

CONTENTS
132ND TECHNICAL MEETINGS
AND PROFESSIONAL EVENTS

<i>Message from the AES President</i>	3
<i>Message from the Convention Chair</i>	5
<i>General Information:</i>	
Registration Information	7
Technical Papers, Workshops, Tutorials, and Special Events Hours	8
Exhibit Hours	8
Membership Services	8
AES Publications	8
132nd Convention Papers	8
Public Transportation Information	9
<i>Opening Ceremonies:</i>	
Awards Presentation and Keynote Speaker	11
<i>Technical Tours</i>	13
<i>Special Events</i>	16
<i>Student Events</i>	21
<i>Technical Council and Technical Committee</i>	
<i>Meetings</i>	25
<i>Standards Committee Meetings</i>	26
<i>Technical Paper Sessions</i>	27
<i>Engineering Briefs</i>	91
<i>Workshops Program</i>	101
<i>Tutorials Program</i>	119
<i>Cumulative Index</i>	137



Jan A. Pedersen

Welcome to the 132nd Audio Engineering Society Convention! For the first time, the European AES Convention is hosted in the beautiful city of Budapest! I am sure that all of us will enjoy a unique combination of all the benefits of an AES Convention and the beautiful sights and life in Budapest. The scene is set for 4 days of extraordinary high quality professional and social life for everyone working with Audio—recording, education, loudspeakers, broadcasting, games, coding, headphones, audio forensics, audio perception, DSP, automotive audio, and much more.

An AES Convention offers a multitude of opportunities—and I will encourage you to also explore other possibilities than the ones you already planned or know from earlier conventions. This time we have 22 Paper/Poster Sessions covering all major areas of audio engineering. Both on Friday and Saturday you can enjoy presentations of Engineering Briefs, which is a new and exciting opportunity to present and share experience and knowledge in various fields of audio.

Fourteen Workshops offer yet another dimension of the convention by facilitating a more interactive way to share information between a panel of experts and the audience. Fifteen Tutorials will also allow everyone to get an in-depth introduction to specific areas of audio engineering. The Students are a vital part of the AES, which is reflected in the special Student Events that take place throughout the convention.

Please make sure to visit the Exhibition that offers the special AES possibility of Science and Business to meet—because you will meet both the business people, R&D people, educators, and all other people working with audio. The Exhibition also offers the potential of first listening to the theory in a paper presentation and later the same day, to see the product in real life and maybe initiate business relations based on this.

I would like to thank Convention Chair Janos Gyori and his great team for organizing this fantastic AES event! This thank you extends to all of the AES staff, headed up by Executive Director Bob Moses, who on a daily basis make the AES the preferred place to be for people working with audio.

I will be present throughout the whole convention at many of the activities described above, and I would appreciate it if you

have time to say hello and maybe share your experience of the convention and the AES.

I wish that you will enjoy this AES Convention and your AES Membership!

JAN A. PEDERSEN
PRESIDENT AUDIO ENGINEERING SOCIETY



Janos Gyori

Welcome to Budapest! During the previous AES conventions in the UK, I travelled several times by the London tube, the first underground railway of the world. I was thinking, since the first underground railway of the European continent was built in Budapest after the one of London, what if the next AES convention could be held in Budapest? The thought became reality, and this is the first time we welcome the AES convention to Hungary.

Budapest has a rich scientific and cultural tradition. Numerous scientists worked here, including Gyorgy Bekesy who was awarded the Nobel Prize in Physiology or Medicine for his research on the function of the cochlea in the mammalian hearing organ. Famous composers such as Bela Bartok lived here; the Music Academy of Budapest was named after its first director, Ferenc Liszt.

I am satisfied to see many practical topics in our 132nd Technical Program. We will hear about the broadcast experts' topic Loudness Control from several points of view. Presenters will share noticeable research results at Paper Sessions. Numerous research topics will be presented in the field of Spatial Audio.

We will find new and interesting topics during eBrief presentations. And for those who are involved in the pop music industry or simply fanatics of vintage technology, it will be surely interesting to take part in the tutorial "The Making of The Beach Boys *Smile* Sessions" by producers Mark Linett and Barry Marshall.

As a Tonmeister, I also expect a lot of fun at the Recording Competition for the students this year. I guess, many of us will have a problem with choosing from the several parallel sessions.

We encourage attendees to take part in the Technical Tours, as well. You will be guided around the Palace of Arts, which was built with a special isolation-technique, being close to a Danube-bridge full of traffic. Bartok-Hall is a part of this complex. Another facility being visited is the Acoustical Laboratory at the Technical University of Budapest. This would be a good choice for visitors interested in measurement in educational practice.

We invite you to the organ recital of Graham Blyth. It will be held in one of the oldest buildings of Pest, the Downtown Church. The more than 1,000-year-old church had almost been destroyed

several times in the storms of history but it has continuously been restored. The organ has recently been rebuilt, as well. In 2011, the year of Liszt, this instrument was the heart of many great concerts.

It is also a special privilege that the organist Graham Blyth will also give the Heyser Memorial Lecture entitled, "In Pursuit of Elegant Simplicity: Life, Luck, and Learning in Music and Audio." Graham, who was the co-founder of Soundcraft in 1973 is still active both on the field of audio engineering and organ design.

The Banquet will be held at a remarkable place of scientific history, in the beautiful halls of the Hungarian Academy of Sciences, from where there is a wonderful view onto the Buda Castle on the other side of the Danube. Come, and taste the famous Hungarian wines and dishes!

I wish you a valuable and pleasant convention-time in Budapest!

JANOS GYORI
CONVENTION CHAIR

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REGISTRATION

Registration Desk Hours:

Thursday, April 26	08:30 – 18:30
Friday, April 27	08:30 – 18:30
Saturday, April 28	08:30 – 18:30
Sunday, April 29	08:30 – 16:30

REGISTRATION FEES:

AES MEMBERS

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

- AES Honorary/Life Members No Fee
- AES Members & Associates 280 Euros
- AES Student Members 80 Euros

Exhibits Only (valid for 3 days):

- AES Members & Associates No Fee
- AES Student Members No Fee

NONMEMBERS

- Full Program 380 Euros
- Student (with I.D.), Full Program 130 Euros

Exhibits Only (valid for 3 days):

- Nonmembers No Fee
- Nonmember Students No Fee

INDIVIDUAL TICKETS:

- One Full Day (Members) 100 Euros
- One Full Day (Nonmembers) 150 Euros
- One Full Day (Member Students) 30 Euros
- One Full Day (Nonmember Students) 50 Euros
- Single Session Pass (Members) 30 Euros
- Single Session Pass (Nonmembers) 40 Euros

Individual tickets may be purchased at the Special Events Desk. You must have obtained a 3-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, Careers, and Education Fairs.

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

Thursday, April 26	09:00 – 18:30
Friday, April 27	09:00 – 18:30
Saturday, April 28	09:00 – 18:30
Sunday, April 29	09:00 – 17:00

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

EXHIBITS

The **Exhibits** are located in Room Aula.

Exhibit Hours

Friday, April 27	10:00 – 19:00
Saturday, April 28	10:00 – 19:00
Sunday, April 29	10:00 – 17:00

MEMBERSHIP

AES Membership Services are located in the registration area. Why not become a member of the Audio Engineering Society? The difference between the full program registration fee for non-members 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.

132ND CONVENTION PAPERS

Single Copy	€ 4
Complete Set on CD-ROM (single copies of 100 papers)	€90

PUBLIC TRANSPORTATION INFORMATION

Budapest's public transportation systems are operated by the company BKV. The service includes buses, trolleybuses, trams, underground trains (Metro), and above-ground suburban trains (HÉV). Buses, trams, and trolleybuses run daily from 4:30 a.m. until 11:00 p.m. The three Metro lines interconnect at Deák tér Station, and service frequencies range from 15 minutes late in the evening to every two minutes at peak times. The HÉV runs to and from Csepel Island and Ráckeve in the south, Szentendre in the north, and Gödöll in the east.

Tickets or passes must be bought before boarding; you cannot buy them from the driver (except on the night bus network). They are available at Metro stations, tobacconists and newsagents, and from vending machines at many bus and tram stops in the city center. The single tickets are valid for one single journey of any length (without changing) on all routes (including the Cogwheel Railway, but excluding those parts of the HÉV that lie outside the metropolitan boundary of Budapest). Special tickets can be purchased that allow a change of route, and that cover the HÉV outside Budapest. There is also a range of pre-paid books of tickets and value-for-money passes on sale.

When using single tickets, a new ticket has to be validated at the beginning of each journey, and this is done by inserting it into the slot in the small red or orange box situated at waist height near the doors of buses, trolleybuses, trams, and HÉV trains. Please note that on the Underground system these boxes are in the stations at the top of the escalator. There are many ticket inspectors: some of them are uniformed and others plain-clothed, but they always wear a red/blue armband and carry a photographic identification badge. They can ask to see tickets and passes on any vehicle and anywhere on the Metro system, including after you have got off—so please remember to validate your ticket and keep hold of it!

Maps sited at the entrances to metro stations give a comprehensive overview of the entire public transport network in Budapest. Conditions of travel are also included, in German and English.

Hungarian citizens (including those who have repatriated to Hungary and persons holding dual nationality) as well as citizens of other member states of the European Union over the age of 65 years can travel free of charge on BKV Budapest public transport services (except for the funicular, the chair-lift and boat trips) when they present to inspectors a personal identity card or document certifying their citizenship.

Opening Ceremonies

AWARDS PRESENTATION AND KEYNOTE ADDRESS

Thursday, April 26, 13:00 – 14:00

Room Bartók

- Opening Remarks:**
- Executive Director Bob Moses
 - President Jan A. Pedersen
 - Convention Chair Janos Gyori

- Program:**
- AES Awards Presentation
 - Introduction of Keynote Speaker by Convention Chair Janos Gyori
 - Keynote Address by John Buckman

Awards Presentation

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

Keynote Speaker

This year's Keynote Speaker is **John Buckman**, founder of Berkeley, CA-based Magnatune.com (subscription music for consumers), a record label known for its eclectic artist roster, its successful application of Creative Commons licensing, and its patently artist-friendly business practices. Buckman, whose egalitarian business practices are distinguished by non-exclusive agreements with musicians, sharing profits equally with them, and allowing them to retain full rights to their own music, has pioneered a philosophy known as "fair trade music." Since founding Magnatune, Buckman has signed more than 250 recording artists across multiple genres including classical, electronica, world, alt rock, jazz, and hard rock. Buckman is also the founder/owner of a string of additional successful small businesses working with music: MoodMixes.com (background music for restaurants), ToneGnome.com (audio engineering services over the Internet), and is about to launch iLicenseMusic.com. He also runs BookMooch.com (over a million books swapped per year), founded and sold Lyris Inc. (email newsletter software), and is chairman of the Electronic Frontier Foundation, a US-based Internet civil liberties organization. Profiled by *Inc. Magazine*, *The Economist*, and other major publications, Buckman is the co-author of an article in *SysAdmin Magazine* entitled "Which OS Is Fastest for High-Performance Network Applications?" and an article in *Linux Journal* entitled "Magnatune, an Open Music Experiment." The title of his keynote address is "Small and Beautiful: Models for Successful Independent Music Businesses."

Buckman will explain how to find profitable niches and how to build a small business where the work can be done by you and your staff—regardless of how many hands you have.

Technical Tours

A total of 5 Technical Tours is planned. They cater to a wide range of interests. Technical Tours are made available on a first come, first served basis. As some of the locations are working environments, we may be obliged to cancel a visit at short notice. To participate, please sign up at the AES Special Events Desk. All Tech Tours are priced at 15 Euros each, and transport will be provided.

TECHNICAL TOUR 1

Hungarian Parliament

1055 Budapest, Kossuth tér 1-3

Entrance: Gate VII

Thursday, April 26, 13:15 – 16:00

After a general visit to the beautiful buildings of the **Hungarian Parliament** lasting about half an hour, the sound- and voting-system will be introduced. There is also an HD TV network for broadcasting and archiving events happening inside the “Temple of the Nation.”

Preliminary registration is required and visitor's names have to be listed until 8:00 the same day for security reasons. There is a security check at entrance.

Capacity limited to 10 persons.

TECHNICAL TOUR 2

Budapest University of Technology and Economics, Laboratory of Acoustics and Studio Technology (LAST)

1117 Budapest, Magyar tudósok körútja 2

Entrance: Main entrance of Building I

Thursday, April 26, 16:30 – 19:30

LAST forms part of Dept. of Telecommunications of the Faculty of Electrical Engineering and Informatics and has a long established tradition in electroacoustics and audio. The current activity of the lab encompasses teaching audio recording, room and musical acoustics, sound quality engineering, and noise and vibration control of large-scale infrastructural projects—to name but a few. After a short overview of teaching and research activities, four demonstrations will be presented: subjective sound quality tests on an aircraft noise simulator, identification of sound sources by means of an acoustic antenna, AV demonstration of differently tempered scales of keyboard instruments, and simulation of air flow in and around a flue organ pipe.

