AES 131st Convention Program

October 20 – 23, 2011

Jacob K. Javits Convention Center, New York, NY, USA

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

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

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

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

(a) The paper was accepted for presentation at the AES 131st Convention.

(b) The first author was a student when the work was conducted and the manuscript prepared.

(c) The student author's affiliation listed in the manuscript is an accredited educational institution.

(d) The student will deliver the lecture or poster presentation at the Convention.

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The Winner of the 131th AES Convention Student Paper Award is:

The Impact of Producers' Comments and Musicians' Self-Evaluation on Performance during Recording Sessions—Amandine Pras, Catherine Guastavino, McGill University, Montreal, Quebec, Canada
Convention Paper 8579

To be presented on Sunday, October 23, in Session 25 —Auditory Perception

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Session P1 **Thursday, Oct. 20**
9:00 am – 11:00 am **Room 1E09**

ROOM ACOUSTICS

Chair: **Peter Mapp**, Peter Mapp Associates, Colchester, UK

9:00 am

P1-1 **New Thoughts on Active Acoustic Absorbers —***John Vanderkooy*, University of Waterloo, Waterloo, Ontario, Canada

This paper continues an earlier exploration of using full-range loudspeakers as both acoustic sources and sinks, in an attempt to reduce the room decay time for bass frequencies. We develop the theory for a point active absorber immersed in the acoustic source field from a point source. This would apply to normal loudspeakers used as either sources or absorbers at frequencies below about 300 Hz, where they act as points. The result extends the theory of Nelson and Elliott for a point absorber interacting with a plane wave. An extra term occurs that has little net effect when averaged over frequency or distance. In rooms such cancellation occurs due to the varying distances from all the source images to the absorber. Impulse responses in several small rooms were measured from a source and an absorber loudspeaker to both a few listening microphones and a microphone mounted at the absorber. The efficacy of the active absorber is assessed and the results are enigmatic.
Convention Paper 8458

9:30 am

P1-2 **Investigations of Room Acoustics with a Spherical Microphone Array—***Samuel W. Clapp,*[1] *Anne E. Guthrie,*[1,2] *Jonas Braasch,*[1] *Ning Xiang*[1]
[1]Rensselear Polytechnic Institute, Troy, NY, USA
[2]Arup, New York, NY, USA

Most room acoustic parameters are calculated with data from omni-directional or figure-of-eight microphones. Using a spherical microphone array to record room impulse responses can open up several new areas of inquiry. It can yield much more information about the spatial characteristics of the sound field, including the diffuseness of the sound field and the directions of individual reflections. A 16-channel microphone array was designed, built, and tested with both simulations and simple, controlled sound events. Room impulse responses were then measured in reverberant rooms used for music from stage and audience positions, and the results were analyzed using beamforming techniques to determine spatial information about the sound field.
Convention Paper 8459

10:00 am

P1-3 **Room Acoustics Using a 2.5 Dimensional Approach with Damping Included—***Patrick Macey*, PACSYS Limited, Nottingham, UK

Cavity modes of a finite bounded region with rigid boundaries can be used to compute the ➡

steady state harmonic response for point source excitation. In cuboid domains this is straightforward. In general regions, determining a set of orthonormal modes is more difficult. Previous work showed that for rooms of constant height, 3-D modes can be computed from the cross section modes, and this used for a fast solution. This approach used modal damping. More realistic damping associated with wall areas could be included using a damped eigenvalue calculation of the cross section modes. This is restrictive on damping formulations. An alternative non-modal approach, using a trigonometric expansion through the height is proposed. This is still faster than 3-D FEM.
Convention Paper 8460

10:30 am

P1-4 Accurate Acoustic Modeling of Small Rooms —*Holger Schmalle*,[1] *Dirk Noy*,[2] *Stefan Feistel*,[1] *Gabriel Hauser*,[2] *Wolfgang Ahnert*,[1] *John Storyk*[2]
[1]AFMG Ahnert Feistel Media Group, Berlin, Germany
[2]WSDG Walters-Storyk Design Group, Basal, Switzerland

Modeling of sound reinforcement systems and room acoustics in large and medium-size venues has become a standard in the audio industry. However, acoustic modeling of small rooms has not yet evolved into a widely accepted concept, mainly because of the unavailable tool set. This work introduces a practical and accurate software-based approach for simulating the acoustic properties of studio rooms based on FEM. A detailed case study is presented and modeling results are compared with measurements. It is shown that results match within given uncertainties. Also, it is indicated how the simulation software can be enhanced to optimize loudspeaker locations and place absorbers and diffusers in order to improve the acoustic quality of the space and thus the listening experience.
Convention Paper 8457

Session P2 — Thursday, Oct. 20
9:00 am – 11:00 am — Room 1E07

RECORDING AND SOUND PRODUCTION

Chair: **Justin Paterson**, University of West London, London, UK

9:00 am

P2-1 Computer Assisted Microphone Array Design (CAMAD)—*Michael Williams*, Freelance Sound Recording Engineer and Lecturer, Sounds of Scotland, Le Perreux sur Marne, France

The basic aim of Microphone Array Design is to create microphone array recording systems with smooth seamless or “Critically Linked” segment coverage of the surround sound field. Each configuration must take into account the interaction of the many design parameters, with the specific coverage of each segment that is required. The difficulty in the manipulation of these many parameters is one of the major obstacles in developing a wide range of microphone arrays that meet the needs of each particular sound recording environment. This paper will outline the various parameters that need to be taken into consideration and explain the basic approach to developing MATLAB-based software that gives a clear and unambiguous display of all the salient characteristics needed to achieve a stable and reliable microphone array, no matter the number of channels involved, or the type of directivity pattern chosen for each microphone.
Convention Paper 8461

9:30 am

P2-2 In Situ Measurements of the Concert Grand Piano—*Brett Leonard*,[1,2] *Grzegorz Sikora*,[1] *Martha de Francisco*[1,2]
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canada

An in situ approach to the acoustical study of the grand piano is presented in which the instrument is coupled with a typical, reflective recording space. By using accurate, tightly controlled automated playback of expertly performed material, a small number of high-quality transducers are employed to capture more than 1300 spatially distributed data points in the process known as acoustic space sampling (AcSS). The AcSS measurement task is performed on two pianos in two unique recording environments. The data is analyzed using accepted acoustic metrics and psychoacoustic predictors. It is shown that certain spatial areas containing salient physical and psychoacoustic measures are highly correlated recording engineer preference.
Convention Paper 8462

10:00 am

P2-3 Beyond Surround Sound—Creation, Coding and Reproduction of 3-D Audio Soundtracks —*Jean-Marc Jot*,[1] *Zoran Fejzo*[2]
[1]DTS, Inc., Scotts Valley, CA, USA
[2]DTS, Inc., Calabasas, CA, USA

We propose a flexible and practical end-to-end solution for creating, encoding, transmitting, decoding, and reproducing spatial audio soundtracks. The soundtrack encoding format is compatible with legacy surround-sound decoders, while enabling the representation of a three-dimensional audio scene, irrespective of the listener’s playback system configuration. It allows for encoding one or more selected audio objects that can be rendered with optimal fidelity and interactively in any target spatial audio format (existing or future). The transmission or storage data rate and the decoder complexity are scalable at delivery time. A 3-D audio soundtrack may thus be produced once and transmitted or broadcast and reproduced as faithfully as possible on the broadest range of target devices.
Convention Paper 8463

10:30 am

P2-4 The Effects of Multiple Arrivals on the Intelligibility of Reinforced Speech—*Timothy J. Ryan*,[1] *Richard King*,[1] *Jonas Braasch*,[2] *William L. Martens*[3]

[1]McGill University, Montreal, Quebec, Canada
[2]Rensselaer Polytechnic University, Troy, NY, USA
[3]University of Sydney, Sydney, NSW, Australia

The effects of multiple arrivals on the intelligibility of speech produced by live-sound reinforcement systems are examined. The intent is to determine if correlations exist between the manipulation of sound system optimization parameters and the subjective attribute speech intelligibility. Investigated variables are signal-to-noise ratio (SNR), delay time between signals arriving from multiple elements of a loudspeaker array, and array type and geometry. Intelligibility scores were obtained through subjective evaluation of binaural recordings, reproduced via headphone, using the Modified Rhyme Test.
Convention Paper 8464

Workshop 1 **Thursday, Oct. 20**
9:00 am – 10:30 am **Room 1E11**

CANCELLED

Broadcast/Media Streaming Session 1
Thursday, October 20 **9:00 am – 10:30 am**
Room 1E10

FACILITY DESIGN—RENOVATION AND RETROFITTING

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

Presenters: *Judy Elliot-Brown*, Rocket Science, NY
John McGowan, WNET TV
Howie Schwartz, Howard Schwartz Recording Inc., New York NY, USA

There continues to be an increasing need for facility upgrade and expansion in the broadcast production and post-production sectors. Numerous technical and social issues are fueling this, including changing delivery and production protocols (i.e., HDTV, surround audio, etc.); improved economic times with respect to media content organization, etc. Typically, facilities have had to expand or improve their audio production rooms by retrofitting an existing room or installing a new room within an existing production complex. This panel will explore specific techniques as well as examples dealing with the construction and more specific acoustic and technology issues, associated with retrofitting and renovating audio production facilities with an eye toward broadcasting production and post production standards.

Live Sound Seminar 1 **Thursday, Oct. 20**
9:00 am – 11:00 am **Room 1E14**

DSP ALGORITHMS

Chair: **Rich Frembes**, Fulcrum Acoustics

Presenters: *Bradford Benn*, Crown
Klas Dalbjörn, Lab Gruppen
Dave Gunness, Fulcrum Acoustics
Langston Holland, Soundscapes
Bennett Prescott, ADRaudio
Dana Troxel, Rane

It's not talked about often, but almost no two brands of digital loudspeaker processors use the same filter definitions. As a result, there is no such thing as a cross-platform, DSP agnostic, optimal crossover setting. This panel looks at how to create a precise set of crossover and equalization filters for the best loudspeaker performance despite so much variability in DSP filter shapes. DSP engineers and loudspeaker experts talk filter definitions, tips on DSP settings conversion and the need for a loudspeaker filter definition standard.

Game Audio Session 1 **Thursday, Oct. 20**
9:00 am – 10:30 am **Room 1E13**

EMERGING TRENDS IN AUDIO FOR GAMES

Presenters: **Michael Kelly**, Principal Audio R&D Engineer, DTS
Steve Martz, Sr. Design Engineer, THX Ltd.

This workshop looks at the current state of technology requirements for audio in game applications and discusses emerging trends and the technical requirements imposed by those trends. The workshop is presented by the Co-chairs of the AES Technical Committee on Audio for Games. The workshop also summarizes some of the topics presented at the recent 41st International Conference on Audio for Games.

Product Design Session 1 **Thursday, Oct. 20**
9:00 am – 11:00 am **Room 1E08**

THE QUIETEST LINK- BETTER NOISE AND CMRR AT LOW COST IN BALANCED CONNECTIONS

Presenter: **Douglas Self**

A balanced connection is usually considered a high-class connection compared with its unbalanced equivalent because of its proven ability to reject noise on the connection ground and other forms of interference. It is therefore rather unfortunate that the average balanced connection is some 14 dB noisier in terms of its own electronic noise. That difference is both clearly audible and hard to explain away to potential customers.

Another limitation of a balanced link is its finite Common-Mode-Rejection-Ratio, the figure that measures how well ground noise is rejected. Achieving a good CMRR is simple if you can use precision resistors but these are uneconomic for most applications.

This session looks at ways to improve both the noise performance and the CMRR of a balanced link without spending a lot of money.

Tutorial 1 **Thursday, Oct. 20**
9:30 am – 11:00 am **Room 1E12**

DELAY FX—WAIT FOR IT

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

The humble delay is the basis for a broad range of effects: comb filtering, flanging, chorus, echo, reverb, pitch shift, and more. One device, one plug-in, and you've got a vast pallet of production possibilities. Join us for this thorough discussion of the technical fundamentals and production strategies for one of the most powerful signal processes in the audio toolbox: delay.

Session P3 **Thursday, Oct. 20**
10:00 am – 11:30 am **1E Foyer**

POSTERS: TRANSDUCERS

10:00 am

P3-1 Inter- and Intra-Individual Variability in Blocked Auditory Canal Transfer Functions of Three Circum-Aural Headphones—*Florian Völk*, AG Technische Akustik, MMK, Technische Universität München, Munich, Germany

In headphone playback, different factors contribute to a deviation of the presented from the intended stimuli. Most important are the headphone transfer functions, their inter-individual differences, and the intra-individual variability due to repeated positioning on the subjects' heads. This paper gives a detailed inspection of the blocked auditory canal transfer characteristics for one specimen of each of three different circum-aural headphone models frequently used in psychoacoustics, two operating on the electrodynamic, one on the electro-static converter principle. It is shown that the variability can have considerable influence on the stimuli presented, especially in the frequency range above 6 kHz. The data indicate headphone specific variability, suggesting the need for the variability to be considered as headphone selection criterion.
Convention Paper 8465

10:00 am

P3-2 Cable Matters: Instrument Cables Affect the Frequency Response of Electric Guitars—
Rafael Cauduro Dias de Paiva,[1,2]
Henri Penttinen[1]
[1]Aalto University School of Electrical Engineering, Espoo, Finland
[2]Nokia Technology Institute INdT, Brasilia, Brazil

This paper presents analysis results of the effects an instrument cable has on the timbre of an electric guitar. In typical, well-designed audio equipment with proper impedance buffers, the effect of the cable can be considered insignificant at audio frequencies. In contrast, magnetic pickups used in electric guitars, act as resonating low-pass filters. The cable is attached to a resonating electrical circuit and hence its impedance characteristics can influence the system. The simulation and measurement results show that the capacitance of an instrument cable affects the frequency response of the system, whereas the effects of the inductance and series or parallel resistance are negligible. The largest shift in the resonant frequency for the measured cables was 1.4 kHz.
Convention Paper 8466

10:00 am

P3-3 An Approach to Small Size Direct Radiation Transducers with High SPL—*Jose Martinez*,[1] *Enrique Segovia*,[2] *Jaime Ramis*,[3] *Alejandro Espí*,[1] *Jesús Carbajo*[3]
[1]Acústica Beyma, S.L., Valencia, Spain
[2]Obras Públicas e Infraestructura Urbana, Alicante, Spain
[3]Ingeniería de Sistemas y Teoria de la Señal, San Vicent del Raspeig, Spain

This work analyses some of the issues related to small size direct radiation loudspeakers design and aims to achieve high SPL using this kind of loudspeaker. In order to reach it, large diaphragm displacements are needed. Structural dynamic behavior of the moving assembly must be emphasized. With the aid of a numerical model implemented with finite elements, it is possible to quantify the influence of changing the number of folds in the suspension (spider), the distance between spiders, and the effect of unbalanced forces inherent to the loudspeaker construction. Numerical model predictions are compared with experimental results having as reference a six-inch loudspeaker development.
Convention Paper 8467

10:00 am

P3-4 High-Order Analog Control of a Clocked Class-D Audio Amplifier with Global Feedback Using Z-Domain Methods—*Pieter Kemp*,[1] *Toit Mouton*,[1] *Bruno Putzeys*[2]
[1]University of Stellenbosch, Matieland, South Africa
[2]Hypex Electronics B.V., Groningen, The Netherlands

The design of a clocked analog controlled pulse-width modulated class-D audio amplifier with global negative feedback is presented. The analog control loop is designed in the z-domain by modeling the comparator as a sampling operation. A method is presented to improve clip recovery and ensure stability during over-modulation. Loop gain is shaped to provide a high gain across the audio band, and ripple compensation is implemented to minimize the negative effect of ripple feedback. Experimental results are presented.
Convention Paper 8468

10:00 am

P3-5 A Novel Sharp Beam-Forming Flat Panel Loudspeaker Using Digitally Driven Speaker System—*Mitsuhiro Iwaide, Akira Yasuda, Daigo Kuniyoshi, Kazuyuki Yokota, Moriyasu Yugo, Kenji Sakuta, Fumiaki Nakashima, Masayuki, Yashiro, Michitaka Yoshino*, Hosei University, Koganei, Tokyo, Japan

In this paper we propose a beam-forming loudspeaker based on a digitally direct driven speaker system (digital-SP); the proposed speaker employs multi-bit delta-sigma modulation in addition to a line speaker array with flat-panel-loudspeakers (FPLS) and a delay circuit. The proposed speaker can be realized only by D flip-flops and digital-SP. The sound direction can easily be controlled digitally. All processes can be performed digitally without the use of analog components such as power amplifiers, and a small, light, thin, high-quality beam-forming speaker system can be realized. The prototype is constructed using an FPGA, CMOS drivers, and a line FPLS array. The 20 dB attenuation for 20 degree direction is measured.
Convention Paper 8469

10:00 am

P3-6 The Influence of the Directional Radiation Performance of the Individual Speaker Modules, and Overall Array, on the Tonal Balance, Quality and Consistency of Sound Reinforcement Systems—*Akira Mochimaru, Paul Fidlin, Soichiro Hayashi, Kevin Manzolini*, Bose Corporation, Framingham, MA, USA

Room acoustic characteristics, such as reflections and reverberation, often change the performance of speaker systems in a room. It has always been challenging to maintain the tonal balance of a single speaker module when multiple modules are used to form speaker arrays. Both phenomena are unique to sound reinforcement speakers and are mainly determined by a combination of the radiation characteristics of an individual speaker module and the interactions between modules, known as arrayability. First, this study reviews the performance of conventional speaker systems. Next, the acoustic characteristics of speaker modules desired for forming an ideal speaker array are discussed. A new category of speaker system, the Progressive Directivity Array, is introduced to realize the theory as a practical solution.
Convention Paper 8470

Thursday, Oct. 20 10:00 am Room 1E05
Technical Committee Meeting on Human Factors in Audio Systems

Game Audio Session 2 Thursday, Oct. 20
10:30 am – 11:30 am Room 1E13

AUDIO SHORTS: TOOLS

Presenters: **Simon Ashby**, VP Product Strategy, Co-founder Audiokinetic
Peter "pdx" Drescher, Twittering Machine
Michael Kelly, Principal Audio R&D Engineer, DTS

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

Shorty #1: Usage and Optimization of Run-Time Convolution Reverb for Games — *Presenter:* Simon Ashby, VP Product Strategy, Co-founder at Audiokinetic
Despite the fact that the current generation of gaming platforms are really powerful, the usage of convolution reverb is just emerging and the perceptions that this technology is too expensive for games holds strong. This talk will demonstrate different techniques on how convolution reverb can be optimized to fit most game resource budgets while underlining the situations where using a standard parametric reverb may be preferable.

Shorty #2: Data Compression for Games — *Presenter:* Michael Kelly, Principal Audio R&D Engineer, DTS
Game audio developers are always trying to squeeze more data into the allotted RAM resources they are given. The presenter will discuss current audio codec's and their application for games as well as future trends.

Shorty #3: Make it Even Smaller! — *Presenter:* Peter "pdx" Drescher, Twittering Machine
When you need to make that data even smaller than any audio codec could ever assist with, you need to develop techniques for creating awesome sounding content in as little space as possible. Considered an authority on audio for mobile platforms, the presenter will divulge some of his techniques for creating a compelling audio experience in as few bits as possible.

Tutorial 2 Thursday, Oct. 20
10:45 am – 12:45 pm Room 1E11

EAR TRAINING FOR MASTERING ENGINEERS

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

This tutorial includes a comprehensive set of tools to improve ear training, focused on what the Mastering Engineer needs to do his job. Dynamic EQ, De-essing, De-woofing, and many other techniques will be shown, aiming to help audience to recognize different ranges of frequency. Mr. Mayo pioneered the art of mastering since 1992 and boasts credits in over 1,500 musical titles on vinyl, CD, DVD, and Blu-ray.

Session EB1 Thursday, Oct. 20
11:00 am – 12:00 noon Room 1E09

RECORDING/PRODUCTION

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

11:00 am

EB1-1 A New Golden Age of Recording—*Thomas T. Chen*, Stockton, CA, USA

The "Golden Age" of recording is said to be in the early 1960s and consists of recordings that were made that have great natural sound that are a treat to the ear. I believe that the room is one of the major contributors to the sound of the "Golden Age" recordings. I have developed a technique for live recording producing good sound and also a means of adding quality room to multitrack recordings.
Engineering Brief 16

11:15 am

EB1-2 What Constitutes Innovation in Music Production?—*Justin Paterson*, London College of Music, University of West London, London, UK

Innovation has often been at the core of record production, yet as production has advanced from Fred Gaisberg through the techniques of Musique Concrète to the plethora of possibilities afforded by the present digital age, the opportunities for genuine innovation might now seem limited. This notion is explored by considering the ontology of production with reference to audio examples, forming a chronological thread that highlights pieces commonly perceived as landmark innovations, their technological backdrops, and the recurrence/evolution of effect and ➥

aesthetic through successive generations of technology, and ultimately a nexus. The perception, attribution and value of "quality" is another factor, and while this is a separate subject in its own right, some discussion of this better contextualizes the topic.
Engineering Brief 17

11:30 am

EB1-3 Achieving Great Sound in the Age of Loudness Wars—*Bryan Martin*,[1] *Adrian Carr*[2]
[1]Sonosphere Mastering, Montreal, Quebec, Canada
[2]AC Mastering, Montreal, Quebec, Canada

There has been much concern and discussion about the ever increasing demand for higher and higher mix and mastering levels. Veteran mastering engineers Adrian Carr and Bryan Martin will take a new perspective to discuss techniques and processes to improve and maintain fidelity in the current "Maximum-Volume-Level" marketplace. Though this presentation is geared toward the working engineers of today, it is of particular importance to the up and coming generation. Since the level of CD's is not decreasing, the young engineer can still be aware of the advantages he has and the pitfalls he faces.
Engineering Brief 18

11:45 am

EB1-4 Phantom Powering the Modern Condenser Microphone Part II: The Effects of Load Impedance on Microphone Performance—
Mark Zaim, Audio-Technica U.S. Inc., Stow, OH, USA

This paper builds on topics discussed in the previous AES paper "Phantom Powering the Modern Condenser Microphone: A Practical Look at Conditions for Optimized Performance." It is not uncommon for microphone manufacturers to measure performance specifications using open load conditions. However, in application microphones are connected to mixing consoles that have much lower input impedances. These operating conditions can affect microphone performance and cause measurements to deviate from published open load conditions. During the previous investigation, changes were seen in several relevant microphone measurements as load impedance was changed. These observations prompted a more in-depth investigation into the effects of load impedance on microphone performance. The specific performance parameters investigated were: sensitivity, self-noise, current consumption, dynamic range, maximum sound pressure level (max SPL), signal to noise ratio (SNR) and frequency response.
Engineering Brief 19

Engineering Brief 20 has been withdrawn

Workshop 2 — Thursday, Oct.20
11:00 am – 12:30 pm — Room 1E08

CAPTURING HEIGHT IN SURROUND

Chair: **Paul Geluso**

Panelists: *Tom Ammermann*
Gary Elko
Brett Leonard
Jens Meyer
Wilfried Van Baelen
Gregor Zielinsky

While a great deal of development has gone into reproducing audio in the horizontal plane, the next step in immersive audio moves into the vertical dimension. New transmission formats, recording techniques, microphone configurations, and processing methods are needed to fill this demand. At this workshop new methods to capture and create height information for a true 3-D listening experience will be presented by a panel of music producers, engineers, and researchers. Production techniques to capture Z axis information for later reproduction in surround playback environments with height channels will be presented. Post-production methods to create 3-D ambiance including up-mix algorithms for 2.0 and 5.1 to Auro 3-D and mono to 5.1 and 8.1, up to 22.2. will be discussed as well. Recordings made using these technologies will be demonstrated during an additional workshop at NYU on Saturday.

Master Class 1 — Thursday, Oct. 20
11:00 am – 1:00 pm — Room 1E07

THE FINE LINE BETWEEN VOICING AND DESIGN

Presenters: **Casey Dowdell**
Brian Zolner

When the design brief of a digital to analog conversion system calls for performance regardless of cost, a series of known concepts can be put in place. When performance is the highest priority, the execution and fine tuning of these concepts changes the design brief and project schedule. A list of the fundamental concepts is presented along with the relevant discussion as to their necessity. Intriguingly, protracted listening during development results in progressive design changes that must occur to reach an extraordinary outcome.

Casey Dowdell, formerly of Lexicon and HP, and Brian Zolner, formerly of The Harman Specialty Group, are co-founders of Bricasti Design.

Broadcast/Media Streaming Session 2
Thursday, October 20 — 11:00 am –12:30 pm
Room 1E10

LISTENER FATIGUE AND RETENTION

Chair: **David Wilson,** CEA

Panalists: *Todd Baker*, SRS Labs
Frank Foti, Omnia
Greg Monti, Citadel Media
Greg Ogonowski, Orban
Sean Olive, Harman
Ellyn Sheffield, Towson University

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 2 **Thursday, Oct. 20**
11:00 am – 1:00 pm **Room 1E14**

COORDINATION AND BAND PLANNING OF LICENSE FREE RF SPECTRUM FOR EVENTS AND INSTALLATIONS

Chair: **Larry Estrin**, Best Audio

Presenters: *Rod Barton*, NFL Game Day Coordinator
Larry Dunn, City Theatrical
Craig Fredrickson
Jackie Green, Audio-Technica
Tom Turkington, CoachComm

The entertainment industry has a growing need for frequency coordination at every level of production from the Super Bowl to community theater and school events etc. Today an ever increasing number of manufacturers are developing and marketing new license free devices that transmit audio, video, communications, and data. Desired coverage areas range from the one room community theater to the largest stadiums and arenas in the world. The panel will discuss a number of these situations and the most popular types of devices and how they work. We will include a discussion of how, in most cases, coordination and spectrum management technologies can allow almost all devices to work without interfering with each other, in the same general atmosphere. The discussions will include current and future spectrum usage. Frequency coordination protects everyone.

Thursday, Oct. 20 **10:00 am** **Room 1E05**
Technical Committee Meeting on Semantic Audio Analysis

Student/Career Development Event
OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1
Thursday, October 20, 11:15 am – 12:45 pm
Room 1E12

Chair: **Philip Parenteau**

Vice Chair: **Ezequiel Morfi**

The first Student Delegate Assembly (SDA) meeting is the official opening of the Convention's student program and a great opportunity to meet with fellow students from all corners of the world. This opening meeting of the Student Delegate Assembly will introduce new events and election proceedings, announce candidates for the coming year's election for the North & Latin American Regions Vice Chair, announce the finalists in the four new recording competition categories, and announce any upcoming events of the Convention. Students and student sections will be given the opportunity to introduce themselves and their activities, in order to stimulate international contacts. Daniel Deboy will address the first meeting of the SDA about the Student Blog and other education initiatives that we are working on.

All students and educators are invited to participate in this meeting. Election results and Recording Competition Awards will be given at the Student Delegate Assembly Meeting–Part 2 on Sunday, October 23.

Tutorial 3 **Thursday, Oct. 20**
12:00 noon – 1:00 pm **Room 1E13**

AFTER MARKET POWER CORDS—"SNAKE OIL" OR LEGITIMATE AUDIO ACCESSORY?

Presenter: **Michael D. Griffin**, Essential Sound Products, Inc., Rochester, MI, USA

Can after-market power cords afford performance benefits over typical "stock" cords or are they all just "snake oil"? In this session, we'll investigate the performance of audio component power supplies and the performance effects of various "stock" power cords. We'll consider the question "is there an opportunity for improved performance" and discuss ways this might be accomplished.

Thursday, Oct. 20 **12:00 noon** **Room 1E05**
Technical Committee Meeting on Automotive Audio

Special Event
AWARDS PRESENTATION AND KEYNOTE ADDRESS
Thursday, October 20, 1:00 pm – 2:15 pm
Room 1E15/16

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

Program:
- AES Awards Presentation
- Introduction of Keynote Speaker
- Keynote Address by Charles Limb

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
- **Karlheinz Brandenburg**
- **Peter Mapp**
- **Richard Sandler**
- **Valerie Tyler**

CITATION AWARD
- **Garry Gottlieb**
- **Andres Mayo**
- **Sandra Requa**

FELLOWSHIP AWARD
- **Marc Aubort**
- **Peter Eastty**
- **Geoff Emerick**
- **Steve Green**
- **Harry Hirsch**
- **Michael Lannie**
- **Aki V. Makivirta**
- **Glenn Meadows**
- **Eugene Patronis**
- **Ronald Prent**
- **Don Puluse**

SILVER MEDAL
- **Saul Walker**

GOLD MEDAL
- **Rupert Neve**
- **Phil Ramone**

HONORARY MEMBERSHIP

- **Frank Laico** [previously received this year]

Keynote Speaker

In 2010, Dr. Charles Limb infiltrated Baltimore's Hip Hop scene for a study on the parallels between that free form art and "traditional" jazz. An Associate Professor in the Department of Otolaryngology at the Johns Hopkins School of Medicine, Limb is a hearing specialist and saxophonist. His groundbreaking work on how the brain develops and assimilates musical creativity has been featured by NPR, PBS, National Geographic, Scientific American, the Smithsonian Institute, the New York Times, Library of Congress, and the American Museum of Natural History. A Faculty Member at the Peabody Conservatory of Music, Limb received his undergraduate degree at Harvard, his medical degree at Yale, and completed his surgical training at Johns Hopkins Hospital. His current research focuses on the neural basis of musical improvisation and the study of music perception in deaf individuals with cochlear implants.

The title of Limb's keynote address is "Sound, Hearing, and Music: A Journey from the Ears to the Brain" and will explore the physical and intellectual intricacies of musical creativity; its inception and perception. At first glance, the free-form art of musical improvisation and the meticulous environment of the research lab seem to make strange bedfellows, But through the application of rigorous scientific methods to the study of musicians, we have been able to get a glimpse of how this remarkable process of artistic creativity takes place in the brain.

Limb's innovative research into the parallel fields of technology and musical creativity exemplifies the AES mission. His study of brain function in improvising musicians is sure to provide valuable insights.

