CONTENTS
117TH TECHNICAL MEETINGS AND PROFESSIONAL EVENTS

Ron Streicher

Welcome to the 117th AES Convention, and welcome back to San Francisco, a city of varied wonders and endless excitement. To fulfill the challenge of this city, the convention planning committee has assembled a roster of events equally varied and exciting, including many features new to this convention —two tracks of technical papers plus poster sessions; two tracks of tutorial sessions; two tracks of exhibitor seminars; workshops; special events and technical tours; a full historical program and exhibit; the Student Delegate Assembly and Education Forum events; and, of course, the business of the Society, the meetings of the Technical Council and Standards Committee Working Groups.

I would like to draw your particular attention to a few of the highlighted Special Events, available to all registered convention attendees: On Thursday at noon will be the Opening Ceremonies and Awards Presentation, featuring keynote speaker Ron Fair; in the evening, the AES Mixer Party at the Moscone Center will be followed by Graham Blyth's Organ Recital at Grace Cathedral. On Friday, the Platinum Producers and Engineers panels. On Saturday, An Afternoon with Bob Moog, and then the Heyser Lecture presented by Walter Murch. On Sunday, the "Road Warriors Live Sound" panel. Complete details of these and all sessions can be found in this Program Book.

If this doesn't already fill your appointment book to overflowing, the Exhibit Floor will provide you an abundant opportunity to view the latest in audio technology and, even more important, to meet and talk with the people behind the front panels and knobs.

So, enjoy your visit to San Francisco and the AES 117th Convention.

… And, oh yes, Happy Halloween!

RON STREICHER,
PRESIDENT AES

John Strawn

The AES 117th Convention will open with a bang, featuring Ron Fair's keynote address. Not only is Ron a world-renowned producer and an engaging speaker, he has some eyebrow-raising insights into the state of music business today and the likely future facing the music business.

Ever since the "talkies" were invented, sound has been a major component of the film industry. This convention offers a superb selection of technical tours, highlighted by a visit to Lucasfilm's Skywalker Sound in Marin County, surely the hottest ticket at the convention. We are also lucky to have Walter Murch, the well-known editor and sound mixer for films, presenting the Heyser lecture, including a performance of Edison's first sound film from 1894.

One of the biggest trends in audio these days is the use of more than two channels. The convention opens with a special symposium on live surround, held here for the first time on the West Coast. There are many events using surround, resulting in three separate rooms equipped with surround systems (we thank Genelec and Meyer for generously providing equipment).

Audio is an important element in computer and video games, and the quality and variety of audio in games continues to develop dramatically. At the 117th Convention we will present gAmEShowcase, giving convention attendees the chance to hear game audio from the leading manufacturers.

Some believe that the AES divides technical papers into two levels of quality, and assigns lower-quality papers to the poster sessions. On the contrary, all papers pass the same review process. Papers with audiovisual demonstrations suitable for a large audience are best in the lecture format. Highly detailed papers are best in poster sessions, to permit greater interaction between author and audience. At the 117th convention we have allotted more time to poster papers, and we will open each poster session with an introduction of the authors. We have spent considerable time and effort designing suitable boards for mounting poster presentations to highlight the important work being presented in this format.

In preparing workshops for the 117th convention, we have emphasized spontaneous interaction between the audience and the presenters; so come and listen, learn, interact, and be spontaneous.

Expanding on recent AES conventions, we have included exhibitor seminars and a full range of tutorials, touching on the most important aspects of audio in two tracks across three days. This is a rare opportunity for attendees to gain knowledge in all areas of audio, and to learn it straight from the masters.

Streaming audio is not limited to internet downloads of MP3 files. Thanks to efforts spanning both coasts of North America, we will be treated to a live streaming feed from Montreal to San Francisco. The AES Technical Committee on Network Audio Systems will demonstrate streaming of six channels of DSD audio, transmitting a live jazz performance at McGill University in Montreal, Canada, in real time, to the University of California San Francisco's Genentech Hall.

Historically, the AES has seen the introduction of many landmark inventions, including the music synthesizer. Bob Moog will entertain us with his insights on Saturday afternoon. There have been significant developments in music, music technology, and entrepeneurship on the West Coast. Our Historical Committee has gathered some of the West Coast pioneers in electronic music for a reunion on Friday afternoon.On Sunday Peter Gotcher, co-founder of DigiDesign, will talk about what it's like to grow a company.

Although I have been honored to serve as chair of this convention, the real work has been done by an extraordinarily gifted and dedicated committee of volunteers, listed on the inside front cover of this program. If you see them (wearing blue ID tags) during the convention, be sure to thank them for their hard work.

As luck would have it, this is Halloween weekend, one of the biggest holidays every year in San Francisco. Above all, enjoy yourself while you are here!

JOHN STRAWN
CONVENTION CHAIR

MOSCONE CENTER
747 Howard Street
San Francisco, CA, USA
Telephone: +1 415 974 4000
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REGISTRATION

Registration Desk Hours:

Wednesday, October 27	4:00 pm – 7:00 pm
Thursday, October 28	8:00 am – 6:00 pm
Friday, October 29	8:00 am – 6:00 pm
Saturday, October 30	8:00 am – 6:00 pm
Sunday, October 31	8:00 am – 4:00 pm

REGISTRATION FEES:

AES MEMBERS (all grades)

Full Program (technical sessions, workshops, seminars, special events, and exhibits):

• AES Honorary/Life Members	No Fee
• AES Members & Associates	$330
• AES Student Members	$110

Full Program plus **Symposium**

• AES Honorary/Life Members	$110
• AES Members & Associates	$440
• AES Student Members	$190

Symposium Only (includes exhibition)**:**

• AES Members & Associates	$150
• AES Student Members	$100

Exhibits Only (valid for 4 days)**:**

• AES Members & Associates	$ 50
• AES Student Members	$ 50

NONMEMBERS

• Full Program	$410
• Full Program plus Symposium	$525
• Student (with I.D.), Full Program	$155
• Student, Full Program & Symposium	$235

Symposium Only (includes exhibition)**:**

• Nonmembers	$175
• Student Nonmembers	$120

Exhibits Only (valid for 4 days)**:**

• Nonmembers	$ 60
• Nonmember Students	$ 60

INDIVIDUAL TICKETS:

Individual tickets for single paper sessions, tutorial seminars, and workshops may be purchased at the Special Events Desk. You must have purchased a 4-day Exhibits Only pass to obtain individual event tickets.

PAYMENTS

The AES will accept the following payments: cash or credit cards (Eurocard/Mastercard/Visa/American Express) are accepted for on-site registration.

All **badges** have access to Special Events, meetings of Technical Committees, Standards Committees, the Historical Committee, and the Education Fair.

TECHNICAL PAPERS, WORKSHOPS, SEMINARS, AND SPECIAL EVENTS HOURS:

Thursday, October 28	9:00 am – 6:30 pm
Friday, October 29	9:00 am – 6:30 pm
Saturday, October 30	9:00 am – 6:30 pm
Sunday, October 31	9:00 am – 4:00 pm

These times are general; please refer to specific sections in this booklet and/or the Convention Planner for more specific information.

EXHIBITS

The **Exhibit Booths** are located on the Exhibit Hall Level. **Demonstration Rooms** are on the East Mezzanine. Please refer to the *117th Convention Exhibitor Directory* for a complete list of exhibitors and their locations.

Exhibit Hours

Thursday, October 28	12:00 noon – 6:00 pm
Friday, October 29	10:00 am – 6:00 pm
Saturday, October 30	10:00 am – 6:00 pm
Sunday, October 31	10:00 am – 4:00 pm

MEMBERSHIP

AES Membership Services are located in the Registration Area in the South Lobby. Why not become a member of the Audio Engineering Society? The difference between the full program registration fee for nonmembers versus AES members equals the AES membership fee for the year and includes subscription to the 10-issue per year *Journal* (*JAES*) and lower rates for AES Publications. If you wish to become an AES member, please pay the nonmember registration fee and contact AES Membership in the publications area. AES members who want to purchase AES lapel pins or membership certificates may do so at the publications area.

AES PUBLICATIONS

Convention papers and other AES publications, CD-ROMs, and CDs are on sale at the AES Publications Shop. Printed copies of any previous convention paper or *JAES* article from the AES Electronic Library may be ordered at the AES Publications Shop. Hours are the same as the Registration Desk. Please note: These are special convention prices for the items listed below. Regular prices will apply after the convention.

117th CONVENTION PAPERS

Single Copy	$ 4
Complete Set (single copies of 145 papers)	$100
Complete Set on CD-ROM (single copies of 145 papers)	$100
Complete Set and CD-ROM (single copies of 145 papers plus disk)	$150

PRESS

Press attendees are invited to register directly at the Press Registration Desk, located in the Registration area. Press passes are delivered only upon presentation of press credentials (press card, sample of publication, letter from editor).

PRESS CENTER

Access to the Press Center is reserved exclusively for journalists and publication staff. Exhibitors are welcome to deliver press-kits and information for the press but are not permitted to collect any literature from other exhibitors.

Press Center Opening Hours

Thursday, October 28	9:00 am – 6:30 pm
Friday, October 29	9:00 am – 6:30 pm
Saturday, October 30	9:00 am – 6:30 pm
Sunday, October 31	9:00 am – 4:00 pm

AES DAILY

The AES official Daily "Convention News" is available to all convention-goers. Three issues are released, the third one serving days 3 and 4 of the convention.

BUSINESS CENTER

The Moscone Center is equipped with several pay phones using phone cards and credit cards. A Business Center provided with telephones, copy and fax machines, scanners, and printers, as well as Internet connections (e-mail) is available at the Service Center on the Exhibition Level.

SHUTTLE BUS SERVICE

Complimentary round-trip shuttle bus service will be provided between the official AES 117th Convention Hotels and Moscone Center, with the exception of the Marriott, Argent, and W hotels, due to their proximity to Moscone Center.

PUBLIC TRANSPORTATION INFORMATION

Located in the urban heart of San Francisco's downtown district, The Moscone Center is easily accessible by transit. BART offers fast, convenient regional service and MUNI Metro provides light rail service throughout the City into the Powell and Embarcadero stations. Caltrain offers a frequent schedule into the nearby station at 4th and Townsend Streets. All of these stations are within a few minutes' walk of The Moscone Center.

Although parking is available at various locations throughout the south of market area, this remains a dense and highly active urban district. Transit use minimizes traffic congestion and eases the demand for parking at peak periods.

Arrival via BART or MUNI Metro: Disembark at the Powell Street station. Exit to 4th and Market Streets. Turn right on 4th Street. Walk two blocks south to Howard Street and turn left at the signal. The Moscone Center is located on Howard between 3rd and 4th Streets, with Moscone South on the right and Moscone North on the left. For fare and schedule information, call the BART Hotline at 415.989.2278.

Arrival via Caltrain: Exiting the station, cross 4th Street and catch the MUNI #15, #30, or #45 bus line and get off at 3rd and Howard. Moscone South will be on the left and Moscone North on the right. Or to walk, turn left and walk up 4th Street five blocks to Howard. Turn right on Howard. Moscone South is on the right and Moscone North on the left. Call the CalTrain Hotline at 800.660.4287 for fare and schedule information.

AES CONVENTION HOTELS

San Francisco Marriott (Hdqtrs.) +1 415 896 1600
The Argent Hotel +1 415 974 6400
Grand Hyatt San Francisco +1 415 398 1234
Handlery Union Square Hotel +1 415 781 7800
Hilton San Francisco +1 415 771 1400
King George Hotel +1 415 781 5050
Hotel Nikko San Francisco +1 415 394 1111
The Pickwick Hotel +1 415 421 7500
Powell Hotel +1 415 398 3200
Renaissance Parc 55 Hotel +1 415 392 8000
W San Francisco +1 415 777 5300

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

Opening Ceremonies

AWARDS PRESENTATION AND KEYNOTE ADDRESS
Thursday, October 28, 11:30 am – 1:00 pm
Room 305/307

Opening Remarks:

- Executive Director Roger Furness
- President Ron Streicher
- Convention Chair John Strawn

Program:

- AES Awards Presentation
- Keynote Address by Ron Fair, President, A & M Records

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

Keynote Address
Ron Fair is a six-time Grammy®-Nominee, president of A&M Records, producer, arranger, musician, and recording engineer. His credits stretch from collaborating with legendary composer/producer Bill Conti on "Gonna Fly Now," the #1 hit theme from *Rocky*, to discovering five-time Grammy® Award-Winner Christina Aguilera and serving as Executive Producer for the superstar's ten-times platinum album *Stripped* and worldwide #1 Grammy®-Winning "Beautiful." He has held positions at RCA as A&R Manager, at Chrysalis Records as Senior Director of A&R, and as Senior VP of A&R at EMI Records, where his credits include the six-times platinum *Pretty Woman* soundtrack. Rejoining RCA Records in 1993 as Senior VP of A&R, Fair was Executive Producer of the five-times platinum *Reality Bites* soundtrack CD, which included Lisa Loeb's #1 hit "Stay." As president of A&M Records since 1999, Ron Fair oversees an artist roster that includes Sting, Sheryl Crow, The Black Eyed Peas, and Vanessa Carlton. He is a 25-year member of NARAS, serving as National Trustee and member of the Board of Governors, where he remains an active committee member.

In his keynote address, Fair will explore the conundrum of multilevel problems and circumstances that have resulted in the disgorgement of over forty percent of the industry, with an eye to future conditions and possible remedies.

Technical Tours

All technical tours have a limited capacity. Tickets will be allocated on a first-come-first served basis. To participate, please sign up at the AES Tours Desk.

TECHNICAL TOUR 1
KQED—Public Broadcasting Radio and TV Station
Thursday, October 28, 9:00 am – 1:00 pm

KQED is San Francisco's primary public broadcasting outlet. First taking to the airwaves 50 years ago this year to broadcast "educational tv," KQED television and radio have always been flagships of the national public broadcasting networks. Still leading the way, KQED first began digital broadcasting in 2000. This tour will encompass both the radio and television facilities. *Ticket Price: $20.*

TECHNICAL TOUR 2
Electronic Arts—Computer Games
Thursday, October 28, 1:00 pm – 5:00 pm

Electronic Arts is the world's largest computer game developer. With nearly $4 billion in revenues last year and offices around the world, EA sets the pace for the industry. Its Redwood City headquarters contains both the corporate offices and one of the larger development studios, which will be the destination of the tour. Production suites, recording studios, and other facilities will be shown and discussed. *Ticket Price: $30.*

TECHNICAL TOUR 3
Meyer Sound Laboratories
Friday, October 29, 9:00 am – 1:00 pm

Meyer Sound Laboratories is arguably the audio industry's highest end manufacturer of professional loudspeakers for sound reinforcement. An independent, family-owned business since its founding 25 years ago, Meyer Sound's manufacturing is unusual in the extreme level of detail and quality control applied, right down to manufacturing its own drivers from the paper cone on up. *Ticket Price: $20.*

TECHNICAL TOUR 4
Fantasy Studios/Saul Zaentz Film Center
Friday, October 29, 1:00 pm – 5:00 pm

Fantasy Studios/Saul Zaentz Film Center is one of the Bay Area's largest recording and postproduction facilities. It is the site where many famous artists have recorded classic albums and where Oscar-winning films like *Amadeus* and the *English Patient* were produced. The tour will encompass both the recording studio complex and the film studio. *Ticket Price: $20.*

TECHNICAL TOUR 5
Ex'Pression College for Digital Arts and

Center for New Music and Audio Technologies (CNMAT), University of California, Berkeley
Saturday, October 30, 9:00 am – 1:00 pm

Ex'Pression College is a private school teaching the cutting edge of current entertainment technology, including audio recording, animation, and sound design for games. The expansive facility is a model for how modern production can be taught. CNMAT is Berkeley's center for computer music and audio research. Headed by composer/researcher David Wessel, who guided Paris' IRCAM during a critical period of growth, CNMAT is pioneering the ground many of us will be treading in years to come. *Ticket Price: $20.*

TECHNICAL TOUR 6
Dolby Laboratories: Exploring Digital Cinema Sound
Saturday, October 30, 11:00 am – 1:00 pm

Dolby Laboratories creates technologies that intensify and enhance the entertainment experience. Visit Dolby's headquarters to learn more about industry developments in the next cinematic frontier—digital cinema. The new digital cinema format allows for enhanced audio capabilities beyond those of current 35 mm film, including additional channels, mix-down parameters, and intelligent loudness adjustments. Learn details of the new format and experience digital cinema in the state-of-the-art Dolby screening room. *Ticket Price: $20.*

TECHNICAL TOUR 7
Skywalker Sound
Saturday, October 30, 12:30 pm – 6:00 pm

Skywalker Sound is the world-famous audio postproduction facility built by filmmaker George Lucas on his Skywalker Ranch. Always at the forward edge of production techniques and technology, Skywalker Sound has been home to more Oscar-winning soundtracks than you can shake a statue at. The facility includes the 300-seat Stag Theatre and the magnificent Scoring Stage, one of the world's best recording rooms. *Ticket Price: $40.*

TECHNICAL TOUR 8
Audium
Saturday, October 30, 1:00 pm – 3:30 pm

Audium is a purpose-built multichannel performance space in San Francisco. Now in its 38th year of operation, Audium was designed as a place where composer Stan Shaff could perform his compositions in a facility loaded with more than 300 loudspeakers to which he guides the music improvisatorially in each performance. For those who think live performance of multichannel sound is new, Mr. Shaff will show you how it's been done for decades. *Ticket Price: $20.*

Historical Program

Friday, October 29 2:30 pm – 3:30 pm
Room 306
CLASSIC MICROPHONES FROM THE GOLDEN AGE OF RADIO

Presenter: **Barry Brose**, Highland Laboratories, San Francisco, CA, USA

Brose will demonstrate some of the old classic microphones —how they were built, how they look, how they were used, and how they sound. The demonstration includes carbon, condenser, dynamic, and velocity microphones. He will also discuss the invention of the cardioid microphone.

Friday, October 29 4:30 pm – 6:00 pm Panel
6:00 pm Technology Demos
Recombinant Media Labs, 763 Brannan St. (six blocks from Moscone Convention Center); show your convention badge for entry.
BAY AREA ELECTRONIC MUSIC PIONEERS: INNOVATIVE CONTRIBUTIONS TO MUSIC SYNTHESIS, AUDIO SIGNAL PROCESSING, AND ELECTRONIC MUSIC

Panelists: **Don Buchla**, Buchla and Associates, Berkeley, CA, USA
John Chowning, CCRMA at Stanford University, Stanford, CA, USA
Roger Linn, Roger Linn Design, Berkeley, CA, USA
Max Mathews, CCRMA at Stanford University, Stanford, CA, USA
Tom Oberheim, Marion Systems
Dave Rossum, E-mu Systems Inc., Scotts Valley, CA, USA
Dave Smith, Dave Smith Instruments, St. Helena, CA, USA

The radiant evolution of electronic music can be attributed to a number of significant developments that took place along the California coastline from the 1960s to the present day. Innovations from the forefathers of electronic music's wild west became the foundation for modern audio signal processing and electro-musical devices. From the tech benches of these pioneers came distinctive discoveries and instrumental technologies including alternative music controllers, audio generating algorithms, computer-controlled music performance, digital audio sampling, musical event sequencing, musical synthesizer design, speech processing, and synthesized rhythm devices. A moderated panel discussion with our distinguished pioneers will be preceded by a short video presentation highlighting their innovations and contributions to electronic music. On-site demonstrations of various electronic music technologies will follow a Q&A session.

Saturday, October 30 12:30 pm – 1:30 pm
Room 306

THE BIRTH OF RADIO BROADCASTING: CHARLES HERROLD AND THE FIRST RADIO STATION

Presenter: **Mike Adams**, Chair Department of TV, Radio, Film & Theater, San Jose State University, San Jose, CA, USA

In 1909 an obscure San Jose inventor named Charles Herrold began building a radiotelephone. For his microphone he used six carbon buttons in a telephone-like handset. The microphone was water-cooled because in order to broadcast, it was wired in series with a DC arc transmitter, modulating the arc current. In the January 1910 issue of the Electro-Importing Company catalogue, this notarized statement by Herrold appeared in an ad for radio parts: "We have been giving wireless telephone concerts to amateur men in the Santa Clara Valley." This document is a "smoking gun," proving Herrold was the first broadcaster. He had invented a radio station and was broadcasting entertainment and information to a small audience ten years before licensed broadcasting and the first use of the word "radio."

Saturday, October 30 5:00 pm – 6:00 pm
Room 305/307

ENIGMA AND THE "ULTRA SECRET": CRACKING WWII CIPHERS AND THE COMMON ORIGINS OF COMPUTERS, DIGITAL AUDIO, INTERNET SECURITY, AND DIGITAL RIGHTS MANAGEMENT

Presenter: **Jon Paul**, Curator of the Crypto-Museum

The German Enigma Machine is a fascinating conjunction of cryptology, WWII history, and the foundation of modern computing and DSPs. The cracking of Enigma was pivotal to the Allied victory in both theaters of War. The quest to break the Enigma and other Axis cipher machines at Bletchley Park England led directly to today's digital computer and digital audio technology. In this presentation we trace the history and cracking of several WWII cipher machines, demonstrate the operation of Enigma, and highlight their connections to modern digital audio technology..

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

Sunday, October 31 12:00 noon – 1:00 pm
Room 306
MY EARLY EXPERIENCES FOUNDING, FUNDING, GROWING, AND SELLING AUDIO COMPANIES

Presenter: **Peter Gotcher**, Independent Venture Capitalist

Introduction: **Roger Linn**, Roger Linn Design

Over the past 20 years, Peter Gotcher has been involved as a founder, venture capital investor, board member, business advisor, etc., for a number of audio-related companies. Peter co-founded Digidesign and was Chairman and CEO for thirteen years, through its IPO and subsequent acquisition by Avid. Peter will profile several success stories, including Digidesign, while providing some overall advice on the issues faced by audio entrepreneurs such as starting companies, obtaining funding, managing growth, and achieving liquidity.

Historical Committee
The Historical Committee will hold its meeting in the Historical Room. Please see posted signs for date, place, and time.

Special Events

AES SYMPOSIUM
SURROUND LIVE!—MORE SURROUND!
Wednesday, October 27, 10:00 am – 6:00 pm
Julia Morgan Ballroom
Merchants Exchange Building
645 California Street

Preconvention Special Event; additional fee applies

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

Panelists: *David Claringbold*, The Sydney Opera, Sydney, Australia
Kurt Graffy, Arup Acoustics
Dave Haydon, Outboard Electronics
Randy Hoffner, ABC Television
Mark Hood, Echo Park Studios, Bloomington, IN
Jonathan Lane-Talaskey, Jay Pritzker Pavilion, Millennium Park
Riccardo Mazza, School of High Musical Specialization, Saluzzo, Italy
Bruce Olson
Bobby Owsinski, Surround Associates
Steve Shull, Acoustic Dimensions

SURROUND LIVE II—MORE SURROUND is the second iteration of the most comprehensive event devoted exclusively to the creation, production, and reproduction of live performance audio in multichannel surround.

Following on the highly successful event held in NYC in 2003, SURROUND LIVE II—MORE SURROUND, will again be offered as a one-day interactive workshop.

Surround Live will bring together working professionals from the performance audio, broadway theater, broadcast, environmental audio, fine arts, and other industry segments as well as the recording technology areas, to discuss the issues and technological challenges created by presenting music, drama, theater, art, and cultural exhibitions in full multichannel surround audio formats to an audience.

Combining formal presentations with an interactive workshop and live/prerecorded performances, attendees will be able to experience the process of creating and presenting multichannel audio for a variety of live applications, and how this differs from the processes associated with multichannel work done in a post-production environment.

A full 5.1+ channel large scale tour sound system will be in place courtesy of Meyer Sound Labs, along with a DIGICO console, and electronic reverberation technology from Lexicon, Inc.. Shure, Inc. will be supply a wide range of wired and wireless microphones for use in the workshop. A live performance will be an integral part of the event.

Anyone who is involved in performance audio, multichannel sound, or is interested in how the two combine, is invited to attend. One of the main themes of the event is how immersing the audience in the sound field and keeping them closer to the sound system can improve the experience for both audience members and performers.

Registration Fees: AES Members: USD 150; Nonmembers USD 175. This fee includes coffee breaks and lunch and access to all four days of the 117th AES Convention Exhibits at the Moscone Convention Center. There are price reductions available to students or if you combine this Symposium with the Full Program registration for the 117th AES Convention.

This event is in part sponsored and supported by Digico, Meyer Sound Labs, Shure, and the TC Group.

Preliminary Events Schedule

9:30 am - 10:00 am	Coffee and Registration
10:00 am – 12:00 noon	Formal Presentations and Demonstrations
12:00 noon -12:45 pm	Lunch
12:45 pm - 1:00 pm	Q&A from morning session
1:00 pm - 3:30 pm	Formal Presentations (cont'd)
3:30 pm – 5:00 pm	Live Performance Demonstration and Mini-workshop on Surround Mixing and Presentation featuring a Special Guest Performer TBA.
5:00 pm - 5:45 pm	Q&A and Conclusion

Preliminary Presentations Schedule

1. Event Introduction and Opening Presentation—*Frederick Ampel*, Technology Visions, Overland Park, KS
2. Multichannel Audio Concepts in Sound Reinforcement—*Kurt Graffy*, Arup Acoustics
3. Multichannel Audio in Religious Facilities/Presentations—Steve Schull, Acoustic Dimensions
4. Use of surround techniques for sound effects in live drama —*Bruce Olson*
5. Live Theater and Surround Audio—*Mark Hood*, Echo Park Studios, Bloomington, IN
6. A Live Surround Orchestral Experience—*Jonathan Lane-Talaskey*, Jay Pritzker Pavilion, Millennium Park
7. LUNCH
8. Operatic Surround—*David Claringbold*, The Sydney Opera, Sydney, Australia
9. Source-oriented Reinforcement/Delay-Imaging—*Dave Haydon*, Outboard Electronics
10. Spatial Positioning and Multimedia Interaction—*Riccardo*

Mazza, School of High Musical Specialization, Saluzzo, Italy

11. Surround Microphone Applications and Techniques—*Bobby Owsinski*, Surround Associates, Studio City, CA.

12. Multichannel Audio in Live Sports Broadcasting—*Randy Hoffner*, ABC Television

13 .Demos and Live Band

FREE HEARING SCREENINGS
CO-SPONSORED BY THE AES,
HOUSE EAR INSTITUTE, AND SHURE, INC.

Thursday, October 28	12:00 noon–6:00 pm
Friday, October 29	10:00 am–6:00 pm
Saturday, October 30	10:00 am–6:00 pm
Sunday, October 31	10:00 am–4:00 pm

Exhibit Hall

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

gAmEShowcase

Thursday, October 28, 12:00 noon – 6:00 pm
Friday, October 29, 10:00 am – 6:00 pm
Saturday, October 30, 10:00 am – 6:00 pm
Sunday, October 31, 10:00 am – 4:00 pm
Games Room (Opposite Room 300)

Come experience the best of game audio today. If you haven't truly listened to video games in a while, now's your chance! This is not your old Commodore 64. Featuring the latest and greatest game consoles, you are invited to kick back and hear what all the fuss is about. Where else can you hear and play games and call it "industry research?"

Sponsors to date include: Dolby, Genelec, Microsoft, Nintendo, and Spherex. More to come. . . .

OPPORTUNITIES FOR THE ENGINEER IN THE DIGITAL BROADCAST WORLD
Thursday, October 28, 1:30 pm – 3:00 pm
Room 306

Panelists **Andy Butler**, PBS
David Layer, NAB
Tony Masiello, XM
Glynn Walden, Viacom/ Infinity
David Wilson, CE

This event will discuss job opportunities and the training needed for the audio professional in today's "Digital Broadcast World." What are the Employers looking for, and what jobs are out there for the audio professional … how your skills can be utilized.

16TH ANNUAL GRAMMY® RECORDING SOUNDTABLE
Thursday, October 28, 4:00 pm – 6:00 pm
Room 305/307

The Producers and Engineers Wing of the Recording Academy® presents: GAME ON! Video Games: The Future of Music and Entertainment

Chair: **Dave Adelson**, Hits Magazine and Producer/Music Correspondent for E! News Live

Panelists: *Buzz Burrowes*, Director of Tools, Technology and Services, Sony Computer Entertainment America
Niles Rodgers, Legendary Songwriter and Producer
Brian Schmidt, Program Manager for Xbox Audio, Media and Voice Technologies
Tommy Tallarico, President of Tommy Tallarico Studios, Inc. and Founder of Game Audio Network Guild (G.A.N.G.)