Capacity limited to 20 persons.

Technical Tours

TECHNICAL TOUR 3

Palace of Arts, Béla Bartók Concert Hall

Architecture, Acoustics

1095 Budapest, Komor Marcell u. 1

Main Entrance

Friday, April 27, 10:30 – 13:30

It is a special privilege that **Palace of Arts** will be introduced by its architect himself, Mr. Gábor Zoboki on this technical tour. Hungary's National Concert Hall, designed with the feel of a Gothic cathedral, is located at the heart of the new Palace of Arts. The quality of the acoustics are guaranteed by the outstanding work of Russell Johnson, a top specialist who conducted well over fifty years of research in this field and participated in the construction of several top opera houses and concert halls. His work in Budapest was characterized by his close cooperation with the architect, the contractor, and leading Hungarian musicians. There was a special isolation technique applied at the front of the complex so that no vibration or noise can be felt or heard inside the Concert Hall. The Hall features variable acoustics, achieved by openable reverberator chambers. Its Studios will be only briefly showed on this tour.

Capacity limited to 25 persons.

TECHNICAL TOUR 4

Focus-Fox Studio

1021 Budapest, Húvösvölgyi út 141

Main Entrance

Friday, April 27, 13:30 – 16:00

To achieve something extraordinary **Focus-Fox** developed a studio with excellent sound and the best available picture quality. Combining the two operations, a shared grading and mixing studio was created and opened in 2010. The elementary acoustic design was made by the famous Canadian Pilchner and Schoustal Acoustic Design Center. The goal was to fulfill Dolby's and THX's strict requirements in an exceptionally comfortable setting. The result is an environment where real dreams come true all the time. The screening room is 43' x 49' x 16.5' (13 x 15 x 5 meters) in size while the Stewart screen itself is 29.5' x 12.5' (9 x 3,80 meters). The screening room is equipped with Baselight 4 color corrector, Christie's 2k digital projector, and a 35mm film projector. For surround mixing a "two operator" ICON console with Pro Tools HD System and an HD Adrenaline from Avid is used. A Dolby 3D system is also available.

Capacity limited to 20 persons.

TECHNICAL TOUR 5

Palace of Arts, Béla Bartók Concert Hall – Studios

1095 Budapest, Komor Marcell u. 1

Main Entrance

Saturday, April 28, 13:30 – 16:30

There are full-featured sound studios for the preparation and mastering of musical recordings at the **Palace of Arts**. The National Concert Hall serves as the main recording area of the studios, but if required, recordings can be made from the theater or the studio's own 70-m² recording area, which otherwise serves as the mastering studio. An audio, video, and duplex communications link between the recording studio and the broadcast studio makes it possible to prepare TV and video recordings or carry out dubbing and voice-over work to a high standard. The broadcasting studio fulfills the radio and television broadcasting requirements (concert and theater hall) of the facility. Both halls are linked to this studio via the house Studer system. The programs mixed on-site can be transmitted to radio and television stations via an outside broadcast truck or over dedicated ISDN lines.

Capacity limited to 20 persons.

Special Events

MIXER PARTY

Thursday, April 26, 18:30 – 19:30

Room Bartók

A Mixer Party will be held on Thursday evening to enable Convention attendees to meet in a social atmosphere to catch up with friends and colleagues from the world of audio. There will be a cash bar.

ORGAN RECITAL BY GRAHAM BLYTH

Saturday, April 28, 19:30 – 20:30

Belvárosi plébánia templom (Inner City Parish Church)

1056 Budapest, Március 15. tér 2

Graham Blyth's traditional organ recital will be given on the new organ of the Inner-City Parish Church, Belvarosi Plebania Templom. The location at the Pest side of Elisabeth Bridge, across from Gellert Hill, is originally the gravesite of the martyr, Bishop Saint Gellért. Dating to the 12th century, this is the oldest—and, in terms of architectural history, the most extraordinary—ecclesiastical building in Pest. It's actually built on even older structures—an 11th-century Romanesque church commissioned by Hungary's first king, Saint Stephen, plus the remains of the Contra Aquincum (a third-century Roman fortress and tower), parts of which are visible. There is hardly any architectural style that cannot be found in some part or another, starting with a single Romanesque arch in its south tower. The single nave, with an equally high transept and unusually slender columns, still has its original Gothic chancel and some frescoes from the 14th-15th centuries. Two side chapels contain beautifully carved Renaissance altarpieces and tabernacles of red marble from the early 16th century. During Budapest's Turkish occupation the church served as a mosque—a mihrab, a Muslim prayer niche, is a reminder. During the 18th century the church was given two baroque towers and its present facade. In 1808 it was enriched with a rococo pulpit. From 1867 to 1875 Franz Liszt lived in a town house a few steps away, where he held regular "musical Sundays" at which Richard and Cosima Wagner were frequent guests. Liszt's own musical Sunday mornings often began in this church. He conducted many masses here, including the first Budapest performance of his Missa Choralis in 1872. The church contains the relics of Saint Gellért, the bishop who was initially buried here in 1046 after pagans pushed him off a hill across the river. And in 2006 the relics of the legendary 11th-century Hungarian king, Saint László, found their way here as well.

The program will include Prelude & Fugue in B minor by Bach, Chorale No.2 in B minor by Franck, and 1st Symphony by Vienne (selected movements).

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

He divides his time between being a designer of professional audio equipment (he is a co-founder and Technical Director of Soundcraft) and organ related activities. In 2006 he was elected a Fellow of the Royal Society of Arts in recognition of his work in product design relating to the performing arts.

FUTURE DIRECTIONS IN MULTICHANNEL

Friday, April 27, 16:00 – 18:00

Room Brahms

There will be two separate presentations of new multichannel systems.

Recording Techniques for Auro 3D /3D Audio 16:00 – 17:00

Chair: **Gregor Zielinsky**, Sennheiser Electronic, Wedemark, Germany

Panelists: *TBA*

Auro3D is a rather recent additional proposal for reproduction of musical recordings including height information. Its comparatively simple, pragmatic approach makes it all the more interesting for adaption to film and home theater loudspeaker setups. Benefits of Auro3D especially for music recordings will be discussed.

For the Tonmeister then, the question arises how to produce 3-D recordings, either founding the mix mostly on spot microphones, or on new main microphone placement techniques, including height microphones. A vast set of different setups were seen at the recent ICSA 2011 Conference on Spatial Audio, in Detmold. Several microphone setups and their sonic differences will be described. The use of microphones, combinations and adjustable polar patterns will be discussed.

As in stereo, with main microphones a further question arises on if and how to delay compensate additional spot microphones,

Special Events

relative to the main microphones. Also, how need the main microphones need to be delayed in themselves.

Including height loudspeakers produces enormous differences in timbre and spatial perception. How can they be described, and explained? Finally, comments will be presented regarding the differences between single-source and multi-source (phantom source) based localisation in 3-D fields.

Mercedes-Benz Signature Sound—A mediaHYPERIUM production mixed and mastered at Skywalker Sound 17:00 – 18:00

Chair: Herbert Walth, mediaHYPERIUM, Los Angeles, CA, USA

Panelists: *Stefan Bock*, msm-studios, Munich, Germany
Mario Fresner, Daimler AG, Sindelfingen, Germany
Leslie Ann Jones, Skywalker Sound, Marin County, California, USA
Norbert Niemczyk, Daimler AG, Sindelfingen, Germany
Wieslaw Woszczyk, McGill University, CIRMA, Montreal, QC, Canada

A “first-of-its-kind” surround sound project: music mixed and mastered for the new Mercedes SL Roadster model. To create the ultimate listening experience the producer and engineers tailored discrete surround mixes to the car’s audio system, its specific acoustic environment, and the position of driver and passenger.

Current car audio systems are creating surround sound using a computer generated process, often with questionable results. The “hands-on” approach of the Signature Sound production concept aims to reflect the true artistic intention of the musical pieces in an immersive listening experience.

Daimler AG has underwritten the production and collaborated on the exclusive, limited music DVD with the title “Mercedes-Benz Signature Sound.”

We will discuss the concept, how it was done and the importance of an artistic approach in surround sound productions.

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

Friday, April 27, 19:00 – 20:00

Room Bartók

Lecturer: **Graham Blyth**

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 132nd AES Convention is Graham Blyth. Blyth was born in 1948, began playing the piano aged 4 and received his early musical training as a Junior Exhibitioner at Trinity College of Music in London, England. Subsequently at Bristol University, where he read Electrical Engineering, he founded the University Music Society, conducting their Chamber Orchestra and Choir. He holds diplomas in Organ Performance from the Royal College of Organists, The Royal College of Music and Trinity College of Music.

In 1973 he founded Soundcraft with Phil Dudderidge, and has been Technical Director from the beginning. Soundcraft has since grown to be one of the most significant designers and manufacturers of audio mixing consoles in the world. In 1988, the company was bought by Harman, whose professional audio companies now include JBL, DBX, Lexicon, AKG, and Studer.

In the late 1980s he renewed his musical studies with Sulemita Aronowsky for piano and Robert Munns for organ. He gives numerous concerts each year, principally as organist and pianist, but also as a conductor and harpsichord player. He made his international debut with an organ recital at St. Thomas Church, New York in 1993 and since then has given concerts on many of the finest organs in Europe, including the Madeleine and St. Etienne du Mont in Paris, and the Liebfrauen Dom in Munich, and in North America, including Grace Cathedral, San Francisco and St. Ignatius Loyola in New York.

He has lived in Wantage, Oxfordshire, since 1984 where he is currently Artistic Director of Wantage Chamber Concerts and Director of the Wantage Summer Festival. In 1995 he built the Challow Park Recital Hall, an 80 seat auditorium with completely variable acoustics, using the Lexicon LARES system, designed by his Harman colleague David Griesinger. This allows for performance and recording of music ranging from String Quartets through to Organ Recitals.

Today he divides his time between audio engineering and organ design activities. In 2003 he founded the Veritas Organ Company to address the top end of the digital classical organ market, specialising in creating new organs by adding digital voices to existing pipes. He is a Fellow of the Royal Society of Arts and the Audio Engineering Society.

The title of his speech is "In Pursuit of Elegant Simplicity: Life, Luck, and Learning in Music and Audio."

Blyth will talk about how he became a design engineer in the audio industry, the development of the mixing console from a personal perspective during his 41 years in the business, and, in particular, his approach to microphone preamp design with illustrated examples. He will also talk about the importance of the analog engineer in a mostly digital world and about the technical

Special Events

and musical challenges in designing high-quality digital classical organs, with audio examples. Blyth is well known to attendees at AES conventions for his popular organ recitals.

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

BANQUET

Saturday, April 28, 21:00 – 23:30

Academy Club

Hungarian Academy of Sciences

9 István Széchenyi square, Budapest

This year, the Banquet will be held at the Academy Club, part of the Hungarian Academy of Sciences. This elegant and exclusive building is a perfect setting for dining and mixing with friends and colleagues. One can also indulge in the breathtaking views of the Buda Castle and enjoy the stunning beauty of Budapest and the Danube illuminated in the evening while listening to intimate live music. An exciting choice of seasonal dishes combining the traditional Hungarian flavors with modern presentation will be offered, together with wines and other drinks.

Situated near to the church where the organ concert will be held, together these two events make for a perfect evening after a day at the convention.

60 Euros for AES members and nonmembers

Tickets will be available at the Special Events desk.

Student Events and Career Development

Student activities are open to all attendees with a full program badge. All attendees with an interest in audio education are welcome to attend.

STUDENT DELEGATE ASSEMBLY MEETING PLACE

Thursday – Sunday 9:00 – 18:30

Foyer

Come visit the SDA Booth to find out about AES student events at chapters around the world. This is also where you will see postings about finalists in the recording competition and be able to purchase tickets to the Student Social.

OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – PART 1

Friday, April 26, 14:30 – 16:00

Room Bartók

Chair: **Magdalena Plewa**

Vice Chair: **Philip Waldenberger**

The first Student Delegate Assembly (SDA) meeting is the official opening of the Convention's student program and a great opportunity to meet with fellow students from all corners of the world. This opening meeting of the Student Delegate Assembly will introduce new events and election proceedings, announce candidates for the coming year's election for the European and International Regions, announce the finalists in the four new recording competition categories, and announce any upcoming events of the Convention. Students and student sections will be given the opportunity to introduce themselves and their activities, in order to stimulate international contacts.

