.1 orchestral recording for their mediums. Topics will include the different footprint limits in the 5.1 format used for games versus the film format—as well as the those involved in streaming bandwidth for both games and movies. Panelists will go over editing in 5.1 for games to accommodate player-driven music as compared to the linear progression standard in film editing, and will also discuss the lack of standards for 5.1 in games versus the established process and standards for film.

Student Event/Career Development
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Thursday, November 4, 5:00 pm – 6:30 pm
Room 120

Chair: **MeiLing Loo**

Vice Chair: **Philip Parenteau**

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. Election results and Recording Competition Awards will be given at the Student Delegate Assembly Meeting–Part 2 on Sunday, November 7, at 2:30.

Thursday, November 4 5:00 pm Room 113
Technical Committee Meeting on Coding of Audio Signals

Session P6 Friday, Nov. 5
9:00 am – 10:30 am Room 220

MICROPHONE PROCESSING

Chair: **Jon Boley**, LSB Audio, Lafayette, IN, USA

9:00 am

P6-1 Digitally Enhanced Shotgun Microphone with Increased Directivity—*Helmut Wittek*,[1] *Christof Faller*,[2] *Christian Langen*,[1] *Alexis Favrot*,[2] *Christophe Tournery*[2]
[1]SCHOEPS Mikrofone GmbH, Karlsruhe, Germany
[2]Illusonic LLC, Lausanne, Switzerland

Shotgun microphones are still state-of-the-art when the goal is to achieve the highest possible directivity and signal-to-noise ratio with high signal fidelity. As opposed to beamformers, properly designed shotgun microphones do not suffer greatly from inconsistencies and sound color artifacts. A digitally enhanced shotgun microphone is proposed, using a second backward-oriented microphone capsule and digital signal processing with the goal of improving directivity and reducing diffuse gain at low and medium frequencies significantly, while leaving the sound color essentially unchanged. Furthermore, the shotgun microphone's rear lobe is attenuated.
Convention Paper 8187

9:30 am

P6-2 Conversion of Two Closely Spaced Omnidirectional Microphone Signals to an XY Stereo Signal—*Christof Faller*, Illusonic LLC, St-Sulpice, Switzerland

For cost and form factor reasons it is often advantageous to use omni-directional microphones in consumer devices. If the signals of a pair of such microphones are used directly, time-delay stereo with possibly some weak level-difference cues (device body shadowing) is obtained. The result is weak localization and little channel separation. If the microphones are relatively closely spaced, time-delay cues can be converted to intensity-difference cues by applying delay-and-subtract processing to obtain two cardioids. The delay-and-subtract processing is generalized to also be applicable when there is a device body between the microphones. The two cardioids could be directly used as stereo signal, but to prevent low frequency noise the output signals are derived using a time-variant filter applied to the input microphone signals.
Convention Paper 8188

10:00 am

P6-3 Determined Source Separation for Microphone Recordings Using IIR Filters— *Christian Uhle*,[1] Josh Reiss[2]
[1]Fraunhofer IIS, Erlangen, Germany
[2]Queen Mary University of London, London, UK

A method for determined blind source separation for microphone recordings is presented that attenuates the direct path cross-talk using IIR filters. The unmixing filters are derived by approximating the transmission paths between the sources and the microphones by a delay and a gain factor. For the evaluation, the proposed method is compared to three other approaches. Degradation of the separation performance is caused by fractional delays and the directivity of microphones and sources, which are discussed here. The advantages of the proposed method are low latency, low computational complexity, and high sound quality.
Convention Paper 8189
Paper presented by Josh Reiss

Session P7 Friday, Nov. 5
9:00 am – 1:00 pm Room 236

LOUDSPEAKER DESIGN AND AMPLIFIERS

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

9:00 am

P7-1 An Improved Beryllium Dome Diaphragm Assembly for Large Format Compression Drivers—*Marshall Buck*,[1] *Gordon Simmons,*[2] *Sam Saye*[2]
[1]Psychotechnology, Inc., Los Angeles, CA, USA
[2]Brush-Wellman, Fremont, CA, USA

We describe the development, manufacture, and testing of a new large format compression driver diaphragm using a beryllium dome and new type of polymer surround that exhibits improved performance. This design promises to give long life and good reliability with little or no change in performance anticipated over the life of the diaphragm. A comprehensive set of tests of Beryllium, Aluminum, and Titanium diaphragm compression drivers is described including frequency response, distortion, and wavelet time domain analysis on a 2-inch plane wave tube. Substantial differences were measured in the performance categories, particularly in the frequency range above 4 kHz.
Convention Paper 8190

9:30 am

P7-2 Point-Source Loudspeaker Reversely-Attached Acoustic Horn: Improvement of Acoustic Characteristics and Application to Some Measurements—*Takahiro Miura, Teruo Muraoka, Tohru Ifukube*, The University of Tokyo, Tokyo, Japan

It is ideal to measure acoustic characteristics by point- source sound. We proposed, at a previous convention, a point-source measurement loudspeaker that is designed to attach the mouse of hyperbolic horn to the diaphragm of the driver unit. The difference of directional intensity of the loudspeaker at the frequency range of 20 – 700 Hz were within 3 dB at any combination of azimuth and elevation. At the frequency range over 700 Hz, differences of azimuthal directional intensity were within 10 dB while that of the ele- ➥

vational ones were within 20 dB. Following these results, difference of directional frequency characteristics is discussed. Then we applied the loudspeaker for the measurement of acoustic characteristics of a hall.
Convention Paper 8191

10:00 am

P7-3 Ironless Motor Loudspeaker: Quantization of the Subjective Enhanced Sound Quality—
Mathias Remy,[1,2] *Guy Lemarquand*,[1] *Daniele Ceruti*,[3] *Gaël Guyader*,[2] *Romolo Toppi*,[3] *Marc-François Six*[4]
[1]Technocentre Renault, Guyancourt, France
[2]Laboratoire d'Acoustique de l'Université du Maine, Le Mans Cedex, France
[3]Faital S.p.A., Fabbrica Italiana Altoparlanti S.p.A., Donato Milanese (MI), Italy
[4]Hutchinson S.A., Chalette-sur-Loing Cedex, France

This paper presents a set of measurements realized on two automotive loudspeakers. These two loudspeakers have the exact same moving part and suspensions parts but different motors. The first one is equipped with a traditional production model motor made of ferrite and iron whereas the second one gets a prototype of ironless motor made totally of permanent magnets. Blind listening tests performed with these two loudspeakers showed a significant advantage of perceived sound quality for the ironless motor loudspeaker. Several types of measurements have been realized in order to try to quantify and explain this sound quality enhancement. Results are given in this paper.
Convention Paper 8192

10:30 am

P7-4 Air Velocity and Pressure Profiles in the Front of an Electrodynamic Loudspeaker—
Danijel Djurek,[1] *Ivan Djurek*,[2] *Antonio Petosic*[2]
[1]Allesandro Volta Applied Ceramics (AVAC) Laboratoryfor Nonlinear Dynamics, Zlatar Bistrica, Croatia
[2]University of Zagreb, Zagreb, Croatia

Air velocity was recorded in front of an electrodynamic loudspeaker by the use of hot wire anemometric technique. Wire temperature response was detected up to 2 kHz and harmonics were analyzed by the use of the King formula. Near field effects were detected at the *z*-axis distances comparable to the loudspeaker diameter. Extended Greenspan theory was applied to explain measured data. It was stressed the importance of air viscosity in damping of Morse convection in near field regime. Near field effects at distances up to 30 cm were discussed in terms of the Morse convection indicated by the imaginary part of air impedance. According to continuity equation of air flow microphone signal was correlated to the fluid velocity divergence.
Convention Paper 8193

11:00 am

P7-5 New Techniques for Evaluating Audio Amplifiers via Measuring for Induced Wow and Flutter and Differential Phase Distortions—*Ronald Quan*, Ron Quan Designs, Cupertino, CA, USA

In the past, mechanical systems were measured for Wow and Flutter or frequency modulation but not amplifiers. Instead, amplifiers are typically measured for intermodulation and harmonic distortion. A new method for audio amplifier/device performance measures frequency modulation effects and differential phase distortion. Frequency and phase detectors are used to evaluate induced frequency and phase modulation from an amplifier under two conditions. The first condition has a low frequency signal inducing the modulation on a high frequency signal. The second condition has a high frequency AM signal inducing the modulation on a lower frequency signal. Practical design topologies for the new test methods are shown and the results of the new testing methods are tabulated.
Convention Paper 8194

11:30 am

P7-6 Analysis of Two-Pole Compensation in Linear Audio Amplifiers—*Harry Dymond, Phil Mellor*, University of Bristol, Bristol, UK

An analysis of the two-pole compensation technique used in three-stage linear audio amplifiers is presented. An expression for the loop-gain of a linear amplifier incorporating two-pole compensation is derived, allowing the designer to easily select the unity loop-gain frequency and zero location by choosing appropriate values for the compensation components. Also presented is a simulation method that allows the designer to observe an amplifier's closed-loop and loop-gain responses in a single pass without requiring modification to the circuit's feedback path; and two separate modifications to the usual two-pole compensation approach that improve phase margin and significantly enhance negative-rail power-supply rejection ratio.
Convention Paper 8195

12:00 noon

P7-7 A Robust Pseudo-Ternary Modulation Scheme for Filter-Less Digital Class D Amplifiers—*Rossella Bassoli, Carlo Crippa, Germano Nicollini*, ST-Ericsson, Monza Brianza, Italy

This paper presents a new pseudo-ternary modulation scheme for bridge-tied-load digital class D amplifiers that is more robust versus output stage distortions compared to existing ternary modulations. The effects of finite rise and fall times and their mismatches have been introduced for a classical ternary modulation scheme, where a large degradation of the dynamic range can be observed and then extended to reported high performance ternary modulators. It is shown that, even if linearization pulses are inserted to cope with the finite rise/fall time problem, also these modulators are somehow impacted by the edge mismatches.
Convention Paper 8196

12:30 pm

P7-8 Switching/Linear Hybrid Audio Power Amplifiers for Domestic Applications, Part 1: The Class-B•D Amplifier—*Harry Dymond, Phil Mellor*, University of Bristol, Bristol, UK

The analysis, design, and testing of a parallel

switching/linear hybrid audio power amplifier rated at 100 W into 8 ohms are presented. The amplifier employs a hysteretically controlled switching stage and high-bandwidth linear amplifier whose high-gain negative-feedback loop controls the output signal. The majority of the output current is provided by the switching stage, enhancing efficiency. The amplifier's fidelity has been tested with standard commercially available equipment, while efficiency has been evaluated across a very wide range of signal and load conditions using a custom active-load and automated test procedure. The combined fidelity and efficiency test results are analyzed, and the suitability for domestic applications of this amplifier configuration discussed.
Convention Paper 8197

Workshop 4 **Friday, November 5**
9:00 am – 11:00 am **Room 130**

WIRELESS AUDIO STREAMING

Chair: **Gary Spittle**, Audio Consultant

Panelists: *Deepen Sinha*, ATC Labs
David Trainor, CSR

High quality audio is streamed wirelessly using many forms of radio channels. These range from satellite broadcasts, to mobile telephone networks, to Bluetooth ecosystems, and proprietary ultra low latency systems. This workshop will discuss the challenges we face, along with some of the techniques used, in delivering high quality audio over these connections. Audio codecs are an essential component of each radio link. It will be shown how they are adapted for the specific audio source material, radio channel, and receiving device in the system. Furthermore, the impact of interference on the channel will be presented in relation to the codec and how the effects can be minimized.

Workshop 5 **Friday, November 5**
9:00 am – 10:45 am **Room 133**

HOW DOES IT SOUND NOW?
THE EVOLUTION OF AUDIO

Chair: **Gary Gottlieb**, Webster University

Panelists: *Ed Cherney*
Mark Rubel
Elliot Scheiner
Al Schmitt

With 27 Grammy awards between them, panelists Al Schmitt, Elliot Scheiner, Ed Cherney, and Mark Rubel are uniquely qualified to address the issues surrounding quality in audio, the one constant through decades of transitions in our business. Moderator Gary Gottlieb (engineer, author and educator) draws from the old Chet Atkins story with the punch line, "How does it sound now?" as these audio all-stars discuss the methodology employed when confronted with new and evolving technology and how we retain quality and continue to create a product that conforms to our own high standards. This may lead to other conversations about the musicians we work with, the consumers we serve, and the differences and similarities between their standards and our own. How high should your standards be? How should it sound now? How should it sound tomorrow?

Broadcast/Media Streaming Session 5
Friday, November 5 **9:00 am – 10:15 am**
Room 206

AUDIO FOR THE OLYMPIC BROADCAST

Presenters: **Michael Nunan**, CTV
Joshua Tidsbury, CTV

Broadcasting the Vancouver 2010 Olympic Winter Games for Canada's Olympic Broadcast Media Consortium was an exercise in large numbers.

- 17 days
- 2450 hours of programming
- 12 TV channels
- 20 radio stations
- 22 languages
- 1400+ staff
- 7 production control rooms
- 6 studios
- 15 crews
- 21 edit suites and more than 30 editors

"The Olympic Suite" music package produced in-house for the Games contained more than 240 cues. Over 400 animated production elements were designed, mixed, and deployed. More than 40 hours of "Feature" content was pre-produced for in-games use. Tackling the audio for this massive undertaking would be daunting under any circumstances—but to do it in support of the first ever Olympic Winter Games to be produced entirely in 5.1 Surround was remarkable. Join us for a look behind the scenes of an amazing sonic adventure in Vancouver. The presentation will provide some insight into infrastructure, training, work flow, sound design, music production, and much more—from pre-production through to the Closing Ceremony. Presenting this session will be 2 members of the CTV Operations and Engineering group: Michael Nunan and Joshua Tidsbury. Between them, Michael and Josh lived every moment of the Games and are pleased to have the opportunity to share their experiences.

Game Audio Session 4 **Friday, November 5**
9:00 am – 10:00 am **Room 120**

CODE MONKEY PART 2: LUA IS NOT A HAWAIIAN PICNIC—THE BASICS OF SCRIPTING FOR DYNAMIC AUDIO IMPLEMENTATION

Presenter: **Kristoffer Larson**, Audio Manager, WB Games, Seattle, WA, USA

Isn't a script what you use for your VO sessions? Why, yes, little Billy, but in the world of dynamic audio it means something different. Scripting can be the cheese in your excellent sound sandwich. This session will teach you scripting basics and how scripting can enhance your dynamic audio implementation. (Hold the mayo.) LUA is not universally used, but it is common enough that you'll benefit from having a basic understanding of it. Take-home examples will be provided.

Live Sound Seminar 3 **Friday, November 5**
9:00 am – 10:45 am **Room 131**

MEASUREMENT MICROPHONES

Chair: **Ray Rayburn**, K2 Audio

Panelists: *David Josephson*, Josephson Engineering ➥

Noland Lewis, ACO Pacific
Karl Winkler, Lectrosonics

What makes measurement microphones different from regular mics? How do you choose, use, and store one? Type 1 or type 2? Free field or pressure response? Wired or wireless? Learn what the experts have to say.

Product Design Session 2 **Friday, November 5**
9:00 am – 10:30 am **Room 132**

AN OVERVIEW OF AUDIO SYSTEM GROUNDING AND INTERFACING

Presenter: **Bill Whitlock**

Equipment makers like to pretend the problems don't exist, but this tutorial replaces hype and myth with insight and knowledge, revealing the true causes of system noise and ground loops. Unbalanced interfaces are exquisitely vulnerable to noise due to an intrinsic problem. Although balanced interfaces are theoretically noise-free, they're widely misunderstood by equipment designers, which often results in inadequate noise rejection in real-world systems. Because of a widespread design error, some equipment has a built-in noise problem. Simple, no-test-equipment, troubleshooting methods can pinpoint the location and cause of system noise. Ground isolators in the signal path solve the fundamental noise coupling problems. Also discussed are unbalanced to balanced connections, RF interference, and power line treatments. Some widely used "cures" are both illegal and deadly.

Friday, November 5 **9:00 am** **Room 113**
Technical Committee Meeting on Semantic Audio Analysis

Session P8 **Friday, Nov. 5**
9:30 am – 11:00 am **Room 226**

POSTERS: AUDIO PROCESSING—PART 1

9:30 am

P8-1 **Near and Far-Field Control of Focused Sound Radiation Using a Loudspeaker Array—** *Sangchul Ko, Youngtae Kim, Jung-Woo Choi*, Samsung Electronics Co. Ltd., Gyeonggi-do, Korea

In this paper a sound manipulation technique is proposed to prevent unwanted eavesdropping or disturbing others in the vicinity if a multimedia device is being used in a public place. This is capable of realizing the creation of a spatial region having highly acoustic potential energy at the listener's position. For doing so, the paper discusses the design of multichannel filters with a spatial directivity pattern for a given arbitrary loudspeaker array configuration. First some limitations in using conventional beamforming techniques are presented, and then a novel control strategy is suggested for reproducing a desired acoustic property in a spatial area of interest close to the loudspeaker array. This technique also allows us to control an acoustic property in an area relatively far from the array with a single objective function. In order to precisely produce a desired shape of energy distribution in both areas, spatial weighting technique is introduced. The results are compared with those from controlling each area separately.
Convention Paper 8198

9:30 am

P8-2 **A Real-Time Implementation of a Novel Psychoacoustic Approach for Stereo Acoustic Echo Cancellation—***Stefania Cecchi, Laura Romoli, Paolo Peretti, Francesco Piazza*, Università Politecnica delle Marche, Ancona, Italy

Stereo acoustic echo cancellers (SAECs) are used in teleconferencing systems to reduce undesired echoes originating from coupling between loudspeakers and microphones. The main problem of this approach is related to the issue of uniquely identifying each pair of room acoustic paths, due to high interchannel coherence. In this paper a real-time implementation of a novel approach for SAEC based on the psychoacoustic effect of missing fundamental is proposed. An adaptive algorithm is employed to track and remove the fundamental frequency of one of the two channels, ensuring a continuous decorrelation without affecting the stereo quality. Several tests are presented taking into account a real-time implementation on a DSP framework in order to confirm its effectiveness.
Convention Paper 8199

9:30 am

P8-3 **Solo Plucked String Sound Detection by the Energy- to-Spectral Flux Ratio (ESFR)—** *Byung Suk Lee*,[1,2] *Chang-Heon Lee*,[3] *Gyuhyeok Jeong,*[1] *In Gyu Kang*[1]
[1]LG Electronics, Inc., Seocho-Gu, Seoul, Korea
[2]Columbia University, New York, NY, USA
[3]Yonsei University, Seoul, Korea

We address the problem of distinguishing solo plucked string sound from speech. Due to the harmonic components present in both types of signals, a low complexity music/speech classifier often misclassifies these signals. To capture the sustained harmonic structures observed in solo plucked string sound, we propose a new feature, the Energy-to-Spectral Flux Ratio (ESFR). The values and the statistics of the ESFR for solo plucked string sound were distinct from those for speech when calculated over windows of 20 to 50 ms. By building a low complexity detector with the ESFR, we demonstrate the discriminating performance of the ESFR feature for the considered problem.
Convention Paper 8200

9:30 am

P8-4 **Separation of Repeating and Varying Components in Audio Mixtures—***Sean Coffin*, Stanford University, Stanford, CA, USA

A large amount of modern pop music contains digital "loops" or "samples" (short audio clips) that appear multiple times during a song. In this paper a novel approach to separating these exactly repeating component waveforms from the rest of an audio mixture is presented. By examining time-frequency representations of the mixture during several instances of a single

repeating component and taking the complex value for each time-frequency bin with the smallest magnitude across all instances we can effectively extract the content that is perceived to be repeating given that the rest of the mixture varies sufficiently. Results are presented demonstrating successful application to commercially available recordings as well as to constructed audio mixtures achieving signal to interference ratios up to 42.8 dB.
Convention Paper 8201

9:30 am

P8-5 High Quality Time-Domain Pitch Shifting Using PSOLA and Transient Preservation— *Adrian von dem Knesebeck, Pooya Ziraksaz, Udo Zölzer*, Helmut-Schmidt-University, Hamburg, Germany

An enhanced pitch shifting system is presented that uses the Pitch Synchronous Overlap Add (PSOLA) technique and a transient detection for processing of monophonic speech or instrument signals. The PSOLA algorithm requires the pitch information and the pitch marks for the signal segmentation in the analysis stage. The pitch is acquired using a well established pitch detector. A new robust pitch mark positioning algorithm is presented that achieves high quality results and allows the positioning of the pitch marks in a frame-based manner to enable real-time application. The quality of the pitch shifter is furthermore enhanced by extracting the transient components before the PSOLA and reapplying them at the synthesis stage to eliminate repetitions of the transients.
Convention Paper 8202

Special Event
FREE HEARING SCREENINGS CO-SPONSORED BY THE AES AND HOUSE EAR INSTITUTE
Friday, November 5 10:00 am–6:00 pm
Saturday, November 6 10:00 am–6:00 pm
Sunday, November 7 10:00 am–4:00 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.

Student Event/Career Development
STUDENT SCIENCE SPOT
Friday, November 5 through Sunday, November 7
10:00 am – 6:00 pm
Concourse

Hey students! Check out the Student Science Spot for cool student designs and projects! See the creative geniuses demonstrate their knowledge in the flesh. Learn about other student events or just hang and meet some new audio friends. This is an amazing opportunity to share your hard work—don't be shy! Hope to see you all there!

Friday, November 5 10:00 am Room 113
Technical Committee Meeting on Audio Recording and Mastering Systems

Tutorial 7 Friday, November 5
10:15 am – 11:15 am Room 120

ANALYSIS AND MODELING OF THE dbx 902 DE-ESSER

Presenter: **Aaron Wishnick**, Izotope Inc., Boston, MA, USA

Analog device modeling is an increasingly important tool in modern audio signal processing. There are a variety of techniques for modeling many different devices. Here we will present an example of modeling one device, the dbx 902 de-esser, a very well regarded hardware de-esser, from start to finish. We will describe a set of techniques for analyzing the hardware unit as a “grey” box to determine its characteristics, incorporating the device’s specifications, and, most importantly, empirical results from probing the unit with test signals. Mathematical models for analyzing the de-esser will be presented, which would also apply to other dynamic range control processors. We will examine how this device differs from other typical implementations of de-essers, and finally, describe a digital emulation. The lessons learned here should be useful to any beginner interested in device modeling.

Workshop 6 Friday, November 5
10:30 am – 12:45 pm Room 206

SINGLE UNIT SURROUND MICROPHONES

Chair: **Eddy B. Brixen**, EBB-Consult

Panelists: *Gary Elko*, mh acoustics LLC
David Josephson, Josephson Engineering
Jim Pace, Sanken Microphones/Plus 24
Pieter Schillebeeckx, Sound Field
Morten Stove, DPA Microphones
Mattias Strömberg, Milab
Helmut Wittek, SCHOEPS Mikrofone GmbH

The workshop will present available single-unit surround sound microphones in a kind of "shoot out." There are a number of these microphones available and more units are on their way. These microphones are based on different principles. However, due to their compact sizes there may/may not be restrictions to the performance. Basically this workshop will present the different products and the ideas and theories behind them.

Workshop 7 Friday, November 5
10:45 am – 12:45 pm Room 132

APPLICATIONS FOR HIGH-QUALITY AUDIO OVER LONG-DISTANCE NETWORKS

Chair: **Nathan Brock**, University of California San Diego

Panelists: *Chris Chafe*, Stanford University
Elizabeth Cohen, Cohen Acoustical
Jeremy Cooperstock, McGill University
Peter Stevens, BBC

The recent deployment of wide-area fiber networks has made low-latency streaming of uncompressed and lightly-compressed audio possible for many users in academia and industry. Applications for such streaming media, and for fast file transfers over such networks, have been explored for the past decade but are not widely known outside of the networking research community. This workshop will present several use cases for such networks in areas including live performance, production and postproduction, archiving, telecommunications, remote pedagogy, and broadcasting.