The 16th Annual GRAMMY Recording SoundTable is presented by the National Academy of Recording Arts & Sciences, Inc. (NARAS) and hosted by AES. Please join top music makers, experts, and creators of video games as they relate experiences and knowledge about getting in—and winning—in this exciting and thriving multimedia industry.

MIXER PARTY
Thursday, October 28, 6:00 pm – 8:00 pm
Esplanade Ballroom

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

Special Events

ORGAN RECITAL BY GRAHAM BLYTH
Thursday, October 28, 8:30 pm – 10:00 pm
Grace Cathedral
1100 California Street (at Taylor Street)
San Francisco, California 94108

Buses will depart from the Moscone Center at 7:45 pm and will return to Moscone and the AES Hotels after the performance.

NOTE: In conjunction with the recital, AES President Ron Streicher will present a tutorial (T6) at the cathedral starting at 6:00 pm. For details see Tutorial Seminars.

Graham Blyth will perform an organ recital at Grace Cathedral. The Alexander Memorial Organ, given by Harriet Crocker Alexander in memory of her husband Charles B. Alexander, was built by the Æolian-Skinner Company, Boston, MA, under the direction of G. Donald Harrison in 1934, with minor alterations in 1952, and additions in 1974 by Casavant Frères. It comprises 125 ranks of pipes spread over seven divisions. A total of 7,347 pipes make it one of the largest organs in the Western United States, and its beauty one of the finest American Classic organs in the world.

Blyth's program features Louis Vierne's 1st Organ Symphony and Cesar Franck's Chorale No. 2 together with works by Buxtehude, Cocker, Elgar, Guilmant, and Mulet.

Graham Blyth received his early musical training as a Junior Exhibitioner at Trinity College of Music in London, UK. 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 the Trinity College of Music. In the late 1980s he renewed his studies with Sulemita Aronowsky for piano and with Robert Munns for organ. Blyth made his international debut with an organ recital at St. Thomas Church, New York, in 1993, and since then has presented concerts on some of the finest organs in those cities visited by the AES, including San Francisco (Grace Cathedral), Los Angeles, Amsterdam, Copenhagen, Munich, Paris (Madeleine Church), and Berlin. He gives numerous concerts each year, principally as an organist and a pianist, but also as a conductor and a harpsichord player. Blyth is founder and technical director of Soundcraft. He divides his time between his main career as a designer of professional audio equipment and music-related activities. He has lived in Wantage, Oxfordshire, UK, since 1984, where he is currently artistic director of Wantage Chamber Concerts and director of the Wantage Festival of Arts. He is also founder and conductor of the Challow Chamber Singers & Players. He is involved with Musicom Ltd., a British company at the leading edge of pipe organ control system and digital pipe synthesis design and has recently formed the Veritas Organ Company to address the aspirational end of the church organ market.

AUDIO PROFESIONAL EN LATINOAMERICA
(Professional Audio in Latin America)
Friday, October 29, 9:00 am – 12:00 noon
Room 306

Chair: **Elmar Leal,** Technical Director, Taller de Arte Sonoro, Caracas, Venezuela

Panelists: *Paul Lima*, Brazil
Horacio Malvicino, Argentina
Andres Mayo, USA
Mercedes Onorato, AES Vice President, Latin American Region
George Petersen, Mix Magazine
David Rodriguez, Discmakers
Jorge Urbano, Sound Check Magazine, Mexico

A successful series of events from 1992 to 1997 comes back to unite the Latin America audio community again. A panel of specialists in different areas of the audio industry in Latin America will discuss important issues such as audio education, career opportunities, musical production trends, music business, industrial audio products, technical advances in premastering and restoration, and distribution of technical information in Spanish. This event also includes an international panel that will focus on the continuing exchanges between all AES sections in reference to education, seminars, and convention organization, in order to share material and achievements and to continue the task of expanding this promising region. Ample time will be reserved for audience participation. This event will be conducted primarily in Spanish.

DIGITAL BROADCAST RADIO FORUM
Friday, October 29, 9:00 am – 12:00 noon
Room 305/307

Moderator: **David Bialik**, Systems Engineering Consultant

Panelists: *David Layer,* NAB NRSC
Michael Lyons, Ibiquity
Tony Masiello, XM
Geir R. Skaaden, Neural Audio
*Mike Starling,*Tomorrow Radio
Fred von Lohmann, Electronic Frontier Foundation
David Wilson, CE

Fourteen years ago the AES had the first forum in NY. Digital Radio has now gone from theory to reality. We will discuss Eureka, IBOC (HD Radio), Satellite Radio, Tomorrow Radio, and the listening environments. This is a chance to become familiar with the newest radio technology that has been successfully adopted in the US.

PLATINUM PRODUCERS
Friday, October 29, 1:00 pm – 2:30 pm
Room 305/307

Moderator: **Ron Fair**, President of A&M Records

Panelists: *Howard Benson*, Producer and A&R, Warner Bros.
Jack Joseph Puig, Producer
Phil Ramone, Producer
Chink Santana, Producer
Mark Wright, Producer, Executive VP A&R, Sony Records

A panel of renowned and highly successful producers will eloquently address the continually evolving technical and creative issues with which they are faced today.

Howard Benson, credits include Hoobastank, My Chemical Romance, Papa Roach, POD.

Jack Joseph Puig, credits include John Mayer, Green Day, No Doubt.

Phil Ramone, credits include Elton John, Peter Cincotti, Billy Joel, Paul Simon, Ray Charles Duets.

Chink Santana, credits include Fat Joe f/Ashanti, Wuts Luv, Irv Gotti, Ja Rule.

Mark Wright, credits include Brooks & Dunn, Gretchen Wilson, Gary Allen, Susan Haynes.

PLATINUM ENGINEERS—THE ALL-STAR MIX ENGINEERS PANEL
Friday, October 29, 3:00 pm – 4:30 pm
Room 305/307

Moderator: **Jack Joseph Puig**, Ocean Way Studios, Los Angeles, CA, USA

Panelists: *Bob Clearmountain*, Mixer/Producer
Tom Lord-Alge, Mixer

Today's music is being recorded in not only commercial facilities but in unconventional places such as warehouses, garages, and bedrooms on a multitude of different mediums making the mix engineers role more important than ever before. This panel of world renown mix engineers will discuss the challenges and techniques of taking an album with multiple mediums, producers, and recording engineers and making it sonically consistent and radio ready.

Jack Joseph Puig, credits include John Mayer, Green Day, No Doubt (to mention a few). Known for his ability to merge the sounds of the past three decades to create something that is thoroughly modern, Puig has produced and mixed hits for John Mayer and Green Day among many others. Calling L.A.'s Ocean Way studios home, he has spent time both as an engineer and as a producer and sees the duality as a plus when dealing with today's Top 40 rock acts who often ask him to mix their albums.

Bob Clearmountain, credits include The Finn Brothers, The Rolling Stones, Bryan Adams, Bruce Springsteen, Joe Cocker, Shawn Colvin, INXS.

Tom Lord-Alge, credits include Avril Lavigne, blink 182, Dave Matthews.

SURROUND SOUND FOR DIGITAL RADIO
Saturday, October 30, 9:00 am – 12:00 noon
Room 305/307

Moderator: **Emil Torick**

Panelists: *Frank Foti,* Telos Omnia
Rocky Graham, Dolby Labs
Alan Kraemer, SRS Labs
Michael Lyons, Ibiquity
Tony Masiello, XM Satellite Radio
Robert Orban, Orban
Robert Reams, Neural Audio

Radio is evolving past the barriers of stereo and is introducing surround sound. We will discuss the various systems and how they work within the various broadcast and bandwidth restraints.

PLATINUM MINDS: FROM STEREO TO SURROUND
Saturday, October 30, 12:30 pm – 2:00 pm
Room 305/307

Moderator: **Nathaniel Kunkel**, Independent Producer

Panelists: *Peter Chaikin*, JBL Professional
Geoff Emeric, Engineer/Producer
Robin Hurley, Warner Strategic Marketing (WSM)
Jeff Levison, Audio Disc Producer
George Massenburg, GML, LLC
Rodney Orpheus, DTS Entertainment
Jim Pace, AID
Ronald Prent, Galaxy Studios

When will surround sound become the everyday way to listen to music? For the last five years surround sound has made huge inroads into studios, personal listening places, cars, and homes. The technology continues to evolve as well as become more friendly and cost effective. CES/CEA is keenly interested in this format as DVD-Video sales are skyrocketing and DVD-Audio and SACD sales are still lagging. The producers, engineers, recording labels, and manufacturers are better working together than ever before to perfect these formats. Last year at AES the Grammy's showcased their panel in surround-only and it was the highest attendance they had in recent years. There is still an immense hunger for the format from the listener's perspective as well as the producer's, engineer's, and label's, who still have not gotten into this provocative format. The panel will discuss how the surround sound-only formats can be used to become the consumer audio format of choice.

AN AFTERNOON WITH BOB MOOG
Saturday, October 30, 2:30 pm – 4:00 pm
Room 305/307

Moderator: **Craig Anderton**, Editor at Large, EQ Magazine

In 1964, Bob Moog showed his first electronic music modules to the AES Convention in New York. Now, forty years later after changing the world of audio forever, Moog returns to the AES for an informal, interactive discussion with the audience. From building the synthesis equipment for the best-selling classical album of all time (*Switched-On Bach*), to his current efforts in pushing the musical state of the art in expressive synthesis, Moog remains an entertaining, engaging personality who continues to play a vital role in the audio community.

SPARS MENTORING SESSION
Saturday, October 30, 4:00 pm – 5:30 pm
Concourse

Moderator: **Paul Gallo**, Executive Director, SPARS

Mentors: Capitol Studios, Los Angeles, Ca, USA
Crawford Audio Services, Atlanta, GA, USA
Emerald, Nashville, TN, USA
Larrabee Studios, Los Angeles, CA, USA
Masterdisk, New York, NY, USA
Remote Recording, New York, NY, USA
Skywalker Sound, San Francisco, CA, USA
Tiki Recording, Glen Cove, NY, USA
Village Recorder, Los Angeles, Ca, USA

The Society of Professional Audio Recording Services (SPARS) is a twenty-five-year-old professional organization dedicated to sharing practical, hands-on business information about audio facility ownership, management, and operations.

This event, which is co-hosted by the AES, features an elite panel of studio owners, managers, and recording industry professionals who will explore strategies for adapting your business to changing times. Mentors will be available for one-on-one or group sessions to mentor and discuss innovative ways to handle unique challenges facing the successful operation of an audio production facility.

Special Events

OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE

Saturday, October 30, 6:30 pm – 8:30 pm
Room 305/307

Lecturer: **Walter Murch**

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 117th AES Convention is Walter Murch. Murch has been editing sound in Hollywood since starting on Francis Ford Coppola's film The Rain People (1969). He edited sound on American Graffiti (1973) and The Godfather Part II (1974), won his first Academy Award nomination for The Conversation (1974), won his first Oscar for Apocalypse Now (1979), and won unprecedented double Oscars for sound and film editing for his work on The English Patient (1996). Most recently he helped reconstruct Touch of Evil to Orson Welles' original notes, and edited The Talented Mr. Ripley. Murch was, along with George Lucas and Francis Coppola, a founding member of northern California cinema. He has directed—Return to Oz (1985)—and longs to do so again, but as an editor and sound man he is one of the few universally acknowledged masters in his field. For his work on the film Apocalypse Now, Walter coined the term "Sound Designer," and along with colleagues such as Ben Burtt, helped to elevate the art and impact of film sound to a new level.

Murch's lecture is entitled, "Edison's First Sound Film and the Three Fathers of Cinema." This event features a presentation of Edison's first sound film from 1894, thought to be lost for many years. Patrick Loughney at the Library of Congress recently rediscovered and repaired the long-lost cylinder, and sent it to Murch to resync with the nitrate picture. He will give an explanation of how this was done, despite the fact that there was no start mark, no standard frame rate for film, nor rpm for the cylinder. Then, using Edison as a springboard, he will examine the hypothetical question: what would have happened to cinema if film+sound had been invented 100 years earlier? Would we have known what to do with it, or were certain other 19th century cultural developments—dynamism in music (Beethoven) and realism in literature (Flaubert)—needed to prepare the way for film to catalyze cinema's true strengths:

the dynamic representation of closely-observed reality.

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

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

Special Events

STREAMING DSD AUDIO COMES TO THE AES
Sunday, October 31, 9:30 am – 11:30 am
University of California San Francisco's Genentech Hall

Buses will depart from the Moscone Center at 8:45 am and will return after the performance.

Moderators: **Jeremy Cooperstock, Wieslaw Woszczyk,**
McGill University, Montreal, Quebec, Canada

The AES Technical Committee on Network Audio Systems will demonstrate streaming of six channels of DSD audio (the 2.8 MHz sampling system used for SACD), transmitting a live jazz performance at McGill University in Montreal, Canada, in real time, to the University of California San Francisco's Genentech Hall. The audience will experience the performance with the latest in audio technology, accompanied by an SDI digital video projection, transmitted using the McGill UltraVideoconferencing system. Technical issues and implications to future recording and performance applications will be discussed. A multisite Halloween performance is also planned.

ROAD WARRIORS – LIVE SOUND
Sunday, October 31, 12:30 pm – 2:00 pm
Room 305/307

Moderator: **Clive Young**

Panelists: *Monty Lee Wilkes*, Journeyman FOH engineer
TBA

This freewheeling panel of touring professionals will cover the latest trends, techniques, and tools that shape modern sound reinforcement. The all-star panel will careen through subject matter ranging from gear to gossip, in what promises to be an entertaining and educational 90 minutes—with the engineers on the business side of the microphone, saying something besides “testing” and “check” for a change!

Special Events

TOM DOWD & THE LANGUAGE OF MUSIC
Sunday, October 31, 2:00 pm – 4:00 pm
Room 303

This 90-minute documentary profiles the life and work of legendary producer/recording engineer Tom Dowd. A former Atlantic Records engineer, Dowd was responsible for some of the most important R&B, rock, and jazz records ever made. Interviews with recording industry icons tell the story of this humble genius, while historical footage, photographs, and classic music tracks capture the magic of the recording studio. It's here that Dowd, the unaffected master, recounts the recording sessions and technical achievements that altered the course of contemporary music. The film includes powerful interviews, photos, and film clips—both historical and contemporary—of Ray Charles, Aretha Franklin, Eric Clapton, Otis Redding, the Allman Brothers, Les Paul, Phil Ramone, Joe Bonamassa, Ahmet Ertegun, and many more musical giants.

This special viewing has been made possible by Dana Dowd. Dana Dowd's expertise in marketing, event coordination, media buying, and logistics has landed her pivotal roles in events for OpSail, Microsoft, and Playboy, among others. In 2003 she began a company named Silver Fox Productions in honor of her late father, Tom Dowd. Since its inception, Silver Fox Productions has become a vital part of preserving her father's legacy by promoting and marketing his music and photography, as well as the film *Tom Dowd & The Language of Music*. Dowd also works closely with her mother Cheryl Dowd, president of Tom Dowd Productions, to continue the preservation of her father's work in engineering and producing.

Student Activities

Student activities are open to all attendees with a full program badge. All attendees with an interest in audio education are welcome to attend. The main program contains a series of 25 entry-level **Tutorial Seminars**, which are of particular interest to students. These are described on pages 139 to 165.

STUDENT DELEGATE ASSEMBLY (SDA) MEETING -1
Thursday, October 28, 10:00 am – 11:30 am
Room 306

All students and educators are invited to participate in this meeting. This opening meeting of the Student Delegate Assembly will introduce new rules and regulations concerning elections, announce candidates for the coming year's election for the North/Latin America Regions, announce the finalists in the recording and design competition categories, hand out the judges' sheets to the nonfinalists, and announce any upcoming events of the convention. Also at this time there will be sign-up sheets for the mentoring sessions.

"STEPS TO THE FUTURE"—ONE ON ONE MENTORING -1
Thursday, October 28, 1:00 pm – 3:00 pm
Room 310

Students are invited to sign-up for an individual meeting with a distinguished mentor from the audio industry. Sign-ups will begin at the first SDA meeting and additional sign-up sheets will be located in the student area at the convention. All students are encouraged to participate in this exciting and rewarding opportunity for individual discussion with industry mentors.

DESIGN COMPETITION
Friday, October 29, 11:30 am – 1:00 pm
Room 310

The design competition is a competition for audio projects made by students at any university or recording school, which will challenge students with an opportunity to showcase their technical skills. This is not for recording projects or theoretical papers. Designs will be judged by a panel of industry experts in design and manufacturing. Multiple prizes will be awarded.

STUDENT POSTER SESSION
Friday, October 29, 11:30 am – 1:00 pm
Outside Rooms 301, 302

This session provides an opportunity for students to present their research work on all subjects covered by the AES.

Student Activities

"STEPS TO THE FUTURE"—ONE ON ONE MENTORING SESSION -2
Friday, October 29, 12:00 noon – 2:00 pm
Room 310

Students are invited to sign-up for an individual meeting with a distinguished mentor from the audio industry. Sign-ups will begin at the first SDA meeting and additional sign-up sheets will be located in the student area at the convention. All students are encouraged to participate in this exciting and rewarding opportunity for individual discussion with industry mentors.

EDUCATION FAIR
Friday, October 29, 12:30 pm – 2:30 pm
Concourse

Institutions offering studies in audio (from short courses to graduate degrees) will be represented in a "table top" session. Information on each school's respective programs will be made available through displays and academic guidance. There is no charge for schools to participate. Admission is free and open to all convention attendees.

EDUCATION FORUM PANEL: "EDUCATION IN THE FIELD OF AUDIO PRODUCTION"
Friday, October 29, 4:00 pm – 6:00 pm
Room 306

Moderator: **David Christensen,** The Art Institute of Seattle, Seattle, WA, USA

Panelists: **Martha de Francesco,** McGill University, Montreal, Quebec, Canada
John Lay, New England Institute of Art, West Brookline, MA, USA
Theresa Leonard, The Banff Centre, Banff, Alberta, Canada
Bob Ludwig, Gateway Mastering & DVD, Portland, Maine, USA
Geoff Martin, Bang & Olufsen, Struer, Denmark
Roy Pritts, University of Colorado, Denver, CO, USA

Audio production post secondary education runs from short term "certificate" programs through postgraduate programs. AES is in a unique position of providing support to these schools. AES provides a networking channel between educators including our new on-line educational forum with active student sections at many of these schools and gives support to these sections. The AES Conventions offer educational events which bring together students and faculty from many schools. This panel is designed to support and expand the AES educational network and focus on common issues faced by audio production schools and their students.

Specifically: (1) Maintaining a facility that will attract students; (2) Attracting industry participation; (3) Keeping curriculum current with industry needs; (4) Providing a broad-based program allowing for multitalented graduates; (5) Internships; and (6) Placement of graduates.

RECORDING COMPETITION—STEREO
Saturday, October 30, 9:00 am – 12:30 pm
Room 306

Finalists selected by the prejudging panel will give brief descriptions and play recordings.

9:00 am – 10:00 am	Classical Category
10:15 am – 11:15 am	Jazz/Folk Category
11:30 am – 12:30 pm	Pop/Rock Category

Judges:
Classical: Martha de Francesco
Jazz/Folk: Jim Anderson, Lynn Fuston
Pop/Rock: Jean-Marie Horvat, Ronald Prent

RECORDING COMPETITION—SURROUND
Saturday, October 30, 2:00 pm – 5:30 pm
Room 308

Finalists selected by the prejudging panel will give brief descriptions and play recordings.

2:00 pm – 3:00 pm	Classical Surround
3:15 pm – 4:15 pm	Nonclassical Surround
4:30 pm - 5:30 pm	Post Surround

Judges:
Classical: Martha de Francesco
Nonclassical Surround: Jim Anderson, Lynn Fuston, Jean-Marie Horvat, Ronald Prent
Post Surround: Tom Holman, Shawn Murphy, Mark Willsher

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

STUDENT DELEGATE ASSEMBLY (SDA) MEETING -2
Sunday, October 31, 9:30 am – 11:30 am
Room 306

The closing meeting of the SDA will elect a new vice chair. Votes will be cast by the designated representative from each recognized AES student section or academic institution in the North/Latin America Regions. Judges' comments and awards will be presented for the Recording and Design Competitions. Plans for future student activities at local, regional, and international levels will be summarized.

AES Meetings

REGIONS AND SECTIONS MEETING
A meeting of the officers of all AES Sections will take place on Sunday, October 31, from 9:00 am – 11:00 am, in Room 303.

HISTORICAL COMMITTEE MEETING
A meeting of the Historical Committee will take place during the convention. Attendance is open to all. Please see posted signs for date, place, and time.

Technical Council and Technical Committee Meetings

The **TECHNICAL COMMITTEES**, coordinated by the AES Technical Council, track trends in audio in order to recommend to the Society special papers sessions, standards projects, publications, and awards in their fields. The TC meetings are open to all convention registrants.

THURSDAY, OCTOBER 28

9:30 am	Network Audio Systems (Room 300)
1:00 pm	Automotive Audio (Room 300)
2:00 pm	High Resolution Audio (Room 300)
4:00 pm	Coding of Audio Signals (Room 302)
4:00 pm	Human Factors in Audio (Room 300)
5:00 pm	Loudspeakers and Headphones (Room 301)

FRIDAY, OCTOBER 29

9:30 am	Studio Practices and Production (Room 300)
11:30 am	Archiving, Restoration, and Digital Libraries (Room 301)
11:30 am	Microphones and Applications (Room 300)
2:00 pm	Semantic Audio Analysis (Room 300)
4:00 pm	Audio for Telecommunications (Room 300)
6:00 pm	Multichannel and Binaural Audio Technologies (Room 304)

SATURDAY, OCTOBER 30

9:30 am	Audio for Games (Room 300)
11:30 am	Perception and Subjective Evaluation of Audio (Room 300)
1:30 pm	Transmission and Broadcasting (Room 300)
3:00 pm	Hearing Conservation (Room 300)
4:30 pm	Audio Recording and Storage Systems (Room 300)
5:00 pm	Signal Processing (Room 302)
5:30 pm	Acoustics and Sound Reinforcement (Room 301)

OPEN HOUSE OF THE TECHNICAL COUNCIL AND THE RICHARD C. HEYSER MEMORIAL LECTURE

Saturday, Oct. 30 6:30 pm – 8:00 pm Room 305/307

Lecturer: **Walter Murch**

For complete details see page 30.

Standards Committee Meetings

STANDARDS COMMITTEE MEETINGS

The AES Standards Committee (AESSC) is the organization responsible for the AES Standards Program. It publishes a number of technical standards, Information documents, and technical reports.

Over 65 working groups and task groups with a fully international membership are engaged in writing standards covering fields that include:

- Digital Audio
- Preservation and Restoration
- Acoustics
- Interconnections
- Networks and File Transfer

STANDARDS MEETINGS

Meetings of Standards Committee working groups take place starting two days prior to the opening of the convention and run throughout the convention.

Standards working group meetings are open to all individuals who are materially and directly affected by the documents that may be issued under the scope of the working group.

The schedule of meetings follows.

Meetings, including plenary meetings of the Standards Committee, are scheduled to take place in Standards Room 1 or Room 2. The schedule is subject to changes and additions. Daily updates may also be obtained in the Standards Facilities/ Writing Room.

Complete information, including scope of working groups and project status, is available at http://www.aes.org/standards.

Contact: standards@aes.org.

TUESDAY, OCTOBER 26

1:30 pm	SC-02-02 Digital Input/Output Interfacing (Room 1)

WEDNESDAY, OCTOBER 27

9:00 am	SC-06-01 Audio File Transfer and Exchange (Room 1)
11:30 am	SC-06-02 Audio Applications Using the High Performance Serial Bus (IEEE 1394) (Room 1)
2:00 pm	SC-05-02 Audio Connectors (Room 1)
4:00 pm	SC-05-05 Grounding and EMC Practices (Room 1)

Standards Committee Meetings

THURSDAY, OCTOBER 28

9:00 am	SC-06-06 Audio Metadata (Room 1)
11:00 am	SC-03-06 Digital Library and Archive Systems (Room 1)
1:00 pm	SC-03-04 Storage and Handling of Media (Room 1)
3:30 pm	SC-03-01 Analog Recording (Room 1)
5:00 pm	SC-03-02 Transfer Technologies (Room 1)

FRIDAY, OCTOBER 29

9:00 am	SC-04-04 Microphone Measurement and Characterization (Room 1)
11:30 am	AESSC Plenary I (Room 1)
2:00 pm	SC-02-01 Digital Audio Measurement Techniques (Room 1)
4:30 pm	SC-06-04 Internet Audio Delivery System (Room 1)
6:00 pm	SC-03-12 Forensic Audio (Room 1)

SATURDAY, OCTOBER 30

9:00 am	SC-04-03 Loudspeaker Modeling and Measurement (Room 1)
11:00 am	SC-04-01 Acoustics and Sound Source Modeling (Room 1)
1:30 pm	SC-04-07 Listening Tests (Room 1)

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

SUNDAY, OCTOBER 31

9:00 am	TBA
11:30 am	AESSC Plenary II (Room 1)

PLEASE NOTE: The AES reserves the right to examine briefcases, literature bags or sacks, and handbags for security reasons.

Recording, video taping or photographing the technical sessions, workshops, or special events is not permitted.

The AES takes no responsibility for the contents of the technical sessions, workshops, tutorial seminars, special events, or exhibitor seminars.

The information contained in this book is subject to change.