All students and educators are invited to participate in this meeting. Election results and Recording Competition Awards will be given at the Student Delegate Assembly Meeting–Part 2 on Sunday, April 29.

Student Events & Career

EDUCATION FORUM

Friday, April 27, 09:00 – 10:30

Room Brahms

Moderators: **Ezequiel Morfi**, Chair, AES SDA, North and Latin American Regions
Colin Pfund, Vice Chair AES SDA, North and Latin American Regions
Magdalena Plewa, Chair SDA, Europe and International Regions
Philip Waldenberger, Vice Chair, AES SDA, Europe and International Regions

Student Roundtable

Come share your unique experience as a student of audio. Bring your thoughts and perspectives to an open discussion to be moderated by the AES Student Delegate Assembly Officers who want to encourage this dialogue. How are you learning about audio? What is unique about your program and its facilities? How do co-curricular activities like the ones sponsored by AES and other organizations contribute to your experience?

Explore strategies for making connections with the professional world and discuss the curriculums and philosophies of your programs. Students, faculty, alumni, industry professionals, and anyone interested in commenting on the state of audio education are welcome to participate.

RECORDING COMPETITION—PART 1

Friday, April 27, 16:30 – 18:30

Room Bartók

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

- Traditional Non-Multitrack Acoustic Recording – 16:30–17:30
- Sound for Visual Media – 17:30–18:30

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

EDUCATION AND CAREER/JOB FAIR

Saturday, April 28, 11:00 – 12:30

Foyer

Institutions offering studies in audio (from short courses to graduate degrees) will be represented in a “table top” session. Information on each school’s respective programs will be made available through displays and academic guidance. There is no charge for schools to participate.

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

RECORDING COMPETITION—PART 2

Saturday, April 28, 16:30 – 18:30

Room Bartók

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

- Traditional Multitrack Recording – 16:30–17:30
- Non-Traditional Multitrack Recording – 17:30–18:30

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

STUDENT DELEGATE ASSEMBLY MEETING—PART 2

Sunday, April 29, 13:00 – 14:30

Room 5

At this meeting the SDA will elect a new vice chair. One vote will be cast by the designated representative from each recognized AES student section in the Europe and International Regions. Judges’ comments and awards will be presented for the Recording Competitions. Plans for future student activities at local, regional, and international levels will be summarized.

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. All meetings are in the **Bartók Meeting Room**.

THURSDAY, APRIL 26

- 09:00 Acoustics and Sound Reinforcement
- 10:00 Semantic Audio Analysis
- 11:00 Audio Recording and Mastering Systems
- 12:00 Automotive Audio
- 14:00 Archiving, Restoration, and Digital Libraries
- 15:00 Coding of Audio Signals
- 16:00 Loudspeakers and Headphones
- 17:00 Audio for Games

FRIDAY, APRIL 27

- 09:00 Human Factors in Audio Systems
- 10:00 Spatial Audio
- 11:00 Audio for Telecommunications
- 12:00 Signal Processing
- 13:00 High Resolution Audio
- 14:00 Fiber Optics for Audio
- 15:00 Network Audio Systems
- 16:00 Studio Practices and Production

SATURDAY, APRIL 28

- 09:00 Electro Magnetic Compatibility
- 10:00 Audio Forensics
- 11:00 Microphones and Applications
- 12:00 Hearing and Hearing Loss Prevention
- 13:00 Transmission and Broadcasting
- 16:00 Sound for Digital Cinema & Television (Formative)

SUNDAY, APRIL 29

- 10:00 Perception and Subjective Evaluation of Audio Signals

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

Friday, April 27 19:00 – 20:00
Room Bartók

Lecturer: **Graham Blyth**

For complete details see page 18.

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 throughout the Convention. They take place in the Fortuna room.

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 can be found at:

<http://www.aes.org/standards/meetings/>.

Contact: standards@aes.org.

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, or special events.

The information contained in this book is subject to change.

Technical Paper Sessions

P1	Applications in Audio	Thursday, April 26 09:30–11:00 Room Lehar
P2	Emerging and Innovative Audio	Thursday, April 26 09:30–11:00 Room Liszt
P3	Music and Modeling	Thursday, April 26 11:00–12:30 Room Lehar
P4	Sound Reinforcement and Studio Technologies	Thursday, April 26 11:00–12:30 Room Liszt
P5	Recording and Production	Thursday, April 26 14:30 –18:30 Room Lehar
P6	Multimodal Apps and Broadcast	Thursday, April 26 14:30–17:00 Room Liszt
P7	Perception	Thursday, April 26 17:00–18:00 Room Liszt
P8	Listening Tests: Part 1	Friday, April 27 09:00–11:00 Room Lehar
P9	Analysis and Synthesis of Sound: Part 1	Friday, April 27 09:00–12:00 Room Liszt
P10	Education	Friday, April 27 11:00–12:00 Room Lehar
P11	Posters: Quality Evaluation and Spatial Audio	Friday, April 27 12:30–14:00 Foyer
P12	Amps and Measurement	Friday, April 27 14:00–16:30 Room Lehar
P13	Analysis and Synthesis of Sound: Part 2; Content Management	Friday, April 27 14:00–16:30 Room Liszt

P14	Listening Tests: Part 2	Friday, April 27 16:30–18:00 Room Lehar
P15	Spatial Audio: Part 1	Saturday, April 28 09:00–11:30 Room Lehar
P16	High Resolution and Low Bit Rate	Saturday, April 28 09:00–12:30 Room Liszt
P17	Audio Effects	Saturday, April 28 11:30–13:00 Room Lehar
P18	Posters: Education and Human Factors; Applications	Saturday, April 28 13:00–14:30 Foyer
P19	Spatial Audio: Part 2	Saturday, April 28 14:30–17:30 Room Lehar
P20	Transducers	Sunday, April 29 09:00–11:30 Room Lehar
P21	Posters: Audio Equipment and Instrumentation	Sunday, April 29 12:00–13:30 Foyer
P22	Quality Evaluation	Sunday, April 29 14:00–16:30 Room Lehar

APPLICATIONS IN AUDIO

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

09:30

P1-1 Efficient Binaural Audio Rendering Using Independent Early and Diffuse Paths—Fritz Menzer, MN Signal Processing, Schwerzenbach, Switzerland

A multi-source binaural audio rendering structure is proposed that efficiently implements plausible binaural reverberation including early reflections and late reverberation. The structure contains delay lines and a feedback delay network that operate independently, modeling early reflections and diffuse reverberation, respectively. Computationally efficient heuristics are presented for the implementation of an HRTF set and for the diffuse reverberation. A real-time implementation on a mobile device is presented.

Convention Paper 8584

10:00

P1-2 The Hand Clap as an Impulse Source for Measuring Room Acoustics—Prem Seetharaman, Stephen P. Tarzia, Northwestern University, Evanston, IL, USA

We test the suitability of hand clap recordings for measuring several acoustic features of musical performance and recording rooms. Our goal is to make acoustic measurement possible for amateur musicians and hobbyists through the use of a smartphone or web app. Hand claps are an attractive acoustic stimulus because they can be produced easily and without special equipment. Hand claps lack the high energy and consistency of other impulse sources, such as pistol shots, but we introduce some signal processing steps that mitigate these problems to produce reliable acoustical measurements. Our signal processing tool chain is fully-automated, which allows both amateurs and technicians to perform measurements in just a few seconds. Using our technique, measuring a room's reverberation times and frequency response is as easy as starting a smartphone app and clapping several times.

Convention Paper 8585

10:30

- P1-3 Subjective Sound Quality Evaluation of a Codec for Digital Wireless Transmission**—*Matthias Frank,¹ Alois Sontacchi,¹ Thomas Lindenbauer,² Martin Opitz²*
¹University of Music and Performing Arts Graz, Graz, Austria
²AKG Acoustics GmbH, Vienna, Austria

This paper presents a subjective evaluation of a proprietary sub-band ADPCM (Adaptive Differential Pulse Code Modulation) codec for digital wireless transmission. The evaluation is carried out with 40 expert listeners and is divided into several experimental stages. First, the audibility threshold for codec artifacts is determined for each frequency sub-band, separately. In the next stage, different configurations are ranked on a scale of subjective sound quality ratings, with the resolutions varied across all bands. Finally, selected configurations corresponding to different quality ratings are compared to signals of analog wireless transmission in a multi-dimensional test. This test reveals the characteristic artifacts for each transmission method. Overall, digital transmission can achieve a better sound quality than analog transmission.

Convention Paper 8586

EMERGING AND INNOVATIVE AUDIO

Chair: **Francis Rumsey**

09:30

P2-1 Virtual Microphones: Using Ultrasonic Sound to Receive Audio Waves—*Tobias Merkel,¹ Hans-G. Lühmann,² Tom Ritter¹*

¹Beuth Hochschule für Technik, Berlin, Germany

²Lütronik Elektroakustik GmbH, Berlin, Germany

A highly focused ultrasound beam was sent through the room. At a distance of several meters the ultrasonic wave was received again with an ultrasonic microphone. The wave field of a common audio source was overlaid with the ultrasonic beam. It was found that the phase shift of the received ultrasonic signal obtains the audio information of the overlaid field. Since the ultrasonic beam itself acts as sound receiver, there is no technical device like membranes necessary at direct vicinity of sound reception. Because this kind of sound receiver is not visible or touchable we call it "Virtual Microphone."
Convention Paper 8587

10:00

P2-2 Implementation and Evaluation of Autonomous Multi-Track Fader Control—*Stuart Mansbridge, Saoirse Finn, Joshua D. Reiss*, Queen Mary, University of London, London, UK

A new approach to the autonomous control of faders for multitrack audio mixing is presented. The algorithm is designed to generate an automatic sound mix from an arbitrary number of monaural or stereo audio tracks of any sample rate and to be suitable for both live and postproduction use. Mixing levels are determined by the use of the EBU R-128 loudness measure, with a cross-adaptive process to bring each track to a time-varying average. An hysteresis loudness gate and selective smoothing prevents the adjustment of intentional dynamics in the music. Real-time and off-line software implementations have been created. Subjective evaluation is provided in the form of listening tests, where the method is compared against the results of a human mix and a previous automatic fader implementation.

Convention Paper 8588

10:30

P2-3 A Voice Classification System for Younger Children with Applications to Content Navigation
—*Christopher Lewis, Christopher Pike, Yves Raimond, BBC R&D, London, UK*

A speech classification system is proposed that has applications for accessibility of content for younger children. To allow a young child to access online content (where typical interfaces such as search engines or hierarchical navigation would be inappropriate) we propose a voice classification system trained to recognize a range of sounds and vocabulary typical of younger children. As an example we designed a system for classifying animal noises. Acoustic features are extracted from a corpus of animal noises made by a class of young children. A Support Vector Machine is trained to classify the sounds into 1 of 12 corresponding animals. We investigated the precision and recall of the classifier for various classification parameters. We investigated an appropriate choice of features to extract from the audio and compared the performance when using mean Mel-frequency Cepstral Coefficients (MFCC), a single-Gaussian model fitted to the MFCCs, as well as a range of temporal features. To investigate the real-world applicability of the system we paid particular attention to the difference between training a generic classifier from a collected corpus of examples and one trained to a particular voice.

Convention Paper 8589

MUSIC AND MODELING

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

11:00

P3-1 Physical Model of the Slide Guitar: An Approach Based on Contact Forces—*Gianpaolo Evangelista*, Linköping University, Campus Norrköping, Sweden

In this paper we approach the synthesis of the slide guitar, which is a particular play mode of the guitar where continuous tuning of the tones is achieved by sliding a metal or glass piece, the bottleneck, along the strings on the guitar neck side. The bottleneck constitutes a unilateral constraint for the string vibration. Dynamics is subject to friction, scraping, textured displacement, and collisions. The presented model is physically inspired and is based on a dynamic model of friction, together with a geometrical model of the textured displacements and a model for collisions of the string with the bottleneck. These models are suitable for implementation in a digital waveguide computational scheme for the 3-D vibration of the string, where continuous pitch bending is achieved by all-pass filters to approximate fractional delays. Friction is captured by nonlinear state-space systems in the slide junction and textured displacements by signal injection at a variable point in the waveguide.

Convention Paper 8590

11:30

P3-2 Measuring Spectral Directivity of an Electric Guitar Amplifier—*Agnieszka Roginska*,¹ *Alex U. Case*,² *Andrew Madden*,¹ *Jim Anderson*¹

¹New York University, New York, NY, USA

²University of Massachusetts Lowell, Lowell, MA, USA

The recorded timbre of an electric guitar amplifier is highly dependent on the position of the microphone. Small changes in the location of the microphone can yield significant spectral differences, particularly at positions very close to the amp. This paper presents densely measured radiation pattern characteristics of an electric guitar amplifier on a 3-D grid in front, beside, behind, and above the amplifier in a hemi-anechoic space. We use this data to analyze the change in spectral differences between the numerous points on

the measurement grid. Differences between acoustically measured and estimated frequency responses (predicted, using interpolation) are used to study the change in the acoustic field in order to gain insight and an understanding of the spectral directivity sensitivity factor of the electric guitar amplifier.