Thursday, Oct. 20 2:00 pm Room 1E02
Standards Committee Meeting on Digital Input/ Output Interfacing, SC-02-02

Session P4 Thursday, Oct. 20
2:30 pm – 4:30 pm Room 1E09

LOUDSPEAKERS

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

2:30 pm

P4-1 Subwoofers in Rooms: Equalization of Multiple Subwoofers—*Juha Backman*, Nokia Corporation, Espoo, Finland, Aalto University, Espoo, Finland

The effectiveness of multiple subwoofers in controlling low-frequency room modes can be improved through designing individual equalization for each of the subwoofers. The paper discusses two strategies toward equalizer design, including individual equalization of multiple subwoofer responses, minimizing the total energy radiated to the room, and optimization for minimizing sound field variation. The frequency responses and sound field distributions obtained using these methods are compared to the results of conventional equalization and modal control only through loudspeaker placement.
Convention Paper 8471

3:00 pm

P4-2 A Systematic Approach to Measurement Limit Definitions in Loudspeaker Production—*Gregor Schmidle*, NTi Audio AG, Schaan, Liechtenstein

A typical end-of-line loudspeaker test comprises ten or more different parameters tested. Each parameter has its own pass/fail limits contributing to the overall test result of the loudspeaker and therefore to the yield of the production line. This paper gives a comprehensive overview about commonly used limit calculation methods and procedures in the industry. It also delivers a systematic guidance for choosing the right limit scheme for maximizing yield, quality, and throughput.
Convention Paper 8472

3:30 pm

P4-3 Inverse Distance Weighting for Extrapolating Balloon-Directivity-Plots—*Joerg Panzer*,[1] *Daniele Ponteggia*[2]
[1]R&D Team, Salgen, Germany
[2]Audiomatica Srl, Firenze, Italy

This paper investigates an extrapolation for missing directivity-data in connection with Balloon-Plots. Such plots display the spherical directivity-pattern of radiators and receivers in form of contoured sound pressure levels. Normally the directivity-data are distributed evenly so that at each point of the display-sphere there would be sufficient data-points. However, there are circumstances where we want to display data that are not evenly distributed. For example, there might be only available the horizontal and vertical scans. The proposed Inverse Distance Weighting method is a means to extrapolate into these gaps. This paper explains this method and demonstrates some examples.
Convention Paper 8473

4:00 pm

P4-4 Mechanical Fatigue and Load-Induced Aging of Loudspeaker Suspension—*Wolfgang Klippel*, Klippel GmbH, Dresden Germany

The mechanical suspension becomes more and more compliant over time changing the loudspeaker properties (e.g., resonance frequency) significantly. This aging process is reproducible and the decay of the stiffness can be modeled by accumulating the apparent power supplied to the suspension part and using an exponential relationship. The free parameters of this model are estimated from empirical data provided by on-line monitoring or intermittent measurements during regular power tests or other kinds of long-term testing. The identified model can be used to predict the load-induced aging for music or test signals having arbitrary spectral properties. New characteristics are being introduced that simplify the quality assessment of suspension parts and separate mechanical fatigue from the initial break-in effect. Practical experiments are performed to verify the model and to demonstrate the diagnostic value for selecting optimal suspension parts providing sufficient long-term stability.
Convention Paper 8474

Session P5 **Thursday, Oct. 20**
2:30 pm - 4:30 pm **Room 1E07**

AUDIO PROCESSING—PART 1

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

2:30 pm

P5-1 Automatic Detection of the Proximity Effect—*Alice Clifford, Josh Reiss*, Queen Mary University of London, London, UK

The proximity effect in directional microphones is characterized by an undesired boost in low frequency energy as source to microphone distance decreases. Traditional methods for reducing the proximity effect use a high pass filter to cut low frequencies that alter the tonal characteristics of the sound and are not dependent on the input source. This paper proposes an intelligent approach to detect the proximity effect in a single capsule directional microphone in real time. The low frequency boost is detected by analyzing the spectral flux of the signal over a number of bands over time. A comparison is then made between the bands to indicate the existence of the proximity effect. The proposed method is shown to accurately detect the proximity effect in test recordings of white noise and other musical inputs. This work has applications in the reduction of the proximity effect.
Convention Paper 8475

3:00 pm

P5-2 Vibrato Detection Using Cross Correlation Between Temporal Energy and Fundamental Frequency—*Henrik von Coler*,[1] *Axel Röbel*[2]
[1]Technical University of Berlin, Berlin, Germany
[2]IRCAM, Paris, France

In this work we present an approach for detecting quasi periodic frequency modulations (vibrato) in monophonic instrument recordings. Since a frequency modulation in physical instruments usually causes an amplitude modulation, our method is based on a block wise cross correlation between the extracted frequency- and amplitude-modulation trajectories. These trajectories are obtained by removing the constant components. The resulting cross correlation curve shows significant positive peaks at vibrato regions and local minima at note boundaries. Our approach has the advantage of working without a previous note boundary detection and needs only a small look ahead. Furthermore no presumptions on vibrato parameters have to be made.
Convention Paper 8476

3:30 pm

P5-3 A Non-Time-Progressive Partial Tracking Algorithm for Sinusoidal Modeling—*Maciej Bartkowiak*,[1] *Tomasz Zernicki*[2]
[1]Poznan University of Technology, Poznan, Poland
[2]Telcordia Poland, Poznan, Poland

In this paper we propose a new sinusoidal model tracking algorithm that implements a non-progressive way of data processing. Sinusoidal partial parameters are estimated in the consecutive frames; however, the order of establishing individual connections between partials is optimized within the whole signal or within a specific time window. In this way, the strongest connections may be determined early, and subsequent predictions of each trajectory evolution are based on a more reliable partial evolution history, compared to a traditional progressive scheme. As a consequence, the proposed non-progressive tracking algorithm offers a statistically significant improvement of obtained trajectories in terms of better classic pattern recognition measures.
Convention Paper 8477

4:00 pm

P5-4 Perceptually Relevant Models for Articulation in Synthesized Drum Patterns—*Ryan Stables, Jamie Bullock, Ian Williams*, Birmingham City University, Birmingham, UK

In this study we evaluate current techniques for drum pattern humanization and suggest new methods using a probabilistic model. Our statistical analysis shows that both deviations from a fixed grid and corresponding amplitude values of drum patterns can have non-Gaussian distributions with underlying temporal structures. We plot distributions and probability matrices of sequences played by humans in order to demonstrate this. A new method for humanization with structural preservation is proposed, using a Markov Chain and an Empirical Cumulative Distribution Function (ECDF) in order to weight pseudorandom variables. Finally we demonstrate the perceptual relevance of these methods using paired listening tests.
Convention Paper 8478

Workshop 3 **Thursday, Oct.20**
2:30 pm – 4:00 pm **Room 1E10**

GOT METADATA? HISTORICAL, CULTURAL, AND FUTURE ISSUES OF INFORMATION ASSOCIATION FOR ARCHIVING AUDIO MATERIALS

Chair: **Thomas Ross Miller**, New York University, New York, NY, USA

Panelists: *Holger Grossmann*, Fraunhofer Institute for Digital Media Techology IDMT, Ilmenau, Germany
Chris Lacinak, AudioVisual Preservation Solutions, New York, NY, USA

Metadata is an integral component of preservation and an essential part of the audio object. Sound recordings without associated metadata are incomplete and might not be properly interpreted, understood, or managed. Meaningful access depends on effective linkage to information stored as metadata. This workshop explores the past, present, and future of metadata standards in archives and preservation. Tom Miller discusses methods and problems involved with studying and digitizing ethnographic wax cylinders and other cultural resources trapped in archaic media. Chris Lacinak analyzes recent studies and advancements focusing on embedded metadata, or metadata stored in the file ➥

itself. Holger Grossman offers new ways of globally linking cross-cultural digital music catalogues for purposes of licensing, sales, and ethnomusicological research. The presentations touch on conceptual issues as well as technical approaches to extracting and associating descriptions such as: delineating segment borders, categorizing annotations from a diverse array of sources, the use of personalized tags, and the evolution of music similarity rankings. We will consider the possibilities for capturing and interrelating different semantics of cognitive music perception, sociomusical effects produced by the algorithmic analysis of style, hybrid techniques combining personal classifications with automated systems, and the future of standardization.

Tutorial 4 **Thursday, Oct. 20**
2:30 pm – 4:00 pm **Room 1E11**

MP3 CAN SOUND GOOD

Presenters: **Schuyler Quackenbush**, Audio Research Labs, Scotch Plains, NJ, USA
Thomas Sporer, Fraunhofer IDMT, Ilmenau, Germany

Why does MP3 get such a bad rap? This tutorial will briefly present the history of MP3 coding as a medium for portable music players and discuss the impact of these early experiences on how MP3 is regarded in the audio community. The main focus of the tutorial is to discuss how to develop a robust way to characterize audio quality: a test method must be double-blind, have listeners with thorough training, use a credible test methodology, and turn the subjective results into a numeric measure of quality using sound statistical principles. To close, examples of several subjective tests of audio codecs will be presented, showing some that are done well and others that are not done well.

Tutorial 5 **Thursday, Oct. 20**
2:30 pm – 4:00 pm **Room 1E12**

ACOUSTICS FOR SOUND REINFORCEMENT

Presenters: **Peter Mapp**, Peter Mapp Associates, Colchester, UK
Kurt Graffy, Arup Acoustics

Traditionally, acoustic design for performance spaces focuses on optimum acoustics for non-reinforced performances. However, a fine concert hall for symphonic music can be very troublesome for reinforced sound. Even performance spaces designed with variable acoustics are mostly optimized for acoustic sources, little consideration is given to the needs for reinforced sound.

This tutorial addresses the different acoustic requirements for reinforced sound as opposed to non-reinforced sound. Topics to be discussed will include:

- Reverberation, support or interference?
- Source directivity, positioning and aiming
- Discrete reflections and how to avoid them
- What to include in a new venue for optimum adaptation for reinforced sound
- Practical measures in existing situations
- Workarounds in acoustic harsh environments - Control of bass reverberation and room acoustic effects at low frequencies

Live Sound Seminar 3 **Thursday, Oct. 20**
2:30 pm – 4:30 pm **Room 1E14**

DATA NETWORKS AS A DIGITAL AUDIO TRANSPORT MECHANISM

Presenter: **Josh Evans**, Lab Gruppen

An exploration of data networks as a digital audio distribution model. How can we share audio and control data, and how can we facilitate audio routing over a network? Several of the current audio network protocols as well as some proprietary topologies will be examined.

Games Audio Session 3 **Thursday, Oct. 20**
2:30 pm – 4:00 pm **Room 1E08**

GAME AUDIO SECRETS: EVERYTHING YOU ALWAYS WANTED TO KNOW ABOUT GAME AUDIO BUT WERE AFRAID TO ASK

Presenters: **Peter "pdx" Drescher**, Twittering Machine
Stephen Harwood Jr., Dynamic Systems Music
Damian Kastbauer, Bay Area Sound
Scott Selfon, Senior Development Lead, Microsoft

Game-curious? Interested in the video game industry, but unsure of what exactly it is that we do here? You are not alone. Video game production values are improving rapidly, creating increased demand for top-notch, experienced audio professionals. But many composers, sound designers, and producers looking to bring their expertise from the world of film and TV into the video game industry are uncertain about what it is they'll be getting themselves into. In this very special extended Q&A session, a panel of distinguished game audio insiders will take questions from the floor, and turn them into answers from behind the curtain. Discussion will be driven by audience participation—all topics are welcome, from interactive music to implementation. Come prepared to inquire, be inspired, and take notes.

Historical Event
LEOPOLD STOKOWSKI AND THE HISTORY OF ANALOG RECORDING
Friday, October 21, 2:30 pm – 4:00 pm
Room 1E13

Presenter: **Robert Auld**

Leopold Stokowski was active as a recording artist from 1917 until 1977—virtually the entire period of the recording of music by analog technology. Further, due to his obsessive interest in the art and technology of recording, he was frequently in contact with engineers and researchers who were working on technical advances. Therefore, it is possible to trace most of the major developments in analog recording over nearly 60 years through examining the recordings of Leopold Stokowski.

Auld's multimedia presentation will include rare audio recordings, still photos, and film clips, all drawn from the extensive activities of Leopold Stokowski over the course of his career. Special attention will be paid to his involvement in the development of multichannel sound recording, including his collaboration with Bell Labs starting in 1932, his work with Walt Disney for the film Fantasia, and his encouragement of recording in quadraphonic sound in the 1970s.

Robert Auld is an audio engineer active in New York City and the northeast region. His clients include National Public Radio, the BBC, Riverside Symphony, Beantown Swing Orchestra, and many others. He has written for Recording Magazine on technical audio issues and has made presentations on historical audio subjects at AES conventions and section meetings. He is a former chairman of the New York Section of the AES.

Special Event
PHIL RAMONE/TONY BENNETT
Thursday, October 20, 2:30 pm – 4:00 pm
Room 1E15/16

Presenter: **Phil Ramone**

Tony Bennett: Duets II

Producer Phil Ramone, co-producer and engineer Dae Bennett, and singer Tony Bennett will discuss the production of their new album, *Tony Bennett: Duets II*. *Duets II* features Mr. Bennett in duet with Norah Jones, Carrie Underwood, Andrea Bocelli, Amy Winehouse, Mariah Carey, Natalie Cole, Sheryl Crow, Josh Groban, Faith Hill, John Mayer, Willie Nelson, Alejandro Sanz, Michael Bublé, K.D. Lang, and Lady Gaga. [*Note:* Mr. Bennett was ill and did not attend this event.]

Session P6 **Thursday, Oct. 20**
3:00 pm – 4:30 pm **1E Foyer**

POSTERS: SPEECH

3:00 pm

P6-1 A Systematic Study on Formant Transition in Speech Using Sinusoidal Glide—*Wen-Jie Wang*,[1] *Benjamin Guo*,[2] *Chin-Tuan Tan*[2]
[1]New York University, School of Medicine, New York, NY, USA
[2]City University of New York, New York, NY, USA

The goal of this study was to use sinusoidal glide to investigate the perceivable acoustic cues in a consonant-vowel /CV/ syllable systematically. The sinusoidal glide is designed to mimic the formant trajectory in a /CV/ syllable with two parts: a frequency glide followed by a constant frequency. The experiment varied the frequency step (with rising and falling glide) and duration of the initial part, and the center frequency and duration of the final part of the sinusoidal glide. We asked six normal hearing subjects to discriminate sinusoidal glides from sinusoids of constant frequency, and found that subjects require a larger frequency step when the duration of the glide is shortened but a smaller frequency step when the center frequency of the final part is lowered, to discriminate the two stimuli. The outcome of this experiment is compared to the outcomes of previous studies using synthesized formants and sinusoidal replicas.
Convention Paper 8479
[Paper not presented but is available for purchase]

3:00 pm

P6-2 Perceived Quality of Resonance-Based Decomposed Vowels and Consonants— *Chin-Tuan Tan*,[1] *Benjamin Guo*,[1] *Ivan Selesnick*[2]
[1]New York University School of Medicine, New York, NY, USA
[2]Polytechnic Institute of New York University, Brooklyn, NY, USA

The ultimate objective of this study is to employ a resonance-based decomposition method for the manipulation of acoustic cues in speech. Resonance-based decomposition (Selesnick, 2010) is a newly proposed nonlinear signal analysis method based not on frequency or scale but on resonance; the method is able to decompose a complex non-stationary signal into a “high-resonance” component and a “low-resonance” component using a combination of low- and high-Q-factors. In this study we conducted a subjective listening experiment on five normal hearing listeners to assess the perceived quality of decomposed components, with the intention of deriving the perceptually relevant combinations of low- and high-Q-factors. Our results show that normal hearing listeners generally rank high-resonance components of speech stimuli higher than low-resonance components. This may be due to a greater salience of perceptually significant formant cues in high-resonance stimuli.
Convention Paper 8480
[Paper not presented but is available for purchase]

3:00 pm

P6-3 Relationship between Subjective and Objective Evaluation of Distorted Speech Signals—*Mitsunori Mizumachi*, Kyushu Institute of Technology, Fukuoka, Japan

It is important for designing a noise reduction algorithm to evaluate the quality of noise-reduced speech signals accurately and efficiently. Subjective evaluation gives accurate evaluation, but requires listening tests with a lot of subjects and time. Then, objective distortion measures are employed as efficient evaluation. However, almost all the distortion measures do not consider the temporal variation of speech distortion. In this paper the temporal aspect of the segmental speech distortion is investigated based on higher-order statistics, that is, variance, skewness, and kurtosis. It is interesting that the skewness of the objective evaluation gives a good explanation for the discrepancy between subjective and objective evaluation.
Convention Paper 8481

3:00 pm

P6-4 Multiple Microphones Speech Enhancement Using Minimum Redundant Array—*Kwang-Cheol Oh*, Samsung Electronics Co., Ltd., Suwon City, Gyeong-Gi Do, Korea

A non-uniformly-spaced multiple microphone array speech enhancement method with small aperture size is proposed and analyzed. The technique utilizes a minimum redundant array structure used in antenna array in order to prevent spatial aliasing for high frequencies and uses the phase difference-based dual microphone speech enhancement techniques to implement small-microphone array. It has highly directive features from the low-frequency to high ➥

frequencies evenly, and its performance is measured with directivity index. The directivity index(DI) for the proposed approach is about 3 dB higher than that of a multiple microphone approach with the phase-based filter.
Convention Paper 8482

Thursday, Oct. 20 3:00 pm Room 1E05
Technical Committee Meeting on Microphones and Applications

Session P7 Thursday, Oct. 20
4:30 pm – 6:30 pm Room 1E09

SOUND FIELD ANALYSIS AND REPRODUCTION—PART 1

Chair: **Sascha Spors**, Deutsche Telekom Laboratories, Berlin, Germany

4:30 pm

P7-1 Two Physical Models for Spatially Extended Virtual Sound Sources—*Jens Ahrens, Sascha Spors*, Deutsche Telekom Laboratories, Technische Universität Berlin, Berlin, Germany

We present physical models for the sound field radiated by plates of finite size and spheres vibrating in higher modes. The intention is obtaining a model that allows for controlling the perceived size of a virtual sound source in model-based sound field synthesis. Analytical expressions for radiated sound fields are derived and simulations of the latter are presented. An analysis of interaural coherence in a virtual listener, which has been shown to serve as an indicator for perceived spatial extent, provides an initial proof of concept.
Convention Paper 8483

5:00 pm

P7-2 Auditory Depth Control: A New Approach Utilizing a Plane Wave Loudspeaker Radiating from above a Listener—*Sungyoung Kim, Hiraku Okumura, Hideki Sakanashi, Takurou Sone*, Yamaha Corporation, Hamamatsu, Japan

One of the distinct features of a 3-D image is that the depth perceived by viewer is controlled so that objects appear to project toward the viewers. However, it has been hard to move auditory imagery near to listeners using conventional loudspeakers and panning algorithms. In this study we proposed a new system for controlling auditory depth, which incorporates two loudspeakers: one that radiates sound from the front of a listener and another that radiates plane waves from above a listener. With additional equalization that removes spectral cues corresponding to elevation, the proposed system generates an auditory image "near a listener" and controls the depth perceived by the listener, thereby enhancing the listener's perception of 3-D sound.
Convention Paper 8484

5:30 pm

P7-3 The SCENIC Project: Space-Time Audio Processing for Environment-Aware Acoustic Sensing and Rendering—*Paolo Annibale,*[1] *Fabio Antonacci,*[2] *Paolo Bestagini,*[2] *Alessio Brutti,*[3] *Antonio Canclini,*[2] *Luca Cristoforetti,*[3] *Emanuël Habets,*[1,4] *J. Filos,*[1] *Walter Kellerman,*[1] *Konrad Kowalczyk,*[1] *Anthony Lombard,*[1] *Edwin Mabande,*[1] *Dejan Markovic,*[2] *Patrick Naylor,*[4] *Maurizio Omologo,*[3] *Rudolf Rabenstein,*[1] *Augusto Sarti,*[2] *Piergiorgio Svaizer,*[3] *Mark Thomas*[4]
[1]University of Erlangen, Erlangen, Germany
[2]Politecnico di Milano, Milan, Italy
[3]Fondazione Bruno Kessler – IRST, Trento, Italy
[4]Imperial College London, London, UK

SCENIC is an EC-funded project aimed at developing a harmonized corpus of methodologies for environment-aware acoustic sensing and rendering. The project focuses on space-time acoustic processing solutions that do not just accommodate the environment in the modeling process but that make the environment help toward achieving the goal at hand. The solutions developed within this project cover a wide range of applications, including acoustic self-calibration, aimed at estimating the parameters of the acoustic system; environment inference, aimed at identifying and characterizing all the relevant acoustic reflectors in the environment. The information gathered through such steps is then used to boost the performance of wavefield rendering methods as well as source localization/characterization/extraction in reverberant environments.
Convention Paper 8485

6:00 pm

P7-4 Object-Based Sound Re-Mix for Spatially Coherent Audio Rendering of an Existing Stereoscopic-3-D Animation Movie—*Marc Evrard,*[1] *Cédric R. André,*[1,2] *Jacques G. Verly,*[1] *Jean-Jacques Embrechts,*[1] *Brian F. G. Katz*[2]
[1]University of Liege, Liege, Belgium
[2]LIMSI-CNRS, Orsay, France

While 3-D cinema is becoming more mainstream, little effort has focused on the general problem of producing a 3-D sound scene spatially coherent with the visual content of a stereoscopic-3-D (s-3D) movie. The perceptual relevance of such spatial audiovisual coherence is of significant interest. In order to carry out such experiments, it is necessary to have an appropriate s-3D movie and its corresponding 3-D audio track. This paper presents the procedure followed to obtain this joint 3-D video and audio content from an exiting animated s-3D film, problems encountered, and some of the solutions employed.
Convention Paper 8486

Session P8 Thursday, Oct. 20
4:30 pm – 6:30 pm Room 1E07

LOUDNESS MEASUREMENT AND PERCEPTION

Chair: **Dan Harris**, Sennheiser Technology and Innovation

4:30 pm

P8-1 Effect of Horizontal Diffusivity on Loudness —*Densil Cabrera*, Luis Miranda, University of Sydney, Sydney, NSW, Australia

This paper examines how the spatial characteristics of a sound field affect its loudness. In an experiment, listeners adjusted the gain of stimuli so as to match the loudness of a reference stimulus. The experiment was conducted using binaural stimuli presented over headphones—using stimuli that simulated the sound from eight sound sources evenly distributed in a circle around the listener. Four degrees of diffusivity were tested, ranging from a single active sound source, to all eight sources producing decorrelated sound with identical power spectra. Four power spectra were tested: broadband pink noise, and low-, mid-, and high-frequency bandpass-filtered pink noise. The paper finds that in modeling the binaural loudness summation of the diffuse stimuli, a binaural gain constant about 1 or 2 dB greater than that of the non-diffuse stimuli provides the least error.
Convention Paper 8487

5:00 pm

P8-2 Locating the Missing 6 dB by Loudness Calibration of Binaural Synthesis—*Florian Völk, Hugo Fastl*, AG Technische Akustik, MMK, Technische Universität München, Munich, Germany

Binaural synthesis is a sound reproduction technology based on the convolution of sound signals with impulse responses defined between a source and a listener's eardrums. The convolution products are typically presented by headphones. For perceptual verification, subjects traditionally remove the headphones to listen to the corresponding real scenario, which is cumbersome and requires a pause between the stimuli. In this paper loudness adjustments are presented using a method that allows for direct comparison by defining the reference scene for a listener wearing headphones. Influences of different headphones and equalization procedures are discussed and an explanation for the difference in auditory canal pressure between headphone and loudspeaker reproduction at the same loudness commonly referred to as the missing 6 dB is deduced.
Convention Paper 8488

5:30 pm

P8-3 Difference between the EBU R-128 Meter Recommendation and Human Subjective Loudness Perception—*Fabian Begnert, Håkan Ekman, Jan Berg*, Luleå University of Technology, Piteå, Sweden

The vast loudness span of broadcast sound can be reduced by the use of loudness meters. In an ideal case, the measured and the perceived loudness would be equal. A loudness meter fulfilling the EBU R-128 recommendation was investigated for its correspondence with perceived loudness. Several sound stimuli with large loudness differences representing five different types of broadcast program material were normalized to have equal meter measured loudness level. Subjects listened to pair-wise presentations of the normalized stimuli, which they subsequently set to have equal perceived loudness. The settings were recorded and analyzed. The results show that the normalization yields both equal as well as different perceived loudness between program types. The maximum difference was ±2.82 dB.
Convention Paper 8489

6:00 pm

P8-4 A Novel Multi-Stage Loudness Control Algorithm for Audio Processing—*Balaji Vodapally,[1] Brijesh Singh Tiwari,[1] Deepen Sinha[2]*
[1]ATC Labs, Noida, India
[2]ATC Labs, NJ, USA

Modern audio broadcast processors are designed to provide an audio with consistent loudness within a specified dynamic range. In general a tight control over the processed audio level is provided by the Automatic Gain Control (AGC) mechanism. Most of the other processing functions are tuned for a pre-defined signal level; this arises the need for multistage loudness control. This paper presents the gain control mechanism employed in various stages of the proposed multi-stage Audio Loudness Control (MALC) and the interaction between various stages. We also present results which explain how the proposed scheme provides better control over loudness against the signal level. The paper also emphasizes the importance of frequency weighing based loudness measure.
Convention Paper 8490

Workshop 4 **Thursday, Oct. 20**
4:30 pm – 6:30 pm **Room 1E08**

SOUND QUALITY EVALUATION

Chair: **Frederik Nagel**, Fraunhofer IIS/International Audio Laboratories, Erlangen, Germany

Panelists: *Søren Bech*, Bang & Olufsen, Denmark
Poppy Crum, Dolby Laboratories, San Francisco, CA, USA
Schuyler Quackenbush, Audio Research Labs, Scotch Plains, NJ, USA
Nick Zacharov, Genelec OY, Finland

The workshop aims at discussing current practices of listening test preparation, execution, and evaluation. It will cover the choice of test items, the selection of participants, and, last, the means of test design and statistical evaluation. Questions such as "What are naive or expert listeners?" "What is a convenience sample and what can we learn from the results?" "What is the difference between small and intermediate artifacts?" "What kind of statistics can be applied when the sample is small? Might other test methods be more adequate?" "Which characteristics should the items under test have?" exist but are hardly topics of publication in audio research. These issues are expected to be discussed among the participants and resulting in proposals to solve the existing problems. Eventually, this workshop aims at providing researchers with a better understanding what happens during listening tests and giving guidelines for a better listening test practice.

Workshop 5 **Thursday, Oct. 20**
4:30 pm – 6:30 pm **Room 1E15/16**

PRODUCING ACROSS GENERATIONS: NEW CHALLENGES, NEW SOLUTIONS—MAKING RECORDS FOR NEXT TO NOTHING IN THE 21ST CENTURY

Chair: **Nicholas Sansano**, New York University, New York, NY, USA

Panelists: *Jason Goldstein*
Dan Knobler, Mason Jar Music
Jesse Lauter
Bob Power
Dan Romer
Jonathan Seale, Mason Jar Music
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.

Broadcast/Media Streaming Session 3
Thursday, October 20 **4:30 pm – 6:00 pm**
Room 1E10

STREAMING WITH HTML5

Chair: **Valerie Tyler**, College of San Mateo

Panelists: Ian Bennett, Microsoft
Greg Ogonowski, Orban

HTML5 is a language for structuring and presenting content for the World Wide Web, a core technology of the Internet. It is the fifth revision of the HTML standard. HTML5 will have many features built into the code. One feature is the media player or how it will handle media being downloaded or streamed. This session will look into the technical considerations for media to be played back as well as the user interfaces.

Game Audio Session 4 **Thursday, Oct. 20**
4:30 pm – 6:30 pmRoom 1E11

REALISTIC INTERACTIVE REVERB PROCESSING FOR GAMES

Moderator: **Steve Martz**, THX Ltd.

Panelists: Dinesh Manocha, University of North Carolina at Chapel Hill, Chapel Hill, NC, USA
Masataka Nakahara, Onfuture Ltd.
Nikunj Raghuvanshi, Microsoft Research

Realistic acoustics is essential for a realistic, immersive game experience. Computing and applying realistic acoustic responses for game scenes, called "interactive reverb," is thus a very important problem. Most games today rely heavily on the audio designer to assess and apply the reverb parameters for different areas of a game scene. This manual process requires a lot of experience, expertise, and labor. Dynamic effects, such as the effect of source / listener positions, or changes in room geometry are almost always neglected.

Due to these reasons, a degree of automation and improvement in acoustic detail has long been needed for games. In the area of room acoustics, automatic calculation methods, as well as perceptual aspects of room responses have been (and continue to be) extensively studied. By utilizing reverb computation based on such techniques, along with artistic input from the audio designer, the quality of game reverb could be vastly improved. In this panel three approaches for interactive reverb computation will be discussed: wave, geometrical, and statistical acoustics. Panelists will discuss the technical aspects, relative strengths and weaknesses, performance requirements/tradeoffs, and also show video demonstrations of the techniques in action. These approaches have been integrated with game engines and also map well to commodity multi-core processors.

(1) *Wave acoustics:* Real-time wave acoustics for games — *Presenter:* Nikunj Raghuvanshi. A novel technique for wave-based sound propagation in games.

(2) *Geometrical acoustics:* GSound + Acoustic Transfer Operator — *Presenter:* Dinesh Manocha. Two recent geometric techniques for interactive sound propagation.

(3) *Statistical acoustics:* Advanced Statistical 3D reverb — *Presenter:* Masataka Nakahara. Statistically calculated reverb using new statistical acoustics theory.

Product Design Session 2 **Thursday, Oct. 20**
4:30 pm – 6:30 pm **Room 1E12**

IS YOUR EQUIPMENT DESIGN A NOISE PROBLEM WAITING TO HAPPEN?