Session P9 **Friday, Nov. 5**
11:00 am – 1:00 pm **Room 220**

LISTENING TESTS

Chair: **Jon Boley**, LSB Audio, Lafayette, IN, USA

11:00 am

P9-1 A Digital-Domain Listening Test for High-Resolution —*John Vanderkooy*, University of Waterloo, Waterloo, Ontario, Canada

There is much debate over whether sampling rates and wordlengths greater than the CD standard are significant for high-quality audio. Tests that have been done require extreme care in selecting compatible devices with known characteristics. I propose tests that use the highest-quality wide-band microphones, only one set of ADCs and DACs, and wide-band reproducing loudspeakers. Real music and artificial signals can be used that have ultrasonic content. The ADCs and DACs are always used at the same extended bit width and high sampling rate, typically 24 bits and 176.4 or 192 kHz. To perform comparative tests at reduced sampling rates and lower bit widths, the digital data is mathematically altered to conform closely to the reduced specification. Files so created can be played back with precise time registration and identical level. ABX tests can be used to quantify if differences are heard, and ensure blindness of tests. Switching of program material can be done in the digital domain, so that relays or other compromising connectivity can be avoided. This paper discusses some remaining difficult issues and outlines the mathematical computations that will be necessary for sample-rate conversion, linear-phase aliasing and reconstruction filters, dithering, and noise shaping of the processed signal.
Convention Paper 8203

11:30 am

P9-2 Variance in Level Preference of Balance Engineers: A Study of Mixing Preference and Variance Over Time—*Richard King, Brett Leonard, Grzegorz Sikora*, McGill University, Montreal, Quebec, Canada

Limited research has been conducted that quantifies how much expert listeners vary over time. A task-based testing method is employed to discern the range of variance an expert listener displays over both short and long periods of time. Mixing engineers are presented with a basic mixing task comprised of one stereo backing track and a solo instrument or voice. By tracking the range in level in which the mixing engineers place a soloist into an accompanying track over a number of trials, trends are observed. Distributions are calculated for three genres of music and variance is calculated over time. The results show that in fact the variance is relatively low, and even lower for the more experienced subjects. These results also provide a baseline for future testing.
Convention Paper 8204

12:00 noon

P9-3 Evaluation of Super-Wideband Speech and Audio Codecs—*Ulf Wüstenhagen, Bernhard Feiten, Jens Kroll, Alexander Raake, Marcel Wältermann*, Deutsche Telekom Laboratories, Berlin, Germany

Increasingly growing usage of headphones for different telephony applications is paired with an increased quality expectation of the user. Recently, different standardization bodies have started work on an enhancement of telephone services. One objective is to improve the quality by providing a codec that supports low-delay super-wideband or fullband quality and in addition show a good quality not only for speech but also for music. Deutsche Telekom Laboratories have evaluated a range of low-delay super-wideband speech and audio codecs in comprehensive listening tests. The tests were conducted using the MUSHRA test method. A mixture of speech and audio conditions were used to check the performance of the codecs for different program types. The results of the listening tests are presented and discussed in the light of future applications.
Convention Paper 8205

12:30 pm

P9-4 Subjective Listening Tests and Neural Correlates of Speech Degradation in Case of Signal-Correlated Noise—*Jan-Niklas Antons,*[1] *Anne K. Porbadnigk,*[1] *Robert Schleicher,*[1] *Benjamin Blankertz,*[1] *Sebastian Möller,*[1] *Gabriel Curio*[2]
[1]Berlin Institute of Technology, Berlin, Germany
[2]Charité-University Medicine, Berlin, Germany

In this paper we examine whether particularly sensitivity of the human cortex to reduction in speech quality is visible in the electroencephalogram (EEG) and whether these measures can be used to improve the behavioral assessment of speech quality. We degraded a speech stimulus (vowel /a/) in a scalable way and asked for a behavioral rating. In addition, the brain activity was measured with EEG. We trained classifiers, who were found capable of distinguishing between events that are seemingly similar at the behavioral level (i.e., no button press), neurally, however, noise contamination is detected, possibly affecting the long-term contentment with the transmission quality.
Convention Paper 8206

Broadcast/Media Streaming Session 6
Friday, November 5 **11:00 am – 1:00 pm**
Room 133

LOUDNESS, METADATA, AND OTHER AUDIO CONCERNS FOR DTV

Chair: **Tomlinson Holman**, University of Southern

California School of Cinematic Arts and Viterbi School of Engineering

Panelists: *Tim Carroll*, Linear Accoustic
David Casey, DTS Inc.
Sterling Davis, Cox Media Group
Thomas Lund, TC Electronics
Steve Lyman, Dolby Laboratories
Jim Starzinski, NBC Universal, Chair ATSC S6-3 Audio Loudness Group
Pat Waddell, Harmonic Inc., Chair ATSC S6 Audio and Video Coding

The introduction of digital television to the U.S. market proceeded in a number of steps: standardization in the early 1990s, first introduction of transmission and sets in the late 1990s, and accelerated adoption over time, culminating as NTSC television was shut off in 2009. The audio standards (that were thought to be easy to do so came first on the testing schedule but turned out to be far more complex than expected) added a number of features to conventional workflows that are only now becoming to be understood in some areas of the vast television production chain. This workshop will discuss what the situation is today from several points of view and how the standards are likely to be promulgated over the next few years.

Live Sound Seminar 4 **Friday, November 5**
11:00 am – 1:00 pm **Room 131**

MEASUREMENT SYSTEMS AND APPLICATIONS

Chair: **John Murray**, Optimum System Solutions

Panelist: *Jamie Anderson*, Rational Acoustics
Ralph Heinz, Renkus-Heinz
Bruce C. Olson, Olson Sound Design
Karl Winkler, Lectrosonics

An investigation of the most popular measurement systems used to align sound systems in the field. Methods and results will be discussed.

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Friday, November 5 11:00 am – 11:45 am
Room 122

Presenter: **Dave Isaac**

MixSensei.com Launch

This seminar integrates engineering skills, musical knowledge, and practical understanding of studio equipment into a fully interactive form of learning that is enriched by the involvement of many high-profile engineers and musicians. PMC user Dave Isaac is a 3-time Grammy winning Producer/ Engineer, artist, and author who has worked with Marcus Miller, Eric Clapton, Michael Jackson, Madonna, Luther Vandross, Puff Daddy, Whitney Houston, Prince, Anita Baker, and many other artists. Dave has developed a video tutorial website (www.mixsensei.com), aimed at the music and audio industry in which he and other Award-winning engineers/ producers and musicians share all their knowledge and "tricks of the trade"!

Friday, November 5 **11:00 am** **Room 113**
Technical Committee Meeting on Human Factors in Audio Systems

Workshop 8 **Friday, November 5**
11:15 am – 12:45 pm **Room 130**

MASTERING: ART, PERCEPTION, TECHNOLOGIES —PART 1

Chair: **Michael Romanowski**, Michael Romanowski Mastering

Panelists: *Gavin Lurssen*, Gavin Lurssen Mastering
Andrew Mendleson, Georgetown Masters
Joe Palmacio, The Place for Mastering
Mike Wells, Mike Wells Mastering

This is a continuation of the Mastering panel from AES 2009 in New York. We will discuss the state of Mastering in 2010. Mastering engineers use technology to achieve the desired results. But what gets little or no discussion is the perceptions and approaches that cause the engineer to make those choices. In this two part series, we want to talk about the art of perception and technology as it pertains to the Mastering industry in 2010 and the future. This particular discussion will focus on mastering technologies and the state of mastering today and looking forward.

Session P10 **Friday, Nov. 5**
11:30 am – 1:00 pm **Room 226**

POSTERS: AUDIO PROCESSING—PART 2

11:30 am

P10-1 MPEG-A Professional Archival Application Format and its Application for Audio Data Archiving—*Noboru Harada,*[1] *Yutaka Kamamoto,*[1] *Takehiro Moriya*,[1] *Masato Otsuka*[2]
[1]NTT Communication Science Labs., Atsugi, Kanagawa, Japan
[2]Memory-Tech Corporation, Tokyo, Japan

ISO/IEC 23000-6 (MPEG-A) Professional Archival Application Format (PA-AF) has just been standardized. This paper proposes an optimized and standard compliant implementation of a PA-AF archiving tool for audio archiving applications. The implementation made use of an optimized MPEG-4 Audio Lossless Coding (ALS) codec library for audio data compression and Gzip for other files. The PA-AF specification was extended to support platform specific attributes of Mac OSs while keeping interoperability among other OSs. Performance test results for actual audio data, such as ProTools HD projects, show that the processing time of a devised PA-AF archiving tool is twice as fast as that of MacDMG and WinZip while the compressed data size is much smaller than that of MacDMG and WinZip.
Convention Paper 8207

11:30 am

P10-2 A Reverberator with Two-Stage Decay and Onset Time Controls—*Keun-Sup Lee, Jonathan S. Abel*, Stanford University, Stanford, CA, USA

An efficient artificial reverberator having two-stage decay and onset time controls is presented. A second-order comb filter controlling the reverberator frequency-dependent decay rates and onset times drives a switched convolution with short noise ➥

sequences. In this way, a non-exponential reverberation envelope is produced by the comb filter, while the switched convolution structure produces a high echo density. Several schemes for generating two-stage decays and onset time controls with different onset characteristics in different frequency-band are described.
Convention Paper 8208

11:30 am

P10-3 Guitar-to-MIDI Interface: Guitar Tones to MIDI Notes Conversion Requiring No Additional Pickups—*Mamoru Ishikawa, Takeshi Matsuda, Michael Cohen*, University of Aizu, Aizu-Wakamatsu, Fukushima-ken, Japan

Many musicians, especially guitarists (both professional and amateur), use effects processors. In recent years, a large variety of digital processing effects have been made available to consumers. Further, desktop music, the "lingua franca" of which is MIDI, has become widespread through advances in computer technology and DSP. Therefore, we are developing a "Guitar to MIDI" interface device that analyzes the analog guitar audio signal and emits a standard MIDI stream. Similar products are already on the market (such as the Roland GI-20 GK-MIDI Interface), but almost all of them need additional pickups or guitar modification. The interface we are developing requires no special guitar accessories. We describe a prototype platformed on a PC that anticipates a self-contained embedded system.
Convention Paper 8209

11:30 am

P10-4 A Mixed Mechanical/Digital Approach for Sound Beam Pointing with Loudspeakers Line Array—*Paolo Peretti,*[1] *Stefania Cecchi,*[1] *Francesco Piazza*,[1] *Marco Secondini,*[2] *Andrea Fusco*[2]
[1]Università Politecnica delle Marche, Ancona (AN), Italy
[2]FBT Elettronica S.p.a., Recanati (M), Italy

Digital steering is often used in line array sound systems in order to tilt the reproduced sound beam in a desired direction. Unfortunately, the working frequency range is limited to low and medium frequencies, thus, sound beams referred to high frequencies can be tilted only by using a mechanical steering involving both an expensive manufacture and a higher environmental impact. The proposed solution is a mixed approach to sound beam steering by considering an on-axis mechanical rotation of each loudspeaker together with the classical digital control applied to the entire system. In this manner the sound beam can be tilted also at high frequency maintaining linear array geometry. Simulations, considering real loudspeaker directivity, will be shown in order to demonstrate the effectiveness of the proposed approach.
Convention Paper 8210

11:30 am

P10-5 The Non-Flat and Continually Changing Frequency Response of Multiband Compressors—*Earl Vickers*, STMicroelectronics, Inc., Santa Clara, CA, USA

Multiband dynamic range compressors are powerful, versatile tools for audio mastering, broadcast, and playback. However, they are subject to certain problems relating to frequency response. First, when excited by a time-varying narrowband input such as a swept sinusoid, they create unwanted magnitude peaks at the band boundaries. Second, and more importantly, the frequency response continually changes, which may have unwanted effects on the long-term average spectral balance. This paper proposes a frequency-domain solution for the unwanted magnitude peaks, whereby slight adjustments to the band boundaries prevent sinusoidal peaks from being midway between two bands. For the second problem, real-time spectral balance compensation may be implemented in either the time or frequency domain.
Convention Paper 8211

11:30 am

P10-6 Volterra Series-Based Distortion Effect—*Finn T. Agerkvist*, Technical University of Denmark, Lyngby, Denmark

A large part of the characteristic sound of the electric guitar comes from nonlinearities in the signal path. Such nonlinearities may come from the input- or output-stage of the amplifier, which is often equipped with vacuum tubes or a dedicated distortion pedal. In this paper the Volterra series expansion for non linear systems is investigated with respect to generating good distortion. The Volterra series allows for unlimited adjustment of the level and frequency dependency of each distortion component. Subjectively relevant ways of linking the different orders are discussed.
Convention Paper 8212

Game Audio Session 5 — Friday, November 5
11:30 am – 1:00 pm — Room 120

DEVELOPING SENSIBLE REFERENCE LEVEL STANDARDS

Chair: **Steve Martz**, Sr. Design Engineer, THX Ltd.

Panelists: *Lance Brown*, Cinematic Game Audio Consultant
Charles Deenen, Senior Creative Director, Audio, Electronic Arts
Ken Felton, Sound Design Manager, SCEA
Tom Hays, Director of Audio Services, Technicolor
Francesco Zambon, Audio Project Lead, Binari Sonori s.r.l.

Particularly in environments where the mix is dynamic and constantly changing, a continuing challenge for game developers is devising (and abiding by) guidelines for appropriate playback levels. While the ever-loudening, highly dynamic-range compressed strategies of the music industry may be appropriate in that world, games can use multiple alternate techniques to "feel" louder and maintain a wide dynamic range without forcing the player to scramble for their remote. This panel will cover the findings of an ongoing multi-platform, multi-studio conversation about what such a set of guidelines would look like, and how we can apply these.

Student Event/Career Development
STUDENT RECORDING CRITIQUES
Friday, November 5, 12:00 noon – 1:00 pm
Room 122

Chairs: **Ian Corbett**, Kansas City Community College, Kansas City, KS, USA
David Greenspan, University of Michigan, Ann Arbor, MI, USA

Students are encouraged to bring in their stereo or surround projects to this non-competitive listening session for feedback and comments from a panel and audience. Students will be able to sign-up for time slots at the first SDA meeting and at the Student Science Spot, on a first come, first served basis. Students who are finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. Bring your stereo or surround work on CD, DVD, or hard disc as clearly-labelled .wav files. The Student Recording Critiques are generously sponsord by PMC.

Friday, November 5 12:00 noon Room 113
Technical Committee Meeting on Perception and Subjective Evaluation of Audio Signals

Friday, November 5 12:00 noon Room 232
Standards Committee Meeting SC-02-12 Audio Applications of Networks

Exhibitor Seminar
FOCUSRITE
Friday, November 5 12:30 pm – 1:30 pm
Room 112

Presenter: **Brad Price**, Audinate Senior Technical Solutions Mgr.

Preparing for the Future: Media Transport over IP Networks

Media transport has until recently been a specialized area of digital technology, isolated from the rapid pace of advancements taking place in IP network computing. New solutions that leverage the latest advancements and cost savings of modern IP networking are appearing and are necessary to "future proof" media transport across the A/V industry.

Friday, November 5 1:00 pm Room 113
Technical Committee Meeting on Spatial Audio

Special Event
LUNCHTIME KEYNOTE: DAVE RAT
Friday, November 5, 1:15 pm – 2:15 pm
Room 132

Live sound is an exciting and rapidly changing industry. Dave Rat, Founder and President of Rat Sound, has seen it all over the last 30 years. He has a unique multi-dimensional perspective as a front-of-house engineer, product designer, industry writer, and president of a major rental and touring company. He is known for questioning common industry practices with healthy doses of pragmatism and scientific curiosity. Dave will share his insights on frequently overlooked needs and challenges facing the live sound touring industry. Dave's style is light-hearted and humorous but the issues he raises are always thought-provoking. If you have any involvement in the live sound industry, you will not want to miss this.

Special Event
SOCIAL MEDIA FOR MUSICIANS AND ENGINEERS —PART 1
Friday, November 5, 1:15 pm – 2:15 pm
Room 130

Presenter: **Bobby Owsinski**

For the first time in history, an artist or band can directly communicate, market, and sell directly to their fans without the need of a record label. This presentation describes the basic elements needed for a successful online social presence, as well as the latest strategies for music and video releases, online promotion and sales, and interaction with fans and clients.

Topics covered include:

- The meaning of Music 3.0
- Why a website is the center of your social media universe
- The elements of a successful site
- The new release schedule
- Your mailing list – old tech, new importance
- The social media world is larger than you think

Exhibitor Seminar
AUDIOMATICA
Friday, November 5 2:00 pm – 3:00 pm
Room 112

Presenters: **Mauro Bigi**
Maurizio Jacchia
Daniele Ponteggia

CLIO Classroom on Electro-Acoustic Measurements

Audiomatica's folk will present the CLIO System through "hands-on" applications. Topics will touch loudspeaker measurements focusing on parameters tests with laser, large-signal time-frequency measurements and quality control applications.

Special Event
PLATINUM MASTERING
Friday, November 5, 2:00 pm – 4:00 pm
Room 134

Moderator: **Bob Ludwig**

Panelists: *Michael Fremer*
Doug Sax

Mastering legend Bob Ludwig will moderate a panel that explores vinyl mastering and disc cutting. Besides Bob, the other panelists will be renowned engineer and disc-cutting expert Doug Sax and vinyl guru Michael Fremer, producer of the DVD *21st Century Vinyl: Michael Fremer's Practical Guide to Turntable Set-Up.* Panelists will discuss how they approach vinyl mastering, disc cutting, and turntable setup. Ludwig will show archival videos of how lacquers are made as well as a clip of him cutting the vinyl master to the hit Genesis album "Invisible Touch."

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Friday, November 5 1:30 pm – 2:30 pm
Room 122

Presenter: **Bobby Fernandez**

The Art of Film Score Mixing

Scoring- and multi-Grammy winning engineer Robert "Bobby" Fernandez will present a case study on film score mixing. As a recording and mixing engineer Bobby has worked on hundreds of movies including many Academy award winning movies like *Million Dollar Baby*, *Unforgiven*, *Spiderman 2 and 3*, *Flags of Our Fathers*, *Master and Commander*, and many more. Most recently he has been working on the Matt Damon movie *Hereafter* and the action movie *The Expendables*. Bobby will present a full session from one of his latest movies and will show us what is involved in the process of recording and mixing a movie score.

Friday, November 5 2:00 pm Room 113
Technical Committee Meeting on High Resolution Audio

Friday, November 5 2:00 pm Room 232
Standards Committee Meeting SC-05-02 Audio Connectors

Session P11 Friday, Nov. 5
2:30 pm – 6:30 pmRoom 220

ACOUSTICAL AND PHYSICAL MODELING

Chair: **Julius O. Smith III**, Stanford University, Stanford, CA, USA

2:30 pm

P11-1 Virtual Acoustic Prototyping—Practical Applications for Loudspeaker Development— *Alex Salvatti*, JBL Professional, Northridge, CA, USA

Acoustic simulations using finite elements have been used in loudspeaker development for over 20 years, with complexity and accuracy accelerating in tandem with the increases in computing power generally available on the engineering desktop. Using user-friendly, modern FEA software, the author presents an overview of methods to build virtual prototypes of both horns and loudspeaker drivers that allows a significant reduction in the number of physical prototypes, as well as reduced development time. A comparison of simulated vs. measured data proves the validity of the methods.
Convention Paper 8213

3:00 pm

P11-2 Simulation of Horn Driver Response by Combination of Matrix Analysis and FEA— *Alex Voishvillo*, JBL Professional, CA, USA

To access performance of a horn driver (compression driver loaded by a horn), measurement of frequency response on-axis and off-axis must be carried out. The measurement process is time-consuming especially if the entire 3-dimensional "balloon" of responses is to be measured. Prediction of directional responses of the horn only (without compression driver) can be performed by the FEA (Finite Elements Analysis) or BEA (Boundary Elements Analysis). However, FEA or BEA of horn only provides relative directional properties of the horn. The SPL responses of horn driver at different angles remain unknown because these responses depend on interaction of electrical, mechanical, and acoustical parameters of the compression driver and the acoustical parameters of the horn. New methods based on a combination of FEA and matrix analysis makes it possible to predict the response of a combination of various compression drivers and horns without actually measuring each combination and even without physically building horns. This method was verified during the development of a new AM series of JBL professional loudspeaker systems and showed high accuracy.
Convention Paper 8214

3:30 pm

P11-3 Dynamic Motion of the Corrugated Ribbon in a Ribbon Microphone—*Daniel Moses Schlessinger*,[1] *Jonathan S. Abel*[2]
[1]Sennheiser DSP Research Laboratory, Palo Alto, CA, USA
[2]Stanford University, Stanford, CA, USA

Ribbon microphones are known for their warm sonics, owing in part to the unique ribbon motion induced by the sound field. Here the motion of the corrugated ribbon element in a sound field is considered, and a physical model of the ribbon motion is presented. The model separately computes propagating torsional disturbances and coupled transverse and longitudinal disturbances. Each propagation mode is implemented as a mass-spring model where a mass is identified with a ribbon corrugation fold. The model is parametrized using ribbon material and geometric properties. Laser vibrometer measurements are presented, revealing stiffness in the transverse and longitudinal propagation and showing close agreement between measured and modeled ribbon motion.
Convention Paper 8215

4:00 pm

P11-4 Modeling of Leaky Acoustic Tube for Narrow-Angle Directional Microphone—*Kazuho Ono,*[1] *Takehiro Sugimoto,*[1] *Akio Ando,*[1] *Kimio Hamasaki*,[1] *Takeshi Ishii,*[2] *Yutaka Chiba,*[2] *Keishi Imanaga*[2]
[1]NHK Science and Technology Research Laboratories, Kinuta Setagaya-ku, Tokyo, Japan
[2]Sanken Microphone Co. Ltd., Suginami-ku, Tokyo, Japan

Line microphones have been popular as narrow directional microphones for a long time. Their structure adopts a leaky acoustical tube with many slits to suppress off-axis sensitivity, together with a directional capsule attached to this tube. Although many microphones of this type are on the market, we have no quantitative theory to explain its behavior, which is very important for effectively designing directivity. We thus modeled the leaky acoustical tube using a distributed equivalent circuit and combined it with the directional capsule's equivalent circuit model. The analysis showed that the model agreed well with the measurement results, particularly at the directional characteristics, while an ordinary model of acoustical tube using delay and sum modeling did not.
Convention Paper 8216

4:30 pm

P11-5 Modeling Viscoelasticity of Loudspeaker Suspensions Using Retardation Spectra— *Tobias Ritter, Finn Agerkvist*, Technical University of Denmark, Kgs. Lyngby, Denmark

It is well known that, due to viscoelastic effects in the suspension, the displacement of the loudspeaker increases with decreasing frequency below the resonance. Present creep models are either not precise enough or purely empirical and not derived from the basis of physics. In this investigation, the viscoelastic retardation spectrum, which provides a more fundamental description of the suspension viscoelasticity, is first used to explain the accuracy of the empirical LOG creep model (Knudsen et al.). Then, two extensions to the LOG model are proposed that include the low and high frequency limit of the compliance, not accounted for in the original LOG model. The new creep models are verified by measurements on two 5.5 loudspeakers with different surrounds.
Convention Paper 8217

5:00 pm

P11-6 Physical Modeling and Synthesis of Motor Noise for Replication of a Sound Effects Library—*Simon Hendry, Josh Reiss*, Queen Mary University of London, London, UK

This paper presents the results of objective tests exploring the concept of using a small number of physical models to create and replicate a large number of samples from a traditional sound effects library. The design of a DC motor model is presented and this model is used to create both a household drill and a small boat engine. The harmonic characteristics, as well as the spectral centroid were compared with the original samples, and all the features agree to within 6.1%. The results of the tests are discussed with a heavy emphasis on realism and perceived accuracy, and the parameters that have to be improved in order to humanize a model are explored.
Convention Paper 8218

5:30 pm

P11-7 Measures and Parameter Estimation of Triodes for the Real-Time Simulation of a Multi-Stage Guitar Preamplifier—*Ivan Cohen*,[1,2] *Thomas Hélie*[1]
[1]Ircam, Paris, France
[2]Orosys R&D, Montpellier, France

This paper deals with the real-time simulation of a multi-stage guitar preamplifier. Dynamic triode models based on Norman Koren's model, and "secondary phenomena" as grid rectification effect and parasitic capacitances are considered. Then, the circuit is modeled by a nonlinear differential algebraic system, with extended state-space representations. Standard numerical schemes yield efficient stable simulations of the circuit and are implemented as VST plug-ins. Measures of real triodes have been realized, to develop new triode models, and to characterize the capabilities of aged and new triodes. The results are compared for all the models, using lookup tables generated with the measures and Norman Koren's model with its parameters estimated from the measures.
Convention Paper 8219

6:00 pm

P11-8 ZFIT: A MATLAB Tool for Thiele-Small Parameter Fitting and Optimization— *Christopher Struck*, CJS Labs, San Francisco, CA, USA

Over the years, many approaches to the calculation of the Thiele-Small parameters have been presented. Most current methods rely upon curve-fitting to the impedance magnitude data for a specific lumped parameter model. A flexible Matlab least-mean-squares optimization tool for complex loudspeaker impedance data is described. Magnitude and phase data are fit to a user-selected lumped parameter model of variable complexity. Appropriate constraints on the optimization help identify if the selected model is of sufficient order or overly complex for the given data. Examples are shown for impedance data from several different loudspeaker drivers.
Convention Paper 8220

Session P12 — Friday, Nov. 5
2:30 pm – 5:30 pm — Room 236

VIRTUAL ROOMS

Chair: **Jean-Marc Jot**, DTS Inc., Scotts Valley, CA, USA

2:30 pm

P12-1 Assessing Virtual Teleconferencing Rooms— *Mansoor Hyder, Michael Haun, Olesja Weidmann, Christian Hoene*, Universität Tübingen, Tübingen, Germany

Spatial audio makes teleconferencing more natural, helps to locate and distinguish talkers in a virtual acoustic environment, and to understand multiple talkers. This paper presents a study on how to design virtual acoustic environments used in 3-D audio teleconferences to maximize localization performance, easiness, and subjective speech quality ratings. We conducted subjective listening-only tests considering different parameters describing the virtual acoustic environment, including acoustic room properties, virtual sitting arrangements, reflections of a conference table, number of concurrent talkers and different voice types of call participants. The experimental results help us to enhance the performance of our open-source, spatial audio teleconferencing solution named "3DTel" by enhancing the quality of its user experience in terms of naturalness and speech quality.
Convention Paper 8221