Convention Paper 8592

12:00

P3-3 Magnitude-Priority Filter Design for Audio

Applications—*Balázs Bank*, Budapest University of Technology and Economics, Budapest, Hungary

In audio, often specialized filter design methods are used that take into account the logarithmic frequency resolution of hearing. A notable side-effect of these quasi-logarithmic frequency design methods is a high-frequency attenuation for non-minimum-phase targets due to the frequency-dependent windowing effect of the filter design. This paper presents two approaches for the correction of this high-frequency attenuation, based either on the iterative update of the magnitude or the iterative update of the phase of the target specification. As a result, the filter follows both magnitude and phase in those frequency regions where it can, while where this is not possible, it focuses on the magnitude. Thus, the new method combines the advantages of traditional complex and magnitude-only filter designs. The algorithms are demonstrated by parallel filter designs, but since the method does not make any assumption on the filter design algorithm used in the iteration, it is equally applicable to other techniques.

Convention Paper 8591

SOUND REINFORCEMENT AND STUDIO TECHNOLOGIES

Chair: **Diemer de Vries**, Amsterdam, The Netherlands

11:00

- P4-1 Full Room Equalization at Low Frequencies with Asymmetric Loudspeaker Arrangements—**
Balázs Bank, Budapest University of Technology and Economics, Budapest, Hungary

For rectangular rooms with symmetric loudspeaker arrangements, full room equalization can be achieved at low frequencies, as demonstrated by previous research. The method is based on generating a plane wave that propagates along the room. However, often the room is not rectangular, and/or a symmetric loudspeaker setup cannot be assured, leading to a deteriorated equalization performance. In addition, the performance of the method drops significantly above a cutoff frequency where a plane wave cannot be generated. These problems are addressed by the proposed method by prescribing only the magnitude in the control points, while the phase is determined by an iterative optimization process starting from the plane wave solution. A true “magnitude-only” variant of the method is also presented. Comparison is given to the plane-wave based methods by introducing asymmetries to the loudspeaker setup in a simulated environment, showing that the new methods result in smaller average magnitude deviations compared to the previous plane-wave based approach.

Convention Paper 8593

11:30

- P4-2 Linear Mixing Models for Active Listening of Music Productions in Realistic Studio Conditions—**
Nicolas Sturmel,¹ *Antoine Liutkus*,² *Jonathan Pinel*,³
Laurent Girin,³ *Sylvain Marchand*,⁴ *Gaël Richard*,²
Roland Badeau,² *Laurent Daudet*¹

¹Université Paris Diderot, Paris, France

²Telecom ParisTech, Paris, France

³GIPSA-Lab, Grenoble INP, Grenoble, France

⁴Université de Bretagne Occidentale, Brest, France

The mixing/demixing of audio signals as addressed in the signal processing literature (the “source separation” problem) and the music production in studio

remain quite separated worlds. Scientific audio scene analysis rather focuses on “natural” mixtures and most often uses linear (convolutive) models of point sources placed in the same acoustic space. In contrast, the sound engineer can mix musical signals of very different nature and belonging to different acoustic spaces, and exploits many audio effects including nonlinear processes. In the present paper we discuss these differences within the strongly emerging framework of active music listening, which is precisely at the crossroads of these two worlds: it consists in giving to the listener the ability to manipulate the different musical sources while listening to a musical piece. We propose a model that allows the description of a general studio mixing process as a linear stationary process of “generalized source image signals” considered as individual tracks. Such a model can be used to allow the recovery of the isolated tracks while preserving the professional sound quality of the mixture. A simple addition of these recovered tracks enables the end-user to recover the full-quality stereo mix, while these tracks can also be used for, e.g., basic remix / karaoke / soloing and re-orchestration applications.

Convention Paper 8594

12:00

P4-3 Capturing Height: The Addition of Z Microphones to Stereo and Surround Microphone Arrays—Paul Geluso, New York University, New York, NY, USA

As surround systems with height channels become more commonplace, new microphone techniques to capture sound in 3-D are needed. In order for the height channels to be effective, they must contain sonic information that is compatible with the 5.1 surround channels in order to improve the listener's sense of three-dimensional sound imaging, space, and immersion. Complex height information can be captured by pairing horizontally oriented microphones with vertically oriented bi-directional microphones. In this paper the author presents a rationale, methodology, and preliminary evaluation of a microphone technique based on this concept.

Convention Paper 8595

RECORDING AND PRODUCTION

Chair: TBD

14:30

P5-1 Automated Horizontal Orchestration Based on Multichannel Musical Recordings—Maximos Kaliakatsos-Papakostas,¹ Andreas Floros,² Michael N. Vrahatis¹

¹University of Patras, Patras, Greece

²Ionian University, Corfu, Greece

Orchestration of computer-aided music composition aims to approximate musical expression using vertical instrument sound combinations, i.e., through finding appropriate sets of instruments to replicate synthesized sound samples. In this paper we focus on horizontal orchestration replication, i.e., the potential of replicating the instantaneous intensity variation of a number of instruments that comprise an existing, target music recording. A method that efficiently performs horizontal orchestration replication is provided, based on the calculation of the instrumental Intensity Variation Curves. It is shown that this approach achieves perceptually accurate automated orchestration replication when combined with automated music generation algorithms.

Convention Paper 8596

15:00

P5-2 The Effect of Scattering on Sound Field Control with a Circular Double-Layer Array of Loudspeakers —Ji-ho Chang, Finn Jacobsen, Technical University of Denmark, Lyngby, Denmark

A recent study has shown that a circular double-layer array of loudspeakers makes it possible to achieve a sound field control that can generate a controlled field inside the array and reduce sound waves propagating outside the array. This is useful if it is desirable not to disturb people outside the array or to prevent the effect of reflections from the room. The study assumed free field condition, however in practice a listener will be located inside the array. The listener scatters sound waves, which propagate outward. Consequently, the scattering effect can be expected to degrade the per-

formance of the system. This paper computationally examines the scattering effect based on the simple assumption that the listener's head is a rigid sphere. In addition, methods to solve the problem are discussed.
Convention Paper 8597

15:30

P5-3 The Equidome, a Personal Spatial Reproduction Array—*James L. Barbour*, Swinburne University of Technology, Melbourne, Australia

Research into spatial sound requires the use of a loudspeaker array capable of reproducing sounds around and above a listening position. Different arrays have been developed that are often limited in their accuracy due to room size, the number of speakers employed, and their positions. They are also expensive to develop, which restricts access for many researchers. A unique array, The Equidome, has been developed with speakers around and above the listening position and it uses audio software capable of locating a sound anywhere over the array, and moving a sound throughout the array. The Equidome is a personal research facility capable of 3-D spatial composition with mono, stereo, multichannel, or ambisonic source files, and reverberation software is also utilized to create immersive sound fields throughout the array.
Convention Paper 8598

16:00

P5-4 A Clipping Detector for Layout-Independent Multichannel Audio Production—*Giulio Cengarle, Toni Mateos*, Fundació Barcelona Media, Barcelona, Spain

In layout-independent audio production, content is produced independently from the number of channels and their location, so that it can be played-back in different multichannel setups. In such contexts, sound is monitored through a playback system that might differ from the potentially many exhibition layouts. Signals combine to the outputs of each playback system in a different way and may produce clipping in some loudspeakers. A method is presented for detecting and quantitatively estimating clipping in the output stage of such systems, based on a suitable definition of a worst-case loudspeaker layout, and associated audio scene rotation and decoding. Practical examples are provided to validate the algorithm.
Convention Paper 8599

16:30

P5-5 Evaluation of Spatial Impression Comparing Surround with Height Channels for 3-D Imagery—

Toru Kamekawa,¹ Atsushi Marui,¹ Toshihiko Date,² Aiko Kawanaka,² Masaaki Enatsu³

¹Tokyo University of the Arts, Tokyo, Japan

²AVC Networks Company, Panasonic Corporation, Osaka, Japan

³marimoRECORDS Inc., Tokyo, Japan

Three-dimensional (3-D) imagery is now widely spreading as one of the next visual formats for Blu-ray or other future media. Since more audio channels are available with future media, the authors aim to find the suitable sound format for 3-D imagery. Semantic Differential method using 24 attributes such as “presence,” “naturalness,” and “preference” was carried out comparing combinations of 3-D and 2-D imagery with 2-channel stereo, 5-channel surround, and 7-channel surround sound (5-channel surround plus 2 height channels). Three factors (spatial factor, preference factor, and quality factor) were extracted from the results of Factor Analysis. Combination of the 3-D imagery with 7-channel surround gives higher scores at all those three factors.

Convention Paper 8600

17:00

P5-6 Microphone Array Design for Localization with Elevation Cues—*Michael Williams, Sounds of Scotland, Le Perreux sur Marne, France*

Analysis of the HRTF characteristics with respect to both azimuth and elevation localization cues would seem to suggest that, while inter-aural time difference and inter-aural level difference information give strong azimuth localization cues, we only have spectral variations in the vertical plane to generate localization cues with respect to elevation. Of course positioning of a second layer of loudspeakers above the horizontal reference plane will already introduce listener spectral differences relative to the horizontal plane reproduction, but the microphone array that feeds this second layer must not generate information that will be in conflict with the localization cues already generated in the horizontal plane of the first layer or main array. In the event of time difference and level difference information being generated between microphones in both layers, localization characteristics must be considered as projected onto the main horizontal plane of localization information.

Convention Paper 8601

17:30

- P5-7 General Integral Equation for the Width-Height Reproduction of a Focused Source Inside—**
Jung-Woo Choi, Yang-Hann Kim, Korea Advanced Institute of Science and Technology, Deajeon, Korea

A general integral formula to reproduce sound field from a virtual sound source located inside secondary sources is proposed. The proposed formula extends the theory on the focused sound source of Wave Field Synthesis for the reproduction of three-dimensional sound field. To resolve the non-existence problem involved with the reproduction of a source inside, an alternative sound field satisfying homogeneous wave equation is derived. The Kirchhoff-Helmholtz integral is reformulated in such a way that the alternative field is reproduced in terms of the secondary sources distributed on a surface. Then the general equation is reduced to simpler forms using various approximations, such as single layer formula reproducing the sound field only by monopole sound sources.

Convention Paper 8602

18:00

- P5-8 Evaluation of a New Active Acoustics System in Performances of Five String Quartets—**
Doyuen Ko, Wieslaw Woszczyk, Sonh Hui Chon, McGill University, Montreal, Quebec, Canada

An innovative electro-acoustic enhancement system, based on measured high-resolution impulse responses, was developed at the Virtual Acoustics Technology (VAT) lab of McGill University and was permanently installed in Multi-Media Room, a large rectangular scoring stage. Standard acoustic measures confirmed that the system was able to effectively improve room acoustic conditions in both spectral and spatial parameters. Subjective evaluation was conducted with twenty professional musicians from five string quartets on three different acoustic conditions. Spatial impression, stage support, and tonal quality were found to be three most dominant perceptual dimensions, while “naturalism of reverberation” was the most salient attribute affecting musicians’ preferences. Results showed a strong preference for enhanced acoustics over natural acoustics of the space.

Convention Paper 8603

MULTIMODAL APPS AND BROADCAST

Chair: **Bozena Kostek**, Gdansk University of Technology,
Gdansk, Poland

14:30

P6-1 Immersive Audiovisual Environment with 3-D Audio Playback—*Javier Gómez Bolaños, Ville Pulkki*, Aalto University, Espoo, Finland

The design of an immersive audiovisual environment for researching the aspects of the perception of spatial sound in the presence of a surrounding moving visual image is presented. The system consists in a visual screen with wide field-of-view based in acoustically transparent screens that span 226° in the horizontal plane and 57° in the vertical plane. In addition, a 3-D multichannel sound reproduction system with 29 active loudspeakers is installed. The total system is optimized for audio playback, and measured acoustical system responses are presented. The system is equipped with a tracking system based on infrared cameras, which enable head-tracking for head phone listening and also interaction based on gestures. This audiovisual system aims to be a tool for researching spatial audio, cross-modal interaction and psychoacoustics, auralization, and gaming.