Technical Paper Sessions

PAPERS

Session A	**Loudspeakers, Part 1**	Thursday, Oct. 28 9:00 am–11:30 am Room 301
Session B	**Lossy and Lossless Audio Coding, Part 1**	Thursday, Oct. 28 9:00 am –11:30 am Room 304
Session C	**Loudspeakers, Part 2**	Thursday, Oct. 28 1:00 pm–5:00 pm Room 301
Session D	**Lossy and Lossless Audio Coding, Part 2**	Thursday, Oct. 28 1:00 pm–4:00 pm Room 302
Session Z1	**Posters: Music Synthesis and Audio Archiving, Storage, and Restoration; Content Management**	Thursday, Oct. 28 2:00 pm–4:00 pm Outside 301/302
Session E	**Audio (Including Telephony) Over Networks**	Friday, Oct. 29 9:00 am–11:30 am Room 301
Session F	**Audio Archiving, Storage, and Restoration; Content Management**	Friday, Oct. 29 9:00 am–11:00 am Room 302
Session Z2	**Posters: Instrumentation and Measurement and Lossy and Lossless Audio Coding**	Friday, Oct. 29 9:00 am–11:00 am Outside 301/302
Session G	**Audio for Computer Games**	Friday, Oct. 29 11:30 am–12:30 pm Room 302
Session H	**Multichannel Sound**	Friday, Oct. 29 2:00 pm–6:00 pm Room 304
Session I	**Psychoacoustics, Perception, and Listening Tests**	Friday, Oct. 29 1:30 pm–6:00 pm Room 302
Session Z3	**Posters: Loudspeakers, Part 1**	Friday, Oct. 29 1:30 pm–3:30 pm Outside 301/302
Session Z4	**Posters: Loudspeakers, Part 2**	Friday, Oct. 29 4:00 pm–6:00 pm Outside 301/302
Session J	**Microphones**	Saturday, Oct. 30 9:00 am–10:30 am Room 301
Session K	**Signal Processing, Part 1**	Saturday, Oct. 30 9:00 am–12:30 pm Room 302

Session Z5	**Posters: Multichannel Sound**	Saturday, Oct. 30 9:30 am–11:30 am Outside 301/302
Session L	**Instrumentation and Measurement**	Saturday, Oct. 30 11:00 am–12:30 pm Room 301
Session M	**Room and Architectural Acoustics; Sound Reinforcement**	Saturday, Oct. 30 1:30 pm–5:30 pm Room 301
Session N	**Signal Processing, Part 2**	Saturday, Oct. 30 1:30 pm–5:00 pm Room 302
Session Z6	**Posters: Psychoacoustics, Perception, and Listening Tests and Spatial Perception and Processing**	Saturday, Oct. 30 2:00 pm–4:00 pm Outside 301/302
Session O	**Audio Recording and Reproduction**	Sunday, Oct. 31 9:00 am–10:30 am Room 301
Session P	**High Resolution Audio**	Sunday, Oct. 31 9:00 am–10:30 am Room 302
Session Z7	**Posters: Signal Processing, Part 1**	Sunday, Oct. 31 9:30 am–11:30 am Outside 301/302
Session Q	**Automotive Audio**	Sunday, Oct. 31 11:00 am–12:30 pm Room 302
Session Z8	**Posters: Signal Processing, Part 2**	Sunday, Oct. 31 1:00 pm–3:00 pm Outside 301/302
Session R	**Spatial Perception and Processing**	Sunday, Oct. 31 1:30 pm – 3:30 pm Room 301
Session S	**Audio-Video Systems**	Sunday, Oct. 31 1:30 pm – 3:30 pm Room 302

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

Session A Thursday, October 28 9:00 am – 11:30 am
Room 301

LOUDSPEAKERS, PART 1

Chair: **Jerry Bauck**, Cooper Bauck Corporation, Tempe, AZ, USA

P A P E R S

9:00 am

A-1 Voice Coil Impedance as a Function of Frequency and Displacement—*Mark Dodd*,[1] *Wolfgang Klippel*,[2] *Jack Oclee-Brown*[1]
[1]KEF Audio UK (Ltd.), Maidstone, Kent; UK
[2]Klippel GmbH, Dresden, Germany

Recent work by Klippel and Voishvillo, et al. has shown the significance of voice coil inductance with respect to the nonlinear behavior of loudspeakers. In such work the methods used to derive distortion require the inductance to be represented by an equivalent circuit rather than the frequency domain models of Wright and Leach. A new technique for measurement of displacement and frequency dependent impedance has been introduced. The complex relationship between coil impedance, frequency, and displacement has been both measured and modeled, using stationary transient FEM, with exceptional agreement. Results show that the impedance model requires that its parameters vary independently with displacement to satisfactorily describe all cases. Distortion induced by the variation of impedance with coil displacement is predicted using a lumped parameter method. This prediction is compared to measurements of the actual distortion. The possibility of using a dynamic transient FE method to predict distortion is demonstrated. The nature of the distortion is discussed.
Convention Paper 6178

9:30 am

A-2 Dynamical Measurement of Loudspeaker Suspension Parts—*Wolfgang Klippel*, Klippel GmbH, Dresden, Germany

The nonlinear stiffness K(x) and the reciprocal compliance C(x) of suspension parts (spider, surrounds, cones) and passive radiators (drones) are measured versus displacement x over the full range of operation. A dynamic, nondestructive technique is developed that excites the suspension parts pneumatically under a similar condition as operated in the loudspeaker. The nonlinear parameters are estimated from the measured displacement and sound pressure signal. This guarantees highest precision of the

results as well as simple handling and short measurement time. The paper develops the theoretical basis for the new technique but also discusses the practical handling, interpretation of the results, and their reproducibility.
Convention Paper 6179

10:00 am

A-3 Modeling Acoustic Room Interaction for Pistonic and Distributed-Mode Loudspeakers in the Correlation Domain—*Neil Harris*, New Transducers Ltd. (NXT), Huntingdon, UK

AES Convention Paper 5732 (AES 114th Convention Amsterdam, The Netherlands) presented a meshless, analytic 3-D solution to the problem of an acoustic source located in a nonanechoic room. By applying the inverse Fourier transform, temporal data was extracted to form a complete time and frequency domain description. This paper uses the same model to investigate the correlation properties of the acoustic radiation. Maps are produced, showing how the correlation varies with position in space. Statistical analysis suggests a possible objective classification of the diffuseness of acoustic fields by scalar quantities such as mean and standard deviation of correlation coefficient values. The distribution of correlation values is seen to follow the beta distribution quite closely.
Convention Paper 6180

10:30 am

A-4 Compensating the Acoustical Loading of Small Loudspeakers Mounted Near Desktops—*Andrew Goldberg*, *Aki Mäkivirta*, *Ari Varla*, Genelec Oy, Iislami, Finland

In professional audio applications, small loudspeakers are often mounted on or near (within the loudspeaker's near field region) large solid surfaces, such as mixing consoles, desktops, and work surfaces. In approximately two-thirds of loudspeakers mounted in such a fashion, the magnitude response is compromised in a predictable and systematic way. An upward deviation of peak value 5.0 dB ± 1.5 dB centered on 141 Hz ± 31 Hz was observable in approximately 80 percent of the cases studied. An additional Room Response Control in active loudspeakers is proposed to compensate for this aberration. A statistical analysis of 89 near-field loudspeakers helps define the correction filter and quantifies the effectiveness of the fixed filter design. Use of the proposed filter in an automated response optimization algorithm for in-situ response equalization is demonstrated.
Convention Paper 6181

11:00 am

A-5 Image Source Model for Loudspeaker Enclosure
—*Juha Backman*, Nokia, Espoo, Finland; Helsinki University of Technology, Espoo, Finland

An application of the image source model for computing the interior sound field of a loudspeaker enclosure is presented. The image source model allows computing the effects of individual reflections, enclosure modes, etc., on the response of the speakers in an efficient manner for rectangular enclosures, both sealed and ported. The effect of the absorbent material can be included without excessively adding the computational complexity. As the model makes no assumptions on the enclosure size, it can be equally well applied for modeling, e.g., in-wall loudspeakers, where a combined model of room and loudspeaker responses can be developed.
Convention Paper 6182

P A P E R S

Session B Thursday, October 28 9:00 am – 11:30 am
Room 304

LOSSY AND LOSSLESS AUDIO CODING, PART 1

Chair: **Michael Goodwin**, Creative ATC, Scotts Valley, CA, USA

9:00 am

B-1 MPEG-4 Scalable to Lossless Audio Coding—
Rongshan Yu,[1] *Ralf Geiger*,[2] *Susanto Rahardja*,[1] *Jürgen Herre*,[3] *Xiao Lin*,[1] *Haibin Huang*[1]
[1]Institute for Infocomm Research, Singapore, Singapore
[2]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany
[3]Fraunhofer Institute for Integrated Circuits IS, Erlangen, Germany;

As the latest extension of MPEG-4 audio coding, MPEG-4 lossless audio coding includes a scalable audio coding solution (SLS) that integrates the functionalities of lossless audio coding, perceptual audio coding, and fine granular scalable audio coding into a single coder framework while providing backward compatibility to MPEG Advanced Audio Coding (AAC) at the bit-stream level. Despite its abundant functionalities, SLS still achieves a compression performance that is comparable to state-of-the-art non-scalable lossless audio coding algorithms. As a result, SLS provides a universal digital audio format for a variety of application domains including professional audio, Internet music, consumer electronics, broadcasting, and others. This paper presents the structure of SLS and its latest developments during the MPEG standardization process.
Convention Paper 6183

9:30 am

B-2 Improved Transient Pre-Noise Performance of Low Bit Rate Audio Coders Using Time Scaling Synthesis—
Brett Crockett, Dolby Laboratories, San Francisco, CA, USA

A new audio coding tool that uses improved time scaling synthesis techniques has been developed, which reduces the duration of pre-noise introduced by low bit-rate audio coding of transient material. When the transient pre-noise reduction processing is used, decoded PCM audio located prior to transient material is processed in the decoder using time scaling synthesis. The synthesized PCM audio is used to remove or reduce the duration of transient pre-noise, improving the perceived quality of low bit-rate audio coded transient material.
Convention Paper 6184

PAPERS

10:00 am

B-3 Subjective Evaluation of MPEG Layer II with Spectral Band Replication—*Gilbert Soulodre, Michel Lavoie*, Communications Research Centre, Ottawa, Ontario, Canada

Spectral Band Replication (SBR) was developed as a means of enhancing the coding of audio signals. It has been recently proposed to use SBR, integrated within the MPEG Layer II codec, as a possible extension to the EUREKA 147 DAB standard. The goal is to provide an equivalent level of subjective quality at a reduced bit rate. In the present paper formal subjective tests were conducted to evaluate the performance of Layer II+SBR at typical DAB bit rates. The tests included Layer II+SBR codecs operating at 128 and 160 kbps, as well as a standard Layer II codec at 128, 160, and 192 kbps. The subjective tests were conducted using the ITU-R BS.1534 (MUSHRA) methodology.
Convention Paper 6185

10:30 am

B-4 Spatial Audio Coding: Next-Generation Efficient and Compatible Coding of Multichannel Audio—
Jürgen Herre,[1] *Christof Faller*,[2] *Sascha Disch*,[1] *Christian Ertel*,[1] *Johannes Hilpert*,[1] *Andreas Hoelzer*,[1] *K. Linzmeier*,[1] *Claus Spenger*,[1] *P. Kroon*[2]
[1]Fraunhofer Institute for Integrated Circuits IS, Erlangen, Germany;
[2]Agere Systems, Allentown, PA, USA

Recently, a new approach in low bit rate coding of stereo and multichannel audio has emerged: Spatial audio coding permits an efficient representation of multichannel audio signals by transmitting a downmix signal along with some compact spatial side information describing the most salient properties of the multichannel sound image. Besides its impressive efficiency allowing multichannel sound at total bit rates of only 64 kbit/s and lower, the approach is also backward compatible to existing transmission systems and thus accommodates a smooth transition toward multichannel audio in the consumer market. The paper gives an overview of the basic concepts and the options provided by spatial audio coding technology. It reports about some recent performance data, first commercial applications and related activities within the ISO/MPEG standardization group.
Convention Paper 6186

PAPERS

11:00 am

B-5 Coding of Spatial Audio Compatible with Different Playback Formats—*Christof Faller*, Agere Systems, Allentown, PA, USA

Recently, various schemes were proposed for parametric coding of stereo and multichannel audio signals. Binaural Cue Coding (BCC) is such a technique. It represents multichannel audio signals as a single downmixed channel plus a small amount of side information. BCC can be applied to mono and stereo backwards compatible coding of multichannel audio signals. In this paper we propose a general paradigm for BCC with multiple transmission channels and show how this can be applied not only to bridging between mono/stereo and multichannel surround but also to bridging between different multichannel surround formats.
Convention Paper 6187

Session C **Thursday, October 28** **1:00 pm – 5:00 pm**
Room 301

P A P E R S

LOUDSPEAKERS, PART 2

Chair: **Marshall Buck**, Psychotechnology, Inc., Los Angeles, CA, USA

1:00 pm

C-1 Do Higher Order Modes at the Horn Driver's Mouth Contribute to the Sound Field of a Horn Loudspeaker? **—***Michael Makarski*, Aachen University, Aachen, Germany

The Boundary Element Method (BEM) is a well-known tool in acoustics for the calculation of radiation from vibrating surfaces. When using BEM for the calculation of horn loudspeakers, the horn surface is described by its surface admittance; the connected driver is modeled by the velocity distribution at the common junction of driver and horn. Measurements of the velocity distribution have shown that higher order modes within the horn throat can be excited by the horn driver (presented at the 116th AES Convention). On the other hand, a two-port description of the driver together with a plane-wave velocity distribution for the BEM calculation leads to good results. It is investigated to what extend higher order modes at the driver's mouth contribute to the sound radiation.
Convention Paper 6188

1:30 pm

C-2 Investigating the Potential Benefits to Both the Objective and Subjective Performance of a Two-Way Loudspeaker Obtained by Using a Wide-Band "Tweeter" to Place the Cross-Over at a Lower than Usual Frequency—*Neil Harris, Alan Hildyard, Valerie Taylor*, New Transducers Ltd. (NXT), Huntingdon, UK

The tweeter in a two-way loudspeaker was replaced by a unit having a natural bandwidth of 300 Hz to 20 kHz. This gave a much greater degree of freedom to the choice of cross-over frequency than would normally be possible. The first part of this paper looks at the potential benefits such freedom could bring to the acoustical performance of the loudspeaker. The second part reports results of early listening tests, which were conducted to discover the most preferred cross-over frequency in the range 700 Hz to 3 kHz.
Convention Paper 6189

PAPERS

2:00 pm

C-3 A Multiple Regression Model for Predicting Loudspeaker Preference Using Objective Measurements: Part II—Development of the Model—*Sean Olive*, Harman International Industries, Inc., Northridge, CA, USA

A new model is presented that accurately predicts listener preference ratings of loudspeakers based on anechoic measurements. The model was tested using 70 different loudspeakers evaluated in 19 different listening tests. Its performance was compared to two models based on in-room measurements with 1/3-octave and 1/20-octave resolution, and two models based on sound power measurements, including the Consumers Union (CU) model, tested in Part One. The correlations between predicted and measured preference ratings were: 1.0 (our model), 0.91 (in-room, 1/20th-octave), 0.87 (sound power model), 0.75 (in-room, 1/3-octave), and -0.22 (CU model). Models based on sound power are less accurate because they ignore the qualities of the perceptually important direct and early reflected sounds. The premise of the CU model is that the sound power response of the loudspeaker should be flat, which we show is negatively correlated with preference rating. It is also based on 1/3-octave measurements that are shown to produce less accurate predictions of sound quality.
Convention Paper 6190

2:30 pm

C-4 Compact Magnetic Suspension Transducer—*Kenneth Kantor, Ioannis Kanellakopoulos, Ali Jabbari*, Tymphany Corporation, Cupertino, CA, USA

The role of compliant parts in the operation of loudspeaker drivers is discussed, and a new method of construction employing a magnetic suspension system is presented. Audio transducers require a complex interaction between moving and non-moving structures, placing conflicting demands on the compliant parts typically employed to interface between them. The limitations of current materials and of manufacturing technology, suggest that replacing flexible and compliant mechanical parts with a system based on magnetic forces might yield several benefits. Such a system, which utilizes a moving magnet balanced between static repulsive forces, is discussed conceptually, analytically and experimentally. Proposed advantages include increased linear excursion, convenient form-factor, reduced wear and fatigue, and the simplification of certain production processes.
Convention Paper 6191

3:00 pm

C-5 Comparative Analysis of Nonlinear Distortion in Compression Drivers and Horns—*Alexander Voishvillo*, JBL Professional, Northridge, CA, USA

Nonlinear effects in horn drivers are the inseparable part of the principle of their operation. In addition to the distortion caused by electrodynamic and mechanical effects, the distortion is generated in the compression chamber by the nonlinear adiabatic compression, modulation of the air's mechanical stiffness, mass, and viscous losses, and by the nonlinear relationship between the particle velocity and the sound pressure. A new more accurate nonlinear model of a compression chamber has been developed. A significant part of distortion is generated in the phasing plug and the horn due to the nonlinear propagation of the high pressure sound waves. Quantitative comparison of the nonlinear effects in a compression chamber and horn is carried out. The comparison is performed by using such criteria as harmonic distortion and two-tone intermodulation distortion.
Convention Paper 6192

3:30 pm

C-6 Maximum Efficiency of Compression Drivers—*D. B. (Don) Keele, Jr.*, Harman International, Northridge, CA, USA

Small-signal calculations show that the maximum nominal efficiency of a horn loudspeaker compression driver is 50 percent and the maximum true efficiency is 100 percent. Maximum efficiency occurs at the driver's resonance frequency. In the absence of driver mechanical losses, the maximum nominal efficiency occurs when the reflected acoustic load resistance equals the driver's voice-coil resistance. The maximum true efficiency occurs when the reflected acoustic load resistance is much higher than the driver's voice-coil resistance. To maximize the driver's broad-band true efficiency, the Bl force factor must be increased as much as possible, while jointly reducing moving mass, voice-coil inductance, mechanical losses, and front air chamber volume. Higher compression ratios will raise high-frequency efficiency but may decrease mid-band efficiency. This paper will explore in detail the efficiency and design implications of both the nominal and true efficiency relationships including gain-bandwidth tradeoffs.
Convention Paper 6193

PAPERS

4:00 pm

C-7 Analysis and Modeling of the Bi-Directional Fluid Flow in Loudspeaker Ports—*Zachary Rapoport, Allan Devantier*, Harman International, Northridge, CA, USA

Bass reflex ports are used in loudspeakers to enhance low frequency performance. At low sound levels the port extends the low frequency response by supplying one of the components of a Helmholtz resonator. At higher sound levels the turbulent intensity in the port increases disrupting the Helmholtz resonance causing distortion, noise, and compression. Although there has been significant work done to reduce these negative effects, no optimal solution has been found. To better understand the flow phenomena within the port, Computational Fluid Dynamics (CFD) was used to model the flow. The flow was simulated for six port profiles over a wide range of sound levels. In order to correlate the results of the CFD work to the real world, the same six ports were prototyped and subjected to several objective and subjective tests.
Convention Paper 6194

4:30 pm

C-8 Comparison of Inverse Filter Real-Time Equalization Methods for Nonminimum Phase Loudspeaker Systems—*Avelino Marques*,[1] *Diamantino Freitas*[2]
[1]Polytechnic Institute of Engineering of Porto (ISEP), Porto, Portugal
[2]Faculty of Engineering of Porto (FEUP), Porto, Portugal

Three time domain digital inverse filter design techniques are considered for nonminimum phase loudspeaker systems equalization, namely: FIR filter obtained with adjustable modeling delay, IIR filter followed by "excess-phase" compensation, and warped filter also followed by "excess-phase" compensation. Off-line inverse filtering results using real measured impulse responses of loudspeaker systems are compared and discussed for each design technique on the basis of the time equalization error, similar response's magnitude flatness, phase linearity and filter order. Real-time inverse filter implementation requirements on a real set-up, using a digital signal processor of the Texas Instruments TMS320 family are also compared based on computational load and memory needs. Results show that loudspeaker equalization with an inverse IIR filter followed by "excess-phase" compensation appears as a good compromise solution.
Convention Paper 6195

Session D **Thursday, October 28** **1:00 pm– 5:00 pm**
Room 302

PAPERS

LOSSY AND LOSSLESS AUDIO CODING, PART 2

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

1:00 pm

D-1 Introduction to Dolby Digital Plus, an Enhancement to the Dolby Digital Coding System—*Louis D. Fielder, Robert L. Anderson, Brett G. Crockett, Grant A. Davidson, Mark F. Davis, Stephen C. Turner, Mark S. Vinton, Phillip A. Williams,* Dolby Laboratories, San Francisco, CA, USA

An extension to the existing Dolby Digital (AC-3) multichannel audio coding standard is described and its new capabilities explored. This coding system is designed with extensive compatibility with the existing standard by retaining most of its metadata and data-framing structure to preserve and extend functionality in existing applications. New features include simultaneous support for multiple program streams, carriage of multichannel signals beyond 5.1 channels, and fine-grained control and support for data rates up to 6.144 Mbps. New coding "tools" including spectral extension, enhanced channel coupling, transient pre-noise processing, and improved filterbank / quantization enhance the performance of earlier AC-3 technology.
Convention Paper 6196

1:30 pm

D-2 Ultra Low Delay Audio Coding with Constant Bit Rate—*Ulrich Krämer, Gerald Schuller, Stefan Wabnik, Juliane Klier, Jens Hirschfeld*, Fraunhofer Institute for Digital Media Technology, Ilmenau, Germany

The Ultra Low Delay (ULD) codec developed at the Fraunhofer IDMT is based on a versatile perceptual audio coding method that achieves very low encoding/decoding delay and is nevertheless capable of high compression ratios. Utilizing a perceptual model for irrelevance reduction, the ULD codec is in principle a variable bit rate codec. To achieve coding with constant bit rate, the use of bit reservoir techniques would result in additional coding delay. This paper presents a rate loop, which ensures constant bit rate coding without increasing coding delay. It is shown that this technique does not decrease the decoded audio quality significantly.
Convention Paper 6197

PAPERS

2:00 pm

D-3 An Analysis of Tandem Error During Audio Transcoding—*Jun Wei Lee,*[1] *Aweke Lemma,*[2] *Michiel van der Veen*[2]
[1]National University of Singapore, Singapore
[2]Royal Philips Electronics, Eindhoven, The Netherlands

In a repeated quantization scenario, apart from the nominal quantization error, an additional tandem error is introduced. The amount of the excess tandem error depends on the characteristics of the quantizers used. In this work, the effect of tandem error and its dependence on the underlying quantizer characteristics are analyzed. A prime example where tandem error leads to increased noise is in audio transcoding. The behavior of tandem-noise for typical audio coding methods such as MPEG 1 Layer 2 is investigated. A method of reducing the tandem errors is proposed. This method involves guiding the quantization process of the first quantizer, assuming prior knowledge of the second quantizer. Results show that the method is able to reduce the amount of tandem error in the repeated quantization scenario.
Convention Paper 6198

2:30 pm

D-4 aacPlus, Only a Low-Bit-Rate Codec?—*Andreas Ehret,*[1] *Kristofer Kjörling,*[2] *Jonas Rödén,*[2] *Heiko Purnhagen,*[1] *Holger Hörich*[1]
[1]Coding Technologies, Nuremberg, Germany;
[2]Coding Technologies, Stockholm, Sweden;

aacPlus, the combination of the well known MPEG AAC and the Spectral Band Replication tool SBR has been introduced as a highly efficient low bit-rate audio codec, representing today's state-of-the-art by providing full bandwidth, near CD audio quality at 48 kbit/s stereo. It is thus suited for applications that demand the highest compression ratios. This paper discusses benefits when using aacPlus at moderate compression ratios in the range of 80 to 128 kbit/s stereo, where so far AAC was the codec of choice. The technological approach for applying SBR in such a scenario is described and subjective evaluations of the presented solution as well as an overview on system and implementation aspects are given.
Convention Paper 6199

3:00 pm

D-5 Efficient Bit Reservoir Design for MP3 and AAC—
Chi-Min Liu,[1] *Li-Wei Chen,*[1] *Ming-Ton Su,*[1] *Wen-Chieh Lee,*[1] *Chung-Han Yang,*[1] *You-Hua Hsiao,*[1] *Zheng-Wen*

Li,[2] *Chu-Ting Chien*[3]
[1]National Chiao-Tung University, Hsin-Chu, Taiwan
[2]InterVideo Digital Technology (Shanghai) Co., Ltd., Shanghai, China
[3]InterVideo Digital Technology, Taipei, Taiwan

Bit reservoir controlling the bits budget among music frames has been the kernel module to have good bit-quality tradeoff in current audio encoders like MP3 and AAC. The approaches of bit reservoirs can be investigated from demand-driven approach and budget-driven one. The demand-driven approach determines the required bits according to the audio contents while the budget-driven one allocates bits according to the bit budgets accumulated in the bit reservoir. Existing bit reservoirs follow basically the budget-driven approach. This paper presents an efficient bit reservoir design with concerns from both demand and budget. The bit reservoir includes a demand estimator to adaptively predict the bits required for each frame. Also, the bit reservoir has a budget regulator to control the bits used according to the codec protocol and the preferred scenario. The new bit reservoir method is included in MP3 and AAC to verify the efficiency through extensive objective and subjective tests.
Convention Paper 6200

3:30 pm

D-6 Design of MPEG-4 AAC Encoder—*Chi-Min Liu,*[1] *Wen-Chieh Lee,*[1] *Chung-Han Yang,*[1] *Kang-Yan Peng,*[1] *Ting Chiou,*[1] *Tzu-Wen Chang,*[1] *Yu-Hua Hsiao*[1]*, Hen-Wen Hue,*[1] *Chu-Ting Chien*[2]
[1]National Chiao-Tung University, Hsin-Chu, Taiwan
[2]InterVideo Digital Technology, Taipei, Taiwan

The state-of-the-art natural audio coder, MPEG-4 AAC, has provided extensive coding modules for achieving high coding efficiency. The modules, which include filter bank, window switch, psychoacoustic model, bit allocation, bit reservoir, lossless coding, temporal noise shaping, and middle/side coding, spans a large design dimension and creates challenges for audio coding technology. In this paper the design of these modules, named as NCTUAAC encoder, is presented to provide adequate audio quality with low computation complexity.
Convention Paper 6201

Session Z1 Thursday, October 28 2:00 pm – 4:00 pm
Outside 301/302

POSTERS: MUSIC SYNTHESIS AND AUDIO ARCHIVING, STORAGE AND RESTORATION; CONTENT MANAGEMENT

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation

2:00 pm

Z1-1 Frequency-Domain Additive Synthesis with an Oversampled Weighted Overlap-Add Filterbank for a Portable Low-Power MIDI Synthesizer—*King Tam*, Dspfactory Ltd., Waterloo, Ontario, Canada

This paper discusses a hybrid audio synthesis method employing both additive synthesis and DPCM audio playback, and the implementation of a miniature synthesizer system that accepts MIDI as an input format. Additive synthesis is performed in the frequency domain using a weighted overlap-add filterbank, providing efficiency gains compared to previously known methods. The synthesizer system is implemented on an ultra-miniature, low-power, reconfigurable application specific digital signal processing platform. This low-resource MIDI synthesizer is suitable for portable, low-power devices such as mobile telephones and other portable communication devices. Several issues related to the additive synthesis method, DPCM codec design, and system tradeoffs are discussed.
Convention Paper 6202

2:00 pm

Z1-2 Virtual Air Guitar—*Matti Karjalainen, Teemu Mäki-Patola, Aki Kanerva, Antti Huovilainen, Pekka Jänis*, Helsinki University of Technology, Espoo, Finland

A combination of hand-held controls and a guitar synthesizer is called the "Virtual Air Guitar." The name refers to playing an "air guitar," i.e., just acting the playing with music playback; the term virtual refers to making a playable synthetic instrument. Sensing of the left hand position is used for pitch control, the right hand movements for plucking, and the finger positions of both hands for the other features of sound production. The synthetic guitar algorithm supports electric as well as acoustic sounds, augmented with sound effects and intelligent mapping from playing gestures to synthesis parameters. The realization of the virtual instrument is described, and sound demonstrations are made available.
Convention Paper 6203

2:00 pm

Z1-3 Visually Controlled Synthesis Using the Sonic Scanner and the Graphonic Interface—*Dan Overholt*, University of California at Santa Barbara, Santa Barbara, CA, USA, Studio for Electro-Instrumental Music, Amsterdam, The Netherlands

This paper describes the concepts, design, implementation, and evaluation of two new interfaces for music performance and composition, and their control of various syntheses algorithms through the visual domain. Both of the interfaces were inspired by the idea of generating music through drawing, but they approach the activity in very different ways. While the Graphonic Interface allows you to make music as you are drawing, the Sonic Scanner needs precomposed graphic material in order to make music. However, both of the devices are real-time controllers that produce sound in an interactive manner, thereby allowing them to be used as performance instruments.
Convention Paper 6204

2:00 pm

Z1-4 Mana, a Tool for Human-Supervised Statistical Analysis in Audio-Content Extraction—*Nicolas Wack*, Pompeu Fabra University, Barcelona, Spain

Nowadays, metadata and audio descriptors extraction (used in classification, for instance) is engaged in a rather blind and brute-force method, computing the most possible and then selecting what works using an often long and boring statistical analysis. Moreover, this analysis barely takes into account the intrinsic sense these descriptors may carry. Mana is a graphical user-interface (GUI) based system that aims at adding a bit of human supervision in the process, combining state-of-the-art classification methods (in audio-content extraction and classification) with an ease of use that provides the user with direct control over descriptors and their significance in classification.
Convention Paper 6205

2:00 pm

Z1-5 Nonlinear Projection Algorithm for Evaluating Multiple Listener Equalization Performance—*Sunil Bharitkar, Chris Kyriakakis*, Audyssey Laboratories, Inc., Los Angeles, CA, USA

In this paper we present an application of a multidimension to two-dimension projection algorithm for visualizing room equalization at multiple locations. The algorithm allows easy visualization of the formation of clusters for

our previously presented pattern recognition based multiple listener room equalization filter. Furthermore, the mapping provides an interesting perspective on the formation of room response clusters. We also compare the results obtained from using the proposed map to the results obtained by using the spectral deviation measure.
Convention Paper 6206

Session E **Friday, October 29** **9:00 am – 11:30 am**
Room 301

AUDIO (INCLUDING TELEPHONY) OVER NETWORKS

Chair: **Karlheinz Brandenburg**, Fraunhofer Institute for Media Technology, Ilmenau, Germany