3:00 pm

P12-2 Stereo Acoustic Echo Cancellation for Telepresence Systems—*Shreyas Paranjpe, Scott Pennock, Phil Hetherington*, QNX Software Systems, Vancouver, BC, Canada

A Telepresence system provides its users the sense that they are virtually present in another physical location. Typically, this means providing ➥

a high quality audio and video communication path. Simply adding video communication to typical audio teleconferencing is not enough. When users see each other, they quickly realize the poor audio performance, such as half duplex behavior, that is commonly implemented. In order to make affordable Telepresence systems for everyone, the challenge is to design high performance audio communication systems that are computationally efficient.
Convention Paper 8222

3:30 pm

P12-3 Early Energy Conditions in Small Rooms and in Convolutions of Small-Room Impulse Responses—*U. Peter Svensson, Hassan el-Banna Zidan*, Norwegian University of Science and Telecommunications, Trondheim, Norway

A simplified prediction model for the early-to-late energy ratio has been tested for small rooms that are typically used in video conferences. Measurements have been carried out in a few rooms and early- and late-energy levels have been compared with predictions according to Barron's model and predicted octave-band levels are typically within 1-2 dB of measured values. Measured impulse responses are then convolved to simulate a video conference setup, and simplified predictions of the early and late energy conditions of convolved impulse responses are compared with (convolved) measurements.
Convention Paper 8223

4:00 pm

P12-4 A Convolution-Based System for Virtual Acoustic Support of Performing Musicians—*Wieslaw Woszczyk, Doyuen Ko, Brett Leonard*, McGill University, Montreal, Quebec, Canada

Musicians performing on stage need to hear a proper balance of themselves and other players in order to achieve a good sense of ensemble and musicality. Their ability is influenced by the quality of the acoustic response of the room. In spaces where the existing acoustic conditions are detrimental to good communication on stage, "electronic architecture" may be used to rebuild the acoustic support for musicians. A system is developed that utilizes measured impulse responses from a variety of superior acoustic spaces to generate, using near zero-latency multichannel convolution, an artificial sound field augmenting that already present. This method of virtual acoustic technology does not amplify (or use again) the energy produced by the existing room; instead it generates desirable room response components from other measured spaces. The adjustable acoustic conditions are set using a comprehensive GUI, transducer arrays, and a layered system architecture.
Convention Paper 8224

4:30 pm

P12-5 Simulating Hearing Loss in Virtual Training—*Ramy Sadek, David Krum, Mark Bolas*, University of Southern California Institute for Creative Technologies, Playa Vista, CA, USA

Audio systems for virtual reality and augmented reality training environments commonly focus on high-quality audio reproduction. Yet many trainees may face real-world situations wherein hearing is compromised. In these cases, the hindrance caused by impaired or lost hearing is a significant stressor that may affect performance. Because this phenomenon is hard to simulate without actually causing hearing damage, trainees are largely unpracticed at operating with diminished hearing. To improve the match between training scenarios and real-world situations, this effort aims to add simulated hearing loss or impairment as a training variable. The goal is to affect everything users hear—including non-simulated sounds such as their own and each other's voices—without overt noticeability, risk to hearing, or requiring headphones.
Convention Paper 8225

5:00 pm

P12-6 OpenAIR: An Interactive Auralization Web Resource and Database—*Simon Shelley, Damian T. Murphy*, University of York, Heslington, York, UK

There have been many recent initiatives to capture the impulse responses of important or interesting acoustic spaces, although not all of this data has been made more widely available to researchers interested in auralization. This paper presents the Open Acoustic Impulse Response (OpenAIR) Library, a new online resource allowing users to share impulse responses and related acoustical information. Open-source software is provided, allowing the user to render the acoustical data using various auralization strategies. Software tools and guidelines for the process of impulse response capture are also provided, aiming to disseminate best practice. The database can accommodate impulse response datasets captured according to different measurement techniques and the use of robust spatial audio coding formats is also considered for the distribution of this type of information. Visitors to the resource can search for acoustical data using keywords and can also browse uploaded datasets on a world map.
Convention Paper 8226

Session P13 — Friday, Nov. 5
2:30 pm – 4:00 pm — Room 226

POSTERS: AUDIO EQUIPMENT AND MEASUREMENT

2:30 pm

P13-1 Neutral-Point Oscillation Control Based on a New Audio Space Vector Modulation (A-SVM) for DCI-NPC Power Amplifiers—*Vicent Sala, Luis Romeral, G. Ruiz*, UPC-Universitat Politecnica de Catalunya, Terrassa, Spain

In this paper the oscillation or flotation in the DC-BUS neutral point in the DCI-NPC (Diode Clamped Inverter – Neutral Point Clamped) amplifiers is presented as one of the most important distortion sources. This perturbation is characterized and studied, as well as its causes and distorting effects. It also presents two techniques of vector modulation for audio. The intelligent use of these techniques in the process of vector

modulation allows the redistribution of the charge of the two capacitors in the DC-BUS, allowing the control of the voltage in the neutral point of the DC-BUS, and therefore, the cancellation of the flotation and its distorting effects. Experimental and simulation results that verify these strategies are presented.
Convention Paper 8227

2:30 pm

P13-2 Vacuum Tube Amplifiers Using Electronic DC Transformers—*Theeraphat Poomalee,*[1] *Kamon Jirasereeamornkul,*[1] *Marian K. Kazimierczuk*[2]
[1]King Mongkut's University of Technology Thonburi, Tung-kru, Bangkok, Thailand
[2]Wright State University, Dayton, OH, USA

This paper proposes a method to synthesis vacuum-tube audio amplifiers using the electronic DC transformers to replace the traditional audio-frequency output transformers usually used in the output stage of the amplifier. The proposed amplifiers can achieve the frequency response from DC-100 kHz if the DC transformers operated at 500 kHz switching frequency and interleave technique are used. The principle of operation, DC model, and various examples are given.
Convention Paper 8228

2:30 pm

P13-3 The Single Stereo Display and Stereo VU Meters— *Michael D. Callaghan*, Radio Station KIIS-FM, Los Angeles, CA, USA

This paper describes the use of a single row of bi-color indicators to replace and overcome the deficiencies of the typical pair of meters used to show left and right signal levels in stereo applications. By using bi-color elements, a total of three colors are actually obtained; a single color when the left channel is driven, a single color when the right channel is driven, and a mixture of the two when both channels are driven. Watching the row of indicators during program operation will indicate three different amplitudes; the left channel volume, the right channel volume, and the difference between the two of them. These amplitudes are immediately obvious and very easy to interpret.
Convention Paper 8229

2:30 pm

P13-4 Frequency Characteristics Measurements of Cylindrical Record Player by the Pulse-Train Method—*Teruo Muraoka, Takahiro Miura, Tohru Ifukube*, The University of Tokyo, Tokyo, Japan

The authors have been engaged in the research of restoration of seriously damaged audio signals employing Generalized Harmonic Analysis (GHA). In this research it is important to know frequency characteristics of sound reproducing equipment to obtain clear sound with proper tonal equalization. The authors previously measured frequency characteristics of several acoustic 78 rpm shellac-record players utilizing the Pulse-Train Method, and successively measured cylindrical record players with same method recently. Frequency characteristics of phonograph record players were measured using frequency test records conventionally, however it is impossible to obtain shellac or cylindrical test records any more. Therefore the authors employed the Pulse-Train Method, which was originally developed for the measurements of phonograph cartridges and cutter heads in 1970s. For the measurement this time, the authors first made a cylindrical record curved a silent sound groove and curved an additional groove perpendicular to the sound groove on the cylinder surface. Pulse-train response was obtained by reproducing the cylindrical record using object record players and reference electric record player. Frequency characteristics of object record players were analyzed applying DFT to measured Pulse-Train waveforms.
Convention Paper 8230

2:30 pm

P13-5 Seeing Sound: Sound Sensor Array with Optical Outputs—*Charles Seagrave*,[1] *Eric Benjamin*[2]
[1]Seagrave Instruments, San Rafael, CA, USA
[2]Surround Research, Pacifica, CA, USA

Characterization of acoustic spaces frequently involves taking SPL measurements at numerous locations within the space. Such measurements typically require relocation of the measurement apparatus or multiple microphones wired to a multiplexer. This approach can be time consuming, especially if it must be repeated after changes in loudspeaker location or acoustical treatments of other modifications. This paper presents methods of visualizing both standing waves in rooms and loudspeaker coverage uniformity in outdoor venues, using an array of sound sensors with optical (visible light) output. This new approach allows for rapid visual observation of sound fields, and simultaneous SPL data collection from multiple positions.
Convention Paper 8231

2:30 pm

P13-6 Effects of Oversampling on SNR Using Swept-Sine Analysis—*Christopher Bennett, Daniel Harris, Adam Tankanow, Ryan Twilley*, Oygo Sound, LLC, Miami, FL, USA

The swept-sine technique is an alternative method to acquire impulse response measurements and distortion component responses. Swept-sine analysis has been under recent investigation for its use in auditory applications. In this paper the researchers seek to show that an improvement in signal-to-noise ratio (SNR) can be achieved by applying oversampling while utilizing swept-sine analysis. Oversampling does not give an improvement in SNR in traditional click impulse response methods; however, due to the noise shaping properties of the post-processing involved in swept-sine analysis, the noise floor can be reduced.
Convention Paper 8232

2:30 pm

P13-7 Rapid In-Place Measurements of Multichannel Venues—*John Vanderkooy*, University of Waterloo, Waterloo, Ontario, Canada

It is often useful to have transfer-function mea- ➥

surements of large venues with an audience present. This precludes multiple chirps or other long-duration signals from being used. This paper studies the use of simultaneous, multiple "orthogonal" maximum-length sequences applied to the loudspeakers, captured by a number of microphones at selected listening positions. Such MLS signals last only a few seconds and are noise-like, being minimally disruptive to an audience, yet they allow full transfer-function system identification between each loudspeaker and microphone. The main detractor of the method is that the effective noise level is high. This paper studies implementation issues and assesses the S/N of such measurements. It turns out that exciting each loudspeaker separately is usually better than simultaneous excitation, except in special circumstances. An example is shown for the simultaneous measurement of two loudspeakers in a room with two microphones.
Convention Paper 8233

2:30 pm

P13-8 Ground Loops: The Rest of the Story—
Bill Whitlock,[1] *Jamie Fox*[2]
[1]Jensen Transformers, Inc., Chatsworth, CA, USA
[2]The Engineering Enterprise, Alameda, CA, USA

The mechanisms that enable so-called ground loops to cause well-known hum, buzz, and other audio system noise problems are well known. But what causes power-line related currents to flow in signal cables in the first place? This paper explains how magnetic induction in ordinary premises AC wiring creates the small voltage differences normally found among system ground connections, even if "isolated" or "technical" grounding is used. The theoretical basis is explored, experimental data shown, and an actual case history related. Little has been written about this "elephant in the room" topic in engineering literature and apparently none in the context of audio or video systems. It is shown that simply twisting L-N pairs in the premises wiring can profoundly reduce system noise problems.
Convention Paper 8234

Workshop 9 Friday, November 5
2:30 pm – 4:00 pm Room 130

LIVE MONITORING AND LATENCY WITH DIGITAL AUDIO NETWORKS

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

Panelists: *Carl Bader*, Aviom
Kevin Gross, AVA Networks
Michael Lester, Shure
Robert Scovill, Avid

The increasing adoption of digital audio networks for live events can impact the latency of audio signals as perceived on stage. Issues related to audio latency when considering personal monitoring and traditional, speaker-based monitoring are discussed. Real cases are shown and detailed, as well as the preferences and habits of performers.

Broadcast/Media Streaming Session 7
Friday, November 5 2:30 pm – 3:45 pm
Room 133

INNOVATIONS IN DIGITAL RADIO

Chair: **David Bialik**, Consultant

Panelists: *Steve Fluker*, Cox Radio
Frank Foti, Telos-Omnia-Axia
David Layer, NAB
Skip Pizzi, Consultant/Radio Ink
Tom Ray, WOR - Buckley Broadcasting
Geir Skaaden, DTS
David Wilson, CEA

This session will discuss the various innovations of the past year plus what is on the horizon. Transmission, playback, production, and reception are some of the topics. This will be a discussion of technology and technique.

Game Audio Session 6 Friday, November 5
2:30 pm – 4:00 pm Room 120

MOBILE GAME AUDIO FOR HEADPHONES AND MICRO-SPEAKERS

Chair: **Steve Martz**, Sr. Design Engineer, THX Ltd.

Panelists: *Peter "pdx" Drescher*, Sound Designer, Twittering Machine
Greg Klas, Sr. Manager, Audio Engineering, Fisher-Price, Inc.
Jeffrey Xia, Sr. Acoustics Engineer, Ole Wolf Electronics

Mobile platforms (phones, toys, portable gaming devices) are typically relegated to using small speakers as a means to recreate an immersive environment for games. These playback devices have certain attributes that require unique approaches when creating content for mobile entertainment. A panel consisting of speaker manufacturers and mobile game creators will discuss the performance characteristics and limitations of headphones and other micro-speakers as they pertain to playback of game audio on those devices as well considerations for designing game content.

Live Sound Seminar 5 Friday, November 5
2:30 pm – 4:15 pm Room 131

WIRELESS MICROPHONES FOR THE FUTURE

Chair: **James Stoffo**, Professional Wireless Systems

Panelists: *Don Boomer*, Line 6
Mark Brunner, Shure
Joe Ciaudelli, Sennheiser
Gino Sigismondi, Shure
Karl Winkler, Lectrosonics

The FCC keeps changing the wireless spectrum available for microphones. The 700 MHz band is already off limits and now there is Super-WiFi and the National Broadband Plan to consider below 700 MHz. Is any part of the spectrum safe? Learn the latest developments from the FCC and how the experts are insuring reliable RF operation now and preparing for the future.

Product Design Session 3 **Friday, November 5**
2:30 pm – 4:00 pm **Room 132**

IEEE 802.1 AUDIO/VIDEO BRIDGING (AVB)

Moderator: **Lee Minich**

Panelists: *Bradford Benn*, Harman
Joerg Bertholdt, XMOS
Jerry Placken, Meyer Sound
Sheldon Radford, Avid
Michael Johas Teener, Broadcom
Dave Theis, Sennheiser
Aidan Williams, Audinate

This panel discussion with key members of the AVnu Alliance will explore the role of the IEEE 802.1 Audio/Video Bridging [AVB] Standards in the professional audio industry, as well as consumer electronics and automotive applications. An in-depth explanation of AVB and related IEEE standards will be presented, followed by a discussion of the advantages of an open non-proprietary technology, the role of silicon makers to ensure a cost-effective solution, and the critical role a compliance program will play to ensure interoperability of AVB devices. Panelists will discuss the direct implications of AVB for system designers and talk about how best to prepare for the new standard. Ample time for questions from the audience will be allowed.

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Friday, November 5 3:00 pm – 4:00 pm
Room 122

Presenter: **Dave Isaac**

Thinking Outside the Box from Inside the Box

Whether it's your home workstation or small studio, a tour bus or a hotel room on the road, how do you record or mix professionally and creatively? David Isaac will help you understand what it is that you need to be able to express yourself and your music! Dave Isaac is a 3-time Grammy winning Producer/ Engineer, artist and author who has worked with Marcus Miller, Eric Clapton, Michael Jackson, Madonna, Luther Vandross, Puff Daddy, Whitney Houston, Anita Baker, Prince …… Dave has developed a video tutorial website (www.mixsensei.com) in which he and other Award winning engineers/producers and musicians share all their knowledge and "tricks of the trade"!

Friday, November 5 **3:00 pm** **Room 113**
Technical Committee Meeting on Electro Magnetic Compatibility

Friday, November 5 **3:30 pm** **Room 232**
Standards Committee Meeting SC-05-05 Grounding and EMC Practices

Broadcast/Media Streaming Session 8
Friday, November 5 **4:00 pm – 5:30 pm**
Room 133

LISTENER FATIGUE AND RETENTION

Chair: **David Wilson**, CEA

Panelists: *Sam Berkow*, SIA
Marvin Caesar
Frank Foti, Omnia
JJ Johnston, DTS Inc.
Sean Olive, Harman
Bill Sacks, Optimod.FM
Thomas Sporer, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

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

Student Event/Career Development
RECORDING COMPETITION SURROUND
Friday, November 5, 4:00 pm – 7:00 pm
Room 206

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. This event presents stereo and surround recordings in these categories:

- Surround Sound for Picture 4:00 pm to 5:00 pm
- Surround Classical 5:00 pm to 6:00 pm
- Surround Non-Classical 6:00 pm to 7:00 pm

The top three finalists in each category, as identified by our judges, present a short summary of their production intentions and the key recording and mix techniques used to realize their goals. They then play their projects for all who attend. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2) on Sunday afternoon.

The competition is a great chance to hear the work of your fellow students at other educational institutions. Everyone learns from the judges' comments even if your project isn't one of the finalists, and it's a great chance to meet other students and faculty.

Look online for the latest list of the generous sponsors supporting the Student Recording Competitions.

Friday, November 5 **4:00 pm** **Room 113**
Technical Committee Meeting on Loudspeakers and Headphones

Game Audio Session 7 **Friday, November 5**
4:15 pm – 5:15 pm **Room 120**

AUDIO SHORTS—SOUND DESIGN

Presenters: **Randy Buck**, Principal, The Sound Department, Austin, TX, USA
Charles Deenen, Senior Creative Director, Audio, Electronic Arts
Kristoffer Larson, Audio Manager, WB Games, Seattle, WA, USA
Marc Schaefgen, Principal/Owner, The Sound Department, Austin, TX, USA
Jay Weinland, Senior Audio Lead, Bungie Studios

Three mini sessions are presented by game audio dudes that guarantee you will walk away with cool new techniques. Twenty minutes each to serve up an in-depth look at topics in sound design that matter most to them. Q&A to follow.

Shorty #1: *Game Audio Sound Sourcing*—What is special about gathering sonic source material for games as opposed to other media? Need for longer ambiences (streams can be up to 5 minutes or more), more variation, more microphone perspectives, components rather than complex events.

Shorty #2: *The Loop Trick and More*—How to create seamless loops of any length and the best way to approach sound design for looping material. And, to loop or not to loop? That is always the question when it comes to rapid fire weapons. Learn a few techniques that go beyond the loop.

Shorty #3: *My Favorite Plugin!*—Three speakers from other sessions will talk about their current favorite plug-in, why they love it so much, and how they use and or abuse it.

Session P14 — Friday, Nov. 5
4:30 pm – 6:00 pm — Room 226

POSTERS: LOUDSPEAKERS AND MICROPHONES

4:30 pm

P14-1 Coaxial Flat Panel Loudspeaker System with Dynamic Push-Pull Drive—*Drazenko Sukalo*, DSLab – Device Solution Laboratory, Munich, Germany

After the successful introduction of the flat television, acousticians are concerned with the design of a "full-range" flat panel loudspeaker. A new design with low manufactured depth, consisting of an array of two conventional cone drivers and a transmission line and the method for driving of them is presented. The main aim was to build a small-sized flat panel box but with extended low frequency response and low distortion output because of the extended liner diaphragm excursion. The PSpice-OrCAD® simulator was used to represent a distributed model of a transmission line. The results of the simulation show the influence of the parameters of the transmission-line enclosure on the impedance curve and resonant frequency of the woofer driver. Among others, this paper is concerned with an active filter design for driving loudspeaker drive units in an appropriate phase relationship in the low frequency region, by means of implementing of DPP drive. A prototype of the flat panel loudspeaker is built according to the described design concept and the results of sound pressure level measurement are presented. The design result from work performed for DSLab and is subject to the referenced patent.
Convention Paper 8235
[Paper was not presented but is available for purchase]

4:30 pm

P14-2 A Novel Universal-Serial-Bus-Powered Digitally Driven Loudspeaker System with Low Power Dissipation and High Fidelity—*Hajime Ohtani,*[1] *Akira Yasuda,*[1] *Kenzo Tsuihiji,*[1] *Ryota Suzuki,*[1] *Daigo Kuniyoshi,*[1] *Junichi Okamura*[2]
[1]Hosei University, Koganei, Tokyo, Japan
[2]Trigence Semiconductor, Chiyoda, Tokyo, Japan

We propose a novel digitally driven loudspeaker system in which a newly devised mismatch shaper method, multilevel noise shaping dynamic element matching, is used to realize high fidelity, high sound power level, and low power dissipation. The unit used for the mismatch shaper method can easily increase the number of sound pressure levels with the aid of an H-bridge circuit, even when the number of sub-speakers is fixed. Further, it reduces the noise caused by quantization and loudspeaker mismatches and decreases the switching loss. The output sound power level equipped with six voice coils is 94 dB/m when a 3.3-V universal-serial-bus power supply is used exclusively. The power efficiency is 95% at 0 dBFS and 75% at –10 dBFS.
Convention Paper 8236

4:30 pm

P14-3 Loudspeaker Rub Fault Detection by Means of a New Nonstationary Procedure Test—*German Ruiz, Vicent Sala, Miguel Delgado, Juan Antonio Ortega*, UPC-Universitat Politecnica de Catalunya, Terrassa, Spain

This paper addresses rub defect loudspeaker detection. The study includes a simulation with a rub model based on classical static coulomb friction added to the loudspeaker nonlinearities parametric model to demonstrate the current signal viability to rub failure detection. The electric current signal is analyzed by means of Zhao-Atlas-Marks distribution (ZAMD). A failure extractor based on relevant harmonic ZAMD frequency regions segmentation and Mahalanobis distance is presented. The simulation and experimental results show the goodness and reliability of rub detection method presented.
Convention Paper 8237

4:30 pm

P14-4 Contributions to the Improvement of the Response of a Pleated Loudspeaker—*Jose Martinez,*[1] *Rita Martinez,*[1] *E. Segovia,*[2] *Jesus Carbajo,*[3] *Jaime Ramis*[3]
[1]Acustica Beyma S.L., Moncada, Valencia, Spain
[2]Obras Públicas e Infraestructura Urbana, Spain
[3]Universidad de Alicante, Alicante, Spain

In this paper we describe some results that have led to the improvement of the response of an Air Motion Transformer loudspeaker. First, it is noteworthy that it has been found an approximate analytical solution to the differential equations system that governs the behavior of the moving assembly of this type of transducer, being this valid when the length of the pleat is much greater than the radius of the cylindrical part. This solution is valid for any type of analysis (static, modal, and harmonic), and the modes are significantly simplified assuming the hypothesis above mentioned. In addition, we have analyzed the influence of the thickness and the shape of perforation of the pole piece in the frequency response of the loudspeaker.
Convention Paper 8238

4:30 pm

P14-5 Exploring the Ultra-Directional Acoustic Response of an Electret Cell Array Loudspeaker—*Yu-Chi Chen,*[1] *Wen-Ching Ko*,[1] *Chang-Ho Liou*,[1,2] *Wen-Hsin Hsiao*,[1] *Chih-Chiang Cheng,*[1] *Wen-Jong Wu,*[1] *Pei-Zen Chang*,[1] *Chih-Kung Lee*[1,3]
[1]National Taiwan University, Taipei, Taiwan
[2]Industrial Technology Research Institute, Hsinchu, Taiwan
[3]Institute for Information Industry, Taipei, Taiwan

In recent years, novel thin-plate loudspeakers have triggered much interest. Applications in areas such as 3C peripherals, automobile audio systems, and home theater have been actively discussed. However, the acoustic directivity of a thin-plate loudspeaker depends on the frequency response. At this time, thin-plate loudspeakers have poor directivity. However, if this limitation can be overcome, thin-plate loudspeakers can find useful applications such as in museums, supermarkets, or exhibition areas that require channeling the sound to a particular area or location without affecting nearby areas or unintended audiences. From previous studies, electret cell arrays have been confirmed to be an excellent flexible flat loudspeaker since it can create high performance sounds in a mid to high frequency range. An electret loudspeaker can generate ultra-directional audible sound by adjusting the array size, amplitude modulation, and layout structure.
Convention Paper 8239

4:30 pm

P14-6 A Soundfield Microphone Using Tangential Capsule Arrays—*Eric Benjamin*, Surround Research, Pacifica, CA, USA

The traditional soundfield microphone is a tetrahedral array of pressure gradient microphones, the outputs of which are linearly combined in order to realize signals that are proportional to co-located microphones, one with omnidirectional sensitivity and three orthogonal microphones with figure-of-eight sensitivity. This configuration works well and has been the basis of commercial products for a number of years. Recently, an alternative array type has been disclosed by Craven, Law, and Travis, comprised of pressure gradient sensors arranged with their principle axes oriented tangentially with respect to the center. Additional analysis has been performed and several prototypes were constructed and evaluated.
Convention Paper 8240

4:30 pm

P14-7 A 2-Way Loudspeaker Array System with Pseudorandom Spacing for Music Concerts —*Yuki Ayabe,*[1] *Saburo Nakano*,[1] *Kaoru Ashihara*,[2] *Shogo Kiryu*[1]
[1]Tokyo City University, Setagaya, Tokyo, Japan
[2]Advanced Industrial Science and Technology, Tsukuba, Japan

A 96-channel loudspeaker array system that allows real-time control of sound field has been developed for live musical concerts. Multiple sound focused at different points can been generated and controlled independently using the system. The variable delay circuits, the controller of the power amplifier, and the communication circuit between the hardware and the computer are implemented in FPGAs. In order to extend the frequency range and reduce the spatial aliasing, the loudspeaker array is assembled by two-way loudspeakers with pseudorandom spacing.
Convention Paper 8241

Workshop 10 **Friday, November 5**
4:30 pm – 6:30 pm **Room 130**

AUDIO NETWORK CONTROL PROTOCOLS

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

Panelists: *Bradford Benn*, Harman Corporation, Los Angeles, CA, USA
Richard Foss, Rhodes University, Grahamstown, South Africa
Jeff Koftinoff, Meyer Sound, Berkeley, CA, USA
Andy Schmeder, CNMAT - University of California Berkeley
Peter Stevens, BBC, London, UK

Digital audio networks have solved a number of problems related to the distribution of audio within a number of contexts, including recording studios, stadiums, convention centers, theaters, and live concerts. They provide cabling ease, better immunity to interference, and enhanced control over audio routing and signal processing when compared to analog solutions. There exist a number of audio network types, and also a number of audio network protocols, that define the messaging necessary for connection management and control of devices within networks. In this workshop a panel of audio network protocol experts will share the features of audio network protocols that they are familiar with and how network protocols might adapt and change over the next few years, bearing in mind the need for interoperability.