Convention Paper 8604

15:00

P6-2 Sensitive Audio Data Encryption for Multimodal Surveillance Systems—*Janusz Cichowski, Andrzej Czyzewski*, Gdansk University of Technology, Gdansk, Poland

Novel algorithms for data processing in audiovisual surveillance systems were developed allowing for a better personal data protection. The solution merging the image and audio encryption for privacy-sensitive data protection employing audio stream is described. The main objectives of this research study including motivation and the state of the art are presented with a comprehensive explanation of audio stream relation to the surveillance. The invertible encryption methodology for privacy preserving using audio container is

applied. The experiments are described and obtained results are reported including prospects for future improvements.

Convention Paper 8605

15:30

P6-3 Loudness Normalization of Wide-Dynamic Range Broadcast Material—*Scott Norcross, Michel Lavoie*, Communications Research Centre, Ottawa, Ontario, Canada

Various techniques are being used by broadcasters to normalize the loudness levels of their programs. For long-form content, EBU R128 recommends that the full program be measured using the algorithm described in ITU-R BS.1770-2. ATSC A/85 recommends instead that only the “Anchor Element” of long-form content need be measured. For narrow dynamic range material, the differences between the two measures are not large, but there can be large differences between the two approaches when the material has a wide dynamic range. This paper compares these two measurement approaches and explores their subjective consequences.

Convention Paper 8606

16:00

P6-4 Creating Mood Dictionary Associated with Music—*Magdalena Plewa, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

The paper presents an attempt to create a dictionary of words related to mood associated with music. Two parts of a listening test were designed and carried out with a group of students, most of them users of social music online services. The audience task was to propose adjectives well-describing music tracks. These words were given in Polish and then compared to their English equivalents. The obtained results show that terms associated with music are language-specific and in addition there is a need to use multi-label mood description.

Convention Paper 8607

16:30

P6-5 Redundancy Optimization for Networked Audio Systems—*Damian Kowalski, Piotr Z. Kozłowski*,
Wrocław University of Technology, Wrocław, Lower
Silesia, Poland

Networked audio systems can be simply defined as a connection of IT and professional audio. Nowadays, we can use protocols developed by IT specialists to ensure system recovery without human intervention. There is a possibility to improve the recovery time of the system after failure by optimizing the protocols responsible for network redundancy. The paper is a summary of research completed at Wrocław University of Technology on May 2011. It contains guidelines on how to optimize the network redundancy in order to achieve the best results.

Convention Paper 8608

[Paper presented by Piotr Kozłowski]

PERCEPTION

Chair: **Bozena Kostek**, Gdansk University of Technology,
Gdansk, Poland

17:00

P7-1 Detection of Two Subwoofers: Effect of Broad-Band-Channel Level and Crossover Frequency—
Jussi Rämö, Sakari Bergen, Julian Parker, Veli-Matti Yli-Kätkä, Ville Pulkki, Aalto University, Espoo, Finland

The use of multiple subwoofers can be advantageous compared to a setup consisting of a single subwoofer due to the cancellation of room modes. We investigate the effect of subwoofer crossover frequency and program material on the perceived localization of bass frequencies using single or dual subwoofers, via a listening test. Test results show that dual subwoofer setups are harder to detect than single subwoofer setups and also exhibit the well-known relationship between crossover frequency and difficulty of localization.
Convention Paper 8609
[Paper presented by Ville Pulkki]

17:30

P7-2 Pitch, Timbre, Source Separation, and the Myths of Loudspeaker Imaging—*David Griesinger, David Griesinger Acoustics, Cambridge, MA, USA*

Standard models for both timbre detection and sound localization do not account for our acuity of localization in reverberant environments or when there are several simultaneous sound sources. They also do not account for our near instant ability to determine whether a sound is near or far. This paper presents data on how both semantic content and localization information is encoded in the harmonics of complex tones and the method by which the brain separates this data from multiple sources and from noise and reverberation. Much of the information in these harmonics is lost when a sound field is recorded and reproduced, leading to a sound image which may be plausible but is not remotely as clear as the original sound field.
Convention Paper 8610

LISTENING TESTS: PART 1

Chair: **Jan Plogsties**, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

09:00

P8-1 Comparison of Localization Performance of Blind and Sighted Subjects on a Virtual Audio Display and in Real-life Environments—*György Wersényi, József Répás*, Széchenyi István University, Győr, Hungary

Localization performance of blind subjects was measured in a virtual audio environment using non-individualized but customized HRTFs. Results were compared with former results of sighted users using the same measurement setup. Furthermore, orientation and navigation tasks in a real-life outdoor environment were performed in order to compare localization ability of sighted and visually impaired including “walking straight” tasks with and without acoustic feedback and test runs using the white cane as an acoustic tool during navigation.

Convention Paper 8611

09:30

P8-2 HELM: High Efficiency Loudness Model for Broadcast Content—*Alessandro Travaglini,¹ Andrea Alemanno,² Aurelio Uncini²*

¹FOX International Channels Italy, Rome, Italy

²University of Rome “La Sapienza,” Rome, Italy

In this paper we propose a new algorithm for measuring the loudness levels of broadcast content. It is called the High Efficiency Loudness Model (HELM) and it aims to provide robust measurement of programs of any genre, style, and format, including stereo and multichannel audio 5.1 surround sound. HELM was designed taking into account the typical conditions of the home listening environment, and it is therefore particularly good at meeting the needs of broadcast content users. While providing a very efficient assessment of typical generic programs, it also successfully approaches some issues that arise when assessing unusual content such as programs heavily based on bass frequencies, wide loudness range programs, and multichannel programs as opposed to stereo ones.

This paper details the structure of HELM, including its channel-specific frequency weighting and recursive gating implementation. Finally, we present the results of a mean opinion score (MOS) subjective test that demonstrates the effectiveness of the proposed method.

Convention Paper 8612

[Paper presented by Andrea Alemanno]

10:00

- P8-3 Defining the Listening Comfort Zone in Broadcasting through the Analysis of the Maximum Loudness Levels—Alessandro Travaglini,¹ Andrea Alemanno,² Fabrizio Lantini³**
¹FOX International Channels Italy, Rome, Italy
²University of Rome “La Sapienza,” Rome, Italy
³Electric Light Studio, Rome (RM), Italy

Over the last few years, the broadcasting industry has finally approached the loudness issue by standardizing its measurement and recommending target loudness levels with which all programs are required to comply. If the recommendations are applied and all programs are normalized at the target level, viewers ought to experience consistent perceived loudness levels throughout transmissions. However, due to the inner loudness modulation of the programs themselves, this is not always the case. In fact, even if the overall program loudness levels perfectly match the required target level, excessive loudness modulations can still generate annoyance to viewers if the foreground sound levels exceed the so-called “comfort zone.” The fact is that we still have no clear data on which metering can provide visual/numeric feedback on the perception of “hearing annoyance.” This paper investigates this issue and aims to provide objective evidence of which parameters would better represent this phenomenon. In particular, we describe an extensive subjective test performed for both the typical Stereo TV and the 5.1 home theater set reproductions and analyze its results in order to verify whether the Maximum Momentary Loudness Level, the Maximum Short Loudness Level, and Loudness Range (LRA) values described in EBU R128 can provide robust and reliable numeric references to generate a comfortable listening experience for viewers. Furthermore, we perform a similar analysis for the loudness descriptors of the algorithm HELM and finally indicate the values of those parameters that show the most consistent and reliable figures.

Convention Paper 8613

10:30

P8-4 The Relative Importance of Speech and Non-Speech Components for Preferred Listening Levels—*Ian Dash,¹ Miles Mossman,² Denis Cabrera²*

¹Consultant, Marrickville, NSW, Australia

²The University of Sydney, Sydney, NSW, Australia

In a prior paper the authors reported on a listening test that attempted to establish the relative importance of speech and non-speech components of a mixed soundtrack when matching loudness to reference audio items. That paper concluded that listeners match loudness by overall content rather than by the loudness of the speech or non-speech components. This paper reports on a follow-up listening test that attempts to establish the relative importance of speech and non-speech components in setting preferred listening level without any external reference. The results indicate that while speech levels are set more consistently than non-speech levels, listeners tend to set the overall levels more consistently than either of these components.
Convention Paper 8614

ANALYSIS AND SYNTHESIS OF SOUND: PART 1

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

09:00

- P9-1 Emergency Voice/Stress-Level Combined Recognition for Intelligent House Applications**
—*Konstantinos Drosos*,¹ *Andreas Floros*,¹
Kyriakos Agkavanakis,² *Nicolas-Alexander Tatlas*,³
*Nikolaos-Grigorios Kanellopoulos*¹
¹Ionian University, Corfu, Greece
²BlueDev Ltd., Patras, Greece
³Technological Educational Institute of Piraeus,
Piraeus, Greece

Legacy technologies for word recognition can benefit from emerging affective voice retrieval, potentially leading to intelligent applications for smart houses enhanced with new features. In this paper we introduce the implementation of a system, capable to react to common spoken words, taking into account the estimated vocal stress level, thus allowing the realization of a prioritized, affective aural interaction path. Upon the successful word recognition and the corresponding stress level estimation, the system triggers particular affective-prioritized actions, defined within the application scope of an intelligent home environment. Application results show that the established affective interaction path significantly improves the ambient intelligence provided by an affective vocal sensor that can be easily integrated with any sensor-based home monitoring system.

Convention Paper 8615

09:30

- P9-2 Loudness Range (LRA)—Design and Evaluation**
—*Esben Skovenborg*, TC Electronic A/S, Risskov,
Denmark

Loudness Range (LRA) is a measure of the variation of loudness on a macroscopic time-scale. Essentially, LRA is the difference in loudness level between the soft and loud parts of a program. In 2009 the algorithm for computing LRA was published by TC Electronic and was then included in the EBU R-128 recommendation for loudness normalization. This paper describes the

design choices underlying the LRA algorithm. For each of its parameters the interval of optimal values is presented, supported by analyses of audio examples. Consequences of setting parameter values outside these intervals are also described. Although the LRA measure has already proven its usefulness in practice, this paper provides background knowledge that could support further refinement and standardization of the LRA measure.

Convention Paper 8616

10:00

- P9-3 Statistical Properties of the Close-Microphone Responses**—*Elias K. Kokkinis, Eleftheria Georganti, John Mourjopoulos*, University of Patras, Patras, Greece

The close-microphone technique is widely used in modern sound engineering practice. It is mainly used to minimize the effect of leakage and room acoustics on the received signal. In this paper the properties of the close-microphone response are investigated from a signal processing point of view, through the respective frequency domain statistical moments. Room impulse response measurements were made in various rooms and source-microphone distances, and statistical moments were calculated over frequency and distance. It is shown that the statistical properties of the impulse responses remain relatively constant for short source-microphone distances and this in turn provides a consistent sound, which was validated through a subjective evaluation test.

Convention Paper 8617

10:30

- P9-4 Reproduction of Proximity Virtual Sources Using a Line Array of Loudspeakers**—*Jung-Min Lee, Jung-Woo Choi, Dong-Soo Kang, Yang-Hann Kim*, Korea Advanced Institute of Science and Technology (KAIST), Daejeon, Korea

For reproducing a desired sound field from virtual sources, Wave Field Synthesis (WFS) assumes that virtual sources are positioned at far-field from the loudspeaker array. This far-field assumption inevitably produces reproduction errors when the virtual source is adjacent to the array. In this paper we propose a method that can render the sound field from virtual sources positioned near the loudspeaker array. The driving functions of loudspeakers are derived for the planar array geometry, and then the surface integral is

reduced to a line integral by utilizing different approximations from WFS. In addition, a modified equation for the discrete loudspeaker distribution is presented. Numerical simulations show that the proposed method can reduce the reproduction error to a practically acceptable level.

Convention Paper 8618

11:00

P9-5 On the Statistics of Binaural Room Transfer Functions—*Eleftheria Georganti,¹ Tobias May,² Steven van de Par,² John Mourjopoulos¹*

¹University of Patras, Patras, Greece

²University of Oldenburg, Oldenburg, Germany

The well-known property of the spectral standard deviation of Room Transfer Functions (RTFs), that is, its convergence to 5.57 dB, is extended to reverberant Binaural Room Transfer Functions (BRTFs). The BRTFs are related to the anechoic Head Related Transfer Functions (HRTFs) and the corresponding RTFs. Consequently, the statistical properties of the RTFs and HRTFs can be systematically related to the statistical properties of the BRTFs. In this paper the standard deviation of BRTFs measured in different types of rooms, for various source/receiver distances and azimuth angles is computed. The derived values are compared to the ones obtained from the single channel RTFs measured at the same positions. Their relationship to the 5.57 dB value is discussed.

