AES 127th Convention Program

October 9 – 12, 2009

Jacob Javits Convention Center, New York

Pre-Convention Special Event
LIVE SOUND SYMPOSIUM: SURROUND LIVE VII
Capturing New Places—New Spaces
Thursday, October 8, 9:00 am – 5:00 pm
Manhattan Center Studios
311 West 34th Street
New York, NY 10001

Preconvention Special Event; additional fee applies

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

Panelists: *Kurt Graffy*
Fred Aldous
Kevin Cleary
Jim Hilson
Michael Pappas
Tom Sahara
Others TBA

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

Once again the extremely popular Surround Live event returns to AES's 127th Convention in New York City. Created and Produced by Frederick Ampel of Technology Visions, this marks the event's seventh consecutive workshop exclusively provided to the AES.

The day long special event will take place on Thursday, October 8, 2009 at the Manhattan Center, from 9 am until 5 pm with lunch provided to registered attendees.

In conjunction with live events, finding a critical listening space for use in monitoring and producing surround sound product is nearly impossible and usually requires commandeering an office or store room and applying a patchwork of field fabricated acoustical solutions with the materials at hand. The problem is compounded by a lack of sufficient acoustical volume, since these spaces are usually required to provide an accurate listening environment for more than one listener and the ability to accurately reproduce wide bandwidth audio. Decisions are made not only about source material content and veracity, but also about how best to process and present the multichannel surround event in a manner that translates the aural experience of a live event to a wider audience. So: ??????? How best to optimize an oftentimes less-than-ideal acoustical environment? Practical criteria and solutions are presented here to help overcome the limitations of monitoring surround sound in acoustically small spaces that are not intended for this purpose.

The program is scheduled to include presentations from:

- Kurt Graffy of ARUP Acoustics, San Francisco, CA, USA—Keynote Speaker
- Fred Aldous, Fox Sports
- Kevin Cleary, ESPN
- Jim Hilson, Dolby Laboratories
- Michael Pappas, KUVO Radio
- Tom Sahara, Turner Networks
- Others TBA

There will also be technology presentations from:

- Antoine Hurtado, Isostem France
- Norm Hurst-Sarnoff Labs.

Preliminary Program:

8:15 am – Registration and Continental Breakfast
9:00 am – Event Introduction – Frederick Ampel and Keynote Address by Kurt Graffy (Arup)
10:30 am – Coffee Break
10:40 am – The Real Words – Fred Aldous, Tom Sahara, Kevin Cleary
12:30–1:00 – Lunch (provided for ticketed participants)
1:00 pm – Mike Pappas, Michael Nunn
2:10 pm – Afternoon Break
2:30 pm – Panel Discussions sponsored by the Sports Video Group

PANEL TOPICS SUBJECT TO REVISION

SURROUND SOUND: HOW FAR WE'VE COME, WHERE WE ARE GOING

Only seven years ago the only live sports produced in Surround Sound was traditionally NFL telecasts. But today nearly everything is in Surround Sound, including esoteric Olympic sports like sailing and curling. But new challenges lie ahead, including production of on-game elements in Surround Sound and also the potential to mix not only for the home theater but for movie cinemas broadcasting live 3-D events. How has the move of Surround Sound production beyond football and NASCAR events changed the nature of sound design? What are the best ways to deliver not only the roar of the crowd and engines to the viewer but the subtle sounds of sport to viewers? Featuring: Fred Aldous, Fox Sports; Dennis Baxter, Olympic Broadcasting; Bob Dixon, NBC Olympics; JJ Johnstone, DTS

PANEL 2: SAYING GOODBYE TO STEREO

Today's Surround Sound mix is compromised by the need to deliver a quality audio signal to the mono and stereo audience. Roger Charlesworth and friends introduce a new goal for the industry: Standardizing on Surround Sound production for all live events, no matter how big or small. What are the advantages, financially and creatively, to producing only in Surround Sound? How can the industry move toward that goal? And when can it realistically be attained?

Platinum Sponsors: Neural Audio, The Sports Video Group, and Sennheiser/K&H

Gold Sponsors: Digico, Ti-Max/Outboard Electronics

Session P1 **Friday, October 9**
9:00 am – 12:30 pm **Room 1E07**

AUDIO PERCEPTION

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

9:00 am

P1-1 Effect of Whole-Body Vibration on Speech. Part I: Stimuli Recording and Speech Analysis
—*Durand Begault*, NASA Ames Research Center, Moffett Field, CA, USA

In space launch operations, speech intelligibility for radio communications between flight deck and ground control is of critical concern particularly during launch phases of flight having a predicted 12 Hz thrust oscillation. The potential effects of extreme acceleration and vibration during launch scenarios may impact vocal production. In this study, the effect of 0.5 and 0.7 g whole body vibration was evaluated on speech production of words (Diagnostic Rhyme Test word list). Six subjects were recorded in a supine position using a specially-designed chair and vibration platform. Vocal warbling, pitch modulation, and other effects were observed in the spectrographic and fundamental frequency analyses.
Convention Paper 7820

9:30 am

P1-2 Comparison of Objective Assessment Methods for Intelligibility and Quality of Speech—*Juan-Pablo Ramirez, Alexander Raake*, Deutscher Telekom Laboratories, TU Berlin, Berlin, Germany

Subjective rating of the quality of speech in narrow-band telecommunication is parametrically assessed by the so-called E-model. Intelligibility of the speech signal transmitted has a significant impact on the opinion of users. The Speech Intelligibility Index quantifies the amount of speech perceptual features available to the listener in conditions with background noise and linear frequency distortion. It has shown to be highly correlated with subjective speech recognition performance. This paper proposes a comparison between both models. It refers to, and details, improvements toward the modeling of quality in wide-band transmission.
Convention Paper 7821

10:00 am

P1-3 A Novel Listening Test-Based Measure of Intelligibility Enhancement—*Markus Kallinger*,[1] *Henning Ochsenfeld*,[1] *Anne Schlüter*[2]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]University of Applied Science Oldenburg, Oldenburg, Germany

One of the main tasks of speech signal processing aims at increasing the intelligibility of speech. Furthermore, in environments with low ambient noise the listening effort can be supported by appropriate algorithms. Objective and subjective measures are available to evaluate these algorithms' performance. However, most of these measures are not specifically designed to evaluate the performance of speech enhancement approaches in terms of intelligibility improvement. This paper proposes a novel listening test-based measure, which makes use of a speech intelligibility test, the Oldenburg Sentence Test (German *Oldenburger Satztest*, OLSA). Recent research results indicate a correlation between listening effort and speech intelligibility. Therefore, we propose to use our measure for both intelligibility enhancement for algorithms being operated at low signal-to-noise ratios (SNRs) and listening effort improvement at high SNRs. We compare the novel measure to results obtained from listening test-based as well as instrumental evaluation procedures. Good correlation and more plausible results in specific situations illustrate the potential of the proposed method.
Convention Paper 7822

10:30 am

P1-4 Which Wideband Speech Codec? Quality Impact Due to Room-Acoustics at Send Side and Presentation Method—*Alexander Raake, Marcel Wältermann, Sascha Spors*, Deutsche Telekom Laboratories, Techniche Universität Berlin, Berlin, Germany

We report on two listening tests to determine the speech quality of different wideband (WB) speech codecs. In the first test, we studied various network conditions, including WB–WB and WB–narrowband (WB–NB) tandeming, packet loss, and background noise. In addition to other findings, this test showed some codec quality rank-order changes when compared to the literature. To evaluate the hypothesis that secondary test factors lead to this rank-order effect, we conducted another speech quality listening test. Here we simulated different source material recording conditions (room-acoustics and microphone positions), processed the material with different WB speech coders, and presented the resulting files monotically in one and diotically in another test. The paper discusses why and how these factors impact speech quality.
Convention Paper 7823

11:00 am

P1-5 Evaluating Physical Measures for Predicting the Perceived Quality of Blindly Separated Audio Source Signals—*Thorsten Kastner*, University of Erlangen-Nuremberg, Erlangen, Germany, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

For blind source separation (BSS) based applications where the aim is the reproduction of the

separated signals, the perceived quality of the produced audio signals is an important key factor to rate these systems. In this paper several signal-derived features are compared to assess their relevance in reflecting the perceived audio quality of BSS signals. The most relevant features are then combined in a multiple linear regression model to predict the perceptual quality. In order to cover a large variety of source signals and different algorithms, the reference ratings are obtained from extensive listening tests rating the BSS algorithms that participated in the Stereo Source Separation Campaigns 2007 (SASSEC) and 2008 (SiSEC). Results are presented for predicting the perceived quality of SiSEC items based on a model that was calibrated using SASSEC material.
Convention Paper 7824

11:30 am

P1-6 Statistics of MUSHRA Revisited—*Thomas Sporer,*[1,2] *Judith Liebetrau,*[1] *Sebastian Schneider*[2]
[1]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[2]TU Ilmenau, Ilmenau, Germany

Listening tests are the final instance when judging perceived audio quality. To achieve reliable and repeatable results, the experimental design and the statistical analysis of results are of great importance. The “triple stimulus with hidden reference” test (Rec. ITU-R BS.1116) and the MUSHRA test (multi-stimulus with hidden reference and anchors, Rec. ITU-R BS.1534, MUSHRA) are well established standardized listening tests. Traditionally, the statistical analysis of both is based on simple parametric statistics. This paper reanalyzes the results from MUSHRA tests with alternative statistical approaches mainly considering the fact that in MUSHRA every listener is not only assigning a score to each item, but is performing an inherent ranking test and a paired comparison test (“better-worse”) between pairs of stimuli. Thus, more statistical information is made visible.
Convention Paper 7825

12:00 noon

P1-7 Statistical Analysis of ABX Results Using Signal Detection Theory—*Jon Boley,*[1] *Michael Lester*[1,2]
[1]LSB Audio, Lafayette, IN, USA
[2]Shure Incorporated, Niles, IL, USA

ABX tests have been around for decades and provide a simple, intuitive means to determine if there is an audible difference between two audio signals. Unfortunately, however, the results of proper statistical analyses are rarely published along with the results of the ABX test. The interpretation of the results may critically depend on a proper statistical analysis. In this paper a very successful analysis method known as signal detection theory is presented in a way that is easy to apply to ABX tests. This method is contrasted with other statistical techniques to demonstrate the benefits of this approach.
Convention Paper 7826

Session P2 **Friday, October 9**
9:00 am – 12:30 pm **Room 1E16**

MUSIC PRODUCTION

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

9:00 am

P2-1 Computational Optimization of a Practical End-Fire Loudspeaker Array—*Andrew Christian*

The mechanics of an array of loudspeakers that focuses coherent acoustical energy in the longitudinal direction and attempts to cancel it in the transverse direction are discussed. A practical situation is discussed in generality, which leads to the creation of two absolute measures over which the performance of an array may be evaluated. A numerical scheme to evaluate the performance of different configurations of the array is proposed and its accuracy verified. A realistic situation is proposed as a test bed. A simulation is run over all practical configurations of the array, which generates graphs showing the features of these configurations. These results are discussed and an optimized design settled upon. Further practical considerations of end-fire arrays are discussed.
Convention Paper 7827

9:30 am

P2-2 Improved Methods for Controlling Touring Loudspeaker Arrays—*Ambrose Thompson*, Martin Audio, London, UK

Users of modern array loudspeakers, used for high level sound reinforcement, demand more precise control of these systems. Current methods of control were examined and found to be inadequate for meeting a new more stringent set of user requirements. We investigate how these requirements may be formed into a mathematical model of the system suitable for numerical optimization. The primary design variable for optimization was the complex transfer functions applied to each acoustic source. We then describe how the optimized transfer functions were implemented with FIR filters on typically available hardware. Finally, comparison was made between the predicted and measured output for a large array.
Convention Paper 7828

10:00 am

P2-3 Investigations on the Inclusion of the LFE Channel in the ITU-R BS.1770-1 Loudness Algorithm—*Scott G. Norcross, Michel C. Lavoie*, Communication Research Centre, Ottawa, Ontario, Canada

The current ITU-R BS.1770-1 loudness algorithm does not include the LFE channel for 5.1-channel audio signals. It has been proposed that the LFE channel should be included in the loudness measurement to improve measurement accuracy and to fully reflect all the channels of a 5.1 audio signal. On the other hand the exclusion of the LFE channel in most downmixing systems may be ➥

one reason not to include it in the loudness measurement. Along with looking at objective pros and cons of adding the LFE channel to the BS.1770-1 loudness algorithm, results of formal subjective tests are used to show the effect of the LFE channel on perceived loudness of multichannel program material.
Convention Paper 7829

10:30 am

P2-4 Automatic Equalization of Multichannel Audio Using Cross-Adaptive Methods—*Enrique Perez-Gonzalez, Joshua Reiss*, Queen Mary, University of London, London, UK

A method for automatically equalizing a multi-track mixture has been implemented. The method aims to achieve equal average perceptual loudness on all frequencies amongst all multi-track channels. The method uses accumulative spectral decomposition techniques together with cross-adaptive audio effects to achieve equalization. The method has applications in live and recorded audio mixing where the audio engineer would like to reduce set-up time, or as a tool for inexperienced users wishing to perform audio mixing. Results are reported that show how the frequency content of each channel is modified, and that demonstrate the ability of the automatic equalization method to achieve a well-balanced and equalized final mix.
Convention Paper 7830

11:00 am

P2-5 Inside Out— Time Variant Electronic Acoustic Enhancement Provides the Missing Link for Acoustic Music Outdoors—*Steve Barbar*, E-coustic Systems, Belmont, MA, USA

No matter how good the acoustic ensemble, moving them from the concert hall to an outdoor stage dramatically changes the listening experience for both the musicians, and those in attendance—usually, not for the better. For the musicians, the loss of reflected and reverberant energy alters communication between members of the ensemble. The physiology of playing the instrument changes as well—without support from reflected and reverberant energy, musicians must compensate. Thus while the outdoor performance experience of many be deemed "good" for both those playing as well those listening, it is not the experience that either desire. This paper describes how time variant electro-acoustic enhancement has been successfully used to dramatically improve the acoustical musical experience for outdoor performance.
Convention Paper 7831

11:30 am

P2-6 Engineering Outreach for Student Chapter Activities—*Scott Porter, Todd Marco, Jeremy Joseph, Jason Morris*, The Pennsylvania State University, State College, PA, USA

The Penn State Audio Engineering Society Student Section has been active since its establishment in late 1991. Recently, the student officers have made a concerted effort to increase the section's visibility and outreach to university students in science and engineering disciplines at both the graduate and undergraduate level. To accomplish this, the authors built around the existing infrastructure by adding new events and programs to engage students at a variety of technical, artistic, and interpersonal levels. In this paper the section's core programming will be briefly discussed and followed by an examination of the additional events that have attracted new science and engineering students to the section.
Convention Paper 7832

12:00 noon

P2-7 Desktop Music Production and the Millennials: A Challenge for Educators, Researchers, and Audio Equipment and Music Software Industry—*Jan-Olof Gullö*, Royal College of Music, Stockholm, Södertörn University, Huddinge, Sweden

Music is very important for today's youth the Millennials. They often listen to music for hours every day and many also produce music by themselves. As a result young people now show different musical abilities compared with earlier generations. New software for music production, combined with the development of less expensive but more powerful computers, has made Desktop Music Production available to a large public. Producers of music production software also show a growing interest in the educational market. This raises questions about what demands this puts on the training and work of teachers in music production and audio education as well as the future challenges to suppliers of music production software and music technology.
Convention Paper 7833

Convention Paper 7834 has been withdrawn.

Tutorial 1 **Friday, October 9**
9:00 am – 10:30 am **Room 1E11**

THE GROWING IMPORTANCE OF MASTERING IN THE HOME STUDIO ERA

Presenter: **Andres Mayo**, Andres Mayo Mastering, Argentina

Artists and producers are widely using their home studios for music production, with a better cost/benefit ratio, but they usually lack technical resources, and the acoustic response of their rooms is unknown. Therefore, there is greater need for a professional mastering service in order to achieve the so-called "standard commercial quality." This tutorial presents a list of common mistakes that can be found in homemade mixes with real-life audio examples taken directly from recent mastering sessions. What can and what can't be fixed at the mastering stage?

Broadcast/Media Streaming Session 1
Friday, October 9 **9:00 am –11:00 am**
Room 1E15

STUDIO DESIGN AND ACOUSTICS: A CASE STUDY

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

Presenters: *Judy Elliott-Brown*, Connectivity, NY, USA
Chris Harmaty, Technical Structures
Dirk Noy, Walters-Storyk Design Group Europe
Marcy Ramos, M. Ramos Associates
Brian Wick, audioEngine

A team of creative design and system integration specialists will scope out a hypothetical media environment. The myriad of variables in site selection, planning, construction, systems integration, acoustics, HVAC, furniture & equipment selection and aesthetics will be considered. While the options may seem limitless, the panel's collaborative process in addressing this open-ended fantasy is guaranteed to produce an abundance of surprising recommendations and conclusions.

Games Audio 1 **Friday, October 9**
9:00 am – 11:30 am **Room 1E08**

EDUCATION IN GAME AUDIO IN THREE PARTS

Co-chairs: **Steve Horowitz**, Nick-Digital, the Code International Inc., New York, NY, USA
Steve Martz, THX, San Francisco, CA USA

Part 1: Game Production 101

Panelists: S*tephen Harwood*, Okiron Music, New York, NY, USA
Tom Salta, Persist Music, Norwalk, CT, USA
Mike Worth, Game Music Inc. and IGDA, Philadelphia, PA, USA

So you want a career in game audio but don't know where to start? This tutorial provides an introduction to the state of the art in games and interactive media. We will outline the basic components of game creation for those seeking employment in the industry. Topics covered include music, sound effects, voice-over recording, and implementation. We will discuss the parallels to film/tv/animation style productions but focus on the uniqueness of game development. The panel will also touch on fundamental concepts such as: formats, game styles, adaptive scoring, and much more.

Part 2: Meet the Team

Panelist: *Alexander Brandon*, Heatwave Interactive, Austin, TX, USA

The audio team is responsible for all aspects of game audio design and implementation. This workshop will introduce the departments of the team and discuss the roles they play in producing the final product. It includes a comprehensive description of the tasks they perform in audio design, content creation, technology development, and implementation of the audio assets. Whether you want to be an audio programmer, composer, sound designer, sound engineer, or team manager, this workshop will help you choose the right job for you.

Part 3: State of the Union

Panelists: *Paul Lipson*, Pyramind Inc. and Game Audio Network Guild, San Francisco, CA, USA
Sam Howard-Spink, NYU Steinhardt, New York, NY, USA
Ufuk Onen, Bilkent University and Sis Productions, Ankara, Turkey
Richard Stevens, Leeds Metropolitan University, Leeds, UK
Michael Sweet, Berklee College of Music, Boston, MA, USA

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 others 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.

Special Event
MOTOWN AT 50: LEON WARE, THE SONGWRITER AND PRODUCER BEHIND MARVIN GAYE'S CLASSIC *I WANT YOU*
Friday, October 9, 9:00 am – 10:30 am
Room 1E12/13

Moderators: **Jason King**
Harry Weinger

Released in 1976, Marvin Gaye's *I Want You* is perhaps the most sensual concept album in the entire Motown catalog. Romantic singles like the title track and "After the Dance" were massively influential, as they laid the framework for the Quiet Storm into the 1980s and beyond and provided inspiration for artists ranging from Todd Rundgren to Madonna. *I Want You* was crafted by visionary singer-songwriter-producer Leon Ware, who also worked in the 1970s with artists like Michael Jackson, Quincy Jones, and Minnie Riperton and later in the 1990s helped neo-soul star Maxwell rise to fame. In this rare conversation in conjunction with Motown's 50th Anniversary year, Leon Ware (currently signed to the Concord/Stax label) deconstructs the complex, groundbreaking production and engineering behind *I Want You*, and he touches on other musical highlights in his impressive multi-decade artistic legacy. Moderated by The Clive Davis Department of Recorded Music Artistic Director Jason King.

Historical Event
HISTORY OF LIVE SOUND
Friday, October 9, 9:00 am – 12:00 noon
Room 1E09

Moderator: **Thomas Shevlin**

Panelists: *Roy Clair*
David Robb
Abe Jacob
Rusty Brutsche

Presented by Toronto-based Sr. Engineer Thomas Shevlin, this session will feature a panel of pioneers of live touring and Broadway sound. Opening with a slideshow reaching back before the 1920s origins of live sound systems, Shevlin and his panelists will discuss what originally drew them to the field, its progression during their careers, and their vision of the future. The session will conclude with a lively Q & A.

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Friday, October 9 9:00 am Room 1E05

Technical Committee Meeting on Automotive Audio

Session P3 Friday, October 9
10:00 am – 11:30 am Foyer 1E

POSTERS: TRANSDUCERS AND AMPLIFIERS

10:00 am

P3-1 Target Modes in Moving Assemblies of Pleated Loudspeaker—*Jose Martínez,*[1] *Fernando Bolaños*,[1] *Enrique G. Segovia Eulogio*,[2] *Jaime Ramis Soriano*[2]
[1]Acustica Beyma S.L., Moncada, Valencia, Spain
[2]Universidad de Alicante, Alicante, Spain

In this paper we present the process followed for the adjustment of a numerical model in finite elements of the mechanical behavior of a pleated loudspeaker, based on the AMT technology (Air Motion Transformer). In this type of transducer, the diaphragm is formed by longitudinal folds. In the internal face of each one of these folds is printed a conductive ribbon. We have obtained first the participation factors and the generalized mass from the results of a natural vibration modal analysis. Next, an analysis is realized taking into account the loss factors of the materials, followed by a forced vibration modal analysis. Finally, a method is described for the characterization of the materials (Young Modulus and Loss Factor), by using modal analysis techniques.
Convention Paper 7835

10:00 am

P3-2 Cone Shape Optimization Based on FE/BE Simulation to Improve the Radiated Sound Field—*Patrick Macey*, PACSYS Limited, Nottingham, UK

An optimization procedure is used in conjunction with finite/boundary element simulation to adjust the shape of an axisymmetric cone, defined as initially straight, and improve the radiated sound field, while keeping the maximum depth as a constraint. The effect of several different objective functions is considered. The optimization procedure is made more feasible by reducing the effect of local minima by artificially high damping applied to some components within the drive unit.
Convention Paper 7836

10:00 am

P3-4 Study and Improvement of DC-Link Perturbations Models for DCI-NPC Power Amplifiers—*Vicent Sala, Luis Romeral*, Universitat Politecnica de Catalunya, Terrassa, Spain

This paper presents the most important sources of distortion in high power DCI-NPC amplifiers. These distortion sources are attributed to power supply perturbations due to the DC-Link midpoint voltage oscillations. We have presented a classic model for assessing the magnitude of these disturbances and justified the need to correct the classic model to accommodate the real load impedance variations. A new model is proposed, and this is compared and studied with the classic model by analytical, experimental, and simulation methods. This paper concludes that in control or cancellation applications it is necessary to use models that include the load impedance variations regarding the frequency.
Convention Paper 7837

10:00 am

P3-5 Active Control Based on an Estimator for the Bus-Pumping Cancellation in the Half-Bridge Class-D Audio Power Amplifiers—*Vicent Sala, Luis Romeral*, Universitat Politecnica de Catalunya, Terrassa, BCN, Spain

This paper presents a new technique to avoid the distortion generated due to midpoint voltage variations on DC-Bus of Class-D Half Bridge audio amplifiers. A usual distortion source on a Class-D amplifier is the Bus-Pumping effect. Bus-Pumping causes characteristic distortion and introduces attenuation on output gain. Due to this effect the amplifier efficiency decreases. By including distortion factors on the Half-Bridge amplifier model a new Hybrid Active Control (HAC) is implemented. The HAC minimizes distortion due to midpoint DC-Bus voltage variations. Simulation and experimental results confirm the theoretical approach of parameter influence on Bus-Pumping and the effectiveness of HAC implemented control. Results show a reduction of general distortion, which allows incremented audio amplifier efficiency.
Convention Paper 7838

10:00 am

P3-6 Simple Amplifier for Single Frequency Subwoofer—*Vladimir E. Filevski*, Audio Expert DOO, Skopje, Macedonia

Frequency mapped amplifier driving single frequency subwoofer is an inexpensive way to add the missing low tones to small satellite loudspeakers. The whole amplifier system consists of a band-pass filter (typically 20–120 Hz), an envelope detector, single frequency (typically 50–60 Hz) constant amplitude generator, mixer (that multiplies output from the envelope detector and single frequency generator), and a conventional power amplifier. The amplifier proposed in this paper unites the functions of the mixer, the generator, and the power amplifier in a single unit and does not need a DC power supply, but it runs on 50/60 Hz AC power supply, without rectifier and without big voltage-smoothing capacitors. With an appropriate MOSFETs the proposed amplifier can run directly on the 120 V/60 Hz mains supply line—without a power transformer; but in that case, it needs a loudspeaker with a sufficiently high impedance on the frequency of 60 Hz in order not to stress output transistors of the amplifier.
Convention Paper 7839

Friday, October 9 10:00 am Room 1E05

Technical Committee Meeting on Fiber Optics for Audio (formative meeting)

Workshop 1 **Friday, October 9**
10:30 am – 12:30 pm **Room 1E11**

FOLLOWING THE SCORE: FILM MUSIC FROM COMPOSITION TO RECORDING TO POSTPRODUCTION

Chair: **Ron Sadoff**, NYU Film Scoring Program, New York, NY, USA

Panelists: *Lawrence Machester*, Late Night with Jimmy Falon, New York, NY, USA
Ira Newborn, NYU Steinhardt, New York, NY, USA
Tim Starnes, NYU Steinhardt, New York, NY, USA
Dominick Tavella

In tracing the progressive stages from composing, recording, mixing, and re-mixing, a film score undergoes transformation through a collaborative process. Featuring discussion and demonstrations, key professionals will reveal how their craft and interactions impact the final product.

Panel: composer Ira Newborn (*Naked Gun, Blues Brothers*), music editor Tim Starnes (*The Lord of the Rings: The Return of the King, The Departed*), score recordist/mixer Lawrence Manchester (*The Good Shepherd, Hollywoodland*), re-recording engineer Dominick Tavella (*The Wrestler*, Ken Burns' *The National Parks*). Moderated by Director of the NYU Steinhardt Film Scoring Program, Ron Sadoff.

Master Class 1 **Friday, October 9**
11:00 am – 1:00 pm **Room 1E15**

BERNERS & ABEL

Presenters: **Johathan Abel**
David Berners

Analog Circuit Emulation for Plug-in Design

David Berners and Jonathan Abel, both of the Center for Computer Research in Music and Acoustics and Universal Audio, are two of the foremost experts in digital modeling and the software emulation of classic analog audio circuitry. They will share insights, approaches, and techniques involved in their creation of some of the production community's most widely used analog emulation algorithms.

Special Event
PLATINUM MASTERING
Friday, October 9, 11:00 am – 12:30 pm
Room 1E12/13

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

Panelists: *Greg Calbi*
Bernie Grundman

In an era when single-track downloads and online streaming dominate the recorded music landscape, we take a moment to celebrate the album as an art form. Mastering engineers behind some of the top recordings in history speak in depth about their work on those projects. They share fresh insights, behind-the-scenes anecdotes, rare photos, session notes, and audio clips from their historic sessions.

Panelists will discuss their work on landmark albums, including "Born in the USA," Michael Jackson's "Thriller," John Lennon's "Rock and Roll," Paul Simon's "Graceland," John Mayer's "Continuum," and more.

Multi-platinum and Grammy-winning mastering engineer Bob Ludwig of Gateway Mastering & DVD (Rolling Stones, Bruce Springsteen, The Police, Nirvana, Rush), will moderate this all star panel.

Friday, October 9 **11:00 am** **Room 1E05**

Technical Committee Meeting on Loudspeakers and Headphones

Broadcast/Media Streaming Session 2
Friday, October 9 **11:30 am – 1:00 pm**
Room 1E08

INNOVATIONS IN DIGITAL BROADCASTING

Chair: **David Bialik**, DKB Broadcast Associates

Panelists: *Ken Hunold*, Dolby
David Layer, NAB
Chriss Scherer, SBE Certifications
Geir Skaaden, DTS Inc.
Mike Starling, NPR
David Wilson, CE

This session will look at some of the innovations in digital broadcasting including the digital conversion of U.S. television, radio for the deaf, HD radio, Surround, IP television, receiver technology, and S.B.E. certification.

Workshop 2 **Friday, October 9**
12:00 noon – 1:00 pm **Room 1E09**

MAX FOR LIVE

Chair: **Yukio King**, Ableton AG, Berlin, Germany

Panelists: *Gerhard Behles*, Ableton AG, Berlin, Germany
Mari Kimura, Juilliard School, New York, NY, USA
David Zicarelli, Cycling '74, San Francisco, CA, USA

The integration of Max/MSP/Jitter into the Ableton Live platform is meant to expand the possibilities of how digital audio workstation software can be used. By empowering a worldwide community of technology-saavy musicians and creatives with the co-developed product Max for Live, Ableton and Cycling '74 are inviting users to increase the interoperability of their projects and embrace community-generated software solutions. This workshop will feature a discussion of Max for Live from the perspective of violinist and composer, Mari Kimura, in the context of her own ongoing work with interactive performance technology. Buttressing Kimura's artistic perspective will be David Zicarelli, CEO of Cycling '74, and Gerhard Behles, CEO of Ableton, who will provide insight into the overall vision of this joint platform.