Workshop 11 **Friday, November 5**
4:30 pm – 6:00 pm **Room 132**

AES42 AND DIGITAL MICROPHONES

Chair: **Helmut Wittek**, SCHOEPS Mikrofone GmbH

Panelists: *Stephan Flock*, DirectOut GmbH
Tom Frey, Sennheiser
Stephan Peus, Georg Neumann GmbH

The AES42 interface for digital microphones is not yet widely used. This can be due to the relatively young appearance of digital microphone technology but also a lack of knowledge and practice with digital microphones and the corresponding interface exists. The advantages and disadvantages have to be communicated in an open and neutral way regardless of commercial interests but on the basis of the actual need of the engineers. Along with an available “White Paper” about AES42 and digital microphones, which is aimed at neutral in-depth information and which was compiled from different authors, the proposed workshop intends to enlighten facts and prejudices on this topic.

Workshop 12 Friday, November 5
4:30 pm – 6:30 pm Room 131

KEEP TURNING IT DOWN! DEVELOPING AN EXIT STRATEGY FOR THE LOUDNESS WARS

Chair: **Martin Walsh**, DTS Inc.

Panelists: *Bob Ludwig*, Gateway Mastering
Thomas Lund, TC Electronic
Susan Roberts, Berklee College of Music

Following on from the popular workshop presented at the 127th AES Convention that delved into topics relating to the nature and the consequences of the loudness wars, our panel of loudness experts and "master" mastering engineers will provide an update on the progress toward ending the war and returning peace, harmony, and dynamic range to the people.

The workshop will focus on alternatives to the practice of overly aggressive dynamic range compression using weapons such as seasoned mastering techniques and gain normalization algorithms and standards. Audience participation is encouraged and all are welcome to voice their own opinion and comments in relation to the issues discussed.

Exhibitor Seminar
NARAS CCD
Friday, November 5 5:00 pm – 6:00 pm
Room 112

Presenter: **John Spencer**

Credit Where Credit Is Due: Metadata!

Accurate metadata is critical for any profitable recording industry model. Most of today's commercial recording projects are "born digital," requiring a new paradigm for how projects are documented, distributed, and archived. This presentation demonstrates the Library of Congress funded multitrack project that created CCD (Content Creator Data), a standardized schema, data dictionary, field set, and free studio collection application for gathering the technical, descriptive, and participant information associated with recording projects. CCD can provide the dynamic, end to end documentation necessary to connect the dots and facilitate e-copyright, e-commerce, and archiving.

Friday, November 5 5:00 pm Room 113
Technical Committee Meeting on Fiber Optics for Audio

Friday, November 5 5:00 pm Room 232
Standards Committee Meeting SC-02-08 Audio File Transfer and Exchange

Game Audio Session 8 Friday, November 5
5:30 pm – 6:30 pm Room 120

CODE MONKEY PART 3: <LEARN>XML</LEARN>

Presenter: **Michael Kelly**, Senior Audio Engineer, Sony Computer Entertainment Europe

This session provides an overview of XML, the eXtensible Markup Language, and explains how it is used within the game-audio production pipeline. This is a chance to pick up some pointers how to read and write in XML. This introductory-level session is for those who work with XML (whether you know it or not!) and want to know more. The session may also be beneficial to those outside the games industry and shows how some popular game-audio tools use XML. The sessions ends with some pointers to more advanced topics for the adventurous.

Broadcast/Media Streaming Session 9
Friday, November 5 5:45 pm – 7:00 pm
Room 133

AUDIO PROCESSING FOR STREAMING: FINDING THE SILVER LINING IN THE INTERNET CLOUD

Chair: **Bill Sacks**, Optimod.FM

Panelists: *Ray Archie*, CBS
Frank Foti, Telos
Greg Ogonowski, Orban
Skip Pizzi, Consultant/Radio Ink

Traditional media people from broadcast and recording have a far different perspective from IT people. Compression now means two far different things to a seasoned audio engineer depending if the conversation is about dynamic range reduction or a data stream's efficiency. We must learn to communicate with one another. We must learn the TCP/IP language and protocols well enough to relate the needs to our IT counterparts and we need to be able to teach them, in their language, what we need them to do with us in order to accomplish our mission. This panel will discuss the evolving relationship of audio and IT and how to improve not just the technical interfaces to learn, but the mutual understanding of our needs.

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Friday, November 5, 7:00 pm – 8:00 pm
Room 134

Lecturer: **Ben Burtt**

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 129th AES Convention is Ben Burtt. Burtt's credits and accomplishments are the stuff of legend; his contributions are integrally woven into a number of iconic and Oscar-winning films. Showing a rare fluency across many disciplines, Burtt has worked in every facet of film production for over 35 years: directing, producing, sound design, sound editing, editing, voicing, animation, visual effects, and voice design in motion pictures, television, specialty, educational, and documentary. He was the Sound Designer for the Star Wars and Indiana Jones series, has been nominated for 12 Academy Awards for sound effects work, and has won four Oscars. He has created globally and instantly recognized sounds like the light saber and characters like Wall-E and R2-D2. Burtt's credits include the IMAX film *Special Effects*, for which he was nominated for directing in the

Best Short Documentary category of the Academy Awards. In 2008 he received the Oscar nominations for Best Sound Editing and Best Sound Mixing for Pixar-Disney's *Wall-E*. In addition, he voiced the main character, Wall-E and Wall-E's robot sidekicks. His most recent sound design was for JJ Abram's *Star Trek*. He is currently at work on Lucasfilm's *Red Tails*, and Disney's upcoming feature *John Carter of Mars*. Burtt holds a BS in Physics from Allegheny College, a Masters in Film Production from the University of Southern California, and a Doctorate of Arts from Allegheny College. He is married to Margaret Burtt and has four children and five, no, make that six wonderful grandchildren. The title of his lecture is, "The Sound Behind the Image."

Much has been documented about the technical history of motion picture sound. We know a lot about the story of microphones, loudspeakers, and optical, magnetic, or digital recording processes. Very little has been said about the aesthetic history: Why do sound people do what we do? What have been the creative achievements? The great ideas? How has sound been used to enhance the image and give vast dramatic power to the feature film?

"The Sound Behind The Image" will walk us through cinema history from the silent film to 1977 when Burtt designed sounds for *Star Wars*. He will talk more about the ART of film sound than the SCIENCE. He will pinpoint and show the moments in American film history that inspired and allowed him to learn his craft in sound design. Burtt believes a Language of Sound developed in the classic era that is still the basis for all our creative sound work today. Let us study, learn, speak, and enjoy that language together.

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

Student Event/Career Development
STUDENT SOCIAL
Friday, November 5, 8:00 pm – ???

Join us for a fun and exciting evening at the Student Social. On Friday night, Nov. 5th, at 8 pm the fun begins! Don't miss this chance to meet and engage with students from all over the world! The social will cost ten dollars per student. Get your ticket at the Student Science Spot or one of the SDA leaders. More information and details will be provided at SDA-1. Or contact vice-chair of the Americas, Philip Parenteau at philip@aes-sda.org

Session P15 **Saturday, Nov. 6**
9:00 am – 1:00 pm **Room 220**

MULTICHANNEL AUDIO PLAYBACK

Chair: **Alex Voishvillo**, JBL Professional, CA, USA

9:00 am

P15-1 Why Ambisonics Does Work—*Eric Benjamin*,[1] *Richard Lee*,[2] *Aaron Heller*[3]
[1]Surround Research, Pacifica, CA, USA
[2]Pandit Littoral, Cooktown, Queensland, Australia
[3]SRI International, Menlo Park, CA, USA

Several techniques exist for surround sound, including Ambisonics, VBAP, WFS, and pair-wise panning. Each of the systems have strengths and weaknesses but Ambisonics has long been favored for its extensibility and for being a complete solution, including both recording and playback. But Ambisonics has not met with great critical or commercial success despite having been available in one form or another for many years. Some observers have gone so far as to suggest that Ambisonics can't work. The present paper is intended to provide an analysis of the performance of Ambisonics according to various psychoacoustic mechanisms in spatial hearing, such as localization and envelopment.
Convention Paper 8242

9:30 am

P15-2 Design of Ambisonic Decoders for Irregular Arrays of Loudspeakers by Non-Linear Optimization—*Aaron Heller*,[1] *Eric Benjamin*,[2] *Richard Lee*[3]
[1]SRI International, Menlo Park, CA, USA
[2]Surround Research, Pacifica, CA, USA
[3]Pandit Littoral, Cooktown, Queensland, Australia

In previous papers, the present authors described techniques for design, implementation, and evaluation of Ambisonic decoders for regular loudspeaker arrays. However, to accommodate domestic listening rooms, irregular arrays are often required. Because the figures of merit used to predict decoder performance are non-linear functions of loudspeaker positions, non-linear optimization techniques are needed. In this paper we discuss the implementation of an open-source application based on the NLopt non-linear optimization software library that derives decoders for arbitrary arrays of loudspeakers, as well as providing a prediction of their performance using psychoacoustic criteria, such as Gerzon's velocity and energy localization vectors. We describe the implementation and optimization criteria and report on listening tests comparing the decoders produced.
Convention Paper 8243

10:00 am

P15-3 Discrete Driving Functions for Horizontal Reproduction Using Wave Field Synthesis and Higher Order Ambisonics—*César D. Salvador*, Universidad de San Martín de Porres, Lima, Peru

Practical implementations of physics-based spatial sound reproduction techniques, such as Wave Field Synthesis (WFS) and Higher Order Ambisonics (HOA), require real-time filtering, scaling, and delaying operations on the audio signal to be spatialized. These operations form the so-called loudspeaker's driving function. This paper describes a discretization method to obtain a rational representation in the *z*-plane from the continuous WFS and HOA driving functions. Visual and numerical comparisons between the continuous and discrete driving functions, and between the continuous and discrete sound pressure fields, synthesized with circular loudspeaker arrays, are shown. The percentage discretization errors, in the reproducible frequency range and in the whole listening area, are in the order of 1%. A methodology for the reconstruction of immersive soundscapes composed with nature sounds is also reported as a practical application.
Convention Paper 8244

10:30 am

P15-4 Reducing Artifacts of Focused Sources in Wave Field Synthesis—*Hagen Wierstorf, Matthias Geier, Sascha Spors*, Technische Universität Berlin, Berlin, Germany

Wave Field Synthesis provides the possibility to synthesize virtual sound sources located between the loudspeaker array and the listener. Such sources are known as focused sources. Previous studies have shown that the reproduction of focused sources is subject to audible artifacts. The strength of those artifacts heavily depends on the size of the loudspeaker array. This paper proposes a method to reduce artifacts in the reproduction of focused sources by using only a subset of loudspeakers of the array. A listening test verifies the method and compares it to previous results.
Convention Paper 8245

11:00 am

P15-5 On the Anti-Aliasing Loudspeaker for Sound Field Synthesis Employing Linear and Circular Distributions of Secondary Sources —*Jens Ahrens, Sascha Spors*, Deutsche Telekom Laboratories, Technische Universität Berlin, Berlin, Germany

The theory of analytical approaches for sound field synthesis like wave field synthesis, nearfield compensated higher order Ambisonics, and the spectral division method requires continuous distributions of secondary sources. In practice, discrete loudspeakers are employed and the synthesized sound field is corrupted by a number of artifacts that are commonly referred to as spatial aliasing. This paper presents a theoretical investigation of the properties of the loudspeakers that are required in order to suppress such spatial aliasing artifacts. It is shown that the employment of such loudspeakers is not desired since the suppression of spatial aliasing comes by the cost of an essential restriction of the reproducible spatial information when practical loudspeaker spacings are assumed.
Convention Paper 8246

11:30 am

P15-6 The Relationship between Sound Field Reproduction and Near-Field Acoustical Holography—*Filippo Fazi, Philip Nelson*, University of Southampton, UK

The problem of reproducing a desired sound field with an array of loudspeakers and the technique known as Near-Field Acoustical Holography share some fundamental theoretical aspects. It is shown that both problems can be formulated as an integral equation that usually defines an ill-posed problem. The example of spherical geometry and planar geometry is discussed in detail. It is shown that for both the reproduction and the acoustical holography cases, the ill-conditioning of the problem is greatly affected by the distance between the source layer and the measurement/control surface.
Convention Paper 8247

12:00 noon

P15-7 Surround Sound with Height in Games Using Dolby Pro Logic IIz— *Nicolas Tsingos,*[1] *Christophe Chabanne,*[1] *Charles Robinson,*[1] *Matt McCallus*[2]
[1]Dolby Laboratories, San Francisco, CA, USA
[2]RedStorm Entertainment, Cary, NC, USA

Dolby Pro Logic IIz is a new matrix encoding/decoding system that enables the transmission of a pair of height channels within a conventional surround sound stream (e.g. 5.1). In this paper we provide guidelines for the use of Pro logic IIz for interactive gaming applications including recommended speaker placement, creation of elevation information, and details on how to embed the height channels within a 5- or 7-channel stream. Surround sound with height is already widely available in home-theater receivers. It offers increased immersion to the user and is a perfect fit for 2-D or stereoscopic 3-D video games.
Convention Paper 8248

12:30 pm

P15-8 Optimal Location and Orientation for Midrange and High Frequency Loudspeakers in the Instrument Panel of an Automotive Interior—*Roger Shively*,[1] *Jérôme Halley*,[2] *François Malbos*,[3] *Gabriel Ruiz*[4]
[1]Harman International, Novi, MI, USA
[2]Harman International, Karlsbad, Germany
[3]Harman International, Chateau du Loir, France
[4]Harman International, Bridgend, Wales, UK

In a follow-up to a previous paper (AES Convention Paper # 8023, May 2010) using the modeling process described there for modeling loudspeakers in an automotive interior, the optimization of midrange and of high frequency tweeter loudspeakers' positions for best acoustic performance in the driver's side (left) and passenger's side (right) of automotive instrument panel is reported on.
Convention Paper 8249

Session P16 **Saturday, Nov. 6**
9:00 am – 12:30 pm **Room 236**

SIGNAL ANALYSIS AND SYNTHESIS

Chair: **Agnieszka Roginska**, New York University, New York, NY, USA

9:00 am

P16-1 Maintaining Sonic Texture with Time Scale Compression by a Factor of 100 or More— *Robert Maher*, Montana State University, Bozeman, MT, USA

Time lapse photography is a common technique to present a slowly evolving visual scene with an artificially rapid temporal scale. Events in the scene that unfold over minutes, hours, or days in real time can be viewed in a shorter video clip. Audio time scaling by a major compression factor can be considered the aural equivalent of time lapse video, but obtaining meaningful time-compressed audio requires interesting practical and conceptual challenges in order to retain the original sonic texture. This paper reviews a variety of existing techniques for compressing 24 hours of audio into just

a few minutes of representative "time lapse" audio and explores several useful modifications and optimizations.
Convention Paper 8250

9:30 am

P16-2 Sound Texture Analysis Based on a Dynamical Systems Model and Empirical Mode Decomposition—*Doug Van Nort, Jonas Braasch, Pauline Oliveros*, Rensselaer Polytechnic Institute, Troy, NY, USA

This paper describes a system for separating a musical stream into sections having different textural qualities. This system translates several contemporary approaches to video texture analysis, creating a novel approach in the realm of audio and music. We first represent the signal as a set of mode functions by way of the Empirical Mode Decomposition (EMD) technique for time/frequency analysis, before expressing the dynamics of these modes as a linear dynamical system (LDS). We utilize both linear and nonlinear techniques in order to learn the system dynamics, which leads to a successful separation of the audio in time and frequency.
Convention Paper 8251

10:00 am

P16-3 An Improved Audio Watermarking Scheme Based on Complex Spectral Phase Evolution Spectrum—*Jian Wang, Ron Healy, Joe Timoney*, NUI Maynooth, Co Kildare, Ireland

In this paper a new audio watermarking algorithm based on the CSPE algorithm is presented. This is an extension of a previous scheme. Peaks in a spectral representation derived from the CSPE are utilized for watermarking, instead of the previously proposed frequency identification. Although this new scheme is simple, it achieves a high robustness besides perceptual transparency and accuracy which is one distinguishing advantage over our previous scheme.
Convention Paper 8252

10:30 am

P16-4 About This Dereverberation Business: A Method for Extracting Reverberation from Audio Signals— *Gilbert A. Soulodre*, Camden Labs, Ottawa, Ontario, Canada

There are many situations where the reverberation found in an audio signal is not appropriate for its final use, and therefore we would like to have a means of altering the reverberation. Furthermore we would like to be able to modify this reverberation without having to directly measure the acoustic space in which it was recorded. In the present paper we describe a method for extracting the reverberant component from an audio signal. The method allows an estimate of the underlying dry signal to be derived. In addition, the reverberant component of the signal can be altered.
Convention Paper 8253

11:00 am

P16-5 Automatic Recording Environment Identification Using Acoustic Features—*Usman Amin Chaudhary, Hafiz Malik*, University of Michigan-Dearborn, Dearborn, MI, USA

Recording environment leaves its acoustic signature in the audio recording captured in it. For example, the persistence of sound, due to multiple reflections from various surfaces in a room, causes temporal and spectral smearing of the recorded sound. This distortion is referred to as audio reverberation time. The amount of reverberation depends on the geometry and composition of a recording location, the difference in the estimated acoustic signature can be used for recording environment identification. We describe a statistical framework based on maximum likelihood estimation to estimate acoustic signature from the audio recording and use it for automatic recording environment identification. To achieve these objectives, digital audio recording is analyzed first to estimate acoustic signature (in the form of reverberation time and variance of the background noise), and competitive neural network based clustering is then applied to the estimated acoustic signature for automatic recording location identification. We have also analyzed the impact of source-sensor directivity, microphone type, and learning rate of clustering algorithm on the identification accuracy of the proposed method.
Convention Paper 8254

11:30 am

P16-6 Automatic Music Production System Employing Probabilistic Expert Systems— *Gang Ren, Gregory Bocko, Justin Lundberg, Dave Headlam, Mark F. Bocko*, University of Rochester, Rochester, NY, USA

An automatic music production system based on expert audio engineering knowledge is proposed. An expert system based on a probabilistic graphical model is employed to embed professional audio engineering knowledge and infer automatic production decisions based on musical information extracted from audio files. The production pattern, which is represented as a probabilistic graphic model, can be "learned" from the operation data of a human audio engineer or manually constructed from domain knowledge. The authors also discuss the real-time implementation of the proposed automatic production system for live mixing application scenarios. Musical event alignment and prediction algorithms are introduced to improve the time synchronization performance of our production model. The authors conclude with performance evaluations and a brief summary.
Convention Paper 8255

12:00 noon

P16-7 Musical Eliza: An Automatic Musical Accompany System Based on Expressive Feature Analysis— *Gang Ren, Justin Lundberg, Gregory Bocko, Dave Headlam, Mark F. Bocko*, University of Rochester, NY, USA

We propose an interactive algorithm that musically accompanies musicians based on the matching of expressive feature patterns to existing archive recordings. For each accompany music segment, multiple realizations with differ- ➥

ent musical characteristics are performed by master music performers and recorded. Musical expressive features are extracted from each accompany segment and its semantic analysis is obtained using music expressive language model. When the performance of system user is recorded, we extract and analyze musical expressive feature in real time and playback the accompany track from the archive database that best matches the expressive feature pattern. By creating a sense of musical correspondence, our proposed system provides exciting interactive musical communication experience and finds versatile entertainment and pedagogical applications.
Convention Paper 8256

Tutorial 8 Saturday, November 6
9:00 am – 10:45 am Room 206

SPATIAL AUDIO REPRODUCTION: FROM THEORY TO PRODUCTION

Presenters: **Frank Melchior**, IOSONO GmbH, Germany
Sascha Spors, Deutsche Telekom AG Laboratories, Germany

Advanced high-resolution spatial sound reproduction systems like Wave Field Synthesis (WFS) and Higher-Order Ambisonics (HOA) are being used increasingly. Consequently more and more material is being produced for such systems. Established channel-based production processes from stereophony can only be applied to a certain extent. In the future, a paradigm shift toward object-based audio production will have to take place in order to cope for the needs of systems like WFS. This tutorial spans the bridge from the physical foundations of such systems, over their practical implementation toward efficient production processes. The focus lies on WFS, however the findings will also be applicable to other systems. The tutorial is accompanied by practical examples of object-based productions for WFS.

Broadcast/Media Streaming Session 10
Saturday, November 6 9:00 am – 10:30 am
Room 133

AUDIO OVER IP: A TUTORIAL

Presenters: **Steve Church**, Telos Systems
Skip Pizzi, Media Technology Consultant & Technology Editor, *Radio Ink* magazine

IP-based networking continues to grow in popularity among broadcast, sound reinforcement, and other audio facilities. Learn the latest from the men who “wrote the book” on AoIP in this session, which will cover topics ranging from general advantages to specific applications of this groundbreaking new technology.

Game Audio Session 9 Saturday, November 6
9:00 am – 11:00 am Room 120

GAME INDUSTRY OVERVIEW

Chair: **Marc Schaefgen**, Principal/Owner, The Sound Department, Austin, TX, USA

Panelists: *Lance Brown*, Cinematic Game Audio Consultant
Charles Deenen, Senior Creative Director, Audio, Electronic Arts
Adam Levenson, Senior Director, Central Audio and Talent, Activision Blizzard

How does the game industry work compared to other media industries? What is the game development process? Unlike other media industries, the deployment platforms for games are constantly evolving as are the tools used to create content for said platforms. What if the TV industry experienced an evolution like HD every five years? How do game developers sail the seas of technical change? How does the technology affect creativity? Where can I fit in if I'm coming from a related media industry? Lots of good questions, those and more answered by a panel of top game audio professionals.

Live Sound Seminar 6 Saturday, November 6
9:00 am – 10:45 am Room 131

SUBWOOFER DIRECTIONALITY

Chair: **Charlie Hughes**, Excelsior Audio

Panelists: *Ales Dravinec*, ADRaudio
Bill Gelow, Bosch/Electrovoice
Dave Rat, Rat Sound

Directional subwoofers and subwoofer arrays can help to keep low frequency energy on the audience, where it’s desired, and away from areas where it’s not. This can be a great help in reducing rumble on stage and increasing gain before feedback. Single enclosure subwoofers with directivity control will be discussed along with arrays of multiple enclosures and their directional properties. Join us to find out how the spacing, loading, and signal processing of individual loudspeaker elements help to yield subwoofer directionality.

Product Design Session 4 Saturday, November 6
9:00 am – 10:45 am Room 132

GROUNDING AND SHIELDING—CIRCUITS AND INTERFERENCE—PART 1

Presenter: **Ralph Morrison**

This first session will discuss the way signals and power are transported. We will discuss the basic meanings of words such as voltage, current, capacitance, and inductance; the role conductor geometries have in controlling where signals and power can travel; the problems of utility and facility design together with the meaning of ground and earth; the interference problems created by transformers and facility wiring; a discussion of shielding as applied to analog circuits and radiating structures. Terms such as differential, balanced, common-mode, normal-mode, and single ended will be explained.

Saturday, November 6 9:00 am Room 113
Technical Committee Meeting on Studio Practices and Production

Saturday, November 6 9:00 am Room 232
Standards Committee Meeting SC-04-04 Microphone Measurement and Characterization

Student Event/Career Development
CAREER/JOB FAIR
Saturday, November 6, 10:00 am – 11:30 am
Concourse

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 visit with representatives from the companies and find out more about job and internship opportunities in the audio industry. Bring your resume!

Saturday, November 6 10:00 am Room 113
Technical Committee Meeting on Hearing and Hearing Loss Prevention

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Saturday, November 6 10:15 am – 11:15 am
Room 122

Presenter: **David Miles Huber**

Intelligent Dance Music in 5.1

David Miles Huber will be presenting highlights from his Grammy-nominated project *Colabs*, as well as his latest release *Parallax Eden* in 5.1 surround sound. DMH will also be sharing insights into his production techniques, philosophies, and "toys-of-the-trades" for making his own style of IDM (intelligent dance music).