Convention Paper 8619

11:30

P9-6 On Acoustic Detection of Vocal Modes—*Eddy B. Brixen,¹ Cathrine Sadolin,² Henrik Kjelin²*

¹EBB-consult, Smorum, Denmark

²Complete Vocal Institute, Copenhagen, Denmark

According to the Complete Vocal Technique four vocal modes are defined: Neutral, Curbing, Overdrive, and Edge. These modes are valid for both the singing voice and the speaking voice. The modes are clearly identified both from listening and from visual laryngograph inspection of the vocal cords and the surrounding area of the vocal tract. For many reasons it would be preferred to apply a simple acoustic analysis to identify the modes. This paper looks at the characteristics of the voice modes from an acoustical perspective based on voice samples from four male and two female subjects. The paper describes frequency domain criteria for the discrimination of the various modes.

Convention Paper 8620

EDUCATION

Chair: **Jan Plogsties**, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

11:00

P10-1 Audio DSP from Scratch for Students of Computing—*Ewa Lukasik*, Poznan University of Technology, Poznan, Poland

The paper presents a set of dedicated programming applications for teaching digital audio basics to the novices with a background in computing. These applications have been used in the Institute of Computing Science Poznan University of Technology for over 15 years. Each application is devoted to an individual subject, is self explanatory, illustrates the given topic, and includes several simple problems to be solved by students. The topics include: the idea of signal energy and RMS, signals orthogonality, signals approximations, simple sinusoidal synthesis, spectrum, sampling theorem, upsampling and downsampling, convolution, idea of digital filters and interpretation of frequency characteristics, FIR and IIR filters—their zeroes and poles, windowing, DFT and its time and frequency resolution, cepstral analysis, and others. This set of programs proved useful as a starting point to more advanced audio projects in the domain of MIR, speech and speaker recognition, speech synthesis, audio coding, audio transmission, and others.

Convention Paper 8621

11:30

P10-2 A Comparison of Audio Frameworks for Teaching, Research, and Development—*Martin Robinson*,¹
*Jamie Bullock*²

¹University of the West of England, Bristol, UK

²Birmingham Conservatoire, Birmingham, UK

This paper compares a range of audio frameworks for the support of teaching, research, and the development of audio applications. The authors employ a range of metrics with which to compare the frameworks including: licensing terms, portability across different architectures, audio data-type support, efficiency of processing code, expressiveness, usability, and community activity. Conclusions are drawn that none of

these frameworks score highly in all of these domains. This suggests that while there are already a large number of such frameworks there remain areas to be addressed. The authors suggest that this might be through the development of existing systems or the development of new frameworks to meet these needs.
Convention Paper 8622

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

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

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

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

(a) The paper was accepted for presentation at the AES 132nd Convention.

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

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

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

* * * * *

**The Winner of the 132nd AES Convention
Student Paper Award is:**

**Robustness of a Mixed-Order Ambisonics
Microphone Array for Sound Field Reproduction**
—*Marton Marschall*,¹ *Sylvain Favrot*,¹ *Jörg Buchholz*²
¹Technical University of Denmark, Lyngby, Denmark
²National Acoustic Laboratories, Chatswood, Australia
Convention Paper 8645

*To be presented on Saturday, April 28 in Session P15
—Spatial Audio: Part 1*

POSTERS: QUALITY EVALUATION AND SPATIAL AUDIO

12:30

- P11-1 Influence of Resolution of Head Tracking in Synthesis of Binaural Audio**—*Mikko-Ville Laitinen, Tapani Pihlajamäki, Stefan Lösler, Ville Pulkki, Aalto University, Espoo, Finland*

The use of head tracking in binaural synthesis of spatial sound increases the quality of reproduction. The required quality of a head-tracking system for this purpose is the topic of interest in this paper. A listening test was performed to evaluate the effect of four common sources of error in head-tracking systems. The listeners rated the naturalness of binaural reproduction in different head-tracking conditions. According to the test, the main requirement for the head-tracking system is to obtain an accurate and unrestricted azimuth angle. Furthermore, lower update rate of the tracking system affects the quality. The results showed prominent dependence on program material, and individual differences were also notable.

Convention Paper 8623

12:30

- P11-2 Objective and Subjective Tests of Consumer-Class Audio Devices**—*Marek Pluta, Pawel Malecki, AGH University of Science and Technology, Krakow, Poland*

The paper presents a new stand for subjective listening tests in the Department of Mechanics and Vibroacoustics of Krakow University of Science and Technology, as well as a procedure and results of preliminary tests that utilize it. The most important difference, compared to the previously used stand, is exclusion of an analog mixing console, which reduces the length of electroacoustic channel, and substituting active studio monitors with 12 pairs of closed headphones in order to provide a larger group of listeners with the same listening conditions. The tests study popular consumer audio reproduction devices, therein standalone and motherboard integrated sound cards, as well as portable players. The research included measurements and comparison of objective parameters, as well as ABX method listening test, with a stu-

dio-class audio interface as a reference device. A group of listeners of various professional profiles, including students of the Academy of Music in Krakow and Acoustic Engineering of Krakow University of Science and Technology, took part in these listening tests. Results are compared to the data obtained using the previous test stand.

Convention Paper 8624

12:30

P11-3 Subjective Evaluations of Perspective Control Microphone Array (PCMA)—*Hyunkook Lee*, University of Huddersfield, Huddersfield, UK

Perspective Control Microphone Array (PCMA) is a technique that allows one to flexibly render spatial audio images depending on the desired virtual listening position in a reproduced sound field. Two subjective listening experiments have been conducted to evaluate the effectiveness of PCMA on the controls over the perceived source/ensemble distance and width attributes. The first experiment verified a hypothesis suggesting that perceived width would decrease as source-listener distance increased, using anechoic trumpet and conga sources convolved with binaural impulse responses of a concert hall. It was shown that the distance and width linearly changed at doubled distances and were negatively correlated. The second experiment tested three reference virtual array configurations of PCMA on the same attributes using the same sources. The results agreed with the perceptual patterns observed in the concert hall situation in that there was a linear decrease in perceived width at an increased perceived distance. The main effect of PCMA configuration was found to be statistically significant. These results seem to strongly validate the effectiveness of PCMA for postproduction and user-interactive applications.

Convention Paper 8625

12:30

P11-4 Objective Profiling of Perceived Punch and Clarity in Produced Music—*Steven Fenton, Jonathan Wakefield*, University Of Huddersfield, Huddersfield, UK

This paper describes and evaluates an objective measurement that profiles a complex musical signal over time in terms of identification of dynamic content and overall perceived quality. The authors have previously identified a potential correlation between inter-band dynamics and the subjective quality of produced music

excerpts. This paper describes the previously presented Inter-Band Relationship (IBR) descriptor and extends this work by conducting more thorough testing of its relationship to perceived punch and clarity over varying time resolutions. A degree of correlation is observed between subjective test scores and the objective IBR descriptor suggesting it could be used as an additional model output variable (MOV) to describe punch and clarity with a piece of music. Limitations have been identified in the measure however and further consideration is required with regard to the choice of threshold adopted based on the range of dynamics detected within the musical extract and the possible inclusion of a gate as utilized in some loudness algorithms.

Convention Paper 8626

12:30

P11-5 A Hybrid Method Combining Synthesis of a Sound Field and Control of Acoustic Contrast—Martin Bo Møller,¹ Martin Olsen,¹ Finn Jacobsen²

¹Bang & Olufsen a/s, Struer, Denmark

²Technical University of Denmark, Lyngby, Denmark

Spatially confined regions with different sound field characteristics, in the following referred to as sound zones, may be desired in some situations. Recently, various sound field control methods for generating separate sound zones have been proposed in the literature. The different algorithms introduce different levels of control over the physical characteristics of the resulting sound fields. This paper introduces a hybrid of two existing methods employed for obtaining sound zones: “Energy Difference Maximization” for control of the sound field energy distribution and “Pressure Matching,” which contributes with synthesis of a desired sound field. The hybrid method introduces a tradeoff between acoustic contrast between two sound zones and the degree to which the phase is controlled in the optimized sound fields.

Convention Paper 8627

12:30

P11-6 Validation of Room Plane Wave Decomposition as a Tool for Spatial Ecogram Analysis of Rooms—Ana Torres,¹ Jose J. Lopez,² Basilio Pueo³

¹Universidad de Castilla La Mancha, Ciudad Real, Castile-La Mancha, Spain

²Technical University of Valencia, Valencia, Spain

³Universidad de Alicante, Alicante, Spain

The classical point based analysis of sound attributes

in room acoustics are not usually enough for analyzing the acoustic complexity of a room. Some spatial attributes that provide more information are been employed. Circular arrays of microphones and the subsequently plane wave decomposition have been proposed in the literature and employed successfully by the authors and others. In this paper we validate our previous work comparing the resulting measured echograms with the simulation of the room acoustics using a basic room acoustics modeling software, ROOMSIM, freely available. The resulting echograms of both methods are compared identifying strengths and weaknesses of the measurement method based on circular arrays and proposing some ideas for a future more in deep analysis.

Convention Paper 8628

AMPS AND MEASUREMENT

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

14:00

P12-1 Current-Driven Switch-Mode Audio Power Amplifiers—*Arnold Knott, Niels Christian Buhl, Michael A. E. Andersen*, Technical University of Denmark, Lyngby, Denmark

The conversion of electrical energy into sound waves by electromechanical transducers is proportional to the current through the coil of the transducer. However virtually all audio power amplifiers provide a controlled voltage through the interface to the transducer. This paper presents a switch-mode audio power amplifier not only providing controlled current but also being supplied by current. This results in an output filter size reduction by a factor of 6. The implemented prototype shows decent audio performance with $THD + N$ below 0.1%.

Convention Paper 8629

14:30

P12-2 Debugging of Class-D Audio Power Amplifiers—*Lasse Crone, Jeppe Arnsdorf Pedersen, Jakob Døllner Mønster, Arnold Knott*, Technical University of Denmark, Lyngby, Denmark

Determining and optimizing the performance of a Class-D audio power amplifier can be very difficult without knowledge of the use of audio performance measuring equipment and of how the various noise and distortion sources influence the audio performance. This paper gives an introduction on how to measure the performance of the amplifier and how to find the noise and distortion sources and suggests ways to remove them. Throughout the paper measurements of a test amplifier are presented along with the relevant theory.

Convention Paper 8630

15:00

- P12-3 Investigation of Crosstalk in Self Oscillating Switch Mode Audio Power Amplifier**—*Thomas Haagen Birch, Rasmus Ploug, Niels Elkjær Iversen, Arnold Knott*, Technical University of Denmark, Lyngby, Denmark

Self oscillating switch mode power amplifiers are known to be susceptible to interchannel disturbances also known as crosstalk. This phenomenon has a significant impact on the performance of an amplifier of this type. The goal of this paper is to investigate the presence and origins of crosstalk in a two-channel self oscillating switch mode power amplifier (class D). A step-by-step reduction of elements in an amplifier built for this task is used for methodically determining the actual presence and origins of crosstalk. The investigation shows that the crosstalk is caused by couplings in the self oscillating pulse width modulation circuits, but also that the output filter has a major impact on the level of crosstalk.

Convention Paper 8631

15:30

- P12-4 Measuring Mixing Time in Non-Sabinian Rooms: How Scattering Influences Small Room Responses**—*Lorenzo Rizzi, Gabriele Ghelfi*, Suono e Vita – Acoustic Engineering, Lecco, Italy

The goal of this work is to optimize a DSP tool for extrapolating from room impulse response information regarding the way in which the transition between early reflections and late reverberation occurs. Two different methods for measuring this transition (usually referred as mixing time, t_{mix}) have been found in literature, both based on statistical properties of acoustic spaces. Appropriate changes have been implemented and the algorithms have been tested on I.R. measured in eight different environments. Particular attention is given to non-Sabinian environments such as small-rooms for music. It has been also measured a relationship between sound diffusion and t_{mix} , showing how the presence of scattering elements contributes to lower t_{mix} altering the statistical properties of I.R.

Convention Paper 8632

16:00

P12-5 Separation of High Order Impulse Responses in Methods Based on the Exponential Swept-Sine—

Stephan Tassart,¹ Aneline Grand^{1,2}

¹ST-Ericsson, Paris, France

²Now with Arkamys, Paris, France

Many real analog systems (e.g., electroacoustic loudspeaker, audio amplifiers, filters, etc.) exhibit weakly nonlinear features when driven by large amplitude signals. A large scale of such electromechanical devices are well modeled by the cascade of Hammerstein models. The exponential swept-sine is a natural excitation vector in order to identify the structural elements from those models. This paper extends the original swept-sine principle to the case of band-limited test vectors, suggests an intermodulation law for the generation of band-limited test vectors, and shows that a long-duration swept-sine can be replaced by a series of slightly phase-shifted short-duration swept-sines. High order impulse responses are separable even in case of temporal overlap with a linear combination of the measurements. The method is demonstrated on examples.