9:00 am

E-1 An RTP Payload Format for MIDI—*John Lazzaro, John Wawrzynek*, University of California, Berkeley, CA, USA

The Real-Time Protocol (RTP) is an extensible transport for sending media streams over Internet Protocol packet networks. We describe a new payload format that extends RTP to transport MIDI (the Musical Instrument Digital Interface command language). The payload format encodes all commands that may legally appear on a MIDI 1.0 DIN cable. The format is suitable for interactive applications (such as the remote operation of musical instruments) and content-delivery applications (such as file streaming). The format may be used over lossy unicast and multicast networks, and defines tools for graceful recovery from packet loss to support use over lossy unicast and multicast networks (including wireless networks). Stream behavior, including the MIDI rendering method, may be specified during session setup. Rendering methods are specified using the extensible Multipurpose Internet Mail Extensions (MIME) registry.
Convention Paper 6207

9:30 am

E-2 Network Time Delay and Ensemble Accuracy: Effects of Latency, Asymmetry—*Chris Chafe, Michael Gurevich*, Stanford University, Stanford, CA, USA

Pairs of musicians were placed apart in isolated rooms and asked to clap a rhythm together. Each person monitored the other's sound via headphones and microphone pickup, which was as close as possible. Time delay from source to listener was manipulated across trials. Trials were recorded and clap onset times were measured with an event detection algorithm. Longer delays produced increasingly severe tempo deceleration, and shorter delays (< 11.5 ms) produced a modest, but surprising, acceleration. The paper's goal is to characterize effects of delay on rhythmic accuracy and identify the region most conducive to ensemble playing. The results have implication for networked musical performance. Network delay is a function of transmission distance and/or internetworking

(routing) delays. The findings suggest that sensitive ensemble performance can be supported over rather long paths (e.g., San Francisco to Denver at about 20 ms, one-way). The finding that moderate amounts of delay are beneficial to tempo stability seems, at first glance, counterintuitive. We discuss the observed effect.
Convention Paper 6208

10:00 am

E-3 Practical Issues in Objective Speech Quality Assessment with ITU-T P.862—*Ville-Veikko Mattila,*[1] *Antti Kurittu*[2]
[1]Nokia Research Center, Tampere, Finland
[2]Nokia Networks, Helsinki, Finland

The ITU-T P.862 Recommendation specifies the Perceptual Evaluation of Speech Quality (PESQ) algorithm that is the current industrial standard for the objective, intrusive assessment of the one-way speech quality of narrowband networks and speech codecs. The practical use of P.862, however, has raised several questions about the robustness, applicability, and accuracy of the algorithm. The current paper presents results from an investigation of these issues. The characteristics of test signals and the interferences of signal interfaces are shown to have a significant effect on the quality assessment with P.862. A measurement procedure is proposed to define the accuracy of P.862 in the comparison of different or unknown technologies. It is concluded that various test factors should be carefully defined so that different P.862 measurements are comparable.
Convention Paper 6209

10:30 am

E-4 Integrated High-Performance Multichannel Audio Interconnection—*Michael Page, Gary Cook, Peter Eastty, Richard Marshall*, Sony Pro-Audio Lab, Oxford, UK

This paper describes a family of related technologies that provide a very high-performance audio interconnection system for professional applications. The first element is an advanced point-to-point audio interconnection based on a 100 Megabit Ethernet physical layer, which is currently undergoing AES standardization. The second element is a complementary gigabit-based high-capacity interconnection for applications such as backbone links. These are all linked together with a router technology that provides both low-latency audio channel routing and packet-switched control data routing. These technologies together provide a flexible, high-bandwidth digital audio infrastructure, which is ideally suited for applications

requiring low, deterministic latency and high reliability.
Convention Paper 6210

11:00 am

E-5 An Internet Protocol (IP) Sound System—*Tom Blank, Bob Atkinson, Michael Isard, James D. Johnston, Kirk Olynyk*, Microsoft Corporation, Redmond, WA, USA

We describe a system that applies Internet concepts and software techniques to deliver audio from source to loudspeakers using common computing hardware. The techniques overcome clocking and jitter problems. Microphones built into each transducer to locate loudspeakers allow the system to identify speaker placement, automatically compensate for off-center listening locations, adjust for inter-channel gain, delay, and do frequency response matching. A research prototype demonstrates the concepts and measures the resulting quality.
Convention Paper 6211

P A P E R S

Session F **Friday, October 29** **9:00 am – 11:00 am**
Room 302

AUDIO ARCHIVING, STORAGE, AND RESTORATION; CONTENT MANAGEMENT

Chair: **David Ackerman**, Harvard University, Cambridge, MA, USA

9:00 am

F-1 Wow Detection and Compensation Employing Spectral Processing of Audio—*Andrzej Czyzewski, Przemyslaw Maziewski, Marek Dziubinski, Andrzej Kaczmarek, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

The engineered algorithms are presented for the detection of parasitic frequency modulation in audio originating from irregularities of sound carrier velocity. The algorithms were developed with special regard to nonperiodic frequency modulation effects found in old movie sound tracks. The proposed algorithms consider the influence of the wow disturbance on the location of formants in time-frequency representation. The dynamic analysis of formant structure behavior underlies discriminating between parasitic frequency changes and natural frequency fluctuations. The compensation of the detected wow-related frequency modulation is accomplished basing on the nonuniform resampling algorithm, driven by the discerned parasite modulation patterns. The details of the proposed wow detection and compensation techniques are presented and achieved results are discussed.
Convention Paper 6212

9:30 am

F-2 Correction of Wow and Flutter Effects in Analog Tape Transfers—*Jamie Howarth*,[1] *Patrick Wolfe*[2]
[1]Plangent Processes, Nantucket, MA, USA
[2]University of Cambridge, Cambridge, UK

In this paper we describe a system whereby analog hardware is combined with the theory of nonuniform sampling in order to correct for wow and flutter effects in analog tape transfers. We show how, in certain instances, the medium itself can provide an accurate measurement of a recording's timing irregularities, in which case digital signal processing techniques permit a playback-rate correction of what is essentially an irregularly sampled audio waveform. Results using both real and synthetic data demonstrate the effectiveness of the method, both in cases of severe

degradation as well as high-quality analog transfers heretofore considered normal.
Convention Paper 6213

10:00 am

F-3 Computationally Efficient Blind Dereverberation of Audio Signals—*Hesu Huang, Chris Kyriakakis*, University of Southern California, Los Angeles, CA, USA

Convolutive noise in the form of reverberation can significantly degrade the quality and intelligibility of the real-world audio recordings. To reduce this type of acoustic noise, we propose a single-microphone dereverberation method based on Constant Modulus Algorithm (CMA)—a blind deconvolution technique. In particular, a new Non-causal Delayless Subband Filtering architecture is designed and combined with CMA to reduce the computational complexity of the overall system. Experimental results show that our method presents a comparable performance to full-band CMA but with less computational cost in dereverberating audio signals.
Convention Paper 6214

10:30 am

F-4 Audio Fingerprints: Technology and Applications—
Douglas Keislar, Erling Wold, Thom Blum, Muscle Fish, a division of Audible Magic Corporation, Berkeley, CA, USA

Audio fingerprinting provides the base technology for a variety of recent applications. Aspects of the underlying fingerprinting technology and some typical applications are presented. An effort is made to discuss both Audible Magic's work as well as that of its competitors.
Convention Paper 6215

PAPERS

Session Z2 Friday, October 29 9:00 am – 11:00 am
Outside 301/302

POSTERS: INSTRUMENTATION AND MEASUREMENT AND LOSSY AND LOSSLESS AUDIO CODING

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation.

9:00 am

Z2-1 A New Digital Measurement for Distortion of Acoustical Devices—*Keiichi Imaoka, Juro Ohga*, Shibaura Institute of Technology, Minato-ku, Tokyo, Japan

There is still no suitable measuring method by a digital processing system for nonlinear distortion of acoustical devices. This paper presents a new amplitude nonlinearity measurement by using a Pink-TSP (time stretched pulse) signal. This method applies a TSP signal, whose frequency band is partially eliminated, to the device. The detected component produced in the rejected band is measured as a distortion.
Convention Paper 6216

9:00 am

Z2-2 Measurement of Small-Size Loudspeaker Units by New Acoustical Loads—*Yusuke Nakano, Juro Ohga*, Shibaura Institute of Technology, Minato-ku City, Tokyo, Japan

This paper describes a new measuring method for small sized loudspeakers by using a tube load. The acoustical loads defined in IEC standards for loudspeaker measurement, both of closed boxes and a baffle, are larger in size than the practical acoustical loads for small loudspeakers, for example, mobile telephone bodies. This paper proposes a tube load for measurement and examines practical methods without any effect by tube resonance.
Convention Paper 6217

9:00 am

Z2-3 Extending Quasi-Anechoic Measurements to Low Frequencies—*Eric Benjamin*, Dolby Laboratories, San Francisco, CA, USA

It is often desirable to make electroacoustic measurements in ordinary working spaces. These measurements would normally be performed in anechoic chambers. Various techniques have been evolved to make what are commonly referred to as “quasi-anechoic” measurements.

These techniques make use of the fact that the initial signal from a loudspeaker-microphone system is anechoic, until the first reflection arrives. By analyzing only that portion of the signal which arrives before the first echo, an anechoic measurement is achieved. These measurements have a low-frequency limitation due to the shortness of the reflection-free time window. Time-frequency tradeoffs in the transformation of the Impulse Response to the frequency domain make it difficult to accurately estimate the response of the device under test. We first characterize the nature of the errors induced by the short time window and then propose a methodology for reducing the error.
Convention Paper 6218

9:00 am

Z2-4 Efficient AAC Single Layer Transcoder—*Chun-Yi Lee, Cheng-Han Yang, Te-Hsueh Lai, Tihao Chiang, Hsueh-Ming Hang*, National Chiao Tung University, Hsinchu, Taiwan

This paper presents a novel algorithm for transcoding the MPEG-4 AAC single-layer bit-streams for bit-rate adaptation purposes. The delivery of multimedia over heterogeneous networks and to the devices with varying capabilities calls for the bit-rate adaptation capability. A previous approach that cascades a pair of full-grown decoder and encoder has very high computational complexity. Our approach can reduce the complexity drastically; however, its coding performance is close to that of the previous cascaded method. In order to achieve this simplification goal, three rate-distortion models/techniques have been employed.
Convention Paper 6219

9:00 am

Z2-5 Effective Tonality Detection Algorithm Based on Spectrum Energy in Perceptual Audio Coder—
Keun-Sup Lee,[1] *Kyu-Chel Yeon,*[2] *Young-Cheol Park,*[1] *Dae Hee Youn*[1]
[1]Yonsei University, Seoul, Korea
[2]LG Electronics Inc., Anyang, Korea

The goal of the perceptual audio coder is to reduce redundancy and irrelevancy of audio signals based on the concept of masking. Several studies on the masking effect reveal that the masking threshold varies as a function of the noise-like or tone-like nature of audio signals. Therefore, the tonality of audio signals significantly influences the quality and efficiency of the perceptual audio coder. In this paper, we proposed a new effective algorithm for tonality measurements using spectrum energy. The perfor-

P A P E R S

mance of the proposed algorithm is comparable to the MPEG audio psychoacoustic model II (PAM-II). However, since the proposed algorithm consists of simple operations plus a few transcendental functions, computational complexity is much lower than the PAM-II. The proposed algorithm was tested with audio signals. DSP implementation showed that the proposed algorithm could be implemented with 2.88 MIPS.
Convention Paper 6220

9:00 am

Z2-6 Audio Patch Method in MPEG-4 HE-AAC Decoder
—*Han-Wen Hsu*,[1] *Chi-Min Liu*,[1] *Wen-Chieh Lee*,[1] *Zheng-Wen Li*[2]
[1]National Chiao Tung University, Hsin-Chu, Taiwan
[2]InterVideo Digital Technology (Shanghai) Co., Ltd., Shanghai, China

This paper extends the previous work on AAC to the HE-AAC. The audio path method consists of two individual parts, zero band dithering and high frequency reconstruction. The zero band dithering can conceal the fishy artifact in the low frequency part that is encoded by a convention AAC encoder. Furthermore, high frequency reconstruction can extend the audio obtained from the SBR to a full bandwidth signal. Intensive experiments have been conducted on various encoders and audio tracks to check the quality improvement and the possible risks in degrading the quality. The objective test measure used is the recommendation system by ITU-R Task Group 10/4.
Convention Paper 6221

9:00 am

Z2-7 Bit-Weighted Inter-Channel Prediction for Subband Audio Coding—*Cheng-Han Yang, Hsueh-Ming Hang*, National Chiao Tung University, Hsinchu, Taiwan

An efficient algorithm for removing interchannel redundancy in subband audio coding is presented in this paper. In our approach, the bit-weighted interchannel prediction is applied to the Modified Discrete Cosine Transform (MDCT) coefficients. Similar to the INT-DCT-based approach, no audio quality degradation will be induced by our method. In addition, the bit rate reduction performance of our method is about 8 percent better than that of the INT-DCT-based approach for the cases where interchannel prediction is useful.
Convention Paper 6222

Session G **Friday, October 29** **11:30 am – 12:30 pm**
Room 302

AUDIO FOR COMPUTER GAMES

Chair: **Martin Wilde**, Motorola, Inc., Schaumburg, IL, USA

11:30 am

G-1 Establishing a Reference Playback Level for Video Games—*Mark Tuffy*, THX Ltd., San Rafael, CA, USA

Over the last two decades, there have been dramatic advances in video game technology. At this time, audio for games has moved from monophonic beeps to full 5.1 surround sound, utilizing Dolby Digital and DTS. While the games industry has embraced these technologies, there are no standards or guidelines in place to ensure that game audio exploits the potential of this delivery mechanism. As a result, there is still the push toward "louder is better." One element key to moving away from "loud" to "quality" is establishing a reference level for playback. This paper suggests such a reference level and why this would be logical for the games industry to adopt.
Convention Paper 6223

12:00 noon

G-2 Can Playing a Computer Game Affect Perception of Audio-Visual Synchrony?—*Peter Ward, Slawomir Zielinski, Francis Rumsey*, University of Surrey, Guildford, Surrey, UK

The investigation aimed to discover the effect of involvement in an interactive task on the perception of audio-visual asynchrony in a computer game environment. An experimental game was designed to test the investigated phenomenon. The experiment tested only audio lag conditions. It was found that within the confines of the experimental method, the threshold of perception was increased in the interactive game condition by approximately 40 ms (±20 ms), which is a small but statistically significant value.
Convention Paper 6224

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Session H **Friday, October 29** **2:00 pm – 6:00 pm**
Room 304

MULTICHANNEL SOUND

Chair: **Geoff Martin**, Bang & Olufsen A/S, Struer, Denmark

2:00 pm

H-1 Customization for Personalized Rendering of Motion-Tracked Binaural Sound—*Joshua Melick, V. Ralph Algazi, Richard Duda, Dennis Thompson*, University of California, Davis, CA, USA

Motion-tracked binaural, or MTB, recordings preserve the dynamic sound localization cues provided by voluntary head motion, making MTB less sensitive than other binaural methods to mismatches between the HRTF of the listener and the HRTF of the recording system. However, MTB performance can be improved by customizing the reproduction process to the listener. In this paper the different types of mismatch and their perceptual consequences are identified. Techniques are presented for partially or completely correcting the mismatches, and properties of these techniques are described.
Convention Paper 6225

2:30 pm

H-2 5.1 and 22.2 Channel Multichannel Sound Productions Using an Integrated Surround Sound Panning System—*Kimio Hamasaki*,[1, 2] , *Setsu Komiyama*,[1] *Hiroyuki Okubo*,[1] *Koichiro Hiyama*,[1] Wataru Hatano[3]
[1]NHK Science & Technical Research Laboratories, Tokyo, Japan
[2]Kyushu University, Fukuoka, Japan
[3]Tamura Corporation, Tokyo, Japan

5.1 surround sound productions and broadcasts have become popular in Japan, and NHK has developed 22.2 multichannel sound system for ultra-high definition video. While two-dimensional or three-dimensional sound reproduction is possible by these multichannel sound systems, the production of contents is more complicated and time consuming than two-channel stereo production. In productions using a conventional surround sound mixing tool, in particular, much time is needed for creating two-dimensional sound effects. Therefore, an integrated surround panning system was developed to enable various surround sound effects to be created easily. This paper introduces the newly developed integrated surround sound panning system, which has innovative functions such as a

distance control and an integrated sound source movement control, and discusses various issues on multichannel sound production.
Convention Paper 6226

3:00 pm

H-3 Examination of Multichannel Sound Field Recomposition Utilizing Frequency-Dependent Interaural Cross Correlation (FIACC)—*Teruo Muraoka, Tomoaki Nakazato, Masaki Ichikawa*, Musashi Institute of Technology, Tokyo, Japan

Locations of loudspeakers were examined utilizing frequency dependent interaural cross correlation (FIACC) for optimum sound field recomposition in the multichannel recording and reproducing process. Experiments were conducted by comparing pairs of FIACC, where one of a pair was measured in an original sound field, and the other was measured in a reproduced sound field. Conclusively, it became clear that ITU's recommendation to the loudspeaker arrangement in 5 channel system is reasonable.
Convention Paper 6227

3:30 pm

H-4 Modeling Auditory Localization of Subwoofer Signals in Multichannel Loudspeaker Arrays—*Jonas Braasch, William L. Martens, Wieslaw Woszczyk*, McGill University, Montreal, Quebec, Canada

For economical reasons, home entertainment surround sound systems are usually equipped with a single subwoofer channel. The main argument for this procedure is the believed inability of the auditory system to localize low frequencies in small reverberant rooms. However, a psychoacoustic localization test that was conducted using a standard 5-channel set-up with subwoofers showed that the listeners were able to determine the lateral displacement left, center, or right of the loudspeaker presenting the test stimulus (an octave-band noise burst at 31.5-Hz, 63-Hz or 125-Hz center frequency). Using a binaural model simulating human perception, recordings of subwoofer signals at different positions were analyzed. As expected, the interaural level differences remained nearly constant for different subwoofer positions in the low frequency range. On the basis of interaural time differences, however, the model was able to predict the position of the loudspeaker regarding the left/right dimension, verifying the outcome of the listening test. The results indicate the importance of considering more than one subwoofer in multichannel audio systems.
Convention Paper 6228

4:00 pm

H-5 Identification and Discrimination of Listener Envelopment Percepts Associated with Multiple Low-Frequency Signals in Multichannel Sound Reproduction—*William Martens, Jonas Braasch, Wieslaw Woszczyk*, McGill University, Montreal, Quebec, Canada

Multichannel surround sound systems that meet the 5.1 channel ITU standard may use a single subwoofer to reproduce signals below 120 Hz. Yet the ITU standard allows for presentation of five different full-range signals to the listener, each extending to a low-frequency limit of 20 Hz. Including multiple low-frequency signals in sound reproduction enables the creation of auditory spatial imagery that features variation in perceptual attributes such as auditory source width (ASW), auditory source distance (ASD), and listener envelopment (LEV). The experiments reported in this paper were designed to determine how identifiable and how discriminable the listener envelopment percepts are in the auditory images resulting from decorrelation between low-frequency signals presented via selected pairs of five loudspeakers positioned according to the ITU standard configuration.
Convention Paper 6229

4:30 pm

H-6 Multichannel Sound Recording Using 3-, 4-, and 5-Channel Arrays for Front Sound Stage Coverage —*Michael Williams*, Sounds of Scotland, Paris, France

In previous papers the Multichannel Microphone Array Design (MMAD) procedure has been used mainly to determine the design of arrays giving complete 360 degree coverage of the sound field. Many sound recording engineers, however, use the main microphone array to cover only the front sound stage and add in early reflections and reverberation either by artificial means (electronic generation) or by using a second array in the reverberation field. This paper describes MMAD procedure applied to only front coverage of the main sound stage using 3-, 4- or 5-channel microphones, covering any desired angle within the front hemisphere and for the usual 1st-order microphone directivities. Various array alignments are described in the form of the arc-of-a-circle with different radius. All arrays described are critically linked (seamless) within the front hemisphere.
Convention Paper 6230

5:00 pm

H-7 Designing High Spatial Resolution Microphones— *Arnaud Laborie, Rémy Bruno, Sébastien Montoya*, Trinnov Audio, Paris, France

Multichannel recording is one of the most important challenges of today's audio techniques. A good surround recording should provide good envelopment feeling, accurate localization, a large sweet spot, and respect for tones—all at the same time. The Fourier-Bessel theory and advanced signal processing allows you to obtain directivities designed from panning laws, which have been designed to optimally drive any multichannel layout. This paper presents the underlying concept of high spatial resolution, the spatial equivalent for high fidelity, and points out why this is a key point to achieve high spatial quality. A very flexible and scalable technology providing high spatial resolution, as well as a high-performance 5.0 microphone featuring a compact and light array of 16 omnidirectional capsules are also presented.
Convention Paper 6231

5:30 pm

H-8 Capturing Manipulation and Reproduction of Sampled Acoustic Impulse Responses—*Ronen Ben-Hador, Itai Neoran*, Waves Audio, Tel Aviv, Israel

We discuss the capturing manipulation and reproduction of impulse responses (IRs) of acoustic spaces. While trying to maintain the accuracy of an IR, other factors such as sound quality and musical character of sound should also be considered. Furthermore, IRs are not limited to preserving the sound of venues but also as a tool in music production. Therefore, the IRs are converted to standard multichannel reproduction formats, such as stereo and ITU 5.0. In order to obtain a flexible reverberation tool, the IRs are manipulated to modify acoustic properties such as reverb time and interchannel decorrelation. A new real-time audio plug-in was developed for which IRs of venues and devices were recorded worldwide. The IRs are convolved with dry audio. The plug-in supports mono, stereo, and surround, at sample-rates up to 96 kHz.
Convention Paper 6232

PAPERS

Session I **Friday, October 29** **1:30 pm – 6:00 pm**
Room 302

PSYCHOACOUSTICS, PERCEPTION, AND LISTENING TESTS

Chair: **Gilbert Soulodre**, Communications Research Centre, Ottowa, Ontario, Canada

1:30 pm

I-1 Preferred Listening Levels and Acceptance Windows for Dialog Reproduction in the Domestic Environment—*Eric Benjamin*, Dolby Laboratories, San Francisco, CA, USA

Audio reproduction systems have as their first goal the ability to reproduce the program sounds at the level desired by the user. There are a number of areas in audio system design where this is critical. The system should be able to reproduce audio at the levels needed by typical users, and presumably should also have the ability to accommodate a substantial portion of the desired reproduction levels. Beyond this simple requirement, some systems should have their behavior optimized for the level at which they will be used. The literature on listening levels has been surveyed, and new data has been gathered to determine what the preferred listening levels are for a variety of listening circumstances. Additional experiments have been done to estimate the range of listening levels that may be acceptable to the typical listener.
Convention Paper 6233

2:00 pm

I-2 Evaluation of Different Loudness Models with Music and Speech Material—*Esben Skovenborg*,[1, 2] *Søren H. Nielsen*[2]
[1]University of Aarhus, Århus, Denmark
[2]TC Electronic A/S, Risskov, Denmark

The evaluation of twelve models of loudness perception is presented. One of the loudness models is based on a novel algorithm, and another is based on a combination of two known measurement techniques. The remaining models are all implementations of common or standardized loudness algorithms. The ability of each model to predict or measure the subjective loudness of speech and music segments is evaluated. The reference loudness is derived from two listening experiments using the speech and music segments as stimuli. Different statistical measures are employed in the evaluation of the models, so that both

the absolute performance of the models and the performance relative to the between-listener disagreement are measured.
Convention Paper 6234

2:30 pm

I-3 Level Detection Tunings and Techniques for the Dynamic Range Compression of Audio Signals—
Ryan Cassidy, Stanford University, Stanford, CA, USA

In a recent work, we reviewed the basics of level detection for dynamic range compression and presented various tunings of level detector parameters for optimal correspondence with well-known and recently proposed facts pertaining to loudness perception. In this paper we review key points and present several extensions. We review the mathematics behind the operation of the popular root-mean-square detector, with special attention paid to the effect of time constants on detector performance. We compare an equal loudness filter, designed in the prior work to compensate for frequency-dependent steady-state loudness perception, to standard weightings. Updated results based on newly standardized loudness contours are also presented.
Convention Paper 6235

3:00 pm

I-4 A New Objective Measure of Perceived Loudness—
Alan Seefeldt, Brett Crockett, Michael Smithers, Dolby Laboratories, San Francisco, CA, USA

The applications of an accurate objective measure of subjectively perceived loudness are many. Accordingly, the ITU-R has initiated a study to identify such a measure for a new recommendation. A new objective loudness measurement based on modifications to a traditional psychoacoustic model of perceived loudness was developed for this study. When compared to subjective loudness matching data generated outside the ITU-R, the new measure is found to perform better than simpler weighted power measurements and the unmodified psychoacoustic model.
Convention Paper 6236

3:30 pm

I-5 Training of Listeners for Evaluation of Spatial Attributes of Sound—*Juha Merimaa, Wolfgang Hess*, Ruhr-Universität Bochum, Bochum, Germany

A group of listeners were engaged in training to learn to evaluate auditory source width (ASW) and listener envel-

PAPERS

opment (LEV). The training consisted of discussions on the perception of spatial sound and visualization of both attributes with drawings. After each session the subjects evaluated a set of stimuli consisting of different source signals simulated in a few chosen acoustical environments. Most subjects developed consistent criteria for their judgments and maintained them throughout the training and a subsequent control experiment conducted two months later. However, considerable individual differences were found. Analysis of the data revealed that the large part of the differences was due to different judgments between the chosen source signals. The training also suggested that some differences could have been caused by the translation from the "spatial image" formed of the stimuli to the unidimensional judgments. A further experiment showed that this was not the case.
Convention Paper 6237

4:00 pm

I-6 Development of a Sound Quality Evaluation System—*Preben Kvist*,[1] *Carsten Thomsen*,[1] *Sangil Park*,[2] *Joonhyun Lee*,[2] *Finn Agerkvist*[3]
[1]DELTA Acoustics & Vibration, Hørsholm, Denmark
[2]Samsung Electronics, Suwon, Korea
[3]Technical University of Denmark, Lyngby, Denmark

This paper describes the development of the first version of the Sound Quality Evaluation System. The purpose of the system is to predict the subjective sound quality of home theater systems based on objective measurements. Sixteen home theater systems were measured in an anechoic room. Several metrics expected to correlate with the subjective quality were proposed and tested. A model for the sound quality was created by mapping the subjective evaluations of the home theater systems with the metrics calculated for each system. Correlation between the subjective listening test and the prediction is presented.
Convention Paper 6238

4:30 pm

I-7 The Effect of Room Acoustics on MP-3 Audio Quality Evaluation—*Daniël Schobben, Steven van de Par*, Philips Research Laboratories, Eindhoven, The Netherlands

The impact of using loudspeaker versus headphone playback on the subjective quality of compressed audio is investigated. It is shown that reverberation—and to a lesser extent cross-talk, which both are introduced naturally in loudspeaker playback—can effectively hide coding artifacts. In double blind listening tests subjects had to rate MP3 coded excerpts at various bit rates. The excerpts

were played back over headphones. Reverberation and cross-talk were introduced artificially to simulate loudspeaker playback so that their impact could be assessed separately. Results show that quality scores of the reverberated excerpts were significantly higher than for the corresponding "dry" excerpts for 64 kb/s bit rate while these differences diminished with increasing bit rate. This indicates that coding artifacts can become less audible in reverberant listening conditions.
Convention Paper 6239

5:00 pm

I-8 The Restitution of Timbre by Loudspeakers in a Listening Room: Perceptual and Physical Measurements—*Mathieu Lavandier, Philippe Herzog, Sabine Meunier*, Laboratoire de Mécanique et d'Acoustique, Marseille, France

This paper deals with the relationships between two parallel evaluations of a panel of loudspeakers: perceptual measurements and physical ones. The sound radiated by the loudspeakers has been recorded. The recordings were submitted to both listening tests and signal analysis. Pair-comparison tests were run using headphones, so the spatial dimension of sound reproduction is not investigated. This first attempt revealed two main perceptual dimensions. They were independent of the tested recording techniques and musical excerpts. We determined a suitable method of analysis for the physical measurements, and then we looked for objective attributes correlated with the perceptual dimensions.
Convention Paper 6240

5:30 pm

I-9 Subjective Test of Class D Amplifiers Without Output Filter—*Finn Agerkvist*,[1] *Lars Fenger*[2]
[1]Technical University of Denmark, Lyngby, Denmark
[2]Bang & Olufsen ICEpower a/s, Lyngby, Denmark

This paper presents subjective tests of the active transducers concept. The tests are designed to determine whether the output filter on class D amplifiers used in an active loudspeaker can be omitted without audible error occurring. The input signal of the amplifiers was limited to 0 to 3 kHz corresponding to that of a woofer unit. A listening panel of seven persons was used in the tests. The tests showed that no errors could be detected.
Convention Paper 6241

Session Z3 Friday, October 29 1:30 pm – 3:30 pm
Outside 301/302

PAPERS

POSTERS: LOUDSPEAKERS, PART 1

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation.