David Miles Huber is a Grammy-nominated producer and musician in the electronic IDM, dance, and surround-sound genres, whose music has sold over the million copies. His latest music and collaborations can be heard on www.davidmileshuber.com

Tutorial 9 Saturday, November 6
10:30 am – 12:30 pm Room 130

DAMPING OF THE ROOM LOW-FREQUENCY ACOUSTICS (PASSIVE AND ACTIVE)

Presenters: **Reza Kashani**, University of Dayton, Dayton, OH, USA
Jim Wischmeyer, Bag End Loudspeaker Systems, Lake Barrington, IL, USA

As the result of its size and geometry, a room excessively amplifies sound at certain frequencies. This is the result of standing waves (acoustic resonances/modes) of the room. These are waves whose original oscillation is continuously reinforced by their own reflections. Rooms have many resonances, but only the low-frequency ones are discrete, distinct, unaffected by the sound absorbing material in the room, and accommodate most of the acoustic energy build up in the room.

In this tutorial, after discussing the low frequency room acoustics, different passive and active bass trapping techniques for adding damping to a room will be talked about and their advantages/disadvantages discussed. The event will conclude by comparing/contrasting damping with equalizing.

Tutorial 10 Saturday, November 6
11:00 am – 1:00 pm Room 132

NETWORKING FOR AUDIO APPLICATIONS

Presenters: **Bradford Benn**, Harman Corporation, Elkhart, IN, USA
Robert Economaki, Cisco, Des Moines, IA, USA
Kevin Gross, AVA Networks, Denver, CO, USA

Computer networks, principally Ethernet and IP networks, are becoming a common means of distributing audio within a facility and over distance. Audio engineers need an understanding of how networks work and how to use them effectively.

The tutorial will introduce participants to terminology; technologies and standards; configuration and design considerations; and troubleshooting concepts. Live demonstrations will reinforce specific topics including Network equipment configuration and management; IP subnetting, addressing and routing; and use of diagnostic tools.

The tutorial will give participants the background required to work effectively with IT professionals implementing and maintaining networked audio distribution.

Workshop 13 Saturday, November 6
11:00 am – 1:00 pm Room 206

PROGRESS IN COMPUTER-BASED PLAYBACK OF HIGH RESOLUTION AUDIO

Chair: **Vicki R. Melchior**, Audio DSP Consultant, Boston, MA, USA

Panelists: *Bob Bauman*, Lynx Studio Technology, Cosa Mesa, CA, USA
James Johnston, DTS Inc., Calabasas, CA, USA
Andy McHarg, dCS Ltd., Cambridge, UK
Daniel Weiss, Weiss Engineering, Zurich, Switzerland

With the continuing decline in discs as music sources and concurrent growth of electronic distribution, computers and network attached storage (NAS) are now rapidly evolving as front end components in place of traditional transports and players. Computers have long been useful within mastering workflows, though not always loved, and their introduction into high quality music systems raises a new range of engineering challenges.

Intrinsic to computers are problems of EMC, switching noise, dirty power, jittered clocks, crosstalk, driver and operating system variability, protocol incompatibilities, and software errors, to name a few. These may directly influence audio quality. Of special importance, for example, are the design as well as system configuration of digital audio interfaces (USB, Firewire, S/PDIF, WiFi, Ethernet etc), D/A conversion, and data processing, along with clocks and power sourcing.

The panel in this workshop are active in the design of these systems and will discuss some of their results and thoughts regarding the most salient factors for optimization of sonic performance in this area.

Broadcast/Media Streaming Session 11
Saturday, November 6 11:00 am – 12:30 pm
Room 133

AUDIO PROCESSING FOR RADIO

Chair: **Tom Ray**, Buckley Broadcasting/WOR Radio, New York, NY, USA

Panelists: *Steve Fluker*, Cox Media, Orlando, FL, USA
Frank Foti, Omnia
Jeff Keith, Wheatstone Corporation
Robert Orban, Orban

There is much discussion as to why radio stations are ➥

"overprocessed"—a term that is true or not depending on your point of view. This panel will be discussing audio processing in the radio environment. There will be a brief discussion of audio processing history, up to and including the advantages of using digital processors. And radio today is not just an analog medium—we will discuss do's and don't's for processing radio in the digital realm—and try taking a look into the future.

Live Sound Seminar 7 Saturday, November 6
11:00 am – 1:00 pm Room 131

LIVE SOUND FOR CORPORATE EVENTS: WHY IT'S NOT "JUST" TALKING HEADS!

Chair: **Ken Newman**, Audio Applications

Panelists: *Rich Halvorson*, Presentation Audio
Michael Jackson, BPIAudio
Kevin McKereghan, BBI Engineering

Corporate sound events require a high degree of perfection in sound delivery, but "the look" often takes precedence over loudspeaker placement. Signal flow and mixing can be very complex. Strategies and solutions will be discussed using example cases.

Saturday, November 6 11:00 am Room 113
Technical Committee Meeting on Transmission and Broadcasting

Saturday, November 6 11:00 am Room 232
Standards Committee Meeting SC-02-01 Digital Audio Measurement Techniques

Game Audio Session 10 Saturday, November 6
11:15 am – 12:45 pm Room 120

AUDIO CAGE MATCH!

Referee: **Steve Horowitz**, Composer/Musician/Referee, The Code International Inc.

Presenters: **Peter "pdx" Drescher**, Sound Designer, Twittering Machine
Larry the O

A spirited "cage match" discussion about sound, music, interactive audio, game soundtracks, hardware, software, and audio production, by two experienced industry veterans. Peter Drescher and Larry the O met at Berklee College of Music in 1976 and have been arguing both sides of any audio issue ever since. A series of questions will be asked by the moderator; each will be discussed for a specific amount of time. Topics might include:

- "I promise never to program a computer to play something I can't" (aaaand ... fight!)
- How much does audio quality really matter?
- Why does MIDI sound bad, and what's it good for anymore?
- Is music supposed to be easy to learn, or difficult?
- Why do game soundtracks have to SUCK so bad!?
- Who needs interactive audio anyway?
- The Myth of Music Ownership: music as "a service you listen to," not "a thing that you buy."
- Have advances in technology actually made audio production better?
- (Stay tuned ... more to come)

Referee will be provided. Trained medical personnel standing by. Remember, this is not a competition, it is only an exhibition—please, no wagering. After all, there is always the remote chance these guys might agree on something!

Special Event
PLATINUM ARTISTS
Saturday, November 6, 11:30 am – 1:00 pm
Room 134

Moderator: **David Goggin**

Panelists: *Bruce Botnick*
Corey Cunningham
KamranV
Ray Manzarek
Veronica Romeo
CJ Vanston

World-renowned recording artists, producers and engineers share their insights into the recording craft. What do artists look for in producers, engineers, and studios? How has recording changed since multitracking burst on the scene? What were some of the magical moments they experience in the studio? The panel will consist of the following:

- Legendary Doors keyboardist Ray Manzarek and band's producer, Bruce Botnick
- Spanish pop star Veronica Romeo and her producer, CJ Vanston
- Corey Cunningham of up-and-coming San Francisco-based rock band Magic Bullets and the band's producer, KamranV
- Moderator David Goggin, noted recording industry author and photographer

Student Event/Career Development
STUDENT RECORDING CRITIQUES
Saturday, November 6, 11:30 am – 12:30 pm
Room 122

Chairs: **Ian Corbett**, Kansas City Community College, Kansas City, KS, USA
David Greenspan, University of Michigan, Ann Arbor, MI, USA

Students are encouraged to bring in their stereo or surround projects to this non-competitive listening session for feedback and comments from a panel and audience. Students will be able to sign-up for time slots at the first SDA meeting and at the Student Science Spot, on a first come, first served basis. Students who are finalists in the Recording Competition are excluded from participating in this event to allow the many non-finalists an opportunity for feedback on their hard work. Bring your stereo or surround work on CD, DVD, or hard disc as clearly-labelled .wav files. The Student Recording Critiques are generously sponsord by PMC.

Saturday, November 6 12:00 noon Room 113
Technical Committee Meeting on Signal Processing

Student Event/Career Development
EDUCATION FAIR
Saturday, November 6, 12:30 pm – 2:00 pm
Concourse

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.

Historical Event
THE LIBRARY OF CONGRESS HISTORICAL AUDIO COLLECTIONS AND NEW FACILITY
Saturday, November 6, 1:00 pm – 2:00 pm
Room 133

Presenter: **Brad McCoy**, Senior Studio Engineer, Library of Congress, Packard Campus for Audio Visual Conservation, Culpeper, VA, USA

The Library of Congress houses vast collections of historical audio recordings in a wide variety of formats. McCoy will play noteworthy examples from the collection and describe the Library's new facility in Culpeper, VA. He will also discuss the audio preservation methods used for these important historical recordings.

Saturday, November 6 1:00 pm Room 113
Technical Committee Meeting on Automotive Audio

Saturday, November 6 1:00 pm Room 232
Standards Committee Meeting SC-04-03 Loudspeaker Modeling and Measurement

Special Event
SOCIAL MEDIA FOR MUSICIANS AND ENGINEERS —PART 2
Saturday, November 6, 1:15 pm – 2:15 pm
Room 130

Presenter: **Bobby Owsinski**

MySpace, Facebook, YouTube, and Twitter are important elements for developing a fanbase, but without the proper strategy they can prove ineffective and take so much time that there's no time left for creating. This presentation shows artists, bands, musicians, and audio professionals the best techniques to utilize social media as a promotional tool without it taking 20 hours a day.

Topics covered include:
- Social Media management strategies
- Optimizing your YouTube presence
- The secrets of viral videos
- Search Engine Optimization basics
- Using Facebook and Twitter for marketing
- The secret behind successful tweets
- Is MySpace dead?
- What's next?

Special Event
LUNCHTIME KEYNOTE: ADAM LEVENSON
Saturday, November 6, 1:15 pm – 2:15 pm
Room 120

Presenter: **Adam Levenson**, Senior Director, Central Audio and Talent, Activision Blizzard

The Trappings of Hollywood

Videogame publishers have been striving to emulate and outshine the entertainment value of blockbuster Hollywood films and primetime TV for decades. Audio presentation ranging from mix, to music, effects, voice, and related storytelling are essential aspects to reaching this elusive goal. What are the aesthetic, technical, and business similarities and differences between producing game audio and other mainstream entertainment media? How have games outdone or fallen short of the audio presentation value delivered in film and TV? Does celebrity talent in acting, scriptwriting, or composing drive sales, enhance, or detract from game play? During the next decade, how will the dramatic increase in entertainment choices and our changing economy influence consumer appetite for the trappings of Hollywood in games?

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Saturday, November 6 1:30 pm – 2:30 pm
Room 122

Presenter: **Darius "Deezle" Harrison**

What Makes it Sound Good?!

Darius "Deezle" Harrison is a 2-time Grammy Award, 4-time ASCAP Rhythm and Soul Award, and a B.E.T Award winner. He will discuss the process of making a song sound good from a producer's perspective, whether using artificial sounds or acoustical instruments. Deezle is mostly known for his work on 3 albums with Lil Wayne, which ultimately garnered him 2 Grammy Awards for Best Rap Album and best Rap Song for "Lollipop." Deezle has become a much sought-after producer who has worked alongside many other artists such as Outkast, Ludacris, Mary J Blige, Usher, Donald Harrison, Jay-Z, and Kerry Hilson. While planning the growth of his Drum Major label and brand, Deezle is currently putting his producing and engineering skills to work on his upcoming album *Superstar*, which will be a collaboration with multiple artists.

Exhibitor Seminar
SCHOEPS
Saturday, November 6 2:00 pm – 3:00 pm
Room 112

Presenter: **Helmut Wittek**

Schoeps SuperCMIT: Practical Questions, Applications, Samples, and Hands-On Demos

This new digital super-shotgun microphone offers possibilities for the sound engineer that have not existed before. The seminar presents this with support from a collection of audio samples and useful comparisons to state-of-the-art microphones. New workflow and answers to practical questions on interfacing and applications are explained. The use of the SuperCMIT's different integral DSP algorithm presets is explained. Recent experience from professional users is presented. Any other related experience, topic, or comment brought up by the audience is appreciated.

Saturday, November 6 2:00 pm Room 113
Technical Committee Meeting on Microphones and Applications

Session P17 Saturday, Nov. 6
2:30 pm – 6:30 pm Room 220

REAL-TIME AUDIO PROCESSING

Chair: **Jayant Datta**, THX, Ltd., San Rafael, CA, USA

2:30 pm

P17-1 A Time Distributed FFT for Efficient Low Latency Convolution—*Jeffrey Hurchalla*, Garritan Corp., Orcas, WA, USA

To enable efficient low latency convolution, a Fast Fourier Transform (FFT) is presented that balances processor and memory load across incoming blocks of input. The proposed FFT ➥

transforms a large block of input data in steps spread across the arrival of smaller blocks of input and can be used to transform large partitions of an impulse response and input data for efficiency, while facilitating convolution at very low latency. Its primary advantage over a standard FFT as used for a non-uniform partition convolution method is that it can be performed in the same processing thread as the rest of the convolution, thereby avoiding problems associated with the combination of multithreading and near real-time calculations on general purpose computing architectures.
Convention Paper 8257

3:00 pm

P17-2 An Infinite Impulse Response (IIR) Hilbert Transformer Filter Design Technique for Audio—*Daniel Harris*,[1] *Edgar Berdahl*,[2] *Jonathan S. Able*[2]
[1]Sennheiser Research Laboratory, Palo Alto, CA, USA
[2]Stanford University, Stanford, CA, USA

Hilbert Transformers have found many applications in the signal processing community, from single-sideband communication systems to audio effects. IIR implementations are attractive for computationally sensitive systems due to their lower number of coefficients. However, as in any advanced filter design problem, their tuning and implementation present a number of design challenges and tradeoffs. Furthermore, while literature addressing these problems exists, designers must draw from several sources to find answers. In this paper we present a complete start-to-finish explanation of how to implement an efficient infinite impulse response (IIR) Hilbert transformer filter. We start from a half-band filter design and show how the poles move as the half-band filter is transformed into summed all-pass filters and then from there into a Hilbert transformer filter. The design technique is based largely on pole locations and creates a filter in the cascaded 1st order allpass form, which is numerically robust.
Convention Paper 8258

3:30 pm

P17-3 Automatic Parallelism from Dataflow Graphs —*Ramy Sadek*, University of Southern California, Playa Vista, CA, USA

This paper presents a novel algorithm to automate high-level parallelization from graph-based data structures representing data flow. Algorithm correctness is shown via a formal proof by construction. This automatic optimization yields large performance improvements for multi-core machines running host-based applications. Results of these advances are shown through their incorporation into the audio processing engine Application Rendering Immersive Audio (ARIA) presented at AES 117. Although the ARIA system is the target framework, the contributions presented in this paper are generic and therefore applicable in a variety of software such as Pure Data and Max/MSP, game audio engines, non-linear editors and related systems. Additionally, the parallel execution paths extracted are shown to give effectively optimal cache performance, yielding significant speedup for such host-based applications.
Convention Paper 8259

4:00 pm

P17-4 The Design of Low-Complexity Wavelet-Based Audio Filter Banks Suitable for Embedded Platforms—*Neil Smyth*, Cambridge Silicon Radio, Belfast, N. Ireland, UK

Many audio applications require the use of low complexity, low power, and low latency filter banks (e.g., real-time audio streaming to mobile devices). The underlying mathematics of wavelet transforms provides these attractive characteristics for embedded platforms. However, commonly used wavelets (Haar, Daubechies) possess coefficients containing irrational numbers that lead to distortion in fixed-point implementations. This paper discusses the development and provides practical performance comparisons of filter banks using wavelet transforms as an alternative to more commonly used sub-banding filter banks in PCM audio coding algorithms. The advantages and disadvantages of wavelets used in such audio compression applications are also discussed.
Convention Paper 8260

4:30 pm

P17-5 Application of Optimized Inverse Filtering to Improve Time Response and Phase Linearization in Multiway Loudspeaker Systems—*Mario Di Cola*,[1] Daniele Ponteggia[2]
[1]Audio Labs Systems, Casoli (CH), Italy
[2]Studio Ponteggia, Terni (TR), Italy

Digital processing has been widely demonstrated to be a very useful technique in improving loudspeaker systems' performances. Particularly interesting is Inverse Filtering applied to loudspeaker systems because it can improve performances and sound quality in terms of transient response and reduced overall phase shift. Inverse Filtering is a processing technique that can be realized with FIR filtering techniques with a specific sequence of taps that need to be synthesized "ad hoc" for a specific transducer and/or for a specific loudspeaker system configuration. Most of the studies on this matter so far, with very few exceptions, have been focused on the "DSP processing" point of view, being generally related to the involved mathematics and relative numerical problems. This paper represents a discussion on the philosophy that should drive the application of this technique to process a loudspeaker system in order to really improve it, and consequently it's been focused on the analysis of the loudspeaker system nature and the understanding of what can really be processed with a 1-dimensional "action." We will discuss what can be synthesized as a "2-port" model of the loudspeaker and then what can be effectively obtained by processing the input signal of a loudspeaker system.
Convention Paper 8261

5:00 pm

P17-6 Filter Design for a Double Dipole Flat Panel Loudspeaker System Using Time Domain

Toeplitz Equations—*Tobias Corbach, Martin Holters, Udo Zölzer*, Helmut-Schmidt-University/ University of the Federal Armed Forces, Hamburg, Germany

Today flat panel loudspeakers are used in multiple applications. Due to their high directivity and their good structural integration properties, flat panel loudspeakers are commonly used for directed acoustic information. A previously proposed system of 2 parallel flat panel dipole loudspeakers with adapted input filtering ensures a high suppression of the backward radiation and only minor influences to the forward radiation side. This paper presents a new approach to the filter computation for this application. It makes use of the time domain convolution, realized by Toeplitz matrices and builds the desired filter impulse responses by a least squares approach. The different filter computations as well as the numerical and measured results are shown.
Convention Paper 8262

5:30 pm

P17-7 A Low Complexity Approach for Loudness Compensation—*Pradeep D. Prasad*, Ittiam Systems Pvt. Ltd., Bangalore, Karnataka, India

The essence of loudness compensation is to maintain the perceived spectral balance of audio content irrespective of the playback volume level. The need for this compensation arises due to the inherent non-linearity in human aural perception manifesting as change in spectral balance. The compensation varies with critical band, original, and playback specific loudness. This results in a computationally intensive approach of estimating original and target specific loudness and calculating required compensation for every frame. A low complexity algorithm is proposed to enable resource constrained devices to efficiently perform loudness compensation. A closed form expression is derived for the proposed compensation followed by an analysis of the quality versus complexity tradeoff.
Convention Paper 8263

6:00 pm

P17-8 MPEG Spatial Audio Object Coding—The ISO/MPEG Standard for Efficient Coding of Interactive Audio Scenes—*Oliver Hellmuth*,[1] *Heiko Purnhagen*,[2] *Jeroen Koppens*,[3] *Jürgen Herre*,[1] *Jonas Engdegård*,[2] *Johannes Hilpert*,[1] *Lars Villemoes*,[2] *Leonid Terentiev,*[1] *Cornelia Falch,*[1] *Andreas Hölzer,*[1] *María Luis Valero*,[1] *Barbara Resch*,[2] *Harald Mundt*,[4] *Hyen-O Oh*[5]
[1]Fraunhofer Institute for Integrated Circuits IIS Erlangen, GE
[2]Dolby Sweden AB, Stockholm, Sweden
[3]Philips Applied Technologies, Eindhoven, The Netherlands
[4]Dolby Germany GmbH, Nürnberg, Germany
[5]LG Electronics, Seoul, Korea

In 2007, the ISO/MPEG Audio standardization group started a new work item on efficient coding of sound scenes comprising several audio objects by parametric coding techniques. Finalized in the summer of 2010, the resulting MPEG "Spatial Audio Object Coding" (SAOC) specification allows the representation of such scenes at bit rates commonly used for coding of mono or stereo sound. At the decoder side, each object can be interactively rendered, supporting applications like user-controlled music remixing and spatial teleconferencing. This paper summarizes the results of the standardization process, provides an overview of MPEG SAOC technology, and illustrates its performance by the results of the recent verification tests. The test includes operation modes for several typical application scenarios that take advantage of object-based processing.
Convention Paper 8264

Session P18 Saturday, Nov. 6
2:30 pm – 6:30 pm Room 236

BINAURAL AND TRANSAURAL AUDIO

Chair: **Durand R. Begault**, NASA Ames Research Center, Moffett Field, CA, USA

2:30 pm

P18-1 Modification of HRTF Filters to Reduce Timbral Effects in Binaural Synthesis, Part 2: Individual HRTFs—*Juha Merimaa*, Sennheiser Research Laboratory, Palo Alto, CA, USA

In the first part of this study [1], a method for designing modified head-related transfer function (HRTF) filters with reduced timbral effects was proposed. Spectral localization cues were effectively scaled down while preserving the interaural time and level differences. For non-individualized HRTFs, the modifications were found to produce no statistically significant changes in localization. This paper continues the investigation using individual HRTFs. It is shown that in this case the reduction in timbral effects comes at a slight listener-dependent cost in localization performance. The filter design thus enables trading off more neutral timbre against more accurate localization.
Convention Paper 8265

3:00 pm

P18-2 On the Improvement of Auditory Accuracy with Non-Individualized HRTF-Based Sounds —*Catarina Mendonça,*[1] *Jorge Santos*,[1] *Guilherme Campos,*[2] *Paulo Dias,*[2] *José Vieira*,[2] *João Ferreira*[1]
[1]University of Minho, Minho, Portugal
[2]University of Aveiro, Aveiro, Portugal

Auralization is a powerful tool to increase the realism and sense of immersion in Virtual Reality environments. The Head Related Transfer Function (HRTF) filters commonly used for auralization are non-individualized, as obtaining individualized HRTFs poses very serious practical difficulties. It is therefore extremely important to understand to what extent this hinders sound perception. In this paper we address this issue from a learning perspective. In a set of experiments, we observed that mere exposure to virtual sounds processed with generic HRTF did not improve the subjects' performance in sound source localization, but short training periods ➥

involving active learning and feedback led to significantly better results. We propose that using auralization with non-individualized HRTF should always be preceded by a learning period.
Convention Paper 8266

3:30 pm

P18-3 Processing and Improving Head-Related Impulse Response Database for Auralization
—*Ben Supper*, Focusrite Audio Engineering Ltd., High Wycombe, UK

To convert a database of anechoic head-related impulse responses [HRIRs] into a set of data that is suitable for auralization involves many stages of processing. The output data set must be precisely corrected to account for some circumstances of the recording. It must then be equalized to remove coloration. Finally, the database must be interpolated to a finer resolution. This paper explains these stages of correction, equalization, and spatial interpolation for a frequently-used data set obtained from a KEMAR dummy head. The result is a useful database of HRIRs that can be applied dynamically to audio signals for research and entertainment purposes.
Convention Paper 8267

4:00 pm

P18-4 Stimulus-Dependent HRTF Preference—
Agnieszka Roginska,[1] *Gregory H. Wakefield*,[2] *Thomas S. Santoro*[3]
[1]New York University, New York, NY, USA
[2]University of Michigan, Ann Arbor, MI, USA
[3]Naval Submarine Medical Research Lab, Groton, CT, USA

Measurement of individual Head Related Transfer Functions (HRTFs) can be inconvenient, expensive, and time consuming. User selected HRTFs can alleviate the complexity of individually measured HRTFs and make better quality 3-D audio available to more listeners. This paper presents the results of a study designed to investigate the use of user-selected HRTFs augmented with customized interaural cues. In the study presented subjects were asked to select HRTFs that resulted in an accurate percept based on three specific criteria: externalization quality, elevation, and front/back discrimination. Subjective tests were conducted using three different stimuli. Results of the experiment are presented.
Convention Paper 8268
Paper presented by Gregory Wakefield.