Convention Paper 8633

**ANALYSIS AND SYNTHESIS OF SOUND: PART 2;
CONTENT MANAGEMENT**

Chair: **Michael Kelly**, DTS Inc., London, UK

14:00

P13-1 Overview of Feature Selection for Automatic Speech Recognition—*Branislav Gerazov, Zoran Ivanovski*, Faculty of Electrical Engineering and Information Technologies, Skopje, Macedonia

The selection of features to be used for the task of Automatic Speech Recognition (ASR) is critical to the overall performance of the ASR system. Throughout the history of development of ASR systems, a variety of features have been proposed and used, with greater or lesser success. Still, the research for new features, as well as modifications to the traditional ones, continues. Newly proposed features as well as traditional feature optimization focus on adding robustness to ASR systems, which is of great importance for applications involving noisy environments. The paper seeks to give a general overview of the various features that have been used in ASR systems, giving details to an extent granted by the space available.

Convention Paper 8634

14:30

P13-2 Evaluating the Influence of Source Separation Methods in Robust Automatic Speech Recognition with a Specific Cocktail-Party Training—*Amparo Marti*,¹ *Maximo Cobos*,² *Jose J. Lopez*¹

¹Universitat Politècnica de València, Valencia, Spain

²Universitat de València, Burjassot, Valencia, Spain

Automatic Speech Recognition (ASR) allows a computer to identify the words that a person speaks into a microphone and convert it to written text. One of the most challenging situations for ASR is the cocktail party environment. Although source separation methods have already been investigated to deal with this problem, the separation process is not perfect and the resulting artifacts pose an additional problem to ASR performance in case of using separation methods

based on time-frequency masks. Recently, the authors proposed a specific training method to deal with simultaneous speech situations in practical ASR systems. In this paper we study how the speech recognition performance is affected by selecting different combinations of separation algorithms both at the training and test stages of the ASR system under different acoustic conditions. The results show that, while different separation methods produce different types of artifacts, the overall performance of the method is always increased when using any cocktail-party training.

Convention Paper 8635

15:00

P13-3 Automatic Regular Voice, Raised Voice, and Scream Recognition Employing Fuzzy Logic—

Kuba Lopatka, Andrzej Czyzewski, Gdansk University of Technology, Gdansk, Poland

A method of automatic recognition of regular voice, raised voice, and scream used in an audio surveillance system is presented. The algorithm for detection of voice activity in a noisy environment is discussed. Signal features used for sound classification, based on energy, spectral shape, and tonality are introduced. Sound feature vectors are processed by a fuzzy classifier. The method is employed in an audio surveillance system working in real-time both in an indoor and outdoor environment. Achieved results of classifying real signals are presented and discussed.

Convention Paper 8636

15:30

P13-4 Enhanced Chroma Feature Extraction

from HE-AAC Encoder—*Marco Fink,¹ Arijit Biswas,² Walter Kellermann¹*

¹University of Erlangen-Nuremberg, Erlangen, Germany

²Dolby Germany GmbH, Nuremberg, Germany

A perceptually enhanced chroma feature extraction during the HE-AAC audio encoding process is proposed. Extraction of chroma features from the MDCT-domain spectra of the encoder and its further enhancement utilizing the perceptual model of the encoder is investigated. The main advantage of such a scheme is a reduced computational complexity when both chroma feature extraction and encoding is desired. Specifically, the system is designed to produce reliable chroma features irrespective of the block switching decision of the encoder. Three methods are

discussed to circumvent the poor frequency resolution during short blocks. All proposed enhancements are evaluated systematically within a well-known state-of-the-art chord recognition framework.

Convention Paper 8637

16:00

P13-5 Hum Removal Filters: Overview and Analysis—

Matthias Brandt, Jörg Bitzer, Jade University of Applied Sciences, Oldenburg, Germany

In this contribution we analyze different methods for removing sinusoidal disturbances from audio recordings. In order to protect the desired signal, high frequency selectivity of the used filters is necessary. However, due to the time-bandwidth uncertainty principle, high frequency selectivity brings about long impulse responses. This can result in audibly resonating filters, causing artifacts in the output signal. Thus, the choice of the optimal algorithm is a compromise between frequency selectivity and acceptable time domain behavior. In this context, different filter structures and algorithms have different characteristics. To investigate their influence on the hum disturbance and the desired signal, we have evaluated three methods using objective error measures to illustrate advantages and drawbacks of the individual approaches.

Convention Paper 8638

LISTENING TESTS: PART 2

Chair: **Thomas Sporer**, Fraunhofer Institute
for Integrated Circuits IIS, Ilmenau, Germany

16:30

- P14-1 Determining the Threshold of Acceptability for an Interfering Audio Program**—*Jon Francombe*,¹
Russell Mason,¹ *Martin Dewhirst*,¹ *Søren Bech*²
¹University of Surrey, Guildford, Surrey, UK
²Bang & Olufsen, Struer, Denmark

An experiment was performed in order to establish the threshold of acceptability for an interfering audio program on a target audio program, varying the following physical parameters: target program, interferer program, interferer location, interferer spectrum, and road noise level. Factors were varied in three levels in a Box-Behnken fractional factorial design. The experiment was performed in three scenarios: information gathering, entertainment, and reading/working. Nine listeners performed a method of adjustment task to determine the threshold values. Produced thresholds were similar in the information and entertainment scenarios, however there were significant differences between subjects, and factor levels also had a significant effect: interferer program was the most important factor across the three scenarios, while interferer location was the least important.

Convention Paper 8639

17:00

- P14-2 Signal Processing Framework for Virtual Headphone Listening Tests in a Noisy Environment**—*Jussi Rämö*, *Vesa Välimäki*,
Aalto University, Espoo, Finland

A signal processing framework is introduced to enable parallel evaluation of headphones in a virtual listening test. It is otherwise impractical to conduct a blind comparison of several headphones. The ambient noise isolation capability of headphones has become an important design feature, since the mobile usage of earphones takes place in noisy listening environments. Therefore, the proposed signal processing framework allows a noise signal to be fed through a filter simulat-

ing the ambient sound isolation at the same time when music is played. This enables the simultaneous evaluation of the timbre and background noise characteristics, which together define the total headphone listening experience. Methods to design FIR filters for compensating the reference headphone response and for simulating the frequency response and isolation curve of the headphones to be tested are presented. Furthermore, a real-time test environment implemented using Matlab and Playrec is described.

Convention Paper 8640

17:30

P14-3 Perceptual Evaluation of Stochastic-Event-Based Percussive Timbres for Use in Statistical

Sonification—*William Martens, Mark*

McKinnon-Bassett, Densil A. Cabrera, University of Sydney, Sydney, NSW, Australia

The results of statistical data analysis have typically been presented using visual displays, but the sonification of data for auditory display, particularly using sound varying along realistic timbral dimensions, can offer an attractive alternative means for rendering such results. It was hypothesized that stochastic-event-based percussive timbres could be useful in communicating the details of statistical data, and so a preliminary study was designed to investigate the use of these timbres in data sonification. This study examined the ability of listeners to estimate the variation in a physical parameter for stimuli selected from a set of recorded percussive events. Specifically, two experiments were executed to determine whether listeners are generally able to estimate the number of small pellets that were present inside a container based upon the sounds that were made when the container was shaken a few times. The experimental results showed that a 6-dB trial-to-trial variation in reproduction level had no significant effect on obtained estimates, whereas variation in spectral energy distribution did significantly affect estimates of the number of small pellets in the shaken container. While the capacity to discriminate between sonification system outputs has been established, investigation of system effectiveness in applications remains to be done.

Convention Paper 8642

SPATIAL AUDIO: PART 1

Chair: **Christof Faller**, Illusonic LLC

09:00

P15-1 Analysis on Error Caused by Multi-Scattering of Multiple Sound Sources in HRTF Measurement—
Guangzheng Yu, Bosun Xie, Zewei Chen, Yu Liu,
South China University of Technology, Guangzhou,
China

A model consisting of two pulsating spherical sound sources and a rigid-spherical head is proposed to evaluate the error caused by multi-scattering of multiple sound sources in HRTF measurement. The results indicate that the ipsilateral error below 20 kHz caused by multi-scattering is within ± 1 dB when the radius of sound sources does not exceed 0.025 m, the source distance to head center is not less than 0.5 m, and the angular interval between the two adjacent sources is not less than 25 degrees. This accuracy basically satisfies the requirement of ipsilateral HRTF measurements. For improving the accuracy in contralateral HRTF measurement, some sound absorption treatments on the source surface are necessary.

Convention Paper 8643

09:30

P15-2 Personalization of Headphone Spatialization Based on the Relative Localization Error in an Auditory Gaming Interface—
Aki Härmä, Ralph van Dinther,
Thomas Svedström, Munhum Park, Jeroen Koppens,
Philips Research Europe, Eindhoven, The Netherlands

In binaural sound reproduction applications using head-related transfer functions (HRTFs) it is beneficial that the properties of the HRTFs correspond to the personal characteristics of the real HRTFs of the user. In this paper we propose a method to choose HRTFs using a relative localization test. This allows us to make the selection of the best HRTFs using a simple auditory interface. It is possible to design the HRTF personalization interface in a consumer device as an auditory game where the task of the user is to place sound objects in relation to each other. Two different interfaces are compared in a listening test. The results of

the tests reported in the current paper are mixed and do not give a conclusive picture on the performance of the proposed system, however, they do give interesting insights about the properties of binaural listening.

Convention Paper 8644

10:00

P15-3 Robustness of a Mixed-Order Ambisonics

Microphone Array for Sound Field Reproduction

—*Marton Marschall*,¹ *Sylvain Favrot*,¹ *Jörg Buchholz*²

¹Technical University of Denmark, Lyngby, Denmark

²National Acoustic Laboratories, Chatswood, Australia

Spherical microphone arrays can be used to capture and reproduce the spatial characteristics of acoustic scenes. A mixed-order Ambisonics (MOA) approach was recently proposed to improve the horizontal spatial resolution of microphone arrays with a given number of transducers. In this paper the performance and robustness of an MOA array to variations in microphone characteristics as well as self-noise was investigated. Two array processing strategies were evaluated. Results showed that the expected performance benefits of MOA are achieved at high frequencies, and that robustness to various errors was similar to that of HOA arrays with both strategies. The approach based on minimizing the error of the reproduced spherical harmonic functions showed better performance at high frequencies for the MOA layout.

Convention Paper 8645

10:30

P15-4 An Algorithm for Efficiently Synthesizing Multiple Near-Field Virtual Sources in Dynamic Virtual Auditory Display

—*Bosun Xie*, *Chengyun Zhang*,
South China University of Technology, Guangzhou,
China

An algorithm for efficiently synthesizing multiple near-field virtual sources in dynamic virtual auditory display (VAD) is proposed. Applying the method of principal component analysis, a set of measured near-field head-related impulse responses (HRIRs) for KEMAR manikin at various source directions and distances are decomposed into a weighted sum of 15 time-domain basic functions along with a mean time-domain function, in which the time-independent weights represent the location dependence of HRIRs. Accordingly, multiple virtual sources synthesis at various locations is implemented by a common bank of 16 filters representing the time-domain basis functions and the mean

time-domain function, in which the adjustable gains of the filters for each input stimulus (as well as an overall gain and delay for each input stimulus) control intended source locations. The computational cost of the proposed algorithm is reduced compared with that of conventional ones. Psychoacoustic experiments via a dynamic VAD with head-tracking validate the performance of the proposed algorithm.

Convention Paper 8646

11:00

P15-5 Scalable Coding of Three-Dimensional Multichannel Sound—Design of Conversion Matrix and Modeling of Unmasking Phenomenon—

Akio Ando, NHK Science and Technical Research Laboratories, Setagaya, Tokyo, Japan, and Tokyo Institute of Technology, Meguro, Tokyo, Japan

We propose two methods for coding and transmitting three-dimensional multichannel sound signals: scalable coding and transmission, and modeling of the quantization error. The first method converts N -channel sound signals into M -channel basic signals and $(N-M)$ -channel additional signals using a matrix operation. The matrix is trained by simulated annealing to minimize its condition number and the energy of additional signals. The unmasking artifact may occur when the N -channel signals are restored from the decoded signals using the inverse matrix. The second method estimates the quantization error signals by the polynomial approximation of the decoded signals. Experimental results showed that the combination of both methods could realize a 1.2 Mbps scalable transmission of 22-channel sounds without a notable sound degradation.