1:30 pm

Z3-1 Acoustic Response Simulation of a Cone-Type Loudspeaker by the Finite Element Method—*Noboru Kyouno*,[1] *Tsuyoshi Usagawa*,[1] *Tatsuo Yamabuchi*,[2] *Yukio Kagawa*[3]
[1]Kumamoto University, Kumamoto, Japan
[2]Toyama University, Toyama, Japan
[3]Akita Prefectural University, Akita, Japan

Acoustic responses of an axisymmetric cone-type loudspeaker mounted in an infinite baffle have been analyzed as an electro-mechano-acoustic transducer by applying the Finite Element Method; conical shell elements to the mechanical system and triangular ring elements to the acoustic system. The outer semi-infinite space where sound is emitted from the cone loudspeaker is treated analytically by applying the Green's function. The mechano-acoustic system of the cone loudspeaker is connected to the electrical system by an electro-mechanical equivalent circuit. The calculated sound pressure responses are compared with measured results, demonstrating that the calculated responses are very good predictors of measured results.
Convention Paper 6242

1:30 pm

Z3-2 Nonlinearities Characterization—*Delphine Bard*, Swiss Federal Institute of Technology, Lausanne, Switzerland

This paper presents transducer nonlinearity analysis in view of compensation of these effects. We describe an experimental method of weak nonlinearity characterization, based on a representation of the nonlinearity by Volterra series and using multitone excitations. Device linearization can be achieved by applying the inverse nonlinearity upstream of the device, on the condition that the nonlinearity law is known. To address the need to distinguish nonlinear effects from linear distortions, an ad hoc experimental method has been developed. The characterization of a weakly nonlinear electroacoustic device with usual methods of measurement (THD, intermodulation) does not illustrate the nonlinearities themselves, but only

some of their effects. This is the reason why this characterization method was developed.
Convention Paper 6243

1:30 pm

Z3-3 Numerical Analysis of Total Harmonic Distortion of a Loudspeaker in a Low-Frequency Range—*F. D. Zong, Z. L. Zhang*, ZheJiang Normal University, ZheJiang, China

Japanese expert Yoshinisa studied the total harmonic distortion in nonlinear phenomena of the loudspeaker in a low frequency range caused by nonlinear mechanic resilience, but he ignored the case of nonlinearity of mechanic resilience and magnetic field. The authors' work focuses on finding the total harmonic distortion of the nonlinear motion of the loudspeaker by means of numerical calculation. They obtained the numerical solution through numerical calculation using MATLAB software, and the corresponding curves about the total harmonic distortion versus frequency through spectrum analysis using SPECTRA PLUS software. They also analyzed the influence of nonlinearity of magnetic field on the total harmonic distortion of the loudspeaker, and drew several useful conclusions.
Convention Paper 6244

1:30 pm

Z3-4 Characteristics of Loudspeakers by a Multilayer Piezoelectric Ceramic—*Jun Fujii, Juro Ohga, Norikazu Sasida*, Shibaura Institute of Technology, Tokyo, Japan

This paper presents the construction and characteristics of a small size loudspeaker with a multilayer piezoelectric ceramic bimorph diaphragm for use with mobile telephones. The multilayer ceramic wafer is characterized by lower operation voltage due to larger capacitance than that of conventional ceramics. This is suitable for mobile equipment with battery operation. Precise measurement of diaphragm parameters and analysis of loudspeaker response are also described in this paper.
Convention Paper 6245

1:30 pm

Z3-5 Radiation of Enclosed Loudspeaker in Baffle: Simulation Model and Results—*Elena Prokofieva*, Linn Products, Glasgow, Scotland, UK

A theoretical investigation of the conventional loudspeaker, placed into a rigid wall, was presented at AES 116th Convention, Berlin, Germany (Convention Paper 6154).

The quasidynamic approach to the loading force and pressure was introduced there. The loudspeaker diaphragm was regarded as a number of concentric rings. The acoustic pressure and surface velocity are predicted using the first two approximations in that paper. These simulations show the difference between the real measurements and the analytical models, used with some standard assumptions from the classic theory of plates. The results suggest that the modeling process needs to become more specified. Hence, to simulate the behavior of a standard suspension, the outer, clamped ring was given the characteristics of rubber with nonlinear deformations. The 4th approximation of the problem is formulated in this paper.
Convention Paper 6246

1:30 pm

Z3-6 Jump Resonance in Audio Transducers—*Ali Jabbari, Andy Unruh*, Tymphany Corporation, Cupertino, CA, USA

The resonance behavior of a driver with low damping is studied. In such a system, the existing nonlinearities can result in jump resonance, a bifurcation phenomenon with two regimes. One regime, accompanied by a sudden decrease in amplitude, is evident when the frequency of excitation is increasing. The other regime, exhibiting a sudden increase in amplitude, is present when the frequency of excitation is decreasing. Jump resonance was experimentally observed in an audio transducer with low damping and subsequently confirmed by analysis and simulation using a detailed dynamic model that includes the most significant sources of nonlinearities. The conclusion of this paper is that the primary cause of jump resonance in audio transducers is the nonlinearity in the driver compliance. The importance of this phenomenon increases as the use of current amplifiers becomes more widespread, since the resulting low system damping makes jump resonance more likely.
Convention Paper 6247

Session Z4 Friday, October 29 4:00 pm – 6:00 pm
Outside 301/302

POSTERS: LOUDSPEAKERS, PART 2

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation.

4:00 pm

Z4-1 Impulse Response and Frequency Response of a Line Loudspeaker Array—*Chao Jiang, Jian Zou, Yong Shen*, Nanjing University, Nanjing, P. R. China

A uniform method is presented to calculate impulse response of an arbitrary point of sound field radiated by a line loudspeaker array. The frequency response is also obtained by applying FFT technology to the impulse response. It is shown that, in any point of a sound field, the frequency response is similar to a low-pass filter, and the cut off frequency varies with the position of the observation point.
Convention Paper 6248

4:00 pm

Z4-2 Real-Time Voice-Coil Temperature and Cone Displacement Control of Loudspeakers—*Constâncio Bortoni, Sidnei Noceti Filho, Rui Seara, Rosalfonso Bortoni*, Federal University of Santa Catarina, Brazil

With loudspeakers operating in a high power environment (common in PA systems), the voice-coil overheating and the excessive cone displacement are the main causes of damage and faults. These drawbacks are related to the low efficiency and cone displacement limitation, respectively. This paper proposes a procedure to measure and control both the voice-coil temperature and cone displacement by using a digital signal processor (DSP). The voice-coil temperature and cone displacement are indirectly obtained from the coil DC resistance variation and the cone acceleration, respectively. This approach takes into account (by measuring) some real characteristics of the loudspeaker, as its inherent nonlinearities. Thus, we can obtain the most from the sound system, since it may now work without the usual safety margin needed for these systems.
Convention paper 6249

4:00 pm

Z4-3 Loudspeaker Transducers with an Alternative Tubular Form Factor—*Andrew Unruh*[1], *Robert True*[2]
[1]Tymphany Corporation, Cupertino, CA, USA
[2]True Technologies, Inc. Pleasant Prairie, WI, USA

Three different multidiaphragm loudspeaker transducers with a tubular form factor are investigated. The transducers consist of a conventional motor structure, a tubular housing, and multiple diaphragms. In one design, sound is generated by the relative motion between a housing that is driven by a single motor and diaphragms that are attached to the housing via flexible surrounds. In a second design, a single motor drives one set of diaphragms and sound is generated by the relative motion between the diaphragms and the fixed housing. In the final design, two motors are used to drive two sets of diaphragms in opposition, and sound is generated by the relative motion between them.
Convention Paper 6250

4:00 pm

Z4-4 Higher Order Harmonic Signature Analysis for Loudspeaker Defect Detection—*Dan Foley*,[1] *Robert Celmer*,[2] *Benjamin Sachwald*,[2] *James Anthony*,[2] *Tony Pagliaro*,[2] *Shane Thompson*[2]
[1]Listen, Inc., Boston, MA, USA
[2]University of Hartford, West Hartford, CT, USA

Loudspeaker assembly faults, such as a rubbing voice coil, bent frame, loose spider, etc., have traditionally been detected using experienced human listeners at the end of a production line. Previous attempts to develop production measurement systems for on-line testing typically analyze only low-order harmonics for the primary purpose of measuring total harmonic distortion (THD), and thus are not specifically designed to detect defective rub, buzz, and ticking sounds. This paper describes a new method wherein the total energy of high-order harmonics groups, for example, 10th through the 20th or 31st through the 40th, are measured and analyzed. By grouping high-order harmonics and resolving their respective total energies, distinct signatures can be obtained that correlate to the root cause of audible rub and buzz distortions (Temme, 2000). The paper discusses loudspeakers tested with specific defects, as well as results of a computer-based electroacoustic measurement and analysis system used for detection.
Convention Paper 6251

Session J **Saturday, October 30** **9:00 am – 10:30 am**
Room 301

MICROPHONES

Chair: **David Josephson**, Josephson Engineering, San Jose, CA, USA

9:00 am

J-1 Long Range Noise Canceling Microphone—*Alexander Goldin*, Alango Ltd., Haifa, Israel

The paper presents Long Range Noise Canceling (LRNC) microphone array technology developed in Alango Ltd. LRNC is a digital signal processing technology utilizing near field signals from two unidirectional or four omnidirectional microphones. It allows differentiation between a user's voice originating in a closed region in front of LRNC microphone and other sounds that are effectively blocked. The pick up range of the LRNC microphone may be as large as 70 cm in front of the microphone and, if necessary, may be easily reduced by changing corresponding software parameters. This unique property makes the LRNC microphone attractive for a variety of voice applications where distant sounds, noises or acoustic echoes must be blocked.
Convention Paper 6252

9:30 am

J-2 Spatial Definition and the PanAmbiophone Microphone Array for 2-D Surround and 3-D Fully Periphonic Recording—*Robert (Robin) Miller III*, FilmakerTechnology, Bethlehem, PA, USA

Higher sampling rates are necessary for high spectral resolution, but it is higher angular resolution and precision that preserve source directionality and therefore higher tonal/timbral quality of that source, termed spatial definition. In acoustic spaces that are extensions of musical instruments, voices, and sources of sound effects (as for movies), tonality is a major contributor to lifelike perception. But in audio reproduction, lifelike tonality is limited by the recording system. A surround microphone has been developed both for more precise 2-D surround (PanAmbio), compatible with ITU 5.1, and for "PerAmbio" 3-D (with height) for the ultimate in tonal reality, distributable using ordinary 6-channel media for either decoderless 2-D replay or 3-D with decoder and five additional loudspeakers.
Convention Paper 6253

10:00 am

J-3 Advanced Simulation of a Condenser Microphone Capsule—*Roger Grinnip III*, Shure Incorporated, Niles, IL, USA

An advanced numerical model of a pressure condenser microphone capsule is presented. The model divides the acoustic space into internal and external domains and couples the dynamic pressure in each domain to the capsule diaphragm motion. The external acoustic space is modeled using the boundary element (BE) method, which allows for arbitrary geometry of the capsule/microphone external surface. The diaphragm is modeled as a circular tensioned membrane of negligible bending stiffness. The internal acoustic space (both the viscous air film and back chamber) is modeled as a cylindrical cavity with negligible axial pressure variation. Flow through the back plate is modeled as an annular array of circular pores with generalized functions locating each pore position. Although the presented model is specialized for a simple pressure condenser microphone, the numerical implementation is sufficiently generic to allow for a large variation in capsule parameters. The complete model, implemented in a software package called VC, is used to generate a simulated response curve that is compared to a response curve taken from an experimental prototype. The results show excellent agreement throughout the measured frequency range, indicating this new coupled model may be used for advanced microphone characterization and design.
Convention Paper 6254

Session K **Saturday, October 30** **9:00 am – 12:30 pm**
Room 302

PAPERS

SIGNAL PROCESSING, PART 1

Chair: **Duane Wise**, DSP Consultant, Boulder, CO, USA

9:00 am

K-1 New Balanced Input Integrated Circuit Achieves Very High Dynamic Range in Real-World Systems—*Bill Whitlock*[1]*, Fred Floru*[2]
[1]Jensen Transformers, Van Nuys, CA, USA
[2]THAT Corporation, Milford, MA, USA

Limited Common-Mode Rejection Ratio (CMRR) in balanced interfaces often limits dynamic range in real-world audio systems. Conventional differential amplifier input circuits suffer serious CMRR degradation when driven by real system signal sources instead of laboratory generators. An ideal audio transformer, because of its extremely high common-mode impedances, is virtually immune to this degradation. A new Integrated Circuit (IC) is described that uses a patented topology to achieve common-mode impedances comparable to those of an ideal transformer. As a result, the IC enables signals with very high dynamic range to be transported without contamination by system ground voltage differences or other sources of common-mode interference. Other features of the IC, relating to audio signal quality and reliability, are also detailed.
Convention paper 6261

9:30 am

K-2 Improved Analog Class-D Amplifier with Carrier Symmetry Modulation—*Bruce Candy*,[1] *S. M. Cox*[2]
[1]Halcro, Torrensville, Australia
[2]University of Adelaide, Adelaide, Australia

A novel analog class-D amplifier has been developed that produces low distortion. The structure follows the well known, prior art class-D structures with negative feedback but includes modulation of the symmetry of the carrier oscillator waveform by a derivative of the input signal. This compensates a nonlinear phase modulation effect that is intrinsic to the prior art structures. The improvement is substantial at very low extra cost.
Convention Paper 6260

10:00 am

K-3 Enhancement of Audio Signals Based on Modulation Spectrum Processing—*Carlos Avendano, Michael*

Goodwin, Creative Advanced Technology Center, Scotts Valley, CA, USA

In this paper we describe a signal processing technique for enhancing audio signals based on manipulation of their modulation spectra. The modification is achieved by filtering the time trajectories of spectral envelopes in different frequency bands. Scaling of higher modulation frequencies with shelving filters is used to modify rapidly-changing acoustic events, thus effectively enhancing transient components without the need for explicit detection. The perceptual effect of such modifications is analogous to the edge processing applied to images, where acoustic details can either be smoothed or sharpened depending on the desired quality of the sound.
Convention Paper 6259

10:30 am

K-4 Design Criteria for Simple Sinusoidal Parameter Estimation Based on Quadratic Interpolation of FFT Magnitude Peaks—*Mototsugu Abe*,[1] *Julius O. Smith III*[2]
[1]SONY Corporation, Tokyo, Japan
[2]Stanford University, Stanford, CA, USA

Due to its simplicity and accuracy, quadratic peak interpolation in a zero-padded Fast Fourier Transform (FFT) has been widely used for sinusoidal parameter estimation in audio applications. While general criteria can guide the choice of window type, FFT length, and zero-padding factor, it is sometimes desirable in practice to know more precisely the requirements for achieving a prescribed error bound. In this paper we theoretically predict and numerically confirm the errors associated with various parameter choices and provide precise criteria for designing the estimator. In particular, we determine (1) the minimum zero-padding factor needed for a given frequency-error bound in quadratic peak interpolation, and (2) the minimum allowable frequency separation for a given window length.
Convention Paper 6256

11:00 am

K-5 Raised Cosine Equalization Utilizing Log Scale Filter Synthesis—*David McGrath, Justin Baird, Bruce Jackson*,
Lake Technology, Surry Hills, New South Wales, Australia

An improved method of audio equalization utilizing raised cosine filters is introduced. Raised cosine filters offer improved selectivity in comparison to traditionally implemented equalization functions while also maintaining beneficial attributes such as a minimum phase response. The raised cosine filter also enables flat summation and asymmetrical filtering characteristics, resulting in an equaliza-

tion system offering capability beyond traditional filter implementations.
Convention Paper 6257

11:30 am

K-6 Real-Time Sound Source Separation: Azimuth Discrimination and Resynthesis—*Dan Barry*,[1] *Bob Lawlor*,[2] *Eugene Coyle*[1]
[1]Dublin Institute of Technology, Dublin, Ireland
[2]National University of Ireland, Maynooth, Ireland

We present a real-time sound source separation algorithm that performs the task of source separation based on the lateral displacement of a source within the stereo field. The algorithm exploits the use of the pan pot as a means to achieve image localization within stereophonic recordings. As such, only an interaural intensity difference exists between left and right channels for a single source. Gain scaling and phase cancellation techniques are used in the frequency domain to expose frequency dependent nulls across the azimuth plane. The position of these nulls in conjunction with magnitude estimation. Grouping techniques are then used to resynthesize separated sources. Results obtained from real recordings show that for music, this algorithm outperforms current source separation schemes.
Convention Paper 6258

12:00 noon

K-7 Enhancement of Audio Signals Using Transient Detection and Modification—*Michael Goodwin, Carlos Avendano*, Creative Advanced Technology Center, Scotts Valley, CA, USA

This paper describes a processing approach that enables perceptually compelling modification of audio signals via accentuation or suppression of transients. The transient detection uses a frequency-domain analysis, which yields a spectral flux parameter. In typical detection methods, such a parameter would be compared with a threshold to derive a binary transient detection function. Here, we instead use an adjustable graded response to arrive at a continuous transient characterization function. This smooth function is then used to drive a nonlinear frequency-domain signal modification. We demonstrate that binary detection is problematic for perceptual manipulation, that the soft-decision technique overcomes these problems, and that our system is able to achieve substantial modification of signal's attributes without introducing significant artifacts.
Convention Paper 6255

Session Z5 Saturday, October 30 9:30 am – 11:30 am
Outside 301/302

POSTERS: MULTICHANNEL SOUND

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation.

9:30 am

Z5-1 An Ear Training System for Identifying Parameters of Artificial Reverberation in Multichannel Audio—*Jason Corey*, University of Michigan, Ann Arbor, MI, USA

Artificial reverberation is a signal processing tool used by recording engineers to help create a sense of spaciousness, depth, and envelopment in a sound recording. A system is proposed for training listeners to detect and identify various aspects of artificial reverberation reproduced in multichannel audio. The training helps increase listeners' auditory sensitivity to parameters of artificial reverberation in sound scene comparisons. The exercises progress from simple matching to identification of the more subtle aspects of artificial reverberation.
Convention Paper 6262

9:30 am

Z5-2 A Novel Multichannel Panning Method for Standard and Arbitrary Loudspeaker Configurations—*Ramy Sadek, Chris Kyriakakis*, University of Southern California, Marina Del Rey, CA, USA

This paper presents a novel panning algorithm called Speaker-Placement Correction Amplitude Panning (SPCAP), which guarantees conservation of loudspeaker power output. The method is appropriate for any loudspeaker arrangement (e.g., ITU 5.1, 10.2, etc.) and scales with the number of loudspeakers. SPCAP works by correcting initial pan values based on loudspeaker placement to achieve constant power output. Because panning occurs over an arbitrary number of loudspeakers (i.e., is not pair-wise), SPCAP provides two significant advantages over discrete panning schemes. First, pan values for current and future surround-sound formats (e.g., 5.1 and 10.2) are guaranteed to conserve power under any lower-resolution setup, making dynamic up/down mixing in nonstandard setups feasible. Second, SPCAP provides a framework for producing wide (non point-source) sounds.
Convention Paper 6263

9:30 am

Z5-3 Multi-Source Low Frequency Room Simulation Using Finite Difference Time Domain Approximations— *Adrian Celestinos, Sofus Birkedal Nielsen*, Aalborg University, Aalborg East, Denmark

Sound level distribution generated by loudspeakers placed in a room can be simulated using numerical methods. The purpose of this paper is to present an application based on finite-difference time-domain approximations (FDTD) for the study of low frequencies in audio reproduction such as ordinary stereo to multichannel surround setups. A rectangular room is simulated by using a discrete model in time and space. This technique has been used extensively and gives good performance at low frequencies. The impulse response can be obtained in addition to the sound level distribution. Simulation of multiple loudspeakers in a room can be achieved to evaluate and visualize their coupling with the room. A high-frequency resolution can be obtained for auralization purposes.
Convention Paper 6264

9:30 am

Z5-4 A Host-Based Real-Time Multichannel Immersive Sound Playback and Processing System—*Ramy Sadek*, University of Southern California, Marina Del Rey, CA, USA

This paper presents ARIA (Application Rendering Immersive Audio). This system provides a means for the research community to easily test and integrate algorithms into a multichannel playback/recording system. ARIA uses a host-based architecture, meaning that programs can be developed and debugged in standard C++ without the need for expensive, specialized DSP programming and testing tools. ARIA allows developers to exploit the speed and low cost of modern CPUs, provides cross-platform portability, and simplifies the modification and sharing of codes. This system is designed for real-time playback and processing, thus closing the gap between research testbed and delivery systems.
Convention Paper 6265

9:30 am

Z5-5 Optimum Loudspeaker System with Subwoofer and Digital Equalization—*Vassilis Tsakiris,*[1] *Chris Orinos*[2]
[1]Crystal Audiovideo Ltd., London, UK
[2]Crystal Audio SA, Vrilissia, Greece

In this paper we investigate the subwoofer concept in relation to the various benefits of digital equalization and the

way it can be used together with today's small sized multichannel loudspeaker systems. We try to systematize a somewhat objective method of comparing between different subwoofer positions and crossover frequencies regarding their optimum response in a listening area. All these show that we can raise the subwoofer frequency at 120 Hz and thus relieve the main loudspeakers from the task of reproducing frequencies down to 80 Hz. Thus it is possible to create a high-end system using slim, line array, main loudspeakers, with all their known advantages, which can be correctly integrated, both aesthetically and acoustically, in any listening room.
Convention Paper 6266

Session L **Saturday, October 30** **11:00 am – 12:30 pm**
Room 301

INSTRUMENTATION AND MEASUREMENT

Chair: **Eric Benjamin**, Dolby Laboratories, San Francisco, CA, USA

P A P E R S

11:00 am

L-1 Determining the Peak Sound Level Capability of Loudspeakers and Sound Systems—*Bill Waslo*, Liberty Instruments, Inc., Liberty Township, Ohio, USA

One of the more relevant characteristics of a sound system is the maximum level at which it will function in its intended environment before the output becomes objectionably distorted. Because of design, construction, or thermal limitations, this characteristic can vary with both the frequency content and the duration of the applied stimulus at each measurement. Further complicating distortion measurement is the variation in frequency response due to reflections in the environment. This paper describes an automated technique using shaped tone-bursts under software control to generate the stimuli, acquire the responses, process and correct the data for room response, and present a graphical representation of the peak sound level capability versus test frequency. Also described is a novel technique for separating noise and distortion energy from stimulus energy in an in-room measurement.
Convention Paper 6267

11:30 am

L-2 Plane Wave Tubes—Uses and Limitations—*Marshall Buck*, Psychotechnology, Inc., Los Angeles, CA, USA

The use of a plane wave tube (PWT) is standard practice for the testing of audio compression drivers, as the damped tube provides an acoustic impedance load for the driver that is similar to an infinitely long horn of the same throat diameter. When properly terminated, it is anechoic. Uses of plane wave tubes for compression driver testing include: (1) Frequency response measurements, (2) distortion measurements; (3) coherence measurements; (4) power testing; (5) power compression testing; (6) listening tests; and (7) driver impedance measurements. The use of a PWT does not replace testing a driver on a horn in an anechoic environment; it is an adjunct to it. A construction method is described that provides an effectively terminated PWT. A calibration method is described that can be

used to assure that the anechoic response of a PWT is within a stated error level. Various microphone locations are evaluated for performance of frequency response and distortion. Error analysis is provided based on modal calculations and experimental data.
Convention Paper 6268

12:00 noon

L-3 Measurement of Audio Equipment with Log-Swept Sine Chirps—*Thomas Kite*, Audio Precision, Beaverton, OR, USA

The log-swept sine chirp provides a way to measure the transfer function and harmonic distortion of an audio device simultaneously. A deconvolution operation separates the linear and nonlinear responses in time. Results on real audio equipment are compared to classical methods and found to agree. An extension for simultaneously measuring crosstalk is suggested.
Convention Paper 6269

Session M **Saturday, October 30** **1:30 pm – 5:30 pm**
Room 301

PAPERS

ROOM AND ARCHITECTURAL ACOUSTICS; SOUND REINFORCEMENT

Chair: **Mendel Kleiner**

1:30 pm

M-1 Acoustic Redesign of the Danish National Gallery **—***Jan Voetmann*, DELTA Acoustics & Vibration, Hørsholm, Denmark

After the inauguration of the expansion of the Danish National Gallery in 1998 a serious acoustic mishap was experienced in the new large exhibition rooms. These interconnected rooms of approximately 33,000 cubic meters (approximately 1.2 million cubic feet) were supposed to also offer a multipurpose acoustical environment for a variety of cultural events. A record-breaking reverberation time of approximately 11 seconds was measured. The acoustic redesign process included not only the necessity of finding acoustical effective solutions; these solutions also had to be invisible or near-invisible due to the architectural requirements. This paper describes how the requirements were met, resulting in a highly acceptable reverberation time of a little more than 2 seconds.
Convention Paper 6270

2:00 pm

M-2 Systematic and Common Errors in Sound System STI and Intelligibility Measurements—*Peter Mapp*, Peter Mapp Associates, Colchester, UK

STI and its derivatives (RaSTI and STIPa) have become the internationally accepted methods for acoustically measuring the potential intelligibility performance of a sound system. However, in practice, many of the measurements carried out in the field to either verify or ascertain sound system and Voice Alarm intelligibility performance are often based on flawed techniques. This paper examines a number of common problems found to affect measurement accuracy. The paper also highlights conditions under which STI and STIPa inherently appear to incorrectly predict intelligibility performance. In particular it is shown that the currently available commercial software programs and instrumentation fail to correctly predict the performance of sound systems exhibiting irregular or band-limited frequency responses when they are operating in reverberant environments under quiet (i.e., high-signal-to-noise ratio)

PAPERS

conditions. Significant discrepancies between the various measurement platforms are also found to occur.
Convention Paper 6271

2:30 pm

M-3 Phase Equalization for Multichannel Loudspeaker-Room Responses—*Sunil Bharitkar*, Audyssey Laboratories, Inc., Los Angeles, CA, USA

Given a multichannel loudspeaker system, in a typical single or multiple listener setup, the combined response of the loudspeakers will exhibit significant fluctuation around the crossover region due to noncoincident positions of any two loudspeakers. This fluctuation manifests as an undesired broad spectral notch or a peak around the crossover region. The spectral notch, for example, introduced around the crossover due to complex addition of the two loudspeaker responses, generally, cannot be compensated with only magnitude response equalization. In this paper we present a recipe for compensating the spectral notch around the crossover region by designing a digital equalization filter using a stable all-pass network.
Convention Paper 6272

3:00 pm

M-4 Assessment of Music Audio Quality in a Sports Stadium—*Scott Willsallen, Densil Cabrera*, University of Sydney, Sydney, Australia

This paper investigates subjective and objective parameters of a sound reinforcement system in a large sports stadium. The sound at fourteen groups of three receiving positions was studied in a subjective listening test, as well as through objective system measurements. For an orchestral music sample, seventeen system tunings were subjectively assessed with fifteen scales. Objective measurements were made at each receiving position. Results showed significant variation for many of the subjective scales between tunings, as well as between receiving positions. To some extent, subjective and objective measurements were related as they describe system tuning and receiving position. Beyond its specific results, this paper highlights a range of difficulties in empirically assessing audio quality for music in a very large venue.
Convention Paper 6273