4:30 pm

P18-5 Comparison between Spherical Headmodels and HRTFs in Upmixing for Headphone-Based Virtual Surround and Stereo Expansion—Part I
—*Sunil Bharitkar, Chris Kyriakakis*, University of Southern California, Los Angeles, CA, USA, Audyssey Labs., Inc., Los Angeles, CA, USA

In this paper, a first of multiple-parts, we compare the performance of headmodels with head-related transfer functions (HRTFs), which have previously been published, using different upmixing techniques for headphone virtual surround. We consider a spherical head, with and without the pinna or the torso model, whereas for the HRTFs we incorporate the CIPIC, Nagoya, and MIT HRTF sets in the up-mixing. The up-mixing technique includes using the Moorer reverberator, a modified Moorer reverberator, and modeling the direct sound, the first several discrete reflections (with adjustable delay and amplitude), and the diffuse field reflections with a tunable frequency dependent decorrelator. Furthermore, since the measured HRTFs can introduce audible coloration we investigate if there is a trade-off between localization and timbre by incorporating complex-domain smoothing of the HRTF time responses. To evaluate the localization and timbre performance between the models we use movie and music content (viz., stereo, ITU downmix, and a commercial downmix method) as well as Gaussian tone noise bursts of critical bandwidth.
Convention Paper 8269

5:00 pm

P18-6 HRTF Measurements with Recorded Reference Signal
—*Marko Durkovic, Florian Sagstetter, Klaus Diepold*, Technische Universität München, Munich, Germany

Head-Related Transfer Functions (HRTFs) are used for adding spatial information in 3-D audio synthesis or for binaural robotic sound localization. Both tasks work best when using a custom HRTF database that fits the physiology of each person or robot. Usually, measuring HRTFs is a time consuming and complex procedure that is performed with expensive equipment in an anechoic chamber. In this paper we present a method that enables HRTF measurement in everyday environments by passively recording the surroundings without the need to actively emit special excitation signals. Experiments show that our method captures the HRTF's spatial cues and enables accurate sound localization.
Convention Paper 8270

5:30 pm

P18-7 Angular Resolution Requirements for Binaural Room Scanning
—*Todd Welti*,[1] *Xinting Zhang*[2]
[1]Harman International, Northridge, CA, USA
[2]State University of New York at Binghamton, Binghamton, NY, USA

Binaural Room Scanning is a method of capturing and reproducing a binaural representation of a room or car, using a dummy head incorporating binaural microphones, and individual measurements made with the dummy head positioned at a number of different head angles. The measurement process can be time-consuming. It is therefore important to know how high the angular resolution needs to be. An experiment was performed to see if the angular resolution could be reduced from the current 1 degree resolution to 15 degree resolution, without causing an audible difference. Using a 3 alternative forced choice method, trained listeners compared 1 degree and 15 degree angular resolution and could not reliably detect the difference.
Convention Paper 8271

6:00 pm

P18-8 Binaural Reproduction of 22.2 Multichannel Sound over Loudspeakers—*Kentaro Matsui, Akio Ando*, NHK Science and Technology Research Laboratories, Tokyo, Japan

NHK has proposed the 22.2 multichannel sound system, which consists of 22 loudspeakers and 2 for LFE producing three-dimensional spatial sound, to be the format for future TV broadcasting. To allow it to be reproduced in homes, we have investigated various reproduction methods that use fewer loudspeakers. We introduce a design of binaural rendering of the 22.2 multichannel sound with three frontal loudspeakers as a minimum configuration model for homes. It can stably process the system inverse filters by dividing them into all-pass and minimum-phase components and successfully compensate the sound quality with a peak suppression method.
Convention Paper 8272

Session P19 Saturday, Nov. 6
2:30 pm – 4:00 pm Room 226

POSTERS: SPATIAL SOUND PROCESSING—PART 1

2:30 pm

P19-1 Estimation of the Probability Density Function of the Interaural Level Differences for Binaural Speech Separation—*David Ayllon, Roberto Gil-Pita, Manuel Rosa-Zurera*, University of Alcalá, Alcalá de Henares (Madrid), Spain

Source separation techniques are applied to audio signals to separate several sources from one mixture. One important challenge of speech processing is noise suppression and several methods have been proposed. However, in some applications like hearing aids, we are not interested just in removing noise from speech but amplifying speech and attenuating noise. A novel method based on the estimation of the Power Density Function of the Interaural Level Differences in conjunction with time-frequency decomposition and binary masking is applied to speech-noise mixtures in order to obtain both signals separately. Results show how both signals are clearly separated and the method entails low computational cost so it could be implemented in a real-time environment, such as a hearing aid device.
Convention Paper 8273

2:30 pm

P19-2 The Learning Effect of HRTF-Based 3-D Sound Perception with a Horizontally Arranged 8-Loudspeaker System—*Akira Saji, Keita Tanno, Li Huakang, Tetsuya Watanabe, Jie Huang*, The University of Aizu, Aizuwakamatsu City, Fukushima, Japan

This paper argues about the learning effects on the localization of HRTF-based 3-D sound using an 8-channel loudspeaker system, which creates virtual sound images. This system can realize sound with elevation by 8 channel loudspeakers arranged on the horizontal plane and convolving HRTF, not using high or low mounted loudspeakers. The position of the sound image that the system creates is difficult to perceive because such HRTF-based sounds are unfamiliar. However, after repetition of the learning process, almost all listeners can perceive the position of the sound images better. This paper shows this learning effect for an HRTF-based 3-D sound system.
Convention Paper 8274

2:30 pm

P19-3 Spatial Audio Attention Model Based Surveillance Event Detection—*Bo Hang, Ruimin Hu, Xiaochen Wang, Weiping Tu*, Wuhan University, Wuhan, China

In this paper we propose a bottom-up audio attention model based on spatial audio cues and subband energy change for unsupervised event detection in stereo audio surveillance. First, the spatial audio parameter Interaural Level Difference (ILD) is extracted to calculate and represent the attention events, which are caused by rapid moving sound source. Then the subband energy change is computed to present the salient energy distribution change in frequency domain. At last, an environment adaptive normalization is used to assess the normalized attention level. Experimental results demonstrate that the proposed audio attention model is effective for audio surveillance event detection.
Convention Paper 8275

2:30 pm

P19-4 Investigating Perceptual Effects Associated with Vertically Extended Sound Fields Using Virtual Ceiling Speaker—*Yusuke Ono, Sungyoung Kim, Masahiro Ikeda*, Yamaha Corporation, Iwata, Shizuoka, Japan

Virtual Ceiling Speaker (VCS) is a signal processing method that creates an elevated auditory image using an optimized cross-talk compensation for a 5-channel reproduction system. In order to understand latent perceptual effects caused by virtually elevated sound imageries, we experimentally compared the perceptual differences between physically and virtually elevated sound sources in terms of ASW, LEV, Powerfulness, and Clarity. The results showed that listeners perceived higher LEV or Clarity by adding physically or virtually elevated early reflections than 5-channel content in either case. It might implicate that attributes related to spatial dimensions were relatively well expressed due to virtually elevated signals using VCS.
Convention Paper 8276

2:30 pm

P19-5 Enhancing 3-D Audio Using Blind Bandwidth Extension—*Tim Habigt, Marko Durkovic, Martin Rothbucher, Klaus Diepold*, Technische Universität München, Minich, Germany

Blind bandwidth extension techniques are used to recreate high frequency bands of a narrowband audio signal. These methods allow ➥

increasing the perceived quality of signals that are transmitted via a narrow frequency band as in telephone or radio communication systems. We evaluate the possibility to use blind bandwidth extension methods in 3-D audio applications, where high frequency components are necessary to create an impression of elevated sound sources.
Convention Paper 8277

Workshop 14 Saturday, November 6
2:30 pm – 3:30 pm Room 130

RETHINKING THE DIGITAL AUDIO WORKSTATION

Chair: **Michael Hlatky**, Accessive Tools GmbH

Panelists: *Jörn Loviscach*, University of Applied Sciences Bielefeld
Guy McNally, Uncut Video Inc.
Bernard Mont-Reynard, SoundHound Inc.
Allan Seago, London Metropolitan University

The DAWs of today very much resemble those of 1989. Yes, the buttons have become nicer and we can record more tracks in parallel, but with the technology advances since then, we should be doing much better. There are, however, not many companies on the market today that have the ability to rethink how their products work. Yet, universities and research centers have brought us an immense collection of new technologies to build new products upon. How about, for instance, real-time online cross-DAW collaboration, leveraging social networks for finding the optimal effect settings, and making DAWs not only bulletproof, but also foolproof? This workshop surveys existing technologies; it looks into possible synergies from other fields of computing sciences, and proposes practical improvements for and/or radical changes to DAW software.

Broadcast/Media Streaming Session 12
Saturday, November 6 2:30 pm – 4:00 pm
Room 133

AUDIO FOR NEWSGATHERING

Chair: **Skip Pizzi**, Media Technology Consultant & Technology Editor, *Radio Ink* magazine

Panelists: *Daniel Mansergh*, Director of Engineering, KQED-FM, San Francisco
Jeff Towne, Co-producer/Engineer, "Echoes" and Tools Editor, Transom.org

The world of broadcast news presents numerous challenges to the recording, production, and presentation of the rich information carried by sound. Most challenging is the proper recording of raw sound for news events as they happen in the field. From faraway battlefields to neighborhood playgrounds, the tools and processes required to capture this sound appropriately and reliably for broadcast are a highly specialized tributary of audio technology. Practicing experts in the field will share tips and techniques honed over years of experience on the beat.

Game Audio Session 11 Saturday, November 6
2:30 pm – 4:30 pm Room 120

CAREERS IN GAME AUDIO

Chair: **Steve Horowitz**, Composer/Musician, The Code International Inc.

Panelists: *Tim Duncan*, DMA; Associate Professor; Director, Digital Audio Technology, Cogswell Polytechnical College
Shiloh Hobel, Sr. Director, Industry and Career Services - Sound Arts, Ex'pression College for Digital Arts
David Javelosa, Professor of Game Development, Santa Monica College
Lennie Moore, Composer
Michael Sweet, Associate Professor, Berklee College of Music

You already know that you want a career in game audio. You even know which position is best for you. The big question is how do you get the training and experience necessary to land that first job? This workshop will present the latest work of the IASIG (Interactive Audio Special Interest Group) and local education institutions to develop standardized school curriculum for games and interactive audio. Programs are springing up all over the world, and this panel will provide the big overview of what training is available now and what is coming in the future. From single overview classes, associate degree programs to a full four year university study we will preview the skill sets and templates that are desired and the recommended path for getting that position on the audio team.

Live Sound Seminar 8 Saturday, November 6
2:30 pm – 4:30 pm Room 131

FILL SPEAKERS IN LIVE SOUND REINFORCEMENT SYSTEMS

Chair: **Tom Young**, Electroacoustic Design Services

Panelists: *Jamie Anderson*, Rational Acoustics
Ales Dravinec, ADRaudio
Ted Leamy, Pro Media/Ultra Sound
Dave Rat, Rat Sound

Sometimes required due to a last minute whim or poor planning and at other times budgeted, selected, configured, and optimized with as much care as the primary loudspeakers, fill speakers are a commonplace fixture in our work in live sound. Aside from the obvious need, fill speakers are also employed very effectively for imaging/localization in some types of productions. Additional care/coordination must sometimes be provided in some applications to reduce the visibility of these (as well as the primary) loudspeakers and without unintended compromises in the electroacoustic performance of the system.

Product Design Session 5 Saturday, November 6
2:30 pm – 4:30 pm Room 132

GROUNDING AND SHIELDING—CIRCUITS AND INTERFERENCE—PART 2

Presenter: **Ralph Morrison**

This second session of Grounding and Shielding Circuits takes off where the first session ended. Under discussion will be: the role of digital logic and processors in the audio world; A/D converters; transmission line basics; impedance control and impedance matching; the relation between rise and fall times and frequency spectrum; the need for ground and power planes; decoupling and filtering on logic structures; the interface between analog and digital circuits; digital analog filters; aliasing errors; balanced digital lines and common-mode rejection; multilayer boards; interference problems such as cross talk, ground bounce, via locations.

Special Event
GRAMMY SOUNDTABLE
Saturday, November 6, 2:30 pm – 4:00 pm
Room 134

Moderator: **Sylvia Massy**

Panelists: *Joe Barresi*
Bob Clearmountain
DJ Khalil
Jimmy Douglass
Nathaniel Kunkel

Sonic Imprints: Songs That Changed My Life

Some songs are hits, some we just love, and some have changed our lives. Our panelists break down the DNA of their favorite tracks and explain what moved them, what grabbed them, and why these songs left a life long impression.

Sylvia Massy broke into the big time with 1993's Undertow by Los Angeles rock band Tool, and went on to engineer recordings for a diverse group of artists including Aerosmith, Babyface, Big Daddy Kane, The Black Crowes, Bobby Brown, Prince, Julio Iglesias, Seal, Skunk Anansie, and Paula Abdul, among many others. Massy also engineered and mixed projects with producer Rick Rubin, including Johnny Cash's *Unchained* album, which won a Grammy award for Best Country Album. She continues to work with artists such as Sublime, the Cliks, and Showbread, most often at Radio Star Studios which she owns and operates in Weed, California. She also writes the "Gear Stories" column for *Mix* magazine.

HISTORICAL COMMITTEE MEETING
Saturday, November 6, 3:00 pm – 4:00 pm
Room 114

Saturday, November 6 3:00 pm Room 113
Technical Committee Meeting on Acoustics and Sound Reinforcement

Saturday, November 6 3:00 pm Room 232
Standards Committee Meeting SC-03-06, SC-03-07 Digital Library and Archive Systems and Audio Metadata (combined)

Exhibitor Seminar
RENKUS-HEINZ, INC.
Saturday, November 6 3:30 pm – 4:30 pm
Room 112

Presenter: **Stefan Feistel**

Latest Developments in AFMG Measurement Tools EASERA and SysTune

This seminar presents the latest developments and trends in the standard-setting AFMG measurement software packages SysTune and EASERA. It focuses on using mobile devices with SysTune, the renowned live sound measurement software, as well as on new windowing options and loudspeaker alignment techniques. Also, new and expanded measurement options of EASERA v1.2 are discussed such as the determination of absorption, reflection and scattering coefficients.

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Saturday, November 6 3:30 pm – 4:30 pm
Room 122

Presenters: **Josh Tidsbury**
Michael Nunan

An Olympic Score

Josh Tidsbury, Systems Specialist and Post Sound Mixer, and Michael Nunan, Post Sound Supervisor and co-producer of the music package of the 2010 Winter Olympic Games for CTV Television in Toronto, Ontario are presenting the CTV *2010 Olympic Suite*. Between them, Michael and Josh lived every moment of the Games and are pleased to have the opportunity to share their experiences. Josh will also present some of his latest projects, which include the famous Canadian "Hockey Theme" that he recorded and mixed (again in collaboration with Michael Numan), combined with some examples of movies like *Stake Land*, which won the Winner of Midnight Madness awards at the recent TIFF 2010.

Workshop 15 Saturday, November 6
3:45 pm – 4:45 pm Room 130

SEMANTIC AUDIO SUCCESS: COMMERCIAL APPLICATIONS OF SEMANTIC AUDIO ANALYSIS

Chair: **Jay LeBoeuf**, Imagine Research

Panelists: *Ching-Wei Chen*, Gracenote, Inc. Emeryville, CA, USA
Aaron Master, SoundHound Inc.
Erling Wold, Audible Magic

Thanks to mobile devices, cloud-computing, and ample storage and computing, we are now seeing a great dawn of products that include semantic audio analysis (SAA) technologies. Without even knowing it, millions of users reap the benefits of semantic audio analysis on a daily basis. Companies include SAA techniques to provide their users with magical experiences: offering intelligent, simple, work flows that listen to and richly interact with a user's speech, music, or environment. This workshop will introduce AES members to some of the commercial uses and applications of semantic audio analysis. A panel discussion will engage companies to describe how SAA technology has powers their innovative products, the technical challenges that they face, and the future of things to come.

Saturday, November 6 4:00 pm Room 113
Technical Committee Meeting on Network Audio Systems

Session P20 Saturday, Nov. 6
4:30 pm – 6:00 pm Room 226

POSTERS: SPATIAL SOUND PROCESSING—PART 2

4:30 pm

P20-1 Inherent Doppler Properties of Spatial Audio—*Martin J. Morrell, Joshua D. Reiss*, Queen Mary University of London, London, UK

The Doppler shift is a naturally occurring phenomenon that shifts the pitch of sound if the emitting object's distance to the listener is not a ➥

constant. These pitch deviations, alongside amplitude change help humans to localize a source's position, velocity, and movement direction. In this paper we investigate spatial audio reproduction methods to determine if Doppler shift is present for a moving sound source. We expand spatialization techniques to include time-variance in order to produce the Doppler shift. Recordings of several different loudspeaker layouts demonstrate the presence of Doppler with and without time-variance, comparing this to the pre-calculated theoretical values.
Convention Paper 8278

4:30 pm

P20-2 A Binaural Model with Head Motion that Resolves Front-Back Confusions for Analysis of Room Impulse Responses—
John T. Strong, Jonas Braasch, Ning Xiang,
Rensselaer Polytechnic Institute, Troy, NY, USA

Front-back confusions occur in both psychoacoustic localization tests and interaural cross-correlation-based binaural models. Head motion has been hypothesized and tested successfully as a method of resolving such confusions. This ICC-based model set forth here simulates head motion by filtering test signals with a trajectory of HRTFs and shifting an azimuth remapping function to follow the same trajectory. By averaging estimated azimuths over time, the correct source location prevails while the front-back reversed location washes out. This model algorithm is then extended to room impulse response analysis. The processing is performed on simulated binaural impulse responses at the same position but different head angles. The averaging allows the model to discriminate reflections coming from the front from those arriving from the rear.
Convention Paper 8279

4:30 pm

P20-3 A Set of Microphone Array Beamformers Implementing a Constant-Amplitude Panning Law—*Yoomi Hur*,[1,2] *Jonathan S. Abel*,[1]
Young-cheol Park,[3] *Dae Hee Youn*[2]
[1]Stanford University, Stanford, CA, USA
[2]Yonsei University, Seoul, Korea
[3]Yonsei University, Wonju, Korea

This paper describes a technique for designing a collection of beamformers, a "beamformerbank," that approximately produces a constant-amplitude panning law. Useful in multichannel recording scenarios, a point source will appear with energy above a specified sidelobe level in at most two adjacent beams, and the beam sum will approximate the source signal. A non-parametric design method is described in which a specified sidelobe level determines beam width as a function of arrival direction and frequency, leading directly to the number and placement of beams at every frequency. Simulation results using several microphone array configurations are reported to verify the performance of proposed technique.
Convention Paper 8280

4:30 pm

P20-4 A 3-D Sound Creation System Using Horizontally Arranged Loudspeakers—
Keita Tanno, Akira Saji, Huakang Li, Jie Huang,
The University of Aizu, Fukushima, Japan

In this research we have studied a 3-D sound creation system using 5- and 8-channel loudspeaker arrangements. This system has a great advantage in that it does not require the users to purchase a new audio system or to reallocate loudspeakers. The only change for creators of television stations, video game makers, and so on is to install the new proposed method for creation of the 3-D sound sources. Head-related transfer functions are used to create the signals of left and right loudspeaker groups. An extended amplitude panning method is proposed to decide the amplitude ratios between and within loudspeaker groups. Listening experiments show that the subjects could perceive the elevation of sound images created by the system as well.
Convention Paper 8281

4:30 pm

P20-5 Locating Sounds Around the Screen—
David Black,[1] *Jörn Loviscach*[2]
[1]Hochschule Bremen, University of Applied Sciences, Bremen, Germany
[2]Fachhochschule Bielefeld, University of Applied Sciences, Bielefeld, Germany

Today's large-size computer screens can display a wealth of information easily enough to overload the user's visual perceptual channel. Looking for a remedy for this effect, we researched into providing additional acoustic cues through surround sound loudspeakers mounted around the screen. In this paper we demonstrate the results of user evaluations of interaction with screen elements using the surround-screen setup. Results of these tests have shown that applying surround-screen sound can enhance response times in a simple task, and that users can localize the approximate origin of a sound when played back with this technique.
Convention Paper 8282
Paper presented by Jörn Loviscach

Broadcast/Media Streaming Session 13
Saturday, November 6 4:30 pm – 6:00 pm
Room 133

STREAM FORMATS FOR CONTENT DELIVERY NETWORKS

Chair: **Ray Archie**, CBS

Panelists: *Benny Fischer*, Limelight
Andy Jones, Stream Guys
Andrew Snook, StreamOn
Sam Sousa, Stream the World

The streaming formats for CDN's panel is about the relationship between distribution and encoding methodologies. Licensing, error-correction, quality vs compression, and consumer-adoption are just a few variables to be discussed by this all-star panel. We hope to shed light about the future of scalable and reliable digital distribution.

Game Audio Session 12 **Saturday, November 6**
4:30 pm – 6:00 pm **Room 206**

MIXING THE DICE WAY—BATTLEFIELD, HDR AUDIO, AND INSTANTIATED MIXING

Presenter: **David Mollerstadt**, EA/DICE

In this session David Mollerstedt presents the detailed concepts behind how High Dynamic Range Audio does adaptive runtime level balancing. It further explains the Instantiated Mixer System in the Frostbite engine that allows for elaborate manipulation of individual sounds and asset groups.

Live Sound Seminar 9 **Saturday, November 6**
4:30 pm – 6:30 pm **Room 134**

SAN FRANCISCO LIVE SOUND IN THE '70S

Chair: **Lee Brenkman**

Panelists: *Radley Hirsch*
Gil Mazzi, Third Ear Sound
John Meyer
Harry Popick

What was it like to run sound gigs in the 1970s, with home-grown loudspeakers, amplifiers pushed to the breaking point, and escalating SPL expectations? Veterans from the San Francisco Bay Area relate their victories and war stories.

Saturday, November 6 4:30 pm Room 232
Standards Committee Meeting SC-04-01 Acoustics and Sound Source Modeling

Workshop 16 **Saturday, November 6**
4:45 pm – 6:15 pm **Room 131**

MASTERING: ART, PERCEPTION, AND TECHNOLOGIES—PART 2

Chair: **Michael Romanowski**, Michael Romanowski Mastering

Panelists: *Gavin Lurssen*, Gavin Lurssen Mastering
Andrew Mendleson, Georgetown Masters
Joe Palmacio, The Place for Mastering
Paul Stubblebine, Paul Stubblebine Mastering
Mike Wells, Mike Wells Mastering

This is a continuation of the Mastering panel from AES 2009 in New York. We will discuss the state of Mastering in 2010 by Mastering Engineers. Mastering engineers use technology to achieve the desired results. But what gets little or no discussion is the perceptions and approaches that cause the engineer to make those choices. In this two part series, we want to talk about the art of perception and technology as it pertains to the Mastering industry in 2010, and the future. In this session, we will talk about perception and the art form of mastering, and how decisions are made based on our approaches and perception in the mastering environment.

Product Design Session 6 **Saturday, November 6**
4:45 pm – 6:15 pm **Room 132**

DESIGNING MICROPHONE PREAMPLIFIERS

Presenter: **Gary Hebert**

Microphone preamplifiers are a fundamental building block in professional audio systems, with a legacy reaching back to the beginning of audio times. Indeed, the mic preamp often defines the signature sound of a piece of equipment or even a recording studio. Its performance must exceed nearly everything else in the system, and as the first line of defense to the outside world it must withstand hostile conditions such as 48V phantom power faults or erroneously patched hot signals.

Today's mic preamp developers face many challenges as new products demand lower cost and power, components that have been depended on for years are discontinued, demands for quality either increase or (worse) are forgotten, and so on. Fortunately, today's developers have more options than ever before to design cost-effective, high-performance, small, green power, sweet sounding mic preamps. This tutorial presents a variety of circuit designs that trade off between cost, size, power consumption, noise, THD, CMR, and other factors.

Tutorial 11 **Saturday, November 6**
5:00 pm – 6:15 pm **Room 130**

THE iPOD GENERATION—THE AUDIO ARTIFACTS THE CONSUMER IS LISTENING TO

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

After a brief introduction to the psychoacoustic principles exploited by data compression (critical bands, minimum audition thresholds, simultaneous masking, and temporal masking), and a brief introduction to the capabilities and limitations imposed by the various components of encoders (frequency lines, windowing, temporal accuracy, compression considerations—CBR versus VBR, bit rates, and stereo modes), common artifacts produced by various encoders will be presented as audio, RTA, or waveform graph examples (stereo image changes, loss of bandwidth, pre and post echoes, double speak, ringing, bass fuzziness, flattening of dynamics, phase shift, "swirlies," frequency content and noise addition, musical content removed and noise added, and the results of more and less adaptive encodings). The presentation will focus on the artifacts created by a variety of current and common music download and dissemination formats. To conclude, consumer awareness (or lack thereof) of these artifacts is discussed, along with signs that "better" is on the way.

Broadcast/Media Streaming Session 14
Saturday, November 6 **5:30 pm – 6:30 pm**
Room 120

CAREERS IN BROADCASTING

Chair: **Chriss Scherer**, CPBE CBNT; Editor, Radio magazine; Past President, Society of Broadcast Engineers

Panelists: *William Blum*, Station Engineer, KBLX-FM
Russell Brown, Chief Engineer, KMTP-TV
Steve Lampen, Multimedia Technology Manager and Product Line Manager, Belden
Kimberly Sacks, Contract Engineer

As technology has evolved, pro audio and broadcasting seem to have diverged. But the skills you use in a pro ➥

audio career are likely applicable to a career in broadcasting, too. The moderator and panelists each have experience in pro audio and broadcasting, and they will share their career insights to show that the two industries have a great deal in common. A Q&A will also be held to clarify the bridge between the two industries. Students and professionals are encouraged to attend.

Special Event
ORGAN RECITAL
Saturday, November 6, 8:00 pm – 9:00 pm
The Cathedral of St. Mary of the Assumption
1111 Gough St., San Francisco, CA

Organist Graham Blyth's recitals are a highlight of every AES convention at which he performs. This year's recital will be held at St. Mary's Cathedral, a modern structure with a panoramic view of San Francisco. The cathedral's Ruffatti organ was designed with the Baroque repertoire in mind, a fact Blyth will reflect with a strong Bach emphasis in the program. He will also play the Fantasia & Fugue on B.A.C.H by Liszt. The second half will feature works by well and not so well known composers who held the position of Organiste Titulaire at some of the famous churches in Paris.