Special Event
FREE HEARING SCREENINGS
Friday, October 9 12:00 noon–6:00 pm
Saturday, October 10 10:00 am–6:00 pm
Sunday, October 11 10:00 am–6:00 pm
Monday, October 12 10:00 am–4:00 pm
Exhibit Hall

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Attendees are invited to take advantage of a free hearing screening. 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 to sign up, please go to Booth 486 in the exhibits area.

Student/Career Development Event
STUDENT BOOTH
Friday, October 9 through Monday, October 12
Exhibits Area

The student booth is a meeting place for all students attending the convention, in addition to being the central point for communication. Students can sign up for mentoring sessions, find out more information about student events at the convention, and receive tickets to the student party. Recording competition finalists will also be posted at the student booth. Students and professionals alike will be able to listen to a variety recording competition entries submitted by AES student members. The student booth will be hosted by NYU Music Technology students and supported by the SDA.

Friday, October 9 12:00 noon Room 1E05

Technical Committee Meeting on Acoustics and Sound Reinforcement

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Friday, October 9, 1:00 pm – 2:30 pm
Room 1E12/13

Opening Remarks:
- Executive Director Roger Furness
- President Jim Anderson
- Convention Chair Agnieszka Roginska

Program:
- AES Awards Presentation
- Introduction of Keynote Speaker
- Keynote Address by William “Bill” McGlaughlin

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

BOARD OF GOVERNORS AWARD:
- **Jorge R. Urbano Calva**
- **Peter Cook**
- **Michael Fleming**
- **Steve Johnson**
- **Toru Kamekawa**
- **Michael C. Kelly**
- **Francisco Miranda Kirchner**
- **Andres Mayo**
- **Juha Merimaa**
- **Mercedes Onorato**
- **Richard Sanders (posthumous)**
- **Joshua Tidsbury**
- **Nick Zacharov**

FELLOWSHIP AWARD:
- **Jean-Marc Jot**
- **Shawn Murphy**
- **Ray Rayburn**
- **Rudolph Van Gelder**
- **Daniel Weiss**

SILVER MEDAL AWARD:
- **Ioan Allen**

DISTINGUISHED SERVICE MEDAL AWARD:
- **Irv Joel**
- **Han Tendeloo**
- **Emil Torick**

Keynote Speaker
Peabody Award-winning radio personality William “Bill” McGlaughlin has been named Keynote Speaker for the 127th AES Convention. The long-time host and music director of American Public Media’s popular *Saint Paul Sunday* radio program, McGlaughlin is a highly regarded broadcaster, educator, composer, and conductor.

In 2003 McGlaughlin began hosting *Exploring Music* a WFMT daily radio program showcasing great works of classical music. Every week the widely popular show explores a single classical music theme in hour-long daily episodes. McGlaughlin’s unique brand of information-laced enthusiasm has drawn praise for the instilling a sense of classical music appreciation within an audience that has grown to over 500,000 listeners. Beyond his career as a broadcaster and music educator, he is an accomplished orchestral musician, classical composer, and conductor

In 2004 *Exploring Music* garnered McGlaughlin, the Lifetime Achievement Award from Fine Arts Radio International. He has co-hosted the nationally syndicated radio series *Center Stage* from Wolf Trap since its inception in 1999, and since 2007 has hosted the nationally syndicated *Concerts from the Library of Congress* radio series. Among his many awards and honors are five ASCAP Awards for Adventurous Programming; the 1990 Deems Taylor Award (ASCAP) for *Saint Paul Sunday*; a 1990 Honorary Doctorate from Westminster College; and a 1995 Peabody Award for *Saint Paul Sunday*.

Entitled “Talent Doesn’t Push Buttons,” McGlaughlin’s address will consider the relationship between the on-air talent and the audio engineers who ensure the high-quality sound and technical support that contribute to a program’s long-term success.

Friday, October 9 1:30 pm Room 1E02

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

Exhibitor Seminar Friday, October 9
2:00 pm – 3:00 pm Room 1E17

ACO PACIFIC, INC.

The SLARMSolution™ and What Does Noise Sound Like? Noise and Health Issues

Presenters: **Les Blomberg,** Noise Pollution Clearinghouse
Noland Lewis, ACO Pacific, Inc.
Virgil Terrell, ACO Pacific - Midwest

What Does Noise Sound Like? How and how not to create angry neighbors. Understand, anticipate, and reduce the impact of noise on neighbors and communities. The SLARMSolution™—a discussion of the SLARM™ and NetSlarm™ systems as an innovative approach to compliance, enforcement, and the control of community, industrial, and entertainment noise issues. ACOustics Begins With ACO™

Session P4 **Friday, October 9**
2:30 pm – 7:00 pm **Room 1E07**

TRANSDUCER MODELING AND DESIGN

Chair: **Siegfried Linkwitz**, Linkwitz Lab, Corte Madera, CA, USA

2:30 pm

P4-1 Modeling the Intermodulation Distortion of a Coaxial Loudspeaker—*Edward Dupont, Stanley Lipshitz*, University of Waterloo, Waterloo, Ontario, Canada

This paper is an attempt to explain the intermodulation distortion of a coaxial loudspeaker driver. Such a loudspeaker, in which the woofer and tweeter are excited at frequencies $f1$ and $f2$ respectively, is known to produce sum and difference frequencies $f\pm = f1 \pm f2$. Generation of these can be attributed to both the nonlinearity of the equations of motion and the Lagrangian boundary behavior of the woofer. A simplified model is used consisting of an acoustic monopole located in front of a baffled planar piston. To characterize the phenomena of interest the second-order equation for pressure is used. An exact integral solution is then given for the $f\pm$ pressure terms. A special case analytic solution is also discussed. Several numerical investigations of the model are performed and compared with experiment.
Convention Paper 7840

3:00 pm

P4-2 Study and Characterization of the Odd and Even Nonlinearities in Electrodynamic Loudspeakers by Periodic Random-Phase Multisines—*Pepe Gil-Cacho,*[1] *Toon van Waterschoot,*[1] *Marc Moonen*,[1] *Søren Holdt Jensen*[2]
[1]Katholieke Universiteit Leuven (KUL), Leuven, Belgium
[2]Aalborg University, Aalborg, Denmark

In acoustic echo cancellation (AEC) applications, often times an acoustic path from a loudspeaker to a microphone is estimated by means of a linear adaptive filter. However, loudspeakers introduce nonlinear distortions that may strongly degrade the adaptive filter performance, thus nonlinear filters have to be considered. In this paper measurements of three types of loudspeakers are conducted to detect, quantify, and qualify nonlinearities by means of periodic random-phase multisines. It is shown that odd nonlinearities are more predominant than even nonlinearities over the entire frequency range. The aim of this paper is then to demonstrate that third-order (cubic) adaptive filters have to be used, which is in clear conflict with the extensive, almost unique, use of second-order (quadratic) Volterra filters.
Convention Paper 7841

3:30 pm

P4-3 The Effect of Sample Variation among Cabinets of a Line Array on Simulation Accuracy—*Stefan Feistel, Wolfgang Ahnert*, Ahnert Feistel Media Group, Berlin, Germany

Most line array systems consist of a number of discrete sound sources. For typical performance criteria of such arrays, such as the homogeneous, controlled radiation of sound or its minimum variation among mechanically identical arrays, it is important that the radiation properties such as sensitivity and directional response of the individual sources are very similar. Based on statistical means, we discuss the effect of sample variation on the overall array performance. We show that for typical modeling applications the influence of sample variation is small and that it can be neglected in most cases as a minor error. Our results are derived by three different methods, a rigorous mathematical analysis, numerical simulations, and exemplary measurements.
Convention Paper 7842

4:00 pm

P4-4 SPICE Simulation of Headphones and Earphones—*Mark Kahrs*, University of Edinburgh, Edinburgh, UK

Unlike loudspeakers, headphones and earphones are not in the mainstream of electroacoustical design. In this paper different designs for headphones and earphones are discussed and simulated with the aid of SPICE, the well-known electrical circuit simulator. These simulations can be used to perform elementary design tradeoffs. One significant difficulty is the lack of component measurements in the open literature. The paper begins with an overview of design aspects of headphones. This is followed by a review of the use of SPICE as an electroacoustical simulator. The following section details various experiments done using SPICE to explore headphone design. The conclusion decries the lack of publicly available information as well as the dearth of components.
Convention Paper 7843

4:30 pm

P4-5 A Preliminary SPICE Model to Calculate the Radiation Impedance of a Baffled Circular Piston—*Scott Porter, Stephen Thompson*, The Pennsylvania State University, State College PA, USA

Acoustic systems often use circular pistons to radiate sound into fluid media. Mathematically, the solution to the radiation impedance of a baffled circular piston is well known. Implementing the exact solution in circuit analysis packages, such as SPICE, however, can be difficult because many commercial packages do not include Bessel and Struve functions. A SPICE subcircuit is presented that calculates the radiation impedance for all frequencies to a good approximation.
Convention Paper 7844

5:00 pm

P4-6 Comparison between Measurement and Boundary Element Modelization of Subwoofers—*Manuel Melon, Christophe Langrenne, Olivier Thomas, Alexandre Garcia*, CNAM, Paris, France

At very low frequency, even large anechoic chambers cannot be used to measure subwoofers accurately. A solution consists in using the Field Separation Method (FSM). This technique allows subtracting the field reflected by the measurement room walls to the measured field, thus recovering the acoustic pressure that would have been radiated under free field conditions. In this paper Field Separation Method is used to measure two subwoofer prototypes. Results are compared to the ones given by a boundary element modelization of the subwoofers. Input velocities required for the modeling are measured by using a laser Doppler vibrometer. Comparisons are performed on the following quantities: on-axis pressure and directivity. Discrepancies between results obtained by these two methods are discussed and explained when possible.
Convention Paper 7845

5:30 pm

P4-7 Design of a Coincident Source Driver Array with a Radial Channel Phase-Plug and Novel Rigid Body Diaphragms—*Mark Dodd,*[1] *Jack Oclee-Brown*[2]
[1]GP Acoustics (UK) Ltd., Maidstone, Kent, UK
[2]KEF Audio (UK) Ltd., Maidstone, Kent, UK

The simple source characteristics that coincident-source driver arrays promise are an attractive design goal, but many engineering obstacles must be overcome to avoid undesirable complex behavior. This paper outlines an innovative design approach that achieves simple source behavior over several octaves and avoids the complex acoustical and vibrational behavior found in traditional drivers. The high frequency section of the driver combines techniques for optimizing the response introduced by Dodd and the radial channel phase-plug design, introduced by the authors. The midrange unit uses a cone with novel geometry allowing it to move as a rigid body to cover an octave above the crossover frequency. The resulting driver and its measured behavior is described in the light of some possible alternative approaches.
Convention Paper 7846

6:00 pm

P4-8 New Induction Drive Transducer Designs—*Marshall Buck,*[1] *Patrick Turnmire,*[2] *David Graebener*[3]
[1]Psychotechnology, Inc., Los Angeles, CA, USA
[2]RedRock Acoustics, Arroyo Seco, NM, USA
[3]Wisdom Audio, LLC, Carson City, NV, USA

Induction motor designs for a loudspeaker typically use a transformer assembly with a fixed primary and moving secondary (driving ring) immersed in a static magnetic field. Induction drive audio transducers can be designed to produce very high efficiency, high output mid/high devices. In another configuration a very linear, long stroke woofer with high output is realizable. Measurements will be provided on prototypes of both types of devices. The midrange exhibits an output of 83 acoustic watts with 300 Watts drive, and a maximum 10 Watt efficiency of 45%. The devices have a very linear stroke with both Bl vs. X and Le vs. X extremely flat. Reliability is enhanced over conventional voice coil drive means due to the elimination of moving lead in wires. A wide range of nominal impedances can be designed by changing the wire size and number of turns in the primary.
Convention Paper 7847

6:30 pm

P4-9 A Thin and Flexible Sound Generator Using an Electro-Active Elastomer—*Takehiro Sugimoto,*[1] *Kazuho Ono,*[1] *Akio Ando,*[1] *Yuichi Morita,*[2] *Kosuke Hosoda,*[2] *Daisaku Ishii*[2]
[1]NHK Science & Technology Research Laboratories, Setagaya-ku, Tokyo, Japan
[2]Foster Electric Co., Ltd., Akishima, Tokyo, Japan

We propose a new sound generator using electroactive elastomer (EAE), which is an elastic and flexible material. Our prototype sound generator is composed of a thin polyurethane elastomer sheet and conducting polymer electrodes. The electrodes are formed on both surfaces of the elastomer sheet and are driven by audio signals with a DC bias voltage. We conducted a transformation analysis of the EAE and found that using of the side-length change is more effective than using the thickness change. An EAE sound generator provides 30 dB higher sound pressure level (SPL) at 1 kHz than one using thickness change. The configuration design and operating conditions suitable for a sound generator are discussed.
Convention Paper 7848

Session P5 **Friday, October 9**
2:30 pm – 7:00 pm **Room 1E16**

VIRTUAL ACOUSTICS

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

2:30 pm

P5-1 The Effect of Whole-Body Vibration on Preferred Bass Equalization in Automotive Audio Systems—*Germain Simon,*[1] *Sean Olive,*[2] *Todd Welti*[2]
[1]Chalmers University, Göteborg, Sweden
[2]Harman International, Northridge, CA, USA

A set of experiments was conducted to study the effect of whole-body vibration on preferred low frequency equalization of an automotive audio system. Listeners indicated their preferred bass equalization for four different music programs reproduced through a high quality automotive audio system auditioned in situ (in the car) and through a headphone-based binaural room scanning system. The task was repeated while the listener experienced different levels of simulated and real whole-body vibrations associated with the automotive audio system itself. The results reveal that the presence of whole-body vibration can reduce the preferred level of bass equalization by as much as 3 dB depending on the program, the level of vibration, and the individual listener.
Convention Paper 7956

3:00 pm

P5-2 Using Programmable Graphics Hardware for Acoustics and Audio Rendering—*Nicolas Tsingos*, Dolby Laboratories, San Francisco, CA, USA

Over the last decade, the architecture of graphics accelerators (GPUs) has dramatically evolved, outpacing traditional general purpose processors (CPUs) with an average 2.25-fold increase in performance every year. With massive processing capabilities and high-level programmability, current GPUs can be leveraged for applications far beyond visual rendering. In this paper we offer an overview of modern programmable GPUs and how they can be applied to acoustics and audio rendering for virtual reality or gaming applications. For tasks ranging from sound synthesis and audio signal processing to numerical acoustic simulations, GPUs massive parallelism and dedicated instructions can offer a 5- to 100-fold performance improvement over traditional CPU implementations. We illustrate such benefits with results from 3-D audio processing and sound scattering simulations and discuss future opportunities for audio and acoustics applications on massively multicore processors.
Convention Paper 7850

3:30 pm

P5-3 Designing Practical Filters for Sound Field Reconstruction—*Mihailo Kolundzija,*[1] *Christof Faller*,[1] *Martin Vetterli*[1,2]
[1]Ecole Polytechnique Fédérale de Lausanne, Lausanne, Switzerland
[2]University of California at Berkeley, Berkeley, CA, USA

Multichannel sound field reproduction techniques, such as Wave Field Synthesis (WFS) and Sound Field Reconstruction (SFR), define loudspeaker filters in the frequency domain. However, in order to use these techniques in practical systems, one needs to convert these frequency-domain characteristics to practical and efficient time-domain digital filters. Additional limitation of SFR comes from the fact that it uses a numerical matrix pseudoinversion procedure, where the obtained filters are sensitive to numerical errors at low levels when the system matrix has high condition number. This paper describes physically-motivated modifications of the SFR approach that allow for mitigating conditioning problems and frequency-domain loudspeaker filter smoothing that allows for designing short time-domain filters without affecting the sound field reconstruction accuracy. It also provides comparisons of sound field reproduction accuracy of WFS and SFR using the obtained discrete-time filters.
Convention Paper 7851

4:00 pm

P5-4 Investigations on Modeling BRIR Tails with Filtered and Coherence-Matched Noise—*Fritz Menzer, Christof Faller*, Ecole Polytechnique Fédérale de Lausanne, Lausanne, Switzerland

This paper investigates to what extent the tails of left and right binaural room impulse responses (BRIRs) can be replaced by white Gaussian noise that has been processed to have the same energy decay relief and interaural coherence as the original BRIRs' tail. For this purpose BRIRs were generated consisting of two parts where the first part is taken from the original BRIR and the second part is filtered and coherence-matched noise. A subjective test was carried out to investigate how the perceived similarity between original and modeled BRIRs decreases as the split point between the parts approaches the direct sound part of the BRIRs. Also, frequency-dependent and conventional frequency-independent interaural coherence matching were compared.
Convention Paper 7852

4:30 pm

P5-5 Localization of Sound Sources in Reverberant Environments Based on Directional Audio Coding Parameters—*Oliver Thiergart, Richard Schultz-Amling, Giovanni Del Galdo, Dirk Mahne, Fabian Kuech*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Methods for spatial audio processing are becoming more important as the variety of multichannel audio applications is permanently increasing. Directional Audio Coding (DirAC) represents a well proven technique to capture and reproduce spatial sound on the basis of a downmix audio signal and parametric side information, namely direction of arrival and diffuseness of the sound. In addition to spatial audio reproduction, the DirAC parameters can be exploited further. In this paper we propose a computationally efficient approach to determine the position of sound sources based on DirAC parameters. It is shown that the proposed localization method provides reliable estimates even in reverberant environments. The approach also allows to trade off between localization accuracy and tracking performance of moving sound sources.
Convention Paper 7853

5:00 pm

P5-6 Distance Perception in Loudspeaker-Based Room Auralization—*Sylvain Favrot, Jörg M. Buchholz*, Technical University of Denmark, Kgs. Lyngby, Denmark

A loudspeaker-based room auralization (LoRA) system has been recently proposed that efficiently combines modern room acoustic modeling techniques with high-order Ambisonics (HOA) auralization to generate virtual auditory environments (VAEs). The reproduction of the distance of sound events in such VAE is very important for its fidelity. A direct-scaling distance perception experiment was conducted to evaluate the LoRA system including the use of near-field control (NFC) for HOA. Experimental results showed that (i) loudspeaker-based auralization in the LoRA system provides similar distance perception to that of the corresponding real environment and that (ii) NFC-HOA provides a significant increase in the range of perceived distances for near sound sources as compared to standard HOA.
Convention Paper 7854

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5:30 pm

P5-7 Dual Radius Spherical Cardioid Microphone Arrays for Binaural Auralization—*Frank Melchior*,[1,2] *Oliver Thiergart*,[3] *Giovanni Del Galdo*,[3] *Diemer de Vries*,[2] *Sandra Brix*[4]
[1]IOSONO GmbH, Erfurt, Germany
[2]TU Delft, Delft, The Netherlands
[3]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[4]Fraunhofer IDMT, Ilmenau, Germany

The direction dependent analysis of impulse response measurements using spherical microphone arrays can deliver a universal basis for binaural auralization. A new method using dual radius open sphere arrays is proposed to overcome limitations in practical realizations of such arrays. Different methods to combine the two radii have been analyzed and will be presented. A plane wave decomposition in conjunction with a high resolution HRTF database is used to generate a binaural auralization, wherein the different designs are simulated under ideal and real conditions. The results have been evaluated in a quality grading experiment. It is shown that the dual radius cardioids design is an effective method to enhance the perceived quality in comparison to conventional spherical array designs.
Convention Paper 7855

6:00 pm

P5-8 Recording Multichannel Sound within Virtual Acoustics—*Wieslaw Woszczyk, Tom Beghin, Martha de Francisco, Doyuen Ko*, McGill University, Montreal, Quebec, Canada

Virtual acoustic environments were implemented in a laboratory based on real-time convolution of multiple high-resolution impulse responses previously measured in real rooms. The quality of these environments has been tested during live music performance and recording in a highly demanding Virtual Haydn Project. The technological and conceptual novelty of this project is to allow one to separate the recording of the direct sound and of the ambient sound into two independent processes, each individually optimized and adjusted. The method offers recording and rehearsal rooms that are guaranteed to be quiet and free from traffic noise and other interference. At present, the technical complexity and system cost are still very high but we can expect that these will be reduced in time.
Convention Paper 7856

6:30 pm

P5-9 The Application of Compressive Sampling to the Analysis and Synthesis of Spatial Sound Fields—*Nicolas Epain, Craig Jin, André Van Schaik*, University of Sydney, Sydney, NSW, Australia

Compressive sampling provides a new and interesting tool to optimize measurements of physical phenomena with a small number of sensors. The essential idea is that close to perfect reconstruction of the observed phenomenon may be possible when it can be described by a sparse set of basis functions. In this paper we show how to apply compressive sampling techniques to the recording, analysis, and synthesis of spatially extended sound fields. Numerical simulations demonstrate that our proposed method can dramatically improve the playback of spatialized sound fields as compared, for example, with High Order Ambisonics.
Convention Paper 7857

Workshop 3 **Friday, October 9**
2:30 pm – 4:30 pm **Room 1E08**

LIES, DAMN LIES, AND STATISTICS

Chair: **Thomas Sporer**, Fraunhofer Institute for Digital Media Technology, Ilmenau, Germany

Panelists: *Jon Boley*, LSB Audio, LLC, Lafayette, IN, USA
Poppy Crum, Johns Hopkins School of Medicine, Baltimore, MD, USA
Jon Shlens, New York University, NY, USA

Listening tests have become an important part of the development of audio systems (CODECS, loudspeakers, etc.). Unfortunately, even the simplest statistics (mean and standard deviation) are often misused. This workshop will start with a basic introduction to statistics, but room will be given to discuss the pertinence of some commonly-used tests and alternative methods will be proposed, thereby making it interesting for more experienced statisticians as well. The following topics will be covered (among others): experimental design, distributions, hypothesis testing, confidence intervals, analysis of paired comparisons and ranking data, and common pitfalls potentially leading to wrong conclusions.

Workshop 4 **Friday, October 9**
2:30 pm – 4:00 pm **Room 1E11**

MASTERING IN AN EVER EXPANDING UNIVERSE (2009 AND BEYOND!)

Co-chairs: **Gavin Lurssen**
Joe Palmaccio, The Place . . . For Mastering, Nashville, TN, USA

Panelists: *Vic Anesini*, Battery Studios, New York, NY, USA
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA
Michael Romanowski, Michael Romanowski Mastering, San Francisco, CA, USA
Mark Wilder, Battery Studios, New York, NY, USA

Change has touched every corner of the professional audio community. This is all too true for the once bedrock institution of mastering. Like its cousin, the multi-room recording studio, mastering engineers and the studios they work in have splintered into many smaller, customized businesses. This panel assembles both those who have made the leap from larger facilities to owner-operator models and those who continue to work at larger facilities. Together they will discuss a state of the state of mastering in 2009. Topics will include: The burgeoning business of artist funded projects. The state of surround mastering. Mastering engineers becoming mixers. The box, the whole box and nothing but mixing in the box. Loudness—is there a backlash against LOUD?

Vinyl is dead? Not so fast! What used to take a day, now takes how long? Is anything like it used to be? Gear, techniques, and acoustics. What is the future of the full service mastering facility? Where are we headed in 2010?

Tutorial 2 — Saturday, October 10
2:30 pm – 4:30 pm — Room 1E16

THE HEARING CONSERVATION SEMINAR

Presenter: **Benjamin Kanters**, Columbia College, Chicago, IL, USA

The Hearing Conservation seminar is a new approach to promoting awareness of hearing loss and conservation. This program is specifically targeted to students and professionals in the audio and music industries. Experience has shown that this group of practitioners easily understands the concepts of hearing physiology as many of the principles and theories are the same as those governing audio and acoustics. Moreover, these people are quick to understand the importance of developing their own safe listening habits, as well as being concerned for the hearing health of their clients and the music-listening public. The seminar is a 2-hour presentation in three units: first, an introduction to hearing physiology, the second, noise-induced loss, and third, practicing effective and sensible hearing conservation.

Presented on behalf of the AES Technical Committee on Hearing and Hearing Loss Prevention.

Master Class 2 — Friday, October 9
2:30 pm – 4:30 pm — Room 1E15

ALEX CASE

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

Yesterday's FX Today

With affordable digital audio tools being continuously invented, refined, improved, extended, and upgraded, we are lucky to be a part of the audio industry at this moment. We have no excuse not to create original, beautiful art. What we do with today's ability to do anything can be informed by the creative and technical achievements expressed in touchstone recordings decades ago. This master class takes a close look at some iconic moments of signal processing innovation in recorded music history, undoing, isolating, and analyzing the effects for our edification.

Live Sound Seminar 1 — Friday, October 9
2:30 pm – 4:15 pm — Room 1E09

SOUND SYSTEM DESIGN AND INSTALLATION CONSIDERATIONS FOR CHURCHES AND HOWS

Moderator: **Bill Thrasher**, Thrasher Design Group, Inc., Kennesaw, GA, USA

Panelists: *Jeff Davidson*, First Baptist Church, Dallas, TX
David Hatmaker, Yamaha Commercial Audio Systems
Blair McNair, Schoolhouse AV
Michael Petterson, Shure, Inc.

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

Student/Career Development Event
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Friday, October 9, 2:30 pm – 3:30 pm
Room 1E06

Chair: **Teri Grossheim**

Vice Chair: **Meiling Loo**

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

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

Special Event
RECORDING THE BEATLES
Friday, October 9, 3:00 pm – 5:00 pm
Room 1E12/13

Moderators: **Brian Kehew**
Kevin Ryan

Kevin Ryan and Brian Kehew are co-authors of *Recording the Beatles*, hailed as the definitive book on the Beatles' studio sessions. *Recording the Beatles* has received high praise for its research quality and depth of detail into Abbey Road recording techniques and equipment. The result of over a decade of painstaking research and extensive interviews, this 500-page volume is the definitive study of how the Beatles' body of music was recorded. The authors will discuss their massive undertaking and share some of the techniques and technology used by the engineers, producers, and artists collectively responsible for one of the most compelling musical legacies of all time. Their presentation will include rare photos of the studios and equipment from Abbey Road in the 1960s.

Friday, October 9 — 3:00 pm — Room 1E05

Technical Committee Meeting on Transmission and Broadcasting

Exhibitor Seminar — Friday, October 9
3:15 pm – 4:15 pm — Room 1E17

RENKUS-HEINZ, INC.

RHAON—Demystifying Digital Networking

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Presenters: **Jonas Domkus**
Ralph Heinz

This seminar will provide a clear definition of digital audio and control networks and explain the RHAON solution. Brief case studies and a hands-on demonstration will highlight the technical, sonic, and cost benefits. RHAON, which offers CobraNet digital audio delivery, networked control and supervision, and user programmable DSP in every box over a single Ethernet Cat5.