Presenter: **Bill Whitlock**, Jensen Transformers

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

Session EB2 **Thursday, Oct. 20**
5:00 pm – 6:30 pm **1E Foyer**

POSTERS

5:00 pm

EB2-1 *Engineering Brief 21 has been withdrawn*

5:00 pm

EB2-2 **Digital Control of an Analog Parametric Equalizer —***Blair Ryan Conner*, Purdue University, West Lafayette, IN, USA

This project focuses on creating a digitally controllable analog band-pass filter with an

adjustable resonant frequency for a middle frequency adjuster in an audio equalization stage. The design of the band-pass filter is a standard series resistor, inductor, and capacitor filter network. An adjustable gyrator circuit simulates an inductor to change the resonant frequency of the filter. Inside the gyrator circuit, a voltage-controlled amplifier is configured to simulate a resistor to change the gyrators simulated inductance. A digital to analog converter controls the gain of the voltage-controlled amplifier to make the analog filter digitally controlled. This circuit successfully acts as a bandpass filter with a digitally controllable resonant frequency.
Engineering Brief 22

5:00 pm

EB2-3 Mash-Up—*Stephen Partridge*, Buckinghamshire New University, High Wycombe, Buckinghamshire, UK

The project will constitute an exploration of the choices made by users of audio post-production/editing applications in the selection and reuse of digital media files. The intended outcome would be to inform a better understanding of the ways in which contemporary users engage with digital media artifacts. The production context upon which this study will be based concerns an ongoing enterprise project.
Engineering Brief 23

5:00 pm

EB2-4 Statistical Analysis of Electro-Acoustic Measurements Sets Using Scilab—*Daniele Ponteggia*, Audiomatica Srl, Firenze, Italy

The production management of electro-acoustic systems require the statistical analysis of measurements data. The analysis process should be sufficiently flexible to match the needs of the production process and the number of measured samples should be large enough to ensure the accuracy in statistical terms. Using an open source numerical computation software (Scilab) is possible to create statistical analysis procedures in a simple and cost effective way. Scilab syntax is simple enough to be acquired within a fairly short time, while data analysis capabilities are very advanced. In this work some sample applications are shown, with minimal code edit the provided examples can be adapted to several real world cases.
Engineering Brief 24

5:00 pm

EB2-5 Latency Measurements of Audio Sigma Delta Analog to Digital and Digital to Analog Converts—*Yonghao Wang*, Queen Mary University of London, London, UK, Birmingham City University, Birmingham, UK

Latency is a well recognized issue when using digital audio workstations for live music processing. Previous research has reported measurements of the latency of the whole audio processing chain based on a "blackbox" approach. This report presents the results of latency measurement of typical compact analog to digital and digital to analog converters (ADC/DACs) in isolation from computer system processing overheads by using a high-speed data acquisition device. The report discusses the testing methods and pitfalls. It confirms that the latency is almost exclusively accounted for by the expected group delay of the digital decimation filters and interpolation filters used in the Sigma-Delta convertor.
Engineering Brief 25

5:00 pm

EB2-6 The Effect of Reverberation on Music Performance—*Elisa Sato, Toru Kamekawa, Atsushi Marui*, Tokyo University of the Arts, Tokyo, Japan

The structure of playing musical instruments consists of 3 basic steps: performing on the instrument to make musical sounds, recognizing spatial information and music in the space through listening to the sounds themselves, and finally returning the information to the performance to adapt their musical experience and consciousness. These steps go on as a loop during the performance of musical instruments, and it is widely known that reverberation effects recognition of the space, i.e. the second step. Adjusting the whole acoustic environment inside the room with sampling reverberation should help musicians to play just as they want to. In this paper the relation between multiple parameters of reverberation and the features of solo violin performance is investigated.
Engineering Brief 26

5:00 pm

EB2-7 Analysis-Synthesis Techniques for Additive Granular Synthesis—*James O'Neill*, University of Miami, Coral Gables, FL, USA

This project explores granular synthesis techniques that utilize various basis functions inspired by existing matching pursuit algorithms. The first algorithm performs a STFT on an input signal and synthesizes a new, granular signal using one-dimensional Gabor atoms. These atoms can be made to virtually reproduce the input signal, but a wide variety of granular effects can be achieved by altering the distribution of the atoms in the time and frequency domains, such as granular time stretching and pitch shifting, along with statistical distribution techniques introduced by the author. The second algorithm utilizes a basis set of generated noise bursts, which can be over-complete or an orthonormal basis for the Hilbert space that corresponds to the analysis window by applying the Gram-Schmidt process to the burst library. The noise functions are then used as grain contents in the synthesis stage, where a variety of effects are created with redistribution methods. Audio examples are provided over headphones.
Engineering Brief 27

5:00 pm

EB2-8 Harmonic Distortion Analysis in a Class-AB Tube Amplifier: The McIntosh MC-240—*Phillip Minnick*, University of Miami, Coral Gables, FL, USA ➡

Some Class-AB tube amplifiers remain in demand for audiophiles due to their linear gain stages, low feedback, and minimal high-order harmonic distortion. The McIntosh MC240 Class-AB tube amplifier is a benchmark, high-quality stereo amplifier from the 1960s. This presentation gives an overview of the restoration of the amplifier to achieve a satisfactory level of performance, examination of the electrical topology in relation to the output signal's distortion characteristics, and detailed analysis of the distortions produced by the amplifier using psychoacoustic-based listening tests as well as standard benchmark tests.
Engineering Brief 28

5:00 pm

EB2-9 The "Williams Star" Microphone Array Support System—*Michael Williams*, Freelance Sound Recording Engineer and Lecturer, Sounds of Scotland, Le Perreux sur Marne, France

With the ever increasing interest in 5-channel recording for home cinema, television, and pure 5-channel audio, the search for an operationally simple, reliable, and high quality surround sound recording system is becoming ever more important. The equal segment 4-channel and 5-channel arrays described in two AES papers by Michael Williams (AES Convention Papers 3157 and 7480) are attracting more and more interest within the audio industry. However the research for a satisfactory operational microphone configuration cannot be dissociated from the purely mechanical problem of finding a suitable microphone array support system. The "Williams Star" Microphone Support System can provide a simple, flexible, and reliable microphone array support system for any equal segment array design. The visual impact of this system has also been reduced to an absolute minimum. Currently only the 4-channel and 5-channel seem to meet operational broadcasting requirements, but a 7-channel format is also provided for future developments with respect to the new Blu-ray format.
Engineering Brief 29

Thursday, Oct. 20 5:00 pm Room 1E02
Standards Committee Meeting on Audio-File Transfer and Exchange, SC-02-08

Thursday, Oct. 20 5:00 pm Room 1E05
Technical Committee Meeting on Archiving, Restoration, and Digital Libraries

Session P9 Friday, Oct. 21
9:00 am – 12:30 pm Room 1E07

APPLICATIONS IN AUDIO

Chair: **Jonas Braasch**, Rensselaer Polytechnic Institute, Troy, NY, USA

9:00 am

P9-1 Simulation-Based Interface for Playing with Sounds in Media Applications—*Insook Choi*,[1] *Robin Bargar*[2]
[1]City University of New York, Brooklyn, NY, USA
[2]Columbia College Chicago, Chicago, IL, USA

Advanced audio processing for interactive media is in demand for a wide range of applications and devices. The requirements for interactive media contexts tend to impose both device-specific and style-specific constraints. The goal of the present research is to develop a robust approach to interactive audio that may be persistent across diverse media contexts. This project adopts a structural approach to the relationship of interactive sounds to interactive graphical media. We refer to this as a model-to-model architecture. Sound production is decoupled from specific media styles, enabling abstractions using feature analysis of simulation output that can be adapted to a variety of media devices. The identifying metaphor for this approach is playing with sounds through graphical representations and interactive scenarios.
Convention Paper 8491

9:30 am

P9-2 Advances in ENF Database Configuration for Forensic Authentication of Digital Media—
Catalin Grigoras, Jeffrey M. Smith, Christopher W. Jenkins, University of Colorado Denver, Denver, CO, USA

When building an Electric Network Frequency (ENF) database for forensic purposes, ensuring that the recorded signal satisfies standards for forensic analysis is crucial. The ENF signal shall be free of clipping and lossy compression distortions, the signal to noise ratio shall be as high as possible, and the acquisition system clock shall be synchronized with an atomic clock. Using an ENF database to compare reference and questioned ENF involves precise measurements of amplitude, spectrum, and zero-crossings in order to accurately time-stamp, discover potential edits, and authenticate digital audio/video recordings. Due to the inherent differences in electronic components, building multiple ENF probes to create multiple databases with matching waveforms can be challenging. This paper addresses that challenge and offers a solution by using MathWorks MATLAB to calculate the best combination of components and produce graphical displays to help give a visual aid to the outcome in order to build a high quality ENF probe. This paper also addresses the challenge of establishing a fail-safe database to safely store the accurately acquired ENF information. This paper concludes that a reliable ENF database is mandatory for both scientific research and for forensic examination.
Convention Paper 8492

10:00 am

P9-3 Virtual Systems Engineering in Automotive Audio—*Alfred J. Svobodnik*, Harman International, Vienna, Austria

The present paper focuses on virtual product development for automotive audio systems. In the core a multidisciplinary simulation environment is used to perform all system engineering tasks in a fully virtual environment. First the theory of a multiphysical simulation model of electrodynamic loudspeakers is described. Subsequently, this model is extended to account for enclosures used as a resonance volume for loudspeakers (especially for

reproduction of low frequency musical content). Furthermore, it is shown how the multiphysical model of loudspeakers and enclosures can be extended to simulate the radiation of sound waves into the car interior. Finally the virtual audio system, described by a multiphysical simulation model, is virtually tuned and auralized long before any piece of hardware exists. Tuning and auralization require that the simulation model is extended toward a multidisciplinary simulation environment as, additionally to engineering analysis methods, paradigms of the following disciplines are added: digital signal processing, psychoacoustics, binaural audio and subjective evaluation. The integration of the human factor (i.e., how audio events are perceived by humans with respect to spectral and spatial effects) is added in the tuning process and it is demonstrated how we can ultimately listen to a virtual audio system by means of advanced auralization techniques based on a binaural playback system. Additionally some remarks on the business benefits of these methods are given and uncertainties in our simulation models, which are inherent to every modeling approach, are addressed as well.
Convention Paper 8493
[Paper not presented but is available for purchase]

10:30 am

P9-4 Acoustical Modeling of Gunshots including Directional Information and Reflections— *Robert Maher*, Montana State University, Bozeman, MT, USA

Audio recordings of gunshots exhibit acoustical properties that depend upon the geometry and acoustical characteristics of nearby reflecting surfaces and the relative orientation of the firearm with respect to the recording microphone. Prior empirical studies have demonstrated the basic principles of gunshot recordings near the firearm and near the target. This paper describes an experiment to model the directional characteristics and reflections of several firearm types for a set of test configurations. The results show that reflections and reverberation can be a significant portion of the total acoustic energy received at the microphone.
Convention Paper 8494

11:00 am

P9-5 Influence of Recording Distance and Direction on the Analysis of Voice Formants —Initial Considerations—*Eddy B. Brixen*, EBB Consult, Smørum, Denmark; *Siham Christensen*

Based on recordings carried out in an anechoic chamber it is investigated to which degree the voice formants are affected by recording distance and direction in the near field (10, 20, 40, 80, and 100 cm) and different directions (on-axis, horizontally 45 and 90 degrees, vertically +/–45 degrees). This paper presents the analysis applied and discusses to what extent the results obtained must be taken into consideration when assessing voice samples for general phonetic research and for automatic voice ID/voice comparisons. It is concluded from the results of the analysis that especially weaker formants are displaced to a not negligible degree.
Convention Paper 8495

11:30 am

P9-6 Musical Movements—Gesture Based Audio Interfaces—*Anthony Churnside, Chris Pike, Max Leonard*, BBC R&D, Media City, Salford, UK

Recent developments have led to the availability of consumer devices capable of recognizing certain human movements and gestures. This paper is a study of novel gesture-based audio interfaces. The authors present two prototypes for interacting with audio/visual experiences. The first allows a user to "conduct" a recording of an orchestral performance, controlling the tempo and dynamic. The paper describes the audio and visual capture of the orchestra and the design and construction of the audio-visual playback system. An analysis of this prototype, based on testing and feedback from a number of users, is also provided. The second prototype uses the gesture tracking algorithm to control a three-dimensional audio panner. This audio panner is tested and feedback from a number of professional engineers is analyzed.
Convention Paper 8496

12:00 noon

P9-7 A Nimble Video Editor that Puts Audio First— *Jörn Loviscach*, Fachhochschule Bielefeld (University of Applied Sciences), Bielefeld, Germany

Video editing software tends to be feature-laden, to respond sluggishly to user input—and to be focused on visuals rather than on auditory aspects. All of this is a burden when the task is to edit material in which audio plays the leading role, such as a talk show, a video podcast, or a lecture recording. This paper presents a highly visual no-frills video editor that is tailored to these tasks. It combines a range of techniques that speed up the process of searching and reviewing. They range from an overview-and-detail display to speech recognition to constant-pitch variable-speed playback. The implementation is heavily multithreaded and fully leverages the computer's main memory to ensure a highly fluid interaction.
Convention Paper 8497

Session P10 Friday, Oct. 21
9:00 am – 12:00 noon Room 1E09

TRANSDUCERS AND AUDIO EQUIPMENT

Chair: **Juha Backman**, Nokia Corporation, Espoo, Finland

9:00 am

P10-1 A Parametric Study of Magnet System Topologies for Miniature Loudspeakers— *Holger Hiebel*, Knowles Electronics Austria GmbH, Vienna, Austria, Graz University of Technology, Graz, Austria

This paper presents an overview of the results of a parametric study on miniature loudspeaker (microspeaker) designs. It compares a specific ➥

microspeaker design with fixed outer dimensions in three different electrodynamic magnet system topologies, namely the centermagnet, ringmagnet, and doublemagnet configurations. The study results are derived from simulations of the Bl-factor, moving mass, and effective radiating area with the voice coil inner diameter being the independent variable. Sound pressure level, electrical quality factor, and resonance frequency in a closed box were calculated and used to create easily understandable charts comparing the three topologies.
Convention Paper 8498

9:30 am

P10-2 A Computational Model of Vented Band-Pass Enclosure Using Transmission Line Enclosure Modeling—*Jongbae Kim, Gyung-Tae Lee, Yongje Kim*, Samsung Electronics Co., Ltd., Suwon City, Gyeong-Gi Do, Korea

In order to predict low frequency performance of loudspeaker systems, the lumped parameter model is very useful. However, it doesn't consider the enclosure geometry. Therefore, in some special cases, it makes a serious deviation between simulation and experimental results. According to the recent slim trend in IT devices, the majority of flat panel TVs and mobile devices adopt not only thin, long enclosures but also front radiating structure with waveguide. This loudspeaker system could be simplified as a vented band-pass enclosure. However due to the negligence of geometry, the effects of vent and driver location can't be considered. This paper discusses a computational model of the complicated vented band-pass enclosure using Backman's low-frequency method for slim type band-pass enclosure models. Simulation results were compared with experimental results to verify the validity of the computational model.
Convention Paper 8499

10:00 am

P10-3 Nonlinear Viscoelastic Models—*Finn T. Agerkvist*, Technical University of Denmark, Lyngby, Denmark

Viscoelastic effects are often present in loudspeaker suspensions. This can be seen in the displacement transfer function that often shows a frequency dependent value below the resonance frequency. In this paper nonlinear versions of the standard linear solid model (SLS) are investigated. The simulations show that the nonlinear version of the Maxwell SLS model can result in a time dependent small signal stiffness while the Kelvin Voight version does not.
Convention Paper 8500

10:30 am

P10-4 Practical Applications of a Closed Feedback Loop Transducer System Equipped with Differential Pressure Control—*Fabio Blasizzo,*[1] *Paolo Desii,*[2] *Mario Di Cola,*[3] *Claudio Lastrucci*[2]
[1]Fabio Blasizzo, Trieste, Italy
[2]Powersoft S.r.l., Firenze, Italy
[3]Audio Labs Systems, Chieti, Italy

A closed feedback loop transducer system dedicated to very low frequency reproduction can be used in several different applications. The use of a feedback control loop can be very helpful to overcome some of the well known transducer limitations and to improve some of the acoustical performances of most of subwoofer systems. The feedback control of this system is based on a differential pressure control sensor. The entire system control is performed by a "Zero Latency DSP" application, specifically designed for this purpose in order to be able to process the system with real time performances. Practical applications to real world examples are being shown with design details and some test results.
Convention Paper 8501

11:00 am

P10-5 Dual Diaphragm Compression Drivers—*Alex Voishvillo*, JBL Professional, Northridge, CA, USA

A new type of compression driver consists of two motors, two diaphragm assemblies, and two phasing plugs connected to the same acoustical load (horn or waveguide). The annular flexural diaphragms are made of light and strong polymer film providing low moving mass. Unique configuration of the phasing plugs provides summation of both acoustical signals and direction of the resulting signal into a mutual acoustical load. Principles of operation of the dual driver are explained using a combination of matrix analysis, finite elements analysis, and data obtained from a scanning vibrometer. Comparison of performance of this driver and conventional driver based on titanium dome diaphragm is performed. New transducer provides increase of power handling, lower thermal compression, smoother frequency response, and decrease of nonlinear distortion and sub-harmonics.
Convention Paper 8502

11:30 am

P10-6 Distortions in Audio Op-Amps and Their Effect on Listener Perception of Character and Quality—*Robert-Eric Gaskell, Peter E. Gaskell, George Massenburg*, McGill University, Montreal, Quebec, Canada

Different operational amplifier topologies are frequently thought to play a significant role in the sonic character of analog audio equipment. This paper explores whether common audio operational amplifiers are capable of producing distortion characteristics within their normal operational range that can be detected by listeners and alter listener perception of character and quality. Differences in frequency response and noise are carefully controlled while the distortion characteristics of the op-amps are amplified. Listening tests are performed in order to determine what differences listeners perceive. Listening tests also examine listener preference for different op-amps for the purpose of exploring what physical measurements best predict differences in perceived audio character and quality.
Convention Paper 8503

Tutorial 6 **Friday, Oct. 21**
9:00 am – 10:30 am **Room 1E11**

EAR TRAINING FOR THE ASPIRING AUDIO PROFESSIONAL

Presenter: **Mark Erickson**, Texas State University, San Marcos, TX, USA

There are no short-cuts. Improving your listening skills takes effort. Without direction, a budding audio professional is more likely to waste time and effort in search of better "ears." This tutorial will enable you to be more productive and reach your goals sooner, with less effort, resulting in a more positive growth experience. Techniques discussed can be integrated into a life-long quest to keep your aural skills sharp or continue your aural development.

Tutorial 7 **Friday, Oct. 21**
9:00 am – 10:30 am **Room 1E08**

FUNDAMENTALS OF AUDIO AND DATA NETWORKS OVER FIBER OPTICS AND CAT5 CABLING

Presenter: **Marc Brunke**, Optocore

It is a pre-requisite to be familiar with audio and data networking fundamentals when working with modern audio, video, and data transmission systems. This presentation clarifies the theory behind networking, starting with the basics of fiber and Cat5 cabling through to conceptual data transmission theories including network design and implementation. Different approaches and different ways of dealing with synchronization and jitter problems will be described, each point will be supported with an example from real life applications taking into consideration all present known available technologies.

Broadcast/Media Streaming Session 4
Friday, October 21 **9:00 am – 10:30 am**
Room 1E10

AUDIO PROCESSING FOR RADIO

Chair: **Steve Fluker**, Cox Radio

Panalists: *Frank Foti*, Omnia Audio
James J. Johnston, Consultant
Robert Orban, Orban
Randy Woods, WPOZ

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

Live Sound Seminar 4 **Friday, Oct. 21**
9:00 am – 11:00 am **Room 1E14**

AC POWER AND GROUNDING

Moderator: **Bruce Olson**, Olson Sound Design

Panelists: Bill Whitlock, Jensen Transformers
Jim VanBergen, NYC

There is much misinformation about what is needed for AC power at production events, much of it having to do with life-threatening advice. This panel will discuss how to properly and safely provide AC power for all concerned, without noise, hum or buzz. The discussion will start at the outlet working backwards to the source of power, while covering systems from a couple of speakers on sticks up to multiple stages in ballrooms, road houses, and event centers. This discussion will include 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.

Game Audio Session 5 **Friday, Oct. 21**
9:00 am – 10:30 am **Room 1E13**

TECHNIQUES IN FIELD RECORDING

Presenters: **Colin Hart**, hartFX
Ric Viers, The Detroit Chop Shop

This tutorial discusses techniques in field recording with an emphasis on game-specific source recording including examples in several areas of location or field recording. Topics include basic gear and concepts in field recording, session documentation, how to get a good sound, gaining access to locations, and field recording on a budget.

Product Design Session 3 **Friday, Oct. 21**
9:00 am – 11:00 am **Room 1E12**

HIGH RESOLUTION AUDIO—A NETWORKED FUTURE?

Chair: **Vicki Melchior**, Technical Consultant, MA, USA

Panelists: John Dawson, Arcam, Cambridge, UK
Aaron Gelter, Harman International, UT, USA
Steven Harris, BridgeCo (SMSC), UK

Though traditionally built on point-to-point connections, high res home audio and video systems are evolving toward full or partial networking. The tantalizing promise of internet access and server sourcing, disc as well as streamed A/V, and easy interconnection are driving more and more audio design in this direction. That includes multichannel audio, home theater, and even stereo. Nevertheless, networks in this market must integrate with traditional A/V and at the same time promote low jitter, high stability DAC clocks, together with appropriately low data delay, tight clock synch between devices, and excellent grounding and shielding.

The panel in this workshop discuss a number of promising standards and protocols (e.g., AVB, HDMI 1.4, Thunderbolt, plus standard uPNP/DLNA Ethernet, wi-fi), and consider their potential and promise relative to current forms of interfacing.

Special Event
YESTERDAY, TODAY, AND FOREVER: THE ART AND SCIENCE BEHIND THE MOTOWN AND VERVE CATALOG REISSUES
Friday, October 21, 9:00 am – 10:30 am
Room 1E15/16

Presenters: **Kevin Reeves**

Andy Skurow
Harry Weinger

The panel will discuss: What is catalog, and how are reissues conceptualized? The vault system: What is the process of finding the assets? And talk about how the technical engineering brings all the elements together for the final package.

Session P11 **Friday, Oct. 21**
9:30 am – 11:00 am **1E Foyer**

POSTERS: PRODUCTION AND ANALYSIS OF MUSICAL SOUNDS

9:30 am

P11-1 Filling the Gaps between the Grains of Down-Pitching PSOLA or Getting the Frog Out of PSOLA—*Adrian von dem Knesebeck, Udo Zölzer*, Helmut-Schmidt-University Hamburg, Hamburg, Germany

An improvement regarding the down-pitching quality of the Pitch Synchronous Overlap Add (PSOLA) technique is presented. The behavior of the common PSOLA algorithm when decreasing the input signal's pitch by one octave or even less is analyzed. A new grain processing algorithm is proposed, which modifies the synthesis grains depending on the synthesis period. The time domain and frequency domain properties of the proposed algorithm are discussed. The presented algorithm improves the perceived quality of the PSOLA algorithm when down-pitching while preserving the low complexity.
Convention Paper 8504

9:30 am

P11-2 Content-Based Approach to Automatic Recommendation of Music—*Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

This paper presents a content-based approach to music recommendation. For this purpose a database that contains more than 50000 music excerpts acquired from public repositories was built. Datasets contain tracks of distinct performers within several music genres. All music pieces were converted to mp3 format and then parameterized based on MPEG-7, mel-cepstral, and time-related dedicated parameters. All feature vectors are stored as csv files and will be available on-line. A study of the database statistical characteristics was performed. Different splits into train and test sets were investigated to provide the most accurate evaluation of the decision-based solutions. Classification time and memory complexity were also evaluated.
Convention Paper 8505

9:30 am

P11-3 A Digital Emulation of the Boss SD-1 Super Overdrive Pedal Based on Physical Modeling—*Martin Holters, Kristjan Dempwolf, Udo Zölzer*, Helmut Schmidt University, Hamburg, Germany

The Boss SD-1 Super Overdrive effect pedal is a classical overdrive circuit for electric guitars. It consists of commonly found building blocks, namely common collector circuits for input and output buffering, the distortion stage as an op-amp circuit with diodes in the feed-back, and a tone-control block as a parametric linear circuit around an op-amp. In this paper we analyze the circuit to derive a digital model where carefully applied simplifications strike a good balance between faithful emulation and computational efficiency. Due to the generality of the analyzed sub-circuits, the results should also be easily transferable to the many similar sub-circuits found in other effect units.
Convention Paper 8506

9:30 am

P11-4 A Triode Model for Guitar Amplifier Simulation with Individual Parameter Fitting—*Kristjan Dempwolf, Martin Holters, Udo Zölzer*, Helmut Schmidt University, Hamburg, Germany

A new approach for the modeling of triodes is presented, featuring simple and physically-motivated equations. The mathematical description includes the replication of the grid current, which is a relevant parameter for the simulation of overdriven guitar amplifiers. If reference data from measurements of practical triodes is available, an individual fitting to the reference can be performed, adapting some free parameters. Parameter sets for individual models are given. To study the suitability for circuit simulations, a SPICE model is created and tested under various conditions. Results of the model itself and when embedded in SPICE simulations are presented and compared with measurements. It is shown that the equations characterize the properties of real tubes in good accordance.
Convention Paper 8507

9:30 am

P11-5 Dereverberation of Musical Instrument Recordings for Improved Note Onset Detection and Instrument Recognition—*Thomas Wilmering, Mathieu Barthet, Mark B. Sandler*, Queen Mary University of London, London, UK

In previous experiments it has been shown that reverberation affects the accuracy of onset detection and instrument recognition. Pre-processing a reverberated speech signal with dereverberation for automatic speech recognition (ASR), where reverberation also decreases efficiency, has been proven effective for mitigating this performance decrease. In this paper we present the results of an experimental study addressing the problem of onset detection and instrument recognition from musical signals in reverberant condition by pre-processing the audio material with a dereverberation algorithm. The experiments include four different onset detection techniques based on energy, spectrum, and phase. The instrument recognition algorithm is based on line spectral frequencies (LSF) and k-means clustering. Results show improvement in onset detection performance, particularly of the spectral-based techniques. In certain conditions we also observed improvement in instrument recognition.
Convention Paper 8508

9:30 am

P11-6 Dimensional Reduction in Digital Synthesizers GUI Using Parameter Analysis
—*Daniel Gómez*,[1] *Juan Sebastián Botero*[2]
[1]Universidad ICESI, Cali, Colombia
[2]Ypisilon Tech, Medellín, Colombia

Digital synthesizers have cognitive overload issues for interaction, specially for novice or intermediate users. A system is developed to generate a custom GUI based on timbre clustering and parameter data analysis of preset programs present in VST synthesizers. The result is a new interface with dynamic control of the amount of variables maintaining full functionality of synthesizer performance. This system is designed to adapt to the degree of knowledge and cognitive control of synthesizer parameters of the user. The results of using diverse clustering techniques for synthesizers sound analysis and an original statistical analysis of preset programs parameter data are exposed. Implications of the use of our system in real world scenarios are reviewed.
Convention Paper 8509

Student/Career Development Event
STUDENT RECORDING CRITIQUES
Friday, October 21, 9:30 am – 10:30 am
Room 1E06

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

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

Friday, Oct. 21 10:00 am Room 1E05
Technical Committee Meeting on Coding of Audio Signals

Special Event
BOHEMIAN: A CONVERSATION WITH JUDY COLLINS
Friday, October 21, 10:30 am – 11:30 am
Room 1E15/16

Presenters: **Judy Collins**
Jason King

It was once said that "if amethysts could sing, they would sound like Judy Collins." Indeed, Judy Collins has been a folk and pop music icon and a powerful influence on several generations of musicians and listeners. Her ethereal performances and top-selling recordings of songs like "Both Sides Now" and "Send in the Clowns" ultimately helped make songwriters like Joni Mitchell, Leonard Cohen, and Stephen Sondheim become household names. Now in the fifth decade of her career, Judy is about to release *Bohemian*, her new album on the heels of her critically acclaimed 2009 release *Paradise*. Judy chats with Jason King, the Artistic Director of NYU's Clive Davis Institute of Recorded Music, about her life, career, and the making of the new album. Clips will be played from the as yet unreleased new CD.

Exhibitor Seminar Friday, Oct. 21
10:45 am – 11:45 am Room 1E06

PMC: MASTERS OF AUDIO SERIES

From A (Abba) to Z (Led Zeppelin)

Presenter: **Leif Mases**, Maselec

The immensely successful producer/engineer Leif Mases will be giving us the inside story of engineering and producing albums from Led Zeppelin, Jeff Beck, and Black Sabbath. In latter years Leif has turned his experience and skills to designing and manufacturing a critically acclaimed line of high end audio tools (compressors, eqs, de-essers, peak limiters, etc.) for the most discerning engineers, whether it is for mastering, recording, or mixing.

Workshop 6 Friday, Oct. 21
11:00 am – 12:30 pm Room 1E10

POUND OF CURE OR OUNCE OF PREVENTION? AUDIO ASSET RESTORATION, MIGRATION, AND PRESERVATION

Chair: **Ed Outwater**, EO3 Consulting, Ketchum, ID, USA

Panelists: *Chuck Ainlay*, Engineer/Producer
Jeff Anthony, Iron Mountain Entertainment Services
Rob Jaczko, Berklee College of Music
Patrick Kraus, Warner Music Group
Bob Ludwig, Mastering Engineer

The task of properly archiving entertainment assets today goes well beyond safely storing film cans, tape boxes, hard drives, data tapes, or whatever physical format houses the content to be preserved. This is especially true in the case of music assets, where audio asset restoration, migration, and preservation are becoming more and more important as the recording industry undergoes a basic transformation in the way it operates. In today's up-loadable, down-loadable, cloud-based, e-delivery- centric music marketplace, the ability to preserve and prepare music assets for secure and immediate delivery directly from the Archive is a key capability, transforming it not only into an ultra- secure repository for valuable music assets, but also quite often the first link in today's digital supply chain. The desired end-product in either case, so important in a market increasingly focused re-packaging and re-releasing content, is an asset that can be successfully utilized for commercial or, perhaps ultimately, for cultural purposes. To keep pace with this rapidly changing marketplace, it has become necessary to utilize not only secure physical storage, but also digital migration, restoration, electronic delivery, and other studio operations to archive effectively.

This workshop/panel discussion will be a review and discussion of a basic set of principles and procedures involved in "rescuing" valuable assets that have not been properly archived, and, following that, a corollary set detailing "how to" properly archive audio assets from the outset. It will be presented by a panel of industry experts with differing perspectives and experiences related to archiving.

Master Class 2 **Friday, Oct. 21**
11:00 am – 1:00 pm **Room 1E08**

HUMAN HEARING 101—HOW IT WORKS (THE SHORT VERSION)

Presenter: **Jim (JJ) Johnston**

In this Master Class we will discuss a variety of subjects at a conceptual level, from Head Related Transfer Functions to the actual filters implemented in the human Cochlea. Along the way, loudness vs. intensity, the law of the first wavefront, localization, auditory masking, the effects of attention, and inter-sensory interaction will be discussed. When you are done with this class, you will have an understanding of how the ear does frequency analysis, and the way that that affects everything else that relates to the human auditory system, from localization, through masking, to the absolute threshold of hearing. In addition to the above, issues of envelopment, direct vs. diffuse sensation, and spatial perception will be discussed as time allows.

Broadcast/Media Streaming Session 5
Friday, October 21 **11:00 am – 1:00 pm**
Room 1E10

STREAMING AND ENCODING

Chair: **David Bialik**

Panalists: Todd Baker, SRS Lab
Kirk Harnack, Telos Alliance
James Johnston, Consultant
Steve Lyman, Dolby
Jan Nordmann, Fraunhofer USA
Greg Ogonowski, Orban
Geir Skaaden, DTS
William Waters, NewTek

This session will discuss various methods of streaming and encoding media and how it can best be delivered to the end user over various platforms.

Live Sound Seminar 5 **Friday, Oct. 21**
11:00 am – 1:00 pm **Room 1E14**

10 THINGS TO GET RIGHT

Moderator: **Karl Winkler**, Lectrosonics

Panelists: *Mark Frink*
Lloyd Kinkade
Bruce Olson
Bennett Prescott
Dave Shadoan

Sound reinforcement systems, whether temporary for touring purposes or fixed for installations, exhibit many of the same elements of success and failure. In order to avoid some of the common pitfalls, this panel aims to cover the 10 most important things about these systems that must be designed and installed correctly. Top practitioners from across the US will discuss these top issues, their philosophies, and pet peeves in audio system design.

Game Audio Session 6 **Friday, Oct. 21**
11:00 am – 12:30 pm **Room 1E13**

AUDIO AS A REAL-TIME INPUT AND FEEDBACK MECHANISM FOR FULL BODY GAMING

Presenters: **Chris Jahnkow**, Sony Computer Entertainment America
Scott Selfon, Microsoft

A panel of game console platform developers discusses the technologies and techniques allowing audio to become an input for games, as well as using it as an output/user feedback mechanism. Topics will include using sound to replace haptic feedback when no tactile system is used, echo cancellation and noise suppression, real-time audio analysis, and gameplay implementations driven by these dynamic and fully player-driven natural user input systems.

Special Event
PLATINUM PRODUCERS
Friday, October 21, 11:00 am – 1:00 pm
Room 1E15/16

Moderator: **David Weiss**, co-founder SonicScoop

Panelists: *Steve Jorda*n (The Verbs, Keith Richards, John Mayer, Rod Stewart, Beyoncé, Buddy Guy)
David Kahne (Sublime, Regina Spektor, Paul McCartney)
Gabe Roth, Founder, Daptone Records (Amy Winehouse, Sharon Jones and the Dap-Kings)

The Producer's Portfolio

Everyone agrees the artist hires the producer to serve the band or singer/songwriter and their music. This panel, however, will address the producer's personal artistic visions, and the growing bodies of work their creative philosophies pilot into reality. Considered a creative artistic force in their own right, each of these producers collaborates fully with their clients both in pre-production and the studio. Participants will explore the artistic sensibilities they've nurtured, how they've expressed themselves in their work, and how that self-assurance and unique perspective has enabled their careers to flourish.

Student/Career Development Event
CAREER/EDUCATION FAIR
Friday, October 21, 11:00 am – 12:30 pm
1E Foyer

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

Institutions offering studies in audio (from short courses to graduate degrees) will be represented in a "table top" **Education Fair**. 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.