Located just minutes from the Golden Gate Bridge, Downtown Financial District, Twin Peaks, and The Marina, St. Mary's Cathedral is in the heart of the city at the top of Cathedral Hill. The Ruffatti Organ, built in 1971 by Fratelli Ruffatti of Padua, Italy, has been acclaimed as one of the finest in the world. It rises impressively from its soaring pedestal platform into a magnificent art form in its own right. It consists of 4842 pipes on 89 ranks and 69 stops.

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.

Session P21 **Sunday, Nov. 7**
9:00 am – 12:30 pm **Room 220**

LOW BIT-RATE AUDIO CODING

Chair: **Marina Bosi**, Stanford University, Stanford, CA, USA

9:00 am

P21-1 Combination of Different Perceptual Models with Different Audio Transform Coding Schemes—Implementation and Evaluation—
Armin Taghipour, Nicole Knölke, Bernd Edler, Jörn Ostermann, Leibniz Universität Hannover, Germany

In this paper four combinations of perceptual models and transform coding systems are implemented and compared. The first of the two perceptual models is based on a DFT with a uniform frequency resolution. The second model uses IIR filters designed in accordance with the temporal/spectral resolution of the auditory system. Both of the two transform coding systems use a uniform spectral decomposition (MDCT). While in the first system the quantizers are directly controlled by the perceptual model, the second system uses a pre- and post-filter with frequency warping for shaping the quantization noise with a temporal/spectral resolution more adapted to the auditory system. Implementation details are given and results of subjective tests are presented.
Convention Paper 8283
Paper presented by Bernd Edler

9:30 am

P21-2 Using Noise Substitution for Backwards-Compatible Audio Codec Improvement—
Colin Raffel, Experimentalists Anonymous, Stanford, CA, USA

A method for representing error in perceptual audio coding as filtered noise is presented. Various techniques are compared for analyzing and re-synthesizing the noise representation. A focus is placed on improving the perceived audio quality with minimal data overhead. In particular, it is demonstrated that per-critical-band energy levels are sufficient to provide an increase in quality. Methods for including the coded error data in an audio file in a backwards-compatible manner are also discussed. The MP3 codec is treated as a case study, and an implementation of this method is presented.
Convention Paper 8284

10:00 am

P21-3 An Introduction to AVS Lossless Audio Coding—*Haiyan Shu, Haibin Huang, Ti-Eu Chan, Rongshan Yu, Susanto Rahardja*, Institute for Infocomm Research, Agency for Science, Technology & Research, Singapore

Recently, the audio video coding standard workgroup of China (AVS) issued a call for proposal for audio lossless coding. Several proposals were received, in which the proposal from the Institute for Infocomm Research was selected as Reference Model (RM). The RM is based on time-domain linear prediction and residual entropy coding. It introduces a novel residual pre-processing method for random access data frames and a memory-efficient arithmetic coder with dynamic symbol probability generation. The performance of RM is found to be comparable to those of MPEG-4 ALS and SLS. The AVS lossless coding is expected to be finalized at the end

of 2010. It will become the latest extension of the AVS-P3 audio coding standard.
Convention Paper 8285

10:30 am

P21-4 Audio Re-Synthesis Based on Waveform Lookup Tables—*Sebastian Heise,*[1] *Michael Hlatky,*[1] *Jörn Loviscach*[2]
[1]Accessive Tools GmbH, Bremen, Germany
[2]Hochschule Bielefeld University of Applied Sciences, Bielefeld, Germany

Transmitting speech signals at optimum quality over a weak narrowband network requires audio codecs that must not only be robust to packet loss and operate at low latency, but also offer a very low bit rate and maintain the original sound of the coded signal. Advanced speech codecs for real-time communication based on code-excited linear prediction provide bandwidths as low as 2 kbit/s. We propose a new coding approach that promises even lower bit rates through a synthesis approach not based on the source-filter model, but merely on a lookup table of audio waveform snippets and their corresponding Mel-Frequency Cepstral Coefficients (MFCC). The encoder performs a nearest-neighbor search for the MFCC features of each incoming audio frame against the lookup table. This process is heavily sped up by building a multi-dimensional search tree of the MFCC-features. In a speech coding application, for each audio frame, only the index of the nearest neighbor in the lookup table would need to be transmitted. The encoder synthesizes the audio signal from the waveform snippets corresponding to the transmitted indices.
Convention Paper 8286

11:00 am

P21-5 A Low Bit Rate Mobile Audio High Frequency Reconstruction—*Bo Hang, Ruimin Hu, Yuhong Yang, Ge Gao*, Wuhan University, Wuhan, China

In present communication systems, high quality audio signals are supposed to be provided with low bit rate and low computational complexity. To increase the high frequency band quality in current communication system, this paper proposed a novel audio coding high frequency bandwidth extension method, which can improve decoded audio quality with increasing only a few coding bits per frame and a little computational complexity. This method calculates high-frequency synthesis filter parameters by using a codebook mapping method, and transmits quantified gain corrections in high-frequency parts of multiplexing coding bit streams. The test result shows that this method can provide comparable audio quality with lower bit consumption and computational complexity compared to the high frequency regeneration of AVS-P10.
Convention Paper 8287

11:30 am

P21-6 Perceptual Distortion-Rate Optimization of Long Term Prediction in MPEG AAC— *Tejaswi Nanjundaswamy,*[1] *Vinay Melkote,*[1] *Emmanuel Ravelli,*[2] *Kenneth Rose*[1]
[1]University of California Santa Barbara, CA, USA
[2]Fraunhofer IIS, Erlangen, Germany

Long Term Prediction (LTP) in MPEG Advanced Audio Coding (AAC) exploits inter-frame redundancies via predictive coding of the current frame, given previously reconstructed data. Particularly, AAC Low Delay mandates LTP, to exploit correlations that would otherwise be ignored due to the shorter frame size. The LTP parameters are typically selected by time-domain techniques aimed at minimizing the mean squared prediction error, which is mismatched with the ultimate perceptual criteria of audio coding. We thus propose a novel trellis-based approach that optimizes the LTP parameters, in conjunction with the quantization and coding parameters of the frame, explicitly in terms of the perceptual distortion and rate tradeoffs. A low complexity "two-loop" search alternative to the trellis is also proposed. Objective and subjective results provide evidence for substantial gains.
Convention Paper 8288

12:00 noon

P21-7 Stereo Audio Coding Improved by Phase Parameters—*Miyoung Kim, Eunmi Oh, Hwan Shim*, Samsung Electronics, Gyeonggi-do, Korea

The parametric stereo coding exploiting phase parameters in a bit-efficient way is a part of MPEG-D USAC (Unified Speech and Audio Coding) standard. This paper describes the down-mixing and up-mixing scheme to further enhance the stereo coding in strong out-of-phase or near out-of-phase signals. The conventional downmixing as a sum of left and right channel for parametric stereo coding has the potential problems, phase cancellation in out-of-phase signals, which results in audible artifacts. This paper proposes the phase alignment by estimated overall phase difference (OPD) parameter and inter-channel phase difference (IPD) parameter. Furthermore, this paper describes the phase modification to minimize the phase discontinuity of down-mixed signal by scaling the size of the stereo channels.
Convention Paper 8289

Session P22 **Sunday, Nov. 7**
9:00 am – 10:00 am **Room 236**

ENHANCEMENT OF AUDIO REPRODUCTION

Chair: **Richard Foss**, Rhodes University, Grahamstown, South Africa

9:00 am

P22-1 Enhancing Stereo Audio with Remix Capability— *Hyen-O Oh,*[1,2] *Yang-Won Jung,*[1] *Alexis Favrot,*[3] *Christof Faller*[3,4]
[1]LG Electronics Inc., Seoul, Korea
[2]Yonsei University, Seoul, Korea
[3]Illusonic LLC, Lausanne, Switzerland
[4]EPFL, Lausanne, Switzerland

Many audio appliances feature capabilities for modifying audio signals, such as equalization, acoustic room effects, etc. However, these modification capabilities are always limited in the ➥

sense that they apply to the audio signal as a whole and not to a specific “audio object.” We are proposing a scheme that enables modification of stereo panning and gain of specific objects inherent in a stereo signal. This capability is enabled (possibly stereo backwards compatibly) by adding a few kilobits of side information to the stereo signal. For generating the side information, the signals of the objects to be modified in the stereo signal are needed.
Convention Paper 8290

9:30 am

P22-2 Automatically Optimizing Situation Awareness and Sound Quality for an Isolating Earphone—*John Usher*, Hearium Labs., San Francisco, CA, USA

Sound isolating (SI) earphones are increasingly used by the general public with portable media players in noisy urban and transport environments. The dangers of these SI earphones are becoming increasingly apparent, and an urgent review of their usage is being recommended by legislators. The problem is that the user is removed from their local ambient scene: a reduction in their “situation awareness” that often leads to accidents involving unheard oncoming vehicles. This paper introduces a new automatic gain control system to automatically mix the ambient sound field with reproduced audio material. A discussion of the audio system architecture is given and an analysis of 20 different warning sounds is used to suggest suitable parameters.
Convention Paper 8291

Tutorial 12 **Sunday, November 7**
9:00 am – 10:45 am **Room 130**

LOUDSPEAKERS AND HEADPHONES—DIAGNOSTICS OF SOUND RADIATION

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

Distributed mechanical parameters describe the vibration and geometry of the sound radiating surface of loudspeaker drive units. This data is the basis for predicting the sound pressure output and a decomposition of the total vibration into modal and sound pressure related components. This analysis separates acoustical from mechanical problems, shows the relationship to the geometry and material properties, and gives indications for practical improvement. The tutorial combines the theoretical background with practical loudspeaker diagnostics illustrated on various kinds of transducers such as woofer, tweeter, compression driver, microspeaker, and headphones.

Workshop 17 **Sunday, November 7**
9:00 am – 10:30 am **Room 132**

PATHS TO HIGH RESOLUTION DOWNLOADS

Chair: **Mike Wells**, Mike Wells Mastering

Panelists: *Virgilio Bacigalupo*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
Tony Berman, Berman Entertainment and Technology Law, San Francisco, CA, USA
Jean Cook, Future Music Coalition
Jeff Price, Tunecore, Brooklyn, NY, USA
John Spencer, BMS/Chase, Nashville, TN, USA

This panel will explore the current state of digital distribution of commercial audio recordings, and how the audio engineering community can get involved with digital distribution channels to create a new direction toward the goal of high resolution downloads. Beginning with an overview of how modern digital distribution channels have come to the current standards of low-resolution MP3 and AAC codecs as standards, we then explore current efforts toward increasing fidelity and quality of distributed encodings within the digital distribution channels, how we can work with digital distributors to improve interaction of audio engineers with the digital distribution community, and finish with action steps that both parties (audio engineering and digital distribution) can take toward improving fidelity and quality of distributed audio assets.

Game Audio Session 13 **Sunday, November 7**
9:00 am – 11:00 am **Room 120**

TAKIN' CARE OF BUSINESS

Chair: **Scott Selfon**, Lead Program Manager, Microsoft Corporation

Panelists: *Rod Abernethy*, Red Note Audio
Simon Amarasingham, CEO, dSonic Inc.
Alistair Hirst, CEO, OMNI Audio
Julien Kwasneski, President, Bay Area Sound, Inc..
Josh Rose, CEO/Co-founder, Flying Wisdom Studios

A panel of four game audio production companies will give insight on how to run a successful business. Moderated by one of their peers, questions will cover a wide range as well as involve audience participation.

Live Sound Seminar 10 **Sunday, November 7**
9:00 am – 10:45 am **Room 131**

ELECTROACOUSTIC ENHANCEMENT SYSTEMS: LARES, CONSTELLATION, AFC, SIAP

Chair: **Roger Schwenke**, Meyer Sound

Panelists: *Steve Barbar*, LARES
Lon Brannies, Yamaha Corporation
Kurt Graffy, ARUP Acoustics
Ian Hunter, The Shalleck Collaborative
Vikram Kirby, Thinkwell
Bruce C. Olson, Olson Sound Design,

Interest in electroacoustic enhancement systems has been increasing steadily for two decades. A comparison of the major systems will be made with example cases introduced.

Student Event/Career Development
RECORDING COMPETITION STEREO
Sunday, November 7, 9:00 am – 1:00 pm
Room 206

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. This event presents stereo recordings in these categories:

- Classical 9:00 am to 10:00 am
- Jazz/Blues 10:00 am to 11:00 am
- World/Folk 11:00 am to 12:00 noon
- Pop/Rock 12:00 noon to 1:00 pm

The top three finalists in each category, as identified by our judges, present a short summary of their production intentions and the key recording and mix techniques used to realize their goals. They then play their projects for all who attend. Meritorious awards are determined here and will be presented at the closing Student Delegate Assembly Meeting (SDA-2) on Sunday afternoon.

The competition is a great chance to hear the work of your fellow students at other educational institutions. Everyone learns from the judges' comments even if your project isn't one of the finalists, and it's a great chance to meet other students and faculty.

Look online for the latest list of the generous sponsors supporting the Student Recording Competitions.

Sunday, November 7 **9:00 pm** **Room 232**
AESSC Plenary Meeting

Session P23 **Sunday, Nov. 7**
9:30 am – 11:00 am **Room 226**

POSTERS: PERCEPTION AND SUBJECTIVE EVALUATION OF AUDIO

9:30 am

P23-1 Simulating Ensemble Rhythmic Interaction Based on Quantifiable Strategy Functions—
Nima Darabi,[1] *U. Peter Svensson*,[1] *Chris Chafe*[2]
[1]Norwegian University of Science and Technology, Trondheim, Norway
[2]Stanford University, Stanford, CA, USA

This paper studies the strategy taken by a pair of ensemble performers under the influence of delay. A general quantifiable measure of strategy taken by performers in an interactive rhythmic performance is represented in a form of a single-parameter strategy function. This is done by imposing an assumption about a decision-making process for "onset generation" by a participant, with one degree of freedom, to the observed data. We present specific examples of such strategy functions, suitable for different scenarios of rhythmic collaboration. By perpendicular projection of strategy functions of an ensemble performing trail onto Cartesian axis a nominal trial was transformed to a "strategy path" to show how the performers change their strategies during the course of a trial. By mathematical induction it was proven that this transformation from the time domain to a "strategy domain" is conditionally reversible, i.e., time vectors of an ensemble trial can be reconstructed by a domino effect having its time-free strategy path and given an initial state. This algorithm is considered to be a means to simulate the ensemble trials based on the overall strategies leading them.
Convention Paper 8292

9:30 am

P23-2 Hearing Threshold of Pure Tones and a Fire Alarm Sound for People Listening to Music with Headphones—
Kaori Sato,[1] *Shogo Kiryu*,[1] *Kaoru Ashihara*[2]
[1]Tokyo City University, Setagaya-ku, Tokyo, Japan
[2]Advanced Industrial Science and Technology, Tsukuba, Japan

When listening to music through headphones, the listeners may be less sensitive to environmental sounds. The sound pressure level of the fire alarm bell sound was measured in an actual internet cafe. The hearing thresholds of pure tones and the fire alarm bell sound were measured for the subjects with headphones. The minimum sound pressure level of the fire alarm bell sound recorded in the cafe was about 40 dB under the worst condition. When the subjects listened to pseudo-music signals through headphones, the hearing threshold of the fire alarm sound increased to about 80 dB.
Convention Paper 8293

9:30 am

P23-3 Psychoacoustic Measurement and Auditory Brainstem Response in the Frequency Range between 10 kHz and 30 kHz—*Motoi Koubori,*[1] *Kaoru Ashihara,*[2] *Mizuki Omata,*[1] *Masaki Kyoso,*[1] *Shogo Kiryu*[1]
[1]Tokyo City University, Setagaya-ku, Tokyo, Japan
[2]Advanced Industrial Science and Technology, Tsukuba, Japan

High-frequency components above 20 kHz can be recorded in recent high-resolution audio media. However, it is argued whether such components can be perceived or not. In this paper a psychoacoustic measurement and auditory brainstem response in the high-frequency range are reported. In the psycho-acoustic measurement, some subjects could perceive the high-frequency sounds above 20 kHz and the auditory brainstem response could be measured for one subject at the frequency of 22 kHz. However, the sound pressure levels of the thresholds were beyond 80 dB in the both measurements. The results were unremarkable. Because auditory brainstem response is a direct signal from the auditory nerve, the nerve seems not to be stimulated by weak high-frequency sounds.
Convention Paper 8294

9:30 am

P23-4 Acoustical Design of Control Room for Stereo and Multichannel Production and Reproduction—A Novel Approach—*Bogic Petrovic, Zorica Davidovic*, BoZo Electronics, MyRoom Acoustics, Beograd, Serbia

This paper describes a new method of acoustic adaptation of control rooms with a goal to satisfy the necessary conditions for a quality control room, able to provide a better mix translation to other systems, with less need for the engineer to adapt, which is compatible for stereo as well as for surround monitoring. Two practical examples of control rooms will be described, which are realized by using the new principles, along with the descriptions and experiences of sound engineers who have worked in them.
Convention Paper 8295

9:30 am

P23-5 New 10.2-Channel Vertical Surround System (10.2-VSS); Comparison Study of Perceived Audio Quality in Various Multichannel Sound Systems with Height Loudspeakers—*Sunmin Kim,*[1] *Young Woo Lee,*[1] *Ville Pulkki*[2]
[1]Samsung Electronics Co. Ltd., Suwon, Korea
[2]Aalto University School of Science and Technology, Aalto, Finland

This paper presents the listening test results of perceived audio quality with several loudspeaker arrangements in order to find the optimal configuration of loudspeakers for a next-generation multichannel sound system. We compare new reproduction formats with NHK 22.2-channel and 7.1-channel setup of Recommendation ITU-R BS.775-2. The subjective evaluations focused on the loudspeaker configurations at the top layer were carried out with test materials generated with different methods, by mixing, and by reproducing B-format recordings. The results show that the perceptual difference in the overall quality achieved with the new 10.2-channel vertical surround system with 3 top loudspeakers and the NHK 22.2-channel system was imperceptible in a grading scale used in the experiment.
Convention Paper 8296

9:30 am

P23-6 Perceptually Motivated Scoring of Musical Meter Classification Algorithms—*Matthias Varewyck, Jean-Pierre Martens*, Ghent University, Ghent, Belgium

In this paper perceived confusions between the four most popular meters 2/4, 3/4, 4/4, and 6/8 in Western music are examined. A theoretical framework for modeling these confusions is proposed and translated into a perceptually motivated objective score that can be used for the evaluation of meter classification algorithms with respect to meter labels that were elicited from a single annotator. Experiments with three artificial and two real algorithms showed that the new score is preferable over the traditional accuracy since the score rewards algorithms that make reasonable errors and seems to be more robust against different annotators.
Convention Paper 8297

9:30 am

P23-7 Toward a Classification of Audiovisual Media Content—*Ulrich Reiter*, Norwegian University of Science and Technology, Trondheim, Norway

This paper describes a qualitative experiment designed to ultimately derive a set of meaningful attributes for the classification of audiovisual media content. Whereas such attributes are available for the classification of video only content, they are missing for audiovisual content. Based on the suggestions made by Woszczyk et al. in their 1995 AES Convention paper [Preprint 4133], we have taken a closer look in a combined set of experiments, one consisting in a quality trade-off decision, and one consisting in arelevance sorting task with respect to these attributes.
Convention Paper 8298

9:30 am

P23-8 The Influence of Texture and Spatial Quality on the Perceived Quality of Blindly Separated

Audio Source Signals—*Thorsten Kastner*, University of Erlangen, Erlangen, Germany, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Blind Audio Source Separation (BASS) algorithms are often employed in applications where the aim is the acoustic reproduction of the separated source signals. The perceived quality of the reproduced signals is therefore a crucial criterion. Two different factors can be roughly distinguished that have influence on the perceived quality of blindly separated source signals. First, the quality of the separation of a desired target source from a signal mixture. Second, the preservation of the spatial image of the source, the spatial position of the target source in the signal mixture as it is perceived by the listener. Based on extensive MUSHRA-style listening tests, results are presented reflecting the influence of both factors on the overall basic audio quality of BASS signals. Further, a nonlinear regression model is set up to parametrize the influence of both factors on the subjective audio quality. A correlation of 0.98 between predicted and measured subjective quality and a root mean square prediction error of 2.7 on a [0,100] MUSHRA-scale was achieved for predicting the basic audio quality from an unknown listening test.
Convention Paper 8299

9:30 am

P23-9 Perceptual Evaluation of Spatial Audio Quality—*Hwan Shim,*[1] *Eunmi Oh,*[1] *Sangchul Ko,*[1] *Sang Ha Park*[2]
[1]Samsung Electronics, Gyeonggi-do, Korea
[2]Seoul National University, Seoul, Korea

With rapid development in multimedia devices, realistic spatial audio is of interest. In this paper we discuss how to evaluate realistic audio experience and then determine major perceptual attributes to deliver realistic audio experience to listeners. We propose eight attributes in the three categories such as “timbre,” “localization,” and “spaciousness.” Each perceptual attribute is evaluated by subjective listening tests using different surround reproduction systems including 10.2 and 22.2 channel systems. The experimental results show which spatial audio attribute is influential for realistic audio experience and which attribute is difficult to reproduce by using current reproduction systems.
Convention Paper 8300

Broadcast/Media Streaming Session 15
Sunday, November 7 9:30 am – 10:45 am
Room 133

GATING METHODS AND THE NEW LOUDNESS RECOMMENDATION EBU R 128

Co-chairs: **Florian Camerer**, ORF - Austrian TV; chairman of EBU group PLOUD
Steve Lyman, Dolby Labs

One of the most fundamental changes in the history of audio in broadcasting is underway: the change of the leveling paradigm from peak normalization to loudness normalization. This session presents two aspects of loudness normalization. The first deals with the evaluation of different gating methods that shall help to further improve the matching of the objective measurement with the subjective impression of loudness. Dolby Labs recently began building a database of wide and narrow dynamic range program samples and evaluating their subjective loudness. The loudness was assessed using the same method that was used by the ITU to develop Recommendation ITU-R BS.1770. The object of the work is to be able to evaluate the effectiveness of any proposed gated or un-gated loudness measurement method on wide and narrow dynamic range program material. Results of the current studies will be presented by Steve Lyman from Dolby Labs. In the second presentation the core document of the EBU working group PLOUD will be introduced in detail: EBU R 128 "Loudness normalization and permitted maximum level of audio signals." Florian Camerer, the chairman of PLOUD, will explain this groundbreaking recommendation as well as the Technical Documents about Loudness Metering and the descriptor Loudness Range. The documents will also be examined with a special focus on their practical implications and consequences. Audio examples will illustrate the concept of loudness normalization.

Session P24 Sunday, Nov. 7
10:15 am – 11:15 am Room 236

AUDIO TRANSMISSION

Chair: **Richard Foss**, Rhodes University, Grahamstown, South Africa

10:15 am

P24-1 Parameter Relationships in High-Speed Audio Networks—*Nyasha Chigwamba,*[1] *Richard Foss*,[1] *Robby Gurdan,*[2] *Brad Klindradt*[2]
[1]Rhodes University, Grahamstown, South Africa
[2]Universal Media Access Networks GmbH, Dusseldorf, Germany

There exists a need to remotely control and monitor parameters within audio devices. It is often necessary for changes in one parameter to affect other parameters. Thus, it is important to create relationships between parameters. The capability for relationships has existed for some time between the parameters within mixing consoles. This paper explores the parameter relationships within mixing consoles, the parameter relationships in current audio networks, and then goes on to propose some fundamental relationships that should exist between parameters. It describes how these relationships have been implemented within the X170 protocol.
Convention Paper 8301

10:45 am

P24-2 Experiment of Sixteen-Channel Audio Transmission Over IP Network by MPEG-4 ALS and Audio Rate-Oriented Adaptive Bit-Rate Video Codec—*Yutaka Kamamoto,*[1,2] *Noboru Harada*,[1,2] *Takehiro Moriya*,[1] *Sunyong Kim,*[2] *Masanori Ogawara,*[2] *Tatsuya Fujii*[2]
[1]NTT Communication Science Laboratories, Atsugi, Kanagawa, Japan
[2]NTT Network Innovation Laboratories, Yokosuka, Kanagawa, Japan

This paper describes an experiment of lossless ➥

audio transmission over the IP network and introduces a prototype codec that combines lossless audio coding and variable bit rate video coding. In the experiment 16-channel acoustic signals compressed by MPEG-4 ALS were transmitted from a live venue to a cafe via the IP network to provide high-quality music. At the cafe, received sound data were decoded losslessly and appropriately remixed for adjustment to the environment at the location. The combination of high-definition video and audio data enables fans to enjoy a musical performance at places other than the live venue at the same time. This experiment motivates us to develop a codec that guarantees audio quality.
Convention Paper 8302

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Sunday, November 7 10:30 am – 11:30 am
Room 122

Presenter: **Crispin Murray**

How to Screw Up a 5.1 Mix (Or How to Avoid It!)

Crispin Murray, Technology Development Manager from the renowned Metropolis Studios in London, UK, will present a listening session in which he explains some of the pitfalls to avoid when mixing in 5.1. He also provides a selection of some great 5.1 surround mixes that have been mastered at Metropolis, which will make for an entertaining hour of music!

Master Class 3 Sunday, November 7
11:00 am – 1:00 pm Room 132

DSP—WHY SO HARD?