Session P6 **Friday, October 9**
3:30 pm – 5:00 pm **Foyer 1E**

POSTERS: PRODUCTION AND ANALYSIS OF MUSICAL SOUNDS

3:30 pm

P6-1 Automatic Cloning of Recorded Sounds by Software Synthesizers—*Sebastian Heise,*[1] *Michael Hlatky*,[1] *Jörn Loviscach*[2]
[1]Hochschule Bremen (University of Applied Sciences), Bremen, Germany
[2]Fachhochschule Bielefeld (University of Applied Sciences), Bielefeld, Germany

Any audio recording can be turned into a digital musical instrument by feeding it into an audio sampler. However, it is difficult to edit such a sound in musical terms or even to control it in real time with musical expression. Even the application of a more sophisticated synthesis method will show little change. Many composers of electronic music appreciate the direct and clear access to sound parameters that a traditional analog synthesizer offers. Is it possible to automatically generate a synthesizer setting that approximates a given audio recording and thus clone a given sound to be controlled with the standard functions of the particular synthesizer employed? Even though this problem seems highly complex, we demonstrate that its solution becomes feasible with computer systems available today. We compare sounds on the basis of acoustic features known from Music Information Retrieval and apply a specialized optimization strategy to adjust the settings of VST instruments. This process is sped up using multi-core processors and networked computers.
Convention Paper 7858

3:30 pm

P6-2 Low-Latency Conversion of Audible Guitar Tones into Visible Light Colors—*Nermin Osmanovic*, Microsoft Corporation, Redmond, WA, USA

Automated sound-to-color transformation system makes it possible to display the corresponding color map that matches the actual note played on the guitar at the same instant. One application of this is to provide intelligent color effect "light show" for live instruments on stage during performance. By using time and frequency information of the input signal, a computer can analyze sound events and determine which tone is currently being played. The knowledge about guitar tone sound event being played on the audio input provides a basis for the implementation of a digital sound-to-light converter. The converter streams live audio input, analyzes frames based on the signal's power threshold, determines fundamental frequency of the current tone, maps this information to color, and displays the targeted light color in real-time. The final implementation includes full screen presentation mode with real time display of both pitch and intensity of the sound.
Convention Paper 7859

3:30 pm

P6-3 ThereмUS: The Ultrasonic Theremin—*André Gomes*,[1] *Daniel Albuquerque,*[2] *Guilherme Campos,*[2] *José Vieira*[2]
[1]Globaltronic-Electrónica e Telecomunicaçoes, Águeda, Portugal
[2]University of Aveiro, Aveiro, Portugal

In the Theremin, the performer's hand movements, detected by two antennas, control the pitch and volume of the generated sound. The ThereмUS builds on this concept by using ultrasonic sensing for hand position detection and processing all signals digitally, a distinct advantage in terms of versatility. Not only can different sound synthesis algorithms be programmed directly on the instrument but also it can be easily connected to other digital sound synthesis or multimedia devices; a MIDI interface was included for this purpose. The ThereмUS also features translucent panels lit by controllable RGB LED devices. This makes it possible to specify sound-color mappings in the spirit of the legendary Ocular Harpsichord by Castel.
Convention Paper 7860

3:30 pm

P6-4 Structural Segmentation of Irish Traditional Music Using Chroma at Set Accented Tone Locations—*Cillian Kelly, Mikel Gainza, David Dorran, Eugene Coyle*, Dublin Institute of Technology, Dublin, Ireland

An approach is presented that provides a structural segmentation of Irish Traditional Music. Chroma information is extracted at certain locations within the music. The resulting chroma vectors are compared to determine similar structural segments. Chroma is only calculated at "set accented tone" locations within the music. Set accented tones are considered to be impervious to melodic variation and are entirely representative of an Irish Traditional tune. Results show that comparing set accented tones represented by chroma significantly increases the structural segmentation accuracy than when set accented tones are represented by pitch values.
Convention Paper 7861

3:30 pm

P6-5 Rendering Audio Using Expressive MIDI—*Stephen J. Welburn, Mark D. Plumbley*, Queen Mary, University of London, London, UK

MIDI renderings of audio are traditionally regarded as lifeless and unnatural—lacking in expression. However, MIDI is simply a protocol for controlling a synthesizer. Lack of expression is caused by either an expressionless synthesizer or by the difficulty in setting the MIDI parameters

to provide expressive output. We have developed a system to construct an expressive MIDI representation of an audio signal, i.e., an audio representation that uses tailored pitch variations in addition to the note base pitch parameters that audio-to-MIDI systems usually attempt. A pitch envelope is estimated from the original audio, and a genetic algorithm is then used to estimate pitch modulator parameters from that envelope. These pitch modulations are encoded in a MIDI file and rerendered using a sampler. We present some initial comparisons between the final output audio and the estimated pitch envelopes.
Convention Paper 7862

3:30 pm

P6-6 Editing MIDI Data Based on the Acoustic Result—*Sebastian Heise,*[1] *Michael Hlatky*,[1] *Jörn Loviscach*[2]
[1]Hochschule Bremen (University of Applied Sciences), Bremen, Germany
[2]Fachhochschule Bielefeld (University of Applied Sciences), Bielefeld, Germany

MIDI commands provide an abstract representation of audio in terms of note-on and note-off times, velocity, and controller data. The relationship of these commands to the actual audio signal is dependent on the actual synthesizer patch in use. Thus, it is hard to implement effects such as compression of the dynamic range or time correction based on MIDI commands alone. To improve on this, we have developed software that silently computes a software synthesizer's audio output on each parameter update to support editing of the MIDI data based on the resulting audio data. Precise alignment of sounds to the beat, sample-correct changes in articulation, and musically meaningful dynamic compression through velocity data become possible.
Convention Paper 7863

3:30 pm

P6-7 Sound Production and Audio Programming of the Sound Installation GROMA—*Judith Nordbrock, Martin Rumori*, Academy of Media Arts Cologne, Cologne, Germany

In this paper we shall be picturing the sound production, in particular the mixing scenarios and the audio programming, of GROMA. GROMA is a permanent urban sound installation that incorporates early historic texts on urbanization combined with environmental sounds from two of Cologne's partner cities, Rotterdam and Liège, in an algorithmic multichannel composition. This installation was inaugurated in 2008 at the location of Europe's largest underground parking lot in Cologne, situated in the area of Rheinauhafen. For producing the sound material, special methods had to be developed, that allow for the fine-grained aesthetical design of the sound in the unconventional venue and that also support the aleatoric combination of the sound situations.
Convention Paper 7864

Live Sound Seminar 2 — Friday, October 9
4:30 pm – 6:15 pm — Room 1E09

MICROPHONE DRESSING

Moderator: **Mary McGregor**, New York, NY, USA

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

Games Audio 2 — Friday, October 9
4:30 pm – 5:30 pm — Room 1E08

LEVELING UP GAME AUDIO IMPLEMENTATIONS

Chair: **Scott Selfon**, Microsoft Corporation, Redmond, WA, USA

Panelists: *Dave Fraser*, Heavy Melody
Scott Gershin

How does a sound implementer solve the conundrum of delivering 40+ hours of interactive, dynamic, non-repetitive game audio that fits into a download, portable, or disc-based title? How can we devise and abide by reference level standards in an environment where mixing is occurring in real time? How do we get real-time DSP effects out of our gear and into the game, so the player hears what we hear? This panel of IESD members (Interactive Entertainment Sound Developers, a professional branch of the Game Audio Network Guild) will discuss these challenges, as well as other aesthetic and procedural considerations faced by sound designers, programmers, and audio directors in developing and delivering an overall sound design vision within the technical and logistical constraints of interactive media.

Exhibitor Seminar — Friday, October 9
4:30 pm – 6:30 pm — Room 1E06

MANHATTAN PRODUCERS ALLIANCE (MANHATPRO)

Welcome to the Digital Apocalypse

Presenters: **Steve Horowitz,** Moderator, The Code International
Joe Carroll, ManhatPro Founder
Frank Ferrucci, Leenalisa Music
Scott Freiman, Second Act Studios
Steve Horelick, Steve H Music
Elizabeth Rose, Elizabeth Rose Music
Sam Howard Spink, NYU
Wade Tonken, Noize Factory
John Van Eps, Van Eps Music

Welcome to the Digital Apocalypse! Advanced tips and tricks for super-sizing your career in a down economy. This fast paced, info-packed presentation is essential for composers and producers in all mediums (music producers, songwriters, educators, and students) who are interested in staying on top of the changing face of the rapidly evolving audio/music business. This high level group of industry pros will each give short presentations on specific topics: new directions in audio studios, education, record labels, licensing deals, and much more. Each mini presentation will focus on a specific area of the business. This is sure to be a freewheeling and highly informative discussion for those just starting out in the business as well as long time industry pros.

* *So You Think You Got Game? Game Audio Production, and Education:* Steve Horelick (Steve H Music), Sam Howard Spink (NYU)

* *Pro Audio Education in the Schools and Beyond: New Education and Production Models:* Joe Carroll (ManhatPro Founder), Wade Tonken (Noize Factory)

* *How to Get and Keep a Gig: The Changing Face of the Music and Audio Biz for Film/TV:* Scott Freiman (Second Act Studios), Frank Ferrucci (Leenalisa Music)

* *Music by the Foot: Advanced Lessons in Music Licensing and Dealing with Music Libraries: John Van Eps (Van Eps Music), Elizabeth Rose (Elizabeth Rose Music) Steve Horowitz - Moderator (The Code Interna-*tional)

Workshop 5 — Friday, October 9
5:00 pm – 6:30 pm — Room 1E15

WHAT WILL PERCEPTUAL AUDIO CODING STAND FOR 20 YEARS FROM NOW?

Chair: **Anibal Ferreira**, University of Porto, Porto, Portugal, ATC Labs, Chatham, NJ, USA

Panelists: *Jürgen Herre*, Fraunhofer IIS, Erlangen, Germany
Youngmoo E. Kim, Drexel University, Philadelphia, PA, USA
Bastiaan Kleijn, KTH-Royal Institute of Technology, Stockholm, Sweden
Mark Sandler, Queen Mary, University of London, UK
Gerald Schuller, Ilmenau University of Technology, Germany

As a follow-up to the Historic Program on “Perceptual Audio Coding—The First 20 Years” that took place during the 125th AES Convention in San Francisco (2008), this workshop focuses on how new trends are likely to reshape the meaning and scope of Perceptual Audio Coding (PAC) during the next two decades. A panel of experts will discuss current issues and will highlight consistent signs existing today and that suggest significant shifts and gains. In particular: how did we get here and what are the main tools of the trade? What new paradigms and divide-and-conquer approaches will shape the future of PAC? Will PAC be driven by auditory scene analysis? Will PAC move from a set of techniques that perform fairly well with general audio, to a large set of compression techniques highly tuned to specific auditory objects (or audio objects/sources) and thus highly efficient? How efficiently can we compress speech or singing and what new advances are likely to take place? How efficiently can we compress individual instrument sounds?

Special Event
PRODUCING ACROSS GENERATIONS: NEW CHALLENGES, NEW SOLUTIONS—MAKING RECORDS FOR NEXT TO NOTHING IN THE 21ST CENTURY
Friday, October 9, 5:00 pm – 7:00 pm
Room 1E12/13

Moderators: **Jesse Lauter**
Nicholas Sansano, New York University, New York, NY, USA

Panelists: *Carter Matschullat*
Bob Power
Mark Ronson
Jeff Silverman
Tony Visconti

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

Historical Event
MERCURY LIVING PRESENCE
Friday, October 9, 5:00 pm — 7:00 pm
Room 1E11

Moderator: **Tom Fine**

Presented by Tom Fine (engineer), this session will trace the technical history of one of the world’s most highly regarded classical music labels. Recognized for a catalog of ground-breaking recordings from the 1950s, ‘60s, and ‘70s the label began to flourish in the late 1940s just as the single-microphone technique was perfected. Engaging pristine audio samples, Fine will trace MLP’s progress from single-mic mono through the 3-spaced-omni stereo technique. He will also discuss the 35mm mag-film recording medium and detail the colorful 1990s CD reissue program.

Friday, October 9 — 5:00 pm — Room 1E05

Technical Committee Meeting on Hearing and Hearing Loss Prevention

Exhibitor Seminar — Friday, October 9
5:00 pm – 6:00 pm — Room 1E17

THAT CORPORATION

THAT New Professional Audio Digitally Controlled Mic Preamp

Presenters: **Rosalfonso Bortoni**
Fred Floru
Gary Hebert
Wayne Kirkwood
Bob Moses
Les Tyler

After years of not-so-clandestine R&D, THAT Corporation is finally ready to unveil its high-performance digitally-controlled microphone preamp. This seminar offers a comprehensive look at the design and offers many tips on how to apply it. The seminar will include a live demonstration.

Broadcast/Media Streaming Session 3
Friday, October 9 — 5:30 pm – 7:00 pm
Room 1E08 — CANCELLED

Session P7 **Saturday, October 10**
9:00 am – 1:00 pm **Room 1E07**

AUDIO IN MULTIMODAL APPLICATIONS

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

9:00 am

P7-1 Listening within You and without You: Center-Surround Listening in Multimodal Displays—*Thomas Santoro*,[1] *Agnieszka Roginska*,[2] *Gregory Wakefield*[3]
[1]Naval Submarine Medical Research Laboratory (NSMRL), Groton, CT, USA
[2]New York University, New York, NY, USA
[3]University of Michigan, Ann Arbor, MI, USA

Listener cognitive performance improvements from implementations of enhanced spatialized auditory displays are considered. Investigations of cognitive decision making driven by stimuli organized in "center-surround" auditory-only and visual-only arrangements are described. These new prototype interfaces employing a center-surround organization ("listening within—listening without") exploit the capability of the auditory and visual modalities for concurrent operation and facilitate their functioning to support cognitive performance in synthetic, immersive environments.
Convention Paper 7865

9:30 am

P7-2 A Loudspeaker Design to Enhance the Sound Image Localization on Large Flat Displays— *Gabriel Pablo Nava,*[1] *Keiji Hirata*,[1] *Masato Miyoshi*[2]
[1]NTT Communication Science Laboratories, Seika-cho, Kyoto, Japan
[2]Kanazawa University, Ishikawa-ken, Japan

A fundamental problem in auditory displays implemented with conventional stereo loudspeakers is that correct localization of sound images can be perceived only at the sweet spot and along the symmetrical axis of the stereo array. Although several signal processing-based techniques have been proposed to expand the listening area, less research on the loudspeaker configuration has been reported. This paper describes a new loudspeaker design that enhances the localization of sound images on the surface of flat display panels over a wide listening area. Numerical simulations of the acoustic field radiated, and subjective tests performed using a prototype panel show that the simple principle of the design effectively modifies the radiation pattern so as to widen the listening area.
Convention Paper 7866

10:00 am

P7-3 A Method for Multimodal Auralization of Audio-Tactile Stimuli from Acoustic and Structural Measurements—*Clemeth Abercrombie, Jonas Braasch*, Rensselaer Polytechnic Institute, Troy, NY, USA

A new method for the reproduction of sound and vibration for arbitrary musical source material based on physical measurements is presented. Tactile signals are created by the convolution of "uncoupled" vibration with impulse responses derived from mechanical impedance measurements. Audio signals are created by the convolution of anechoic sound with binaural room impulse responses. Playback is accomplished through headphones and a calibrated motion platform. Benefits of the method include the ability to make multimodal, side-by-side listening tests for audio-tactile stimuli perceived in real music performance situations. Details of the method are discussed along with obstacles and applications. Structural response measurements are presented as validation of the need for measured vibration signals in audio-tactile displays.
Convention Paper 7867

10:30 am

P7-4 "Worms" in (E)motion: Visualizing Emotions Evoked by Music—*Frederik Nagel*,[1] *Reinhard Kopiez*,[2] *Oliver Grewe*,[3] *Eckart Altenmüller*[2]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]Hanover University of Music and Drama, Hanover, Germany
[3]Studienstiftung des deutschen Volkes e. V., Bonn, Germany;

Music plays an important role in everyday human life. One important reason for this is the capacity of music to influence listeners' emotions. This study describes the application of a recently developed interface for the visualization of emotions felt while listening to music. Subjects (n = 38) listened to 7 musical pieces of different styles. They were asked to report their own emotions, felt in real-time, in a two-dimensional emotion space using computer software. Films were created from the time series of all self-reports as a synopsis. This technique of data visualization allows an appealing method of data analysis while providing the opportunity to investigate commonalities of emotional self-reports as well as differences between subjects. In addition to presenting the films, the authors of this study also discuss its possible applications in areas such as social sciences, musicology, and the music industry.
Convention Paper 7868

11:00 am

P7-5 Enhanced Automatic Noise Removal Platform for Broadcast, Forensic, and, Mobile Applications—*Shamail Saeed,*[1] *Harinarayanan E.V.*,[1] *Deepen Sinha*,[2] *Balaji V.*[1]
[1]ATC Labs, Noida, India
[2]ATC Labs, Chatham, NJ, USA

We present new enhancements and additions to our novel Adaptive/Automatic Wide Band Noise Removal (AWNR) algorithm proposed earlier. AWNR uses a novel framework employing dominant component subtraction followed by adaptive Kalman filtering and subsequent restoration of the dominant components. The model parameters for Kalman filtering are estimated utilizing a multi-component Signal Activity Detector ➥

(SAD) algorithm. The enhancements we present here include two enhancements to the core filtering algorithm, including the use of a multi-band filtering framework as well as a color noise model. In the first case it is shown how the openness of the filtered signal improves through the use of a two band structure with independent filtering. The use of color noise model, on the other hand, improves the level of filtering for wider types of noises. We also describe two other structural enhancements to the AWNR algorithm that allow it to better handle respectively dual microphone recording scenarios and forensic/restoration applications. Using an independent capture from a noise microphone the level of filtering is substantially increased. Furthermore for forensic applications a two/multiple pass filtering framework in which SAD profiles may be fine tuned using manual intervention are desirable.
Convention Paper 7869

11:30 am

P7-6 Catch Your Breath—Musical Biofeedback for Breathing Regulation—*Diana Siwiak, Jonathan Berger, Yao Yang*, Stanford University, Stanford, CA, USA

Catch Your Breath is an interactive musical biofeedback system adapted from a project designed to reduce respiratory irregularity in patients undergoing 4D-CT scans for oncological diagnosis. The medical application system is currently implemented and undergoing assessment as a means to reduce motion-induced distortion in CT images. The same framework was implemented as an interactive art installation. The principle is simple—the subject's breathing motion is tracked via video camera using fiducial markers, and interpreted as a real-time variable tempo adjustment to a MIDI file. The subject adjusts breathing to synchronize with a separate accompaniment line. When the subjects breathing is regular and at the desired tempo, the audible result sounds synchronous and harmonious. The accompaniment's tempo gradually decreases, which causes breathing to synchronize and slow down, thus increasing relaxation.
Convention Paper 7870
[Paper not presented but available for purchase]

12:00 noon

P7-7 Wavefield Synthesis for Interactive Sound Installations—*Grace Leslie,*[1,2] *Diemo Schwarz,*[1] *Olivier Warusfel,*[1] *Frédéric Bevilacqua,*[1] *Pierre Jodlowski*[1]
[1]IRCAM, Paris, France
[2]University of California, San Diego, La Jolla, CA, USA

Wavefield synthesis (WFS), the spatialization of audio through the recreation of a virtual source's wavefront, is uniquely suited to interactive applications where listeners move throughout the rendering space and more than one listener is involved. This paper describes the features of WFS that make it useful for interactive applications, and takes a recent project at IRCAM as a case study that demonstrates these advantages. The interactive installation GrainStick was developed as a collaboration between the composer Pierre Jodlowski and the European project Sound And Music For Everyone Everyday Everywhere Everyway (SAME) at IRCAM, Paris. The interaction design of GrainStick presents a new development in multimodal interfaces and multichannel sound by allowing users control of their auditory scene through gesture analysis performed on infrared camera motion tracking and accelerometer data.
Convention Paper 7871

12:30 pm

P7-8 Eidola: An Interactive Augmented Reality Audio-Game Prototype—*Nikolaos Moustakas, Andreas Floros, Nikolaos-Grigorios Kanellopoulos*, Ionian University, Corfu, Greece

Augmented reality audio represents a new trend in digital technology that enriches the real acoustic environment with synthesized sound produced by virtual sound objects. On the other hand, an audio-game is based only on audible feedback rather than on visual. In this paper an audio-game prototype is presented that takes advantage of the characteristics of augmented reality audio for realizing more complex audio-game scenarios. The prototype was realized as an audiovisual interactive installation, allowing the further involvement of the physical game space as the secondary component for constructing the game ambient environment. A sequence of tests has shown that the proposed prototype can efficiently support complex game scenarios provided that the necessary advanced interaction paths are available.
Convention Paper 7872

Session P8 **Saturday, October 10**
9:00 am – 12:30 pm **Room 1E16**

DATA COMPRESSION

Chair: **Christof Faller,** Ecole Polytechnique Fédérale de Lausanne, Lausanne, Switzerland

9:00 am

P8-1 Wireless Transmission of Audio Using Adaptive Lossless Coding—*David Trainor*, APTX (APT Licensing Ltd.), Belfast, Northern Ireland, UK

In audio devices, such as smartphones, media players, and wireless headsets, designers face the conflicting requirements of elevating coded audio quality and reducing algorithm complexity, device power dissipation, and transmission bit-rates. As a result, there are significant challenges in providing highest-quality real-time audio streaming between devices over wireless networks. Mathematically-lossless audio coding algorithms are an attractive means of maximizing coded audio quality. However, in the context of wireless audio transmission between portable devices, characteristics of such algorithms such as modest levels of bandwidth reduction, encoding complexity, and robustness to data loss need to be carefully controlled. Such control can be elegantly engineered by incorporating real-time adaptation and scaling into the audio coding algorithm itself. This paper describes a lossless

audio coding algorithm called apt-X Lossless, which has been designed with scalability and automated adaptation as its principal characteristics.
Convention Paper 7873

9:30 am

P8-2 Quantization with Constrained Relative Entropy and Its Application to Audio Coding—*Minyue Li, W. Bastiaan Kleijn*, KTH – Royal Institute of Technology, Stockholm, Sweden

Conventional quantization distorts the probability density of the source. In scenarios such as low bit rate audio coding, this leads to perceived distortion that is not well characterized by commonly used distortion criteria. We propose the relative entropy between the probability densities of the original and reconstructed signals as an additional fidelity measure. Quantization with a constraint on relative entropy ensures that the probability density of the signal is preserved to a controllable extent. When it is included in an audio coder, the new quantization facilitates a continuous transition between the underlying concepts of the vocoder, the bandwidth extension, and a rate-distortion optimized coder. Experiments confirm the effectiveness of the new quantization scheme.
Convention Paper 7874

10:00 am

P8-3 Enhanced Stereo Coding with Phase Parameters for MPEG Unified Speech and Audio Coding—*JungHoe Kim,*[1] *Eunmi Oh*,[1] *Julien Robilliard*[2]
[1]Samsung Electronics Co. Ltd., Gyeonggi-do, Korea
[2]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

The proposed technology is concerned with a bit-efficient way to deliver phase information. This technology is to encode only interchannel phase difference (IPD) parameter and to estimate overall phase difference (OPD) parameter at the decoder with transmitted interchannel phase difference and channel level difference. The proposed technology reduces the bit rate for phase parameters compared to the case that both IPD parameters and OPD parameters are transmitted as specified in MPEG Parametric Stereo. The entropy coding scheme for phase parameters is improved utilizing the wrapping property of the phase parameters. We introduce phase smoothing at the decoder and adaptive control of quantization precision for phase parameters to minimize annoying artifacts due to abrupt changes of quantized phase parameters. The proposed phase coding can improve stereo sound quality significantly and it was accepted as a part of the MPEG-D USAC (Unified Speech and Audio Coding) standard.
Convention Paper 7875

10:30 am

P8-4 An Enhanced SBR Tool for Low-Delay Applications—*Michael Werner*,[1] *Gerald Schuller*[1,2]
[1]Ilmenau University of Technology, Ilmenau, Germany
[2]Fraunhofer IDMT, Ilmenau, Germany

An established technique to reduce the data rate of an audio coder is Spectral Band Replication (SBR). The standard SBR tool is made for applications where encoding/decoding delay is of no concern. Our goal is to obtain an SBR system with little algorithmic delay for use in real-time applications, such as wireless microphones or video conferencing. We already developed a low delay SBR tool (LD-SBR) but it produces a relatively high amount of side information. This paper presents an enhanced SBR tool for low delay applications that uses techniques from LD-SBR in combination with Codebook Mapping (CBM). This leads to an enhanced low delay SBR tool with a reduced amount of side information without degrading audio quality.
Convention Paper 7876

11:00 am

P8-5 Audio Codec Employing Frequency-Derived Tonality Measure—*Maciej Kulesza, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

A transform codec employing efficient algorithm for detection of spectral tonal components is presented. The tonality measure used in the MPEG psychoacoustic model is replaced with the method providing adequate tonality estimates even if the tonal components are deeply frequency modulated. The reliability of the hearing threshold estimated using a psychoacoustic model with standardized tonality measure and the proposed one is investigated using objective quality testing methods. The proposed tonality estimator is also used as a basis for detector of noise-like signal bands. Instead of quantizing the noise-like signal components according to the usual transform coding scheme, the signal bands containing only noise-like components are filled with locally generated noise in the decoder. The results of the listening tests revealing usefulness of employed tonality estimation method for such a coding scenario are presented.
Convention Paper 7877

11:30 am

P8-6 New Approaches to Statistical Multiplexing for Perceptual Audio Coders Used in Multi-Program Audio Broadcasting—
Deepen Sinha,[1] *Harinarayanan E.V.,*[2] *Ranjeeta Sharma*[2]
[1]ATC Labs, Chatham, NJ, USA
[2]ATC Labs, Noida, India

In the case of multi-program audio broadcasting or transmission joint encoding is an attractive proposition. It has previously been reported that the conventional joint encoding benefits conventional perceptual audio coders in this scenario. However, previous attempts to such statistical multiplexing have focused primarily on joint bit

allocation. Here we show that such an approach is not sufficient to realize the promise of statistical multiplexing. Rather a successful strategy has two essential ingredients including a highly accurate psychoacoustic model and a coordination mechanism that goes beyond joint allocation. We describe methods to achieve these objectives and also present objective and subjective coding results.
Convention Paper 7878

12:00 noon

P8-7 Subjective Evaluation of MP3 Compression for Different Musical Genres—*Amandine Pras, Rachel Zimmerman, Daniel Levitin, Catherine Guastavino*, McGill University, Montreal, Quebec, Canada

MP3 compression is commonly used to reduce the size of digital music files but introduces a number of artifacts, especially at low bit rates. We investigated whether listeners prefer CD quality to MP3 files at various bit rates (96 kb/s to 320 kb/s), and whether this preference is affected by musical genre. Thirteen trained listeners completed an AB comparison task judging CD quality and compressed. Listeners significantly preferred CD quality to MP3 files up to 192 kb/s for all musical genres. In addition, we observed a significant effect or expertise (sound engineers vs. musicians) and musical genres (electric vs. acoustic music).
Convention Paper 7879

Workshop 6 Saturday, October 10
9:00 am – 10:15 am Room 1E15

WHAT SHAPE WILL USER INTERFACES FOR AUDIO TAKE IN THE FUTURE?

Chair: **Ellen Haas**, US Army Research Laboratory, Abeerdeen Proving Ground, MD, USA

Panelists: *Jeremy Cooperstock*, McGill University, Montreal, Quebec, Canada
Jörn Loviscach, University of Applied Sciences, Bielefeld, Germany

An international panel of researchers and engineers will describe trends and envisioned possibilities in interfaces, applications, and systems associated with audio. Whereas the present generation of interfaces for acquisition, manipulation, and presentation of audio can be characterized as a largely homogeneous collection of physical knobs, dials, buttons, or their graphical equivalents, we foresee possible advantages in an evolutionary move toward interfaces encompassing technologies including body- and brain-based sensors, virtual environments, spatial and context awareness. This workshop will explore a variety of such interfaces that may be relevant for applications ranging from audio production to information display to mobile audio experience. Audience members will be encouraged to share their design challenges with the panel in order to stimulate discussion of known and potential issues. Participants will gain a knowledge of research directions, as well as trends that could influence product design.

Workshop 7 Saturday, October 10
9:00 am – 10:30 am Room 1E11

VIRTUAL HAYDN: RECORDING AND PERFORMING IN VIRTUAL ACOUSTICS

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

Panelists: *Tom Beghin*, McGill University, Montreal, Quebec, Canada
Martha de Francisco, McGill University, Montreal, Quebec, Canada
Doyuen Ko, McGill University, Montreal, Quebec, Canada

Keyboardist, record producer, and virtual acoustics architect have joined forces to apply “virtual acoustics” for the first time on a cutting-edge commercial recording of Virtual Haydn to be released in September 2009 on four Blu-ray discs by Naxos. More than fourteen hours of music are performed on seven instruments in nine “virtual rooms.” Featured rooms range from Haydn’s own study in his Eisenstadt home to the Holywell Music Room in Oxford, England. They have been acoustically measured, digitally mapped, then precisely recreated in the recording studio allowing the performer to interact with his virtual surroundings in real time, as if he was actually performing in those rooms. The panelists will present interdisciplinary challenges of this project and celebrate the 2009 Haydn Bicentenary at AES.

Master Class 3 Saturday, October 10
9:00 am – 11:00 am Room 1E12/13

Dennis & Olhsson

Presenters: **Bob Dennis**
Bob Olhsson

Mastering the Motown Sound

Bob Dennis and Bob Olhsson were the engineers charged with mastering the records that helped Motown become a new sound heard around the world. In conjunction with Motown’s 50th Anniversary, this master class presents a rare opportunity to learn how it was done. These engineers were able to cut the loudest and cleanest records of the early 60s establishing the “Loud and Clear” standard with the Supremes first #1 hit. A simple psychoacoustic principle was used for level manipulation in both mixing and mastering to raise the perceived loudness done primarily without the use of compression and limiting. Mr. Dennis and Mr. Olhsson will show how the Motown approach can be applied to mastering today.

Broadcast/Media Streaming Session 4
Saturday, October 10 9:00 am – 10:30 am
Room 1E08

DIGITAL AUDIO NETWORKS IN THE STUDIO

Chair: **Neil Glassman**

Panelists: *Tag Borland*, Logitek
Kelly Parker, Wheatstone
Greg Shay, Telos-Omnia-Axia
Ted Staros, Harris

This session explores the use of digital audio networking within audio origination facilities by examining the underlying technologies, deployment issues, and implications for workflow. Networked audio takes advantage of established technologies to easily and economically create real-time audio networks using standard telecommunications cabling and components. With digital audio networking, multiple audio signals can be sent over a single connection and the degradation and delay that multiple conversions introduce are reduced, the routing, distribution, and mixing of audio signals is performed in the digital domain and conversion only takes place at the edges of the network. Most systems are capable of managing both audio and control/status data, while some also incorporate video, telco, and other data services.

Live Sound Seminar 3 **Saturday, October 10**
9:00 am – 10:45 am **Room 1E09**

GOOD MIC TECHNIQUES—IT'S NOT JUST FOR THE STUDIO

Moderator: **Dean Giavaras**, Shure Incorporated, Niles, IL, USA

Panelists: *John Convertino*
Dave Hewitt
Scott Lehrer
Stephan Scherthan
Robert Scovill
Dave Shadoan

One critical factor for a good sounding live event is selecting, placing, and using the right microphones for the particular application. This workshop will feature a panel of experts who will discuss their tips, tricks, and experience for getting the job done right at the start of the signal path.