Friday, Oct. 21 **11:00 am** **Room 1E05**
Technical Committee Meeting on Audio Recording and Mastering Systems

Product Design Session 4 **Friday, Oct. 21**
11:30 am – 1:00 pm **Room 1E12**

REAL-TIME AUDIO PROCESSING CAPABILITIES OF MICROCONTROLLERS AND APPLICATION PROCESSORS

Presenter: **Paul Beckmann**

Microcontrollers and application processors have been steadily improving in performance to the point where they are being seriously considered by many audio product developers. Although they don't have all of the features found in traditional DSP processors, their performance has been boosted via DSP specific instructions and ever increasing clock speeds. This workshop carefully compares the architectures of microcontrollers and application processors with traditional DSPs with an eye toward their suitability for real-time audio processing. Measurements of the performance of individual low-level functions and complete signal chains on actual hardware are presented and we show what types of products can be realistically supported by each processor type.

Session EB3 **Friday, Oct. 21**
12:00 noon – 1:00 pm **Room 1E09**

APPLICATIONS OF AUDIO ENGINEERING

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

12:00 noon

EB3-1 Foreign-Language Dubbing Practices—
Theodor Stojanov, Independent Writer and Post Production Sound Editor, Montreal, Quebec, Canada

The interdisciplinary nature of foreign language dubbing involves the collaboration of actors, directors, producers, translators, sound and video departments, and often develops into an elaborate production project in its own right. Increasingly, particularly with motion pictures, the dubbing industry faces new challenges as studios are required to work at the pace of the original production and follow any and all modifications, leading ultimately to the global release of a film everywhere simultaneously, an economic model that has been gathering momentum in recent years. The technology for foreign dialogue dubbing has undergone a great optimization of the production work-flow to accommodate such new requirements, and this presentation will show various practices currently in use throughout Europe and North America. I will discuss the idealized work-flow versus technology-specific solutions, cost optimization, media security, and the advantages of certain types of technologies on foreign-language synchronization. Areas of development in foreign-language dubbing and internationalization will be discussed.
Engineering Brief 30

12:15 pm

EB3-2 Application of Wave Field Synthesis and Analysis on Automotive Audio—*Michael J. Strauß*,[1] *Peter Gleim*[2]
[1]Fraunhofer IDMT, Ilmenau, Germany
[2]Audi AG, Ingolstadt, Germany

High-quality audio equipment in cars enjoys rising popularity. For a remarkable number of people the car compartment represents the primary listening environment for enjoying music playback. Besides excellent sound quality spatial audio capabilities can also be expected from today's top systems. In this talk we will give insight into the implementation of a spatial audio system based on Wave Field Synthesis that was realized inside an SUV. Audi AG and Fraunhofer IDMT together present the outcome of their research collaboration named "Audi Sound Concept." The Audi Q7 prototype has, on the one hand, remained a series production vehicle on the exterior, while on the other hand been rebuilt into a HiFi-Studio in the interior. The talk will include a systematic overview, a description of the playback-system, and some remarks about sound field analysis based on array measurements.
Engineering Brief 31

12:30 pm

EB3-3 Spooky Sounds: Interactive Audio Systems and Design for a Themed Attraction in an Academic Environment—*Bruce Ellman*,[1] *John Huntington*[2]
[1]NYC College of Technology/CUNY, Sunnyside, NY, USA
[2]NYC College of Technology/CUNY, Brooklyn, NY, USA

The high-tech, interactive Gravesend Inn haunted hotel attraction features a large, distributed, audience-triggered sound system to implement a design both startling and evocative. Sound Designer Bruce Ellman and Systems Engineer John Huntington will discuss the challenges faced in developing this system and using it to teach Entertainment Technology Students.
Engineering Brief 32

12:45 pm

EB3-4 High Performance Architectural and Electro Acoustic Isolation Solutions—*Dave Kotch, John Storyk*, Walters-Storyk Design Group

Increasing demands for community noise abatement, most specifically for architectural spaces such as night clubs and performance venue, have resulted in a variety of interesting design solutions to insure high performance sound attenuation, with FSTC results in excess of 80dB. This has often been accomplished with a combination of architectural construction design as well as electro-acoustic systems design such as directional subs, low frequency harmonic processing, and other systems integration devices. This paper will explore these designs and recent field results.
Engineering Brief 33

Exhibitor Seminar **Friday, Oct. 21**
12:00 noon – 1:00 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

Intelligent Dance Music in 5.1

Presenter: **David Miles Huber**

DMH is a 2 x Grammy-nominated (albums *Colabs* and *Paralax Eden*) producer and musician in the electronic IDM, dance, and surround-sound genres, whose music has sold over one million copies. David will take the audience on a musical journey and present his latest Grammy nominated project *Chamberland*. For further info on David and his work check out www.davidmileshuber.com

Friday, Oct. 21 **12:00 noon** **Room 1E02**
Standards Committee Meeting on Audio Connectors, SC-05-02

Friday, Oct. 21 **12:00 noon** **Room 1E05**
Technical Committee Meeting on Transmission and Broadcasting

Lunchtime Keynote
KARLHEINZ BRANDENBURG
Friday, October 21, 1:00 pm – 2:00 pm
Room 1E10

The MP3 Story

In the not-too-distant past we were faced with the challenge of transmitting high-quality audio over phone lines. While this seemed impossible at the time, ideas from psychoacoustics and signal processing and work by many researchers helped the seemingly impossible to become reality: mp3 and other audio codecs enabled the seamless transport of audio over legacy copper phone lines. However, the mp3 story did not end there. The internet was being transformed from a text-based medium into a major carrier for sound of all kinds, including music. This meant changes not only for the payload (from text to audio), but also new dangers for the audio quality delivered to music lovers. And it changed business models for music sales dramatically, shaking the foundations of the music industry. This is the story of MP3.

Student/Career Development Event
SPARS SPEED COUNSELING WITH EXPERTS—MENTORING ANSWERS FOR YOUR CAREER
Friday, October 21, 1:00 pm – 2:15 pm
Room 1E13

Counselors: **Niko Bolas**
Flash Ferruccio
David Glasser
Neil Goldberg
Roy Hendrickson
Eric Johnson
Fred Kevorkian
Lawrence Manchester
Randy Merrill
Leslie Mona-Mathus
Tom Paul
Mark Rubel
Tom Salta
Rick Senechal
Shaun Wall

This seminar is specially suited for students, recent graduates, and those interested in career advice. Hosted by SPARS in cooperation with AES, G.A.N.G., Post NY Alliance, and other audio organizations, career related Q&A sessions will be offered to participants in a speed group counseling format. A dozen students will interact with 2–3 working professionals in specific audio engineering fields or categories every 15 minutes. Audio engineering fields/categories include record production, mastering, audio post, film music, gaming, and live sound.

Exhibitor Seminar **Friday, Oct. 21**
1:00 pm – 2:00 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

How Low Can You Go……..?

Presenter: **Darius "Deezle" Harrison**

Deezle will present on the importance of "true" low end, either from an 808, acoustic bass, or kick drum and how it determines the sound of the music. Deezle is a 4-time ASCAP Rhythm and Soul Award and a B.E.T Award winner. He is best known for his work with Lil Wayne, which ultimately garnered him 2 Grammy Awards for Best Rap Album and Best Rap Song for "Lollipop." Deezle has worked with Outkast, Ludacris, Mary J Blige, Usher, Donald Harrison, Jay-Z, and Kerry Hilson. While finishing tracks for the new DMX album, Deezle also found time to produce an album for the immensely talented singer/actress Keke Palmer.

Friday, Oct. 21 **1:00 pm** **Room 1E05**
Technical Committee Meeting on Audio for Games

Hot Lunch
MICROPHONE CONTROLLER FOR VOCAL FX
Friday, October 21, 1:30 pm – 2:15 pm
Room 1E12

Presenter: **Daniel Schlessinger**

A prototype vocal microphone has been developed that offers intimate and expressive control over vocal effects. For instance, vocalists can control looping or reverb effects from push buttons on the mic or even pan his/her voice between a pair of outputs simply by moving the mic. With this prototype, our idea is to put controls where the vocalist can easily reach them, letting them map the buttons and fader and motion sensors via MIDI to any outboard effects unit of their choice. Without being tied to a foot pedal or having to reach down to the front panel of an effects box, the artists are provided with additional creative possibilities with a complete freedom on stage to control their sound.

Friday, Oct. 21 **1:30 pm** **Room 1E02**
Standards Committee Meeting on Grounding and EMC Practices, SC-05-05

Session P12 **Friday, Oct. 21**
2:00 pm – 4:00 pm **Room 1E09**

LOUDSPEAKER REPRODUCTION

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

2:00 pm

P12-1 Size and Shape of Listening Area Reproduced by Three-Dimensional

Multichannel Sound System with Various Numbers of Loudspeakers—*Ikuko Sawaya, Satoshi Oode, Akio Ando, Kimio Hamasaki*, NHK Science and Technology Research Laboratories, Setagaya, Tokyo, Japan

A wide listening area is necessary so that several people can listen to a multichannel sound program together. It is considered that the size of the listening area depends on the number of loudspeakers. To examine the relationship between the number of loudspeakers and the size of listening area that maintains spatial impression at the center of a three-dimensional multichannel sound, two subjective evaluation experiments were carried out. The first experiment showed that the size of the listening area increases by increasing the number of loudspeakers. The second experiment showed that the shape of the listening area is dependent on the locations of loudspeakers. On the basis of the experimental results, a new parameter for estimating the shape of listening area is proposed.
Convention Paper 8510

2:30 pm

P12-2 Numerically Optimized Touring Loudspeaker Arrays—Practical Applications—*Ambrose Thompson, Jason Baird, Bill Webb*, Martin Audio, High Wycombe, UK

We describe the implementation of a user guided automated process that improves the quality and consistency of loudspeaker array deployment. After determining basic venue geometry a few easily understood goals for regions surrounding the array are specified. The relative importance of the goals is then communicated to the optimization algorithm in an intuitive manner. Some representative examples are presented, initially optimized with default coverage goals. We then impose extra requirements such as changing the coverage at the last moment, avoiding noise sensitive regions and demanding a particularly quiet stage.
Convention Paper 8511

3:00 pm

P12-3 Vertical Loudspeaker Arrangement for Reproducing Spatially Uniform Sound—
Satoshi Oode,[1] *Ikuko Sawaya,*[1] *Akio Ando,*[1] *Kimio Hamasaki,*[1] *Kenji Ozawa*[2]
[1]Japan Broadcasting Corporation, Setagaya, Tokyo, Japan
[2]University of Yamanashi, Kofu, Yamanashi, Japan

It was recently recognized that the loudspeaker arrangement of multichannel sound systems can be vertically expanded to improve the spatial impression. This paper discusses the relationship between the vertical interval between loudspeakers placed in a semicircle above ear height and the impression of spatially uniform sound, as part of a study of three-dimensional multichannel sound systems. In total, 24 listeners evaluated the spatial uniformity of white noise reproduced by loudspeakers arranged in vertical semicircles at equal intervals of 15°, 30°, 45°, 60°, 90° or 180°, and with azimuthal angles of 0°, 45°, or 90°. Loudspeakers arranged with vertical intervals that were less than 45° were found to reproduce spatially uniform sound for all of the azimuthal angles tested.
Convention Paper 8512

3:30 pm

P12-4 Multichannel Sound Reproduction in the Environment for Auditory Research—*Mark A. Ericson*, Army Research Laboratory, Aberdeen Proving Ground, MD, USA

The Environment for Auditory Research (EAR) is a new U.S. Army Research Laboratory facility at Aberdeen Proving Ground, Maryland, dedicated to spatial sound perception and speech communication research. The EAR is comprised of four indoor research spaces (Sphere Room, Dome Room, Distance Hall, and Listening Laboratory), one outdoor research space (Open EAR), and one common control center (Control Room). Digital audio signals are routed through state-of-the-art RME Hammerfall DSP and Peavey MediaMatrix® hardware to over 600 loudspeakers and microphone channels throughout the facility. The facility's acoustic environments range from anechoic, through various soundscapes, to real field environments. The EAR facility layout, the audio signal processing capabilities, and some current research activities are described.
Convention Paper 8513

Session P13 — Friday, Oct. 21
2:00 pm – 4:30 pm — Room 1E07

LOW BIT-RATE CODING—PART 1

Chair: **Stephan Preihs**, Leibniz Universität Hannover, Hannover, Germany

2:00 pm

P13-1 Performance of MPEG Unified Speech and Audio Coding—*Schuyler Quackenbush,*[1] *Roch Lefebvre*[2]
[1]Audio Research Labs, Scotch Plains, NJ, USA
[2]Université de Sherbrooke, Sherbrooke, Quebec, Canada

The MPEG Unified Speech and Audio Coding (USAC) standard completed technical development in July, 2011 and is expected to issue as international standard ISO/IEC 23003-3, Unified Speech and Audio Coding in late 2011. Verification tests were conducted to assess the subjective quality of this new specification. Test material consisted of 24 items of mixed speech and music content and performance was assessed via subjective listening tests at coding rates ranging from 8 kb/s for mono material to 96 kb/s for stereo material. The mean performance of USAC was found to be better than that of MPEG High Efficiency AAC V2 (HE-AAC V2) and Adaptive Multi-Rate Wide Band Plus (AMR-WB+) at all tested operating points. For most bit rates tested this performance advantage is large. Furthermore, USAC provides a much more consistent quality across all signal content types than the other systems tested. This paper summarizes the results of the verification tests.
Convention Paper 8514

2:30 pm

P13-2 Improved Error Robustness for Predictive Ultra Low Delay Audio Coding—*Michael Schnabel*,[1] *Michael Werner*,[2] *Gerald Schuller*[1]
[1]Ilmenau University of Technology, Ilmenau, Germany
[2]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

This paper proposes a new method for improving the audio quality of predictive perceptual audio coding in the context of the Ultra Low Delay (ULD) coding scheme for real time applications. The commonly used auto-regressive (AR) signal model is leading to an IIR predictor in the decoder. For random access of the transmission as well as for transmission errors, a reset of the predictor states, in both encoder and decoder is used. The resets reduce the prediction performance and thus the SNR, especially during stationary signal parts, since the resulting noise peaks could become audible. This paper shows that using adaptive reset intervals, that are chosen according to psychoacoustic rules, improves the audio quality.
Convention Paper 8515

3:00 pm

P13-3 AAC-ELD v2—The New State of the Art in High Quality Communication Audio Coding—*Manfred Lutzky, María Luis Valero, Markus Schnell, Johannes Hilpert*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Recently MPEG finished the standardization of a Low Delay MPEG Surround tool that is tailored for enhancing the widely adopted AAC-ELD low delay codec for high-quality audio communication into AAC-ELD v2. In combination with the Low Delay MPEG Surround tool, the coding efficiency for stereo content outperforms competing low delay audio codecs at least by a factor of 2. This paper describes the technical challenges and solutions for designing a low delay codec that delivers a performance that is comparable to that of existing state of the art compression schemes featuring much higher delay, such as HE AAC v2. It provides a comparison to competing proprietary and ITU-T codecs, as well as a guideline for how to select the best possible points of operation. Applications facilitated by AAC-ELD v2 in the area of broadcasting and mobile video conferencing are discussed.
Convention Paper 8516

3:30 pm

P13-4 QMF-Based Harmonic Spectral Band Replication—*Haishan Zhong*,[1] *Lars Villemoes*,[2] *Per Ekstrand*,[2] *Sascha Disch*,[3] *Frederik Nagel*,[3] *Stephan Wilde*,[3] *Kok Seng Chong*,[1] *Takeshi Norimatsu*[1]
[1]Panasonic Corporation
[2]Dolby Sweden AB
[3]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Unified speech and audio coding (USAC) is the next step in the evolution of audio codecs that are standardized by the Moving Picture Experts Group (MPEG). USAC provides consistent quality for music, speech and mixed material, by extending the strength of an audio codec by speech codec functions. Two novel flavors of Spectral Band Replication (SBR) were introduced to enhance the perceptual quality of SBR: the Discrete Fourier Transform (DFT) based and the Quadrature Mirror Filterbank (QMF) harmonic SBR. The DFT-based SBR has higher frequency resolution for the harmonic transposition process, resulting in good sound quality. The QMF-based SBR has significantly lower computational complexity. This paper describes the detailed technical aspects of the low complexity QMF-based harmonic SBR tool within USAC. A complexity comparison and the listening test results are also presented in the paper.
Convention Paper 8517

4:00 pm

P13-5 Perceptually Optimized Cascaded Long Term Prediction of Polyphonic Signals for Enhanced MPEG-AAC—*Tejaswi Nanjundaswamy, Kenneth Rose*, University of California Santa Barbara, Santa Barbara, CA, USA

MPEG-4 Advanced Audio Coding uses the long term prediction (LTP) tool to exploit inter-frame correlations by providing a segment of previously reconstructed samples as prediction for the current frame, which is naturally useful for encoding signals with a single periodic component. However, most audio signals are polyphonic in nature containing a mixture of several periodic components. While such polyphonic signals are themselves periodic with overall period equaling the least common multiple of the individual component periods, the signal rarely remains sufficiently stationary over the extended period, rendering the LTP tool ineffective. Further hindering the LTP tool is the typically employed parameter selection based on minimizing the mean squared error as opposed to the perceptual distortion criteria defined for audio coding. We thus propose a technique to exploit the correlation of each periodic component with its immediate past, while taking into account the perceptual distortion criteria. Specifically, we propose cascading LTP filters corresponding to individual periodic components, designed appropriately in a two stage method, wherein an initial set of parameters is estimated backward adaptively to minimize the mean squared prediction error, followed by a refinement stage where parameters are adjusted to minimize the perceptual distortion. Objective and subjective results validate the effectiveness of the proposal on a variety of polyphonic signals.
Convention Paper 8518

Workshop 7 **Friday, Oct. 21**
2:00 pm – 3:30 pm **Room 1E08**

HIGHLY-DIRECTIONAL MICROPHONES FOR SOUND RECORDING

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

Panelists: *Gary Elko*, mh acoustics
Christof Faller, Illusonic LLC
Michael Militzer, Microtech Gefell
Klas Strandberg, Telinga

Times are changing quickly with regard to highly directional microphones both in theory and in practice. Most systems aim at optimal speech transmission and intelligibility in communication applications. However, there are also new approaches that sound engineers can consider using for certain recording purposes as well (e.g. the Eigenmike® array, the KEM 970, and the SuperCMIT). Along with improved communication microphone solutions, they will compete with conventional methods such as short and long interference tubes as well as parabolic mirrors.

So which method fits which application best? Can these new microphone types be used for applications such as music recording, location sound, nature recording? Each has different advantages with regard to frequency response, directivity, flexibility, price, robustness, predictability of results, and practicability. The workshop will compare them on the basis of technical data, practical experiences and sample recordings.

Live Sound Seminar 6 — Friday, Oct. 21
2:00 pm – 4:00 pm — Room 1E14

THEATRICAL MICROPHONE DRESSING

Presenter: **Mary McGregor**

Fitting actors with wireless microphone elements and wireless transmitters has become a detailed art form. From ensuring the actor is comfortable and the electronics is safe and secure, to getting the proper sound with minimal detrimental audio effects, all while maintaining the visual illusion, 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."

Exhibitor Seminar — Friday, Oct. 21
2:00 pm – 3:00 pm — Room 1A01

COMMUNITY PROFESSIONAL LOUDSPEAKERS

dSPEC – More Than Just DSP. System Design in Zero Time

Presenters: **Chris Barrow** and **Ron Neely**

Community introduces its first ever Networked Loudspeaker Processor, dSPEC226, and its companion software Resyn. Learn about a faster, easier way to design and commission a DSP-based Community Loudspeaker System. Learn about advanced loudspeaker protection features, CONEQ-enhanced audio with FIR filters, and the Resyn Engineered Workflow featuring Effortless Ethernet.

Exhibitor Seminar — Friday, Oct. 21
2:00 pm – 3:00 pm — Room 1E06

PMC: MASTERS OF AUDIO SERIES

If You Have Them, Use Them…..Properly!

Presenter: **Ronald Prent**

With the aid of what he calls "a choreography for six speakers," the multi award-winning surround sound engineer Ronald Prent shows how music in surround can be a source of great emotion. Building a mix from scratch, Ronald will deliver insight into the creative skill set required to make a great surround mix. Using multiple examples from his ongoing career in creative mixing he will provide an entertaining hour of music!

Friday, Oct. 21 — 2:00 pm — Room 1E05
Technical Committee Meeting on Spatial Audio

Session P14 — Friday, Oct. 21
2:30 pm – 4:00 pm — 1E Foyer

POSTERS: AUDIO PROCESSING

2:30 pm

P14-1 Acoustic Channel Decorrelation with Phase Modification for Stereo Acoustic Echo Cancellation—*Jae-Hoon Jeong, So-Young Jeong, Woo-Jeong Lee, Jung-Eun Park, Jeong-Su Kim, Yongje Kim*, Samsung Electronics Co., Ltd., Suwon, Korea

In this paper we propose a novel acoustic channel decorrelation method, which prevents poor convergence in stereophonic echo cancellation. In order to minimize audio quality degradation while maximizing channel decorrelation, the proposed method provides a subband phase modification of each channel, which depends on the amount of inter-channel phase difference so that redundant phase alteration is avoided. In addition, we introduce a phase control parameter for each subband to preserve perceptual stereo image of each channel. The performances of the proposed method are verified with MUSHRA subjective audio quality test, impulse response misalignment, and echo return loss enhancement. The results show that the proposed method has a good decorrelation performance while minimizing signal distortion and stereo image change.
Convention Paper 8519

2:30 pm

P14-2 Least-Squares Local Tuning Frequency Estimation for Choir Music—*Volker Gnann, Markus Kitza, Julian Becker, Martin Spiertz*, RWTH Aachen University, Aachen, Germany

Choir conductors often have to deal with the problem that the tuning pitch of a choir tends to decrease gradually over time. For that reason, we present an algorithm that measures and displays the tuning frequency evolution for polyphonic choir music over time. Basically, it analyzes the short-time Fourier spectrogram, picks out the most important peaks, and determines their frequencies. From these frequencies, the algorithm calculates the concert A pitch that leads to the smallest least-squares-error when the measured frequencies are sorted into a semitone grid.
Convention Paper 8520

2:30 pm

P14-3 A Low Latency Implementation of a Non Uniform Partitioned Overlap and Save

Algorithm for Real Time Applications—
Andrea Primavera, Stefania Cecchi, Laura Romoli, Paolo Peretti, Francesco Piazza,
Universita Politecnica delle Marche, Ancona, Italy

FIR convolution is a widely used operation in the digital signal processing field, especially for filtering operations in real time scenarios. In this context low computationally demanding techniques for calculating convolutions with low input/output latency become essential, considering that the real time requirements are strictly related to the impulse response length. In this paper a multi-threaded real time implementation of a Non Uniform Partitioned Overlap and Save algorithm is proposed with the aim of lowering the workload required in applications like reverberation, also exploiting the human ear sensitivity. Several results are reported in order to show the effectiveness of the proposed approach in terms of computational cost, taking into consideration different impulse responses and also introducing comparisons with existing techniques of the state of the art.
Convention Paper 8521

2:30 pm

P14-4 3-D Audio Depth Rendering Method for 3-DTV —*Sunmin Kim, Young Woo Lee, Yongje Kim*,
Samsung Electronics Co., Ltd., Suwon, Korea

This paper proposes a novel 3-D audio depth rendering method with stereo loudspeakers in order to enhance an immersion of 3-D video contents. The 3-D audio depth rendering system for 3-DTV consists of an audio depth index that estimates the distance of an audio object between TV and a listener, and a distance control algorithm based on the audio depth index. Two kinds of audio depth index estimation algorithms are presented. One utilizes a disparity map of stereo image, and the other tracks loudness of stereo audio signal. Listening tests show that the proposed audio depth rendering system allows the listener to feel the depth of the sound corresponding to a pop-up effect of the 3-D image.
Convention Paper 8522

2:30 pm

P14-5 Virtual Height Speaker Rendering for Samsung 10.2-Channel Vertical Surround System—*Young Woo Lee*,[1] *Sunmin Kim*,[1] *Hyun Jo*,[2] *Youngjin Park*,[2] *Yongje Kim*[1]
[1]Samsung Electronics Co., Ltd., Suwon, Korea
[2]KAIST, Daejeon, Korea

This paper proposes the virtual sound elevation rendering algorithm that can give a listener an impression of virtual 10.2 channel loudspeakers. The proposed algorithm requires 10.2 channel input signals and the conventional 7.1 channel loudspeaker system (ITU-R BS.775-2). The proposed virtual height speaker rendering consists of generic head-related transfer function (HRTF), which is calculated using 45 individualized HRTFs of CIPIC datasets, and a mixing algorithm using four loudspeakers among 7.1 channels. For subject evaluation, three kinds of playbacks are compared with various materials: original 10.2 channel signal, down-mixed 7.1 channel signal, and the proposed 7.1 channel signal in terms of source positioning, envelopment, and overall sound quality.
Convention Paper 8523
[Paper presented by Sunmin Kim]

2:30 pm

P14-6 Non Linear Convolution and its Application to Audio Effects—*Lamberto Tronchin*,
University of Bologna, Bologna, Italy

The non linear convolution could be applied to enhance the linear convolution on acoustic musical instruments and audio devices. In this paper a novel technology, based on exponential sine sweeps measurements, is presented. The non linear convolution is based on the Volterra series approach and enables real time non linear emulation of acoustic devices (as valve amplifiers and musical instruments). The new developed tool (a VST plugin developed in C++ in JUCE framework) will be presented. The emulation of musical instruments will be compared with real recordings. The results will be finally analyzed and commented.
Convention Paper 8524

Tutorial 8 **Friday, Oct. 21**
2:45 pm – 4:15 pm **Room 1E12**

DEMYSTIFYING AUDIO SOUND CONTROL PROTOCOLS

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

Starting from an introduction to the concepts of network discovery, control, and connection management and an examination of existing approaches including MIDI, AES24, OSC, IEC 62379, and XFN among others, this tutorial on the XFN connection management and control protocol presents core features concepts and functionality. There will be a discussion of options for achieving interoperability between devices using XFN as well as other protocols. A demonstration will show how XFN is used in conjunction with a graphical interface to perform discovery, connection management and control of network audio devices. A protocol essentially similar to the XFN protocol has been proposed for standardization in the AES under project AES-X170. Attendees will be updated on the status and progress of this project.

Broadcast/Media Streaming Session 6
Friday, October 21 **2:30 pm – 4:00 pm**
Room 1E10

TELEVISION LOUDNESS AND METADATA

Chair: **Skip Pizzi**, NAB

Panelists: *Richard Cabot*, Qualis Audio
Paul Keller, Harris Corp.
Steve Lyman, Dolby
Andrew Mason, BBC R&D
Stephan Schreiner, Fraunhofer IIS
Robert Seidel, CBS

Digital television is now well established in the US market, where most analog TV broadcasting was eliminated in 2009. The ATSC DTV audio system includes numer-

ous features not previously available, including the ability to normalize audio loudness at the receiver (via the AC-3 dialnorm metadata feature). This process has not been uniformly applied, however, causing numerous listener complaints that ultimately resulted in federal legislation mandating the matching of TV commercial and program audio loudness. This new law—the Commercial Audio Loudness Mitigation (CALM) Act—is expected to become effective in January 2013, and FCC rules are currently being crafted toward its implementation, with significant input from ATSC and the industry on how to best comply with the Act's requirements. Learn the latest about controlling TV audio loudness and complying with the new mandates, from broadcasters, manufacturers, and vendors in this timely and important session.

Game Audio Session 7 **Friday, Oct. 21**
2:30 pm – 4:00 pm **Room 1E11**

EMERGING MARKET: PROCESSING PLUG-IN'S FOR GAMES

Presenters: **Mike Caviezel**, Microsoft
Alex Westner, iZotope

Game Audio Systems are rapidly approaching the processing capabilities of DAW's. Real-time processing in games has become more than just reverb and a bit of low-pass. There are many uses for real-time DSP and as in the DAW, there are many flavors of plug-ins that can accomplish similar tasks. In the past developers have been happy, pleased, even surprised to get a filter, or a reverb, or a limiter. Now we are starting to see options in the types of plug-ins integrated into today's game audio pipelines. Plug-in developers are starting to see a potential new market for their technology and some of them have already been working with game developers to incorporate these technologies. This session looks at some of these Plug-in/Game developer relationships and how they can benefit both parties.

Special Event
BEN FOLDS—THE BEST IMITATION OF MYSELF
Friday, October 21, 2:30 pm – 4:15 pm
Room 1E15/16

Presenters: **Ben Folds**
Errol Kolosine

International recording star Ben Folds discusses his career, influences, and the music business, as well as his new retrospective, Best Imitation of Myself, with Errol Kolosine, followed by a Q & A with attendees. We'll examine his time with The Ben Folds Five, as a solo artist, and also as a collaborator.

Friday, Oct. 21 **3:00 pm** **Room 1E02**
Standards Committee Meeting on Digital Library and Archive Systems and Audio Metadata (joint meeting), SC-03-06 and SC-03-07

Exhibitor Seminar **Friday, Oct. 21**
3:30 pm – 4:30 pm **Room 1A01**

THAT CORPORATION

De-Integrating Integrated Circuit Preamps

Presenter: **Les Tyler**

For many years, IC suppliers—including THAT—have offered several highly integrated analog microphone preamplifiers. These ICs are convenient and easy to use, and generally include the input-stage feedback resistors plus an output differential amplifier. While convenient, these included elements reduce the preamps' flexibility and restrict performance in some areas compared with that obtainable from discrete designs. In this seminar, we present a new-generation of analog IC microphone preamplifiers that offer more flexibility and higher performance through selective omission of internal items. We will review the benefits and detriments of this approach, and offer many practical tips on how to get the most out of these new ICs.

Exhibitor Seminar **Friday, Oct. 21**
3:30 pm – 4:30 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

Breaking the Rules—Surround Sound Recording Part 1

Presenter: **Morten Lindberg**

Surround Sound is a sculpture, where stereo can be described as a flat canvas.

Learn how to record in surround from a true master. Morten Lindberg is a 9-time Grammy nominated / winning balance engineer and producer with vocals, choirs and strings as his core area of expertise.

Workshop 8 **Friday, Oct. 21**
3:45 pm – 4:45 pm **Room 1E08**

THE MOBILE GENERATION OF MUSIC CREATION AND PRODUCTION

Chair: **Jay LeBoeuf**, Imagine Research

Panelists: *Mark Ethier*, iZotope
Michael Gitig, Gobbler
Henrik Lenberg, SoundCloud

Thanks to mobile devices, cloud computing, and innovative software algorithms, we are seeing a further democratization of the music creation and production process. Users of innovative iPad/iPhone applications, cloud-based audio sharing, and collaboration sites are turning virtually everyone into a content producer. This panel explores the products, technology, and external factors that are enabling this revolution.