3:30 pm

M-5 Line Array Performance at Mid and High Frequencies
—*Henrik Staffeldt*,[1] *Ambrose Thompson*[2]

[1]HS Consulting, Copenhagen, Denmark
[2]Martin Audio Ltd., London, UK

This paper focuses on the direct sound frequency response of line arrays—rectilinear or curved—at mid and high frequencies (1 kHz to 10 kHz), which is arguably the most important range and one that is relatively easy to measure. In this frequency range a line array may produce irregular on- and off-axis frequency responses in the audience area, which is difficult to predict using simpler models. The irregularities, which appear as frequency varying attenuation, depend in a complicated way on array configuration and air absorption. Array performance prediction software usually models a line array as a number of directive point sources placed on a line or curve. The directive point source model has been used to simulate line arrays to study the frequency response behavior of line arrays at mid and high frequencies. The results of the study are compared with frequency response predictions calculated by new software including multichannel array controller simulations and measured complex spherical polar data for a specific 3-way line array cabinet. The predictions are compared to direct sound frequency response measurements on line arrays using the same 3-way cabinet to show the degree of accuracy with which directive point source models can predict the frequency responses of line arrays.
Convention Paper 6274

4:00 pm

M-6 Head-Tracked Auralization of Acoustical Simulation—
Christoph Moldrzyk,[1] *Wolfgang Ahnert,*[2] *Stefan Feistel,*[2] *Tobias Lentz,*[3] *Stefan Weinzierl*[1]
[1]Technical University of Berlin, Berlin, Germany
[2]ADA Acoustic Design Ahnert, Berlin, Germany
[3]Aachen University, Aachen, Germany

A desirable feature of modern acoustical simulation programs is the easy, fast, and reliable auralization of prediction results. To be considered as a serious tool, the auralization results should be equivalent to human perception in reality. In this paper we consider a new auralization technique based on a head-tracked headphone system with high spatial resolution and real-time convolution. We discuss the measurement of directional head-related transfer functions, the calculation of directional binaural impulse responses, and the realization as a real-time convolution software. A listening test was performed, comparing reality, measurement, and prediction results for a sample room.
Convention Paper 6275

P A P E R S

4:30 pm

M-7 Directional Measurement of Airborne Sound Transmission Paths Using a Spherical Microphone Array—*Bradford Gover*, National Research Council, Ottawa, Ontario, Canada

A spherical microphone array has been used to perform directional measurements of airborne sound transmission between rooms. With a source and array on opposite sides of a wall, omnidirectional impulse responses were measured to each of the array microphones. Beamforming resulted in a set of directional impulse responses, which were analyzed to find the distribution of arriving sound energy at the array position during various time ranges. Weak spots in the separating wall are indicated as directions of increased arriving sound energy. The system was able to identify minor defects in a test wall in between two reverberation chambers and also to identify leaks in the wall of an actual meeting room.
Convention Paper 6276

5:00 pm

M-8 Electronic Bass Trap—*Reza Kashani*,[1] *James Wischmeyer*[2]
[1]University of Dayton, Dayton, OH, USA
[2]Bag End Loudspeakers, Barrington, IL, USA

Bass traps, regardless of their effectiveness in abating bass acoustic coloration in a room have two, somewhat undesirable attributes: (1) large size and (2) lack of adaptability. An alternative to the use of bass traps, discussed in this paper, is incorporating a properly devised, feedback control scheme into a powered subwoofer, making the subwoofer to exhibit the same dynamics as that of a bass trap. This patent pending, active coloration control solution, which can be viewed as an "electronic bass trap" adds acoustic damping to the low-frequency modes of a room. In addition to a powered subwoofer, the electronic bass trap uses a microphone and an op-amp circuit. Numerical and experimental results indicate the effectiveness of the electronic bass trap in adding acoustic damping to the low-frequency standing wave(s) in a room.
Convention Paper 6277

Session N **Saturday, October 30** **1:30 pm – 5:00 pm**
Room 302

SIGNAL PROCESSING, PART 2

Chair: **Bob Adams**, Analog Devices, Norwood, MA, USA

1:30 pm

N-1 Moore's Law and Digital Audio—What Have We Done with All the Transistors?—*Peter Eastty*, Pro-Audio Lab, Sony, Oxford, UK

This paper investigates what has happened to the many transistors used in digital audio engineering. Improvements in signal quality have certainly accrued but advances in ease of use, reliability, availability, serviceability, power consumption, delay and cost have far outweighed, from a practical standpoint, advances in audio quality. Many examples are given, a novel signal visualization technique is described, and based upon the history, some predictions are made.
Convention Paper 6278

2:00 pm

N-2 Dither Myths and Facts—*Stanley Lipshitz, John Vanderkooy*, University of Waterloo, Waterloo, Ontario, Canada

Twenty-five years after the discovery of the desirable attributes of nonsubtractive triangular probability density function dither for signal quantization, some misunderstandings, myths, and half-truths still abound regarding what dither does and does not do. The increased use of dithered sigma-delta modulators has recently brought some of these questions to the fore. Some of these errors are relatively easy to explain and correct, while others are considerably more subtle in nature, but nevertheless also need to be addressed. This paper attempts to explain and clarify these matters, with the aid of copious time-domain, frequency-domain, and statistical-domain illustrations. It assumes that the reader already has a good knowledge of the theory of dithered quantization.
Convention Paper 6279

2:30 pm

N-3 Description of Limit Cycles in Feedback Sigma Delta Modulators—*Derk Reefman,*[1] *Joshua Reiss,*[2] *Erwin Janssen,*[1] *Mark Sandler*[2]
[1]Philips Research, Eindhoven, The Netherlands
[2]Queen Mary, University of London, London, UK

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The authors have recently developed a framework for analysis of limit cycle behavior in feedforward sigma delta modulators (SDMs). However, the dynamics of feedback SDMs appear to be completely different. Here, we extend that framework to include limit cycles in feedback SDMs. We prove that for DC inputs, periodic output implies state space periodicity. An outcome of this is that for an Nth order SDM, at least N-1 initial conditions must be fixed in order to have limit cycle behavior. We present expressions for the minimum disturbance of the input or initial conditions that is needed to break up a limit cycle. These expressions are notably different from the analogous expressions for feedforward SDMs. We show that dithering the quantizer is a suboptimal approach to removing limit cycles, and limit cycle stability is determined. Examples are provided that illustrate the theoretical results, and these results are also compared with those found for feedforward SDM designs. It is shown that, with respect to limit cycle behavior, it makes little difference whether feedforward or feedback designs are used.
Convention Paper 6280

3:00 pm

N-4 Implementation of "Tree" and "Stack" Algorithms for Look-Ahead Sigma Delta Modulators—*James Angus*, University of Salford, Salford, UK

Look-Ahead Sigma-Delta modulators look forward k samples before deciding to output a "one" or a "zero." The Viterbi algorithm is then used to search the trellis of the exponential number of possibilities that such a procedure generates. This paper describes alternative tree-based algorithms. Tree-based algorithms are simpler to implement because they do not require backtracking to determine the correct output value. They can also be made more efficient using "Stack" algorithms. Both the tree algorithm and the more computationally efficient stack algorithms are described. Implementations of both algorithms are described in some detail; in particular, the appropriate data structures for both the trial filters and score memories.
Convention Paper 6281

3:30 pm

N-5 Novel Subwoofer Signal Conditioner Design Using a Fully Programmable Analog Array and Software Tools— *Ian Macbeth*,[1] *Tegid Roberts*[2]
[1]Anadigm Ltd., Crewe, UK
[2]Cadarn Consulting Ltd., Eynsham, UK

The control and real-time software programmability of low-frequency audio signals in the analog domain is inaccu-

rate, cumbersome, and expensive. In the digital domain there are issues relating to low-frequency distortions, latency, and design time. A fully programmable analog array IC methodology is presented that combines the benefits of DSP programmability with analog signal processing by way of a case study demonstrating software design tools and custom software configuration models. This single-chip subwoofer conditioner solution implements a subsonic filter, adjustable audio compressor, Linkwitz transform filter, and low-pass output filter with full software control. Performance measurements of this implementation, as well as further enhancements to the software models, are also discussed.
Convention Paper 6282

4:00 pm

N-6 Characterization of Spherical Loudspeaker Arrays— *Peter Kassakian, David Wessel*, University of California, Berkeley, CA, USA

The synthesis and rotational control of radiation patterns produced by spherical arrays of loudspeakers is studied. We identify operating regions, in terms of complexity of patterns and frequency ranges, over which patterns can be accurately synthesized. By considering an inner product space of far-field patterns, we can reason geometrically about approximation errors when using the systems to synthesize and control target responses. Bounds for normalized error across subspaces, in particular subspaces corresponding to the control operation of rotation, are calculated using singular value decomposition. The bounds can be interpreted as the best and worst-case errors encountered when dynamically steering the patterns.
Convention Paper 6283

4:30 pm

N-7 Interpolating Linear- and Log-Sampled Convolution —*D. B. (Don) Keele, Jr.*, Harman International, Northridge, CA, USA

This paper describes a class of FIR filter/convolvers based on interpolation that allow sparse specification of the filter's impulse-response waveform or equivalently its frequency spectrum in both linear- and log-spaced domains. Interpolation allows the filter's impulse response or frequency response to be specified in significantly fewer samples. This is turn means that far fewer filter taps are required. Linear- and log-sampled interpolating filter/convolvers can further be categorized into two types: Type 1, interpolation in time, and Type 2, interpolation in frequency. Type 1 provides direct specification of the filter's

impulse response in linear or log time, while Type 2 allows direct specification of the complex (real-imaginary) frequency response of the filter in linear or log frequency. Each form of filter vastly reduces the number of filter taps but greatly increases the processing complexity at each tap. Efficient implementations of the log-spaced filter-convolvers are presented that use multiple asynchronous sample-rate converters. This paper is a continuation of the author's "Log Sampling" paper presented to the AES in Nov. 1994 [AES 97th Convention, Preprint 3935]. This paper represents work in progress with a conceptual description of the convolution technique with minimal mathematical development.
Convention Paper 6284

Session Z6 Saturday, October 30 2:00 pm – 4:00 pm
Outside 301/302

POSTERS: PSYCHOACOUSTICS, PERCEPTION, AND LISTENING TESTS AND SPATIAL PERCEPTION AND PROCESSING

PAPERS

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation.

2:00 pm

Z6-1 A Comparison of Speech Intelligibility between the Callsign Acquisition Test and the Modified Rhyme Test—*Daniel Hicks*,[1] *Mohan Rao*,[1] *Tomasz Letowski*[2]
[1]Michigan Tech University, Houghton, MI, USA
[2]U.S. Army Research Laboratory, Aberdeen Proving Ground, MD, USA

The Callsign Acquisition Test (CAT) is a new speech recognition test developed by the U.S. Army to examine speech intelligibility in a military environment. This paper compared speech intelligibility results of the Callsign Acquisition Test with another test used widely in industrial applications, the Modified Rhyme Test, using listening tests and objective speech metrics. A group of 24 listeners between the ages of 18 and 25 participated in the study. Six different types of recorded background noises radiating from an armored personnel carrier; helicopter; jet engine; mid-size car; subway train; and standard pink noise were used in the study. Test results demonstrated that the differences in the mean speech recognition scores obtained for CAT and MRT across all selected background noises were not statistically significant. However, the effect of noise and interactions between the noise and the test were statistically significant. A correlation of the measured scores with the spectral content of the background noise revealed somewhat higher scores for MRT compared with CAT under selected background noises that have most of the frequency content above 500 Hz. In contrast, slightly higher scores for CAT were noticed for selected noises having predominantly low-frequency components below 250 Hz.
Convention Paper 6285

2:00 pm

Z6-2 Adjustment of the Parameters Proposed for the Objective, Perceptual Based Evaluation Methods of Compressed Speech and Audio Signals—*Piotr Kozlowski, Andrzej Dobrucki*, Wroclaw University of Technology, Wroclaw, Poland

This document presents results of continuation of research about objective methods, which use psychoacoustics knowledge for estimation of the quality of audio signals. The software written especially for this research is presented. This program allows for implementation of the different published methods for evaluation of the quality of perceptual coded audio signals. Protocols PAQM, PSQM, NMR, PEAQ, PESQ are implemented up to now.

All of these algorithms are used for simulation of the auditory system. The software is open for the addition of new protocols such as the plug-ins. There is a possibility to change and improve earlier published protocols. Authors proposed in earlier works how to improve objective protocols, e.g., by changing pitch scale. Suggested adjustment of internal parameters of signal processing, which improves results of objective evaluation, is presented. The criterion of optimization is the difference between results of subjective and objective evaluation.
Convention Paper 6286

2:00 pm

Z6-3 New Techniques Assisting Cochlear Implants Fitting—*Adam Walkowiak*,[1] *Andrzej Czyzewski,*[2] *Artur Lorens*,[1] *Bozena Kostek*[1, 2]
[1]Institute of Physiology and Pathology of Hearing, Warsaw, Poland
[2]Gdansk University of Technology, Gdansk, Poland

Measurement of Spread of Excitation (SoE) provides a potential method of assessment of cochlear implant users' benefit. To provide maximum benefit for the cochlear implant users the speech processor should be fitted to the patients' need. One objective method that could deliver important information for fitting is Neural Response Telemetry (NRT). This method helps to estimate an amplitude of electrical current that is required to elicit hearing sensation via cochlear implant. It is also possible to determine Spread of Excitation—the longitudinal spread of electrically evoked neural excitation in the cochlea, based on NRT results. The parameters of the Spread of Excitation in the individual patient may help to explain the patients' performance and indicate in which way sound processing strategies could be modified to improve one's benefit. In this paper measured profiles of SoE are shown and some preliminary analyses are presented.
Convention Paper 6287

2:00 pm

Z6-4 Influence of Head-Tracking to Spatial Perception—
Wolfgang Hess, Ruhr-University, Bochum, Germany

To validate a head-tracking system in comparison to a loudspeaker arrangement regarding perception of auditory source width (ASW), i.e., the horizontal extension of the auditory event(s), experiments by representing narrow-band and broadband noise stimuli of certain degrees of correlation were conducted. The excitation signals were presented by (i) pairs of loudspeakers and (ii) headphones with head-tracking in an anechoic environment. The evaluation by five trained subjects resulted in a good correspondence of the outcomes of the loudspeaker and the head-tracking experiments, i.e., the head-tracking system had a negligible influence on spatial perception.
Convention Paper 6288

2:00 pm

Z6-5 Frequency Dependence of Perceptual Sound Image Distance Using Direct-to-Reverberant Ratio Control Method—*Yasushige Nakayama*,[1] *Kaoru Watanabe*,[1] Satoshi Kuwata[2]
[1]NHK Science & Technical Research Laboratory, Tokyo, Japan
[2]Denon, Ltd., Fukushima, Japan

A 3-dimensional sound image controlling method is described, which controls the intensity level of sound images to arrange them near and far away from a listener (the "control" means amplitude panning for distance). The images are created with two loudspeakers arranged near and far away from the listener or with a loudspeaker array system. A subjective evaluation was performed to examine the perceptual distance of the octave band and white noise by changing the direct-to-reverberant energy ratio. It was found that the distance produced by the direct-to-reverberant energy ratio has a frequency dependency, and that the distance for more than 5,660 Hz is not significantly different when the ratio is changed. In addition, a 3-D audio coding method utilizing this result was developed. An experiment using the tool showed that the bit-rate efficiency of the method, which unifies frequency components above 5,660 Hz, can be increased by more than 30 percent compared with dual-mono transform coding, without degradation in 3-D audio reproduction.
Convention Paper 6289

2:00 pm

Z6-6 Virtual Sound Algorithm for Wide Stereo Sound Stage—*Sunmin Kim, Joon Lee, Seong Cheol Jang, Sangil Park*, Samsung Electronics, Suwon, Korea

This paper provides a wide stereo algorithm that widens the stereo sound stage for a two-channel loudspeaker lay-

out. The design method consists of a binaural synthesis, a crosstalk canceller, and a direct filter. This creates multiple virtual loudspeakers and allows them to spread out in the front. Consequently, the proposed algorithm, which includes the widening of the sound stage and the timbre preservation, is designed in the form of a 2-by-2 filter matrix. The filter order is minimized for easy implementation while maintaining the performance.
Convention Paper 6290

2:00 pm

Z6-7 Audio Signal Decorrelation Based on a Critical Band Approach—*Maurice Bouéri, Chris Kyriakakis*, University of Southern California, Los Angeles, CA, USA

This paper presents a new method for decorrelating audio signals by applying a random time shift to each critical band. The resulting signals exhibit significantly lower interaural cross correlation (IACC). The effects of this type of decorrelation on perceived envelopment and loss of phantom image will be presented.
Convention Paper 6291

2:00 pm

Z6-8 A Simplified Scene-Based Paradigm for Use in Training Applications—*Rafael Kassier, Tim Brookes, Francis Rumsey*, University of Surrey, Surrey, UK

Despite recent consumer uptake of surround sound systems and the existence of a number of studies into spatial audio attributes, there is currently no system to train listeners in the detection and discrimination of the spatial attributes of reproduced sound. Timbral ear training has been shown to increase response consistency in listening tests, but a number of obstacles must be negotiated in order to successfully implement an ear training system for spatial aspects of sound reproduction. The first of these is the determination of which spatial audio attributes are appropriate. This paper describes the formulation of a new spatial audio paradigm by testing previously documented attributes against specific selection criteria and including, modifying or rejecting them accordingly.
Convention Paper 6292

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

Session O **Sunday, October 31** **9:00 am – 10:30 am**
Room 301

PAPERS

AUDIO RECORDING AND REPRODUCTION

Chair: **Derk Reefman**, Philips Research, Eindhoven, The Netherlands

9:00 am

O-1 Specifying the Jitter Performance of Audio Components—*Chris Travis*,[1] *Paul Lesso*[2]
[1]Sonopsis Ltd., Wotton-under-Edge, Gloucestershire, UK
[2]Wolfson Microelectronics, Edinburgh, UK

The question of sample-clock quality is a perennial one for digital audio equipment designers. Yet most chip makers provide very little information about the jitter performance of their products. Consequently, equipment designers sometimes get burnt by jitter issues. The increasing use of packet-based communications and class-D amplification will throw these matters into sharp relief. This paper reviews various ways of characterizing and quantifying jitter, and refines several of them for audio purposes. It also attempts to present a common, unambiguous terminology. The focus includes wideband jitter, baseband jitter, jitter spectra, period jitter, long-term jitter, and jitter signatures. Comments are made on jitter transfer through phase-locked loops and on the jitter susceptibility of audio converters.
Convention Paper 6293

9:30 am

O-2 High Performance Discrete Building Blocks for Balanced Audio Signal Processing—*Bruno Putzeys*, Grimm Audio, Eindhoven, The Netherlands

To audio systems designers, the “fully differential op amp” is a relatively new entry. Two discrete-circuit variations on the theme are presented, one of which provides effectively floating outputs.
Convention Paper 6294

10:00 am

O-3 Partial Unmixing for Personalized Audio—*Mark Dolson,* Creative Advanced Technology Center, Scotts Valley, CA, USA

Immersive audio for interactive gaming is necessarily processed and mixed in real time as it is being rendered on the game audio playback platform. It is generally assumed

that music and movie soundtracks require no comparable processing during playback because listeners typically provide no real-time input that might affect the final rendering. In reality, prepackaged audio is being delivered to music and movie playback platforms in increasingly diverse forms. The result is that mismatches between the spatial audio format, bit depth, and frequency range of the content and that of the playback system pose an emerging problem for which sophisticated playback processing may be an appropriate response. This paper presents a formal statement of the mismatch problem and proposes a unified solution using frequency-domain processing to perform "partial unmixing" of the prepackaged content. Last, we show how this can enable a new music/movie listening experience rooted in the concept of "personalized audio."
Convention Paper 6295

Session P **Sunday, October 31** **9:00 am – 10:30 am**
Room 302

PAPERS

HIGH RESOLUTION AUDIO

Chair: **Mark Sandler**, Queen Mary, University of London, London, UK

9:00 am

P-1 A New Method of Applying High Levels of Dither to Delta-Sigma Modulators—*James Angus*, University of Salford, Salford, Greater Manchester, UK

This paper presents a new method of applying high levels of dither to sigma-delta modulators. In particular, it clarifies the position of the overload point in one-bit sigma-delta modulation systems and presents several overload control methods with comparisons of their efficacy. It then goes on to examine the problem of applying dither to one-bit systems and describes a new approach for applying high levels of dither. It also examines the effect of different dither probability density distributions and shows that simple bilevel dither can be effective at lower levels than other probability density distributions. It presents results that show that dither can be applied at a high enough level to be effective in one-bit sigma-delta modulation systems.
Convention Paper 6296

9:30 am

P-2 Scaleable Multichannel DSD Coding—*Malcolm Hawksford*, University of Essex, Colchester, Essex, UK

DSD is a 1-bit coding scheme based upon sigma-delta modulation. In the commercial realization of this technology exploiting DVD optical disc storage, six discrete channels are accommodated each with a constant bit rate of 2.8224 Mb/s, a specification that cannot be changed within the context of the SACD release format. However, a method of embedding additional data in the DSD bitstream is shown to be feasible with the aim of increasing the number of channels to twelve. The technique retains full compatibility with SACD and only requires modest processing to decode an additional six channels.
Convention Paper 6297

10:00 am

P-3 Perceptual Discrimination of Very High Frequency Components in Musical Sound Recorded with a Newly Developed Wide Frequency Range

PAPERS

Microphone—*Kimio Hamasaki*,[1, 2] *Toshiyuki Nisiguchi*,[1] *Kazuho Ono*,[1] *Akio Ando*[1]

[1]NHK Science & Technical Research Laboratories, Tokyo, Japan
[2]Kyushu University, Fukuoka, Japan

Subjective evaluation tests on perceptual discrimination between musical sounds with and without very high frequency (above 20 kHz) components have been conducted. To make a precise evaluation, the test system is designed to exclude any influence from very high frequency components in the audible frequency range. Moreover, various sound stimuli are originally recorded by a newly developed very wide frequency range microphone in order to contain enough components in very high frequency range. Tests showed that some subjects might be able to discriminate between musical sounds with and without very high frequency components. This paper describes these subjective evaluations, and discusses the possibility of such discrimination as well as the high-resolution audio recording of music.
Convention Paper 6298

Session Z7 Sunday, October 31 9:30 am – 11:30 am
Outside 301/302

POSTERS: SIGNAL PROCESSING, PART 1

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation.

P A P E R S

9:30 am

Z7-1 High-Ordered Adaptive FIR Filters for Acoustical Echo Cancellation—*Devon A. Bergman*,[1] *Michael Scordilis*[2]
[1]Dolby Laboratories, San Francisco, CA, USA
[2]University of Miami, Miami, FL, USA

A common problem in acoustical echo cancellation is the continuous change in the room's impulse response. These changes need to be monitored and adapted to in order to create an enjoyable echo-free environment. In the experiment, measurements were taken to find an optimal FIR filter that would allow for fast convergence of the adaptive filter as well as a significant echo return loss. Finally, considerations were also taken to create a smooth adaptation from one configuration to the next.
Convention Paper 6299

9:30 am

Z7-2 Class D Amplifier with Zero Switching Ripple—
Eric Mendenhall, Audio Power Electronics, Dove Canyon, CA, USA

Class D amplifiers are used for their high efficiency, but they have some undesirable characteristics, one being the residual switching frequency ripple. This paper shows a method of switching frequency ripple reduction by means of ripple steering. With this technique a second output is constructed, into which the switching ripple is steered, substantially relieving the main output from a major artifact of Class D operation.
Convention Paper 6300

9:30 am

Z7-3 FPGA Implementation of an Audio Processor—*Sevag Balkorkian*, American University of Beirut, Beirut, Lebanon

Implementing hardware design in Field Programmable Gate Arrays (FPGA) is a formidable and an interesting task especially when considering digital signal processing (DSP) applications. Hardware design skills and strong background in signal processing are required. Sometimes problems arise in realizing hardware implementation for a

simple design of systems where the theoretical concept is plausible; care should be taken to account for minute design details. The objective of this paper is to present the design of a digital audio signal processor that performs multi-effect processing and, at the same time, is capable of real time configurability on a single FPGA chip. The design is specific to certain algorithmic tasks; there is no need for general purpose architecture, and it can be characterized as a system on chip application. It is configurable and able to change coefficients utilizing Look up Tables and is capable of performing filtering and echo/delay generation.
Convention Paper 6301

9:30 am

Z7-4 Real-Time Power Supply Compensation for Noise-Shaped Class D Amplifier—*Lingli Zhang, John Melanson, Johann Gaboriau, Mel Hagge, Randy Boudreaux*, Cirrus Logic, Austin, TX, USA

This paper presents a pure digital real-time power supply compensation scheme for both single-ended and bridge tied-load configured noise-shaped class D amplifiers. Using the appropriate power supply measurement circuitry, the scaled AC and DC components of the power supply voltage rail(s) are fed back into the PWM controller to modify the feedback path and the direct path of the noise shaper. All delays through the feedback loop have been minimized such that the ripple cancellation of the output stage is accomplished in real time. A two-chip ADC/PWM controller with this compensation scheme achieves 40 dB power supply rejection of a 60 Hz ripple and 100 dB system dynamic range.
Convention Paper 6302

9:30 am

Z7-5 Digital Audio Power Amplifier for DSD Data Streams—
Francis Prime,[1] *Malcolm Hawksford*[2]
[1]KEF Audio, Maidstone, Kent, UK
[2]University of Essex, Colchester, Essex, UK

A digital power amplifier topology is proposed optimized specifically for use with DSD-type data streams. The configuration enables direct interfacing of DSD data with no requirement for intermediate signal processing or analog-to-digital conversion. The output architecture exploits a classic H-bridge configuration and uses a novel form of ac data coupling to simplify internal interface circuitry. Wide range gain control is enabled through modulation of the output-stage power supply voltage that also improves power efficiency at low gain settings. Consideration is giv-

en to finite pulse rise time and a modified DSD data format is investigated.
Convention Paper 6303

9:30 am

Z7-6 Implementation of High-Order Convolution Algorithms with Low Latency on Silicon Chips—*Rolf Anderegg*,[1] *Uli Franke*,[1, 2] *Norbert Felber*,[1] *Wolfgang Fichtner*[1]
[1]Swiss Federal Institute of Technology, Zurich, Switzerland
[2]Weiss Engineering Ltd., Zurich, Switzerland

Audio signal processing often requires modeling of large rooms (e.g., churches) with impulse responses of several seconds duration. Direct convolution of the sound stream with such long responses exceeds the capacity of common signal processors by far. Using the Fast Fourier Transform instead reduces the number of operations logarithmically but introduces unacceptable latency. Segmenting the processing into initial short blocks and subsequent longer ones lets one trade latency versus computation power as presented in previous AES papers. Hardware-wise the reduction of operations comes at the cost of large storage with high memory bandwidths. Dedicated application-specific integrated circuits (ASIC) are predestined to perform the rather regular processing, freeing the processors for other tasks. This paper shows suitable architectures for integration on silicon of optimized fast-convolution algorithms. Possible optimizations for fast-convolution algorithms are examined. Based on these findings different architectures for integration on ASIC/FPGA (Field Programmable Gate Array) of such algorithms are developed, analyzed, and compared. The paper is concluded by presenting an exemplary ASIC implementation.
Convention Paper 6304

9:30 am

Z7-7 Scalability in the Modified Discrete Cosine Transform Filter Bank—*Chul-Jae Yoo*,[1] *Hyung-Myung Kim*[2]
[1]LG Electronics, Seoul, Korea
[2]KAIST, Taejon, Korea

Modified Discrete Cosine Transform (MDCT) filter bank, or often called the Time Domain Aliasing Cancellation (TDAC) filter bank, is widely used in audio coding systems. The last step of the conventional MDCT filter bank is the dual overlap add procedure to restore an uncompressed original signal. The last step can be generalized using the multiple overlap add procedure, in which the input and output block size can be reduced as the number of overlapped windows increases. The MDCT system with

multiple overlap add can reveal scalability features in proportion to the number of overlapped windows when it is used along with the fixed bit adaptive quantization capable of maintaining nearly the same SNR irrespective of the input level. It has been shown that the proposed structure is scalable in block units with the same data rate as the conventional system, and that it shows slight SNR improvement over the conventional one.
Convention Paper 6305

Session Q **Sunday, October 31** **11:00 am – 12:30 pm**
Room 302

PAPERS

AUTOMOTIVE AUDIO

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

11:00 am

Q-1 Construction of a Car Stereo Audio Quality Index—
Andrea Azzali,[1] *Angelo Farina*,[1] *Guido Rovai*,[2] *Giovanni Boreanaz*,[3] *Giorgio Irato*[3]
[1]University of Parma, Parma, Italy
[2]Fiat Auto, Orbassano, Torino, Italy
[3]Vehicle - NVH, CRF (Fiat Research Center), Orbassano, Torino, Italy

A measurable index (IQSB) quantifying perceived quality of car stereos has been developed to forecast aural appreciation. Results of panel interviews and listening tests (in a special "auralization room") have been correlated with the analysis of corresponding binaural recordings. Two outputs were obtained. First, a model of the subjectively most relevant features was identified in terms of statistically significant "verbal descriptors." Second, a single-figure index was constructed, function of objective measurable quantities related with audio performance, and well correlating with the average verbal evaluation (both of "naïve" and "expert" listeners). This tool is of great importance for the automotive industry because it allows for the direct quantification of the audio system performance, a significant part of the perceived quality of the product.
Convention Paper 6306

11:30 am

Q-2 Measurement of Speech Transmission Index Inside Cars Using Throat-Activated Microphone and Its Correlation with Drivers' Impressions—*Fabio Bozzoli, Angelo Farina*, University of Parma, Parma, Italy

One of the most used intelligibility parameters is the Speech Transmission Index. The techniques for determining it employs an artificial speaker and listener. Inside cars, where signal-to-noise ratio is particularly low, the value of STI is mainly influenced by this ratio. Determining the sound power of real speakers is the only way for piloting correctly the artificial mouth. We have implemented a technique that is based on a throat-activated microphone, and it is able to find the level of a real speaker's voice inside noisy spaces in the effective conditions. We have

primarily studied the speech inside cars and have discovered how the value defined by typical configurations may be extremely different from real ones. In this way, we have been able to produce more reliable excitation signals. Using this "raised signal," we have tested one car and have tried to find a good correlation between drivers' impressions and objective values.
Convention Paper 6307

12:00 noon

Q-3 High Power Step-Up Converter for Car Subwoofers—
Giovanni Franceschini,[1] *Alberto Bellini*,[1] *Emilio Lorenzani*,[1] *Matteo Cavatorta*,[1] *Francesco Violi*[2]
[1]University of Parma, Parma, Italy
[2]ASK Industries, Reggio Emilia, Italy

This paper presents an original DC/DC step-up converter topology for high power car audio applications with a battery supply. A prototype was realized and tested for the power supply of a TANDEM subwoofer box. Audio signals are characterized by high dynamic variations, thus they can be amplified only relying on power supplies with high dynamic capabilities and low output ripple. The latter constraint can be achieved only using closed-loop switch-mode DC/DC converters at high switching frequency. So doing converter efficiency is reduced, and EMC problems arise. In summary, power supply efficiency and supply voltage quality are key features of the converter design. In this paper the above-mentioned issues were tackled relying on an open loop topology. The original solution is the adoption of a three-phase transformer within a full-bridge converter topology. The proposed architecture will be referred to as 3 boost power supply. Experimental results confirm that the 3 boost power supply topology allows achievement of higher efficiency, a lower ripple factor, and a unitary transformer utilization factor. It turns out that it is an efficient power supply for a car audio subwoofer system, specifically for a digital output stage, where the quality of the supply level is a key element. The proposed architecture is patent pending.
Convention Papers 6308

Session Z8 Sunday, October 31 1:00 pm – 3:00 pm
Outside 301/302

P A P E R S

POSTERS: SIGNAL PROCESSING, PART 2

NOTE: During the first 10 minutes of the session all authors will give a brief outline of their presentation.