Saturday, October 10 9:00 am Room 1E05

Technical Committee Meeting on Audio Recording and Mastering Systems

Saturday, October 10 9:00 am Room 1E02

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

Saturday, October 10 9:00 am Room 1E03

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

Session P9 **Saturday, October 10**
10:00 am – 11:30 am **Foyer 1E**

POSTERS: SPATIAL AUDIO

10:00 am

P9-1 Surround Sound Track Productions Based on a More Channel Headphone—*Florian M. Koenig*, Ultrasone AG & Florian König Enterprises GmbH, Germering, Germany

Stereo headphones have become very popular in the last few years. One reason was the development around mp3 and its huge market acceptance. Meanwhile, portable surround sound devices could be the successor of "stereo" applications in consumer electronics, multimedia, and games. Some problems are 3-D sound obstructive: humans need individual reproduced binaural signals (HRTF ~ outer ear shape) due to all types of headphone uses. Additionally, infants sound coloration cognition is different from adults, who investigate! Commercial headphones need an adaptive realistic 3-D image over all headphone sound sources (TV, CD, mp3 / mobile phone) with a minimum of elevation effect and a virtual distance perception. We realized surround sound mixings with a 4.0 headphone 5.1 loudspeaker compatible; they can be demonstrated as well.
Convention Paper 7880

10:00 am

P9-2 Reconstruction and Evaluation of Dichotic Room Reverberation for 3-D Sound Generation—*Keita Tanno, Akira Saji, Huakang Li, Tatsuya Katsumata, Jie Huang*, The University of Aizu, Aizu-Wakamatsu, Fukushima, Japan

Artificial reverberation is often used to increase reality and prevent the in-the-head localization in a headphone-based 3-D sound system. In traditional methods, diotic reverberations were used. In this research, we measured the impulse responses of some rooms by a Four Point Microphone method, and calculated the sound intensity vectors by the Sound Intensity method. From the sound intensity vectors, we obtained the image sound sources. Dichotic reverberation was reconstructed by the estimated image sound sources. Comparison experiments were conducted for three kinds of reverberations, i.e., diotic reverberations, dichotic reverberations, and dichotic reverberations added with Head-Related Transfer Functions (HRTF). From the results, we could clarify the 3-D sounds reconstructed by dichotic reverberations with Head-Related Transfer Functions have more spatial extension than other methods.
Convention Paper 7881

10:00 am

P9-3 Reproduction 3-D Sound by Measuring and Construction of HRTF with Room Reverberation—*Akira Saji, Keita Tanno, Jie Huang*, The University of Aizu, Aizu-Wakamatsu, Fukushima, Japan

In this paper we propose a new method using HRTFs that contain room reverberations (R-HRTF). The reverberation is not added to the dry sound source separated with HRTF but contained at their measured process in the HRTFs. We measured the HRTFs in a real reverberant environment for directions of azimuth 0, 45, 90, 135 (left side) and elevation from 0 to 90 (step of 10 degrees) degrees, then constructed a 3-D sound system with the measured R-HRTF with headphones and examined if the sound reality is improved. As a result, we succeed in creating a 3-D spatial sound system with more reality compared with a traditional HRTFs sound system by signal processing.
Convention Paper 7882

10:00 am

P9-4 3-D Sound Synthesis of a Honeybee Swarm —*Jussi Pekonen, Antti Jylhä*, Helsinki University of Technology, Espoo, Finland

Honeybee swarms are characterized by their buzzing sound, which can be very impressive close to a hive. We present two techniques and their real-time sound synthesis of swarming honeybees in 3-D multichannel setting. Both techniques are based on a source-filter model using a sawtooth oscillator with all-pole equalization filter. The synthesis is controlled by the motion of the swarm, which is modeled in two different ways: as a set of coupled individual bees or with a swarming algorithm. The synthesized sound can be spatialized using the location information generated by the model. The proposed methods are capable of producing a realistic honeybee swarm effect to be used in, e.g., virtual reality applications.
Convention Paper 7883

10:00 am

P9-5 An Investigation of Early Reflection's Effect on Front-Back Localization in Spatial Audio— *Darrin Reed, Robert C. Maher*, Montana State University, Bozeman, MT, USA

In a natural sonic environment a listener is accustomed to hearing reflections and reverberation. It is conceived that early reflections could reduce front-back confusion in synthetic 3-D audio. This paper describes an experiment to determine whether or not simulated reflections can reduce front-back confusion for audio presented with nonindividualized HRTFs via headphones. Although the simple addition of a single-order reflection is not shown to eliminate all front-back confusions, some cases of lateral reflections from a side boundary can be shown to both assist and inhibit localization ability depending on the relationship of the source, observer, and reflective boundary.
Convention Paper 7884

10:00 am

P9-6 Some Further Investigations on Estimation of HRIRs from Impulse Responses Acquired in Ordinary Sound Fields—*Shouichi Takane*, Akita Prefectural University, Akita, Japan

The author's group proposed the method for estimation of Head-Related Transfer Functions (HRTFs) or its corresponding impulse response referred to as Head-Related Impulse Responses (HRIRs) [Takane et al., Proc. JCA(2007)]. In this paper the proposed method was further investigated in two parameters affecting the performance of the estimation. The first parameter is the order of AR coefficients, indicating how many past samples are assumed to be related to the current sample. It was found that Signal-to-Deviation Ratio (SDR) was improved by using the proposed method when the order of AR coefficients was about a half of the cutout points. The second parameter is the number of samples used for the computation of AR coefficients. It was shown from the results that SDR was greatly improved when this number corresponds to the duration of the response. This indicates the proposed method properly works in ideal situations.
Convention Paper 7885

10:00 am

P9-7 Virtual Ceiling Speaker: Elevating Auditory Imagery in a 5-Channel Reproduction— *Sungyoung Kim,*[1] *Masahiro Ikeda,*[1] *Akio Takahashi,*[1] *Yusuke Ono*,[1] *William L. Martens*[2]
[1]Yamaha Corp., Iwata, Shizuoka, Japan
[2]University of Sydney, Sydney, NSW, Australia

In this paper we propose a novel signal processing method called Virtual Ceiling Speaker (VCS) that creates virtually elevated auditory imagery via a 5-channel reproduction system. The proposed method is based on transaural crosstalk cancellation using three channels: center, left-surround, and right-surround. The VCS reproduces a binaurally elevated signal via two surround loudspeakers that inherently reduce transaural crosstalk, while the residual crosstalk component is suppressed by a center channel signal that is optimized for natural perception of elevated sound. Subjective evaluations show that the virtually elevated auditory imagery maintains similar perceptual characteristics when compared to sound produced from an elevated loudspeaker. Moreover, the elevated sound contributes to an enhanced sense of musical expressiveness and spatial presence in music reproduction.
Convention paper 7886

10:00 am

P9-8 Spatial Soundfield Reproduction with Zones of Quiet—*Thushara Abhayapala, Yan Jennifer Wu*, Australian National University, Canberra, Australia

Reproduction of a spatial soundfield in an extended region of open space with a designated quiet zone is a challenging problem in audio signal processing. We show how to reproduce a given spatial soundfield without altering a nearby quiet zone. In this paper we design a spatial band stop filter over the zone of quiet to suppress the interference from the nearby desired soundfield. This is achieved by using the higher order spatial harmonics to cancel the undesirable effects of the lower order harmonics of the desired soundfield on the zone of quiet. We illustrate the theory and design by simulating a 2-D spatial soundfield.
Convention Paper 7887

Saturday, October 10 10:00 am Room 1E05

Technical Committee Meeting on High Resolution Audio

Workshop 8 Saturday, October 10
10:30 am – 11:15 am Room 1E15

INTERACTING WITH SEMANTIC AUDIO—BRIDGING THE GAP BETWEEN HUMANS AND ALGORITHMS

Chair: **Michael Hlatky**, Hochschule Bremen (University of Applied Sciences), Bremen, Germany

Panelists: *Rebecca Fiebrink*, Princeton University, Princeton, NJ, USA

Peter Grosche, Max-Planck-Institut für Informatik, Saarbrücken, Germany
Sebastian Heise, Hochschule Bremen, Bremen, Germany
Jay LeBoeuf, Imagine Research, San Francisco, CA, USA
Jörn Loviscach, Fachhochschule Bielefeld, Bielefeld, Germany
Michela Magas, Goldsmiths, University of London, London, UK
Vincent Verfaille, McGill University, Montreal, Quebec, Canada

Technologies under the heading "Semantic Audio" have undergone a fascinating development in the past few years. Hundreds of algorithms have been developed; first applications have made their way from research into possible mainstream application. However, the current level of awareness among prospective users and the amount of actual practical use do not seem to live up to the potential of semantic audio technologies. We argue that this is more an issue concerning interface and interaction than a problem concerning the robustness of the applied algorithms or a lack of need in audio production. The panelists of this workshop offer ways to improve the usability of semantic audio techniques. They look into current applications in off-the-shelf products, discuss the use in a variety of specialized applications such as custom-tailored archival solutions, demonstrate and showcase their own developments in interfaces for semantic audio, and propose future directions in interface and interaction development for semantic audio technologies ranging from audio file retrieval to intelligent audio effects.

The second half of this workshop (11:30 am – 1:00 pm, Foyer 1E) includes hands-on interactive experiences provided by the panel.

Broadcast/Media Streaming Session 5
Saturday, October 10 10:30 am – 12:00 noon
Room 1E08

IP AUDIO—OUT OF THE STUDIO: CONNECTING ANYWHERE

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

Panelists: *Steve Church*, Telos
Chris Crump, Comrex
Robert Marshall, Source Elements
Alvin Sookoo, Musicam USA
Rolf Taylor, APT
Kevin Webb, Tieline Technology

Once the work came to the studio, now the studio has to connect to the world. Whether your audio is voice-over talent calling from their lake-front studio, a correspondent reporting from Afghanistan, a reporter on the scene of a breaking story, a technician monitoring a remote transmitter, or a spirited talk-show discussion with phone-in panelists and callers, the demands on connectivity for broadcasters and producers have dramatically increased. Now the same technology connecting the desktop or mobile device to your favorite network provides the potential for an "anything/anywhere" audio path. But does it work, what have the manufacturers done to address audio quality and quality of service, how universal is IP, what kind of hardware and software is needed, does it solve my problem, and does it really work? The answers will be supplied by some of the leading manufacturers whose hardware and software make the IP connections work. This is a discussion for networks, broadcasters, producers, studios, and facilities who want to know more about how to extend their professional network.

Workshop 9 Saturday, October 10
11:00 am – 1:00 pm Room 1E11

MICROPHONES—WHAT TO LISTEN FOR AND WHAT SPECS TO LOOK FOR

Chair: **Eddy B. Brixen**, EBB-consult, Smorum, Denmark

Panelists: *Gary Baldassari*, DPA Microphones, USA
Jason Corey, University of Michigan, MI, USA
David Josephson, Josephson Engineering
Douglas McKinnie, Middle Tennessee State University, TN, USA
Ossian Ryner, Danish Broadcasting, Denmark

When selecting microphones for a specific music recording, it is worth knowing what to expect and what to listen for. Accordingly it is good to know what specifications that would be optimum for that microphone. This workshop explains the process of selecting a microphone both from the aesthetical as well as the technical point of view. What to expect when placing the microphone is explained and demonstrated. This is not a "I feel like..." presentation. All presenters on the panel are serious and experienced engineers and tonmeisters. The purpose of this workshop is to encourage and teach young engineers and students to take advantage by taking a closer look at the specifications the next time they are going to pick a microphone for a job.

Live Sound Seminar 4 Saturday, October 10
11:00 am – 12:45 pm Room 1E09

EXPLORING THE LOW END— MEASURING AND ALIGNING SUBWOOFERS

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

Panelists: *Derek Featherstone*, FOH Mixer, The Dead
Ralph Heinz, Renkus Heinz
Albert Lecesse, Audio analysts
Jim Wischmeyer, Bag End

In measuring and setting up sound systems, no topic raises more questions (and eyebrows) than how to measure and optimize the performance of subwoofers. Whether considering how to align a subwoofer to the low frequency section of a cluster or trying to create an array of subwoofers that offers "pattern" control—figuring out which measurements will be helpful and how to effectively understand and use the measured data can be a daunting task. This live sound seminar will use real world data, measured live during the event, to explore these and other issues related to measuring and optimizing a system's low end performance.

Student/Career Development Event
CAREER/JOB FAIR
Saturday, October 10, 11:00 am – 1:00 pm
Foyer 1E

The Career/Job Fair will feature several companies from the exhibit floor. All attendees of the convention, students and professionals alike, are welcome to come talk ➥

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

Saturday, October 10 11:00 am Room 1E05

Technical Committee Meeting on Audio for Games

Workshop 10 Saturday, October 10
11:30 am – 1:00 pm Room 1E15

CAN 21ST CENTURY TECHNOLOGY OUTDO THE AUDIO ACHIEVEMENTS OF THE 20TH?

Chair: **Terry Manning**, Compass Point Studios

Panelists: *Oliver Archut*, TAB/Funkenwerk, Gaylord, KS, USA
Larry Janus, Tube Equipment Corporation
Jeff Roberts, Latch Lake Music, Eagan, MN, USA

This workshop will discuss continuing, and outdoing, audio design achievements of the 20th Century into the 21st. Our panel of experts will discuss criteria that made classic gear great, and how the designs created them and the philosophies implemented them are to continue into future. Attention is paid to the designs themselves, the needs and uses in the studio, and the difficulties encountered. Archut, Manning, and Roberts collaborated in construction of the CS-1 Microphone, which hearkens back to classics, yet is also innovative. With Janus taking the lead, all contributed to the Moonray Preamplifier, and are working on other innovative products.

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

Moderator: **Mr. Bonzai**

Panelists: *Jonatha Brooke*
Bob Clearmountain
Scott Jacoby
Kevin Killen
Maiysha
Duncan Sheik

Top recording artists share their perspectives on the recording process—how they choose producers and engineers, what they look for in recording facilities, how the recording environment informs the creative process. Artists are joined on the panel by producers and engineers with whom they've worked.

Saturday, October 10 11:30 am Room 1E02

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

Saturday, October 10 11:30 am Room 1E03

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

Exhibitor Seminar Saturday, October 10
11:45 am – 12:45 pm Room 1E17

SADIE

Native or Hardware Solutions? Which Is Best for Audio Recording and Production?

Presenters: **Graham Boswell, Mark Evans**

Audio professionals now face a choice between hardware-based DSP systems and native systems that harness computer power and utilize your soundcard for I/O. As a result, native solutions are cheaper to produce and offer a more cost effective alternative, but is there still a place for hardware solutions possibly offering a more robust signal path and lower latency? And what about workflow?

Broadcast/Media Streaming Session 6
Saturday, October 10 12:00 noon – 1:00 pm
Room 1E08

MOBILE TV

Chair: **Jim Kutzner**, PBS

The Open Mobile Video Coalition and the broadcasting industry, the broadcast and consumer equipment vendors, and the Advanced Television Systems Committee are nearing the completion of the Mobile DTV broadcast standard. Trials and testing are underway, and many stations are moving to implementation. This presentation will be an overview of developments to date and where we are heading in the next year.

Saturday, October 10 12:00 noon Room 1E05

Technical Committee Meeting on Archiving Restoration and Digital Libraries

Special Event
LUNCHTIME KEYNOTE:
DAVE GIOVANNONI OF FIRST SOUNDS
Saturday, October 10, 1:00 pm – 2:00 pm
Room 1E15

Before Edison, Part 2—Recovering (and Reinterpreting) the World's Earliest Sound Recordings

First Sounds rewrote history last year when it recovered one of mankind's first recordings of its own voice, made in Paris in 1860—advancing by 17 years the invention of audio recording. Attendees at the 125th AES Convention were the first to hear what was then believed to be the world's second-oldest retrievable sound. This year First Sounds founder David Giovannoni returns to AES to report the most recent discoveries and introduce even older sounds. He'll tell of finding a seminal cache of documents that trace (literally) the development of the phonautograph from proof of concept to laboratory instrument. He'll describe the technical challenges of evoking sound from primative recordings made to be seen, not heard. And he'll recount how the inventor's own voice was revealed after posing for a year as the phantasm of a young woman.

First Sounds is an informal collaborative of sound historians, audio engineers, archeophonists, and other individuals who freely contribute their time, expertise, and resources to make mankind's earliest audio recordings audible to all people for all time. For more information go to www.firstsounds.org

David Giovannoni is the founder and President of AudiGraphics, Inc., a firm that provides management tools to National Public Radio, the Corporation for Public Broadcasting, and hundreds of public radio stations nationwide. In 2005 he turned over his operational responsibilities in order to pursue his avocation for historical sound recordings. In the last three years his historic CD reissues and liner notes on Archeophone Records have earned him five Grammy nominations and one Grammy. He is a principal at First Sounds, a collaboration of experts dedicated to making the earliest sound recordings available to all people for all time. First Sounds gained international attention last year when it identified and played back sound recordings made in Paris in 1860—17 years before Edison invented the phonograph.

Student/Career Development Event
STUDIO SLAVE TO AUDIO PROFESSIONAL: WORKSHOP ON INTERNSHIPS AND JOBS
Saturday, October 10, 1:00 pm – 2:30 pm
Room 1E06

Presenters: **David Bowles**, Swineshead Productions LLC
Gary Gottlieb, Webster University, Webster Groves, MO, USA
John Krivit, New England Institute of Art, Brookline, MA, USA

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

Saturday, October 10 1:00 pm Room 1E05

Technical Committee Meeting on Studio Practices and Production

Exhibitor Seminar Saturday, October 10
1:00 pm – 2:00 pm Room 1E17

SENNHEISER ELECTRONIC CORP. USA

Digital Microphones

Presenters: **Wolfgang Fraissinet**
Jonathan Freed
Stephan Peus
Gregor Zielinsky

Join Neumann and Sennheiser as they present this panel discussion designed to show you just how user-friendly digital microphones can be. From ease of use—to unparalleled quality—to cost saving opportunities—learn why digital microphones are the solution to today's challenging market.

Saturday, October 10 1:30 pm Room 1E02

Standards Committee Meeting SC-05-02, Audio Connectors

Broadcast/Media Streaming Session 7
Saturday, October 10 2:00 pm – 3:30 pm
Room 1E08

AUDIO FOR NEWS GATHERING

Chair: **Skip Pizzi**, Media Technology Consultant & Contributing Editor, Radio World

Panelists: *Andrew Butterworth*, Connectivity Specialist, BBC News
Robert Duncan, Foreign Desk Operations, NPR
Mike Noseworthy, Senior Audio Engineer, NBC
Network News Field Operations
Chris Tobin, Broadcast Technologist, CBS Radio, New York, NY, USA

Broadcast journalists inhabit a small, unique, and sometimes dangerous corner of the audio world. The technologies used in this space for audio field recording, and for the transport of live and recorded audio from remote sites to broadcast production centers, have expanded greatly in recent years. This ever-challenging environment is now served by a variety of new options, while a few well-used methods are being phased out. From EV-DO to BGAN, AMR-WB to HE-AAC, CF to SDHC, this session will get beyond the acronyms and explore current and emerging components of audio field recording and "backhaul," along with the applications for which each is best suited—all presented by highly experienced professionals from around the industry.

Live Sound Seminar 5 Saturday, October 10
2:00 pm – 2:45 pm Room 1E09

WHITE SPACE AND TVBD UPDATE

Moderator: **Henry Cohen**, Production Radio Rentals, White Plains, NY, USA

Panelists: *Mark Brunner*, Shure Incorporated, Niles, IL, USA
Joe Ciaudelli, Sennheiser Electronic Corporation, Old Lyme, CT, USA
Edgar Reihl, Shure Incorporated, Niles, IL, USA

The DTV conversion is mostly complete; the impact of this and related FCC decisions is of great concern to wireless microphone users. Exactly what is the status of the 700 MHz spectrum and equipment? What types of interference will the proposed Television Band Devices likely create, and what are the proposed methods of mitigation? Will there be an FCC crack-down on unlicensed microphone use? This panel will discuss the latest FCC rule decisions and decisions still pending.

Games Audio 3 Saturday, October 10
2:00 pm – 3:00 pm Room 1E11

THE ART AND BUSINESS OF GAME MUSIC

Chair: **Tom Salta**, Persist Music, Norwalk, CT, USA

Panelist: *Paul Lipson*

Composer Tom Salta, who has spoken internationally on the subject of video game music and is known for his engaging and informative presentations, along with a panel of distinguished game audio composers, will discuss a myriad of creative and business issues facing the modern composer for video games. Tom is the regional director for the Game Audio Network Guild. G.A.N.G. is the largest organization in the world for game audio professionals, with members in over thirty countries. Topics to include the differences between composing music for games vs. film, TV, and other "linear" media, technical parameters of scoring games, types of Implementation (layered, triggered, etc.), and live demonstration of adap- ➥

tive music using F-MOD. Effective techniques for pitching to clients, layering live and fake orchestras, working with reference music, etc.

Student/Career Development Event
EDUCATION FAIR
Saturday, October 10, 2:00 pm – 4:00 pm
Foyer 1E

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.

Saturday, October 10 2:00 pm Room 1E05

Technical Committee Meeting on Microphones and Applications

Exhibitor Seminar Saturday, October 10
2:15 pm – 3:15 pm Room 1E17

THAT CORPORATION

More Analog Secrets Your Mother Never Told You

Presenters: **Rosalfonso Bortoni**
Fred Floru
Gary Hebert
Wayne Kirkwood
Bob Moses
Les Tyler

In 2007, and again in 2008, THAT Corporation revealed a variety of important analog design techniques in its popular "Analog Secrets Your Mother Never Told You" AES seminar. This year we expose more design secrets including "extreme performance" microphone preamps, protecting audio inputs/outputs during phantom power faults, and maintaining high dynamic range in low cost applications.

Session P10 Saturday, October 10
2:30 pm – 5:00 pm Room 1E07

CONSUMER AUDIO

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

2:30 pm

P10-1 The Wii Remote as a Musical Instrument: Technology and Case Studies—*Paul D. Lehrman*, Tufts University, Medford, MA, USA

The inexpensive and ubiquitous remote for the Nintendo Wii game system uses a combination of technologies that are highly suited for music generation and control. These include position tracking, tilt and motion measurement in three dimensions, a two-dimensional joystick (with its companion "Nunchuk"), and multiple buttons. A new accessory, the "MotionPlus," adds gyroscopic sensing and another, the "Balance Board" adds body-position sensing. Use of the system in several musical performance contexts is examined including conducting a synthetic orchestra and playing expressive single and multi-user instruments.
Convention Paper 7888

3:00 pm

P10-2 Measurement Techniques for Evaluating Microphone Performance in Windy Environments—*Simon Busbridge*,[1,2] *David Herman*[2]
[1]University of Brighton, Brighton, UK
[2]AudioGravity Ltd., Brighton, UK

The traditional solution for controlling microphone wind response (foam windshield) is of limited benefit in miniature applications. The use of ECM and MEMS type microphones is typically associated with DSP-type solutions to reduce the unwanted output from air mass flow. Such solutions vary widely in their effectiveness. The situation is compounded by the range of techniques in current use to evaluate microphone wind response. This paper discusses the essential elements necessary for consistent microphone wind measurements and proposes a standard measurement technique that will be of use to all developers and manufacturers concerned with controlling microphone wind noise. Practical implementation of the technique and results obtained for a range of microphones are presented.
Convention Paper 7889

3:30 pm

P10-3 Psychoacoustical Bandwidth Extension of Lower Frequencies—*Judith Liebetrau,*[1] *Daniel Beer*,[1] *Matthias Lubkowitz*[2]
[1]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[2]TU Ilmenau, Ilmenau, Germany

Nowadays, small and flat loudspeakers are requested by the market for home and mobile entertainment. One major problem of such small devices is the reproduction of lower frequencies due to physical limitations. A lack of low frequencies has a negative impact on the perceived audio quality. To obtain good audio quality even with small loudspeakers, the utilization of psychoacoustical effects is conceivable. Basic principles of psychoacoustical bandwidth extension, concretely implemented algorithms and parameter settings for a considerable extension of bandwidth are explained. Furthermore, a listening test method for evaluating perceived audio quality in comparison to bass extension is described. Based on that, an assessment of increased bandwidth extension and sound coloration is done and a conclusion is drawn.
Convention Paper 7890

4:00 pm

P10-4 Reducing the Complexity of Sub-band ADPCM Coding to Enable High-Quality Audio Streaming from Mobile Devices—*Neil Smyth, David Trainor*, APTX (APT Licensing Ltd.), Belfast, Northern Ireland, UK

The number of consumer audio applications demanding high quality audio compression and communication across wireless networks continues to grow. Although the consumer is increasingly demanding higher audio quality, devices such as portable media players and wireless headsets also demand low computational com-

plexity, low power dissipation, and practical transmission bit-rates to help conserve battery life. This paper discusses research undertaken to lower the complexity of existing high-quality sub-band ADPCM coding schemes to better satisfy these conflicting criteria.
Convention Paper 7891

4:30 pm

P10-5 An Interactive Audio System for Mobiles— *Yohan Lasorsa, Jacques Lemordant*, INRIA, Rhône-Alpes, France

This paper presents an XML format for embedded interactive audio, deriving from well-established formats like iXMF and SMIL. We introduce in this format a new paradigm for audio elements and animations synchronization, using a flexible event-driven system in conjunction with graph description capabilities to replace audio scripting. We have implemented a sound manager for J2ME smartphones and the iPhone. Guidance applications for blind people based on this audio system are being developed.
Convention Paper 7892

Workshop 11 **Saturday, October 10**
2:30 pm – 4:00 pm **Room 1E15**

REDUNDANCY IN AUDIO NETWORKS

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

Panelists: *Kevin Gross*, AVA Networks/QSC Audio Products, USA
Al Walker, KlarkTeknik/Midas, UK
Jérémie Weber, AuviTran, Meylan, France
Varuni Witana, Audinate, Australia

Redundancy within a data network focused on digital audio transport has specific requirements compared to redundancy for usual data networks, and for this reason several practical solutions offer dual network ports. Ideas and solutions from current formats will be presented, detailing requirements of specific use cases, such as digital audio transport for live events or for install. The discussion will also include non-IP audio networks; different topologies, and switchover issues will be presented.

Special Event
GRAMMY RECORDING SOUNDTABLE
Saturday, October 10, 2:30 pm – 4:00 pm
Room 1E12/13

Moderator: *Nile Rodgers*

Panelists: *Chuck Ainlay*
Chris Lord-Alge
Serban Ghenea
Tony Maserati

MWA / Mixing with Attitude

It's the eternal challenge: How to create mixes that stand out from the pack, have power and personality, please the artist, producer, and label, sound great at home, in the car, on the radio—and—(oh no!) on MP3. Translating excitement from a recording to its final mix is a high pressure art form, and our panelists are constantly working at the top of their game. Join us for opinions, tips, tricks, and anecdotes as they share their secrets for success.

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

Live Sound Seminar 6 **Saturday, October 10**
3:00 pm – 4:45 pm **Room 1E09**

PRACTICAL ADVICE FOR WIRELESS SYSTEM USERS

Moderator: **James Stoffo**

Panelists: *Joe Ciaudelli*, Sennheiser Electronic Corporation, Old Lyme, CT, USA
Doug Totel, Shure Incorporated, Niles, IL, USA
Karl Winkler, Lectrosonics, Rio Rancho, NM, USA

From houses of worship to wedding bands to community theaters, there are small to medium-sized wireless microphone systems and IEMs in use by the hundreds of thousands. Unlike the Super Bowl or the Grammys, these smaller systems often do not have dedicated technicians, sophisticated frequency coordination, or in many cases even the proper basic attention to system setup. This panel will begin with a basic discussion of the elements of properly choosing components, designing systems, and setting them up in order to minimize the potential for interference while maximizing performance. Topics covered will include antenna selection and placement, coax, spectrum scanning, frequency coordination, gain structure, system monitoring, and simple testing/troubleshooting procedures. Briefly covered will also be planning for upcoming RF spectrum changes.

This session will be preceded by an update on the current state of RF spectrum and TVBDs.

Games Audio 4 **Saturday, October 10**
3:00 pm – 4:00 pm **Room 1E11**

PORTABLE SURROUND SOUND BY HEADPHONES —MAIN RESEARCH AND SUITABLE PRACTICE

Chair: **Florian Koenig**, Ultrasone/FKE

Panelists: *Juha Backan*, Nokia, Helsinki, Finland
Steffan Diedrichsen, Apple
Robert Schulein

Consumer exposure to surround sound systems and the increasing availability or complex music, games, and entertainment products are ushering in a new generation spatializing advanced head-related reproduction of sound. The complete process of 3-D sound recording and reproduction of sound needs to be coordinated between developers and (game) creators of this equipment. This workshop will explain a variety of established methods in 2.0 binaural or more channel surround sound implications for consumer headphones and their interfaces. Synthesized dynamic binaural pannings or experiences with surround processing in mobiles will also be discussed. Further key points are: 3-D production techniques, manual binaural user control, perspectives in head tracking—just for realistic front/back auditory events with variable distance perceptions or ALL individuals?