Session P15 **Friday, Oct. 21**
4:00 pm – 6:30 pm **Room 1E09**

SOUND FIELD ANALYSIS AND REPRODUCTION—PART 2

Chair: **Juha Merimaa**, Sennheiser Research Laboratory, Palo Alto, CA, USA

4:00 pm

P15-1 Broadband Analysis and Synthesis for Directional Audio Coding Using A-Format Input Signals—*Archontis Politis, Ville Pulkki*, Aalto University, Espoo, Finland

Directional Audio Coding (DirAC) is a parametric

non-linear technique for spatial sound recording and reproduction, with flexibility in terms of loudspeaker reproduction setups. In the general 3-dimensional case, DirAC utilizes as input B-format signals, traditionally derived from the signals of a regular tetrahedral first-order microphone array, termed A-format. For high-quality rendering, the B-format signals are also exploited in the synthesis stage. In this paper we propose an alternative formulation of the analysis and synthesis, which avoids the effect of non-ideal B-format signals on both stages, and achieves improved broadband estimation of the DirAC parameters. Furthermore, a scheme for the synthesis stage is presented that utilizes directly the A-format signals without conversion to B-format.
Convention Paper 8525

4:30 pm

P15-2 Beamforming Regularization, Scaling Matrices, and Inverse Problems for Sound Field Extrapolation and Characterization: Part I—Theory—*Philippe-Aubert Gauthier, Éric Chambatte, Cédric Camier, Yann Pasco, Alain Berry*, Université de Sherbrooke, Sherbrooke, Québec, Canada, and McGill University, Montreal, Québec, Canada

Sound field extrapolation (SFE) is aimed at the prediction of a sound field in an extrapolation region using a microphone array in a measurement region. For sound environment reproduction purposes, sound field characterization (SFC) aims at a more generic or parametric description of a measured or extrapolated sound field using different physical or subjective metrics. In this paper an SFE method recently introduced is presented and further developed. The method is based on an inverse problem formulation combined with a beamforming matrix in the discrete smoothing norm of the cost function. The results obtained from the SFE method are applied to SFC for subsequent sound environment reproduction. A set of classification criteria is proposed to distinguish simple types of sound fields on the basis of two simple scalar metrics. A companion paper presents the experimental verifications of the theory presented in this paper.
Convention Paper 8526

5:00 pm

P15-3 Beamforming Regularization, Scaling Matrices and Inverse Problems for Sound Field Extrapolation and Characterization: Part II—Experiments— *Philippe-Aubert Gauthier, Éric Chambatte, Cédric Camier, Yann Pasco, Alain Berry*, Université de Sherbrooke, Sherbrooke, Québec, Canada, and McGill University, Montreal, Québec, Canada

Sound field extrapolation (SFE) is aimed at the prediction of a sound field in an extrapolation region using microphone array. For sound environment reproduction purposes, sound field characterization (SFC) aims at a more generic or parametric description of a measured or extrapolated sound field using different physical or subjective metrics. In this paper experiments with a recently-developed SFE method (Part I—Theory) are reported in a first instance. The method is based on an inverse problem formulation combined with a recently proposed regularization approach: a beamforming matrix in the discrete smoothing norm of the cost function. In a second instance, the results obtained from the SFE method are applied to SFC as presented in Part I. The SFC classification method is verified in two environments that recreate ideal or complex sound fields. At the light of the presented results and discussion, it is argued that SFE and SFC proposed methods are effective.
Convention Paper 8527

5:30 pm

P15-4 Mixed-Order Ambisonics Recording and Playback for Improving Horizontal Directionality—*Sylvain Favrot*,[1] *Marton Marschall*,[1] *Johannes Käsbach*,[1] *Jörg Buchholz*,[2] *Tobias Weller*[1]
[1]Technical University of Denmark, Lyngby, Denmark
[2]Macquarie University, Sydney, NSW, Australia

Planar (2-D) and periphonic (3-D) higher-order Ambisonics (HOA) systems are widely used to reproduce spatial properties of acoustic scenarios. Mixed-order Ambisonics (MOA) systems combine the benefit of higher order 2-D systems, i.e., a high spatial resolution over a larger usable frequency bandwidth, with a lower order 3-D system to reproduce elevated sound sources. In order to record MOA signals, the location and weighting of the microphones on a hard sphere were optimized to provide a robust MOA encoding. A detailed analysis of the encoding and decoding process showed that MOA can improve both the spatial resolution in the horizontal plane and the useable frequency bandwidth for playback as well as recording. Hence the described MOA scheme provides a promising method for improving the performance of current 3-D sound reproduction systems.
Convention Paper 8528

6:00 pm

P15-5 Local Sound Field Synthesis by Virtual Acoustic Scattering and Time-Reversal— *Sascha Spors, Karim Helwani, Jens Ahrens*, Deutsche Telekom Laboratories, Technische Universität Berlin, Berlin, Germany

Sound field synthesis techniques like Wave Field Synthesis and near-field compensated higher order Ambisonics aim at synthesizing a desired sound field within an extended area using an ensemble of individually driven loudspeakers. Local sound field synthesis techniques achieve an increased accuracy within a restricted local listening area at the cost of stronger artifacts outside. This paper proposes a novel approach to local sound field synthesis that is based upon the scattering from a virtual object bounding the local listening area and the time-reversal principle of acoustics. The physical foundations of the approach are introduced and discussed. Numerical simulations of synthesized sound fields are presented as well as a comparison to other published methods.
Convention Paper 8529

Broadcast/Media Streaming Session 7
Friday, October 21 4:00 pm – 5:30 pm
Room 1E10

AUDIO ENGINEERING SUPPORTING PEOPLE WITH DISABILITIES: A WORKSHOP

Chair: **Eric Small**, Modulation Sciences

Panelists: *Alison Greenwald Neplokh*, Chief Engineer, FCC Media Bureau
Steve Lyman, Dolby
Joel Snyder, Audio Description Project, American Council of the Blind

On October 8, 2010, President Obama signed the Twenty-First Century Communications and Video Accessibility Act (CVAA) into law. The CVAA updates federal communications law to increase the access of persons with disabilities to modern communications and entertainment technology, including new digital, broadband, and mobile innovations. Since much of this law applies to the visually and hearing handicapped, it will create many challenges and opportunities for the audio engineering community. Graphic user interfaces must be made audible while audible interfaces must be made visual- and hearing aid-friendly. This workshop will explore CVAA from the viewpoint of an engineer and regulator who helped write the law and a creator of the content that implements one aspect of the law.

Friday, Oct. 21 4:00 pm Room 1E05
Technical Committee Meeting on High Resolution Audio

Session P16 Friday, Oct. 21
4:30 pm – 6:30 pm Room 1E07

LOW BIT-RATE CODING—PART 2

Chair: **Christof Faller**, Illusonic, Switzerland

4:30 pm

P16-1 A Subband Analysis and Coding Method for Downmixing Based Multichannel Audio Codec—*Shi Dong, Ruimin Hu, Weiping Tu, Xiang Zheng*, Wuhan University, Wuhan, Hubei, China

In the present downmixing based multichannel coding standard, the downmixing process causes the “tone leakage” problem by mixing different channels into one channel. In this paper a novel multichannel analysis method is proposed to reduce “tone leakage” phenomenon with additive side information, the basic idea is to find subbands with the largest spectrum difference and coding their spectrum envelope information. By analyzing the decoded signals, leakage tones are identified and attenuated, then the original ones are reconstructed, which meantime retain the original interchannel level difference (ICLD) of subbands unchanged. Results show our method can improve subjective quality compared with HE-AAC (v2) codec with bit rate increasing slightly.
Convention Paper 8530
[Paper not presented but is available for purchase]

5:00 pm

P16-2 Characterizing the Perceptual Effects Introduced by Low Bit Rate Spatial Audio Codecs—*Paulo Marins*, Universidade de Brasília, Brasília, Brazil

This paper describes a series of experiments that was carried out aiming to characterize the perceptual effects introduced by low bit rate spatial audio codecs. An initial study was conducted with the intention of investigating the contribution of selected attributes to the basic audio quality of low bit rate spatial codecs. Furthermore, another two experiments were performed in order to identify the perceptually salient dimensions or the independent perceptual attributes related to the artifacts introduced by low bit rate spatial audio coding systems.
Convention Paper 8531

5:30 pm

P16-3 Error Robust Low Delay Audio Coding Based on Subband-ADPCM—*Stephan Preihs, Jörn Ostermann*, Institute for Information Processing, Leibniz Universität Hannover, Hannover, Germany

In this paper we present an approach for error robust audio coding at a medium data rate of about 176 kbps (mono, 44.1 kHz sampling rate). By combining a delay-free Adaptive Differential Pulse Code Modulation (ADPCM) coding-scheme and a numerically optimized low delay filter bank we achieve a very low algorithmic coding delay of only about 0.5 ms. The structure of the codec also allows for a high robustness against random single bit errors and even supports error resilience. Implementation structure, results of a listening test, and PEAQ (Perceptual Evaluation of Audio Quality) based objective audio quality evaluation as well as tests of random single bit error performance are given. The presented coding-scheme provides a very good audio quality for vocals and speech. For most of the critical signals the audio quality can still be denoted as acceptable. Tests of random single bit error performance show good results for error rates up to 10-4.
Convention Paper 8532

6:00 pm

P16-4 The Transient Steering Decorrelator Tool in the Upcoming MPEG Unified Speech and Audio Coding Standard—*Achim Kuntz, Sascha Disch, Tom Bäckström, Julien Robillard*, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

Applause signals are still challenging to code with good perceptual quality using parametric multichannel audio coding techniques. To improve the codec performance for these particular items, the Transient Steering Decorrelator (TSD) tool has been adopted into the upcoming Moving Picture Experts Group (MPEG) standard on Unified Speech and Audio Coding (USAC) as an amendment to the MPEG Surround 2-1-2 module (MPS). TSD improves the perceptual quality of signals which contain rather dense spatially distributed transient auditory events like in applause type of signals. Within TSD, tran- ➥

sient events are separated from the core decoder output and a dedicated decorrelator algorithm distributes the transients in the spatial image according to parametric guiding information transmitted in the bitstream. Listening tests show a substantial improvement in subjective quality.
Convention Paper 8533

Session P17 Friday, Oct. 21
4:30 pm – 6:00 pm 1E Foyer

POSTERS: AUDIO EQUIPMENT AND MEASUREMENT

2:30 pm

P17-1 Swept Sine Grains Applied to Acoustical Measurements Using Perceptual Masking Effects—*Joel Preto Paulo*,[1,2] *J. L. Bento Coelho*[2]
[1]ISEL- Instituto Superior de Engenharia de Lisboa, Lisbon, Portugal
[2]IST-CAPS—Instituto Superior Tecnico, TU Lisbon, Lisbon, Portugal

The swept sine technique has proven to lead to accurate estimations of the room impulse response, even in situations of low SNR, non-linearity, and time-variance of the system under test. Regarding the distributive property of the convolution, the swept sine signal can be split into several segments, namely grains, and sent separately to the room. At the reception, the full frame is assembled from the grains by applying the overlap-add based procedure. Choosing appropriate windows, values to the truncation and the overlap of the captured grains, the degree of degradation on the final results, measured by the amount of produced noise, can be controlled. Therefore, each grain can be matched according to the tempered musical scale, which can be applied on musical compositions, or used in perceptual models, taking into account the human auditory masking effects, to compose a test signal frame for acoustical measurements. The possibility of polyphony musical compositions for use on acoustical measurements is assessed and discussed.
Convention Paper 8534

2:30 pm

P17-2 Multichannel Impulse Response Measurement in Matlab—*Braxton Boren, Agnieszka Roginska*, New York University, New York, NY, USA

This paper describes ScanIR, an application for flexible multichannel impulse response measurement in Matlab intended for public distribution. The application interfaces with the PortAudio API using Psychtoolbox-3, a toolkit in Matlab allowing high-precision control of a multichannel audio interface. ScanIR contains single-channel, binaural, and multichannel input modes, and it also allows the use of multiple output test signals. It is hoped that this application will prove useful to researchers using Matlab for physical or psychological acoustic measurements.
Convention Paper 8535

2:30 pm

P17-3 Why Do Tube Amplifiers Have Fat Sound while Solid State Amplifiers Don't?—
Shengchao Li, Wintersweet Electronics, LLC, Potomac, MD, USA

I propose an explanation to why tube amplifiers sound better than solid state amplifiers in certain circumstances. The explanation is, the interaction of (1) the nonlinearity of the output tube, (2) output impedance of the amplifier, and (3) the nonlinearity of the output transformer inductance caused by core material B-H curve, results in a frequency selective nonlinear feedback system that softly limits the speaker cone excursion for low frequency music signals with excessive amplitude, while has little effect on high frequency music signals or low frequency music signals with low to moderate amplitude. Better yet, when low frequency music signals with excessive amplitude is superposed with high frequency music signals, this system selectively limits low frequency music signals and has little effect on the superposed high frequency music signals. Comparing to that of a typical solid state amplifier, this mechanism trades some amplifier nonlinearity for less speaker nonlinearity, resulting in less overall nonlinearity of the music sound waves people's ears perceive.
Convention Paper 8536

Workshop 9 Friday, Oct. 21
4:30 pm – 6:30 pm Room 1E13

NEODYMIUM; COPING WITH THE CONSEQUENCES OF SUPPLY AND DEMAND ELASTICITY

Chair: **Spiro Iraclianos**

Panelists: *John Ebert*, Yungsheng
Alexander King, Ames Laboratory
Stan Trout, Molycorp.

The use of Neodymium Iron Boron magnets has been increasing over the last few decades to address the demands of compact high energy motor applications. Loudspeakers have been adapting the "super magnets" to delicately balance the size/performance ratio demands in today's marketplace. In recent months, the availability of Neodymium has been shrinking while the cost of the magnets, which use the rare earth element, has skyrocketed. The workshop will serve as a forum for industry professionals, from all aspects of the value chain, to gather and collectively discuss and begin to address the facts and myths surrounding the global issue that has caused significant paradigm shifts in the way we think about the future of the industry.

Live Sound Seminar 7 Friday, Oct. 21
4:30 pm – 6:30 pm Room 1E14

SUBWOOFER ARRAYS AND BEAM STEERING

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

Panelists: *Paul Bauman*, JBL
Doug Fowler, Fowler Audio Services LLC
Charlie Hughes, Excelsior Audio
Martyn Rowe, Martin Audio

Nothing seems more topical than the topic of control of low frequency energy via array design. The problem of excess low frequency energy will be addressed with subwoofer and array design, placement, and DSP control. The methods of achieving good coverage for the audience and eliminating bass build up on stage or other areas will be discussed.

Product Design Session 5 **Friday, Oct. 21**
4:30 pm – 6:00 pm **Room 1E12**

ALTERNATIVE MEASUREMENT TECHNIQUES

Presenter: **Tom Kite**, Audio Precision

Modern audio analyzers offer many approaches to taking measurements. Some approaches provide great speed and others high accuracy. Which is best for your circuit? Are steady tones appropriate for codecs? Are chirps appropriate for compressors? This session will compare and contrast alternative approaches such as multitones, chirps, and bursts to traditional techniques. Learn the benefits and pitfalls of various approaches. Get the answers and the understanding behind them.

Special Event
PLATINUM MASTERING
Friday, October 21, 4:30 pm – 6:30 pm
Room 1E15/16

Moderator: **Bob Ludwig**, Gateway Mastering Studios, Portland, ME, USA

Panelists: *Adam Ayan*, Gateway Mastering Studios, Portland, ME, USA
Eric Boulanger, The Mastering Lab
Chris Gehringer, Sterling Sound, New York, NY, USA
Scott Hull, Masterdisk Studios, New York, NY, USA
Barak Moffitt, EVP EMI Label Group (Capitol Studios), Hollywood, CA, USA
Darcy Proper, Wisseloord Studios, Hilversum, The Netherlands

You Have Questions, We Have Answers

The art of music mastering continues to evolve. Mastering has always been one of the more secretive parts of the record making chain, there are still only a few good books written about it and most good mastering engineers learn by apprenticeship which is difficult to find. We have assembled a platinum-quality group of mastering engineers and a record label executive, who will discuss the latest trends and techniques in mastering, discuss the state of the record industry and most importantly take a lot of questions from the audience.

Student/Career Development Event
RECORDING COMPETITION—1
Friday, October 21, 4:30 pm – 6:30 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. This event presents stereo and surround recordings in these categories:

- Sound for Visual Media – 4:30 pm to 5:30 pm
  Judges: Richard King, Stephen Harwood, Geoff Mitchell
- Traditional Acoustic Recording – 5:30 pm to 6:30 pm
  Judges: David Bowles, Martha de Francisco, Ulrike Schwarz

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

Tutorial 9 **Friday, Oct. 21**
5:00 pm – 6:30 pm **Room 1E08**

NOISE ON THE BRAIN—
HEARING DAMAGE ON THE OTHER SIDE

Presenter: **Poppy Crum**, Dolby Laboratories, San Francisco, CA, USA

Did you know that drinking a glass of orange juice every day may actually protect your hearing?

Most discussions of hearing damage focus on what happens to the cochlea and inner ear. While this understanding is crucial to predicting and avoiding trauma that can lead to hearing loss, both acoustic and chemical stimuli can also have significant effects on higher brain areas. In some cases, thresholds and audiograms can look completely normal but listeners may have great difficulty hearing a conversation in a noisy environment. This session will explore the latest research regarding the effects of acoustic and chemical trauma, and how this damage manifests throughout the auditory pathway as changes in hearing sensitivity, cognition, and the experience of tinnitus. We will also consider recent research in chemically preserving hearing and combating these conditions with supplements as common as Vitamin C!

Exhibitor Seminar **Friday, Oct. 21**
5:00 pm – 6:00 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

Intelligent Dance Music Live Performance in 5.1

Presenter: **David Miles Huber**

DMH (2 x Grammy-nominated producer and musician) will perform his latest dance-oriented project "Chamberland" live in surround. Come prepared to have fun!

Friday, Oct. 21 **5:00 pm** **Room 1E05**
Technical Committee Meeting on Signal Processing

Tutorial 10 **Friday, Oct. 21**
5:30 pm – 6:30 pm **Room 1E10**

LIBRARY OF CONGRESS COLLECTIONS
& NATIONAL JUKEBOX

Presenter: **Brad McCoy**, Library of Congress, Culpeper, VA, USA

This tutorial will highlight several of the Library of Con- ➥

gress collections, especially ones with New York interest (Tony Schwartz, some very early Frank Sinatra, others) to use as examples of audio preservation, workflow, and metadata issues at the Library of Congress. The tutorial will also include a quick look at the National Jukebox.

Special Event
OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE
Friday, October 21, 7:00 pm – 9:00 pm
Room 1E15/16

Lecturer: **John Atkinson**

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

The Richard C. Heyser distinguished lecturer for the 129th AES Convention is John Atkinson. Atkinson's formal education was in the sciences—he graduated from the University of London in 1972 with an honors degree in physics and chemistry, and from the University of London's School of Education in 1974 with a postgraduate qualification in the teaching of high-school science—but his passion was always for music. A musician (primarily on bass guitar, but also on recorder, clarinet, violin, and viola da gamba), a sound recordist, and an audiophile, Atkinson pursued all three areas simultaneously in the 1960s and '70s, before finally settling down in magazine publishing in 1976, when he joined the UK's Hi-Fi News & Record Review as an editorial assistant. He became HFN/RR's editor in 1982, and had almost doubled its circulation by 1986, when he moved to the US to become editor of Stereophile, the position he still occupies. Atkinson expanded Stereophile's publishing schedule to monthly, and by 1998, when he and his business partner sold the magazine to Petersen Publications, had tripled its circulation. Starting in 1989, Atkinson introduced a program of measuring the components reviewed in Stereophile, and since then has measured 750 loudspeakers, 500 amplifiers of all kinds, and almost 300 digital products, all under standardized conditions. He has also continued his recording activities, and to date has produced, engineered, edited, mastered, and/or performed on more than 40 commercially released recordings. Atkinson has been a member of the AES since 1981; a committed generalist, he is very likely the only audio magazine editor who has also panned for gold and made his own transistors. The title of his lecture is, "Where Did the Negative Frequencies Go?"

Even a cursory reading of the academic literature suggests that all that matters in audio has already been investigated and ranked in importance. But John Atkinson's 40-year career of performing music, engineering, and producing recordings, reviewing audio components, and editing audio magazines has led him to believe that some things might be taken too much for granted, while other things have been ignored or misunderstood. The title of his lecture is a metaphor: All real numbers have two roots, yet we routinely discard the negative root on the grounds that it has no significance in reality. When it comes to understanding the perception of music, perhaps some of the things we discard as audio engineers merit further examination. This lecture will cover both audio recording and playback technologies; while it might not offer definitive answers, perhaps it will raise some interesting questions.

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

Broadcast/Media Streaming Session 8
Friday, October 21 7:30 pm – 9:00 pm
PC Richards Theater

A HALF CENTURY OF FM STEREO

Co-chairs: **David Bialik**
Scott Fybush, Fybush.com/Inside Radio

Panelists: *Richard Burden*
Frank Foti, Omnia Audio
Richard Mertz, Cavell and Mertz
Arno Meyer, Belar
Robert Orban, Orban
Skip Pizzi, NAB
Bill Sacks, Orban
Eric Small, Modulation Sciences
Jeff Smith, Clear Channel
Herb Squire, Former Chief Engineer of WQXR

Radio's first four decades were strictly monophonic. Despite some early experimentation by FM inventor Edwin Howard Armstrong, even "high-fidelity" FM remained mono until 1961, when the FCC chose among competing systems to declare a standard for FM stereo broadcasting.

To celebrate the fiftieth anniversary of the first commercial FM stereo broadcasts in the United States, this panel discussion will bring together some of the industry's top broadcast engineers and equipment designers. Topics will include FM stereo's birth, the merits of the competing systems that sought FCC approval, the initial technical challenges faced by the new medium, the development of high-density FM audio processing, and the future of analog FM stereo in an increasingly digital world.

This event is being organized by the NY Section of the AES and will be held in the PC Richard & Sons Theater at the Clear Channel Studios, 32 Avenue of the Americas. Seating is limited and tickets are required.

The Audio Engineering Society thanks Clear Channel Communications for providing the theater for this event.

Session P18 Saturday, Oct. 22
9:00 am – 12:00 noon Room 1E09

HEADPHONE PLAYBACK

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

9:00 am

P18-1 The Effects of Headphones on Listener HRTF Preference—Braxton Boren, Agnieszka Roginska, New York University, New York, NY, USA

Listener-selected HRTFs have the potential to provide the accuracy of an individualized HRTF without the time and resources required for HRTF measurements. This study tests listeners' HRTF preference for three different sets of headphones. HRTF datasets heard over the noise-cancelling

Bose Aviation headset were selected as having good externalization more often than those heard over Sennheiser HD650 open headphones or Sony MDR-7506 closed headphones. It is thought that the Bose headset's frequency response is responsible for its superior externalization. This suggests that in systems where high quality headphones are not available, post-processing equalization should be applied to account for the effect of the headphones on HRTF reproduction.
Convention Paper 8537

9:30 am

P18-2 Head Orientation Tracking Using Binaural Headset Microphones—*Hannes Gamper, Sakari Tervo, Tapio Lokki*, Aalto University School of Science, Aalto, Finland

A head orientation tracking system using binaural headset microphones is proposed. Unlike previous approaches, the proposed method does not require anchor sources, but relies on speech signals from the wearers of the binaural headsets. From the binaural microphone signals, time difference of arrival (TDOA) estimates are obtained. The tracking is performed using a particle filter integrated with a maximum likelihood estimation function. In a case study the proposed method is used to track the head orientations of three conferees in a meeting scenario. With an accuracy of about 10 degrees the proposed method is shown to outperform a reference method, which achieves an accuracy of about 35 degrees.
Convention Paper 8538

10:00 am

P18-3 Observing the Clustering Tendencies of Head Related Transfer Function Databases—*Areti Andreopoulou, Agnieszka Roginska, Juan Bello*, New York University, New York, NY, USA

This study offers a detailed description of the clustering tendencies of a large, standardized HRTF repository, and compares the quality of the results to those of a CIPIC database subset. The statistical analysis was implemented by applying k-means clustering on the log magnitude of HRTFs on the horizontal plane, for a varying number of clusters. A thorough report on the grouping behavior of the filters as the number of clusters increases revealed a superiority of the HRTF repository in describing common behaviors across equivalent azimuth positions, over the CIPIC subset, for the majority of the HRTF datasets.
Convention Paper 8539

10:30 am

P18-4 Individual Perception of Headphone Reproduction Asymmetry—*Juha Merimaa,*[1] *V. Ralph Algazi,*[2] *Richard O. Duda*[3]
[1]Sennheiser Research Laboratory, Palo Alto, CA, USA
[2]University of California Davis, Davis, CA, USA
[3]Menlo Park, CA, USA

With headphone listening, the naturally occurring left/right asymmetry in head and ear shapes can produce frequency-dependent variations in the perceived location of a sound source. In this paper the perception of such asymmetry is studied by determining the interaural level differences required to center a set of narrow-band stimuli with different center frequencies. It is shown that the asymmetry varies from one listener to another. Some of the asymmetry can be explained with asymmetry in transmission of sound from the headphones to the entrances of a listener's ear canals. However, the perceived asymmetry for individual listeners is also correlated between different headphone types including in-ear headphones that couple directly to the ear canals. The asymmetry is relatively stable over different times of wearing the headphones. The effect of correcting for the asymmetry ranges from imperceptible to substantial depending on the individual subject.
Convention Paper 8540

11:00 am

P18-5 Binaural Reproduction of Stereo Signals Using Upmixing and Diffuse Rendering—
Christof Faller,[1] *Jeroen Breebaart*[2]
[1]Illusonic LLC, St-Sulpice, Swtizerland
[2]ToneBoosters, Eindhoven, The Netherlands

In this paper benefits and challenges related to binaural rendering for conventional stereo content are explained in terms of width of the sound stage, timbral changes, the perceived distance, and the naturalness of phantom sources. To resolve some of the identified issues, a two-stage process consisting of a spatial decomposition followed by dedicated post processing methods is proposed. In the first stage, several direct sound source signals and additional ambience components are extracted from the stereo content. These signals are subsequently processed with dedicated algorithms to render virtual sound sources by means of HRTF or BRIR convolution and to render an additional diffuse sound field with the correct inter-aural coherence properties based on the extracted ambience signals. It is argued that this approach results in a wider sound stage, more natural and accurate spatial imaging of sound sources, and resolves the "here and now" versus the "there and then" duality for room acoustic simulation in binaural rendering methods.
Convention Paper 8541

11:30 am

P18-6 Sound Quality Evaluation Based on Attributes —Application to Binaural Contents—*Sarah Le Bagousse,*[1] *Mathieu Paquier,*[2] *Catherine Colomes,*[1] *Samuel Moulin*[1]
[1]Orange Labs, Cesson Sévigné, France
[2]Laboratoire d'Informatique des Systemes Compexes, Brest, France

The audio quality assessment is based on standards that mainly evaluate the overall quality to the detriment of more accurate sound criteria. On the other hand, a major problem of an assessment based on sound criteria is their meaning and their understanding that have to be the same for each listener. A previous study clustered a list of sound attributes in three main categories called "timbre," "space," "defaults." ➥

The work presented here is based on those previous results and aims at tuning a subjective test methodology of spatial audio quality. So the three families were included in a test dedicated to the assessment of spatial audio quality with binaural contents. The test was based on the MUSHRA method but using three anchors specifically to each attribute and without explicit reference. The original version was added as the hidden reference. The aim of the listening test described in this paper was to verify the relevance of those three attributes and their influence on the overall quality.
Convention Paper 8542

Workshop 10 Saturday, Oct. 22
9:00 am – 11:00 am Room 1E08

WHAT EVERY SOUND ENGINEER SHOULD KNOW ABOUT THE VOICE

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

Panelists: *Henrik Kjelin*, Complete Vocal Institute, Denmark
Cathrine Sadolin, Complete Vocal Institute, Denmark

The purpose of this workshop is to teach sound engineers how to listen to the voice before they even think of microphone picking and knob-turning. The presentation and demonstrations are based on the "Complete Vocal Technique" (CVT) where the fundamental is the classification of all human voice sounds into one of four vocal modes named Neutral, Curbing, Overdrive, and Edge. The classification is used by professional singers within all musical styles, and has in a period of 20 years proved easy to grasp in both real life situations and also in auditive and visual tests (sound examples and laryngeal images/laryngograph waveforms). These vocal modes are found in the speaking voice as well. Cathrine Sadolin, the developer of CVT, will involve the audience in this workshop, while explaining and demonstrating how to work with the modes in practice to achieve any sound and solve many different voice problems like unintentional vocal breaks, too much or too little volume, hoarseness, and much more. The physical aspects of the voice will be explained and laryngograph waveforms and analyses will be demonstrated. Eddy Brixen will explain essential parameters in the recording chain, especially the microphone, to ensure reliable and natural recordings.

Workshop 11 Saturday, Oct. 22
9:00 am – 10:30 am Room 1E10

MAX FOR LIVE: A DISCUSSION OF COMMUNITY-DRIVEN SOFTWARE DEVELOPMENT AND ITS RELEVANCE IN TODAY'S CREATIVE LANDSCAPE

Chair: **Stefan Brunner**, Max for Live Product Manager, Ableton AG

Panelists: *Michael Carter*, Preshish Moments
Bruce Odland, Sound Artist/Composer/Wooster Group
Rob Sussman, Cycling '74

Two years after the release of Max for Live by Ableton and Cycling '74, an ever-growing community of developers are deciding to leverage this platform, paving the way for new software and hardware interoperability inside and outside of Ableton Live. A panel of creative professionals, ranging from artists to producers, will discuss the creative potential of the community-driven Max for Live platform in developing hardware and software solutions to enable projects large and small. The panel will present an in-depth look at a series of Max for Live devices that reflect the Max for Live's diverse spectrum of uses, ranging from new creative avenues of sound manipulation to enabling easy cross-media performance applications.

Tutorial 11 Saturday, Oct. 22
9:00 am – 12:00 noon Room 1E07

TECHNOLOGY AND PRACTICE OF DISK MASTERING FOR VINYL PRODUCTS

Presenters: **Paul Gold**
Albert Grundy

This tutorial on the technology and practice of disk mastering will cover topics including: technology of the 33 1/3 12-inch LP record; the blank disk and cutting stylus; available space and lines per inch; mechanics of the groove: lateral, vertical and orthogonal motions; the dynamic cutter head stylus amplitude, velocity and acceleration; evolution of the recording characteristic; time constants and corner frequencies; limitations imposed by 10 octave bandwidth. The second part of the tutorial will look at applying this information to cutting parameters after auditioning the program.

Broadcast/Media Streaming Session 9
Saturday, October 22 9:00 am – 10:30 am
Room 1E10

SOUND EFFECTS OF THE *WITCHES OF LUBLIN*

Presenters: *Butch D'Ambrosio*, SFX Artist
Sylvaana Pinto, SFX Artist
David Shinn, Master Engineer
Mark Wiener, SFX Artist
Sue Zizza, Director/Producer

This session will be presented in 3 parts:

(a) Recording live effects—how we used our home and recorded late at night—see photos on the website in NEWS.

(b) Recording a large cast all together and still maintaining isolation for intercuts and post editing

(c) Recording live music—not as a studio production—all isolated—but as is.

Live Sound Seminar 8 Saturday, Oct. 22
9:00 am – 11:00 am Room 1E14

THINGS I LEARNED ON THE BROADWAY LADDER: LESSONS FOR EVERY AUDIO CAREER

Presenter: **Jim Van Bergen**

Panelists: *Megan Henniger*
Abe Jacob
Jessica Paz
Joshua Reid

Learn what the pros learned the hard way. What they

don't teach you in school and what applies to theatrical audio at all levels: high school, college, community, regional, and commercially. Industry secrets, practices to keep, and ways of working that will keep you working. A session to help educate and guide the next generation.