Presenter: **Peter Eastty**, Oxford Digital, Oxford, UK

If you've ever wondered why audio DSP programming is so hard when the algorithms are so simple, this is the place for you. Hundreds of strange and wonderful audio processors have been developed over the past four decades and the presenter has struggled with dozens of them. In order to learn from our mistakes this master class will tour examples of gross bad practice (suitably anonymized to protect the guilty) and in doing so we'll extract some general principles useful to those who will design audio DSPs in the future. As a practical example of what can be achieved, we'll go from simulator based algorithm development to listening to production quality code in a matter of minutes.

Workshop 18 Sunday, November 7
11:00 am – 1:00 pm Room 130

AUDIO SYSTEM MEASUREMENT AND SUBJECTIVE EVALUATION

Chair: **Kurt Graffy**, ARUP

Panelists: *Charlie Hughes*, Excelsior Audio, Gastonia, NC, USA
Peter Mapp, Peter Mapp Associates
Brian McCarty, Coral Sea Studios
Floyd Toole, Consultant, Harman International

With the proliferation of measurement systems that provide both spectral and temporal analysis replacing RTAs (Real Time Analyzers), are the current standards for defining venue or cinema system/room tuning and/or coverage parameters still valid? Just because we can measure it is it subjectively significant? What are we measuring and what does/can it tell us?

Broadcast/Media Streaming Session 16
Sunday, November 7 11:00 am – 12:30 pm
Room 133

AUDIO PERFORMANCE IN STREAMING

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

Panelists: *J. Todd Baker*, SRS Labs
Alex Kosiorek, Cleveland Institute of Music
Jan Nordmann, Faunhofer

"This program is available over the air, on your computer, or on your mobile device."

It's a simple sentence, repeated by broadcasters all over the country. Like many simple sentences, it raises more questions than it answers. Battling bandwidth restrictions and with playback monitors ranging from full-range systems to ear buds, streaming presents a challenging environment for delivering high-quality audio. With the obligation to deliver programs via streaming media, how does a broadcaster maintain the highest audio quality throughout the delivery chain, and how does a broadcast engineer evaluate the audio quality to maximize their program's audio impact? Are there accepted best practices and is anyone creating regulations or standards for program evaluation? Our panel will address practices, standards, and discuss new delivery formats in a lively presentation.

Live Sound Seminar 2 Sunday, November 7
11:00 am – 1:00 pm Room 131

ECONOMICS-DRIVEN CHANGE OF TOURING

Chair: **Ken Lopez**, University of Southern California, CA, USA

Panelists: *Sam Berkow*, SIA Acoustics
David Morgan
Robert Scovill, AVID, Tom Petty
Dave Shadoan, Sound Image

The economics of touring have changed greatly over the last decade. So has the equipment technology. Both factors are a catalyst for change. The session will start with the viewpoint of a tour accountant with commentary from industry executives. Are ego and fear holding back change or are other factors involved? Find out.

Session P25 Sunday, Nov. 7
11:30 am – 12:30 pm Room 236

AUDIO IN EDUCATION

Chair: **Richard Foss**, Rhodes University, Grahamstown, South Africa

11:30 am

P25-1 The Contributions of Thomas Edison to Music Education—*Kevin D. Kelleher*, Stephen F. Austin State University, Nacogdoches, TX, USA

With the invention of the phonograph in 1877, Thomas Edison initiated an expansion of the musical experience. His device provided new learning opportunities for both amateur and professional musicians, in addition to people who claimed no musical background. Advertized as a musical educator, Edison's phonograph instructed families in the home and children at school. As a result of the recording feature of Edison's machine, distinct new methods of studying music emerged. Recordings, for example, were utilized to facilitate distance instruction, and the Edison School Phonograph offered music educators the ability to record their pupils. Recording at home, moreover, was marketed with publications that included detailed descriptions and instructive pictures of recording techniques.
Convention Paper 8303

12:00 noon

P25-2 Shaping Audio Engineering Curriculum: An Expert Panel's View of the Future—*David Tough*, Belmont University, Nashville, TN, USA

Audio engineering programs are being created and expanded at 4-year universities across the United States due to increasing demand for the subject at the university level. The purpose of this online study was to ask an expert panel of engineers to create a ranking of essential core competencies and technologies needed by audio engineering technology programs 10 years into the future (2019). A panel of 52 audio experts and industry leaders were selected as a purposive sample and an online, modified Delphi methodology was employed. The 3-round process produced 160 competencies that can be used by administrators to construct future curriculum and technologies needed for their AET programs.
Convention Paper 8304

Game Audio Session 14 Sunday, November 7
11:30 am – 1:00 pm Room 120

PHYSICS PSYCHOSIS

Presenters: **Stephen Hodde**, Associate Audio Designer, Volition, Inc. (THQ)
Damien Kastbauer, Technical Sound Designer, Bay Area Sound
Jay Weinland, Audio Lead, Bungie Studios

This session will dig into what can become a very complex implementation problem, physics. Solutions can range from simple to extremely complex depending on the demands of the game, the computational resources available, and the ambition of the audio team. Three developers will delve into the solutions they have used, discuss the pros and cons, and where they'd like to go in the future.

Special Event
PLATINUM PRODUCERS AND ENGINEERS
Sunday, November 7, 11:30 am – 1:00 pm
Room 134

Moderator: **Paul Verna**

Panelists: *Niko Bolas*
Joe Chiccarelli
Ross Hogarth

The recording industry and the technology that empowers it have undergone seismic shifts over the past decade. Despite these upheavals, the roles of the producer and engineer have remained vital to the recording process. How do today's top studio professionals stay focused in a fragmented, rapidly changing landscape? What challenges and opportunities do the struggles of the broader recording industry present? How do producers and engineers promote quality to an audience that seems more interested in convenience? These are just some of the questions that panelists Joe Chiccarelli (Frank Zappa, My Morning Jacket, Counting Crows), Niko Bolas (Neil Young, Warren Zevon, Spinal Tap), and Ross Hogarth (Lyle Lovett, John Mellencamp, Jewel) will entertain.

Exhibitor Seminar
NARAS CCD
Sunday, November 7 11:30 am – 12:30 pm
Room 112

Presenter: **John Spencer**

Credit Where Credit Is Due: Metadata!

Accurate metadata is critical for any profitable recording industry model. Most of today's commercial recording projects are "born digital," requiring a new paradigm for how projects are documented, distributed, and archived. This presentation demonstrates the Library of Congress funded multitrack project that created CCD (Content Creator Data), a standardized schema, data dictionary, field set, and free studio collection application for gathering the technical, descriptive, and participant information associated with recording projects. CCD can provide the dynamic, end to end documentation necessary to connect the dots and facilitate e-copyright, e-commerce, and archiving.

Sunday, November 7 12:00 noon Room 113
Technical Council Meeting

Special Event
LUNCHTIME KEYNOTE: IAN MOORE
Sunday, November 7, 1:15 pm – 2:15 pm
Room 133

Presenter: **Ian Moore**, Recording Artist

I'd Rather Have More dBs than Blue LEDs

Audio product designers are generally focused on making new equipment with better specifications, lower price tags, and with the latest trendy bells and whistles. That's just fine, but please don't forget that some folks depend on this gear for their livelihoods, and often what makes sense in the R&D labs doesn't make sense in the back of an 18-wheeler on tour, or in the studio at 3 am. Ian Moore has been recording music, touring, and supporting his family with his music his entire adult life. He has a few things to say to the product developers who make his tools.

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Sunday, November 7 12:30 pm – 1:30 pm
Room 122

Presenter: **Stefan Bock**

Pure Audio Blu-ray—The Ultimate Listening Experience

Stefan Bock, from Mastering Studio Munich (www.msm-studios.com), is on the forefront of the development of ➥

the revolutionary Pure Audio Blu-ray. The interest in this format from all over the world, namely from labels, studios, the hi-fi industry and the media, is becoming stronger and stronger. 2Ls Pure Audio Blu-ray releases are getting enthusiastic reviews, sell well above expectations and their latest release has received a nomination for a Grammy award! For more info: www.pureaudio-bluray.com

Exhibitor Seminar
RENKUS-HEINZ, INC.
Sunday, November 7 1:00 pm – 2:00 pm
Room 112

Presenter: **Stefan Feistel**

Acoustic Modeling Advances in AFMG's Software Suite EASE

This workshop demonstrates new tools related to the industry standard of acoustic modeling, EASE 4.3. In particular, AFMG SoundFlow for the simulation of absorption and transmission properties of multi-layer walls is introduced. The seminar also presents AFMG Reflex, a software developed to predict the sound scattering performance of diffusers. Finally, AURA Cloud is discussed, an expansion of the famous EASE room-acoustic module AURA, which allows for fast and accurate raytracing calculation by means of a remote computer cloud.

Exhibitor Seminar
PMC: MASTERS OF AUDIO SERIES
Sunday, November 7 2:00 pm – 3:00 pm
Room 122

Presenter: **Jochen Veith**

Too Much, Too Little? What to Do!

Every (home) studio will have its fair share of acoustical problems. World-renowned German acoustician Jochen Veith (JV-Acoustics) will give you some insights into how to deal with low frequency problems and will give you a better understanding on how to create a better listening environment. Jochen has designed hundreds of mixing, recording, movie, and broadcast studios over the last 20 years for major studios and artists throughout the world including Coldplay, Metropolis Studios London, Max Martin Stockholm, Walt Disney, Koch International, BMG Ariola, NBC Universal, and many more.

Session P26 **Sunday, Nov. 7**
2:30 pm – 5:00 pm **Room 220**

AUDITORY PERCEPTION

Chair: **Poppy Crum**, Dolby Laboratories

2:30 pm

P26-1 Progress in Auditory Perception Research Laboratories—Multimodal Measurement Laboratory of Dresden University of Technology— *M. Ercan Altinsoy, Ute Jekosch, Sebastian Merchel, Jürgen Landgraf*, Dresden University of Technology Dresden, Germany

This paper presents the general ideas and implementation details of the MultiModal Measurement Laboratory (MMM Lab) of Dresden University of Technology. This lab combines VR equipment for multiple modalities (auditory, tactile, vestibular, visual) and is capable of presenting high-performance, interactive simulations. The goals are to discuss the progress in auditory perception research laboratories in recent years and the technical parameters, which should be considered for the implementation of reproduction systems for different modalities.
Convention Paper 8305

3:00 pm

P26-2 Families of Sound Attributes for the Assessment of Spatial Audio—*Sarah Le Bagousse*,[1] *Mathieu Paquier*,[2] *Catherine Colomes*[1]
[1]Orange Labs – France Télécom R&D, Cesson Sévigné, France
[2]LISyC - Université de Bretagne Occidentale, Brest, France

Over the last years, studies have highlighted many features liable to be used for the characterization of sounds by several elicitation methods. These various experiments have resulted in the production of a long list of sound attributes. But, as their respective meaning and weight are not alike for assessors and listeners, the analysis of the results of a listening test based on sound criteria remains complex and difficult. The experiments reported in this paper were aimed at shortening the list of attributes by clustering them in sound families from the results of two semantic tests based on either a free categorization (i) or use of a multi-dimensional scaling method (ii).
Convention Paper 8306

3:30 pm

P26-3 Listening Tests for the Effect of Loudspeaker Directivity and Positioning on Auditory Scene Perception—*David Clark*, DLC Design, Northville, MI, USA

Using stereo playback in a typical living room, subjects were exposed to six loudspeaker configurations under double-blind conditions and asked if the auditory scene was better or worse than that presented by a reference stereo system. For all configurations, the auditory scene was judged to be plausible, but mean scores were lower than those for the reference. The reference comprised symmetrically-placed conventional box loudspeakers with subwoofers.
Convention Paper 8307

4:00 pm

P26-4 Parametric Modeling of Human Response to a Sudden Tempo Change—*Nima Darabi, Peter Svensson, Jon Forbord*, Norwegian University of Science and Telecommunications, Trondheim, Norway

A human-computer interactive subjective test was held in which 12 users tapped with a suddenly changing metronome by hand-clapping and finger-tapping. Up-sampled recorded trials with different interpolation methods were used to measure their internal timekeeper's tempo in response to each tempo step. An iterative pre-

diction error minimization method was applied on the step response signals, to identify the underlying human users' tempo-changing system related to this sensori-motor synchronization task. Experimental data indicated that the system is fairly LTI and would most likely resemble a second order damped harmonic oscillator. Fit ratio comparison showed that a delayed two-pole one-zero underdamped oscillator (P2DUZ) could be the trade-off between complexity and efficiency of the model. The related parameters for each user (as a set of their memory related built-in factors) were also extracted and shown to be slightly individual-dependent.
Convention Paper 8308

4:30 pm

P26-5 Increasing Intelligibility of Multiple Talkers by Selective Mixing—*Piotr Kleczkowski, Magdalena Plewa, Marek Pluta*, AGH University of Science and Technology, Krakow, Poland

Five tracks of speech signal were recorded. One of the tracks, the target track, consisted of spoken numbers, so that by counting the number of correctly heard words the degree of comprehension of the target talker could be quantified in each trial. Two types of mixes of all five tracks were performed: a simple mix and a selective mix. The latter mix is a development of the processing technique known as binary masking. A large group of subjects (54) listened to both types of mixes and it was found that selective mixing slightly increased the intelligibility of the target talker.
Convention Paper 8309

Session P27 — Sunday, Nov. 7
2:30 pm – 5:00 pm — Room 236

ROOM ACOUSTICS

Chair: **Søren Bech**, Bank & Olufsen a/s, Struer, Denmark

2:30 pm

P27-1 First Results from a Large-Scale Measurement Program for Home Theaters—
Tomlinson Holman, Ryan Green, University of Southern California, Los Angeles, CA, USA, Audyssey Laboratories, Los Angeles, CA, USA

The introduction of one auto-equalization system to the home theater market with an accompanying reporting infrastructure provides methods of data collection that allows research into many practical system installations. Among the results delivered are histograms of room volume, reverberation time vs. volume and frequency, early arrival sound frequency response both equalized and unequalized, and steady-state frequency response both equalized and unequalized. The variation in response over the listening area is studied as well and sheds light on contemporary use of the Schroeder frequency.
Convention Paper 8310

3:00 pm

P27-2 Improving the Assessment of Low Frequency Room Acoustics Using Descriptive Analysis —*Matthew Wankling, Bruno Fazenda, William J. Davies*, University of Salford, Salford, UK

Several factors contribute to the perceived quality of reproduced low-frequency audio in small rooms. Listeners often use descriptive terms such as "boomy" or "resonant." However a robust terminology for rating samples during listening tests does not currently exist. This paper reports on an procedure to develop such a set of subjective descriptors for low frequency reproduced sound, using descriptive analysis. The descriptors that resulted are Articulation, Resonance, and Bass Content. These terms have been used in listening tests to measure the subjective effect of changing three objective room parameters: modal decay time, room volume, and source/receiver position. Reducing decay time increased Articulation while increased preference is associated with increased Articulation and decreased Resonance.
Convention Paper 8311

3:00 pm

P27-3 Subjective Preference of Modal Control Methods in Listening Rooms—*Bruno M. Fazenda, Lucy A. Elmer, Matthew Wankling, J. A. Hargreaves, J. M. Hirst*, University of Salford, Greater Manchester, UK

Room modes are well known to cause unwanted effects in the correct reproduction of low frequencies in critical listening rooms. Methods to control these problems range from simpleloudspeaker/listener positioning to quite complex digital signal processing. Nonetheless, the subjective importance and impact of these methods has rarely been quantified subjectively. A number of simple control methods have been implemented in an IEC standard listening environment. Eight different configurations were set-up in the room simultaneously and could therefore be tested in direct comparison to each other. A panel of 20 listeners were asked to state their preferred configuration using the method of paired comparison. Results show clear winners and losers, indicating an informed strategy for efficient control.
Convention Paper 8312

4:00 pm

P27-4 Wide-Area Psychoacoustic Correction for Problematic Room Modes Using Non-Linear Bass Synthesis—*Adam J. Hill, Malcolm O. J. Hawksford*, University of Essex, Colchester, UK

Small room acoustics are characterized by a limited number of dominant low-frequency room modes that result in wide spatio-pressure variations that traditional room correction systems find elusive to correct over a broad listening area. A psychoacoustic-based methodology is proposed whereby signal components coincident only with problematic modes are filtered and substituted by virtual bass components to forge an illusion of the suppressed frequencies. A scalable and hierarchical approach is studied using the Chameleon Subwoofer Array (CSA), and subjective evaluation confirms a uniform large-area performance. Bass synthesis exploits parallel nonlinear and phase vocoder generators ➥

with outputs blended as a function of transient and steady-state signal content.
Convention Paper 8313

4:30 pm

P27-5 Beyond Coding: Reproduction of Direct and Diffuse Sounds in Multiple Environments— *James D. Johnston,*[1] *Jean-Marc Jot,*[2] *Zoran Fejzo,*[3] *Steve Hastings*[2]
[1]DTS, Inc., Kirkland, WA, USA
[2]DTS, Inc., Scotts Valley, CA, USA
[3]DTS, Inc., Calabasas. CA, USA

For many years, the difference in perception between perceptually direct sounds (i.e., sounds with a specific direction) and perceptually diffuse sounds (i.e., sounds that "surround" or "envelop" the listener) have been recognized, leading to a variety of approaches for simulating or capturing these perceptual effects. Here, we discuss a system using separation of direct and diffuse signals, or for synthetic signals (e.g., those made by modern production methods) synthesis of the diffuse signal in one of several ways, in order to enable the reproduction system, after measuring the characteristics of the playback system, to provide the best possible sensation from that particular set of playback equipment.
Convention Paper 8314

Workshop 19 Sunday, November 7
2:30 pm – 4:30 pm Room 206

THE CHALLENGE OF PRODUCING BLU-RAY

Chair: **Stefan Bock**, msm-studios Munich

Panelists: *Markus Hinz*, Minnetonka Audio Software, Minnetonka, MN, USA
John McDaniel, dts, Los Angeles, CA, USA
Joe Rice, MX Production, San Francisco, CA, USA
Mark Waldrep, AIX Media Group, Los Angeles, CA, USA

Blu-ray is catching on at an interesting rate nowadays, either as a storage medium, a platform for high definition video and audio, or even a super high quality format for audio-only titles, such as Pure Audio Blu-ray. Do mixing and mastering engineers need to change their workflow to incorporate such formats? What is the challenge of working for Blu-ray compared to other surround formats? How can lossless codecs be implemented on Blu-ray? How can Blu-ray discs be authored according to the AES X-188 draft? The panel will present the audience different authoring concepts as it can be found in current commercial products.

Game Audio Session 15 Sunday, November 7
2:30 pm – 4:00 pm Room 120

GAME AUDIO FOR THE VISUALLY IMPAIRED

Presenter: **Michelle Hinn**

Good sound design is very important for placing a player in an interactive environment. It is even more critical for those that are visually impaired. Panelists will discuss methods and strategies for sound design that allows all players to participate in the action.

Live Sound Seminar 12 Sunday, November 7
2:30 pm – 4:30 pm Room 131

OFF THE GRID: GENERATOR POWER

Chair: **Kenneth Fause**, Auerbach Pollock Friedlander

Panelists: *Bruce C. Olson*, Olson Sound Design,
Randall Venerable, Generators Unlimited

Got power? Generators bring a whole new dimension into show planning. Industry veterans and generator experts will share their lessons learned from years of operating off the grid.

Product Design Session 7 Sunday, November 7
2:30 pm – 3:45 pm Room 132

AUDIO MANUFACTURING IN A GLOBAL ECONOMY

Moderator: **Steve Macatee**, Rane Corporation

Panelists: *Rene Jaeger,* LOUD Technologies
Mike Klasco, Menlo Scientific
Pat Quilter, QSC Audio Products
Les Tyler, THAT Corporation

Once upon a time nearly all pro audio manufactures made their own products. Then the global economy brought us into the age of multinational contract manufacturers, and now it is not uncommon for products from competing vendors to go down the same conveyor belt. This workshop will explore how manufacturing in the global economy has affected the audio field from a number of perspectives, including: economics, quality, and innovation. Panelists represent manufactures from several corners of the industry.

Student Event/Career Development
STUDENT DELEGATE ASSEMBLY MEETING —PART 2
Sunday, November 7, 2:30 pm – 4:00 pm
Room 131

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 Competitions. Plans for future student activities at local, regional, and international levels will be summarized and discussed among those present at the meeting.

Master Class 4 Sunday, November 7
3:30 pm – 4:30 pm Room 133

HYBRID MIXING: A STEP BY STEP CLASS ON MIXING THE ALL-AMERICAN REJECTS HIT SINGLE "GIVES YOU HELL"

Presenter: **Eric Valentine**, Undertone Audio, Los Angeles, CA, USA

Eric Valentine will walk through the process of mixing "Gives You Hell." He will discuss all of the techniques, plug-ins, outboard gear, and external summing used in the process. Valentine will start with the unmixed material and go through the process of transforming it into the final mixed version that many folks may be familiar with.

The audience will be able to participate by asking questions throughout the process and will be invited to improve on the finished version when it is done.

Product Design Session 8 **Sunday, November 7**
4:00 pm – 5:00 pm **Room 132**

STANDARDS—HOW THEY MATTER

Presenter: **Mark Yonge**

As technologies become more complex and interdependent, the role of engineering standards to ensure compatibility has become more important. While core technologies within a product may be proprietary, the peripheral technologies that interface with other formal or informal system elements must be specified clearly. Using a non-standard interface or protocol could compromise the acceptance of a product in the market. Audio technology is no different in principle but the requirements of professional audio present their own challenges. This tutorial will offer some practical examples. It will also consider the importance of international standards in a global marketplace, and outline the approach and processes of the Audio Engineering Society Standards Committee.

Tutorial 13 **Sunday, November 7**
4:15 pm – 5:15 pm **Room 133**

COMPARATIVE LISTENING: WHAT CAN WE REALLY HEAR?

Presenter: **Eric Valentine**, Undertone Audio, Los Angeles, CA, USA

Learn objective comparative listening techniques while participating in a series of experiments that will dispel or confirm extraordinary claims made by equipment manufacturers and industry professionals.

Throughout his 20+ years as a record maker Eric Valentine has continually heard extraordinary claims about hearing the performance differences between a huge variety of tools and products used in the industry. Frequently when asked, he found people are making claims based on a listening experience that is neither scientific or objective. As the industry continues to push for better and better performance from the equipment we use in many cases (digital converters, external clocking device, cables or even mic pres) the differences have become very minute and are impossible to evaluate in a casual way. Opinions derived from these casual listening tests can be the motivation behind purchases that involve many thousands of dollars. Valentine will explain how to apply traditional scientific method to listening tests; how psychological influences play a roll; and what it all means when choosing the tools and methods for record making. The goal of this tutorial is to have all the attendees leave with techniques and information that help them make confident, objective decisions when choosing equipment to buy or use, while participating in a fun interactive series of listening experiments.

Workshop 20 **Sunday, November 7**
4:30 pm – 6:00 pm **Room 130**

RETURN TO QUALITY IN AUDIO PRODUCTION

Co-chairs: **Andres Mayo**, Andres Mayo Mastering, Buenos Aires, Argentina
Ronald Prent, Galaxy Studios, Mol, Belgium

Panelists: *Francisco Miranda*, Engineer/Studio Owner, Mexico City, Mexico
Dave Reitzas, Mixer/Producer, Los Angeles, CA, USA
Jeff Wolpert, Producer/Educator, Toronto, Ontario, Canada

We are witnessing a return to the search for better quality in current audio productions, with engineers and producers more concerned about long lasting recordings instead of just thinking about MP3 and Internet delivery. In LA, London, and Mexico (just to name a few) great sounding studios have recently opened, and established ones are regaining clientele thanks to new and improved recording and mastering systems. This panel will discuss the paradigm shift that is affecting industry professionals positively throughout the globe.

Live Sound Seminar 11 **Sunday, November 7**
4:30 pm – 6:00 pm **Room 131**

NETWORKED AUDIO FOR LIVE SOUND

Chair: **Jonathan Novick**, Audio Precision

Panelists: *Carl Bader*, Aviom
Kevin Gross, AVA Networks
Lee Minich, Lab X Technologies
David Scheirman, JBL Professional
Steve Seable, Yamaha

Are audio networks the panacea we all hoped for or is it the peril we all fear? When it comes to live sound networks offer plenty of advantages. However, there are also tradeoffs. No two networks are alike and each offers unique benefits. Should you jump in now or wait for more standardization? FIgure out if networking makes sense for your live business.

Product Design Session 9 **Sunday, November 7**
5:00 pm – 6:00 pm **Room 132**

LICENSING 3RD PARTY TECHNOLOGY: HOW TO MAKE IT WORK FOR YOU

Chair: **Alex Westner**

Panelists: *Morten Lave*
Brian Oppegaard

In the classic product design dilemma, an undersized development team is tasked with designing and building a complete product that satisfies a laundry list of market requirements in a scant mount of time. In today's rapidly moving markets, leveraging third party technologies can be an important part of any product development methodology, whether agile, iterative, linear, or waterfall. When it is not possible to develop a mature technology in-house within the constraints of the schedule, licensing technology from a third party is a viable option. This tutorial will discuss the engineering challenges of working with third parties—from design requirements to technology delivery to developer support. The benefits of third party technology integration are achieved through smart planning and close collaboration with the vendor.