Saturday, October 10 3:00 pm Room 1E05
Technical Committee Meeting on Audio Forensics

Session P11 Saturday, October 10
3:30 pm – 5:00 pm Foyer 1E

POSTERS: SOUND IN REAL SPACES

3:30 pm

P11-1 Acoustics of National Parks and Historic Sites: The 8,760 Hour MP3 File—*Robert Maher*, Montana State University, Bozeman, MT, USA

According to current U.S. National Park Service (NPS) management policies, the natural soundscape of parks and historic sites is a protected resource just like the ecosystems, landscapes, and historic artifacts for which the parks were formed. While several NPS sites have been studied extensively for noise intrusions by tour aircraft and mechanized recreation, most parks and historic sites do not yet have an acoustic baseline for management purposes. A recent initiative of the NPS Natural Sounds Office is to obtain continuous audio recordings of specific sites for one entire year. This paper explores the engineering and scientific issues associated with obtaining, archiving, and cataloging an 8,760 hour long audio recording for Grant-Kohrs Ranch National Historic Site.
Convention Paper 7893

3:30 pm

P11-2 Improved Speech Dereverberation Method Using the Kurtosis-Maximization with the Voiced/Unvoiced/Silence Classification—
Jae-Woong Jeong,[1] *Se-Woon Jeon*,[1] *Young-Cheol Park*,[2] *Seok-Pil Lee*,[3] *Dae-Hee Youn*[1]
[1]Yonsei University, Seoul, Korea
[2]Yonsei University, Wonju, Korea
[3]Korea Electronics Technology Institute (KETI), Sungnam, Korea;

In this paper we present a new speech dereverberation method using the kurtosis-maximization based on the voiced/unvoiced/silence (V/UV/S) classification. Since kurtosis of the UV/S sections are much smaller than V sections, adaptation of the dereverberation filter using these sections often results in slow and nonrobust convergence, and, in turn, poor dereverberation. The proposed algorithm controls adaptation of the dereverberation filter using the results of V/UV/S classification, together with kurtosis measure of the input speech. For the selective control of adaptation, both hard decision and voice likelihood measure based on various features together with kurtosis were tried, and then, the step-size of the adaptive algorithm was varied according to various control strategies. The proposed algorithm provides better and more robust dereverberation performance than the conventional algorithm, which was confirmed through the experiments.
Convention Paper 7894

3:30 pm

P11-3 ASPARAGUS: Autonomous Robotic Explorer for Acoustic Measurement of Classrooms and Seminar Halls—*Suthikshn Kumar*, PESIT, Bangalore, India

It is important to make measurement of acoustic parameters such as sound intensity, reverberation time, background noise level for classrooms and seminar halls. This helps the designer in improving the acoustics of the classroom and seminar halls for speech intelligibility. In this paper we report on the autonomous robotic explorer ASPARAGUS that has been applied for classroom and seminar hall acoustic parameters measurement. We discuss on the design of the robotic explorer and show how the measurements can be accurately carried out.
Convention Paper 7895
[Poster not presented but available for purchase]

3:30 pm

P11-4 A Survey of Broadcast Television Perceived Relative Audio Levels—*Chris Hanna, Matthew Easley*, THAT Corporation, Milford, MA, USA

Perceived television volume levels can vary dramatically as audio changes both within a given broadcast channel and between broadcast channels. This paper surveys the broadcast audio levels in two large metropolitan areas (Atlanta and Boston). Both analog and digital broadcasts are monitored from cable and satellite providers. Two-channel perceived loudness is measured utilizing the ITU-R Rec. BS.1770 loudness meter standard. Statistical data is presented showing the severity and nature of the perceived loudness changes. Finally, dynamic volume control technology is applied to the most severe recordings for perceived loudness comparisons.
Convention Paper 7896

3:30 pm

P11-5 Optimizing the Re-enforcement Effect of Early Reflections on Aspects of Live Musical Performance Using the Image Source Model—*Michael Terrell, Joshua Reiss*, Queen Mary, University of London, London, UK

The image source method is used to identify early reflections which have a re-enforcement effect on the sound traveling within an enclosure. The distribution of absorptive material within the enclosure is optimized to produce the desired re-enforcement effect. This is applied to a monitor mix and a feedback prevention case study. In the former it is shown that the acoustic path gain of the vocals can be increased relative to the acoustic path gain of the other instruments. In the latter it is shown that the acoustic path from loudspeaker to microphone can be manipulated to increase the perceived signal level before the onset of acoustic feedback.
Convention Paper 7897

3:30 pm

P11-6 The Influence of the Rendering Architecture on the Subjective Performance of Blind

Source Separation Algorithms—*Thorsten Kastner*, University of Erlangen-Nuremberg, Erlangen, Germany, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Blind Source Separation algorithms often include a time/frequency (t/f) decomposition / filterbank as an important part allowing for frequency selective separation of the input signal. In order to investigate the importance of the t/f processing architecture for the achieved subjective audio quality, a set of blindly separated audio signals were taken from the Stereo Audio Source Separation Campaign (SASSEC) 2007 and rated in a MUSHRA listening test. The test furthermore included material that was generated by using the separated signals to drive an enhanced time/frequency rendering architecture, as it is offered by MPEG Spatial Audio Object Coding (SAOC). In this way, the same basic separation core algorithm was applied together with different t/f rendering architectures. The listening test reveals an improved subjective quality for the SAOC post-processed items.
Convention Paper 7898

3:30 pm

P11-7 Real-Time Implementation of Robust PEM-AFROW-Based Solutions for Acoustic Feedback Control—*Simone Cifani, Rudy Rotili, Emanuele Principi, Stefano Squartini, Francesco Piazza*, Università Politecnica delle Marche, Ancona, Italy

Acoustic feedback is a longstanding problem in the audio processing field, occurring whenever sound is captured and reproduced in the same environment. Different control strategies have been proposed over the years, among which a feedback cancellation technique based on the prediction error method (PEM) has revealed to be performing on purpose. Recent studies have shown that the integration of a suppression or a noise reduction filter in the system loop might be beneficial from different perspectives. In this paper a real-time implementation of the aforementioned algorithm is presented, which exploits the partitioned-block frequency-domain (PBFD) technique to allow the system to work also with long acoustic paths. NU-Tech software platform has been used on purpose for real-time simulations, performed over synthetic and real acoustic conditions.
Convention Paper 7899

3:30 pm

P11-8 Perception-Based Audio Signal Mixing in Automotive Environments—*Wolfgang Hess*, Harman/Becker Automotive Systems, Karlsbad-Ittersbach, Germany

Information and announcement presentation in noisy environments such as vehicles requires dynamic adjustment of signals for optimal information audibility and speech intelligibility. Not only variant ambient noises, but, in particular, their combination with today's vehicle infotainment systems capability to reproduce a variety of entertainment signal sources, make information presentation difficult. Most different input level ranges as well as a variety of compressions ratios of audio signals have to be considered. A further challenge is the dynamic, loudness-dependent binaural intelligibility level difference of the human auditory system. The algorithm presented in this paper solves these issues described here by dynamically mixing information and announcement signals to entertainment signals. Entertainment signals are attenuated as little as possible, and information or announcement signals are added in loudness as demanded. As a result, optimal announcement intelligibility and information audibility is achieved.
Convention Paper 7900

3:30 pm

P11-9 Visualization and Analysis Tools for Low Frequency Propagation in a Generalized 3-D Acoustic Space—*Adam J. Hill, Malcolm O. J. Hawksford*, University of Essex, Colchester, Essex, UK

A toolbox is described that enables 3-D animated visualization and analysis of low-frequency wave propagation within a generalized acoustic environment. The core computation exploits a Finite-Difference Time-Domain (FDTD) algorithm selected because of its known low frequency accuracy. Multiple sources can be configured and analyses performed at user-selected measurement locations. Arbitrary excitation sequences enable virtual measurements embracing both time-domain and spatio-frequency domain analysis. Examples are presented for a variety of low-frequency loudspeaker placements and room geometries to illustrate the versatility of the toolbox as an acoustics design aid.
Convention Paper 7901

Workshop 12 **Saturday, October 10**
3:30 pm – 4:30 pm **Room 1E08**

TEACHING ELECTRONICS TO AUDIO RECORDING STUDENTS, WHY BOTHER?

Chair: **Michael Stucker**, Indiana University, IN, USA

Panelists: *Eric Brengle*, Swinghouse Studios, CA, USA
Eddie Ciletti, Tangible Technology, MN, USA
Dale Manquen, MANCO, CA, USA
Walter Sear, Sear Sound, NY, USA
Bill Whitlock, Jensen Transformers, CA, USA

There is little doubt that recording engineers can benefit from a knowledge of electronics, so it makes sense to include it in an audio recording curriculum. However, with recording technique, music theory, acoustics, listening, and computer technology demanding more and more course time, how can we integrate electronics into an audio recording curriculum yet ensure that it remains pertinent and applicable to the audio industry of today?

This workshop aims to get feedback from professionals about what level of electronics knowledge they would like from graduates entering the field and encourage an enhanced dialog between the recording and education communities. The workshop will identify goals and outcomes for a practical electronics curriculum, determine an ideal skill set for students that are entering the 21st century marketplace, and investigate an instruction methodology to achieve those ends. The panel will consist of professionals and educators and will explore issues such as teaching materials, lecture vs. lab-based instruction, project based learning, a core knowledge base, and methods for building a continuing dialog between professionals and educators.

Saturday, October 10 3:30 pm Room 1E02

Standards Committee Meeting SC-03-02, Transfer Technologies

Exhibitor Seminar Saturday, October 10
3:30 pm – 4:30 pm Room 1E17

ACO PACIFIC, INC.

The SLARMSolution™ and What Does Noise Sound Like? Noise and Health Issues

Presenters: **Les Blomberg,** Noise Pollution Clearinghouse
Noland Lewis, ACO Pacific, Inc.
Virgil Terrell, ACO Pacific - Midwest

What Does Noise Sound Like? How and how not to create angry neighbors. Understand, anticipate, and reduce the impact of noise on neighbors and communities. The SLARMSolution™—a discussion of the SLARM™ and NetSlarm™ systems as an innovative approach to compliance, enforcement, and the control of community, industrial, and entertainment noise issues. ACOustics Begins With ACO™

Student/Career Development Event
STEREO RECORDING COMPETITION—1
Saturday, October 10, 4:00 pm – 7:00 pm
Room 1E11

The Student Recording Competition is a highlight at each convention. A distinguished panel of judges participates in critiquing finalists of each category in an interactive presentation and discussion. Student members can submit stereo recordings in these categories:

- **Classical** 4:00 to 5:00
  Judges: *Martha DeFrancisco, Richard King, Leszek Wojcik*
- **Jazz/Blues** 5:00 to 6:00
  Judges: *Jim Anderson, Michael Bishop, Dave Hewitt*
- **Folk/World** 6:00 to 7:00
  Judges: *Darcy Proper, Mark Rubel*

The top three finalists in each category present a short summary of their techniques and intentions, and play their project for all who attend. Meritorious awards will be presented later at the closing Student Delegate Assembly Meeting (SDA-2) on Monday Afternoon.

It's a great chance to hear the work of your colleagues from other educational institutions. Everyone learns from the judges' comments even if your project isn't one of the finalists. You'll meet other students and faculty. This year, the event is followed immediately by the Student Social and Student Dinner, so don't miss it.

Sponsors include AEA, Genelec, JBL, Lavry Engineering, Schoeps, Shure, and Ron Streicher.

Saturday, October 10 4:00 pm Room 1E05

Technical Committee Meeting on Coding of Audio Signals

Workshop 13 Saturday, October 10
4:30 pm – 6:00 pm Room 1E15

1080p AND MP3: WE GOT THE PICTURE. WHAT HAPPENED TO THE SOUND?

Chair: **Michael Fremer**

Panelists: *John Atkinson*, Stereophile
Steve Berkowitz, Sony/BMG
Greg Calbi, Sterling Sound
Alan Douches, West West Side Music, New Windsor, NY, USA
Randy Ezratty
Bob Ludwig, Gateway Mastering & DVD, Portland, ME, USA
EveAnna Manley, Manley Audio Labs

Over the past decade, video has gone from 480i 4:3 to 1080p 16:9—the bigger the screen, the better. Audio? Not so much! The bigger the audio system, the more you are mocked for hearing quality that supposedly doesn't exist. The "mainstream" audio standard has degraded from 44.1 K/16 bit (arguably insufficient to begin with) to low bit rate MP3, now considered "more than adequate" by mainstream listeners. As a consequence, people "hear" music while engaging in other activities. Actively listening to music (to the exclusion of all other activities) once commonplace, is almost unheard of (pun intended) in 21st Century life. This has had terrible consequences for both professional and consumer audio and the engineers working in both areas. Most consumers have never heard their favorite music on a good audio system and because of "big box" retailing can't even experience high quality 5.1 soundtrack sound on a good home theater system when they shop. How does the industry reconnect consumers to good sound? And why are young people snapping up vinyl records when they can download MP3s cheaply or for free?

Saturday, October 10 4:30 pm Room 1E03

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

Exhibitor Seminar Saturday, October 10
4:45 pm – 5:45 pm Room 1E17

RENKUS-HEINZ

Iconyx and IC Live—Bringing Digital Beam Steering to Live Sound

Presenters: **Jonas Domkus**
Ralph Heinz

Featuring an IC Live system for the presentation, we will compare its benefits with traditional solutions and show how it resolves challenging acoustic spaces by enabling you to digitally aim sound, minimizing echoes and improving intelligibility. This seminar will show the theory behind digitally steerable loudspeaker technology and demonstrates the end results.

Tutorial 3 Saturday, October 10
5:00 pm – 6:30 pm Room 1E16

PERCUSSION ACOUSTICS AND QUANTITATIVE DRUM TUNING

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

Intricate tuning of acoustic drums can have a significant influence on the quality and contextuality of the instrument during a recording session. In this tutorial presentation, waveform and spectral analysis will be used to show that quantitative tuning and acoustic benchmarking is a viable possibility. The principle acoustic properties of popular drums will be shown by live demonstration and aspects relating to drum tuning will be highlighted. In particular, demonstration and analysis of the following tuning issues will be covered: Achieving a uniform pitch across the drum head; Tuning drums to a desired musical pitch; Manipulating overtones and generating rich timbres; Controlling attack and decay profiles; Tuning the drum kit as a pitched musical instrument.

Live Sound Seminar 7 Saturday, October 10
5:00 pm – 6:45 pm Room 1E09

THE GREENING OF THE BAND: GREEN SOLUTIONS FOR TOURING FOR THE LIVE ENGINEER

Moderator: **John Krivit**, New England Institute of Art, Brookline, MA, USA

Panelists: *Tom Bensen*, VP & GM, RF Productions of NY
Robert Goldstein, President, Maryland Sound International
Albert Leccese, President, Audio Analysts
David Shadoan, President, Sound Image
Neal Turley, Sustainable Waves

Green solutions for touring have been embraced by the live sound business with artists such as Dave Matthews Band, John Mayer, Guster, Jack Johnson, Bonnie Raitt, Barenaked Ladies, and Sheryl Crow all incorporating practices that lessen the environmental impact that touring can inflict. As these measures drift further into the mainstream of the industry, it becomes essential for all touring professionals to stay informed of new professional standards. Join AES Education Committee Vice Chair John Krivit along with a panel of experts, advocates, and artists to find out how you can reduce the carbon footprint of your tour.

Special Event
TECnology HALL OF FAME
Saturday, October 10, 5:00 pm – 6:30 pm
Room 1E12/13

The Mix Foundation for Excellence in Audio was established in 1991 to recognize and promote excellence in the audio arts and sciences. The 25th Annual Technical Excellence & Creativity Awards, the industry's foremost recognition of achievement in audio production and product design, will be presented at this event. The TEC Awards reception will begin at 5:00 pm in the South Concourse and the ceremony will follow at 6:00 pm in Room 1E12/13. As a charitable event, tickets are required for both the reception and the ceremony and may be purchased prior to the event at Booth 127 during the convention subject to availability.

Historical Event
SIGNIFICANT TECHNICAL CONTRIBUTIONS OF RCA CORP.
Saturday, October 10, 5:00 pm – 7:00 pm
Room 1E08

Moderator: **Cliff Rogers**

Panelists: *Fred Barnum*, L3 Communications
Hans Dietza

From Nipper and "His Masters Voice" to the World Trade Center "Antenna" RCA covered the spectrum … . All from Camden NJ.

Talking Machines, by the millions—led to Radios, by the millions—led to RCA Broadcast Division

The Broadcast Division maintained an unbelievable esprit de corps, among their employees—we were routinely challenged to development of state-of-the-art products that met the challenging needs of this fast growing industry, with unbelievable support. The resultant products were of unquestioned quality; were marketed, sold, and supported by an extremely well organized, well trained, and expertly supported, domestic and international sales and service organization. This resulted in an outstanding industry that to this day has been unmatched. Incidentally, the RCA Broadcast Division was one of the most profitable divisions of the RCA Corporation. It operated from roughly from 1930 until 1985. Many of the former RCA Broadcast employees report that the time that they spent at RCA Broadcast, was the most invigorating time of their lives. They served in many aspects of the broadcast industry including, Engineering, Marketing, Development, Sales, Advertising, Service, and the world famous RCA Princeton Laboratories.

Our opportunity here at AES is to look back, and review some of the greatest activities of the RCA Broadcast Division. The list is endless, however, our emphasis here will be directed toward the RCA Broadcast Audio products including microphones, amplifiers, loudspeakers, theater systems, and RCA photophone activities.

Look at the Industry and how it grew…We all learned…Let's look back…What did we learn…Let's now look ahead…

Saturday, October 10 5:00 pm Room 1E05

Technical Committee Meeting on Audio for Telecommunications

Saturday, October 10 5:00 pm Room 1E02

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

Special Event
ORGAN RECITAL BY GRAHAM BLYTH
Saturday, October 10, 7:30 pm – 9:00 pm
Church of Saint Mary the Virgin
145 West 46th Street, NY

Graham Blyth's traditional organ recital will be given at the Church of Saint Mary the Virgin, founded in 1868.

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

One of the main goals in American Classic organ building was for a satisfactory organ ensemble sound, ➥

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

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

César Franck wrote only 12 major works for organ but despite this small output he is considered by many to be the greatest composer of organ music since J.S. Bach. This may well have a lot to do with Liszt's public declaration that Franck was a worthy successor to the great man having just heard him play his Six Pieces for Organ at a Recital in Paris.

Graham will play the Three Chorales by Franck—in E major, B minor, and A minor—preceeding each with one of the “Great” Preludes & Fugues of J.S.Bach—E minor BWV 548, B minor BWV 544, and A minor BWV 543.

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.

Student/Career Development Event
SOCIAL AND DINNER
Saturday, October 10, 7:30 pm – ???
New York University
35 W. 4th St., 6th Floor

NYU and AES-SDU welcomes all students to the NYU Music Technology department at NYU. The student party is an important event where students can begin and continue their networking experience! It will be held at NYU and equipment will be available for demonstrations of student work. Bring your mix to share! Dinner is included. Please buy tickets at the student booth or contact meiling@aes-sda.org

Session P12 **Sunday, October 11**
9:00 am – 1:00 pm **Room 1E07**

TRANSDUCERS MANUFACTURING AND EQUIPMENT

Chair: **Alexander Voishvillo**

9:00 am

P12-1 Time Varying Behavior of the Loudspeaker Suspension: Displacement Level Dependency
—*Finn Agerkvist*,[1] *Bo Rhode Petersen*[2]
[1]Technical University of Denmark, Lyngby, Denmark
[2]Esbjerg Institute of Technology, Aalborg University, Aalborg, Denmark

The compliance of the loudspeaker suspension is known to depend on the recent excitation level history. Previous investigations have shown that the electrical power as well as displacement and velocity plays a role. In this paper the hypothesis that the changes in compliance are caused mainly by how much the suspension has been stretched, i.e., the maximum displacement, is investigated. For this purpose the changes in compliance are measured when exposing the loudspeaker to different levels and types of electrical excitation signals, as well as mechanical excitation only. For sinusoidal excitation the change in compliance is shown to depend primarily on maximum displacement. But for square pulse excitation the duration of the excitation also plays an important role.
Convention Paper 7902

9:30 am

P12-2 Fast Measurement of Motor and Suspension Nonlinearities in Loudspeaker Manufacturing
—*Wolfgang Klippel*,[1] *Joachim Schlechter*[2]
[1]University of Technology Dresden, Dresden, Germany
[2]KLIPPEL GmbH, Dresden, Germany

Nonlinear distortions are measured at the end of the assembling line to check the loudspeaker system and to make a pass/fail decision. However, the responses of single components and total harmonic distortion have a low diagnostic value because they are difficult to interpret and do not reveal the particular cause of the defect. A new measurement technique is presented that measures the nonlinearities of motor and suspension system directly. The results are single-valued parameters (e.g., voice coil offset in mm), which

are directly related with the geometry and large signal parameters of the loudspeaker system. The measurement is only based on the measurement of the electrical signals at the speaker's terminals giving full robustness against ambient noise. The accuracy of the measurement results is investigated while performing measurements using short stimuli between 0.2 and 1.3 seconds. The paper discusses new possibilities for on-line diagnostic during end-of-line testing and the integration into production control to increase the yield of the production.
Convention Paper 7903

10:00 am

P12-3 A Novel Technique for Detecting and Locating Loudspeaker Defects—*Yi Yang, Junfeng Wei, Haihong Feng, Zhoubin Wen*, Chinese Academy of Sciences, Beijing, China

A novel technique for the measurement of rub and buzz using a fast tracking high pass filter is presented first in this paper. In the tests of 100,000 loudspeaker samples on a production line, the very low missed detection was 0.006% and the false alarm rate was 4.68% with this method compared with human hearing tests. Then a method consisted of detecting loudspeaker defect and estimating loudspeaker displacement without using a laser displacement sensor is launched. Only the current response of the loudspeaker is used to predict loudspeaker displacement. And there is less than 3% error in phase component according to the direct laser measurement. Several experiments proved this method was very effective.
Convention Paper 7904

10:30 am

P12-4 Practical Measurement of Loudspeaker Distortion Using a Simplified Auditory Perceptual Model—*Steve Temme,*[1] *Pascal Brunet*,[1] *D. B. (Don) Keele Jr.*[2]
[1]Listen Inc., Boston, MA, USA
[2]DBK Associates and Labs, Bloomington, IN, USA

Manufacturing defects in loudspeaker production can often be identified by an increase in rub and buzz distortion. This type of distortion is quite noticeable because it contributes an edgy sound to the reproduction and is annoying because it often sounds separate or disembodied from the fundamental signal. The annoyance of rub and buzz distortion is tied intimately to human perception of sound and psychoacoustics. To properly implement automated production-line testing of loudspeaker rub and buzz defects, one has to model or imitate the hearing process using a sufficiently accurate perceptual model. This paper describes the results of a rub and buzz detection system using a simplified perceptual model based on human masking thresholds that yields excellent results.
Convention Paper 7905

11:00 am

P12-5 The Audio Performance Comparison and Effective Error Correction Method of Switching Amplifiers—*Jae Cheol Lee, Haekwang Park, Donghyun Lim, Joonhyun Lee, Yongserk Kim*, Samsung Electronics Co., Ltd.

This paper introduces various open-loop and closed-loop switching amplifiers and then reviews merits and demerits when they are applied to consumer electronic products. Audio specifications in products that open-loop and closed-loop switching amplifiers are adopted to are measured and analyzed as to whether they have weak points. After that, the paper proposes a simple and effective method for the error control. The proposed method has the outstanding audio performance in consumer electronics products.
Convention Paper 7906

11:30 am

P12-6 Investigation of Switching Frequency Variations and EMI Properties in Self-Oscillating Class D Amplifiers—*Dennis Nielsen,*[1] *Arnold Knott*,[1] *Gerhard Pfaffinger*,[2] *Michael Andreas E. Andersen*[1]
[1]Technical University of Denmark, Lyngby, Denmark
[2]Harman/Becker Automotive Systems GmbH, Straubing, Germany

Class D audio amplifiers have gained significant influence in sound reproduction due to their high efficiency. One of the most commonly used control methods in these amplifiers is self-oscillation. A parameter of key interest in self-oscillating amplifiers is the switching frequency, which is known for its variation. Knowledge of switching frequency variations is of great importance with respect to electromagnetic interference (EMI). This paper will investigate whether the switching frequency is depended on modulation index and audio reference frequency. Validation is done using simulations, and the results are compared with measurements performed on a 50 W prototype amplifier. The switching frequency is tracked through accurate spectrum measurements, and very good compliance with simulation results are observed.
Convention Paper 7907

12:00 noon

P12-7 Design Optimizations for a High Performance Handheld Audio Analyzer—*Markus Becker*, NTi Audio AG, Schaan, Liechtenstein

All types of advanced mobile devices share certain design challenges. For example, incorporating a powerful embedded processor system to support comprehensive functionality via a full featured easy-to-use human interface at low power consumption. But designing a multi-function handheld audio analyzer adds further challenges based upon requirements for extremely low noise floor, wide measurement range, compatibility with measuring microphones and other demands, including standards compliance. Additional requirements include the efficient display of complex data onto a restricted size display, and efficient and safe operation in many different locations and environments. These place further design burdens on the user interface and the instrument package, respectively.
Convention Paper 7908

➥

12:30 pm

P12-8 The 48 Volt Phantom Menace Returns— *Rosalfonso Bortoni, Wayne Kirkwood*, THAT Corporation, Milford, MA, USA

Hebert and Thomas presented a paper at the 110th AES Convention [Convention Paper 5335] that described the "phantom menace" phenomenon wherein microphone phantom power faults can damage audio input circuitry. Now, a few years later, this paper brings and provides new information about the phantom menace fault mechanisms, analyzes common protection circuits, and introduces a new protection scheme that is more robust. In addition, new information is presented relating these input protection schemes to audio performance and recommendations are made to minimize noise and distortion.
Convention Paper 7909

Session P13 Sunday, October 11
9:00 am – 1:00 pm Room 1E16

SPATIAL AUDIO

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

9:00 am

P13-1 Microphone Array Optimization for a Hearing Restoration Headset—*Marty Johnson,*[1] *Philip Gillett,*[1] *Efrain Perini,*[1] *Alessandro Toso*,[1] *Daniel Harris*[2]
[1]Virginia Tech, Blacksburg, VA, USA
[2]Sennheiser Research Laboratory, Palo Alto, CA, USA

Subjects wearing communications or hearing protection headsets lose the ability to localize sound accurately. Here we describe a hearing restoration headset designed to restore a user's natural hearing by processing signals from an array of microphones using a filter-and-sum technique and presenting the result to the user via the headset's speakers. The filters are designed using a phase compensation technique for mapping the microphone array manifolds (or directional transfer functions) onto the target HRTFs. To optimize the performance of the system, a 3-D numerical model of a KEMAR mannequin with headset was built and verified experimentally up to 12 kHz. The numerical model was used to optimize a three microphone array that demonstrated low reconstruction error up to 12 kHz.
Convention Paper 7910

9:30 am

P13-2 Optimized Parameter Estimation in Directional Audio Coding Using Nested Microphone Arrays—*Giovanni Del Galdo, Oliver Thiergart, Fabian Kuech, Maja Taseskma, Divya Sishtla V.N.*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Directional Audio Coding (DirAC) is an efficient technique to capture and reproduce spatial sound on the basis of a downmix audio signal, direction of arrival, and diffuseness of sound. In practice, these parameters are determined using arrays of omnidirectional microphones. The main drawback of such configurations is that the estimates are reliable only in a certain frequency range, which depends on the array size. To overcome this problem and cover large bandwidths, we propose concentric arrays of different sizes. We derive optimal joint estimators of the DirAC parameters with respect to the mean squared error. We address the problem of choosing the optimal array sizes for specific applications such as teleconferencing and we verify our findings with measurements.
Convention Paper 7911

10:00 am

P13-3 Modification of HRTF Filters to Reduce Timbral Effects in Binaural Synthesis— *Juha Merimaa*, Sennheiser Research Laboratory, Palo Alto, CA, USA

Using head-related transfer functions (HRTFs) in binaural synthesis often produces undesired timbral coloration. In this paper a method for designing modified HRTF filters with reduced timbral effects is proposed. The method is based on reducing the variation in the root-mean-square spectral sum of a pair of HRTFs while preserving the interaural time difference and interaural level difference. In formal listening tests it is shown that the coloration due to the tested non-individualized HRTFs can be significantly reduced without altering the resulting localization.
Convention Paper 7912

10:30 am

P13-4 An Active Multichannel Downmix Enhancement for Minimizing Spatial and Spectral Distortions—*Jeffrey Thompson, Aaron Warner, Brandon Smith*, DTS, Inc., Agora Hills, CA, USA

With the continuing growth of multichannel audio formats, the issue of downmixing to legacy formats such as stereo or mono remains an important problem. Traditional downmix methods use fixed downmix coefficients and mixing equations to blindly combine N input channels into M output channels, where N is greater than M. This commonly produces unpredictable and unsatisfactory results due to the dependence of these passive methods on input signal characteristics. In this paper an active downmix enhancement employing frequency domain analysis of key inter-channel spatial cues is described that minimizes various distortions commonly observed in downmixed audio such as spatial inaccuracy, timbre change, signal coloration, and reduced intelligibility.
Convention Paper 7913

11:00 am

P13-5 Physical and Perceptual Properties of Focused Virtual Sources in Wave Field Synthesis—*Sascha Spors, Hagen Wierstorf, Matthias Geier, Jens Ahrens*, Deutsche Telekom

Laboratories, Techniche Universität Berlin, Berlin, Germany

Wave field synthesis is a well established high-resolution spatial sound reproduction technique. Its physical basis allows reproduction of almost any desired wave field, even virtual sources that are positioned in the area between the loudspeakers and the listener. These are known as focused sources. A previous paper has revealed that focused sources have a number of remarkable physical properties, especially in the context of spatial sampling. This paper will further investigate on these and other physical artifacts. Additionally, results of perceptual experiments will be discussed in order to offer a conclusion on the perceptual relevance of the derived artifacts in practical implementations.
Convention Paper 7914

11:30 am

P13-6 Localization Curves for a Regularly-Spaced Octagon Loudspeaker Array—*Laurent S. R. Simon, Russell Mason, Francis Rumsey*, University of Surrey, Guildford, Surrey, UK

Multichannel microphone array designs often use the localization curves that have been derived for 2-0 stereophony. Previous studies showed that side and rear perception of phantom image locations require somewhat different curves. This paper describes an experiment conducted to evaluate localization curves using an octagon loudspeaker setup. Interchannel level differences were produced between the loudspeaker pairs forming each of the segments of the loudspeaker array, one at a time, and subjects were asked to evaluate the perceived sound event's direction and its locatedness. The results showed that the localization curves derived for 2-0 stereophony are not directly applicable, and that different localization curves are required for each loudspeaker pair.
Convention Paper 7915

12:00 noon

P13-7 Fixing the Phantom Center: Diffusing Acoustical Crosstalk—*Earl Vickers*, STMicroelectronics, Santa Clara, CA, USA

When two loudspeakers play the same signal, a “phantom center” image is produced between the speakers. However, this image differs from one produced by a real center speaker. In particular, acoustical crosstalk produces a comb-filtering effect, with cancellations that may be in the frequency range needed for the intelligibility of speech. We present a method for using phase decorrelation to fill in these gaps and produce a flatter magnitude response, reducing coloration and potentially enhancing dialog clarity. This method also improves headphone compatibility and reduces the tendency of the phantom image to move toward the nearest loudspeaker.
Convention Paper 7916

12:30 pm

P13-8 Frequency-Domain Two- to Three-Channel Upmix for Center Channel Derivation and Speech Enhancement—*Earl Vickers*, STMicroelectronics, Santa Clara, CA, USA

Two- to three-channel audio upmix can be useful in a number of contexts. Adding a front center loudspeaker provides a more stable center image and an increase in dialog clarity. Even in the absence of a physical center loudspeaker, the ability to derive a center channel can facilitate speech enhancement by making it possible to boost or filter the dialog, which is usually panned to the center. Two- to three-channel upmix can also be a first step in upmixing from two to five channels. We propose a frequency-domain upmix process using a vector-based signal decomposition, including methods for improving the selectivity of the center channel extraction. A geometric interpretation of the algorithm is provided. Unlike most existing frequency-domain upmix methods, the current algorithm does not perform an explicit primary/ambient decomposition. This reduces the complexity and improves the quality of the center channel derivation.
Convention Paper 7917

Broadcast/Media Streaming Session 8
Sunday, October 11 9:00 am – 10:30 am
Room 1E08

LIP SYNC ISSUE

Chair: **Jonathan Abrams**, Nutmeg Post

Panelists: *Aldo Cugnini*, AGC Systems LLC
Graham Jones, National Association of Broadcasters
Steve Lyman, Dolby Laboratories

Lip sync remains a complex problem, with several causes and few solutions. From production, through transmission, and ultimately 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.