1:00 pm

Z8-1 Increased Correlation in Blind Audio Watermark Detection—A Blessing in Disguise?—*Krishna Kumar S.*,[1, 2] *Thippur Sreenivas*[3]
[1]Indian Institute of Science, Bangalore, India
[2]Centre for Development of Advanced Computing, Trivandrum, India
[3]Indian Institute of Science, Bangalore, India

Audio watermarks are often made signal-dependent to keep them inaudible in the host signals. Blind watermark detectors, which do not have access to the unwatermarked signal, may seem handicapped, because an approximate watermark has to be rederived from the watermarked signal. Referring to the exact watermark scenario as a semi-blind detector, some reduction in performance is anticipated in blind detection over semi-blind detection. The present paper is an experimental investigation into this issue, applied to a typical correlation-based audio watermark detection scheme. It is found, surprisingly, that the statistical performance of the blind detector is better than that of the semi-blind detector. It is found that the rederived watermark is better correlated to the host signal and hence leads to better detection performance. It is confirmed that this happens only if the embedded watermark is the same as the examined watermark.
Convention Paper 6309

1:00 pm

Z8-2 Comparison of Effectiveness of Musical Sound Separation Algorithms Employing Neural Networks
—*Bozena Kostek, Marek Dziubinski, Piotr Dalka*, Gdansk University of Technology, Gdansk, Poland

In this paper several algorithms are presented, developed for musical sound separation. The proposed techniques for the decomposition of mixed sounds are based on the assumption that pitch of the sounds contained in the mix is known, i.e., inputs of the algorithms are pitch tracks of the signals contained in the mixture. The estimation process of phase and amplitude contours representing harmonic components is based on the limited number of

inner product operations, performed on the signal with the use of complex exponentials matching pitch characteristics of the separated signals, and not on the discrete spectral representations calculated via DFT. In this paper examples of separation results are presented and each algorithm performance is analyzed. The effectiveness of separation algorithms consists in calculation of feature vectors (FVs) derived from musical sounds after the separation process is performed and then in feeding them the Neural Network (NN) for automatic musical sound identification. The experimental results are shown and discussed. A comparison of effectiveness of all presented algorithms is also included, and conclusions are derived.
Convention Paper 6310

1:00 pm

Z8-3 Distortion Audibility in Inverse Filtering—*Scott Norcross, Gilbert Soulodre, Michel Lavoie*,
Communications Research Centre, Ottawa, Ontario, Canada

Previous studies have shown that certain inverse filtering methods introduce audible artifacts that can degrade the audio signal. To correct some of these artifacts various techniques such as regularization, smoothing, and increasing the length of the inverse filter have been proposed. While these methods help in some cases they may also produce other artifacts or distortions that degrade the audio quality. In the present study formal subjective tests were conducted to systematically investigate modeled distortions similar to those found in inverse filtering. Parameters of the distortions, such as spectral shape, length, and time profiles were varied for the subjective tests. The results of the tests can be used to better understand the audibility of these artifacts and to create a perceptual model that can be used to design subjectively improved inverse filters.
Convention Paper 6311

1:00 pm

Z8-4 Harmonic Sound Source Separation Using FIR Comb Filters—*Mikel Gainza*,[1] *Bob Lawlor*,[2] *Eugene Coyle*[1]
[1]Dublin Institute of Technology, Dublin, Ireland
[2]National University of Ireland, Maynooth, Ireland

A technique for separating harmonic sound sources using FIR comb filters is presented. First, a preprocessing task is performed by a multipitch estimator to detect the pitches that the signal is composed of. Then, a method based on the Short Time Fourier Transform (STFT) is utilized to interactively extract the harmonics belonging to a given

source by using FIR comb filters. The presented approach improves upon existing sinusoidal model approaches in terms of the perceptual quality of the extracted signal.
Convention Paper 6312

1:00 pm

Z8-5 AES Technical Committee on Signal Processing Educational CD Project—*Robert Maher*, Montana State University, Bozeman, MT

The AES Technical Committee on Signal Processing is developing a compact disc with educational material and demonstrations intended for students, educators, and working digital audio engineers. The material includes examples of quantization and dither, basic psychoacoustics, and practical DSP. The multimode CD will have both audio tracks and a CD-ROM section with a Web-browser interface. The CD will be produced for sale by the AES Publications office.
Convention Paper 6313

Session R **Sunday, October 31** **1:30 pm – 3:30 pm**
Room 301

PAPERS

SPATIAL PERCEPTION AND PROCESSING

Chair: **Durand Begault**, NASA Ames Research Center, Mountain View, CA, USA

1:30 pm

R-1 Telescopic Spatial Radio—*Norman Jouppi, Subu Iyer*, Hewlett Packard, Palo Alto, CA, USA

We have developed a system we call Telescopic Spatial Radio (TSR). This system transforms monaural transmissions from geographically distributed loudspeakers into a spatial audio presentation using binaural techniques, which preserve the actual physical angles between participants. TSR instantly augments the user's situational awareness with the headings of the speaking users. The system leverages orientation measuring, location tracking, and signal processing capabilities that are rapidly decreasing in cost. TSR has many potential applications ranging from emergency and aviation communication to a richer consumer experience. We have developed a prototype system using laptop computers, GPS, and electronic compasses. The system allows users to select HRTFs from a library and operates over a computer network.
Convention Paper 6314

2:00 pm

R-2 Dynamic Cross-Talk Cancellation for Binaural Synthesis in Virtual Reality Environments—*Tobias Lentz, Gottfried Behler*, Aachen University, Aachen, Germany

To create a virtual reality environment with true immersion a precise spatial audio reproduction system is required. Since the placement of large loudspeaker arrays, which are needed for wave field synthesis systems, may be impossible for some environments, alternative solutions must be found. One application of this kind is a multi-screen VR system where the stereoscopic video images envelope the user. In such a case the presented binaural approach has many advantages. This paper describes the virtual sound source imaging by binaural synthesis and the reproduction over loudspeakers with a dynamic (tracked) cross-talk cancellation system that only needs three to four loudspeakers to cover all listening positions.
Convention Paper 6315

2:30 pm

R-3 Dancing Music: Integrated MIDI-Driven Synthesis and Spatialization for Virtual Reality—*Masahiro Sasaki, Michael Cohen*, University of Aizu, Aizu-Wakamatsu, Fukushima-ken, Japan

We describe a unique MIDI-based spatial sound system featuring a network-driven bank of four RSS-10s (Roland Sound Space Processors) driving a eight-transducer circumferential loudspeaker array in a "3-D Theater," enabling a three-dimensional dynamic musical space. Sound sources can be choreographed by adding dynamic positional gestures to standard MIDI files. Our sequencing system interprets such files, partitioning their data into two streams: one for MIDI tonal events, sent to synthesizers, and the other for positional data, sent simultaneously to sound spatializers, clients in a multimodal heterogeneous collaborative virtual environment. This research achieves programmed and interactive musical control of sound spatialization, synchronizable with stereographic 3-D contents, for spatializing music to give real presence to an audience.
Convention Paper 6316

3:00 pm

R-4 Integration of Measurements of Interaural Cross-Correlation Coefficient and Interaural Time Difference within a Single Model of Perceived Source Width—
Russell Mason, Tim Brookes, Francis Rumsey, University of Surrey, Guildford, Surrey

A measurement model based on the interaural cross-correlation coefficient (IACC) that attempts to predict the perceived source width of a range of auditory stimuli is currently under development. It is necessary to combine the predictions of this model with measurements of interaural time difference (ITD) to allow the model to provide its output on a meaningful scale and to allow integration of results across frequency. A detailed subjective experiment was undertaken using narrow-band stimuli with a number of center frequencies, IACCs and ITDs. Subjects were asked to indicate the perceived position of the left and right boundaries of a number of these stimuli by altering the ITD of a pair of white noise comparison stimuli. It is shown that an existing IACC-based model provides a poor prediction of the subjective results but that modifications to the model significantly increase its accuracy.
Convention Paper 6317

PAPERS

Session S **Sunday, October 31** **1:30 pm – 3:30 pm**
Room 302

AUDIO-VIDEO SYSTEMS

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

1:30 pm

S-1 Sound Editing Workflows and Technologies for Digital Film—The Nonlinear Soundtrack—*John McKay*, Virtual Katy, Wellington, New Zealand

Advancements in nonlinear editing technology have enabled directors to modify their film project at any point during the post process. This freedom provides significant creative flexibility. However, the technologies for sound and film editing are not fully integrated and pose a challenge for sound editors keeping sync with film edits and changes. This paper introduces new workflows and technologies that enable sound editors to work in tandem with the changing film and automate manual processes in a collaborative nonlinear environment. These new workflows and changing technologies will be described using a real-world motion picture case study—*Lord of the Rings*.
Convention Paper 6318

2:00 pm

S-2 5-1 Surround Sound Productions with Multiformat HDTV Programs—*Kazutugu Uchimura, Hiroshi Kouchi, Shinichiro Ogata*, NHK Broadcasting Center, Tokyo, Japan

In the international co-production of HD (high definition) programs, there are some problems in postproduction. These problems originate in the frame-rate relationship between 24 p and 23.976 p. Generally, 23.976 p shooting is used to ensure compatibility with TV systems such as NTSC, but some problems occur when transferring to the PAL system or films. In this paper, in a coproduction with China called *The Ancient Routes of Tea & Horses*, we accomplished 5.1 surround sound that was compatible with both 24 p and 59.94 i HD images. This paper describes the production techniques and problems, and some future challenges. We believe the techniques will be useful for media combinations in the future.
Convention Paper 6319

2:30 pm

S-3 PC-Based Sound Reproduction System Linked to Virtual Environments Rendered by VRML—*Kentaro*

Matsui, Hiroyuki Okubo, Setsu Komiyama, NHK Science & Technical Research Laboratories, Tokyo, Japan

A PC-based sound reproduction system, called PC-VRAS control, has been developed that is linked to virtual environments rendered by Virtual Reality Modeling Language (VRML). It provides spatially synchronized three-dimensional sound with the VRML scene. A listener can explore the VRML scene at will. The surrounding sound is synchronized and automatically resynthesized with each step taken by the listener in real time. This control is built on ActiveX technologies and runs on the Internet Explorer browser window. All processing is done by software that runs on a standard personal computer, so there is no need for any special device.
Convention Paper 6320

3:00 pm

S-4 A Headphone-Free Head-Tracked Audio Telepresence System—*Norman Jouppi, Subu Iyer, April Slayden*, Hewlett Packard, Palo Alto, CA, USA

We have developed a headphones-free bidirectional immersive audio telepresence system. The primary user of the system experiences four-channel audio from a remote location while sitting or standing in a 360-degree surround projection display cube. The display cube incorporates numerous acoustic enhancements, including tilted screens, an anechoic ceiling, and loudspeakers ported through slits in the display cube edges. Head-tracking based on near-infrared video technology obtains both the user's head position and orientation. Users can then vary the orientation of their projected voice at the remote location merely by rotating their own head. Similarly, the arrival time and volume of sound channels transmitted from the remote location is varied automatically in the display cube based on the position of the user's head, to help maintain proper perceived interaural time and level differences between multiple channels.
Convention Paper 6321

Workshops

W-1	**Subjective Evaluation Methods for Multichannel Automotive Sound Systems**	Thursday, Oct. 28 9:00 am – 11:00 am Room 308
W-2	**Field Recording in the Wild**	Thursday, Oct. 28 1:00 pm – 3:00 pm Room 308
W-3	**High Resolution Audio in an Age of Universal Playback**	Thursday, Oct. 28 4:00 pm – 6:00 pm Room 308
W-4	**Spatial Coding of Surround Sound: A Progress Report**	Friday, Oct. 29 9:00 am – 11:00 am Room 308
W-5	**FireWire in Studios—Benefits and Challenges**	Friday, Oct. 29 11:30 am – 1:30 pm Room 308
W-6	**Mastering for Low Bit-Rate Perceptual Codecs**	Friday, Oct. 29 2:00 pm – 4:00 pm Room 308
W-7	**How We Made Games in the Hallway Rock!**	Friday, Oct. 29 4:30 pm – 6:30 pm Room 308
W-8	**Architectural Acoustics for Film and Broadcast Studios**	Saturday, Oct. 30 9:00 am – 11:00 am Room 308
W-9	**Which Audio Recording and Storage Medium for What Purpose?**	Saturday, Oct. 30 11:30 am – 1:30 pm Room 308
W-10	**The Power of Loudspeaker Models**	Saturday, Oct. 30 2:00 pm – 4:00 pm Room 306
W-11	**Signal Processing and Hearing Aids**	Saturday, Oct. 30 4:30 pm – 6:30 pm Room 306
W-12	**The Role of Multiple Low-Frequency Signals in the Perception of Reproduced Sound**	Sunday, Oct. 31 9:00 am – 11:00 am Room 308
W-13	**Future Challenges for the All-Software Studio: Scalability, Stability, Usability**	Sunday, Oct. 31 11:30 am – 1:30 pm Room 308
W-14	**Lossless Audio Coding: MPEG and De Facto Standards**	Sunday, Oct. 31 2:00 pm – 4:00 pm Room 308

Workshops

Workshop 1
Thursday, October 28 **9:00 am – 11:00 am**
Room 308
Cars on Display in Rotunda

SUBJECTIVE EVALUATION METHODS FOR MULTICHANNEL AUTOMOTIVE SOUND SYSTEMS

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

Panelists: *Kristina Busenitz*, Harman Becker Automotive Systems
Dave Clark, DLC Design, Wixom, MI, USA
Laurie Fincham, THX, Ltd.., San Rafael, CA, USA
Hans Lahti, Volvo, Göteborg, Sweden
Mark Ziemba, Panasonic Automotive, Southfield, MI, USA

Audio systems fitted to cars have become progressively more sophisticated, with both high quality audio and movie playback now commonplace. Systems that decode 5.1 channel material from both movie and music sources have appeared this year and are set to be widespread in the near future. This workshop will explore the issues associated with the reliable evaluation of these systems. These include the experiment design, data analysis, dependency of system architecture, seat position, source material, etc. There will be evaluation clinics in cars running through the convention and these will be explained with the results to be reported at the next AES Convention in Barcelona. This study is intended to illuminate the issues discussed and provide data to eventually help draft a recommended practice.

Workshop 2
Thursday, October 28 **1:00 pm – 3:00 pm**
Room 308

FIELD RECORDING IN THE WILD

Chair: **Bernie Krause,** Wild Sanctuary, Glen Ellen, CA, USA

Panelists: *Rob Danielson*, University of Wisconsin, Milwaukee, WI, USA
Charlie Fox, University of Regina, Regina, Saskatchewan, Canada
Ann Kroeber, Sound Mountain, Berkeley, CA, USA
Judy Rocchio, National Park Service, Oakland, CA, USA

Natural soundscapes comprise one of the most fragile and quickly disappearing acoustic constituencies of our planet. This workshop will address the issues of what we once had, what we have lost within the lifetimes of most field recordists capturing natural soundscapes, and the extraordinary lengths to which recordists are compelled to go to capture what's left.

Workshop 3
Thursday, October 28 **4:00 pm – 6:00 pm**
Room 308

HIGH RESOLUTION AUDIO IN AN AGE OF UNIVERSAL PLAYBACK

Chair: **Vicki Melchior,** Audio Signal Processing Consultant, San Anselmo, CA, USA

Panelists: *John Atkinson,* Stereophile Magazine, USA
Rainer Finck, Marantz, Germany
Malcolm O. Hawksford, University of Essex, Colchester, UK
Hajime Kawai, Texas Instruments, Japan
George Massenburg, GML, LLC, TN, USA

Is it possible to maintain high quality, high resolution audio when audio is played back through players/receivers that out of necessity must support not only high resolution audio (likely both PCM and SACD) but also video and a plethora of other audio formats? Can the fundamental issues of conversion, processing, clocking, etc., readily be addressed across the demands of multiple A/V formats? This workshop seeks to address the following issues: What defines high resolution? Are multiformat players compatible with high resolution? What problems afflict them (e.g., clocking, conversion, quality of ICs, jitter, optimization for video, analog design)? Are the design issues fundamental or merely implementation-specific (i.e., dependent only on cost effectiveness)? What system design approaches are preferable?

Workshop 4
Friday, October 29 **9:00 am – 11:00 am**
Room 308

SPATIAL CODING OF SURROUND SOUND: A PROGRESS REPORT

Chair: **Mark Davis,** Dolby Laboratories, Inc., San Francisco, CA, USA

Panelists: *Roger Dressler,* Dolby Laboratories, San Francisco, CA, USA
David Griesinger, Lexicon, Inc., Bedford, MA, USA
Jürgen Herre, Fraunhofer IS, Erlangen, Germany
Robert Reams, Neural Audio, Kirkland, WA, USA
Gilbert Soulodre, Communications Research Centre, Ottawa, Ontario, Canada
Mark Vinton, Dolby Laboratories, Inc., San Francisco, CA – USA

The rise in popularity of surround sound in recent years has resulted in a duality of formats, with both stereo and 5.1-channel surround enjoying wide acceptance. This, in turn, has made it desirable to implement a common data format from which either presentation format can be derived. While it is a relatively straightforward matter to downmix a 5.1-channel program to stereo, it is more compact to provide a stereo format that can be upmixed to surround. Two traditional approaches are the “blind” upmixing of a pure stereo source, e.g., without side-chain information, and upmixing an “encoded” stereo file (“Lt/Rt”), which has been derived from a 5.1-channel source via a specific encoding downmix matrix. Systems employing augmented downmixed channels with side-chain information have demonstrated the feasibility of conveying compatible stereo/5.1-surround programs at data rates of 64 Kbps or less and have become the focus of a new MPEG standardization process. This workshop will provide an overview of the principles of such systems, and examine some of the outstanding problems and proposed solutions of systems currently under development.

Workshop 5
Friday, October 29 **11:30 am – 1:30 pm**
Room 308

FIREWIRE IN STUDIOS—BENEFITS AND CHALLENGES

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

Panelists: *Jun-ichi Fujimori,* Yamaha Corporation, Hamamatsu, Japan
Morten Lave, TC Applied Technologies, Markham, Ontario, Canada
Bob Moses, Island Digital Media Group, Vashon Island, WA, USA
Mark Olleson, Yamaha, Ltd., London, UK
Tim Thompson, Kurzweil Music Systems, Waltham, MA, USA

Experts in the field will highlight the benefits and practical challenges of implementing FireWire (IEEE-1394) into studios. Discussion topics will include: Hardware connections and limitations; Audio transmission, sample rates, word lengths, synchronization, and jitter; Device control, MIDI and other protocols; Software control, plug-ins, and integration into Digital Audio Workstations; Networks and mLAN—how it addresses issues of connection management.

Workshop 6
Friday, October 29 **2:00 pm – 4:00 pm**
Room 308

MASTERING FOR LOW BIT-RATE PERCEPTUAL CODECS

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

Panelists: *John Arthur,* Apple Computer, Inc., Cupertino, CA, USA
Bob Katz, Digital Domain, Alamonte Springs, FL, USA
John Loose, Dolby Laboratories, Inc., San Francisco, CA, USA

Most DVD-V clients are interested in perceptual audio coding (e.g., Dolby Digital and DTS 5.1 surround). The convenience of MP3 and AAC (e.g., iTunes) has attracted a far wider audience than any high resolution audio format could ever hope for. Satellite radio, working with extremely low bit-rates, has also had good success. All sound engineers need to know about codecs and what kinds of signal processing needs to be done in order to achieve commercially acceptable sounding low bit-rate streams. The discussion will include: A quick overview of perceptual codecs and their history; What bit-rates are sufficiently high that *NO* premastering is necessary?; Mastering for medium bit-rate streams; Extreme signal processing for the lowest bit-rate streams. Audio examples will be provided as applicable.

Workshop 7
Friday, October 29 **4:30 pm – 6:30 pm**
Room 308

HOW WE MADE GAMES IN THE HALLWAY ROCK!

Chair: **Martin Wilde,** Motorola, Inc., Schaumburg, IL, USA

Panelists: *Murray Allen,* San Francisco, CA, USA
Buzz Burrowes, Sony Corporation, San Francisco, CA,USA
Jack Buser, Dolby Laboratories Inc., San Francisco, CA, USA
Brian Schmidt, Microsoft Corporation, Redmond, WA, USA

Attendees will likely have experienced the audio capabilities of many different games and gaming platforms at the convention. The audience will hear first-hand from the companies providing this equipment how each of them went about making this as good an audio exhibit as it could be. The audio design and considerations for this event are numerous. Topics for discussion will include: Design of the walk-up gaming stations; Loudspeaker design and performance considerations for games; PC solutions for excellent game audio; Acoustical treatment challenges of this hostile situation.

Workshop 8
Saturday, October 30 **9:00 am – 11:00 am**
Room 308

ARCHITECTURAL ACOUSTICS FOR FILM AND BROADCAST STUDIOS

Chair: **David R. Schwind,** Charles M. Salter Associates, Inc., San Francisco, CA, USA

Panelists: *George L. Augsburger,* Perception, Inc., Los Angeles, CA, USA
Russ Berger, Russ Berger Design Group, Dallas, TX, USA
Tom Holman, TMH Corporation, Los Angeles, CA, USA
Jan Voetmann, DELTA Acoustics, Odense, Denmark

The digital era is bringing numerous changes to film and television audio. This workshop will review aspects of good architectural acoustic design practice for new facilities. Topics for discussion, in the form of project case studies, will include: Planning considerations and design criteria; Internal and external sound isolation; Use of Noise Criteria (NC) and Room Criteria (RC); Noise reduction; Structure-borne noise; Proper HVAC system design; Room shape and its influence on acoustics.

Workshop 9
Saturday, October 30 **11:30 am – 1:30 pm**
Room 308

WHICH AUDIO RECORDING AND STORAGE MEDIUM FOR WHAT PURPOSE?

Chair: **Derk Reefman,** Philips Research, Eindhoven, The Netherlands

Panelists: *Chris Chambers,* BBC Research, London, UK
Kimio Hamasaki, NHK, Tokyo, Japan
Nicolas Hans, Dalet, Paris, France
Kieran Maloney, Exabyte Corporation, Boulder, CO USA
Masaaki Shinmachi, Fostex, Tokyo, Japan

Many people have their preferences for a certain type of storage medium but without unanimous scientific/technical reasoning. The AES Technical Committee on Audio Recording and Storage Systems (TC-ARSS) has also identified the need to discuss various audio data storage media. This workshop tries to bring together experts on different storage media, such that a discussion of the pros and cons of different media can be started that has roots in real science. In addition, the user perspective will be detailed by an end-user panelist. Areas of discussion and storage media will include: digital tape; optical disk; hard disk; distributed network/local storage; end-users for both recording and archiving; views on the future.

Workshop 10
Saturday, October 30 **2:00 pm – 4:00 pm**
Room 306

THE POWER OF LOUDSPEAKER MODELS

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

Panelists: *Andrew Bright,* Nokia Research Centre, Tampere, Finland
David Clark, DLC Design, Wixom, MI, USA
Jürgen Ringlstetter, Harman-Becker, Straubing, Germany
Richard Small, Harman-Becker, Martinsville, IN, USA

With the progress in loudspeaker modeling new parameters are introduced that describe essential properties of loudspeakers more concisely. The workshop gives an overview of linear, nonlinear and thermal models and associated parameters and shows the practical application for loudspeaker design, diagnostics, system development, and active loudspeaker control. The validity of the models and reproducibility of parameters are discussed and illustrated with practical demonstrations. The workshop reveals the borderline between established theory and unmodeled but significant effects, which are the subject of further research.

Workshop 11
Saturday, October 30 **4:30 pm – 6:30 pm**
Room 306

SIGNAL PROCESSING AND HEARING AIDS

Chair: **Brent Edwards,** Starkey Laboratories, Berkeley, CA, USA

Panelists: *Andrew Dittberner,* GN ReSound, Chicago, IL, USA
Mead Killion, Etymotic Research, Elk Grove Village, IL, USA
Neeraj Magotra, Texas Instruments, Dallas, TX, USA
Robert Shannon, House Ear Institute, Los Angeles, CA, USA
Fan-Gang Zeng, University of California at Irvine, Irvine, CA, USA

The focus of the hearing industry has been on the development of signal processing features enabled by DSP technology, such as noise reduction, feedback cancellation, and environment recognition. The purpose of this panel discussion will be to discuss where the industry should be focusing its effort with regard to signal processing development, what the panel foresees in the development of DSP algorithms in the future, what problems currently exist with signal processing in hearing aids, and what should be the forward-looking strategy for DSP research and development. The discussion will involve issues, current problems, and future development regarding: compensation for hearing impairment; Improved functionality of hearing aids; quality issues related to signal processing; subjective and objective measures of benefit; and limitations to what signal processing can do today and in the future.