Product Design Session 6 **Saturday, Oct. 22**
9:00 am – 10:30 am **Room 1E12**

DESIGN OF A DYNAMIC RANGE COMPRESSOR

Presenter: **Josh Reiss**

Dynamic range compression, despite being one of the most widely used audio effects, is still poorly understood, and there is little formal knowledge and analysis of compressor design techniques. In this tutorial we describe several different approaches to digital dynamic range compressor design. Digital implementations of several classic analog approaches are given, as well as designs from recent literature and new approaches that address possible issues. Several design techniques are analyzed and compared, including RMS and peak-based approaches, feedforward and feedback designs, and linear and log domain level detection. We explain what makes the designs sound different, and provide distortion-based metrics to analyze their quality. Finally, we provide recommendations for high performance compressor design.

Special Event
GRATEFUL DEAD EUROPE 72
Saturday, October 22, 9:00 am – 10:45 am
Room 1E15/16

Moderator: **Sam Berkow**

Panelists: *David Glasser*, Mastering Engineer
Jamie Howarth, Tape Transfers and Plangent Processes Speed Correction
Gary Lambert
Jeffrey Norman, Mixer

In the spring of 1972, the Grateful Dead toured Europe with a new (and modified) Ampex MM1100 tape recorder in tow. The tour, and subsequent LP release, find the Dead at one of their creative peaks. In September 2011 Rhino Records will release a massive CD box set containing every note recorded on this 22-show tour—over 70 hours of music on 73 discs, surely one of the most ambitious rock and roll box sets to date. This Event will bring together the production and engineering team who have been working on restoring, mixing, and mastering this historic music. Musical excerpts from the project will be presented, and the project workflow and creative challenges will be discussed.

Saturday, Oct. 22 **9:00 am** **Room 1E02**
Standards Committee Meeting on Audio Applications of Networks, SC-02-12

Session P19 **Saturday, Oct. 22**
9:30 am – 11:00 am **1E Foyer**

POSTERS: RECORDING AND REPRODUCTION

9:30 am

P19-1 A Revised Approach to Teaching Audio Mixing Techniques: Applying the Deliberate Practice Model—*John Holt Merchant, III*, Middle Tennessee State University, Murfreesboro, TN, USA

An overview of the Mixing Techniques course currently offered at Middle Tennessee State University, which was designed to help students develop substantive foundational knowledge and technological competencies regarding the aesthetic and technological aspects of audio mixing techniques by applying the tenets of the Deliberate Practice model. Relevant studies in human performance, characteristics of Millennial students, and pedagogy for developing mental models of audio engineering systems are considered as they apply to recording arts course and curricular design. The results of this study suggest that implementing rigorous, formal practice of foundational skills in audio mixing courses significantly improves students' capabilities.
Convention Paper 8543

9:30 am

P19-2 Sound Field Recording and Reproduction Using Transform Filter Designed in Spatio-Temporal Frequency Domain—*Shoichi Koyama, Ken'ichi Furuya, Yusuke Hiwasaki, Yoichi Haneda*, NTT Cyber Space Laboratories, NTT Corporation, Tokyo, Japan

A method of transform from the received signals of a microphone array to the driving signals of a loudspeaker array for sound field reproduction is investigated. Our objective is to obtain the driving signal of a planar or linear loudspeaker array only from the sound pressure distribution acquired by a planar or linear microphone array. We derive a formulation of the transform from the received signals of the microphone array to the driving signals of the loudspeaker array. The transform is achieved as the mean of a filter in a spatio-temporal frequency domain. Results of measurement experiments in an anechoic room are presented to compare the proposed method with the method based on the conventional least mean square algorithm. The reproduction accuracies were found to be almost the same, but the filter size and amount of calculation required for the proposed method were much smaller than those for the least mean square algorithm based one.
Convention Paper 8544

9:30 am

P19-3 Practical Digital Playback of Gramophone Records Using Flat-Bed Scanner Images—
Baozhong Tian,[1] *Samuel Sambasivam*,[2] *John L. Barron*[3]
[1]West Virginia University Institute of Technology, Montgomery, WV, USA
[2]Azusa Pacific University, Azusa, CA, USA
[3]The University of Western Ontario, London, Ontario, Canada

We present an Optical Audio Reconstruction (OAR) system to play the audio from gramophone records based on image processing. OAR uses an off-the-shelf scanner to achieve an affordable and practical method of reconstructing audio. Converting the analog records to a digital format is important for preserving many historical recordings. We scan the records using ➡

a high resolution large format flat-bed scanner. The images were then segmented and the grooves were tracked to simulate the movement of the stylus. The sound signal was converted from the groove track positions. Our OAR algorithm was able to reconstruct the audio successfully from scanned records. We demonstrated that a fast OAR system producing good quality sound can be built economically.
Convention Paper 8545

9:30 am

P19-4 CAIRA—A Creative Artificially-Intuitive and Reasoning Agent as Conductor of Telematic Music Improvisations—*Jonas Braasch, Doug Van Nort, Selmer Bringsjord, Pauline Oliveros, Anthony Parks, Colin Kuebler*, Rensselear Polytechnic Institute, Troy, NY, USA

This paper reports on the architecture and performance of the Creative Artificially-Intuitive and Reasoning Agent CAIRA as a conductor for improvised avant-garde music. CAIRA listening skills are based on a music recognition system that simulates the human auditory periphery to perform an Auditory Scene Analysis (ASA). Its simulation of cognitive processes includes a comprehensive cognitive calculus for reasoning and decision-making using logic-based reasoning. In a specific application, CAIRA is used as conductor for live music performances with distributed ensembles, where the musicians are connected via the internet. CAIRA uses a visual score and directs the ensemble members based on tension arc estimations for the individual music performers.
Convention Paper 8546

9:30 am

P19-5 Discrimination between Phonograph Playback Systems—*Jason A. Hockman, David M. Weigl, Catherine Guastavino, Ichiro Fujinaga*, McGill University, Montreal, Quebec, Canada

Digitization of phonograph records is an important step toward the preservation of our cultural history and heritage. The phonograph playback systems (PPS) required for this digitization process are comprised of several components in a variety of price ranges. We report on the results of two listening tests intended to ascertain the extent to which expert listeners can discriminate between PPS of different price ranges. These results are intended to determine the extent to which component selection affects the discrimination between PPS and to provide a set of guidelines for the purchase of PPS components for the digitization of phonograph record collections.
Convention Paper 8547

9:30 am

P19-6 Vibration Analysis of Edge and Middle Exciters in Multiactuator Panels—*Basilio Pueo,*[1] *José Vicente Rico,*[1] *José Javier López*[2]
[1]University of Alicante, Alicante, Spain
[2]Technical University of Valencia, Valencia, Spain

Multiactuator panels (MAPs) are an extension of the distribute mode technology in order to be employed for Wave Field Synthesis (WFS) applications. In this paper a special type of semi-clamped boundary condition that exhibited proper results in recent convention papers by the authors, is used for setting a MAP ready for use. For that purpose, the surface velocity over specific points has been measured with a Laser Doppler Vibrometer, paying special attention to representative exciter locations: central area, edge, and corners. For an understanding of the sound-generating behavior of the panel, measures were taken both for exciters accommodated in a roughly centered line, as in the rest of MAP prototypes and for an identical array of exciters positioned at the upper side of the panel, in which the behavior was still acceptable for WFS purposes. Experiments were also conducted to analyze the role that the exciter coupling ring, used to physically attach transducer to panel, has on the vibrational behavior and radiated sound.
Convention Paper 8548

9:30 am

P19-7 Challenges in 2.4 GHz Wireless Audio Streaming—*Robin Hoel, Tomas Motos*, Texas Instruments Inc., Oslo, Norway

Based on the experiences accumulated during the development of a family of 2.4 GHz wireless audio streaming ICs, the paper presents challenges in providing high-quality, uninterrupted audio streaming in the 2.4 GHz ISM band. It discusses the impairments to expect in a dense indoor radio environment, gives an overview of the main interfering radio standards in the 2.4 GHz band that any such system must coexist with, and outlines methods and techniques that are essential in order to overcome these difficulties. To exemplify, some of the critical design choices that were made for the first device in the family are presented along with thoughts on possible future improvements.
Convention Paper 8549

9:30 am

P19-8 Ray-Traced Graphical User Interfaces for Audio Effect Plug-ins—*Benjamin Doursout,*[1] *Jörn Loviscach*[2]
[1]ESIEA—École supérieure d'informatique, électronique, automatique, Laval, France
[2]Fachhochschule Bielefeld (University of Applied Sciences), Bielefeld, Germany

On the computer, effects and software-based music synthesizers are often represented using graphical interfaces that mimic analog equipment almost photorealistically. These representations are, however, limited to a fixed perspective and do not include more advanced visual effects such as polished chrome. Leveraging the flexibility of the audio plug-in programming interface, we have created software that equips a broad class of synthesis and effect plug-ins with interactive, ray-traced 3-D replicas of their user interface. These 3-D models are built automatically through an automated analysis of each plug-ins' standard 2-D interface. Our experiments show that interactive frame rates can be achieved even with low-end graphics cards. The methods presented may also

be used for an automatic analysis of settings and for realistic interactive simulations in the design phase of hardware controls.
Convention Paper 8550

Special Event
SPARS: LESSONS FROM SAVVY OWNERS
Saturday, October 22, 9:30 am – 10:30 am
Room 1E13

Moderator: **Kirk Imamura**

Panelists: *Stephen Joseph Antonelli*, PopMark Media
Kevin Hill, PopMark Media
Lisa Horan, PopMark Media
Chris Mara, Welcome to 1979
Dan Workman, SugarHill Studios

Hosted by SPARS, examples of recording studios that are expanding their business beyond traditional recording will be presented in a panel discussion format. There will be panelists representing three recording businesses, each with a unique business angle/strategy. The panel discussion is intended for audio engineers who are currently running a recording business looking for ideas to expand their business, people who are considering starting a recording business, or those who are interested in what new services recording businesses are offering now.

Workshop 12 **Saturday, Oct. 22**
10:00 am – 1:00 pm
NYU Steinhardt
35 West 4th St., NY
James L. Dolan Recording Studio

CREATING, ENHANCING, AND REPRODUCING SURROUND WITH HEIGHT INFORMATION

Presenters: **Tom Ammermann**
Paul Geluso
Brett Leonard
Lasse Nipkow
Wilfried Van Baelen
Gregor Zielinsky

Listening demos that support Workshop 2 "Capturing Height In Surround" will be presented at the NYU Steinhardt James L. Dolan Recording Studios. Classical, jazz, electronic, and rock recordings that incorporate height information for a true 3-D listening experience will be presented. New technologies including Auro 3D, Illusonic's 3-D room signal generator, Space Builder, and new height capturing microphone techniques developed at NYU will be demonstrated.

Several recordings originally produced in Auro 3D will be presented. All microphone set ups will be shown and discussed including photo and video documentation of the recording sessions. Spatial impression through decorrelated room signals and additional room signals generated by Illusonic's 3-D room signal generator will be demonstrated as well.

Space Builder is the culmination of a research collaboration between McGill University and NHK Broadcasting for the design and implementation of a flexible, realistic ambiance designer for the 22.2 audio standard that accompanies NKH's Ultra High Definition 4320p broadcast standard. Works mixed by Richard King and George Massenburg using Space Builder will be presented in NYU's multichannel sound research lab.

Student/Career Development Event
STUDENT RECORDING CRITIQUES
Saturday, October 22, 10:00 am – 11:00 am
Room 1E06

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

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

Saturday, Oct. 22 **10:00 am** **Room 1E03**
Regions and Sections Forum

Saturday, Oct. 22 **10:00 am** **Room 1E05**
Technical Committee Meeting on Electro Magnetic Compatibility

Game Audio Session 8 **Saturday, Oct. 22**
10:45 am – 11:45 am **Room 1E13**

HISTORY OF AUDIO IMPLEMENTATION TOOLSETS

Presenter: **Damian Kastbauer**, Bay Area Sound

Through the use of audio/visual examples and a pictorial overview this session will chronicle the history of publicly available and proprietary audio middleware toolsets covering: features, functionality, trends, and techniques in an attempt to understand the current state of audio implementation tools being used in today's game development environments. An understanding of the fundamentals that have developed over the past 15 years of audio toolset creation will be explored; in addition to the benefits of middleware and the continued need for specialized proprietary tools.

Workshop 13 **Saturday, Oct. 22**
11:00 am – 1:00 pm **Room 1E08**

AUTHENTICATION OF FORENSIC AUDIO —THEN AND NOW

Chair: **Jeff Smith**, National Center for Media Forensics

Panelists: *Eddy B. Brixen*, EBB Consult
Jonathan Broyles, Image and Sound Forensics
Catalin Grigoras, National Center for Media Forensics

The authentication of recorded events is a common request of forensic audio examiners during the course of formal litigation. With the transition of recorded media moving away from analog cassette tapes, forensic scientists working in this field have had to work creatively to address new challenges. This workshop will not only present a historical overview of forensic tape authentication as it has been employed since the Watergate scandal, new and emerging techniques and methods will be pre-

sented for addressing challenges in the digital domain. Attendees of this workshop will expect to learn a general overview of audio authentication, the nowadays challenges facing forensic audio specialists, and the scientific developments in this field.

Workshop 14 **Saturday, Oct. 22**
11:00 am – 1:00 pm **Room 1E11**

LOUDNESS WARS: THE TIDES HAVE CHANGED

Chair: **Thomas Lund**, TC Electronic, Denmark

Panelists: *John Atkinson*, Stereophile Magazine, New York, NY, USA
Bob Katz, Digital Domain, Orlando, FL, USA
Bob Ludwig, Gateway Mastering Studios, Portland, ME, USA
Susan Rogers, Berklee College of Music, Boston, MA, USA

2011 will go down in history as the year when ITU-R BS.1770-2 was introduced, and the sample peak measurement got retired. The old measurement is responsible for the ruining of 15 years of music heritage at the source, while the new one takes away the loudness advantage squashed productions had over dynamic content. Picking up where the panel left last year, you can expect an updated tour de force in listening examples, theory, and discussions. Find out how the changing tide will influence you when heard from a perceptual, a mixing, a mastering, and a consumer perspective. From Bach to Beyoncé, Monty to Matrix, headphone to HD playback.

Broadcast/Media Streaming Session 10
Saturday, October 22 **11:00 am – 12:30 pm**
Room 1E10

BACKHAUL IN THE TWENTY-FIRST CENTURY

Chair: **Frank Bolognino**, Modular Monolith Systems

Panelists: *Bruce Berensen*, Sirius/XM
Kirk Harnack, Telos Alliance
Chris Tobin, Musicam USA

Backhaul, or "the Clean Feed," is transmitting point-to-point to an individual television / radio station or broadcast network. In the IP world of the 21st Century, new digital codecs and cellular technology have changed the way broadcasters get their feeds back to Master Control.

Some of the items we will discuss include new hardware and software technologies; reliability; cost-effectiveness; compatibility; and latency, bandwidth, and network restriction issues.

The panel will also tell of some of their experiences in planning remotes, interfacing with I.T., and some of the live events they have done using these new technologies.

Live Sound Seminar 9 **Saturday, Oct. 22**
11:00 am – 1:00 pm **Room 1E14**

PRODUCTION WIRELESS SYSTEMS PRACTICAL APPLICATIONS AND PRACTICES

Moderator: **James Stoffo**, Independent Entertainment Production RF Coordinator

Panelists: *Joe Ciaudelli*, Sennheiser USA
Mark Gubser, GSC
Tim Vear, Shure, Inc.
Karl Winkler, Lectrosonics

From houses of worship and wedding bands to regional theaters and corporate audio, 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.

Product Design Session 7 **Saturday, Oct. 21**
11:00 am – 12:30 pm **Room 1E12**

BUILDING ANALOG IN THE 2010S

Presenter: **Bruce Hofer**, Audio Precision, Inc.

The design of high performance analog circuits does not happen by accident. It requires very careful attention to detail, choice of components, topology, and physical layout. This session will explore a number of different factors that can limit analog performance along with some recommended solutions. Numerous examples will be presented including measurements and test results.

Special Event
PLATINUM ENGINEERS
Saturday, October 22, 11:00 am – 1:00 pm
Room 1E15/16

Moderators: **Janice Brown**, SonicScoop Co-founder
Justin Colletti, Engineer/Producer/Journalist

Panelists: *Dave Fridmann* (The Flaming Lips, MGMT, Neon Indian)
Peter Katis (The National, Jónsi, Interpol)
Chris Shaw (Bob Dylan, Public Enemy, Weezer)
Damian Taylor (Bjork, The Prodigy)

Creative Engineering—The Studio as an Instrument

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

Saturday, Oct. 22 **11:00 am** **Room 1E02**
Standards Committee Meeting on Sigital Audio Measurement Techniques, SC-02-01

Saturday, Oct. 22 11:00 am Room 1E05
Technical Committee Meeting on Acoustics and Sound Reinforcement

Exhibitor Seminar Saturday, Oct. 22
11:30 am – 12:30 am Room 1E06

PMC: MASTERS OF AUDIO SERIES

Mastering *Come on Over* by Shania Twain, "The World's Best Selling Country Music Album."

Presenter: **Glenn Meadows**

Come On Over was the third studio album recorded by Canadian singer Shania Twain, released on November 4, 1997. It became the world's best-selling country music album and the best-selling studio album ever released by a female artist in any genre. Glenn Meadows presents the story behind this album and the influence mastering had during the process. Glenn will also be giving tips on what makes a good mix for mastering.

Session EB4 Saturday, Oct. 22
12:00 noon – 12:45 pm Room 1E09

SIGNAL PROCESSING

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

12:00 noon

EB4-1 A New Method for Evaluating Loudspeaker Efficiency in the Frequency Domain—*Joe Jensen*, Technical University of Denmark, Lyngby, Denmark

The Constant Input Power (CIP) frequency response is proposed as a new method to evaluate loudspeaker efficiency in the frequency domain. Through a simulation study it is demonstrated how the CIP response can be a valuable tool when designing loudspeakers for which high efficiency is a priority.
Engineering Brief 34

12:15 pm

EB4-2 Wave Field Synthesis by Multiple Line Arrays—*Matthew Montag, Colby Leider*, University of Miami, Coral Gables, FL, USA

Wave field synthesis (WFS) is a spatial audio rendering technique that produces a physical approximation of wavefronts for virtual sources. Large loudspeaker arrays can simulate a virtual source that exists outside of the listening room. The technique is traditionally limited to the horizontal plane due to the prohibitive cost of planar loudspeaker arrays. Multiple-line-array wave field synthesis is proposed as an extension to linear WFS. This method extends the virtual source space in the vertical direction using a fraction of the number of loudspeakers required for plane arrays. This paper describes a listening test and software environment capable of driving a loudspeaker array according to the proposed extension, as well as the construction of a modular loudspeaker array that can be adapted to multiple-line configurations.
Engineering Brief 35

12:30 pm

EB4-3 Playback Disappointment in Linear PCM Recording Systems—*John "Beetle" Bailey*, The Drive Shed - Recording Studios, Toronto, Ontario, Canada

As an "in-the-trenches" music recording engineer, my workflow has evolved to essentially an all in-the-box approach, with the exception of some external DSP. After spending many hours on a mix, I always get a strong feeling that when I finally print my mix to a stereo track in my workstation, or to a hardware-based digital recorder and play back the resulting 24-Bit-96-kHz WAV file, that it's just not the same. A sense of disappointment. It seems to lack depth, reverb tails fall off, the transient response seems dulled, and an overall "graininess" to the mix. This paper and presentation will demonstrate the differences by way of 5.6-MHz DSD null tests and explore the difference between a live digital stream, and the disappointing playback of that digital stream that has been captured by a recorder in WAV file format. Upon demonstrating the problem, I will discuss possible solutions and workarounds I have used.
Engineering Brief 36

Game Audio Session 9 Saturday, Oct. 22
12:00 noon – 1:00 pm Room 1E13

GAME AUDIO PROGRAMMING FOR ANDROID: FMOD vs JAVA

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

Using FMOD audio middleware to produce soundtracks for Android games has certain advantages over using Java methods built into the operating system . . . and vice versa. The author implemented the sound for a pinball app both ways and compares the techniques.

Saturday, Oct. 22 12:00 noon Room 1E05
Technical Committee Meeting on Loudspeakers and Headphones

Workshop 15 Saturday, Oct. 22
2:00 pm – 4:00 pm Room 1E08

MASTERING IN AN EVER EXPANDING UNIVERSE 2011

Chair: **Joe Palmaccio**, The Place . . . for Mastering, Nashville, TN, USA

Panelists: *Vic Anesini*, Battery Studios, New York, NY, USA
Adam Ayan, Gateway Mastering, Portland, ME, USA
Dave Kutch, The Mastering Palace, New York, NY, USA
Gavin Lurssen, Lurssen Mastering, Los Angeles, CA, USA
Andrew Mendelson, Georgetown Masters, Nashville, TN, USA
Michael Romanowski, Romanowski Mastering, San Francisco, CA, USA
Mark Wilder, Battery Studios, New York, NY, USA

Mastering continues to evolve in both form and function. The panel, made up of today's top-tier mastering engineers will discuss this evolution. Technique, technology, best business practices, and aesthetics will be discussed. Panelists will present their own point of view on specific topics. Audio samples and visual aids will be used to demonstrate the thoughts and techniques of the various panelists. At the conclusion of the panel presentations, attendees will be invited to join the discussion and interact with the panelists.

Topics to be discussed include: Preparing a Mix—Best practices for how to deliver mixes to mastering. The DAW—Far beyond a digital editor, computers, storage, and cloud-based tools have radically changed the toolset of ME's and mix engineers. An examination of how to optimize the workstation to achieve the best sonics will be presented. Loudness—A demonstration of why loud mixes are not the same as high quality mastering. The Client—With continued decentralization of the music business, "the client" is no longer represented by a small list of major record label roles. As "who is the client?" expands, education becomes a critical component of running a successful business. A discussion of servicing and maintaining the client will be presented. The Artform—Beyond the technical creation of production masters, aesthetics play a lead role as ME's consider their individual approach to mastering. Get inside the head of a mastering engineer to learn what drives his creative decision making process. The State of Processing Audio—Have computer software and processing finally given the mastering engineer a technical, subjective, and creative equivalent to purpose built analog tools. Formats—CD, DVD, Downloads, and Vinyl are only the beginning of format options today. As the number of deliverable formats expands, so does mastering studio workflow. What represents a master and how is it best delivered.

Hot Lunch
DEMYSTIFYING FIBER OPTICS FOR AUDIO
Saturday, October 22, 1:00 pm – 1:45 pm
Room 1E07

Presenter: **Ronald Ajemian**

This presentation will cover fiber optic fundamentals for pro-audio and explain the principles behind fiber optic technology so that an audience with no prior knowledge will benefit. Mathematics will also be kept to a minimum. You will come away knowing fiber optic jargon and will also be able to answer these (5) main questions:

What is fiber optics?
What are the advantages of fiber optics?
Where is fiber optics being used in audio/video?
How can you apply fiber optics to audio/video?
What types of cables and connectors can be used for audio/video?

The presentation will wrap up with two short video clips. The first clip is on a new fiber optic cable breakthrough called the ClearCurve courtesy of Corning. The second clip will show how easy it is to put a fiber optic connector together using new state of the art tools.

Q & A to follow after tutorial. Bring your notebooks, pens, and pencils. A hard copy of a glossary of terms will be given out.

Exhibitor Seminar **Saturday, Oct. 22**
1:00 pm – 2:00 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

The "Red Hot" Engineer/Producer

Presenter: **Dave Schiffman**

The Red Hot Chili Peppers, LimpBizkit, Nine Inch Nails, Alanis Morrisette, System of a Down, Rage against the Machine, Audioslave, Weezer, and Johnny Cash have all benefited from the production and engineering skills of Dave Schiffman. Dave will share some secrets of mixing rock and alternative music and he will play examples from some of the varied projects he has worked on including LimpBizkit (*Golden Cobra*), Active Child (*You Are All I See*), Cass McCombs (*Wit's End*), and Thrice (*Major/Minor*).

Saturday, Oct. 22 **1:00 pm** **Room 1E02**
Standards Committee Meeting on Loudspeaker Modeling and Measurement, SC-04-03

Saturday, Oct. 22 **1:00 pm** **Room 1E05**
Technical Committee Meeting on Audio for Telecommunications

Lunchtime Keynote
JANE IRA BLOOM
Saturday, October 22, 1:15 pm – 2:15 pm
Room 1E11

***Wingwalker*: Jane Ira Bloom in Conversation with Ashley Kahn**

Jane Ira Bloom is a soprano saxophonist, a composer, and a pioneer in the use of live electronics and movement in jazz. She is the winner of the 2007 Guggenheim Fellowship in music composition, the 2007 Mary Lou Williams Women in Jazz Award for lifetime service to jazz, the Jazz Journalists Association Award, the Downbeat International Critics Poll for soprano saxophone, and the Charlie Parker Fellowship for jazz innovation. Bloom was the first musician commissioned by the NASA Art Program and has an asteroid named in her honor by the International Astronomical Union. She has recorded and produced 14 albums of her music and has composed for the American Composers Orchestra, the St. Luke's Chamber Ensemble, and the Pilobolus Dance Theater, integrating jazz performers in new settings. Bloom is on the faculty of the New School for Jazz & Contemporary Music in NYC. She will be in conversation with journalist and author, Ashley Kahn, and will discuss her latest release, the critically acclaimed CD, *Wingwalker*.

Exhibitor Seminar **Saturday, Oct. 22**
1:30 pm – 2:30 pm **Room 1A01**

TC ELECTRONIC

TC Electronic Presents: When Is it Too Loud?

Presenters: **Thomas Lund** and **Steve Strassberg**

TC Electronic's Thomas Lund and Steve Strassberg will present a focused a seminar covering the new loudness standards and real time solutions for broadcast, podcasts, music creation, mobile audio, and beyond.

Broadcast/Media Streaming Session 11
Saturday, October 22 **2:00 pm – 3:30 pm**
Room 1E10

LIP SYNC ISSUE

Chair: **Jonathan S. Abrams**, Nutmeg Post

Panelists: *Aldo Cugnini*
Adam Goldberg, AGP, LLC
Steve Lyman, Dolby Laboratories
Jackson Wiegman, Evertz

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

Join us as our panel addresses these questions and possible solutions.

Product Design Session 8 **Saturday, Oct. 22**
2:00 pm – 4:30 pm **Room 1E12**

HOT AND NONLINEAR—LOUDSPEAKERS AT HIGH AMPLITUDES

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

Nonlinearities inherent in electro-dynamical transducer and the heating of the voice coil and magnetic system limit the acoustical output, generate distortion and other symptoms at high amplitudes. The large signal performance is the result of a deterministic process and predictable by lumped parameter models comprising nonlinear and thermal elements. The tutorial gives an introduction into the fundamentals, shows alternative measurement techniques and discusses the relationship between the physical causes and symptoms depending on the properties of the particular stimulus (test signal, music). Selection of meaningful measurements, the interpretation of the results and practical loudspeaker diagnostic is the main objective of the tutorial which is important for designing small and light transducers producing the desired output at high efficiency and reasonable cost.

Student/Career Development Event
EDUCATION FORUM PANEL
Saturday, October 22, 2:00 pm – 3:30 pm
Room 1E13

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

Presenters: *Bill Crabtree*
Michael Fleming

Teaching the Teachers—A Round Table Discussion Among Audio Educators

While audio itself—in all her disciplines—advances at breakneck speed, the educators supporting it must make equivalent progress. AES conventions are reliable catalysts for earnest discussions among audio educators. However the convention, rich with so many activities, always seems to end too soon. Curriculum, personnel, and facilities must offer both time-proven fundamentals and cutting edge innovations. It happens in multiple modes: the classroom, the studio, the lab, online, campus committee meetings, and through industry relationships. We've all found solutions here, and nuggets of wisdom there; we wrestle with challenges and unknowns elsewhere. Join this discussion as we seek to define and prioritize the key issues facing educators and create a vision for the most effective way to address them in future AES activities—through conventions, conferences, online interactions, and more. Share your ideas for the most essential forms of research and the best ways to present the results: publications, tutorials, workshops, and other collaborations. What are the topics educators need to discuss, and what is the best format for sharing our advancements? AES provides the essential community for sharing and learning among audio educators. Help us design the next steps for increasing our productivity, accelerating our innovation, enriching our camaraderie, and enhancing our quality as educators.

Exhibitor Seminar **Saturday, Oct. 22**
2:00 pm – 3:00 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

Patricia Barber's *Café Blue* Remixed - 2011

Presenters: **Jim Anderson** and **Bob Ludwig**

Patricia Barber's classic audiophile album was released in 1994 to great acclaim and has been re-released in many different formats over the years. In 2011, producer Michael Friedman and original engineer Jim Anderson went to Capitol Studios and reworked the project for a new vinyl release. Jim Anderson and Mastering Engineer Bob Ludwig will play tracks from the new version and discuss the process of renewing a classic.