What is the perspective of the NAB and SMPTE regarding lip sync? Where do the latency issues exist? What are the recommendations from the ITU and ATSC? What correction techniques exist? How does video display design affect lip sync? Are there mechanisms for ensuring lip sync is maintained when the signal reaches your television? Join us as our panel addresses these questions and more.

Games Audio 5 Sunday, October 11
9:00 am – 10:30 am Room 1E09

SOUNDS FROM THE THIRD DIMENSION! 3-D SIMULATION TECHNIQUES IN GAME DEVELOPMENT

Presenter: **Scott Selfon**, Microsoft Corporation, Redmond, WA, USA

High fidelity, non-repetitive sound effects, and ambience are only the beginning of a compelling in-game sound experience. Titles are increasingly exploring sophisticated real-time sound manipulation to enhance both realism and immersion. This talk will explore these aspects as they relate to the “virtual world” of the game's universe—dynamic simulation of position, distance, interaction with ➥

game geometry, and environmental reverberation techniques will be discussed. This talk will additionally cover simulation challenges and aesthetic considerations particular to interactive gameplay.

Historical Event
HISTORY OF BELL LABS
Sunday, October 11, 9:00 am – 11:30 am
Room 1E15

Moderator: **Noah Simon**

Presenter Noah Simon will take attendees back to 1915 when nascent behemoth AT&T initiated trans-continental telephone service. This innovation spawned a research division that evolved into Bell Labs. The bustling scientific playground was responsible for a torrent of audio innovations including the condenser microphone, moving coil loudspeakers and microphones, and significant contributions to magnetic recording, sound for film, high-quality disc reproduction, and stereo recording. The presentation will employ recordings, photos, and other media to illustrate a vivid timeline of one of America's most innovative companies.

Student/Career Development Event
SURROUND RECORDING COMPETITION
Sunday, October 11, 9:00 am – 12:00 noon
Room 1E11

The Student Recording Competition is a highlight at each convention. A distinguished panel of judges participates in critiquing finalists of each category in an interactive presentation and discussion. Student members can submit surround recordings in these categories:

- **Classical** 9:00 to 10:00
  Judges: *Martha DeFrancisco, Richard King*
- **Non-Classical** 10:00 to 11:00
  Judges: *Robert Friedrich, Richard King, Ronald Prent*
- **Sound for Picture** 11:00 to 12:00
  Judges: *Michael Barry, Lawrence Manchester, Ira Newborn, Ron Sadoff, Tim Starnes*

The top three finalists in each category present a short summary of their techniques and intentions, and play their project for all who attend. Meritorious awards will be presented later at the closing Student Delegate Assembly Meeting (SDA-2) on Monday Afternoon.

It's a great chance to hear the work of your colleagues from other educational institutions. Everyone learns from the judges' comments even if your project isn't one of the finalists. You'll meet other students and faculty. Don't miss it.

Sponsors include AEA, Genelec, JBL, Lavry Engineering, Schoeps, Shure, and Ron Streicher.

Sunday, October 11 9:00 am Room 1E05

Technical Committee Meeting on Network Audio Systems

Sunday, October 11 9:00 am Room 1E03

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

Session P14 Sunday, October 11
10:00 am – 11:30 am Foyer 1E

POSTERS: SIGNAL PROCESSING

10:00 am

P14-1 A New Distance Measurement Method for Distance-Based Howling Canceller—*Akira Sogami, Arata Kawamura; Youji Iiguni*, Osaka University, Toyonaka, Osaka, Japan

In this paper we propose a new distance measurement method for a distance-based howling canceller. We have previously proposed a howling canceller that uses only distance information between the loudspeaker and the microphone. The howling canceller suppresses howling based on the distance measured by a sonic wave. The conventional measurement method however has a noise while on distance measurement. To solve the problem we propose a new distance measurement method that suppresses the noise. Simulation results in a practical environment show that the proposed distance measurement method can almost exactly estimate the distance more silently than the conventional method.
Convention Paper 7918

10:00 am

P14-2 New Technology for Hearing Stimulation Employing the SPS-S Method—*Andrzej Czyzewski*,[1] *Henryk Skarzynski*,[2] *Bozena Kostek*,[1,2] *Piotr Odya,*[1] *Piotr Suchomski*,[1] *Piotr Skarzynski*[3]
[1]Gdansk University of Technology, Gdansk, Poland
[2]Institute of Physiology and Pathology of Hearing, Warsaw, Poland
[3]Sense Organs Institute, Nadarzyn, Poland

A prototype of a the new Compact Audio Therapy Unit (CATU) is presented that can process any audio signal inside a very compact device working in real time, employing advanced digital filtration, signal keying, manipulating playback rate, various spectral modifications of the signal, repeating phrases, and others. It was designed to provide a platform for the therapy with the new Method of the Aural Perception Stimulation (SPS-S). The design for wearability allows one to use the device effectively in normal everyday life conditions, e.g., outdoors. The compact and versatile processing device can potentially open a new era in patients and trainees mobility.
Convention Paper 7919

10:00 am

P14-3 Frequency Characteristics Measurements of 78 rpm Acoustic Record Players by the Pulse-Train Method—*Teruo Muraoka, Takahiro Miura, Tohru Ifukube*, University of Tokyo, Tokyo, Japan

The authors have been engaged in the research for the restoration of seriously damaged audio signals, employing Generalized Harmonic Analysis (GHA). In this research, it is important to know frequency characteristics of sound reproducing equipments to realize proper sound

reproduction. However, frequency characteristics of the ancient acoustic record players such as "Credenza," etc., are significant but not clear: especially, the frequency characteristics when records are actually reproduced. However it can solely be measured with frequency record and vibrator-method is not used any more. In fact, no shellac-made 78 rpm frequency record can be manufactured today: the traditional measurement techniques are inapplicable. On the other hand, one of the authors previously developed Pulse-Train measurement for phonograph cartridges, in order to obtain their frequency characteristics of amplitude and phase performances. This method is applicable so long as any pulse waveform is curved on record surface. Thus the authors employed this method. Radial directional groove was curved on a surface of shellac disc-record, and Pulse-Train response is obtained by reproducing the record with an acoustic record player. Some examples will be exhibited in the report.
Convention Paper 7920

10:00 am

P14-4 MDCT for Encoding Residual Signals in Frequency Domain Linear Prediction— *Sriram Ganapathy*,[1] *Petr Motlicek*,[2] *Hynek Hermansky*[1,2]
[1]Johns Hopkins University, Baltimore, MD, USA
[2]Idiap Research Institute, Martigny, Switzerland

Frequency domain linear prediction (FDLP) uses autoregressive models to represent Hilbert envelopes of relatively long segments of speech/audio signals. Although the basic FDLP audio codec achieves good quality of the reconstructed signal at high bit-rates, there is a need for scaling to lower bit-rates without degrading the reconstruction quality. Here, we present a method for improving the compression efficiency of the FDLP codec by the application of the modified discrete cosine transform (MDCT) for encoding the FDLP residual signals. In the subjective and objective quality evaluations, the proposed FDLP codec provides competent quality of reconstructed signal compared to the state-of-the-art audio codecs for the 32 – 64 kbps range.
Convention Paper 7921

10:00 am

P14-5 State-Space Biquad Filters with Low Noise and Improved Efficiency for Pipelined DSPs—*David McGrath*, Dolby Laboratories, Sydney, NSW, Australia

A State-Space filter structure is presented, along with simplified equations for mapping the coefficients of arbitrary biquad filter coefficients to the State-Space structure. This procedure allows low noise implementation of an arbitrary second-order filter transfer function. A block-processing variant of the State-Space structure is described, with the added benefit that greater efficiency can be achieved on some classes of modern pipelined DSP processors.
Convention Paper 7922

10:00 am

P14-6 A Bandlimited Oscillator by Frequency-Domain Synthesis for Virtual Analog Applications—*Glen Deslauriers, Colby Leider*, University of Miami, Coral Gables, FL, USA

Problems posed by the bandlimited synthesis of audio signals have long been addressed by the music and audio engineering communities. However, few of the proposed solutions have the flexibility necessary to accurately model and produce the variety of waveform functions present in an analog oscillator. Preferably, an additive technique would be employed as the ideal method of alias-free synthesis. Inverse Fourier Transform synthesis is one method that is often discussed but less-frequently utilized. Here we propose modifications to the method and implementation of Inverse Fourier Transform synthesis as a viable basis for the creation of a software oscillator for a Virtual Analog instrument. Design results show the quality to outperform a variety of currently implemented methods.
Convention Paper 7923

10:00 am

P14-7 Digital Simulation of Phonograph Tracking Distortion—*Richard Tollerton*, Isomorphic Software, Inc., San Francisco, CA, USA

Phonograph tracking distortion results from the misalignment of a playback cartridge with respect to the cutting head. While it has been researched for decades, it remains a source of mystery: it cannot be isolated, it has not been accurately simulated, and its importance remains undecided. Here, a PCM simulation of horizontal and vertical tracking distortion of extremely high quality is presented, operating on the principle of phase modulation, allowing tracking distortion to be evaluated in isolation with real musical content. In this context, tracking distortion is equivalent to digital audio sampling jitter, with the jitter spectrum equal to the signal spectrum. Implications of this connection, as well as simulation accuracy, preliminary listening test results, and potential applications are discussed.
Convention Paper 7924

Sunday, October 11 10:00 am Room 1E05

Technical Committee Meeting on Electro Magnetic Compatibility

Broadcast/Media Streaming Session 9
Sunday, October 11 11:00 am – 1:00 pm
Room 1E08

LISTENER FATIGUE AND LONGEVITY

Chair: **David Wilson**, CEA

Presenters: *Sam Berkow*, SIA Acoustics
Marvin Caesar, Aphex
Tim Carroll, Linear Acoustic
James D. (JJ) Johnston, DTS Inc.
Hannes Müsch, Dolby
Ted Ruscitti
Ellyn Sheffield

This panel will discuss listener fatigue and its impact on listener retention. While listener fatigue is an issue of interest to broadcasters, it is also an issue of interest to ➥

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

Live Sound Seminar 8 **Sunday, October 11**
11:00 am – 12:45 pm **Room 1E09**

10 THINGS TO GET RIGHT WITH SOUND REINFORCEMENT SYSTEMS

Moderator: **Peter Mapp**, Peter Mapp Associates, Colchester, Essex, UK

Panelists: *Mark Engerbretson*, QSC Audio Products, Costa Mesa, CA, USA
Chris Foreman, Community Professional Loudspeakers, Chester, PA, USA
Kurt Graffy, Arup Acoustics

This Live Sound Event will discuss the ten most important things to get right when designing/operating sound reinforcement and PA systems. However, as attendees at the event will learn, there are many more things to consider than just these ten golden rules, and that the order of importance of these often change depending upon the venue and type of system. We aim to provide a practical approach to sound system design and operation illustrated with many practical examples and case histories. Each panelist has many years of practical experience and between them can cover just about any aspect of sound reinforcement and PA systems design, operation, and technology. Come along to an event that aims to answer questions you never knew you had.

Sunday, October 11 **11:00 am** **Room 1E05**

Technical Committee Meeting on Semantic Audio Analysis

Sunday, October 11 **11:00 am** **Room 1E02**

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

Exhibitor Seminar **Sunday, October 10**
11:00 am – 1:00 pm **Room 1E06**

MANHATTAN PRODUCERS ALLIANCE (MANHATPRO) Business Development and Beyond

Presenter: **Matt Grippo**, CTO, Elephant Ventures; Member & Board of Directors, ManhatPro

Bring your energy, enthusiasm, business ideas and questions as ManhatPro board member Matt Grippo focuses on YOU! Take this unique opportunity to meet Matt and spend some time learning some tips and tricks for business development. You'll participate in our panel teams, discuss your personal career goals and get a chance to meet some ManhatPro members.

Exhibitor Seminar **Sunday, October 10**
11:00 am – 12:30 pm **Room 1E17**

RENKUS-HEINZ, INC.

AFMG SysTune 1.1 Live Sound Measurement and DSP Control

Presenter: **Stefan Feistel**

This seminar is designed to offer a very concise but detailed explanation of what customers can expect from SysTune 1.1 upgrade, AFMG's live sound measurement software. This upgrade brings major additions, including new DSP Plug-In functions, several new performance enhancements, Virtual EQ, improved handling of axis settings, default view limits, and better results for music and speech signals.

Workshop 14 **Sunday, October 11**
11:30 am – 1:00 pm **Room 1E15**

CONSIDERATIONS IN CHOOSING A DIGITAL AUDIO CONVERTER

Chair: **Bruce Whisler**

Panelists: *Bob Bauman*, Lynx Studio Technology, Cosa Mesa, CA, USA
Ian Dennis, Prism Sound, Stretham, Cambridgeshire, UK
Michal Jurewicz, Mytek Digital, New York, NY, USA
John Siau, Benchmark Media Systems, Syracuse, NY, USA

Analog-to-digital conversion is a critical operation in the digital recording of audio. Whether the task at hand is the preservation of historical analog materials or the recording of new digitally-born material, the accuracy and transparency of the A/D converter should be of utmost concern to the audio engineer. This workshop features a panel of experts in the design of digital audio converters. Topics to be covered include sampling and filtering methodologies, sample clock issues, and the performance compromises that must be considered in the design of digital audio converters.

Special Event
PLATINUM PRODUCERS AND ENGINEERS
Sunday, October 11, 11:30 am – 1:00 pm
Room 1E12/13

Moderator: **Paul Verna**

Panelists: *Mike Clink*
Sylvia Massy
Dave Reitzas
Bruce Swedien
Tony Visconti

The producers and engineers on this panel have helped bring to life the musical visions of a dizzying array of recording icons, including Guns N' Roses, Whitesnake, Survivor, Madonna, Celine Dion, Natalie Cole, David Bowie, T. Rex, Angelique Kidjo, Duke Ellington, Michael Jackson, George Benson, Tool, Johnny Cash, and System of a Down. These professionals share insights from decades producing, engineering, and mixing some of the most enduring recordings in history.

Student/Career Development Event
STEREO RECORDING COMPETITION—2
Sunday, October 11, 12:00 noon – 1:00 pm
Room 1E11

The Student Recording Competition is a highlight at each convention. A distinguished panel of judges participates in critiquing finalists of each category in an interactive presentation and discussion. Student members can submit stereo recordings in this category:

- **Pop/Rock**
  Judges: *Susan Rogers, Mark Rubel*

The top three finalists in each category present a short summary of their techniques and intentions, and play their project for all who attend. Meritorious awards will be presented later at the closing Student Delegate Assembly Meeting (SDA-2) on Monday Afternoon.

It's a great chance to hear the work of your colleagues from other educational institutions. Everyone learns from the judges' comments even if your project isn't one of the finalists. You'll meet other students and faculty. This year, the event is followed immediately by the Student Social and Student Dinner, so don't miss it.

Sponsors include AEA, Genelec, JBL, Lavry Engineering, Schoeps, Shure, and Ron Streicher.

Sunday, October 11 12:00 noon Room 1E05
Technical Committee Meeting on Signal Processing

Special Event
LUNCHTIME KEYNOTE:
KEES IMMINK
Sunday, October 11, 1:00 pm – 2:00 pm
Room 1E15

Beethoven, Shannon, and the Compact Disc

In the last twenty-five years, digital electronics has replaced essentially all analog consumer electronics products, and the word "digital" has become synonymous with quality. Digital has penetrated all realms of human activities and has profoundly enriched everyday lives around the world. The invention of the CD with their superior sound quality and scratch-free durability marked the beginning of the shift from analog to digital music technology, the "digital revolution."

An audio compact disc (CD) holds up to 74 minutes, 33 seconds of sound, just enough for a complete mono recording of Ludwig von Beethoven's Ninth Symphony (Alle Menschen werden Brüder) at probably the slowest pace it has ever been played, during the Bayreuther Festspiele in 1951 conducted by Wilhelm Furtwängler.

Each second of music requires about 1.5 million bits, which are represented as tiny pits and lands ranging from 0.9 to 3.3 micrometers in length. More than 19 billion channel bits are recorded as a spiral track of alternating pits and lands over a distance of 5.38 kilometers (3.34 miles), which are scanned at walking speed, 4.27 km per hour.

We will discuss the various crucial technical decisions made that would determine the technical success or failure of the new medium.

Kees Immink is president and founder of Turing Machines Inc. and an adjunct professor at the Institute for Experimental Mathematics, Essen, Germany. Immink, a native of Rotterdam, who obtained his Ph.D. degree from the Eindhoven University of Technology, has progressed the digital audio revolution from its very beginning. Over the course of his career, Immink has contributed to the development of a wealth of digital recording products, including the Compact Disc, DAT, DCC, DVD, and the Blu-Ray disc. Immink's work is documented in over 150 technical papers and over 1000 U.S. and foreign patents. Immink, a former AES president, received widespread applause for his contributions to digital audio; he is a recipient of the AES Gold Medal, IEEE Edison Medal, and SMPTE Progress Medal.

Special Event
DIGITAL MICROPHONES LIVE RECORDING SESSION
Sunday, October 11, 1:30 pm – 5:00 pm
New York University
Loewe Theater and Music Technology Studios
35 West 4th St., NY

Moderator: **Gregor Zielinsky**, Sennheiser

Panelists: *Stephan Flock*, RME
Stephan Peus, Neumann
Helmut Wittek, Schoeps

A live recording session will be presented, recording a small Symphony Orchestra at NYU, New York. Using digital microphones from Schoeps, Neumann, and Sennheiser a whole setup of a digital mic system is shown and explained. Additionally, AES 42 Interfaces from RME are presented and in operation. Participants will then have the opportunity to listen and work with the digital microphones. The new control room of the NYU will be used.

There will be a bus going from Javits to NYU and tickets will be required. They can be picked up at the Technical Tours desk.

Session P15 Sunday, October 11
2:00 pm – 5:30 pm Room 1E07

DIGITAL AUDIO EFFECTS

Chair: **David Berners**, Universal Audio, Inc., Santa Cruz, CA, USA

2:00 pm

P15-1 Discrete Time Emulation of the Leslie Speaker—*Jorge Herrera, Craig Hanson, Jonathan S. Abel*, Stanford University, Stanford, CA, USA

A discrete-time emulation of the Leslie loudspeaker acoustics is described. The midrange horn and subwoofer baffle are individually modeled, with their rotational dynamics separately tracked, and used to drive time-varying FIR filters applied to the input. The rotational speeds of the horn and baffle are approximated by first-order difference equations having different time constants for acceleration and deceleration. Several time-varying FIR filter methods were explored, all based on impulse responses tabulated over a dense set of horn and baffle angles. In one method, the input sample scales an interpolated impulse response at the current horn or baffle angle, which is added to the output. An example model of a Leslie 44W is presented.
Convention Paper 7925

2:30 pm

P15-2 A Novel Transient Handling Scheme for Time Stretching Algorithms—*Frederik Nagel,*

Andreas Walther, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Changing either speed or pitch of audio signals without affecting the respective other is often used for music production and creative reproduction, such as remixing. It is also utilized for other purposes such as bandwidth extension and speech enhancement. While stationary signals can be stretched without harming the quality, transients are often not well maintained after time stretching. The present paper demonstrates a novel approach for transient handling in time stretching algorithms. Transient regions are replaced by stationary signals. The thereby removed transients are saved and re-inserted to the time dilated stationary audio signal after time stretching.
Convention Paper 7926

3:00 pm

P15-3 The Switched Convolution Reverberator—
Keun-Sup Lee,[1] *Jonathan S. Abel*,[1] *Vesa Välimäki*,[2] *David P. Berners*[3]
[1]Stanford University, Stanford, CA, USA
[2]Helsinki University of Technology, Espoo, Finland
[3]Universal Audio, Inc., Santa Cruz, CA, USA

An artificial reverberator having low memory and small computational cost, appropriate for mobile devices, is presented. The reverberator consists of an equalized comb filter driving a convolution with a short noise sequence. The reverberator equalization and decay rate are controlled by low-order IIR filters, and the echo density is that of the noise sequence. While this structure is efficient and readily generates high echo densities, if a fixed noise sequence is used, the reverberator has an unwanted periodicity at the comb filter delay length. To overcome this difficulty, the noise sequence is regularly updated or “switched.” Several structures for updating the noise sequence, including a leaky integrator sensitive to the signal crest factor, and a multiband architecture, are described.
Convention Paper 7927

3:30 pm

P15-4 An Emulation of the EMT 140 Plate Reverberator Using a Hybrid Reverberator Structure—
Aaron Greenblatt,[1] *Jonathan S. Abel,*[1,2] *David P. Berners*[1,2]
[1]Stanford University, Stanford, CA, USA
[2]Universal Audio Inc., Scotts Valley, CA, USA

A digital emulation of the Elektromesstechnik (EMT) 140 plate reverberator is presented. The EMT 140 consists of a signal plate and a moveable damping plate; it is approximately linear and time invariant, and its impulse response is characterized by a whip-like onset and high echo density. Here, the hybrid reverberator proposed by Stewart and Murphy, in which a short convolution is run in parallel with a feedback delay network (FDN), is used to model the plate. The impulse response onset is only weakly dependent on the damping control and is modeled by the convolution; the FDN is fit to the impulse response tail. The echo density, equalization, and decay rates are matched at the transition between the convolution and FDN.
Convention Paper 7928

4:00 pm

P15-5 Simulation of a Guitar Amplifier Stage for Several Triode Models: Examination of Some Relevant Phenomena and Choice of Adapted Numerical Schemes—
Ivan Cohen,[1,2] *Thomas Hélie*[1]
[1]IRCAM, Paris, France
[2]Orosys R&D, Montpellier, France

This paper deals with the simulation of a high gain triode stage of a guitar amplifier. Triode models taking into account various “secondary phenomena” are considered and their relevance on the stage is analyzed. More precisely, both static and dynamic models (including parasitic capacitances) are compared. For each case, the stage can be modeled by a nonlinear differential algebraic system. For static triode models, standard explicit numerical schemes yield efficient stable simulations of the stage. However, the effect due to the capacitances in dynamic models is audible (Miller effect) and must be considered. The problem becomes stiff and requires the use of implicit schemes. The results are compared for all the models and corresponding VST plug-ins have been implemented.
Convention Paper 7929

4:30 pm

P15-6 Over-Threshold Power Function Feedback Distortion Synthesis—
Tom Rutt, Coast Enterprises, LLC, Asbury Park, NJ, USA

This paper describes an approach to nonlinear distortion synthesis, which uses Over-Threshold Power Function (OTPF) Feedback. The linear gain of an OTPF Feedback distortion synthesizer (using a high gain amplifier) is determined by a linear feedback element. When the output signal becomes greater than a positive threshold value, or less than a negative threshold value, additional OTPF feedback is applied to the distortion synthesizer. The action of this OTPF Feedback Distortion synthesis closely emulates the soft limiting input/output response characteristics of vacuum tube triode grid limit distortion. An important feature of an OTPF feedback distortion synthesizer is that it always behaves as an instantaneous soft limiter and never results in clipping of the output signal peak levels (even at maximum allowable peak input signal levels), if its nonlinear gain constants are set optimally. The paper also describes both circuit and software plug-in realizations of distortion synthesizers that employ Over-Threshold Cubic Function Feedback.
Convention Paper 7930

5:00 pm

P15-7 Dynamic Panner: An Adaptive Digital Audio Effect for Spatial Audio—
Martin Morrell, Joshua D. Reiss, Queen Mary University of London, London, UK

Digital audio effects usually have their parameters controlled by the user, whereas adaptive digital audio effects, or A-DAFx, have some parameters that are driven by the automatic processing and extraction of sound features in the

signal content. In this paper we introduce a new A-DAFx, the Dynamic Panner. Based on RMS measurement of its incoming audio signal, the sound source is panned between two user defined points. The audio effect is described and discussed, detailing the technicalities of all the control parameters and the creative context in which the effect can be used. Objective results that can be obtained from the effect are also presented. The audio effect has been implemented as both stereo and 3-dimensional effects using Max/MSP.
Convention Paper 7931

Workshop 15 **Sunday, October 11**
2:00 pm – 4:00 pm **Room 1E11**

BLU-RAY AS A HIGH RESOLUTION AUDIO FORMAT FOR STEREO AND SURROUND

Chair: **Stefan Bock**, msm-studios, Germany

Panelists: *Gary Epstein*, Dolby, USA
Tom McAndrew, DTS, USA
Johannes Müller, msm-studios, Germany
Ronald Prent, Galaxy Studios, Belgium

The decision for the Blu-ray disc as the only HD packaged media format also offering up to 8 channels of uncompressed high resolution audio has at least eliminated one of the obstacles of getting high-res stereo and surround sound music to the market. The concept of utilizing Blu-ray as a pure audio format will be explained and Blu-ray will be positioned as successor to both SACD and DVD-A. The operational functionality and a double concept of making it usable both with and without screen will be demonstrated by showing some products that are already on the market.

Special Event
BEHIND THE GLASS: AUDIO PRODUCTION IN THE 21ST CENTURY
Sunday, October 11, 2:00 pm – 4:00 pm
Room 1E12/13

Moderator: **Howard Massey**

Panelists: *Tony Brown*
Simon Climie
Ryan Hewitt
George Massenburg
Ann Mincieli
Russ Titelman

Howard Massey is a leading audio industry consultant, technical writer, and author of *Behind the Glass* and *Behind the Glass Volume II*, two collections of in-depth interviews with top record producers and audio engineers widely used in recording schools the world over. He also co-authored legendary Beatles engineer Geoff Emerick's acclaimed 2006 memoir, *Here, There, and Everywhere: My Life Recording the Music of the Beatles*.

Join us as we gather together some of the top names from the recently released *Behind The Glass Volume II* for a lively discussion about the state of the industry today and a look ahead at the way that production techniques are likely to evolve in the years to come. Topics will include new media and online distribution; the application of legacy (analog) technology in an increasingly digital world; and the critical role of the producer/engineer team in driving both the creative and technical aspects of the record-making process.

Student/Career Development Event
MENTORING SESSION
Sunday, October 11, 2:00 pm – 4:00 pm
Room 1E06

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

Sunday, October 11 **2:00 pm** **Room 1E05**
Technical Committee Meeting on Perception and Subjective Evaluation of Audio Signals

Sunday, October 11 **2:00 pm** **Room 1E02**
Standards Committee Meeting SC-03-06, Digital Library and Archive Systems

Exhibitor Seminar **Sunday, October 10**
2:00 pm – 3:00 pm **Room 1E17**

DSCOPE

Solving the Problem of Reflections with Desktop Speakers

Presenters: **Dan Foley**
Jim Tuomy

Jim Tuomy presents a patented new design for desktop loudspeaker enclosures that avoids the problem of reflections from the desk top, inspired by and developed using the adjustable time window of Prism Sound's dScope Series III audio analyzer.

Master Class 4 **Sunday, October 11**
2:30 pm – 4:30 pm **Room 1E16**

BOB HODAS

Presenter: **Bob Hodas**

Wrangling the Room

Acoustician and studio musician Bob Hodas has traveled the world tuning over a thousand rooms from Tokyo for Sony Entertainment, to London for Abbey Road, and Stateside for Blackbird and Lucasfilm. The focus of the master class will be the loudspeaker/room interface and optimizing system performance. A slide presentation will demonstrate the frequency and phase response of loudspeakers placed properly and improperly in a room. Integrating subwoofers with the main speaker system for both stereo and surround systems will be covered, as will be equalization and what types of acoustic treatments to use and how to determine their proper placement.