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Workshop 12
Sunday, October 31 **9:00 am – 11:00 am**
Room 308

THE ROLE OF MULTIPLE LOW-FREQUENCY SIGNALS IN THE PERCEPTION OF REPRODUCED SOUND

Chair: **William Martens,** McGill University, Montréal, Canada

Panelists: *Jonas Braasch,* McGill University, Montréal, Canada
David Griesinger, Lexicon, Inc., Bedford, MA, USA
Geoff Martin, Bang & Olufson, Struer, Denmark
Robert (*Robin) Miller III*, Filmaker Technology, Bethlehem, PA, USA
Todd Welti, Harman International Industries, Northridge, CA, USA (see Convention Paper 6322)

This workshop will examine the relative value of reproducing more than a single channel low-frequency (i.e., subwoofer) signal in two-channel and multichannel stereophonic sound reproduction. The emphasis will be on the differences that people can hear when presented with two or more subwoofer signals, rather than on optimizing bass management schemes for conventional 5.1 channel surround sound. Topics for discussion include: At what frequency can low-frequency sound be presented via a single reproduction channel without degrading spatial imagery? (And conversely, at what frequency does low-frequency sound need be presented via more than one channel?); Besides the detectability of differences, what are the perceptual consequences of reproducing more than a single channel low-frequency (i.e., subwoofer) signal in two-channel and multichannel stereophonic sound reproduction? What is the best way to handle two (or more) LFE signals? What is best when there are none? What should be recommended when considering costs versus benefits?

Workshop 13
Sunday, October 31 **11:30 am – 1:30 pm**
Room 308

FUTURE CHALLENGES FOR THE ALL-SOFTWARE STUDIO: SCALABILITY, STABILITY, USABILITY

Chair: **Dana Massie**, Waves, Inc., London, UK

Panelists: *Todor Fay*, New Blue, Inc., La Jolla, CA, USA
Adrian Freed, Center for New Music and Audio Technologies, Berkeley, CA, USA
David Gibbons, Digidesign, Daly City, CA, USA
Gerhard Lengeling, Apple Computer, Cupertino, CA, USA
Roger Linn, Roger Linn Design, Berkeley, CA, USA
Guy McNally, Pinnacle Systems, Mountain View, CA, USA
Andy Moorer, Adobe Systems, Inc, Panacea, FL, USA

With the all-software studio, our fundamental mode of interaction with the studio has changed. Studio modules are now often software plug-ins. We can now edit, transform, and manipulate with unprecedented ease and quality and produce digital effects that have no counterpart in the analog world. However, there are new challenges: Can any professional studio tolerate a workstation crash from an unstable operating system in the middle of a billable session? Can a professional studio tolerate gaps in audio from real time delays in the delivery of audio packets? Likewise, can any studio tolerate latency of many tens of milliseconds in an effects loop for live monitoring of a record track? How can a studio incrementally add more processing power? User interfaces can also be nonstandard and complex, requiring big investments in training, and creating obstacles to entry for new engineers.

Workshop 14
Sunday, October 31 **2:00 pm – 4:00 pm**
Room 308

LOSSLESS AUDIO CODING: MPEG AND DE FACTO STANDARDS

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

Panelists: *Peter Craven,* Algol Applications, Steyning, UK
Ralf Geiger, Fraunhofer IDMT, Ilmenau, Germany
Tillman Liebchen, Technical University of Berlin, Berlin, Germany
Werner Oomen, Philips, Eindhoven, The Netherlands

Lossless audio coding is becoming increasingly used in the context of "high-definition audio" and archiving. Besides established schemes in consumer applications (e.g., MLP for DVD-Audio), new technology is being developed within ISO/MPEG. This workshop will provide an overview of widely deployed current systems for lossless audio coding and their applications. Special attention will be given to the technology developed by the ongoing MPEG work on this topic that provides a number of novel features compared to existing systems, such as combined lossy/lossless audio coding, fine grain scalability, and compression of floating point audio.

Tutorial Seminars

TS-1	**Physics of Sound and Hearing**	Thursday, Oct. 28 9:00 am – 11:00 am Room 303
TS-2	**Subjective Microphone Comparisons**	Thursday, Oct. 28 9:00 am – 11:00 am Room 302
TS-3	**Acoustics, Part 1**	Thursday, Oct. 28 1:30 pm – 3:30 pm Room 303
TS-4	**Audio Postproduction**	Thursday, Oct. 28 1:30 pm – 3:30 pm Room 304
TS-5	**Acoustics, Part 2**	Thursday, Oct. 28 4:00 pm – 6:00 pm Room 303
TS-6	**Live Organ Concert Recording**	Thursday, Oct. 28 6:00 pm – 8:30 pm Grace Cathedral
TS-7	**The Basics of Digital Audio: A Seminar with Demonstrations**	Friday, Oct. 29 9:00 am – 11:00 am Room 303
TS-8	**Microphone Techniques for Music**	Friday, Oct. 29 9:00 am – 11:00 am Room 304
TS-9	**All About Looping and Time-Stretch Technologies**	Friday, Oct. 29 11:30 am – 1:30 pm Room 303
TS-10	**Center Channel Challenge**	Friday, Oct. 29 11:30 am – 1:30 pm Room 304
TS-11	**Digital Plumbing for Studio, Broadcast, and Live Audio**	Friday, Oct. 29 2:00 pm – 4:00 pm Room 303
TS-12	**Design and Use of Microphone Preamplifiers**	Friday, Oct. 29 2:00 pm – 4:00 pm Room 301
TS-13	**Planning for Success in Live Sound Reinforcement: Practical Considerations for Performance Audio**	Friday, Oct. 29 4:30 pm – 6:30 pm Room 303
TS-14	**Designing with Delta-Sigma Converters**	Saturday, Oct. 30 4:30 pm – 6:30 pm Room 301

TS-15	**Audio Testing According to AES-17, A Guided Tour to Implementing the Standard**	Saturday, Oct. 30 9:00 am – 11:00 am Room 303
TS-16	**Surround Sound for Motion Pictures—Dramaturgical Goals, Tools, and Concepts**	Saturday, Oct. 30 9:00 am – 11:00 am Room 304
TS-17	**Grounding and Shielding**	Saturday, Oct. 30 11:30 am – 1:30 pm Room 303
TS-18	**Dynamic Range Compression**	Saturday, Oct. 30 11:30 am – 1:30 pm Room 304
TS-19	**Analog Design in a Digital Environment**	Saturday, Oct. 30 2:00 pm – 4:00 pm Room 303
TS-20	**Reverberation Acoustics, Analysis, and Synthesis**	Saturday, Oct. 30 2:00 pm – 4:00 pm Room 304
TS-21	**DSP Part 1: Architectures and Hardware**	Saturday, Oct. 30 4:30 pm – 6:30 pm Room 303
TS-22	**Mastering for Stereo and Surround**	Saturday, Oct. 30 4:30 pm – 6:30 pm Room 304
TS-23	**Synchronization**	Sunday, Oct. 31 9:00 am – 11:00 am Room 304
TS-24	**DSP Part 2: Algorithms and Software**	Sunday, Oct. 31 11:30 am – 1:30 pm Room 303
TS-25	**Acoustic Issues Concerning Small Studio Environments-**	Sunday, Oct. 31 11:30 am – 1:30 pm Room 304

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Tutorial Seminar 1
Thursday, October 28 **9:00 am – 11:00 am**
Room 303

PHYSICS OF SOUND AND HEARING

Presenter: **J. J. Johnston**, Microsoft, Redmond, WA, USA

In this tutorial we will present some basic acoustic and psychoacoustic issues and phenomenon. We will relate them to the practice of digital audio. In the process, we will discuss:

- Low SPL's and how they relate to the noise at the eardrum.
- High SPL's and the nonlinearity of atmospheric transmission.
- Low and High frequencies and how they propagate.
- How one ear hears; outer, middle, and inner ear; and the cochlear filters.
- What is masking and why do I care about it?
- How binaural hearing, HRTFs, and soundfields interact.
- Why is SNR "mostly useless"?

Tutorial Seminar 2
Thursday, October 28 **9:00 am – 11:00 am**
Room 302

SUBJECTIVE MICROPHONE COMPARISONS

Chair: **Jürgen Wahl,** Sennheiser/Neumann, Van Nuys, CA, USA

The purpose of this seminar is to analyze the variables that make it so difficult to predict a microphone's performance in actual applications, and to understand why microphones with seemingly identical technical specifications sound differently, even when used under the same circumstances.

The seminar will demonstrate how to concentrate on less complex segments of performance behavior. For example, when evaluating electronic performance, we can concentrate on good signal-to-noise ratio, low self-noise during very quiet passages, and distortion components in the nonlinear operating range. To evaluate the microphone's acoustic behavior we listen for the imaging of instruments, how it captures room acoustics, reverberation, ambience, and distant instruments. When we analyze the tonal characteristic of the microphone under test, we may include the natural frequency response for all instruments, the extended frequency range, the transient response, the uniform polar pattern, the detailed resolution of harmonic components, and how the microphone works together with other microphones.

Tutorial Seminar 3
Thursday, October 28 **1:30 pm – 3:30 pm**
Room 303

ACOUSTICS, PART 1

Presenter: **Anthony Grimani**, Performance Media Industries, Ltd., Fairfax, CA, USA

The acoustical properties of a room highly influence the perceived sonic quality of any monitoring system. Tutorials 3 and 5 will focus on applications for small room acoustics, such as recording studios, project studios, listening rooms, screening rooms, home theaters. The information will cover all the details that affect acoustics and will provide simple recipes for improving quality and consistency. Basic knowledge of audio theory, wave theory, and acoustics are recommended. This is the first of two acoustics tutorials and covers topics such as: sound isolation through decoupling, mass, and damping; background noise control; vibration and rattle control.

Tutorial Seminar 4
Thursday, October 28 **1:30 pm – 3:30 pm**
Room 304

AUDIO POSTPRODUCTION

Presenter: **Tomlinson Holman**, USC Cinema-Television and TMH Corporation, Los Angeles, CA, USA

Audio produced for accompanying pictures has some specialized considerations that do not appear elsewhere in the field. This tutorial will emphasize these topics. Among them are audio/video synchronization, dialogue and sound effects editing and mixing, and specialized forms of tools used elsewhere in audio such as equalization, noise reduction, and so forth. Multichannel sound is the norm here, and since the sound accompanies a picture, issues of picture/sound spatial matching apply. Another factor is the multiple stages of the workflow needed to tame the potentially hundreds of tracks in use.

Tutorial Seminar 5
Thursday, October 28 **4:00 pm – 6:00 pm**
Room 303

ACOUSTICS, PART 2

Presenter: **Anthony Grimani**, Performance Media Industries, Ltd., Fairfax, CA, USA

The acoustical properties of a room highly influence the perceived sonic quality of any monitoring system. Tutorials 3 and 5 will focus on applications for small room acoustics, such as recording studios, project studios, listening rooms, screening rooms, home theaters. The information will cover all the details that affect acoustics and will provide simple recipes for improving quality and consistency. Basic knowledge of audio theory, wave theory, and acoustics is recommended. This is the second of two acoustics tutorials. Covered topics include: room dimensioning for optimized bass resonance; acoustical treatments for optimized reflections, echoes, and decay time, which include resistive absorbers, Helmholtz resonator absorbers, tympanic and pistonic resonator absorbers, and various forms of diffusers.

Tutorial Seminar 6
Thursday, October 28 **6:00 pm – 8:30 pm**
Grace Cathedral

LIVE ORGAN CONCERT RECORDING

Chair: **Ron Streicher,** Pacific Audio-Visual Enterprises, Pasadena, CA, USA

This tutorial will center on the setup and recording of Graham Blyth's Organ Concert at Grace Cathedral. It will be a 'hands-on" session, with direct involvement of any and all who attend. Participants should come prepared to work. We will do everything required to setup and record this concert—from unloading the van of equipment through to reloading following the concert. All aspects of this process will be thoroughly discussed during the setup process:

- Microphone selection and placement
- Microphone stand security and safety
- Cable routing and taping for safety
- Equipment configuration: the recording chain
- Choice of recording formats for stereo and surround
- ProTools session setup for stereo and surround
- Multitrack (DA78HR) equipment setup
- The "on-location monitoring dilemma" (i.e., headphones or loudspeakers)
- Techniques for logging the session (i.e., the "paperwork")
- Striking the session: cable coiling the RIGHT WAY.

Tutorial Seminar 7
Friday, October 29 **9:00 am – 11:00 am**
Room 303

THE BASICS OF DIGITAL AUDIO:
A Seminar with Demonstrations

Presenter: **Stanley Lipshitz and John Vanderkooy**, Audio Research Group, University of Waterloo, Canada

This is an introductory-level seminar aiming to explain and demonstrate with "live" examples the two fundamental aspects of any digital audio system—sampling and quantization. These two operations will be discussed and illustrated in real-time using a custom-built sampler and quantizer. This will enable us to present some of the pathologies of such systems, which should not normally be audible, and also show that, when properly implemented, a digital system has analog characteristics. This will make the presentation interesting to newcomers and "old pros" alike.

Topics to be covered will include:

- Sampling only (without quantization)
- Sampling artifacts (aliases & images)
- Reconstruction
- Quantization only (without sampling)
- Quantization errors
- Dither

The demonstrations will enable the audience to hear and see what is going on, both good and bad.

Tutorial Seminar 8
Friday, October 29 **9:00 am – 11:00 am**
Room 304

MICROPHONE TECHNIQUES FOR MUSIC

Chair: **Bruce Bartlett,** Crown International, Elkhart, IN, USA

This presentation covers the theory and practice of microphone techniques for music recording (microphone choice and placement). The first half of the presentation explains microphone transducer types, frequency response, polar patterns, and the advantages of each. Some basic guidelines are presented that apply to all microphone techniques. The second half describes several microphone techniques in common use for specific instruments and vocals. Both multimiking and stereo miking are covered.

Tutorial Seminar 9
Friday, October 29 **11:30 am – 1:30 pm**
Room 303

ALL ABOUT LOOPING AND TIME-STRETCH TECHNOLOGIES

Presenter: **Craig Anderton,** EQ Magazine

The ability to change the pitch and timing of digital audio is essential for various types of music, particularly modern electronic, and dance music. Several technologies have been applied to this problem, but as yet, there is no “one size fits all” solution. This tutorial explains the different technologies, which are best suited to which applications, how to optimize stretching technologies, workarounds for existing problems, and how these technologies are applied in musical contexts. Extensive usage of real world audio examples expand on and illustrate the concepts presented in this tutorial.

Tutorial Seminar 10
Friday, October 29 **11:30 am – 1:30 pm**
Room 304

CENTER CHANNEL CHALLENGE

Presenters: **Jeff Levison,** DTS Entertainment, Agoura Hills, CA, USA

The center channel has long been the audio image anchor for the cinema but has found difficulty fitting into easy use for multichannel music. This tutorial will examine a variety of mixing techniques for the center channel and its incorporation in popular music by evaluating artistic stereo goals and translating them into multichannel.

Tutorial Seminar 11
Friday, October 29 **2:00 pm – 4:00 pm**
Room 303

DIGITAL PLUMBING FOR STUDIO, BROADCAST, AND LIVE AUDIO

Presenter: **Michael Poimboeuf,** Digidesign, Daly City, CA, USA

This seminar presents an overview of commonly used digital audio interconnection technologies including cabling, circuits, modulation theory, and analysis techniques for performance measures. The interconnect technologies we study include AES-3 (AES/EBU), IEC60958 (S/PDIF), and IEEE802.3 (Ethernet). Cabling includes shielded 110-ohm twisted pair, UTP and ScTP "category" cable (Cat5/5e/6), and 75-ohm coaxial cable. Circuits include clock/data recovery and PLLs. Analysis techniques and performance measures range from eye-diagrams, and transport jitter measurements, to bit error rate (BER) estimation based on signal-to-noise measures such as NEXT, FEXT, and Alien Crosstalk. We conclude the tutorial with a discussion of new interconnection technology currently in development.

Tutorial Seminar 12
Friday, October 29 **2:00 pm – 4:00 pm**
Room 301

DESIGN AND USE OF MICROPHONE PREAMPLIFIERS

Presenter: **John LaGrou,** Millennia Music & Media Systems, Placerville, CA, USA

Microphone preamplifiers are one of the few essential analog functions in an increasingly digital audio world. In this tutorial we will discuss microphone preamplifier application and technology. Can we have too many mic preamps in our studio? Attend this tutorial and find out.

Tutorial Seminar 13
Friday, October 29 **4:30 pm – 6:30 pm**
Room 303

PLANNING FOR SUCCESS IN LIVE SOUND REINFORCEMENT: Practical Considerations for Performance Audio

Presenter: **David Scheirman**, JBL Professional, Northridge, CA, USA
Including a Panel of Leading Live Sound Professionals

Of interest to students, working system operators, and audio system/product designers alike, this seminar presents an overview of the challenges faced by sound system technicians and operators who must deal with complex interactions between performers, audiences, acoustical spaces, and environmental challenges. Topics include how to plan in advance for successful system deployments, understanding gain stages and the signal path from microphone to loudspeaker, and various decision paths taken by practitioners who balance audio engineering theory with practical applications in the field. Various tips and tricks related to stage plots, input lists, mixing console configurations, and loudspeaker array formats will highlight the session. This seminar will investigate these issues in depth, drawing upon the combined experiences of several working live sound professionals including a high-profile independent concert sound mixer, a leading rental sound company owner, and a noted system engineer.

Tutorial Seminar 14
Friday, October 31 **4:30 pm – 6:30 pm**
Room 301

DESIGNING WITH DELTA-SIGMA CONVERTERS

Presenter: **Steve Green**, Cirrus Logic/ Crystal Semiconductor, Austin, TX, USA

The performance of integrated analog-to-digital and digital-to-analog converters integrated circuits continues to improve as new techniques and processes become available to the IC design engineer. Many subtleties must be understood and addressed in order to realize optimal performance of these devices.

Tutorial Seminar 15
Saturday, October 30 **9:00 am – 11:00 am**
Room 303

AUDIO TESTING ACCORDING TO AES-17, A GUIDED TOUR TO IMPLEMENTING THE STANDARD

Chair: **Richard Cabot,** Qualis Test Systems, USA

The AES digital audio measurement standard has existed for more than a decade and is accepted as the definitive document when assessing the performance of professional digital audio equipment. This seminar will cover the various performance parameters and the measurements used to quantify them. Their origin, the rationale behind their use and the limitations of their application will be discussed. Each implementation of each measurement in the standard will be diagrammed in terms of the equipment required for implementation. Where several implementation approaches are possible, the advantages and disadvantages of each will be considered.

Tutorial Seminar 16
Saturday, October 30 **9:00 am – 11:00 am**
Room 304

SURROUND SOUND FOR MOTION PICTURES DRAMATURGICAL GOALS, TOOLS AND CONCEPTS

Presenter: **Florian Camerer,** ORF, Austrian TV, Vienna, Austria

5.1 surround sound is primarily established as a means to heighten the experience of the recipient of a combined audio-visual event, mainly in the movie world and in its wake the success of the DVD in home theaters. There exists a set of dramaturgical tools for raising the listening/viewing experience through proper use of the additional channels. Nevertheless many concepts are exclusively based on long-term work experience of "what works and what doesn't work." In this tutorial several surround-sound- design-concepts are traced down to deeper psychological roots. New terms are introduced along with advanced solutions going beyond what is possible (dramaturgically) in the cinematic world. The main body of examples of the author come from the TV-documentary area—one of the mixes will even be dismantled into the different stem mixes to illustrate the process of sound-track-crafting with special focus on surround sound.

Tutorial Seminar 17
Saturday, October 30 **11:30 am – 1:30 pm**
Room 303

GROUNDING AND SHIELDING

Presenter: **Bill Whitlock,** Jensen Transformers, Inc., Van Nuys, CA

Many designers and installers of audio/video systems think of grounding and interfacing as a "black art." Do signal cables really "pick up" noise, presumably from the air like a radio receiver? Equipment manufacturers, installers, and users rarely understand the real sources of system noise and ground loop problems, routinely overlooking or ignoring basic laws of physics. Although myth and misinformation are epidemic, this tutorial brings insight and knowledge to the subject.

Signals accumulate noise and interference as they flow through system equipment and cables. Both balanced and unbalanced interfaces transport signals but are also vulnerable to coupling of interference from the power line and other sources. The realities of ac power distribution and safety are such that some widely used noise reduction strategies are both illegal and dangerous. Properly wired, fully code-compliant systems always exhibit small but significant residual voltages between pieces of equipment as well as tiny leakage currents that flow in signal cables. The unbalanced interface has an intrinsic problem, common-impedance coupling, making it very vulnerable to noise problems. The balanced interface, because of a property called common-mode rejection, can theoretically nullify noise problems. Balanced interfaces are widely misunderstood, and their common-mode rejections suffer severe degradation in most real-world systems. Many pieces of equipment, because of an innocent design error, have a built-in noise coupling mechanism dubbed the "pin 1 problem" by Neil Muncy. A simple troubleshooting method that uses no test equipment will be described. It can pinpoint the exact location and cause of system noise. Most often, devices known as ground isolators are the best way to eliminate noise coupling. Signal quality and other practical issues are discussed as well as how to properly connect unbalanced and balanced interfaces to each other. While immunity to RF interference is a part of good equipment design, it must often be provided externally. Finally, power line treatments such as technical power, balanced power, power isolation transformers, and surge suppression are discussed.

Tutorial Seminar 18
Saturday, October 30 **11:30 am – 1:30 pm**
Room 304

DYNAMIC RANGE COMPRESSION

Presenters: **David Berners** and **Jonathan Abel**, Universal Audio & CCRMA Stanford University, CA, USA

This tutorial section covers automatic gain control (AGC), with applications including dynamic range compressors, limiters, gates, and companders. Topics include signal level detection techniques, gain control topologies, and DSP modeling techniques. Parameters such as attack, release, and ratio will be discussed in relation to perceptual color and application. Properties of various technologies including opto-, FET-, and VCA- based compression will be covered, also in relation to desired behavior for various applications. Program dependence, lookahead, and sidechaining will also be discussed. Time permitting, the tutorial will cover de-essing, multichannel compression, and multiband compression.

Tutorial Seminar 19
Saturday, October 30 **2:00 pm – 4:00 pm**
Room 303

ANALOG DESIGN IN A DIGITAL ENVIRONMENT

Presenters: **Dennis Bohn, Rick Jeffs, Paul Mathews,**
Rane Corporation, Mukilteo, WA, USA

This tutorial presents a fast-paced overview of the problems faced by an analog audio designer working in the mixed analog-digital environment found in most professional audio products. A typical mixed analog-digital audio product is examined with respect to the analog design elements necessary to maintain pristine audio performance while satisfying international EMC and safety compliance. Topics include how to bring in low-level signals, maintain fidelity and SNR, provide high gain and buffering, supply clean power, and deliver high quality signals on the output side, all within the context of a hostile environment both inside and outside the product housing. Examples of gotchas and dos and don'ts in chassis design, circuit design, and circuit board layout highlight the session.

Tutorial Seminar 20
Saturday, October 30 **2:00 pm – 4:00 pm**
Room 304

REVERBERATION ACOUSTICS, ANALYSIS, AND SYNTHESIS

Presenters: **Jonathan Abel** and **David Berners,** Universal Audio & CCRMA Stanford University, Stanford, CA, USA

This tutorial presents a review of reverberation analysis and synthesis methods, along with a discussion of reverberation acoustics and psychoacoustics. Although the details of reverberated sound are extremely sensitive to environment and geometry particulars, human listeners appear to pay attention only to certain statistical features, such as decay rate as a function of frequency. In this tutorial, we explore these features and their physical origins, and present signal processing structures to synthesize them. The psychoacoustics of early reflections and the late-field envelope are reviewed. Artificial reverberators based on convolution and on feedback delay networks that synthesize desired early reflection and late-field features, are described.

Tutorial Seminar 21
Saturday, October 30 **4:30 pm – 6:30 pm**
Room 303

DSP -1: ARCHITECTURES AND HARDWARE

Presenter: **Dennis Fink,** Mathematical Systems Design & Universal Audio

DAWs, rack mount signal processors, DSP cards, and effects pedals tend to use DSP chips, microprocessors or microcontrollers as their engines. In this tutorial, we will look at and compare system architectures for these products and the hardware devices used to implement them. We will focus on the common attributes of DSP chips required by audio algorithms and the transport of digital signals in and out of the chip. A brief comparison of commercially available DSP chips and a historical timeline of DSP hardware will be presented as well. This is a detailed overview for those of us not actively working as digital audio hardware engineers.

PLEASE NOTE: Daylight Saving Time ends at 2 am October 31; please remember to set your watches and clocks back one hour.

Tutorial Seminar 22
Saturday, October 30 **4:30 pm – 6:30 pm**
Room 304

MASTERING FOR STEREO AND SURROUND

Presenter: **Bob Ludwig**, Gateway Mastering, Portland, ME, USA

Mastering remains a somewhat misunderstood topic. For all the press on the recording process there are only a few good books available that discuss the mastering process. Mastering is the final creative step in the record making chain. It determines how the final CD or DVD will sound. Then it becomes the first step in the manufacturing chain, creating the production master from which all the replicated discs will be made. Bob will discuss many aspects of mastering with emphasis on the fact that mastering is not just making an original recording louder. There are many creative decisions that need to be made. He will focus on a few specific titles he has worked on and cover what was involved in their mastering.

Tutorial Seminar 23
Sunday, October 31 **9:00 am – 11:00 am**
Room 304

SYNCHRONIZATION

Presenter: **Fred Katz,** Omega Recording Studios, Rockville, MD, USA

Perhaps no subject in audio engineering is more misunderstood by students and working professionals alike than synchronization. Its principles are derived from seemingly unrelated fields such as television and video production, the math involved is unwieldy and often counterintuitive, and when it is applied successfully its results are transparent and easily overlooked. Yet the topic of synchronization is relevant to almost every type of recording project, especially those involving digital audio, video or multimedia, and MIDI and sampling. This presentation will attempt to clarify the fundamental principles of synchronization and will show how they can be applied by examining several common synchronization scenarios involving analog, digital, and video tape, MIDI, Pro Tools, and digital audio networks.

Tutorial Seminar 24
Sunday, October 31 **11:30 am – 1:30 pm**
Room 303

DSP -2: ALGORITHMS AND SOFTWARE

Presenter: **Dennis Fink,** Mathematical Systems Design & Universal Audio

Digital equalizers, mixers, crossovers, and limiters found in plug-ins or hardware all use variants of a common set of algorithms. This tutorial looks at these audio processes and the time and frequency domain algorithms used to implement them. We will talk about analog versus digital processing, algorithm times, sample- versus buffer-processing, DSP utilization, and parameter control. We will look at high level and assembly language programming and consider the five major algorithm classes: IIR filters, FIR filters, interpolator/decimators, modulators, and FFTs. This is a detailed overview of what audio DSP algorithm designers and programmers do.

Tutorial Seminar 25
Sunday, October 31 **11:30 am – 1:30 pm**
Room 304

ACOUSTIC ISSUES CONCERNING SMALL STUDIO ENVIRONMENTS

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

Contrary to many beliefs, small studio environments present a special breed of acoustic issues, some in common but many unique to the small size of both recording and listening rooms. This tutorial will investigate many of these issues, including: low frequency control; loudspeaker placement; ergonomic design, including listening positioning, furniture design, and other object effects on room acoustic response; currently available design, prediction, and measuring tools. Also included in this tutorial will be demonstrations of some of these applications and real world experiences with pictures and diagrams of many "small studio" environments, tips, and tricks.

Exhibitor Seminars

Exhibitor Seminars are presentations by Exhibitors at the 116th Convention giving more in-depth information about their products than they are able to give in their booth. It is a unique opportunity for exhibitors to be able to explain the technical background and features of a product to an audience in a seminar style.

Please refer to the separate Exhibitor Seminar booklet for participating companies, their seminar descriptions, and their location.

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