Saturday, Oct. 22 **2:00 pm** **Room 1E05**
Technical Committee Meeting on Audio Forensics

Session P20 **Saturday, Oct. 22**
2:30 pm – 4:30 pm **Room 1E09**

AUDIO PROCESSING—PART 2

Chair: **James (JJ) Johnston**, Consultant

2:30 pm

P20-1 Digital Low-Pass Filter Design with Analog-Matched Magnitude Response—*Michael Massberg*, Brainworx Music & Media GmbH, Leichlingen, Germany

Using the bilinear transform to derive digital low-pass filters from their analog prototypes introduces severe warping of the response once the cutoff frequency approaches the Nyquist limit. We show how to design pre-warped first and second order low-pass prototypes that, after application of the bilinear transform, provide a better match with the magnitude response of the analog original than applying the bilinear transform directly. Result plots are given and compared for different cutoff frequencies and Q factors.
Convention Paper 8551

3:00 pm

P20-2 Performance Evaluation of Algorithms for Arbitrary Sample Rate Conversion—*Andreas Franck*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Arbitrary sample rate conversion (ASRC) enables changes of the sampling frequency by flexible, time-varying ratios. It can be utilized advantageously in many applications of audio signal processing. Consequently, numerous algorithms for ASRC have been proposed. However, it is often difficult to choose a minimal-cost algorithm that meets the requirements of a specific application. In this paper several approaches to ASRC are reviewed. Special emphasis is placed on algorithms that enable optimal designs, which minimize the resampling error with respect to a selectable norm. Evaluations are performed to assess the computational efficiency of different algorithms as a function of the achievable quality. These analyses demonstrate that structures based on oversampling and resampling filters specifically adapted to this structure yield significant performance improvements over established algorithms.
Convention Paper 8552

3:30 pm

P20-3 Acoustic Echo Cancellation for Surround Sound Using Spatial Decorrelation—
Namgook Cho, Jaeyoun Cho, Jaewon Lee, Yongje Kim, Samsung Electronics Co., Ltd., Korea

One of the main challenges for a stereophonic acoustic echo canceller is that it suffers from poor convergence, which is caused by strong correlation between input signals. We have proposed a new decorrelation technique that adopts spatial decorrelation to address the problem without altering the input signals. Here, we extend the results to a more generic setting, i.e., a 5.1-channel surround system. In the scheme, the input signals are decomposed and projected into the signal subspace and the noise subspace. When the decorrelated signals are fed to the adaptive filters, the interchannel coherence between the input signals decreases significantly, which provides performance improvement in echo reduction. Experiments in a real-world environment and performance comparison with state-of-the-art techniques are conducted to demonstrate the effectiveness of the proposed technique.
Convention Paper 8553

4:00 pm

P20-4 A Fixed Beamforming Based Approach for Stereophonic Audio-Conference Systems—
Matteo Pirro, Stefano Squartini, Francesco Piazza, Università Politecnia delle Marche, Ancona, Italy

Hands-free communications systems require to primarily reduce the impact of the inevitably occurring acoustic echo. Moreover, in the recent past, a certain attention has been devoted to algorithmic frameworks able to provide stereophonic acoustic rendering and so augmenting the pleasantness of the audio-conference experience. In this paper the authors propose an optimally designed fixed Beamformer (BF) based solution for Stereophonic Acoustic Echo Cancellation (SAEC), with a twofold objective in mind: reducing the echo power and maximizing the stereophonic spatial feeling. Up to the author's knowledge this is new to the literature, and the achieved experimental results seem to confirm the effectiveness of the approach. It must be underlined, on purpose, that preliminary subjective listening tests have been carried out to evaluate the attainable audio stereo-recording quality. Moreover, the proposed solution allows to significantly reduce the overall computational cost required by the SAEC framework: indeed, BF implementation asks for few extra filtering operations with respect to the baseline approach from one hand but make the usage of the decorrelation module unessential.
Convention Paper 8554

Session P21 — Saturday, Oct. 22
2:30 pm – 4:00 pm — 1E Foyer

POSTERS: PERCEPTION

2:30 pm

P21-1 Influence of Different Test Room Environments on IACC as an Objective Measure of Spatial Impression or Spaciousness—*Marco Conceição*,[1,2] *Dermot Furlong*[1]
[1]Trinity College Dublin, Dublin, Ireland
[2]Escola Superior de Musica, Artes e Espectáculo–IPP, Porto, Portugal

To investigate the perceptual impression of spaciousness, a physical measure, which relates to listener spaciousness experience, was used. A variable setup was introduced that made possible the control of spaciousness in different rooms. IACC measurements were made using a frontal loudspeaker for the direct sound, with a second loudspeaker for an angled single early reflection positioned in the horizontal plane. Measurements were performed under controlled conditions in which a dummy-head measured Inter Aural Cross Correlation was captured for variable sound fields. It was possible to conclude that there is a similar trend in IACC results from the repetition of the experiments in different rooms. That is, measurement room acoustic details are not crucial to observed trends in IACC variation.
Convention Paper 8555

2:30 pm

P21-2 The Relationship between Interchannel Time and Level Differences in Vertical Sound Localization and Masking—*Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Listening experiments were conducted with a pair of vertically arranged loudspeakers. A group of subjects measured the level of delayed height channel signal at which any subjective effects of the signal became completely inaudible (masked threshold) as well as that at which the perceived sound image was localized fully at the lower loudspeaker (localized threshold), at nine different delay times ranging from 0 to 50 ms. The sound sources were anechoic recordings of bongo and cello performance excerpts. At the delay times up to 5 ms, source type did not have a sig-

nificant effect for both threshold results and neither threshold varied significantly as the delay time increased. In this time range the average level reduction required for a full image shift was 6~7 dB while that for masking was 9~10dB. At the higher delay times, on the other hand, both thresholds decreased as the delay time increased and the difference between the two sources in both threshold results was significant. Furthermore, the relationship between the two thresholds varied depending on the source type.
Convention Paper 8556

2:30 pm

P21-3 Observations on Human Sound Source Localization in the Mid-Sagittal Plane and in Anechoic Space—*Daniela Toledo*, COWI AS, Oslo, Norway

A group of 5 subjects who showed consistently biased sound source localization in the mid-sagittal plane with real sound sources and under anechoic conditions is presented. Three of these subjects were also tested with virtual sound sources synthesized with their own individual head-related transfer functions. Localization under both conditions showed similar trends, even though they could not be considered as equivalent. This suggests that binaural technology was close to emulating the aural experience that the subjects had with real sound sources. These cases are presented to discuss different issues inherent to binaural synthesis, like the way in which the technology is validated and the assumptions that serve as a basis for the technology.
Convention Paper 8557

2:30 pm

P21-4 Reducing the Cost of Audiovisual Content Classification by Experiment—*Ulrich Reiter*, Norwegian University of Science and Technology – NTNU, Trondheim, Norway

A set of subjective attributes of audiovisual media content, originally suggested by Woszczyk et al. in a 1995 AES Convention paper was examined for suitability for AV content classification tasks. Subjective experiments indicate that the 4x4 matrix of dimensions and attributes as suggested in the original paper can be reduced to a more compact 3x2 design for classification purposes without loss of information about the perceptual properties of the content. This can significantly reduce the cost of content classification by experiment.
Convention Paper 8558

2:30 pm

P21-5 Quality and Performance Assessment of Wave Field Synthesis Reproducing Moving Sound Sources—*Michele Gasparini, Paolo Peretti, Laura Romoli, Stefania Cecchi, Francesco Piazza*, Universita Politecnica delle Marche, Ancona, Italy

Reproduction of moving sound sources by Wave Field Synthesis (WFS) arises some specific problems that make static source approach ineffective. An algorithm for the reduction of artifacts and natural representation of the movement effect, based on the synthesis of two virtual sources moving together, has been proposed in a previous work by the same authors. In this paper a practical implementation of the algorithm is presented and evaluated in terms of workload and subjective sound source localization. The influence of some processing parameters on the computational cost has been studied. The sound quality concerning true spatial reproduction is assessed through listening tests and a comparison with previous approaches is reported.
Convention Paper 8559

Tutorial 12 **Saturday, Oct. 22**
2:30 pm – 4:00 pm **Room 1E07**

THE BASICS OF ARCHIVAL PRESERVATION

Presenter: **James Sam**, Hoover Institution, Stanford, University, Stanford, CA, USA

The audio preservation program at the Hoover Institution Archives of Stanford University is a real-world implementation of archival best practices. Two large collections of the Archives are the Radio Free Europe / Radio Liberty and Commonwealth Club of California collections. They are eerily similar in their recording formats and time spans despite being generated on two different continents, yet the archival approach remains the same. Mr. Sam will discuss this approach using these examples and its implications for both legacy and new recordings. He will describe preservation methods used, employing fascinating examples from the collections.

Live Sound Seminar 10 **Saturday, Oct. 22**
2:30 pm – 4:30 pm **Room 1E14**

AN INTERVIEW WITH ROBERT SCOVIL AND DAVE NATALE

Moderator: **Keith Clark**

Panelists: *Robert Scovil*
Dave Natale

Two FOH giants discuss technique, philosophy, and over sixty years combined experience mixing premier musical acts on a wide range of sound systems. With Keith Clark of Pro Sound Web.

Special Event
GRAMMY SOUNDTABLE
Saturday, October 22, 2:30 pm – 4:00 pm
Room 1E15/16

Moderator: **Chris Lord-Alge**

Panelists: *Ken "Duro" Ifill*
Steve Lillywhite
Ann Mincieli
Phil Ramone
Al Schmitt

Some songs are hits, some we just love, and some have changed our lives. Our panelists break down the DNA of their favorite tracks and explain what moved them, what grabbed them, and why these songs left a life-long impression. Back by popular demand, this reprise is a New York-centric version of the GRAMMY SoundTable ➥

presented in San Francisco during the 129th Convention, and is guaranteed to make you feel good about being in the recording business.

Workshop 16 Saturday, Oct. 22
2:45 pm – 4:45 pm Room 1E11

RECORDING SURROUND SOUND MUSIC

Presenter: **Morten Lindberg**, Lindberg Ltd., Oslo, Norway

Balance Engineer and Recording producer Morten Lindberg presents 2L's approach to music in extreme surround sound. Playing original 5.1 masters in 352.8 kHz/24 bit, showing photos and stage layout from recording sessions and discussing the resources behind "the Nordic Sound." Morten Lindberg has produced nine GRAMMY-nominations since 2006. Six of these in categories "Best Engineered Album" and "Best Surround Sound Album."

Saturday, Oct. 22 3:00 pm Room 1E02
Standards Committee Meeting on Acoustics and Sound Source Modeling, SC-04-01

Saturday, Oct. 22 3:00 pm Room 1E05
Technical Committee Meeting on Fiber Optics for Audio

Broadcast/Media Streaming Session 12
Saturday, October 22 3:30 pm – 5:00 pm
Room 1E10

CANCELLED

Exhibitor Seminar Saturday, Oct. 22
3:30 pm – 4:30 pm Room 1E06

PMC: MASTERS OF AUDIO SERIES

From Bedroom to Dubbing Stage

Presenter: **Daniel Porter**

Daniel Porter from Auralex Acoustics will demonstrate the importance of the right acoustics in any space where critical decisions are made during the creation of music.

Workshop 17 Saturday, Oct. 22
4:00 pm – 6:30 pm Room 1E08

LOW-DELAY AUDIO CODING FOR HIGH-QUALITY COMMUNICATION

Chair: **Manfred Lutzky**, Fraunhofer Institute for Integrated Circuits IIS

Panelists: *Bernhard Grill*, Fraunhofer IIS
Milan Jelinek, VoiceAge Corporation
Sascha Spors, Deutsche Telekom Laboratories

After many years of narrow-band (300 Hz–3.4 kHz) voice communication history, a strong trend toward higher speech and audio quality for communication has emerged. Driven by the industry, enhanced services and products become available, such as wide band audio for mobile phones, room telepresence systems, and Apple's iPhone/FaceTime mobile videoconferencing. This workshop presents latest research results on the requirements for these applications from the European research project TA2. The underlying key technology is discussed, i.e., new low delay codecs—standardized by MPEG and ITU-T—providing high quality with full audio bandwidth and stereo transmission even at low data rates. Finally, one of the most promising ongoing standardization activities, the 3GPP Enhanced Voice Service (EVS) is presented as an outlook from a technology and service perspective.

Games Audio Session 10 Saturday, Oct. 22
4:00 pm – 5:30 pm Room 1E13

MAKING MUSIC FOR GAMES: THE STATE OF THE ART

Presenters: **Simon Ashby**, VP Product Strategy, Co-founder at Audiokinetic
Jason Graves, Composer, Jason Graves Music
Tom Salta, Composer, Persist Music

This renowned panel of experts pulls the curtain back on the process of making music for blockbuster games. Get the inside perspective on the latest trends in the composition, production, and integration of music for interactive entertainment.

Saturday, Oct. 22 4:00 pm Room 1E05
Technical Committee Meeting on Perception and Subjective Evaluation of Audio

Saturday, Oct. 22 4:15 pm Room 1E03
Historical Committee Meeting

Tutorial 13 Saturday, Oct. 22
4:30 pm – 6:30 pm Room 1E09

TELEPHONOMETRY: THE PRACTICAL ACOUSTICS OF HANDSETS, HEADSETS, AND MOBILE DEVICES

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

This tutorial introduces the basic concepts of Telephonometry with respect to electroacoustic measurements on analog and digital telephones. Both subjective and objective methods are discussed and the historical concept of loudness rating and standardized methods for its calculation are reviewed. Standard objective measurements of send, receive, sidetone, and echo response are explained. The selection and use of appropriate instrumentation, including ear and mouth simulators, is also described. Techniques for the evaluation of handsets, headsets, speakerphones, and other hands-free devices are presented. Applications of these measurements to analog, digital, cellular, and VOIP devices are shown. Various methods specified in the ITU-T, IEEE, TIA, ETSI, and 3GPP standards are explained.

Master Class 3 Saturday, Oct. 22
4:30 pm – 6:30 pm Room 1E07

THE HYBRID EDIT/MIX APPROACH TO FEATURE FILM SOUND

Presenter: **Skip Lievsay**

Skip Lievsay has worked on the sound-design and mix-

ing of such blockbuster films as True Grit, No Country For Old Men, I Am Legend, It Might Get Loud, and Men In Black. Skip will illustrate his "Hybrid Edit/Mix" approach to post-production sound that erases the traditional lines between sound design, editing, ADR, and the final mix, as all these actions are performed concurrently with the picture edit. The process provides the director with a thorough appreciation of the sound during the crucial early stages of post-production. Most of the mixing happens during this period rather than at the final dubbing stage. Live demonstrations will be given using a ProTools workstation.

Special Event
LEGENDS OF NASHVILLE SOUND
Saturday, October 22, 4:30 pm – 6:30 pm
Room 1E15/16

Moderator: **Cosette Collier**

Panelists: *Wes Bulla*, Belmont University
Jim Kaiser
Mike Poston, Audio Services Group
Bil VornDik, Producer/Engineer

The Nashville section of AES present a continuation of their previous presentation, "Legends of Nashville Sound/History of Nashville's Recording Studios." This latest installment features the engineers, producers, and studio musicians responsible for the "Nashville Sound." The Nashville studio eras that will be featured are "The RCA Years" (beginning early 50s), "The Monument Studio Years" (beginning early 60s), "The Woodland Years" (beginning mid to late 60s), and "The Jack 'Cowboy' Clement Years" (beginning early 70s).

Exhibitor Seminar **Saturday, Oct. 22**
4:30 pm – 5:30 pm **Room 1A01**

PELONIS SOUND AND ACOUSTICS, INC.

Don't Believe Everything You Hear

Presenter: **Chris Pelonis**

An in-depth look at what can go wrong if you are not hearing the truth in studio monitoring systems.

Saturday, Oct. 22 **4:30 pm** **Room 1E02**
Standards Committee Meeting on Microphone Measurement and Characterization, SC-04-04

Broadcast/Media Streaming Session 13
Saturday, October 22 **5:00 pm – 6:30 pm**
Room 1E10

IMPROVING THE STREAMING AUDIENCE EXPERIENCE

Chair: **Bill Sacks**, Orban

Panelists: *Carl Edwards*, Pandora
Steven Harris, BridgeCo
Rusty Hodge, SomaFM
Greg Ogonowski, Orban
Skip Pizzi, NAB

In today's highly competitive digital media environment the listener/viewer is being presented with millions of choices. They have no patience for glitchy players or phasey sounding tin horn unprocessed audio that is overly dynamic for the medium. Marshall McLuhan said this best when he said, "The medium is the message." Our audiences are now experiencing high quality audio on seamless players and if your stream is not one of these better quality streams, those fickle users will be lost forever with one bad experience.

Live Sound Seminar 11 **Saturday, Oct. 22**
5:00 pm – 7:30 pm **Room 1E14**

TVBDS AND THE GEO-LOCATION DATABASE: HOW THEY WORK AND THE NEW PROTOCOLS FOR OPERATING WIRELESS MICROPHONES

Chair: **Henry Cohen**, Production Radio

Panelists: *Jesse Caulfield*, Key Bridge Global
Joe Ciaudelli, Sennheiser
Ira Keltz, Federal Communications Commission
Michael Marcus, Marcus Spectrum Solutions
Steve Mendelsohn, ABC, NY Jets
David Pawlik, Skadden Arps
Edgar Reihl, Shure, Inc.
James Stoffo, independent entertainment production RF coordinator

Television band devices (TVBD) are a reality and the first of the geo-location databases directing TVBD operations are in the testing phase. Operating wireless microphones, IEMs, intercoms, and cueing in this new environment requires understanding how the databases work and the rules governing both licensed and unlicensed wireless production equipment. This panel brings together a diverse group of individuals intimately involved from the beginning with TVBDs, databases, and the new FCC rules as well as seasoned veterans of medium to large scale wireless microphone deployments to discuss how the databases operate, how to use the database for registering TV channel usage, and best procedures and practices to insure minimal problems.

Product Design Session 9 **Saturday, Oct. 22**
5:00 pm – 6:30 pm **Room 1E12**

SPECIFYING CLASS D SOLUTIONS

Presenter: **Brian Oppegaard**, SpeakerPower

Class D offers high-efficiency and light weight. Turnkey solutions and reference designs abound but how does one choose the most appropriate one? There is a lot more to it than price, power, size and weight. This panel of experts will discuss the considerations one needs to make.

Student/Career Development Event
RECORDING COMPETITION—2
Saturday, October 22, 5:00 pm – 7:00 pm
Room 1E11

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

• Traditional Multitrack Studio Recording 4:30 pm to 5:30 pm
Judges: Jim Anderson, Brandie Lane, Mark Rubel ➥

- Modern Multitrack Studio Recording 5:30 pm to 6:30 pm

  Judges: John Merchant, Ronald Prent, Darcy Proper

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

Saturday, Oct. 22 5:00 pm Room 1E05
Technical Committee Meeting on Hearing and Hearing Loss Prevention

Games Audio Session 11 Saturday, Oct. 22
5:30 pm – 6:30 pm Room 1E13

USING SPEECH RECOGNITION FOR GAMES

Presenter: **Scott Selfon**, Microsoft

Voice offers a powerful natural user input mechanism, with the possibility of completely intuitive and seemingly omnipotent control for the player. How should speech be designed to best take advantage of this capability, to set up the player for maximum success, and to create magical experiences even when the player's spoken word isn't always perfectly understood? This lecture will focus on best practices for designing, testing, and tuning grammars and overall player engagement for intuitive and successful speech recognition experiences.

Student/Career Development Event
STUDENT PARTY
Saturday, October 22, 7:30 pm – 10:00 pm
NYU's James L. Dolan Music Recording Studio
35 West 4th St., 6th Floor

Audio Students! Join us for a fun and exciting evening at the AES Student Party to be held at NYU's James L. Dolan Music Recording Studio, a 7,500 square foot multifunctional teaching, recording, and research space. This is one of the most technologically advanced audio teaching facilities in the United States and a great place for pizza and music. The studio is located on the sixth floor of 35 West 4th Street but tickets must be purchased in advance, either at the first meeting of the Student Delegate Assembly or at the AES Student Chapters Booth outside of the Exhibit Hall.

Special Event
ORGAN CONCERT
Saturday, October 22, 8:00 pm – 9:30 pm
Central Synagogue
652 Lexington Avenue at 55th Street, New York

Organist Graham Blyth's concerts are a highlight of every AES convention at which he performs. This year's recital will be held at Central Synagogue. He will play Fantasia & Fugue in G minor by J.S. Bach; "Come, Sweet Death" by J. S. Bach arr. Virgil Fox; Tuba Tune by Norman Cocker; Prelude, Fugue & Variation by Cesar Franck; and Sonata No. 1 in D minor by Alexandre Guilmant.

Constructed by the renowned firm of Casavant Frères of St. Hyacinthe, Canada and completed in 2002, the Gabe M. Wiener Memorial Organ consists of two distinct, interconnected instruments: a Bimah Organ (Casavant Opus 3812) located on both sides of the bimah and used primarily during services to accompany the cantor, choir, and congregation; and a larger Gallery Organ (Casavant Opus 3813) located in the elevated rear gallery and used both for services and concerts. It contains two consoles and 4,345 pipes, 55 stops, and 74 ranks, located in the front and back of the sanctuary. The Bimah Organ, with Choeur, Echo, and Pédale divisions (groups of pipes) was installed and voiced in July 2001, in time for the rededication of the sanctuary on September 9, 2001. The Gallery Organ, with Grand Orgue, Récit, Positif, Solo, and Pédale divisions, was installed and voiced in March 2002. Both coordinate in style and materials with the design of the restored sanctuary. The entire instrument was dedicated at a concert on April 10, 2002, by concert organist David Higgs and the Orpheus Chamber Orchestra. Both organs can be played from separate movable consoles: the Bimah console, which has three keyboards, and a Gallery console that has four. Either can control the entire organ. The Bimah console is equipped with 40 pistons, 31 couplers, and 30 toe studs. The Gallery console is equipped with 80 pistons, 24 couplers, and 34 toe studs. Both consoles have solid-state combination systems with 128 levels of memory, MIDI connections, transposers, and many other amenities. The organ contains two very special stops created specifically for Central Synagogue: a Trompette Shofar, that replicates the sound of the traditional shofar, used for services on Rosh Hashanah and Yom Kippur; and a Klezmer Clarinette, that reproduces the sound of a klezmer clarinet with great brilliance and clarity, believed to be the first such organ stop in the world. Both are used to enrich the accompaniment of contemporary anthems and liturgical music. The instrument also contains a rich array of other reed registers, including a Trompette-de-Fête that can sound out over the entire organ, and a 32-foot Contre-Bombarde in the pedal division that provides floor-shaking bass to the full ensemble. The organ was designed by Pierre Dionne, President of Casavant Frères, and Jacquelin Rochette, Associate Tonal Director, in conjunction with George B. Stauffer and Shelly Palmer, who served as organ consultants for Central Synagogue. It is the product of three years of planning and a cumulative total of 21,000 work-hours by Casavant's artisans and musicians.

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

He divides his time between being a designer of professional audio equipment (he is a co-founder and Technical Director of Soundcraft) and organ related activities. In 2006 he was elected a Fellow of the Royal Society of

Arts in recognition of his work in product design relating to the performing arts.

Session P22 **Sunday, Oct. 23**
9:00 am – 12:30 pm **Room 1E09**

LISTENING TESTS

Chair: **Duncan Williams**, University of Oxford, UK

9:00 am

P22-1 Comparison of Subjective Assessments Obtained from Listening Tests through Headphones and Loudspeaker Setups—*Vincent Koehl,*[1] *Mathieu Paquier,*[1] *Simeon Delikaris-Manias*[2]
[1]University of Brest (UEB), Plouzané, France
[2]National Engineering School of Brest (UEB), Plouzané, France

Sound reproduction over headphones is, because of its convenience, indifferently used to reproduce and assess a large variety of audio contents. Nevertheless, it is not yet proven that differences between sound sequences are equally perceived when played back through headphones as using dedicated loudspeaker systems. This study aims at evaluating whether differences and preferences between excerpts are equally perceived using these two reproduction methods. Various types of audio contents, issued by two different recording systems, were then to be compared on both headphones and loudspeaker setups. The results indicate that the two reproduction methods provided consistent similarity and preference judgments. This suggests that the features involved in similarity and preference assessments were preserved when reproducing these excerpts over headphones.
Convention Paper 8560

9:30 am

P22-2 A Subjective Validation Method for Musical Instrument Emulation—*Leonardo Gabrielli,*[1] *Stefano Squartini,*[1] *Vesa Välimäki*[2]
[1]Università Politecnica delle Marche, Ancona, Italy
[2]Aalto University, Espoo, Finland

This paper deals with the problem of assessing the distinguishability between the sound generated by an acoustical or electric instrument and an algorithm designed to emulate its behavior. To accomplish this, several previous works employed subjective listening tests. These are briefly reviewed in the paper. Different metrics to evaluate test results are discussed as well. Results are reported for listening tests performed on the sound of the Clavinet and a computational model aimed at its emulation. After discussing these results a guideline for subjective listening tests in the field of sound synthesis is proposed to the research community for further discussion and improvement.
Convention Paper 8561

10:00 am

P22-3 Exploratory Studies on Perceptual Stationarity in Listening Tests—Part I: Real World Signals from Custom Listening Tests —*Max Neuendorf,*[1] *Frederik Nagel,*[1,2]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories, Erlangen, Germany

Many recent publications related to audio coding use the recommendation “MUltiple Stimuli with Hidden Reference and Anchor” (MUSHRA; ITU-R BS.1534-1) for the evaluation of subjective audio quality. Judging the quality of multiple conditions can be inconclusive if the employed test excerpts exhibit more than one prevalent artifact. Two papers investigate the impact of time varying artifacts in both, synthetic and real world signals and claim "perceptual stationarity" as a requirement for test sequences used in MUSHRA tests. This first part deals with commonly used test signals. These often have a length of 10 to 20 seconds and frequently contain time varying perceptual artifacts. Ratings of those items are compared to ratings of cutouts that are predominantly perceptually stationary over time.
Convention Paper 8562

10:30 am

P22-4 Exploratory Studies on Perceptual Stationarity in Listening Tests—Part II: Synthetic Signals with Time Varying Artifacts—*Frederik Nagel,*[1,2] *Max Neuendorf*[1]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]International Audio Laboratories, Erlangen, Germany

Many recent publications related to audio coding use the recommendation “MUltiple Stimuli with Hidden Reference and Anchor” (MUSHRA; ITU-R BS.1534-1) for the evaluation of subjective audio quality. Judging the quality of multiple conditions can be inconclusive if the employed test excerpts exhibit more than one prevalent artifact. Two papers investigate the impact of time varying artifacts in both, synthetic and real world signals and claim "perceptual stationarity" as a requirement for test sequences used in MUSHRA tests. This second part focuses on the alternation of multiple types of artifacts in one item and the differences in the ratings compared to items which only contain one of the respective types. It furthermore discusses the meaning of the temporal position of artifacts in an item.
Convention Paper 8563

Convention Paper 8564 has been withdrawn

11:00 am

P22-5 The Practical Effects of Lateral Energy in Critical Listening Environments—*Richard King,*[1,2] *Brett Leonard,*[1,2] *Grzegorz Sikora*[1]
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canada

Limited information exists on the practical effects of lateral reflections in small rooms design for high-quality sound reproduction and critical listen-

ing. A study is undertaken to determine what affect specular and diffuse lateral reflections have on a trained listener. A task-based methodology is employed in which a highly trained subject is asked to perform a task commonly seen in their daily work. The physical conditions of the listening environment are altered to minimize, maximize, and diffuse side-wall reflections. Results correlate the presence of strong lateral energy with an initial reduction of subjects' ability to complete the task within normal tolerances, but adaptation soon occurs, restoring the subjects to practically normal pace and accuracy.
Convention Paper 8565

11:30 am

P22-6 The Effects of Monitoring Systems on Balance Preference: A Comparative Study of Mixing on Headphones versus Loudspeakers
—*Richard King*,[1,2] *Brett Leonard*,[1,2] *Grzegorz Sikora*[1]
[1]McGill University, Montreal, Quebec, Canada
[2]Centre for Interdisciplinary Research in Music, Media and Technology, Montreal, Quebec, Canada

The typical work-flow of the modern recording engineer often necessitates the use of a number of different monitoring systems over the course of a single project, including both loudspeaker-based systems and headphones. Anecdotal evidence exists that suggests different outcomes when using headphones, but there is little quantified, perceptually-based data to guide engineers in the differences to expect when working between monitoring systems. By conducting controlled, in situ measurements with recording engineers performing mixing tasks on both headphones and loudspeakers, the practical effects of both monitoring systems are shown.
Convention Paper 8566

12:00 noon

P22-7 The Effect of Head Movement on Perceived Listener Envelopment and Apparent Source Width—*Anthony Parks, Jonas Braasch*, Rensselear Polytechnic Institute, Troy, NY, USA

This study investigates the effect of head movement in the evaluation of LEV and ASW under 15 different concert hall conditions simulated over eight loudspeakers using Virtual Microphone Control. The conditions consist of varying ratios of front-to-back energy and varying levels of cross-correlated reverberant energy. Head movements are monitored in terms of angular rotation using a head tracker while listeners are prompted to assign subjective ratings of LEV and ASW. The tests are repeated while listeners are asked to keep their heads fixed. Head movements are analyzed and results of the tests are compared.
Convention Paper 8567

Workshop 18 Sunday, Oct. 23
9:00 am – 11:00 am Room 1E11

PANNING FOR MULTICHANNEL LOUDSPEAKER SYSTEMS

Chair: **Ville Pulkki**, Aalto University, Helsinki, Finland

Panelists: *Jan-Mark Batke*, Technicolor, Hannover, Germany
Craig Jin, University of Sydney, Sydney, NSW, Australia
Sascha Spors, Deutsche Telekom Laboratories, Technische Universität Berlin, Berlin, Germany

Amplitude panning is the most used method to position virtual sources over layouts where the number of loudspeakers is between two and about forty. The method is really simple, it provides a nice spatial effect, and does not color the sound prominently. This workshop reviews the working principle and psychoacoustic facts of amplitude panning for stereophony and for multichannel layouts. The panelists will describe some recent improvement suggestions to amplitude panning, which target some shortcomings of amplitude panning in spatial accuracy. A lively discussion is assumed on pros and cons of such processing.

Broadcast/Media Streaming Session 14
Sunday, October 23 9:00 am – 10:30 am
Room 1E10

CONSIDERATIONS FOR FACILITY DESIGN

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

Panelist: *Dirk Noy*, Walters Storyk Design Group

Examples of what we will discuss include:

1. Pre-design: Translating your business plan into spaces and places. The right shapes for the creative workplace leasehold / building. Important Criteria for comparing potential leaseholds/ buildings—Can you get there? Circulation and pathways for People, Stuff & Content; Is the structure and infrastructure ready for your business? Can your workflow work there? An Essential Design Methodology: Due Diligence, Analysis, Concepts, Policy, Design, Manage, Realize. The Project Team: Who you need, how to hire and manage them. Budget and Schedule, every step of the way.
2. Design: Trends in the changing multi-media workplace. Simple techniques to manage acoustical and technical performance with minimum expense. The importance of Coordination.
3. Purchasing: No need to re-invent any wheels—standard processes, contracts, and business structures.
4. Construction Administration: Roles of Project Team members. Managing changes in scope and schedule. One place, seven contractors, one on-air date.
5. Future Proofing: How to plan for upgrades, new developments in fiber optics and changes in production/ post production formats and procedures. How evaluate new technology introductions, how to separate meaningful innovations from "smoke and mirror" trends. 5.1 yes. 3-D? Odorama? The jury may still be out.

Attendees will leave with a good overview of best practices, opportunities and pitfalls related to capital project development for the industries represented at AES.

Games Audio Session 12 Sunday, Oct. 23
9:00 am – 11:00 am Room 1E13

EDUCATION: PREREQUISITES FOR A CAREER IN GAME AUDIO

Presenters: **Karen Collins**, University of Waterloo

Stephen Harwood Jr., IASIG; Dynamic Systems Music
David Javelosa, Santa Monica College
Michael Sweet, Berklee College of Music

The video game industry has grown and evolved tremendously in recent years, and the opportunities for further expansion are limitless. Much of this great potential is the result of an ever-increasing demand for technological innovation. Corollary to this, students wishing to enter the games industry must have a solid education in related technologies and production techniques. To address this need, the IASIG has recently published a Game Audio Curriculum Guideline. In this session, a panel of IASIG Education Working Group members—accomplished industry veterans and professional educators—will share their experience and perspectives on a broad range of professional topics, with a focus on how students can prepare for, begin, and develop a successful career in game audio. The discussion will also include an outline of the curriculum guidelines and suggestions for educators and institutions.