Broadcast/Media Streaming Session 10
Sunday, October 11 **2:30 pm – 4:00 pm**
Room 1E08

AUDIO PROCESSING FOR INTERNET STREAMING

Chair: **David K. Bialik**, DKB Broadcast Associates

Presenters: *Ray Archie*, CBS

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Marvin Caesar, Aphex
Steve Dove, Wheatstone
Frank Foti, Omnia
Thomas Lund, TC Electronics
Geir Skaaden, DTS, Inc.

With the growing popularity of streaming over the Internet, content originators want to be able to deliver intelligible, robust audio that is competitive with other mediums. Given the unique restraints and what the audience listens with, internet streaming requires an audio processing chain unique to radio and television. This session will discuss various techniques and technologies to enhance the audio content to be delivered over the World Wide Web.

Live Sound Seminar 9 **Sunday, October 11**
2:30 pm – 4:15 pm **Room 1E09**

AUTOMIXING FOR LIVE SOUND

Moderator: **Michael "Bink" Knowles**, San Francisco, CA, USA

Panelists: *Bruce Cameron*, Sound Diva
Dan Dugan, Dan Dugan Sound Design
Kevin Maxwell
Enrique Perez Gonzalez, PhD student
Peter Schneider, Gotham Sound

This session offers the live sound mixer a comprehensive discussion of automixer considerations with panelists covering the strengths and weaknesses of automixer topologies and algorithms with respect to their sound quality and ease of use in the field. Analog and digital systems will be compared. Real world applications will be presented and discussed. Questions from the audience will be encouraged.

Games Audio 6 **Sunday, October 11**
2:30 pm – 3:30 pm **Room 1E15**

IMPLEMENTING SOUND AND MUSIC FOR GAMES USING THE UNREAL 3 ENGINE

Presenter: **Richard Stevens**

This tutorial will explore and demonstrate the principles of audio for games and illustrate their technical implementation within the Unreal 3 engine. Through a live demonstration within the Unreal games engine itself the following aspects of games audio will be explored and demonstrated. The implications and challenges of interactivity with reference to memory, repetition and variation. Potential solutions to these issues through use of sample rates, randomization, wavetable synthesis, filtering, concatenation and layering for variation within sound cues, together with an illustration of sound propagation and attenuation over distance conventions within games. The tutorial will also cover the challenges of interactivity for music in games including simple musical transitions, aligned transitions, and generative musical techniques.

Exhibitor Seminar **Sunday, October 10**
3:15 pm – 4:15 pm **Room 1E17**

PRISM

Eliminating Pops and Glitches and Other Digital Audio Horrors

Presenters: **Doug Ordon**
Simon Woollard

Systems storing and transmitting digital audio are often plagued by intermittent pops and glitches or other problems such as channel status (metadata) implementation and interpretation. Simon Woollard presents a range of techniques aimed at ensuring clean and reliable digital audio recording, storage and transmission.

Session P16 **Sunday, October 11**
3:30 pm – 5:00 pm **Foyer 1E**

POSTERS: MULTICHANNEL SOUND AND IMAGING

3:30 pm

P16-1 Evaluation of a Multipoint Equalization System Based on Impulse Responses Prototype Extraction—*Stefania Cecchi,*[1] *Lorenzo Palestini,*[1] *Paolo Peretti,*[1] *Laura Romoli,*[1] *Francesco Piazza*,[1] *Alberto Carini*[2]
[1]Università Politecnica delle Marche, Ancona, Italy
[2]Universita' di Urbino, Urbino, Italy

In this paper a frequency domain multipoint equalization algorithm, which combines fractional octave smoothing of measured impulse responses (IRs) in multiple locations and the extraction of a representative prototype, is presented. The proposed approach is evaluated considering different methods to combine the IRs for the prototype extraction and to obtain the inverse filter for equalization, using sets of impulse responses measured in realistic environments. With respect to previous works, the influence on the equalization performance of the number of considered positions and of the equalization zone size is deeply investigated. Also a comparison with the single point equalization approach is reported. Finally, the multipoint equalization robustness is evaluated also on positions different from those used for the equalizer estimation.
Convention Paper 7932
[Paper presented by Laura Romoli]

3:30 pm

P16-2 Acoustic Design of NHK HD-520 Multichannel Postproduction Studio—A New Approach for Room Acoustic Design Using Multi-Layered Random Diffusers—*Yasushi Satake,*[1] *Kazuhiro Makino,*[1] *Yasuhiro Sakiyama,*[1] *Hideo Tsuro*,[1] *Akira Fukada,*[2] *Ryota Ono*,[2] *Kazutsugu Uchimura*,[3] *Junichi Mikami*,[4] *Masamichi Otani*,[2] *Ikuko Sawaya*[5]
[1]Nittobo Acoustic Engineering Co., Ltd., Tokyo, Japan
[2]NHK (Japan Broadcasting Corporation), Shibuya-ku, Tokyo, Japan
[3]NHK Media Technology, Shibuya-ku, Tokyo, Japan
[4]NHK Integrated Technology, Shibuya-ku, Tokyo, Japan
[5]NHK Science & Technical Research Laboratories, Ketagaya-ku, Tokyo, Japan

In this paper a novel approach for room acoustic design adopting the renewal project of NHK HD-520 multichannel postproduction studio is introduced. HD-520 studio is designed for direct

surround loudspeaker arrangements based on ITU-R BS.775-1 and adopting an acoustically transparent screen. Generally, there are three important keys for acoustic design of multichannel postproduction studios. The first is to obtain stable and flat low frequency responses; the second, smooth panning and accurate phantom sound image. And the third, natural sounds that have well-balanced frequency characteristics. To resolve these problems and produce a superior monitoring environment, a new approach for room acoustic design using multi-layered random diffusers with cylindrical diffusers of different sizes (MLRD) is applied for this project. First of all, an outline and the acoustic design concept for the renewal of NHK HD-520 studio are introduced. Second, the concrete method for room acoustic design for the purpose of certifying the high quality for monitoring of both audio and picture is introduced. The preferable acoustic characteristics were shown in the measurement results, and a high reputation has been given by engineers for the comfortable work area for a surround monitoring environment.
Convention Paper 7933

3:30 pm

P16-3 Robust Interchannel Correlation (ICC) Estimation Using Constant Interchannel Time Difference (ICTD) Compensation—*Dongil Hyun,*[1] *Jeongil Seo,*[2] *Youngcheol Park,*[3] *Daehee Youn*[1]
[1]Yonsei University, Seoul, Korea
[2]Electronics and Telecommunications Research Institute (ETRI), Daejeon, Korea
[3]Yonsei University, Wonju, Korea

This paper proposes an interchannel correlation (ICC) estimation method that can enhance the performance of the spatial audio coding such as Parametric Stereo of HE-AACv2 and MPEG Surround. Conventional ICC estimation methods assume that phase differences between two channel signals are constant in parameter bands and those phase differences are compensated to maximize the ICC. The proposed method introduces robust ICC estimation by compensating constant interchannel time difference (ICTD). ICTD is estimated from interchannel phase difference (IPD) and linear phases corresponding to ICTD are compensated. ICTD is Simulation results show that the proposed method provides more accurate ICC's than the conventional methods.
Convention Paper 7934

Convention Paper 7935 has been withdrawn

3:30 pm

P16-5 Measurement of Audio System Imaging Performance—*David Clark*, DLC Design. Northville, MI, USA

Mixing engineers assign different sounds to different channels of stereo or multichannel media with the expectation that the listener will experience the intended directional aspects of the sound. For standard +/–30 degree arrangement of playback speakers and listener, this expectation is usually realized. As non-standard influences, such as listening off-centerline, are introduced, localization and other aspects of spatial rendition are degraded. This paper describes a measurement system for quantifying these degradations in common perceptual dimensions such as image direction, width, distance, and stability.
Convention Paper 7936

3:30 pm

P16-6 Matching Perceived Auditory Width to the Visual Image of a Performing Ensemble in Contrasting Multi-Modal Environments—*Daniel L. Valente,*[1] *Shane A. Myrbeck,*[2] *Jonas Braasch*[3]
[1]Boys Town National Research Hospital, Omaha, NE, USA
[2]Arup Acoustics, San Francisco, CA, USA
[3]Rensselaer Polytechnic Institute, Troy, NY, USA

Participants were given an audio-visual matching test, in which they were instructed to align the acoustic width of a performing ensemble to a varying set of audio and visual cues. Participants are asked to assess a vocal ensemble that is positioned with varying visual width in five contrasting physical spaces with monotonically-increasing Reverberation Times. Each performance to be assessed begins with forced auditory-visual mismatch (the acoustical location of the sound sources not matching that of the visual imagery), and participants were instructed to align the acoustic presentation to the visual imagery of the performance. The results show that the participants' matching ability is based on the source-distance as well as the spacing of the ensemble.
Convention Paper 7937

3:30 pm

P16-7 Stereo Music Source Separation for 3-D Upmixing—*Hwan Shim,*[1] *Jonathan Abel,*[1] *Koeng-Mo Sung*[2]
[1]Stanford University, Stanford, CA, USA
[2]Seoul National University, Kwanak-Gu, Seoul, Korea

A method for 3-D upmixing based on stereo source separation and a primary-ambient decomposition is presented. The method separately renders primary and ambient components, and separately pans sources derived from the primary signal. Since all separated sources appear in the upmixed output, it is more important that the source separation method be free of audible artifacts than achieve a complete separation of the sources present. Typically, the mixing vector amplitude or energy is allocated to the various sources present, for instance all given to the most likely source, or allocated to each source in proportion to its likelihood. However, these choices produce “musical” noise and source motion artifacts in the upmixed signal. Here, two sources are selected according to the mixing vector direction, and the mixing vector energy is allocated by inverting the panning matrix associated with the selected sources. Listening tests show an upmix with separated sources and few audible artifacts.
Convention Paper 7938

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3:30 pm

P16-8 Automated Assessment of Surround Sound —*Richard C. Cabot*, Qualis Audio, Lake Oswego, OR, USA

The design of a real time electronic listener, optimized for surround sound program assessment, is described. Problems commonly encountered in surround audio production and distribution are automatically identified, including stereo/mono downmix compatibility, balance, metadata inconsistencies, channel interchange, loudness, excessive or inadequate level, and the presence of hum. Making measurements that correlate with audibility, displaying the results in a form easily understood by non-audio personnel created numerous design challenges. The technology used to solve these challenges, particularly that of downmix compatibility, will be described.
Convention Paper 7939

Games Audio 7 **Sunday, October 11**
3:30 pm – 4:30 pm **Room 1E15**

VO FOR VIDEOGAMES: PROCESS AND PIPELINE

Chair: **Morla Gorrondona**

Panelists: *Chip Beaman*
Alexander Brandon
David Chan
Greg deBeer

The Game Audio Network Guild (G.A.N.G.) is the largest organization in the world for game audio professionals, with members in over thirty countries, and the GANG Voice Actor Coalition (GVAC) is the Guild's newest pro-only branch. GVAC members and co-chairs will present a panel discussion about voice acting and dialog production in video games. This exclusive panel features some of the biggest name in modern VO, and will feature lively discussion on both creative and business best practices in the video game industry. Topics include running a successful recording session, technical considerations, recorded examples, reels, casting, union considerations, on-the-gig issues, scripts, interactive vs. linear, getting the best performances out of an actor, budget, and planning, etc.

Broadcast/Media Streaming Session 11
Sunday, October 11 **4:00 pm – 5:30 pm**
Room 1E08

LOUDNESS AND AUDIO PROCESSING FOR BROADCAST

Chair: **Dom Bordonaro**, Cox Radio

Panelists: *Marvin Caesar*, Aphex
Tim Carroll, Linear Acoustic
Frank Foti, Telos-Omnia-Axia
Ken Hunold, Dolby
James D. (JJ) Johnston, DTS Inc.
Thomas Lund, TC Electronics
Andrew Mason, BBC R&D
Jim Starzinski, NBC

The panelists in this session are experts in the field of audio processing and all have various techniques and opinions on how best to tame the audio levels in broadcasting—from the control of loud commercials in television to the competitive loudness of radio stations. Audio processing has always been science with a good measure of art thrown in, so this discussion will be interesting not only from a technical viewpoint, but also from an artistic angle. We will discuss Dialnorm and the use of meta-data for television and loudness and dynamic range for radio.

After the panelists have made their presentations, the audience is invited to participate in a question and answer session, which should be an interesting mix of technology and technique.

Student/Career Development Event
EDUCATION FORUM PANEL
Sunday, October 11, 4:00 pm – 6:00 pm
Room 1E06

Moderators: **Alex Case**, University of Massachusetts Lowell, MA, USA
John Krivit, New England Institute of Art, Brookline, MA, USA

Continuing the thread of very productive dialog started at the AES Convention in Munich last May, all educators are invited to join this open discussion on how we teach students about sound quality and high production standards. In an age when many graduates will not have the chance to work with the finest equipment in the finest rooms, what are the responsibilities of the faculty and what are the expectations for the facilities at institutions of higher learning? Come share your approaches and learn from others as we all seek to teach quality. An open forum for educators, students are also welcome to attend and participate.

Sunday, October 11 **4:00 pm** **Room 1E05**
Technical Committee Meeting on Human Factors in Audio Systems

Sunday, October 11 **4:00 pm** **Room 1E02**
Standards Committee Meeting SC-03-07, Audio Metadata

Workshop 16 **Sunday, October 11**
4:30 pm – 6:00 pm **Room 1E15**

INTRODUCTION TO AUDIO FORENSICS

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

Panelists: *Eddy B. Brixen*, EBB-consult, Smorum, Denmark
Robert Maher, Montano State University, Bozeman, MT, USA
Tom Owen, Owl Investigations, NJ, USA
Jeffrey Smith, University of Colorado, Denver, CO, USA

This is a non-technical introduction to the field of Audio Forensics, sponsored by the Technical Committee on Audio Forensics, intended for persons new or interested in the field. Audio Forensic professionals associated with the AES will review basic methods, challenges, and history related to the field. Opportunities for education (and certification) will also be discussed.

This workshop is sponsored by the Technical Committee on Audio Forensics.

Live Sound Seminar 10 **Sunday, October 11**
4:30 pm – 6:15 pm **Room 1E09**

NETWORKING DIGITAL AUDIO IN LIVE SOUND

Moderator: **Jim Risgin**, On Stage Audio

Panelists: *Brad Benn*, Crown
Rick Kreifeldt, Harman
Albert Lecesse, Audio Analysts
Marc Lopez, Yamaha
Varuni Witana, Audinate Director of Engineering

This session will discuss the challenges and planning involved with deploying digital audio in the sound reinforcement environment. The panel will cover not just the use of digital audio but some of the factors to be considered during the design and application of digital networking within the audio system. While no one solution fits every application, after this panel discussion you will be better able to understand the issues at hand.

Special Event
LES PAUL—CHASING SOUND
Sunday, October 11, 4:30 pm – 64300 pm
Room 1E11

Moderator: **David Bowles**

Panelists: *John Paulson*, Director, Cinematographer, and Editor
Robert Sullivan, Sound Recordist

A 90-minute performance documentary on the life and accomplishments of the legendary Les Paul. A true inventor, Paul was father of the solid-body electric guitar, inventor of overdubbing and multi-track recording, king of the '50s pop charts, and architect of rock 'n' roll. In addition Les Paul's connection with the AES dates back to its inception. This film was produced in High Definition by John Paulson Productions, LLC, Icon Television Music, and WNET/Thirteen American Masters. After the film, the panelists will be interviewed on artistic and technical details of how this project came into being, after which a limited number of questions will be taken from the audience.

Tutorial 4 **Sunday, October 11**
5:00 pm – 6:30 pm **Room 1E16**

ELECTROACOUSTIC MEASUREMENTS OF HEADPHONES AND EARPHONES

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

This tutorial reviews basic electroacoustic measurement concepts: gain, sensitivity, sound fields, signals, linear, and nonlinear systems. 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. A brief introduction to Noise Canceling devices is also presented.

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

KEVIN KILLEN

Presenter: **Kevin Killen**

Platinum Mixing in the Box

Is it possible to mix a million selling record in a bedroom? Kevin Killen (U2, Sugarland, Peter Gabriel) has. His background in the analog domain and transition to the digital environment has enhanced his reputation as a world-class mixer. Discover some of his techniques and how technology, budgets, and past experiences have shaped his approach to the ever evolving challenges of the marketplace.

Sunday, October 11 **5:00 pm** **Room 1E05**
Technical Committee Meeting on Spatial Audio

Broadcast/Media Streaming Session 12
Sunday, October 11 **5:30 pm – 7:00 pm**
Room 1E08

PRODUCTION OF A SOAP OPERA

Presenters: **Dominick Maldari**, All My Children
Bill Mozer, One Life to Live
R.T. Smith

There's a lot more than meets the eye when it comes to mixing a soap opera. It may be considered a dying art, but the production mixer must deal with overworked actors having difficulty following a script, complicated, 3-dimensional blocking, lighting grids that make booming the action more difficult, and botched blocking. All this every day after being handed only a script, and the most rudimentary information as to what's in store.

In addition, the A-1 must determine when booming is not feasible and the use of RF body mic-ing is the lesser of two evils. That's where an agile A-2 assist comes into play. These mics have the added difficulty of having to be hidden when in use. His techniques of doing this are very important as time is always of the essence.

Finally, when all is said and done, the tapes are sent to post-production to have sound effects and music added to the final product, as well as whatever effects are deemed necessary. The postproduction engineer also makes fixes to the relatively few and inevitable flaws that occur in this high speed taping environment.

Dominick Maldari, 5-time Emmy award winner in this endeavor, will give an overview and also discuss the mixing aspects. Emmy award nominee Bill Mozer will go over the audio assist's techniques, and R.T. Smith will talk about the art of postproduction.

Tutorial 5 **Sunday, October 11**
6:00 pm – 7:00 pm **Room 1E07**

AUDIO PRESERVATION AT THE NATIONAL AUDIO-VISUAL CONSERVATION CENTER (NAVCC)

Presenter: **Brad McCoy**

This tutorial will discuss audio preservation at the Library of Congress' National Audio-Visual Conservation Center (NAVCC), which was recently completed in Culpeper, VA. It will also give an overview of the NAVCC, a state- ➥

of-the-art facility for storing and preserving recorded sound, video, and film materials.

Sunday, October 11 6:00 pm Room 1E02

Standards Committee Meeting SC-03-12, Forensic Audio

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Sunday, October 11, 7:15 pm – 9:00 pm
Room 1E12/13

Lecturer: **Phil Ramone**

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 127th AES Convention is Phil Ramone. The title of his address is "Technology and Music Meet Again."

Ramone's presentation will focus on the continuing partnership that technology and music have played through out the years. The evolution of the recording industry will be explored showing how technological innovations have lead to revolutionize the experience of music for the artist and fan a like. The exponential pace of technology will be spotlighted, leading to how it has left our entire industry in chaos.

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

Session P17 Monday, October 12
9:00 am – 12:00 noon Room 1E16

AUDIO NETWORKS

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

9:00 am

P17-1 Performance Metrics for Network Audio Systems: Methodology and Comparison—*Nicolas Bouillot, Mathieu Brulé, Jeremy R. Cooperstock*, McGill University, Montreal, Quebec, Canada

Network audio transmission is becoming increasingly popular within the broadcast community, with applications to Voice over IP (VoIP) communications, audio content distribution, and radio broadcast. Issues of end-to-end latency, jitter, and overall quality, including glitches of the delivered signal, all impact on the value of the technology. Although considerable literature exists comparing audio codecs, little has been published to compare systems in terms of their real-word performance. In response, we describe methods for accurately assessing the quality of audio streams transmitted over networks. These methods are then applied to an empirical evaluation of several audio compression formats supported by different streaming engines.
Convention Paper 7940

9:30 am

P17-2 An Integrated Connection Management and Control Protocol for Audio Networks—*Richard Foss,*[1] *Robby Gurdan,*[2] *Bradley Klinkradt,*[2] *Nyasha Chigwamba*[2]
[1]Rhodes University, Grahamstown, South Africa
[2]Universal Media Access Networks (UMAN), Dusseldorf, Germany

With the advent of digital networks that link audio devices, there is a need for a protocol that integrates control and connection management, allows for streaming of all media content such as audio and video between devices from different manufacturers, and that provides a common approach to the control of these devices. This paper proposes such a protocol, named XFN (currently being standardized as part of the AES X170 project). XFN is an IP-based peer to peer network protocol in which any device on the network may send or receive connection management, control, and monitoring messages. Essential to the XFN protocol is the fact that each parameter in a device can be addressed via a hierarchical structure that reflects the natural layout of the device.
Convention Paper 7941

10:00 am

P17-3 Mixing Console Design Considerations for Telematic Music Applications—*Jonas Braasch,*[1] *Chris Chafe,*[2] *Pauline Oliveros,*[1] *Doug Van Nort*[1]
[1]Rensselaer Polytechnic Institute, Troy, NY, USA
[2]Stanford University, Stanford, CA, USA

This paper describes the architecture for a new mixing console that was especially designed for telematic live music collaborations. The prototype mixer is software-based and programmed in Pure Data. It has many traditional features but also a number of extra modules that are important for telematic projects: transmission test unit, latency meter, remote data link, auralization unit, remote sound level calibration unit, remote monitoring, and a synchronized remote audio recording unit.
Convention Paper 7942

10:30 am

P17-4 Comparison of Receiver-Based Concealment and Multiple Description Coding in an 802.11-Based Wireless Multicast Audio Distribution Network—*Marcus Purat, Tom Ritter*, Beuth Hochschule für Technik Berlin, Berlin, Germany

This paper presents aspects of a study of different methods to mitigate the impact of packet loss in a wireless distribution network on the subjective quality of compressed high fidelity audio. The system was simulated in Matlab based on parameters of an 802.11a WLAN in multicast-mode and the Vorbis codec. To aid

the selection of the most appropriate packet loss concealment strategy not only the additional bandwidth, the processing requirements or the latency need to be considered. One important differentiating factor is the perceived subjective audio quality. Therefore an accurate estimate of the subsequent audio quality is required. Several simulation-based methods using psychoacoustic models of the human hearing system to quantify the subjective audio quality are compared.
Convention Paper 7943

11:00 am

P17-5 Audio-Over-IP Acceptance Test Strategy— *Matthew J. O'Donnell*, BSkyB (British Sky Broadcasting), London, UK

Ensuring the integrity of an application that delivers audio-over-IP through Ethernet demands thorough acceptance testing during the development cycle, due to the effect of the potentially volatile "Best Effort" nature of IP transport upon performance of the application. This paper investigates attributes of protocols used on top of IP that must be taken into account during development and their impact on an audio-over-IP's Quality of Experience to the end user.
Convention Paper 7944

11:30 am

P17-6 Long-Distance Uncompressed Audio Transmission over IP for Postproduction— *Nathan Brock,*[1] *Michelle Daniels*,[1] *Steve Morris*,[2] *Peter Otto*[1]
[1]University of California, San Diego, La Jolla, CA, USA
[2]Skywalker Sound, Marin County, CA, USA

The highly distributed nature of contemporary cinema postproduction has led many to believe that high-speed networking of uncompressed audio could significantly improve workflow efficiency. This paper will provide an overview of several significant issues with long-distance networking, including synchronization, latency, bandwidth limitations, and control protocols. We will present a recent networked postproduction demonstration, in which audio assets in Seattle, San Francisco, and San Diego along with local video assets were streamed to and controlled from a single DAW. These results are expected to lead to persistent wide-area networked postproduction environments to remotely access and control audiovisual assets.
Convention Paper 7945

Session P18 **Monday, October 12**
9:00 am – 10:30 am **Room 1E07**

ANALYSIS AND SYNTHESIS OF SOUND

Chair: **Sunil Bharitkar**, Audyssey Labs/USC, Los Angeles, CA, USA

9:00 am

P18-1 Audio Bandwidth Extension Using Cluster Weighted Modeling of Spectral Envelopes— *Nikolay Lyubimov, Alexey Lukin*, Moscow State University, Moscow, Russian Federation

This paper presents a method for blind bandwidth extension of band-limited audio signals. A rough generation of the high-frequency content is performed by nonlinear distortion (waveshaping) applied to the mid-range band of the input signal. The second stage is shaping of the high-frequency spectrum envelope. It is done by a Cluster Weighted Model for MFCC coefficients, trained on full-bandwidth audio material. An objective quality measure is introduced and the results of listening tests are presented.
Convention Paper 7946

9:30 am

P18-2 Applause Sound Detection with Low Latency —*Christian Uhle*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

This paper presents a comprehensive investigation on the detection of applause sounds in audio signals. It focuses on the processing of single microphone recordings in real-time with low latency. A particular concern is the intensity of the applause within the sound mixture and the influence of the interfering sounds on the recognition performance which is investigated experimentally. Well-known feature sets, feature processing, and classification methods are compared. Additional low-pass filtering of the feature time series leads to the concept of sigma features and yields further improvements of the detection result.
Convention Paper 7947

10:00 am

P18-3 Loudness Descriptors to Characterize Wide Loudness-Range Material—*Esben Skovenborg, Thomas Lund*, TC Electronic A/S, Risskov, Denmark

Previously we introduced the concept of loudness descriptors—key numbers to summarize loudness properties of a broadcast program or music track. This paper presents the descriptors: Foreground Loudness and Loudness Range. Wide loudness-range material is typically level-aligned based on foreground sound rather than overall loudness. Foreground Loudness measures the level of foreground sound, and Loudness Range quantifies the variation in loudness. We propose to use these descriptors for loudness profiling and alignment, especially when live, raw, and film material is combined with other broadcast programs, thereby minimizing level-jumps and also preventing unnecessary dynamics-processing. The loudness descriptors were computed for audio segments in both linear PCM and perceptually coded versions. This evaluation demonstrates that the descriptors are robust against nearly-transparent transformations.
Convention Paper 7948

Workshop 17 **Monday, October 12**
9:00 am – 11:00 am **Room 1E15**

SCALABLE AUDIO CODING—PLEASING ALL OF THE PEOPLE ALL OF THE TIME?

Chair: **Noel McKenna**, APTX, Belfast, Northern Ireland, UK

Panelists: *Rick Beaton*, DTS Inc., CA, USA
Jürgen Herre, Fraunhofer IIS, Erlangen, Germany
Gary Spittle, Cambridge Silicon Radio, Cambridge, UK
David Trainor, APTX, Belfast, Northern Ireland, UK

Scalability is becoming an important characteristic for audio coding, particularly with the increasing variety of audio coding applications and equipment. The concept that more unified approaches to audio coding can elegantly scale and adapt across a wide range of deployment scenarios, while still delivering good levels of coding performance, is very appealing. Scalable audio coding is often considered in the context of scaling the coded bit-rate, but many other important forms of coding scalability exist. A panel of experts will discuss coding scalability across various characteristics, including coded bit-rate, computational complexity, robustness, and type of audio content. Consideration will be given to the challenges that real-world applications present for scalable audio coding and how future coding techniques might evolve to promote higher levels of scalability.

Broadcast/Media Streaming Session 13
Monday, October 12 **9:00 am – 11:00 am**
Room 1E11

SOUND EFFECTS: RECORDING AND MIXING FOR DIFFERENT MEDIA

Co-chairs: **David Shinn**, SueMedia Productions
Sue Zizza, SueMedia Productions

Presenters: *Butch D'Ambrosio*
Marc Wiener

Sound Effects and Sound Design artists, Sue Zizza and David Shinn, of SueMedia Productions, return to AES with an updated demonstration of performance and recording techniques for sound effects and foley, as well as multitrack recording and mixing techniques, for a variety of entertainment media. Included in their presentation will be practical demonstrations on the use of sound effects in audio sound design, including working alongside talent when creating and recording effects, microphone selection for sound effect recording, multitrack and surround sound recording and mixing.

Live Sound Seminar 11 **Monday, October 12**
9:00 am – 10:45 am **Room 1E09**

STATE OF THE ART LOUDSPEAKER DESIGN FOR LIVE SOUND

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

Panelists: *Doug Jones*, Danley Sound Labs
Dave Gunness, Fulcrum Acoustic
Charlie Hughes, Excelsior Audio Design & Services, LLC

Following last year's roundtable discussion with six leading live sound loudspeaker design engineers, this year we have invited three designers back to discuss more specific areas of interest. Dave Gunness (Fulcrum Acoustic) will dispel a few myths and offer his insights into some aspects of how loudspeakers are designed and optimized today. Tom Danley (Danley Sound Labs) will focus on the GH 60, an innovative passive fullrange device with asymmetrical coverage levels. Charlie Hughes (Excelsior Audio Design and Services) will discuss HF horn pattern flip and determining crossover parameters between subwoofers and whatever is above.

Games Audio 8 **Monday, October 12**
9:00 am – 11:00 am **Room 1E08**

INTERACTIVE SPEECH FOR GAMES

Chair: **Steve Martz**, THX, San Francisco, CA, USA

Panelists: *Hope Dippel*, SCEA, San Francisco, CA, USA
Dieter Piltz, Blue Castle Games, Vancouver, BC, Canada

Sports games present one of the greatest challenges in game audio—creating realistic play-by-play and color analysis from a virtual sports announcer. Seamless transitions and accurate descriptions are required for a believable experience in an adaptive environment. This workshop will cover topics pertaining to the compilation and realization of interactive speech in games. We will begin with asset creation— scripting, dialog recording, and working with celebrity voice talent. Then we will cover the implementation process—editing, mastering, phrase-stitching, asset management, and speech systems. Panelists will discuss their methods and experiences in the creation of interactive speech content for sport game titles.