Sunday, Oct. 23 9:00 am Room 1E02
Standards Committee Meeting, AESSC Plenary

Sunday, Oct. 23 9:00 am Room 1E05
Technical Committee Meeting on Studio Practices and Production

Session P23 Sunday, Oct. 23
9:30 am – 11:00 am 1E Foyer

POSTERS: SPATIAL AUDIO PROCESSING—PART 1

9:30 am

P23-1 System Theory of Binaural Synthesis— *Florian Völk*, AG Technische Akustik, MMK, Technische Universität München, Munich, Germany

Binaural synthesis is widely used as an efficient tool for the simulation of acoustical environments. Different headphones together with artificial as well as human heads are employed for the transfer function measurements involved, having considerable influence on the synthesis quality. Within this paper, a detailed system theoretical analysis of the signal paths and systems involved in a typical data based binaural synthesis scenario is given. The components to be equalized are identified, and equalization methods for every scenario are discussed. Further, restrictions and necessities when using artificial or human recording heads and for headphone selection are given. Most important results are the necessity of blocked auditory canal measurements and the selection of proper headphones for completely correct individual binaural synthesis.
Convention Paper 8568

9:30 am

P23-2 A Quantization Method Based on Binaural Perceptual Characteristic for Interaural Level Difference—*Heng Wang, Ruimin Hu, Weiping Tu, Xiaochen Wang*, Wuhan University, Wuhan, Hubei, China

In this paper we study the mechanism exists of perceptual redundancy in spatial parameters and remove redundancy from energy domain to parameter domain. We establish a binaural perceptual model based on frequency dependent of Interaural Level Difference (ILD) and use this model to direct quantization of ILD. It solves the problem that the perceptual redundancy of spatial parameter is difficult to remove. The new quantization strategy merely quantizes the perceived variable quantity of ILD to reduce the coding bit rate. Experimental results showed that this method can bring down the parametric bit rate by about 15% compared with parametric stereo, while maintaining the subjective sound quality.
Convention Paper 8569
[Paper not presented but is available for purchase]

9:30 am

P23-3 Estimation of Head-Related Impulse Responses from Impulse Responses Acquired for Multiple Source Positions in Ordinary Sound Field— *Shouichi Takane, Koji Abe, Kanji Watanabe, Sojun Sato*, Akita Prefectural University, Akita, Japan

In this paper a new method for estimation of Head-Related Impulse Responses (HRIRs) from the impulse responses acquired in ordinary sound field is proposed. Estimation of a single HRIR from the impulse response in the same direction acquired in the ordinary sound field was proposed, and the estimation performance was shown to be not enough in some directions [S. Takane, 127th AES Convention, Paper No. 7885 (2009)]. In order to improve the estimation accuracy, the impulse responses acquired at multiple source positions are used, and the Auto Regressive (AR) coefficients of the HRIRs are assumed common for all source positions in the proposed method. The results of the example estimation of the HATS' HRIRs showed that the estimation accuracy was significantly improved comparing with our previously proposed method.
Convention Paper 8570

9:30 am

P23-4 Toward the Creation of a Standardized HRTF Repository—*Areti Andreopoulou, Agnieszka Roginska*, New York University, New York, NY, USA

One of the main spatial audio topics, nowadays, involves working toward an efficient individualization method of Head Related Transfer Functions. A major limitation in this area of research is the lack of a large and uniform database that will incorporate as many individualized properties as possible. This paper presents the MARL-NYU file format for storing HRTF datasets, and investigates the necessary normalization steps that assure a uniform and standardized HRTF repository, by compiling selected datasets from four HRTF databases.
Convention Paper 8571

9:30 am

P23-5 On the Synthetic Binaural Signals of Moving

Sources—*Nara Hahn, Doo-Young Sung, Koeng-Mo Sung*, Seoul National University, Seoul, Korea

Binaural signals of moving sources are synthesized using head-related impulse responses. The ear signals are synthesized such that the physical properties are correctly contained. The source signal at each time instance is filtered by the instantaneous head-related impulse response, and this wavelet is superimposed at the external ear. A number of properties of synthetic binaural signals are investigated. The spectral shift and head shadowing effect are analyzed in the time domain and in the time-frequency domain. The interpolation/extrapolation methods are employed to compute unmeasured head-related impulse responses. Artifacts caused by these processes are briefly reviewed.
Convention Paper 8572

9:30 am

P23-6 The Role of Head Related Transfer Functions' Spectral Features in Sound Source Localization in the Mid-Sagittal Plane— *Daniela Toledo*, COWI AS, Oslo, Norway

The individual nature of HRTFs is responsible for localization errors when non-individual HRTFs are used in binaural synthesis: localization performance is degraded if the spectral characteristics of the directional filters used do not match the individual characteristic of the listener's HRTFs. How similar the HRTFs should be to avoid degradation in the performance is still unknown. This investigation focuses on identifying and parameterizing spectral characteristics of HRTFs that are relevant as localization cues in the mid-sagittal plane. Results suggest that parameters computed from three spectral features of simplified versions of HRTFs help explaining sound source localization in that plane. Those parameters could be used for individualizing non-individual HRTFs.
Convention Paper 8573

Tutorial 14 **Sunday, Oct. 23**
9:30 am – 11:00 am **Room 1E12**

UNTANGLING THE COMB FILTER

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

All audio experts are familiar with the comb filter. This tutorial gets precise—describing, defining, visualizing, and quantifying this ever-present, always-important property of audio. The radical alteration to frequency content caused by comb filtering is shown to be a problem that can be avoided or minimized in some instances, a negligible factor in others, and a positive production technique worthy of emphasis in yet other situations. Through a more complete understanding of this effect, we can track and mix with confidence, employing comb filtering for practical, technical, and creative sonic benefits.

Historical Event
A TRIBUTE TO WALTER SEAR
Sunday, October 23, 9:30 am – 11:00 am
Room 1E08

Moderator: **Noah Simon**

Panelists: *Steven Durr*, Acoustical Engineer
Roberta Findley, Walter's partner at the studio and filmmaker
Brian Kehew, Historian for Bob Moog Foundation
Sean Lennon
Ray Noguerra

Since his untimely passing in April 2010, Walter Sear's world-famous Sear Sound recording studio has continued to thrive as a champion of analog fidelity. With its reputation for meticulously selected and maintained equipment and a superbly trained staff, Sear Sound has attracted such clients as Paul McCartney, Wilco, and Norah Jones. Walter Sear's encyclopedic knowledge of and unbridled passion for audio quality set a Platinum standard for studios around the world. This panel will address Sear's incomparable life and legendary accomplishments from audio engineering and music composition, to his experimentation with theremins and synthesizers. Noah Simon is a Brooklyn-based engineer/producer/arranger and long-time AES member. He has worked with artists such as Eric Anderson, Scout, Bill Frisell, Susan Tedeschi, and Shawn Colvin.

Student/Career Development Event
STUDENT RECORDING CRITIQUES
Sunday, October 23, 10:00 am – 11:00 am
Room 1E06

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

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

Workshop 19 **Sunday, Oct. 23**
11:00 am – 1:00 pm **Room 1E08**

LISTEN PROFESSIONALLY OR TRAIN YOUR EAR!

Chair: **Sungyoung Kim**, Yamaha Corp., Hamamatsu, Japan

Panelists: *Jason Corey*, University of Michigan, Ann Arbor, MI, USA
Kazuhiko Kawahara, University of Kyusyu, Fukuoka, Japan
Atsushi Marui, Tokyo National Institute of Fine Art and Music, Tokyo, Japan
Sean Olive, Harman International, Northridge, CA, USA

It has been generally accepted that critical listening ability is essential for audio engineers. Compared to traditional training methods, recent training programs provide multiple trainees with fast acquisition of such listening ability through a systematic curriculum optimized for the required task. Moreover, due to the fast processing power of personal computers, an individual can access and experience these programs without hardware limitations. Considering the interests and growth of ear training in the audio and music communities, it is timely and impor-

tant to have a chance to share and discuss the opinions from the experts about necessary features and methods that assist trainees in acquiring the critical listening ability with efficiency, both for personal and group training. For this purpose, the workshop invites panelists from all around the world, who will share their in-depth experience, know-hows, and insights in ear training. While the workshop locally aims to guide the attendees of the workshop to better understanding the concept of ear training and practical tips, it also globally aims to consider the importance of "listening" in the current video-oriented society.

Broadcast/Media Streaming Session 15
Sunday, October 23 11:00 am – 12:30 pm
Room 1E10

NEW INITIATIVES IN DIGITAL AUDIO PLAYBACK AND AUTOMATION FOR RADIO

Chair: **Paul McLane**, Radio World

Panelists: *Don Backus*, Enco
Hari Samrat, Broadcast Electronics
Brad Young, Wide Orbit
Jeff Zigler, RCS

As the line blurs between traditional radio and internet streaming, automation providers are exploring new technologies to serve broadcasters. From enhanced and simplified user interfaces to the ability to control a number of streams from a single workstation automatically, many initiatives are focused on the operator. Others, like running automation on virtualized hardware and increased utilization of cloud computing are focused on the engineering side. Learn how leading providers are addressing these and other issues to assist broadcasters in serving a rapidly changing market.

Live Sound Seminar 12 Sunday, Oct. 23
11:00 am – 1:00 pm Room 1E14

CONTINUING ADVANCEMENTS IN GREEN TECHNOLOGY FOR PRO AUDIO AND CONCERT SOUND TOURING

Moderator: **Tom Bensen**, RF Productions, NY, USA and Outline, NA

Presenters: *Klas Dalbjorn*, Lab.Gruppen
Claudio Lastrucci, Powersoft
Jim Meyer, Clair Global
David W. Robb, Acoustic Dimensions
Felix Robinson, AVI-SPL

This workshop will feature industry experts from both equipment manufacturing and professional users/installers addressing and discussing the challenges that face our industry to conserve energy, reduce the carbon footprint, and contain or reduce the hidden costs of productions and installations.

Games Audio Session 13 Sunday, Oct. 23
11:00 am – 1:00 pm Room 1E13

CAREERS IN GAME AUDIO

Presenters: **Stephen Harwood Jr.**, IASIG; Dynamic Systems Music
Sam Howard-Spink, New York University
Duncan Watt, 38 Studios
Mike Worth, Play Eternal

Game audio excellence demands a coordinated team effort. In addition to sound design, composition, and production supervision, game audio requires skill sets that are rarely encountered elsewhere, including interactive audio programming and implementation. This broad array of work types provides for an equally broad range of career opportunities. Whatever your background and area of specialized expertise might be, there is room for you in this rapidly growing industry. In this session, a panel of accomplished industry veterans will share their experience and perspectives, with a focus on how to begin and develop a successful career in game audio. Audience members will take away a comprehensive understanding of the many opportunities available to audio professionals in the video game industry, as well as valuable suggestions and insights into how to land that first gig.

Product Design Session 10 Sunday, Oct. 23
11:00 am – 1:00 pm Room 1E12

AN OVERVIEW OF AUDIO SYSTEM GROUNDING AND INTERFACING

Presenter: **Bill Whitlock**, Jensen Transformers

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

Sunday, Oct. 23 11:00 am Room 1E05
Technical Committee Meeting on Networked Audio Systems

Tutorial 15 Sunday, Oct. 23
11:30 am – 1:00 pm Room 1E11

REALITY IS NOT A RECORDING / A RECORDING IS NOT REALITY

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

The former New York Times film critic, Vincent Canby, wrote "all of us have different thresholds at which we suspend disbelief, and then gladly follow fictions to conclusions that we find logical." Any recording is a "fiction," a falsity, even in its most pure form. It is the responsibility, if not the duty, of the recording engineer, and producer, to create a universe so compelling and transparent ➥

that the listener isn't aware of any manipulation. Using basic recording techniques, and standard manipulation of audio, a recording is made, giving the listener an experience that is not merely logical but better than reality. How does this occur? What techniques can be applied? How does an engineer create a convincing loudspeaker illusion that a listener will perceive as a plausible reality? Recordings will be played.

Exhibitor Seminar **Sunday, Oct. 23**
11:30 am – 1:00 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

Breaking the Rules—Surround Sound Recording Part 2

Presenter: **Morten Lindberg**

Surround Sound is a sculpture, where stereo can be described as a flat canvas.

Morten will continue to share his experience in creating the best possible surround recording while "breaking the traditional rules." Morten Lindberg is a 9-time Grammy nominated / winning balance engineer and recording producer with vocals, choirs, and strings as his core area of expertise.

Lunchtime Keynote
SKIP PIZZI
Sunday, October 23, 1:00 pm – 2:00 pm
Room 1E10

The Future of Broadcasting in a Connected World

Much industry news nowadays cites the impending death of broadcast and the rise of online as the preferred distribution method for audio and video content. Is this truly the case, or have rumors of broadcasting's demise been greatly exaggerated?

Student/Career Development Event
STUDENT DELEGATE ASSEMBLY MEETING —PART 2
Sunday, October 23, 1:00 pm – 2:30 pm
Room 1E13

At this meeting the SDA will elect a new vice chair. One vote will be cast by the designated representative from each recognized AES student section in the North & Latin American Regions. Judges' comments and awards will be presented for the Recording Competitions. Plans for future student activities at local, regional, and international levels will be summarized.

Hot Lunch
ACOUSTIC & AUDIO APPS
Sunday, October 23, 1:30 pm – 2:15 pm
Room 1E14

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

The introduction of the iPhone, iPod, and iPad has led to the generation of several acoustic and audio measurement apps for these highly portable and popular devices. Applications range from simple sound level (SPL) measurement to slightly more sophisticated 1/3 octave real time analysis to advanced FFT (Fast Fourier Transform) programs with full complex transfer function computation. This latter category of app can enable high resolution frequency response and phase measurements to be made or the system impulse response to be derived—which opens up a whole further raft of acoustic measurements to be made. Other apps make making full use of the host device's computing power enable a wide range of test signals to be generated or even highly sophisticated measures such as STI and STIPA speech intelligibility to be performed. But are these apps valid tools for the audio pro or merely gimmicks for the geeks and nerds ? Come to the hot lunch session and find out.

Exhibitor Seminar **Sunday, Oct. 23**
1:30 pm – 2:30 pm **Room 1E06**

PMC: MASTERS OF AUDIO SERIES

No Ordinary Audio Test Instrument

Presenters: **Richard Cabot, Doug Ordon, Simon Woollard**

As the BS.1770 standard for measurement attains widespread use, so does the need to test loudness meters for conformance to the latest standard.

Prism Sound's Simon Woollard and Doug Ordon, together with Qualis Audio founder Dr. Richard Cabot will be demonstrating and discussing the need for loudness measurement in both stereo and surround environments using the Prism Sound dScope and the Qualis Audio Sentinel surround sound audio monitor.

Broadcast/Media Streaming Session 17
Sunday, October 23 **2:00 pm – 5:00 pm**
Room 1E03

SOCIETY OF BROADCAST ENGINEERS EXAMS

SBE exams will take place on October 23 at 2:00 pm during the annual AES convention. Applicants are encouraged to apply before the exams by going to www.sbe.org and accessing the certification applications. You may apply on-site for the Certified Broadcast Technologist or the Certified Broadcast Networking Technologist exams. If you wish to apply for the broadcast engineer, senior engineer or specialist certifications then you would need to pre- register by October 17. If you have any questions contact Megan Clappe, certification director at mclappe@sbe.org

Session EB5 **Sunday, Oct. 23**
2:15 pm – 3:00 pm **Room 1E09**

PERCEPTION

Chair: **Brett Leonard**, McGill University, Montreal, Quebec, Canada

2:15 pm

EB5-1 Consumer Attitudes Toward Digital Audio Quality —*Ainslie Harris*, Robert Gordon University, Aberdeen, Scotland, UK

This paper builds upon an engineering brief submitted to the 130th AES Convention (Harris 2011). Where the May 2011 brief outlined initial findings from focus groups that were conducted, considering questions about preferred audio quality from the point of view of

attitudes and consumer behavior, this brief focuses on an outline for future research, discussing important questions for consideration, and proposed methodology.
Engineering Brief 37

2:30 pm

EB5-2 The Effect of Downmixing on Measured Loudness —*Scott G. Norcross, Michel C. Lavoie*, Communications Research Center, Ottawa, Ontario, Canada

ITU-R BS.1770 has become the standard for loudness measurement in broadcasting. The measurement algorithm is equally adapted to 5.1 channel audio signals as to a 2-channel downmix. Due to the manner by which the channels are summed, loudness differences can occur between the 5.1 channel signal and that of the stereo downmix. These differences are dependent on the inter-channel characteristics of the 5-channel mix. This engineering brief will outline the differences that can occur with different signals and provide data using real-world broadcast signals.
Engineering Brief 38

2:45 pm

EB5-3 Prediction of Valence and Arousal from Music—*Albertus den Brinker,*[1] *Ralph van Dinther,*[1] *J. Skowronek*[2]
[1]Philips Research Laboratories Eindhoven, Eindhoven, The Netherlands
[2]Technical University Berlin, Berlin, Germany

Mood is an important attribute of music, and knowledge on mood can be used as a basic ingredient in music recommender and retrieval systems. Moods are assumed to be dominantly determined by two dimensions: valence and arousal. An experiment was conducted to attain data for song-based ratings of valence and arousal. It is shown that subject-averaged valence and arousal can be predicted from music features by a linear model.
Engineering Brief 39

Session P24 **Sunday, Oct. 23**
2:30 pm – 4:00 pm **1E Foyer**

POSTERS: SPATIAL AUDIO PROCESSING—PART 2

2:30 pm

P24-1 A Selection Model Based on Perceptual and Statistics Characteristic for Interaural Level Difference—*Heng Wang, Ruimin Hu, Weiping Tu, Ge Gao*, Wuhan University, Wuhan, Hubei, China

In present mobile communication systems, a low bit rate audio signal is supposed to be provided with high quality. This paper researches the mechanism that exists of perceptual and statistics redundancy in binaural cues and establishes a selection model by joint perceptual and statistics characteristic of ILD. It does not quantize the values of ILD where the frequency bands can't easily be perceived by human ears according to the selection model. Experimental results showed that this method can bring down the parametric bit rate by about 15% compared with parametric stereo, while maintaining the subjective sound quality.
Convention Paper 8574
[Paper not presented but is available for purchase]

2:30 pm

P24-2 Perceptual Evaluation of a Spatial Audio Algorithm Based on Wave Field Synthesis Using a Reduced Number of Loudspeakers— *Frank Melchior,*[1] *Udo Heusinger,*[1] *Judith Liebetrau*[2]
[1]IOSONO GmbH, Erfurt, Germany
[2]Technical University Ilmenau, Ilmenau, Germany

With 3-D picture being the driving force of today's motion picture production, there is a growing need for adequate audio solutions, e.g., spatial audio algorithms for reproduction with flexible loudspeaker setups. While these reproduction systems will have to fulfill high quality demands, the amount of loudspeakers needed should be kept as low as possible to optimize commercial aspects. One suitable algorithm from a quality point of view is wave field synthesis (WFS), which, however, requires a huge amount of speakers if implemented as stated in literature. This paper presents the results of a perceptual evaluation of a new algorithm based on WFS. A listening experiment compared state-of-the-art WFS, the new algorithm, and Vector Base Amplitude Panning regarding their perceived localization and coloration.
Convention Paper 8575

2:30 pm

P24-3 Design and Evaluation of an Interactive Simulated Reverberant Environment—*Alan C. Johnson, Kevin Salchert, Andreas Sprotte-Hansen*, New York University, New York, NY, USA

There are many existing approaches to the challenge of simulating a reverberant field. Most of these methods are designed to operate on a signal that has been recorded in a relatively anechoic environment and seek to add in the simulated reverberation of a chosen space. This paper describes a low-cost, scalable approach for directly converting an acoustically dry space into a reverberant space of a larger size, with a number of configurable parameters. This is accomplished by harnessing the mutual feedback among microphones and loudspeakers arranged in the space. The result is a simple, tunable, and interactive system for creating a convincing reverberant environment. Several novel applications of such a system are also discussed.
Convention Paper 8576

2:30 pm

P24-4 Multi-Touch Room Expansion Controller for Real-Time Acoustic Gestures—*Andrew Madden, Pia Blumental, Areti Andreopoulou, Braxton Boren, Shengfeng Hu, Zhengshan Shi, Agnieszka Roginska*, New York University, New York, NY, USA

This paper describes an application that pro- ➥

vides real-time high accuracy room acoustics simulation. Using a multi-touch interface, the user can easily manipulate the dimensions of a virtual space while hearing the room's acoustics change in real-time. Such an interface enables a more fluid and intuitive control of our application, which better lends itself to expressive artistic gestures for use in such activities as sound design, performance, and education. The system relies on high accuracy room impulse responses from CATT-Acoustic and real-time audio processing through Max/MSP and provides holistic control of a spatial environment rather than applying generic reverberation via individual acoustic parameters (i.e., early reflections, RT60, etc.). Such an interface has the capability to create a more realistic effect without compromising flexibility of use.
Convention Paper 8577

2:30 pm

P24-5 Adaptive Crosstalk Cancellation Using Common Acoustical Pole and Zero (CAPZ) Model—Common Pole Estimation—*Hanwook Chung,[1] Sang Bae Chon,[2] Nara Hahn,[1] Koeng-Mo Sung[1]*
[1]Seoul National University, Seoul, Korea
[2]Samsung Electronics Co. Ltd., Suwon, Gyeonggido, Korea

In this paper we introduce adaptive crosstalk cancellation using a common acoustical pole and zero (CAPZ) model for a head-related transfer function (HRTF). As the CAPZ model for HRTF provides an interpretation of the HRTF wherein zeros describe the spatial difference caused by the acoustical propagation path and common poles describe the characteristics of the human auditory system, we designed the proposed model to follow the zero components of the CAPZ model. By estimating common poles and simulations, we verified that the proposed model exhibits enhanced performance compared with a conventional finite impulse response model of HRTF.
Convention Paper 8578

Tutorial 16 — Sunday, Oct. 23
2:30 pm – 4:30 pm — Room 1E08

DRUM AND PERCUSSION PROGRAMMING

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

Drum programming has been evolving at the heart of many studio productions for some 30 years. Over this period, technological opportunities for enhanced creativity have multiplied in numerous directions, from sequenced MIDI one-shots to sample loops, and from DAW cut and stretch techniques to deterministic beat-slicer plug-ins, etc. The palette of sounds available today ranges from ever more realistic to ever more synthetic/exotic. This tutorial will embrace all of these techniques and more and include numerous live demonstrations. Although introducing all key concepts from scratch, its range and hybridization should provide inspiration even for experienced practitioners, leading up to the state-of-the-art. A number of genres will be covered from the pseudo-realism of jazz and funk, to the exploitation of synthetic textures in "intelligent dance music."

Master Class 4 — Sunday, Oct. 23
2:30 pm – 4:30 pm — Room 1E15/16

SUPERSTAR SESSIONS

Presenter: **Tony Visconti**

Tony Visconti rose from the streets of Brooklyn to become a key architect of the sound of British Rock via studio work with such legends as David Bowie, T-Rex, Thin Lizzy, The Moody Blues, Badfinger, and Morrissey. Beyond his production and mixing skills, Visconti is an accomplished composer, musician, and arranger. This Master Class will present techniques and experiences culled from a lifetime of collaboration with some of the world's most celebrated artists, as well as his on-going work on new releases such as Future Is Medieval from The Kaiser Chiefs.

Broadcast/Media Streaming Session 16
Sunday, October 23 — 2:30 pm – 4:00 pm
Room 1E10

MEDIA FILE MANAGEMENT: STORAGE, BACKUP, AND RETRIEVAL OF YOUR ASSETS

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

Panelists: *Jim Boas*, RorkeData–An Avnet Company
Tim Claman, Avid
Michael Gitig, Gobbler

How's your media? Is it safe? Is it accessible?
Recording and production are rapidly moving into a file-based workflow with all the project materials—audio files, edits, plug-ins, automation, and video files—existing on spinning disks. During the project it's necessary to keep track of the elements and after completion it's vital to preserve all the work. But a project rarely remains untouched after submission so there is an increased need for media access and identification after the back-up to grab and re-purpose elements for a new job or revision. Our panel will discuss technologies including fiber, Ethernet, and cloud-based storage and retrieval for addressing the need to manage the exploding volume of media assets generated every day.

Live Sound Seminar 13 — Sunday, Oct. 23
2:30 pm – 4:30 pm — Room 1E14

MULTITRACK RECORDING FOR THE LIVE SOUND ENGINEER

Chair: **Sam Berkow**, SIA Acoutics

Panelists: *Ted Leamy*, ProMedia/UltraSound
Robert Scovil, Avid
Adam Shulman

Multitrack recording of performances is now commonplace for many artists, whether for internal reference and archiving or for potential release. This poses a new set of parameters on both the FOH and monitor mix engineers which at times are diametrically opposed. The best house sound does not not necessarily produce the best recordings and sometimes results in the FOH engineer to mix

two shows simultaneously; one for live, a second for the recording. This panel will discuss the issues confronted by mix engineers and the best techniques and practices to insure both an optimal live show and recording.

Product Design Session 11 **Sunday, Oct. 23**
2:30 pm – 4:00 pm **Room 1E12**

NETWORKED AUDIO GIZMOS—ETHERNET AUDIO TODAY AND TOMORROW

Chair: **Steve Macatee**, Rane Corporation

Panelists: *Bradford Benn*, Crown International
Hardy Martin, Innovative Electronic Designs, Inc.
Chris Pane, Lab X Technologies (The Ethernet AVB Camp)

What problems are integrators faced with when setting up many different Ethernet devices on the same network? Is your design approach causing the problem or a victim of someone else's? What obstacles deter configuration software and devices from establishing and maintaining communications on simple and complex Ethernet networks? The pros and cons of UDP vs TCP protocols for control and metering messages. How to get through routers, VPNs, and blocked ports. Is your design approach making the IT department your friend or foe? Can you make it easier for customers to get through their firewall(s)? What challenges do older products encounter on ever-changing, modern networks? Can one product's communications method interfere with already established device communications? Will Ethernet AVB finally solve all problems known to man?

Historical Event
AMERICAN CLASSICAL RECORDING— FROM 1 MICROPHONE TO 24 TRACKS
Sunday, October 23, 2:30 pm – 4:00 pm
Room 1E11

Presenter: **Tom Fine**

This Historical Event is an exploration of the history of recording techniques and equipment from the 1954 heyday of monophonic full-range high fidelity. Beginning with a single microphone, evolving to early stereo's "golden age of recording," and continues with the increasing complexity of the late 1960s and early '70s, when Columbia and EMI won Grammys for classical recordings made with as many as 32 mics and 24 tracks. This event is highlighted by a rare, comparative listening session featuring Grammy-winning recordings of Ravel's Daphnis et Chloe recorded over three decades. Thomas Fine is a member of ARSC and owner of an analog-to-digital audio transfer studio. With a primary base of archival and education/ institutional clients, Fine specializes in transferring magnetic and grooved-disc media to high resolution digital formats. He is an avid collector of music recordings and student of recording-industry history.

Student/Career Development Event
THE $5.1 LOUDSPEAKER— SOUND FROM A SHOEBOX
Sunday, October 23, 2:30 pm – 3:30 pm
Room 1E13

Moderator: **Dan Domme**, Penn State, Graduate Program in Acoustics, PA, USA

Moving-coil loudspeakers are an exciting means to engage others in audio. Because of their widespread use, loudspeakers are a familiar system combining many of the fundamental physical principles of audio: electricity, magnetism, mechanics, and acoustics. This workshop presents a loudspeaker-enclosure system that can be easily built from scratch. The system is mostly constructed from common household supplies, making it low-cost and accessible to a wide audience.

Session P25 **Sunday, Oct. 23**
3:00 pm – 5:30 pm **Room 1E09**

AUDITORY PERCEPTION

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

3:00 pm

P25-1 The Impact of Producers' Comments and Musicians' Self-Evaluation on Performance during Recording Sessions—*Amandine Pras, Catherine Guastavino*, McGill University, Montreal, Quebec, Canada

When recording in the studio, musicians repeat the same musical composition over and over again without the presence of an audience. Furthermore, recording technologies transform the musical performance that musicians hear in the studio. We conducted a field experiment to investigate whether record producers' comments and musicians' self-evaluation helped musicians improve from one take to another during recording sessions. Twenty-five jazz players, grouped into five ensembles, participated in recording sessions with four record producers. Two types of feedback between takes were varied independently: with or without comments from a record producer and with or without musicians' self-evaluation after listening to the takes in the control room. Our results show that both external comments and self-evaluation give the ensemble a common ground but also make musicians too self-conscious.
Convention Paper 8579

3:30 pm

P25-5 Some New Evidence that Teenagers May Prefer Accurate Sound Reproduction—*Sean Olive*, Harman International Industries, Inc., Northridge, CA, USA

A group of 18 high school students with no prior listening experience participated in two separate controlled listening tests that measured their preferences between music reproduced in (1) MP3 and lossless CD-quality file formats, and (2) music reproduced through four different consumer loudspeakers. Overall, the teenagers preferred the sound quality of the CD-quality file format, and the most accurate, neutral loudspeaker. Together, these tests provide some new evidence that teenagers can discern and appreciate a better quality of reproduced sound when given the opportunity to directly compare it against lower quality options.
Convention Paper 8583

4:00 pm

P25-2 The Influence of Camera Focal Length in the Direct-to-Reverb Ratio Suitability and Its Effect in the Perception of Distance for a Motion Picture—*Luiz Fernando Kruszielski, Toru Kamekawa, Atsushi Marui*, Tokyo University of the Arts, Tokyo, Japan

In order to study the possible influence of a camera focal length in the auditory distance perception, two experiments were conducted. In the first experiment, participants were asked to adjust the amount of reverb according to the presented visual image, which had different focal lenses distances, and therefore different backgrounds. In the second experiment, participants were asked to rate the egocentric sense of distance to the sound source and the suitability of the sound for the visual image in a pairwise comparison. The results have shown that the overall sense of distance is mainly dependent on the focal length; however, if the image foreground object has the same size, the focal length can alter the perception of sound distance.
Convention Paper 8580

4:30 pm

P25-3 Automatic Soundscape Classification via Comparative Psychometrics and Machine Learning—*Krithika Rajagopal*,[1,2] *Phil Minnick*,[1] *Colby Leider*[1]
[1]University of Miami, Coral Gables, FL, USA
[2]Audio Precision, Beaverton, OR, USA

Computational acoustical ecology is a relatively new field in which long-term environmental recordings are mined for meaningful data. Humans quite naturally and automatically associate environmental sounds with emotions and can easily identify the components of a soundscape. However, equipping a computer to accurately and automatically rate unknown environmental recordings along subjective psychoacoustic dimensions, let alone report the environment (e.g., beach, barnyard, home kitchen, research lab, etc.) in which the environmental recordings were made with a high degree of accuracy is quite difficult. We present here a robust algorithm for automatic soundscape classification in which both psychometric data and computed audio features are compared and used to train a Naive Bayesian classifier. An algorithm for classifying the type of soundscape across different categories was developed. In a pilot test, automatic classification accuracy of 88% was achieved on 20 soundscapes, and the classifier was able to outperform human ratings in some tests. In a second test classification accuracy of 95% was achieved on 30 soundscapes.
Convention Paper 8581

5:00 pm

P25-4 Effect of Whole-Body Vibration on Speech. Part II: Effect on Intelligibility—*Durand Begault*, NASA Ames Research Center, Mofett Field, CA, USA

The effect on speech intelligibility was measured for speech where talkers reading Diagnostic Rhyme Test material were exposed to 0.7 g whole body vibration to simulate space vehicle launch. Across all talkers, the effect of vibration was to degrade the percentage of correctly transcribed words from 83% to 74%. The magnitude of the effect of vibration on speech communication varies between individuals, for both talkers and listeners. A "worst case" scenario for intelligibility would be the most "sensitive" listener hearing the most "sensitive" talker; one subject's intelligibility was reduced by 26% (97% to 71%) for one of the talkers.
Convention Paper 8582
[Paper presented by Richard King]

Tutorial 17 — Sunday, Oct. 23
4:00 pm – 5:30 pm — Room 1E11

MASTERING FOR VINYL

Presenter: **Scott Hull**, Masterdisk, New York, NY, USA

Scott Hull has been mastering for vinyl and digital for 28 years, seeing formats come and go, and come back again. In the last few years there has been a renewed interest in producing vinyl among modern artists. What has to be considered when you mix/master your music for vinyl? Scott will dig deep into the quality control issues and discuss several sure ways to sound great on your first pressing.

Topics will include: Why contemporary CD mastering techniques do not produce the best sounding vinyl records. Long Sides—the relationship between volume, duration, and quality. The Turntable—what does yours sound like? The quality control process: mixing— mastering—plating—pressing. The realities of the current vinyl market. Modern trends in record making.

Tutorial 18 — Sunday, Oct. 23
4:00 pm – 5:30 pm — Room 1E10

AUDIO METADATA: UNDERSTANDING AES57 AND AES60

Presenter: **David Ackerman**

AES will publish in the next couple of days AES57 and AES60, two metadata standards pertinent to the archiving community. This tutorial will explain the proper use of these documents in the archiving of audio collections.