Student/Career Development Event
CONTINUING EDUCATION OPTIONS IN AUDIO
Monday, October 12, 9:00 am – 10:30 am
Room 1E06

Panelists: **Alex Case**, University of Massachusetts Lowell, MA, USA
Terron Darby, Pro Media Training, New York, NY, USA
Elsa Lankford, Towson University, Baltimore, MD, USA
George Massenburg, Massenburg Design Works Nashville, TN, USA
Konrad Strauss, Jacobs School of Music, Indiana University,IN, USA

The last 10 years have been an exciting time for the audio industry, the explosion of new technology and media has created a myriad of opportunities for audio service providers. However, as audio, video, and web oriented media production converge into a single field, many audio professionals find that they must learn new skills to remain competitive and diversify their service offerings. This workshop will bring together professionals and educators from a variety of backgrounds to discuss options for audio professionals seeking to learn new skills, further their education, and incorporate new technologies into their production services.

Monday, October 12 **9:00 am** **Room 1E02**
Standards Committee Meeting: AESSC Plenary

Session P19 **Monday, October 12**
10:00 am – 11:30 am **Foyer 1E**

POSTERS: ARRAYS

10:00 am

P19-1 Control of Acoustic Radiation Pattern in a Dual-Dipole Array—*Mincheol Shin, Philip Nelson*, University of Southampton, Highfield, Southampton, UK

This paper proposes a control strategy for generating various acoustic radiation patterns associated with a "personalized sound field" using an acoustic source array. A dual-dipole array, which models each loudspeaker as an imperfect dipole and a source signal control algorithm, which effectively considers both energy difference maximization and radiation efficiency are introduced in order to obtain better radiation patterns for generating the desired sound field. With the proposed dual dipole array and control algorithm, a wide controllability from a comparatively low to a high frequency range is obtained, although the array size is small enough to implement in mobile applications. Conceptually novel control strategy proposed enables to cancel out the backward radiation efficiently. The performance of the dual dipole source array is verified by using computer simulations.
Convention Paper 7949

10:00 am

P19-2 A Novel Beam-Forming Loudspeaker System Using Digitally Driven Speaker System—*Kyosuke Watanabe, Akira Yasuda, Hajime Ohtani, Ryota Suzuki, Naoto Shinkawa, Tomohiro Tsuchiya, Kenzo Tsuihiji*, Hosei University, Koganei, Tokyo, Japan

In this paper we propose a beam-forming loudspeaker based on digitally direct driven loudspeaker system (digital-SP); the proposed speaker employs multi-bit delta-sigma modulation in addition to a line speaker array and a delay circuit. The proposed speaker can be realized only by D flip-flops and digital-SP. Delay elements are introduced between the mismatch shaper and sub-loudspeakers, and beam-forming is realized without degradation of the noise shaping performance of the multi-bit DSM. By using a small amount of additional hardware, we can easily control the sound direction. If a beamforming loudspeaker can be driven digitally, all processes can be performed digitally without the use of analog components such as power amplifiers, and a small, light, high-quality speaker system can be realized. The prototype is constructed using an FPGA, CMOS drivers, and a line speaker array. The effectiveness has been confirmed by using measurement data. A gain of 8 dB or more is measured relative to the normal digital speaker system. An attenuation of 14.7 dB for 40° direction is measured.
Convention Paper 7950

10:00 am

P19-3 Speaker Array System Based on Equalization Method with a Quiet Zone—*Soonho Baek,*[1]

Myung-Suk Song,[1] Seok-Pil Lee,[2] Hong-Goo Kang[1]
[1]Yonsei University, Seoul, Korea
[2]Korea Electronics Technology Institute, Seongnam, Korea

This paper proposes an equalization-based loudspeaker array system to form a consistent sound spot to listeners under reverberant environment. To overcome the poor sound quality of conventional beamforming methods in a reverberant environment, the proposed method designs a novel criterion to reproduce as close as possible sound to the original source at the target point as well as to make null at specified points located in the quiet zone. Simulation results with a 16-channel loudspeaker array system confirm the superiority of the proposed method. In addition, we also verify that the sound pressure level of the quiet zone depends on the number and the area of quiet points.
Convention Paper 7951

10:00 am

P19-4 On the Secondary Source Type Mismatch in Wave Field Synthesis Employing Circular Distributions of Loudspeakers—*Jens Ahrens, Sascha Spors*, Deutsche Telekom Laboratories, Techniche Universität Berlin, Berlin, Germany

The theory of wave field synthesis has been formulated for linear and planar arrays of loudspeakers but has been found to be also applicable with arbitrary convex loudspeaker contours with acceptable error. The main source of error results from the fact that the required properties of the employed loudspeakers are dictated by the Neumann Green's function of the array geometry under consideration. For nonlinear and nonplanar arrays a systematic error arises that is a result of the mismatch between the spatio-temporal transfer function of the loudspeakers and the Neumann Green's function of the loudspeaker contour under consideration. We investigate this secondary source type mismatch for the case of circular distributions of loudspeakers.
Convention Paper 7952

10:00 am

P19-5 A Configurable Microphone Array with Acoustically Transparent Omnidirectional Elements—*Jonathan S. Abel,*[1] *Nicholas J. Bryan,*[1] *Travis Skare,*[1] *Miriam Kolar,*[1] *Patty Huang,*[1] *Darius Mostowfi,*[2] *Julius O. Smith III*[1]
[1]Stanford University, Stanford, CA, USA
[2]Countryman Associates, Inc., Menlo Park, CA, USA

An acoustically transparent, configurable microphone array with omnidirectional elements, designed for room acoustics analysis and synthesis and archaeological acoustics applications, is presented. Omnidirectional microphone elements with 2 mm-diameter capsules and 1 mm-diameter wire mounts produce a nearly acoustically transparent array, and provide a simplified mathematical framework for processing measured signals. The wire mounts are fitted onto a 1.6 cm-diameter tube forming the microphone stand, with the microphones arranged above the tube so that acoustic energy can propagate freely across the array. The wire microphone mounts have some flexibility, and the array may be configured. Detachable arms with small speakers are used to estimate the element positions with an accuracy better than the 2 mm microphone diameter.
Convention Paper 7953
[Paper presented by Nicholas Bryan]

10:00 am

P19-6 Microphone Array Synthetic Reconfiguration—*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 methods for processing signals recorded at a microphone array so as to estimate the signals that would have appeared at the elements of a different, colocated microphone array, i.e., "translating" measurements made at one microphone array to those hypothetically appearing at another array. Two approaches are proposed, a nonparametric method in which a fixed, low-sidelobe beamformer applied to the "source" array drives virtual sources rendered on the "target" array, and a parametric technique in which constrained beamformers are used to estimate source directions, with the sources extracted and rendered to the estimated directions. Finally, a hybrid method is proposed, which combines both approaches so that the extracted point sources and residual can be separately rendered. Experimental results using an array of 2 mm-diameter microphones and human HRTFs are reported as a simple example.
Convention Paper 7954

10:00 am

P19-7 Design and Optimization of High Directivity Waveguide for Vertical Array—*Mario Di Cola,*[1] *Dario Cinanni,*[2] *Andrea Manzini,*[2] *Tommaso Nizzoli,*[2] *Daniele Ponteggia*[3]
[1]Audio Labs Systems, Casoli, CH, Italy
[2]18 Sound - Division of A.E.B, Srl, Cavriago, RE, Italy
[3]Studio Ponteggia, Temi, TR, Italy

Vertically arrayed loudspeaker systems have become widely used for several applications: concert sound, large scale systems, corporate events, and so on. In this kind of system the design of a proper acoustic waveguide is a key point for the system's performances. An acoustic waveguide, properly designed for this purpose, should be optimized for several features at the same time: acoustic loading properties, proper driver-throat matching, minimum internal reflection, low distortion, and, most of all, proper wavefront curvature optimization for good arrayability. An example of a practical approach to the design, dimensioning, and optimization of acoustic waveguide will be shown through loudspeaker system designing principles together with computer simulations and measured final results.
Convention paper 7955

Session P20 — **Monday, October 12**
10:30 am – 1:00 pm — **Room 1E07**

LOUDSPEAKERS IN ROOMS

Chair: **Sunil Bharitkar**, Audyssey Labs/USC, Los Angeles, CA, USA

10:30 am

P20-1 Investigation of Bonello Criteria for Use in Small Room Acoustics—*Todd Welti*, Harman International Industries Inc., Northridge, CA, USA

The Bonello Criteria are a set of conditions that are an attempt to use rectangular room dimensions as a general predictor of room modal response quality. Though intuitively satisfying, and often used in room design, the Bonello Criteria make certain assumptions that are almost never met in listening rooms, and the approach has never been systematically validated. An investigation using a computer model and a large number of possible room dimensions was made to see if meeting the Bonello Criteria should result in improved low frequency acoustical responses. Overall, the Bonello Criteria correlates only weakly to the Variance of Spatial Average and there is no correlation to Mean Spatial Variance.
Convention Paper 7849

11:00 am

P20-2 Subwoofers in Rooms: Effect of Absorptive and Resonant Room Structures—*Juha Backman*, Nokia Corporation, Espoo, Finland

The room-loudspeaker interaction at low frequencies, where individual modes can be easily identified, needs careful consideration when flat response and controlled spatial distribution are desired. The methods for controlling low frequency response are loudspeaker placement, use of multiple subwoofers, use of absorptive materials, and specific to low-frequency acoustics, use of resonators. The effects of various types of resonators and absorptive surfaces are computed using FEM for single and multiple subwoofer configurations in symmetrical and asymmetrical rooms, indicating that taking both the symmetry of the mode to be controlled and the loudspeaker placement into account when placing the resonators and/or absorbers is needed for optimal results.
Convention Paper 7957

11:30 am

P20-3 In Situ Measurements of Acoustic Absorption Coefficients Using the Surface Pressure Method—*Scott Mallais, John Vanderkooy*, University of Waterloo, Waterloo, Ontario, Canada

This paper revisits a method for determining the acoustic reflection factor by use of two pressure measurements: one at a surface under study and the other at a rigid surface in the same location of a room. The rigid surface is approximated in situ by placing a steel sheet in front of the surface under study. Measurements are made with and without the sheet at the same location. The ratio of these measurements is used to determine the acoustic reflection factor of the surface. The principle and limitations of this method are discussed, and experimental results will be given for a rigid surface, a resonant surface, and an absorptive surface, measured in different environments.
Convention Paper 7958

12:00 noon

P20-4 The Challenge to Find the Optimum Radiation Pattern and Placement of Stereo Loudspeakers in a Room for the Creation of Phantom Sources and Simultaneous Masking of Real Sources—*Siegfried Linkwitz*, Linkwitz Lab, Corte Madera, CA, USA

Stereo sound reproduction relies upon the creation of an illusion. Ideally the two loudspeakers and the room disappear, leaving only a phantom acoustic scene to be listened to. The polar frequency response of a loudspeaker determines the angular distribution of room reflections and their spectral content. The placement of the loudspeakers relative to the room surfaces determines the initial delay of the reflections. Together they affect the formation of phantom sources. A proven loudspeaker and room configuration is proposed as starting point for listening tests to determine the optimum loudspeaker radiation pattern. It is an invitation to extend our understanding of the psychoacoustic processes that are involved with stereo listening in a room and to replace anecdotal with scientific evidence.
Convention Paper 7959

12:30 pm

P20-5 The Subjective and Objective Evaluation of Room Correction Products—*Sean E. Olive, John Jackson, Allan Devantier, David Hunt*, Harman International, Northridge, CA, USA

A panel of eight trained listeners gave comparative ratings for five different room correction products based on overall preference and spectral balance. The same loudspeaker/subwoofer without correction was included as a hidden anchor. The results show significant differences among the room correction products in terms of listener preference and perceived spectral balance. The subjective ratings are largely explained by a combination of anechoic and in-room frequency response measurements made on the combined acoustic response of the room correction/loudspeaker.
Convention Paper 7960

Workshop 18 — **Monday, October 12**
11:00 am – 1:00 pm — **Room 1E15**

INTELLIGENT DIGITAL AUDIO EFFECTS

Chair: **Christian Uhle**, Fraunhofer IIS, ERlangen, Germany

Panelists: *Christof Faller*, Illusonic LLC, Lausanne, Switzerland
Joshua Reiss, Queen Mary University of London, London, UK

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Udo Zölzer, Helmut Schmidt University, Hamburg, Germany

Intelligent Digital Audio Effects (I-DAFx) process audio signals in a signal-adaptive way by using a high-level analysis of the input. Examples are the control of time parameters in tremolos, auto-wahs and vibrato effects using beat tracking, the application of blind source separation techniques to upmixing for multichannel reproduction, and signal-adaptive audio effects using feature extraction. Tools for automated live mixing have been developed recently that analyze the multichannel input of a mixer to control the effects, leveling, and panning. These and other techniques are in the scope of this workshop. It presents an overview of I-DAFx and demonstrations of practical implementations with sound examples.

Live Sound Seminar 12 **Monday, October 12**
11:00 am – 12:45 pm **Room 1E09**

AC POWER AND GROUNDING

Moderator: **Bruce C. Olson**, Olson Sound Design, Brooklyn Park, MN, USA

Panelists: *Paul Garrity*, Auerback Pollock Friedlander, New York, NY, USA
Dave Stevens, Cirque du Soleil, Las Vegas, NV, USA
Bill Whitlock, Jensen Transformers, Chatsworth, CA, USA

There is a lot of misinformation about what is needed for AC power for gigs. Much of it has to do with life-threatening advice. This panel will discuss how to provide AC power properly so that you do not kill performers or technicians. Oh, and we can also do it safely without causing noise problems. The only question is, why doesn't everybody know this!

We'll start at the outlet and make our way back to the source of power, while covering systems from a couple of boxes on sticks up to multiple stages in ballrooms, road houses, and event centers. We will discuss large scale installed systems, including multiple transformers and company switches, service types, generator sets, 1ph, 3ph, 240/120 208/120, and explain what all the lingo means. Get the latest information on grounding and typical configurations by this panel of industry veterans.

Special Event
AUDIO EAR TRAINING: LEARNING THE MOST ESSENTIAL AUDIO SKILL!
Monday, October 12, 11:00 am – 12:30 pm
Room 1E11

Moderator: **David Moulton**

Developing the aural skills needed to quickly and accurately identify audio qualities by ear alone is one of the most productive educational efforts a student can make in the field of audio today. Forty years ago, David Moulton adapted music ear training techniques used in music conservatories to teach his first audio students. He has implemented versions of this in all of the colleges he has taught at, as well as for the National Public Radio Music Recording Workshops in the 1980s. In 1993, he authored *Golden Ears*, a stand-alone CD course of audio ear training. In this tutorial, Dave will share some of the things he has learned along the way about how we hear, what it means, and how we can use it for fun and profit.

Exhibitor Seminar **Monday, October 12**
11:00 am – 1:00 pm **Room 1E17**

RENKUS-HEINZ

Speech Intelligibility Prediction with EASE 4.3

Presenter: **Stefan Feistel**

The seminar will give an overview of the new features in EASE 4.3 and discuss several aspects of speech intelligibility theory, measurement, and modeling. It will focus on the various methods EASE 4.3 uses to predict STI in accordance with IEC 60268-16 and their advantages and limitations.

Broadcast/Media Streaming Session 14
Monday, October 12 **11:30 am – 1:00 pm**
Room 1E08

STREAM PLAYBACK AND DISTRIBUTION

Chair: **Ray Archie**, CBS Radio

Presenters: *Majd Naciri*, Intel
Ben Terrell, Reciva
Jean-Francois Gadoury, Stream the World
Harry Johnson, vTuner

With radio broadcasting, encoding and stream delivery are key factors affecting a broadcaster's distribution strategy. In today's environment, all "connected" devices are potential "radios." These devices range from Internet radios to mobile phones to IP-enabled televisions/set-top-boxes to MOBLIN-powered Mobile Internet Devices. This session will focus on presentations of various solutions to these new challenges and will conclude with a panelist discussion about strategy and the future this fast changing landscape. Quality of Service and current problems as well as possible solutions will be examined.

Special Event
LUNCHTIME KEYNOTE:
ASHLEY KAHN
Monday, October 12, 1:00 pm – 2:00 pm
Room 1E09

Kind of Blue

At 50, *Kind of Blue* is many things: the best-selling classic jazz album of all time. The best-known album by today's most listened-to jazz musician Miles Davis. The one jazz album owned by a majority of non-jazz-focused music fans. The only jazz album to appear consistently at the top of greatest-album lists charted by rock, R&B, and pop music publications. Yet besides the session masters' one assembled reel, one safety with minimal studio dialog, and a few black-and-white photos, not much survives from the two historic 1959 sessions that produced *Kind of Blue*. Using PowerPoint images, audio examples of alternate takes, studio chatter, and examples of how the 3-track technology of the time, Ashley Kahn, author of the bestselling book *Kind of Blue: The Making of the Miles Davis Masterpiece* will present an informative and entertaining fly-on-the-wall perspective on the creation of Miles Davis's classic recording.

Ashley Kahn is an author, music journalist, and radio producer whose voice is often heard on National Public Radio's "Morning Edition," and whose books include

Kind of Blue: The Making of the Miles Davis Masterpiece; The House That Trane Built: The Story of Impulse Records; and *A Love Supreme: The Story of John Coltrane's Signature Album*. He also teaches courses on music history at the Clive Davis Department of Recorded Music at New York University. During a thirty-year career in the music business, Kahn has served as a music editor at VH1, the primary editor of *Rolling Stone: The Seventies* (Little, Brown), a deejay on a variety of radio stations, and—for a ten-year stint—tour manager for a multitude of music groups, including: Paul Simon, Peter Gabriel, Ladysmith Black Mambazo, Debbie Harry & the Jazz Passengers, and Britney Spears. He is currently working on a 70-year history of the well-known Blue Note jazz label, titled *Somethin' Else: The Story of Blue Note Records and the Birth of Modern Jazz*.

Tutorial 6 **Monday, October 12**
1:30 pm – 3:30 pm **Room 1E07**

PARAMETRIC DIGITAL REVERBERATION

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

This tutorial is intended for audio algorithm designers and students interested in the design and applications of digital reverberation algorithms. Artificial reverberation has long been an essential tool in the studio and is a critical component of modern desktop audio workstations and interactive audio rendering systems for gaming, virtual reality, and telepresence. Feedback delay networks yield computationally efficient tunable reverberators that can reproduce natural room reverberation decays with arbitrary fidelity, configurable for various spatial audio encoding and reproduction systems and formats.

This session will include a review of early digital reverberation developments and of the physical, perceptual, and signal properties of room reverberation; a general and rational method for designing digital reverberators, and a discussion of reverberation network topologies; flexible and accurate parametric tuning of the reverberation decay time; analysis and simulation of existing reverberation decays; the Energy Decay Relief of a reverberation response and its interpretation; practical applications in audio production and game audio rendering, illustrated by live demonstrations.

Games Audio 9 **Monday, October 12**
1:30 pm – 2:30 pm **Room 1E16**

INTERACTIVE MUSIC TECHNIQUES FOR GAMES

Presenter: **Scott Selfon**, Microsoft Corporation, Redmond, WA, USA

With music for current console and PC games consisting primarily of prerecorded music tracks, how does a composer or audio programmer construct a believably interactive score that reflects and emphasizes the state of the game? This tutorial will quantify some of the techniques in use by games today to construct dynamic and interactive musical scores, delivering musically believable and appropriate scores from finite recorded materials. Additionally, the talk asks and attempts to answer the question of whether and how real-time generated music (via MIDI or other similar protocols) can be a viable technique for presenting compelling game music.

Historical Event
RECORDING THE JAZZ BIG BANDS
Monday, October 12, 1:30 pm – 3:30 pm
Room 1E11

Moderator: **Robert Auld**, AuldWorks

Robert Auld of AuldWorks will trace the rise of the big bands in relation to the development of electrical audio recording. For 35 years, these mutually beneficial art forms produced superb recordings that exemplify the "golden age" of early stereo. A professional trumpet player and live sound engineer for the big band jazz concerts at the Manhattan School of Music, Robert Auld has worked with artists ranging from Wynton Marsalis to the Vanguard Orchestra. In 1997 Mr. Auld published *The Art of Recording The Big Band*, a historical and critical survey based on his 25 years of experience on stage and on the console. He has now rethought and expanded this treatise into a full multi-media presentation covering the period from the 1920s to the present.

Workshop 19 **Monday, October 12**
2:00 pm – 4:00 pm **Room 1E15**

AUDIO MYTHS—DEFINING WHAT AFFECTS AUDIO REPRODUCTION

Chair: **Ethan Winer**, RealTraps, New Milford, CT, USA

Panelists: *Jason Bradley*, Intel Corporation, Hillsboro, CO, USA
Poppy Crum, Johns Hopkins School of Medicine, Baltimore, MD, USA
James Johnston, DTS, Inc., Kirkland, WA, USA

Human auditory memory and perception are frail, and expectation bias and placebo effect are stronger than many care to admit. The result is endless arguments over basic scientific principles that have been understood fully for more than fifty years—the value of ultra-high sample rates and bit depths, the importance of dither, clock jitter, and ultra-low distortion, and so forth. If you move your head even one inch, the frequency response changes substantially due to acoustic comb filtering. Masking makes it difficult to hear artifacts even 40 dB below the music, yet some people are convinced they can hear artifacts 100 dB down or lower. Therefore, the purpose of this workshop is to identify what really matters with audio reproduction and what does not.

Workshop 20 **Monday, October 12**
2:00 pm – 4:00 pm **Room 1E08**

TURN IT DOWN! CONSEQUENCES OF THE EVER-ESCALATING LOUDNESS WARS

Chair: **Martin Walsh**, DTS Inc., Scotts Valley, CA, USA

Panelists: *Bob Katz*, Digital Domain, Altamonte Springs, FL, USA
Bob Ludwig, Gateway Mastering and DVD, Portland, ME, USA
Thomas Lund, TC Electronic A/S, Risskov, Denmark
Susan Rogers, Berklee College of Music, Boston, MA, USA

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The adage that "louder is better" has been put to the ultimate test in recent years. Modern recordings have gradually been mastered louder and louder over the past twenty or so years using techniques such as aggressive dynamic range compression. In some of the latest album masters, this ever-increasing loudness "arms race" is reaching its limit, whereby the average level is almost the same as the peak level. This leads to audible distortion and listener fatigue. In this workshop we will describe the history of the "loudness wars" along with their impact on modern mastering. We will also provide examples of extreme loudness mastering (or re-mastering). A tutorial will be provided by an invited "master" mastering engineer on how one can provide maximum loudness without incurring audible distortion. Finally, we open the discussion to the floor with a variety of topics relating to today's loudness wars and the possibilities of brokering a peace process in the future.

Special Event
SOCIETY OF BROADCAST ENGINEERS CERTIFICATION EXAMS
Monday, October 12, 2:00 pm – 5:00 pm
Room 1E02

The Society of Broadcast Engineers will offer SBE certification exams to AES Convention attendees. SBE Certification is the recognized benchmark of technical ability in broadcasting. Exams for any of the SBE certification levels will be available. Pre-registration through the SBE National office is preferred, but on-site registration will be available at the convention. Fees can be paid via cash, check or credit card (Visa, Master Card or American Express). SBE membership is not required to hold SBE Certification. Please arrive at least ten minutes prior to the exam period.

Master Class 6 **Monday, October 12**
2:30 pm – 4:30 pm **Room 1E16**

BILL WHITLOCK

Presenter: **Bill Whitlock**

The Design of High-Performance Balanced Audio Interfaces

High signal-to-noise ratio is an important goal for most audio systems. However, AC power connections unavoidably create ground voltage differences, magnetic fields, and electric fields. Balanced interfaces, in theory, are totally immune to such interference. For 50 years, virtually all audio equipment used transformers at its balanced inputs and outputs. Their high noise rejection was taken for granted and the reason for it all but forgotten. The transformer's extremely high common-mode impedance—about a thousand times that of its solid-state "equivalents"—is the reason. Traditional input stages will be discussed and compared. A novel IC that compares favorably to the best transformers will be described. Widespread misunderstanding of the meaning of "balanced" as well as the underlying theory has resulted in all-too-common design mistakes in modern equipment and seriously flawed testing methods. Therefore, noise rejection in today's real-world systems is often inadequate or marginal. Other topics will include tradeoffs in output stage design, effects of non-ideal cables, and the "pin 1 problem."

Live Sound Seminar 13 **Monday, October 12**
2:30 pm – 4:45 pm **Room 1E09**

INNOVATIONS IN LIVE SOUND

Moderator: **Ken Lopez**, University of Southern California, Los Angeles, CA, USA

Panelists: *Bill Hanley*, Hanley Audio Systems
Albert Leccese, Audio Analysts
John Monitto, Meyer Sound Laboratories
David Scheirman, JBL Professional
Robert Scovil, Digidesign

New techniques and products are often driven by changes in need and available technology. Today's sound professional has a myriad of products to choose from. That wasn't always the case. What drove the creation of today's products? What will drive the products of tomorrow? Sometimes a look back is the best way to get a peek ahead. A panel of industry pioneers and trailblazers will take a look back at past live sound innovations with an emphasis on the needs and constraints that drove their development and adoption.

Student/Career Development Event
STUDENT DELEGATE ASSEMBLY MEETING —PART 2
Monday, October 12, 2:30 pm – 4:00 pm
Room 1E06

The closing meeting of the SDA will host the election of a new vice chair. Votes will be cast by a designated representative from each recognized AES student section or academic institution in the North/Latin America Regions present at the meeting. Judges' comments and awards will be presented for the Recording and Design Competitions. Plans for future student activities at local, regional, and international levels will be summarized and discussed among those present at the meeting.

Workshop 21 **Monday, October 12**
4:00 pm – 6:00 pm **Room 1E11**

PIMP YOUR MIX

Co-chairs: **Bob Brockmann**
Ryan West

Grammy winning mixers Bassy Bob Brockmann (Christina Aguilera, Babyface, Fugees) and Ryan West (Rihanna, Kanye, John Legend) bring a popular feature of their Elements of Mixing seminar, "Pimp your Mix," to the AES. Ryan and Bassy take a production that has already been mixed by an upcoming producer and tear it apart and put it back together using state-of-the-art and vintage analog gear and some of their fave plug-ins to add luster, drama, impact, and clarity to the mix. Ryan and Bassy demonstrate some of their tips on how to make the vocal shine, how to organize the bass frequencies, and how to use balancing, EQ, and reverb to create a sense of space and clarity in the mix. Attendees are encouraged to ask questions during the makeover process, and Ryan and Bassy will do a Q and A about the mix after the print.

Workshop 22 **Monday, October 8**
4:00 pm – 6:00 pm **Room 1E08**

MICROPHONES FOR EXTREME ENVIRONMENTS

Chair: **David Josephson**, Josephson Engineering, Santa Cruz, CA, USA

Panelists: *D. J. Atkinson*, NTIA Institute for Telecommunications Sciences, Boulder, CO, USA
James Johnston, DTS Inc., Kirkland, WA, USA
Kenneth Link, Dauntless Fire Company, Glendale, CA, USA
Ashley Strickland, Plainfield Fire Department, Plainfield, IN, USA
Don Wright, Glendale Fire Department, Glendale, CA, USA

Microphones are essential components in many places outside broadcast and studio recording. The quality and reliability of a microphone can be a matter of life and death rather than just a question of flavor of sound. Firefighters in a burning house have to rely on clear communication system and face new challenges with the use of low bitrate coders in their radios. The same need applies to divers, astronauts, pilots, tank drivers, bike riders, etc. However, many microphone designs now in use work poorly in high noise environments and cannot be depended on to play the role of being a lifeline. Why is that? Couldn't this be changed? In this workshop a number of people who have studied the problem faced by emergency responders try to identify the special requirements for microphones used under these conditions.

Tutorial 7 **Monday, October 12**
4:00 pm – 5:00 pm **Room 1E07**

THE MANIFOLD JOYS, USES, AND MISUSES OF POLYNOMIAL MAPPING FUNCTIONS IN SIGNAL PROCESSING

Presenter: **Robert Bristow-Johnson**, audioImagination, Burlington, VT, USA

In digital audio signal processing, we "do math" upon samples of audio signals, but also with parameters associated with audio signals. These parameters can perhaps be loosely divided into two groups: numbers that are important and recognizable to the user of the DSP algorithm (loudness in dB, frequency in Hz, pitch or bandwidth in octaves, RT60 in seconds), and numbers that are of direct concern to the DSP algorithm itself (coefficients, thresholds, displacements in samples). There are relationships between these groups of parameters that involve mysteriously designed transcendental functions that may not be available to the processor being used in the implementation. This is about how to implement such mappings without the use of the ubiquitous look-up table (LUT) by fitting polynomials to functions, where it's useful in audio DSP, a little about the different fitting criteria, and methods of fitting.

Games Audio 10 **Monday, October 12**
4:00 pm – 5:30 pm **Room 1E15**

TECHNIQUES OF AUDIO LOCALIZATION FOR GAMES

Chair: **Fabio Minazzi**, Binari Sonori Srl

Panelists: *Hope Dippel*, Sony Computer Entertainment America
Francesco Zambon, Binari Sonori Srl

The production of game soundtracks shares many technologies and techniques with the production of audio for other media. At the same time, videogames are software products, with nonlinear and dynamic behaviors, and these features of the media affect the way audio is produced and localized. This session deals with the full cycle of the dialog production for videogames and is divided into two parts: Part 1 describes the original language soundtrack production; Part 2 is devoted to soundtrack localization with special attention to the differences between the dubbing of a film and the localization of game-audio.