Convention Paper 8647

HIGH RESOLUTION AND LOW BIT RATE

Chair: **Vesa Välimäki**, Aalto University, Espoo, Finland

09:00

P16-1 Time Domain Performance of Decimation Filter Architectures for High Resolution Sigma-Delta Analog to Digital Conversion—Yonghao Wang,^{1,2} Joshua D. Reiss¹

¹Queen Mary University of London, London, UK

²Birmingham City University, Birmingham, UK

We present the results of a comparison of different decimation architectures for high resolution sigma delta analog to digital conversion in terms of pass-band, transition band performance, simulated signal-to-noise ratio, and computational cost. In particular, we focus on the comparison of time domain group delay response of different filter architectures including classic multistage FIR, cascaded integrator-comb (CIC) with FIR compensation filters, particularly multistage polyphase IIR filter, cascaded halfband minimum phase FIR filter, and multistage minimum phase FIR filter designs. The analysis shows that the multistage minimum phase FIR filter and multistage polyphase IIR filter are most promising for low group delay audio applications.

Convention Paper 8648

09:30

P16-2 A Delta-Sigma Modulator Using Dual NTF for 1-Bit Digital Switching Amplifier—Jungmin Choi, Jaeyong Cho, Haekwang Park, Samsung Electronics, Suwon, Korea

In this paper a fifth-order single-loop single-bit delta-sigma modulator (DSM), which is constructed by cascade-of-integrator, feed-back (CIFB) form for a 1-bit digital audio switching amplifier is proposed. High order DSM can achieve high signal-to-noise ratio (SNR), but it has probability that the oscillation occur. To achieve high SNR and improve the stability of the modulator for a large input range, we propose the DSM which is composed of dual noise transfer function (NTF). The

one is high SNR mode that maximizes SNR of DSM and the other is stable mode that enhances stability of DSM. The proposed architecture is simulated in the register transfer level (RTL) and implemented in the FPGA board.

Convention Paper 8649

10:00

P16-3 9 Years HE AAC—Technical Challenges Using an Open Standard in Real-World Applications—

Martin Wolters,¹ Gregory McGarry,² Andreas Schneider,¹ Robin Thesing¹

¹Dolby Germany GmbH, Nuremberg, Germany

²Dolby Australia Pty Ltd., Sydney, Australia

The technical work on creating the MPEG HE AAC standard was finished nine years ago. Since then the format has become very popular in specific markets and devices such as PCs and mobile phones mainly due to its high-compression efficiency. However, creating a reliable eco-system based on this open standard remains a technical challenge. In this paper results of several compatibility tests, which were conducted over the last two years on both mobile phones and broadcast receivers, are presented. The problems encountered and recommended solutions are described.

Convention Paper 8650

10:30

P16-4 Subjective Tests on Audio Mix Dedicated to MP3 Coding—*Szymon Piotrowski,¹ Magdalena Plewa²*

¹AGH University of Science and Technology, Krakow, Poland

²Gdansk University of Technology, Gdansk, Poland

Over the past years the Internet has become very popular as a means of distributing audio. MP3 coded audio is present in the Internet, in a bus, in broadcast. Sound engineers agree that there can often be a lack of control over the downstream processing that is applied to final material. The aim of the presented work is to compare audio mixes dedicated to CD format and converted to MP3 format with MP3-dedicated productions to and evaluate them.

Convention Paper 8651

11:00

P16-5 New Enhancements for Improved Image Quality and Channel Separation in the Immersive Sound Field Rendition (ISR) Parametric Multichannel Audio Coding System—Hari Om Aggrawal,¹ Deepen Sinha²

¹ATC Labs, Noida, India

²ATC Labs, NJ, USA

Consumer audio applications such as satellite broadcasts, HDTV, multichannel audio streaming, gaming, and playback systems are highlighting newer challenges in low-bit-rate parametric multichannel audio coding. This paper describes the continuation of our research related to the Immersive Sound field Rendition (ISR) parametric multichannel encoding system. We focus on the recent enhancements for the surround and center channel generation components of the ISR system. The emphasis being on improving the fidelity and quality of reconstructed 5/5.1-channel audio so that it achieves a level of transparency desirable for high end applications. Furthermore, it is being attempted to improve the robustness of the coding scheme to various difficult signals and listening environments by reducing inter-channel leakage to a minimum. We describe challenging case, various algorithmic improvements to the ISR algorithm to address these and also discuss the subjective impact of these algorithmic improvements.

Convention Paper 8652

11:30

P16-6 Novel Decimation-Whitening Filter in Spectral Band Replication—Han-Wen Hsu, Chi-Min Liu,
National Chiao Tung University, Hsinchu, Taiwan

MPEG-4 high-efficiency advanced audio coding (HE-AAC) has adopted spectral band replication (SBR) to efficiently compress high-frequency parts of the audio. In SBR, the linear prediction is applied to low-frequency subbands to clip the tonal components and smooth the associated spectrum for replicating to high-frequency bands. Such a process is referred to as the whitening filtering. In SBR, to avoid the alias artifact from spectral adjustment, a complex filterbank instead of real filterbank is adopted. For the QMF subbands, this paper analyzes that the linear prediction defined in SBR standard results in the predictive biases. A new whitening filter, called the decimation-whitening filter, is proposed to eliminate the predictive bias and provide advantages in terms of noise-to-signal ratio measure,

frequency resolution, energy leakage, and computational complexity for SBR.
Convention Paper 8653

12:00

**P16-7 MPEG Unified Speech and Audio Coding—
The ISO/MPEG Standard for High-Efficiency Audio
Coding of All Content Types—***Max Neuendorf,¹
Markus Multrus,¹ Nikolaus Rettelbach,¹ Guillaume
Fuchs,¹ Julien Robilliard,¹ Jérémie Lecomte,¹
Stephan Wilde,¹ Stefan Bayer,¹ Sascha Disch,¹
Christian Helmrach,¹ Roch Lefebvre,² Philippe Gour-
nay,² Bruno Bessette,² Jimmy Lapierre,² Kristofer Kjör-
ling,³ Heiko Purnhagen,³ Lars Villemoes,³ Werner
Oomen,⁴ Erik Schuijers,⁴ Kei Kikuri,⁵ Toru Chinen,⁶
Takeshi Norimatsu,⁷ Chong Kok Seng,⁷ Eumi Oh,⁸
Miyoung Kim,⁸ Schuyler Quackenbush,⁹ Bernhard
Grill¹*

¹Fraunhofer IIS, Erlangen, Germany

²Université de Sherbrooke, Sherbrooke, Quebec,
Canada

³Dolby Sweden AB, Stockholm, Sweden

⁴Philips Research Laboratories, Eindhoven,
The Netherlands

⁵NTT DOCOMO, INC., Yokosuka, Kanagawa, Japan

⁶Sony Corporation, Shinagawa, Tokyo, Japan

⁷Panasonic Corporation

⁸Samsung Electronics, Suwon, Korea

⁹Audio Research Labs, Scotch Plains, NJ, USA

In early 2012 the ISO/IEC JTC1/SC29/WG11 (MPEG) finalized the new MPEG-D Unified Speech and Audio Coding standard. The new codec brings together the previously separated worlds of general audio coding and speech coding. It does so by integrating elements from audio coding and speech coding into a unified system. The present publication outlines all aspects of this standardization effort, starting with the history and motivation of the MPEG work item, describing all technical features of the final system, and further discussing listening test results and performance numbers that show the advantages of the new system over current state-of-the-art codecs.

Convention Paper 8654

AUDIO EFFECTS

Chair: **Christof Faller**, Illusonic LLC

11:30

P17-1 Amplitude Manipulation for Perceived Movement in Depth—*Sonia Wilkie, Tony Stockman, Joshua D. Reiss*, Queen Mary University of London, London, UK

The presentation of objects moving in depth toward the viewer (looming) is a technique used in film (particularly those in 3-D) to assist in drawing the viewer into the created world. The sounds that accompany these looming objects can affect the extent to which a viewer can perceptually immerse within the multidimensional world and interact with moving objects. However the extent to which sound parameters should be manipulated remains unclear. For example, amplitude, spectral components, reverb and spatialization can all be altered, but the degree of their alteration and the resulting perception generated, need greater investigation. This paper presents the results from an investigation into one of the sound parameters used as an audio cue in looming scenes by the film industry, namely amplitude, reporting the degree and slope of their manipulation.

Convention Paper 8655

12:00

P17-2 Virtual 5.1 Channel Reproduction of Stereo Sound for Mobile Devices—*Kangeun Lee, Changyong Son, Dohyung Kim, Shihwa Lee*, Samsung Advanced Institute of Technology, Suwon, Korea

With rapid development in mobile devices, consumer demand for a premium sound experience is growing. In this paper a method for 5.1 channel upmixing is introduced. First, the primary and ambience components are separated and the primary signal is decomposed into the standard 5.1 channel direction. Since all separated sources should appear in the upmixed output, we designed the new masking scheme for panning coefficient. In order to reproduce the 5.1 channel sound field in the earphone or headset, the decomposed multi-channel signal is virtually rendered by means of HRTF. The proposed method was compared with conventional upmixing methods and demonstrated better spatiali-

ty image and panning effects with very low complexity requirements, which allows easy implementation on a wide variety of platforms.

Convention Paper 8656

12:30

P17-3 From Short- to Long-Range Signal Tunneling—

Alexander Carôt,¹ Horst Aichmann²

¹Hochschule Anhalt, Köthen/Anhalt, Germany

²Agilent Technologies, Kronberg, Germany

Research results in superluminal pulse transmission indicate propagations with velocities faster than the speed of light. This has so far been considered a short-range effect to be applied within distances of several centimeters. Based on these results and the corresponding theory of quantum tunneling the authors revisit such experiments with distances of several meters. They show that long-range superluminal signal transmission is possible and that it is only restricted by the actual signal bandwidth.

Convention Paper 8657

**POSTERS: EDUCATION AND HUMAN FACTORS;
APPLICATIONS**

13:00

**P18-1 Optimizing Teaching Room Acoustics:
A Comparison of Minor Structural Modifications
to Dereverberation Based on Smoothed Responses**

—*Panagiotis D. Hatziantoniou,¹ Nicolas-Alexander
Tatlas²*

¹University of Patras, Patras, Greece

²Institute of Piraeus, Aigaleo-Athens, Greece

In this work a comparison between traditional acoustical treatment such as building material substitution and digital room acoustics dereverberation is presented for teaching rooms. Measured responses for a number of listening positions for two rooms are shown as well as relevant parameters namely T30, EDT, C-50, and STI. Corresponding values are calculated by employing a simulation model in order to verify its accuracy; minor changes are then introduced to the model aiming to improve speech intelligibility and a new set of parameters is obtained. Finally, dereverberation achieved from inversion of appropriately modified measured room responses based on Complex Smoothing is employed and the acoustical parameters from the filtered impulse responses are derived.

Convention Paper 8658

13:00

**P18-2 Designing an Audio Engineer's User Interface
for Microphone Arrays**—*Stefan Weigand,^{1,2}*

Thomas Görne,² Johann-Markus Batke¹

¹Technicolor, Research and Innovation, Hannover,
Germany

²University of Applied Science (HAW), Hamburg,
Germany

Microphone arrays are rarely used in artistic recordings, despite the benefits they offer. We think this is due to a lack of user interfaces considering audio engineers' needs, enabling them to address the features in an easy and suited way. This paper contributes to solving this problem by outlining guidelines for such interfaces. A graphical user interface (GUI) for microphone arrays employing Higher Order Ambisonics (HOA), incorporating the audio engineers' aims and expecta-

tions, has been developed by analyzing their common activities. The presented solution offers three operation modes, covering the most frequent tasks in (professional) audio productions, thus making it more likely to engage audio engineers in using microphone arrays in the future.

Convention Paper 8659

13:00

- P18-3 User Interface Evaluation for Discrete Sound Placement in Film and TV Post-Production—**
Braham Hughes, Jonathan Wakefield, University of Huddersfield, Huddersfield, UK

This paper describes initial experiments to evaluate the effectiveness of different 3-D input interfaces combined with visual feedback for discrete sound placement in film and TV post-production. The experiments required the user to control the 3-D position of a sound object for a moving target object within a video clip on screen using a range of physical interfaces with and without visual feedback. Inclusion of visual feedback had a statistically significant impact on the accuracy of the tracking of the target object. The Wii remote controller appeared to perform the best in the tests and in the user preference ranking. The traditional desk-based input method performed worst in all tests and the user preference ranking.

Convention Paper 8660

13:00

- P18-4 KnuckleTap—Exploring the Possibilities of Audio Input in a Mobile Rhythmic Notepad Application—**
Julian Rubisch,¹ Michael Jaksche²

¹breakingwav.es, Vienna, Austria

²University of Applied Sciences, St. Poelten, Austria

Apart from some significant contributions in the scientific community and a few notable product innovations, audio input for musical interaction purposes on mobile devices has until now been widely neglected as a control parameter. This situation does not correspond to the fact that musical ideas are often vocalized and developed by musicians using their voice or other sounding objects, nor does it take advantage of contemporary mobile devices' most rigid and reliable sensor—the microphone. As an example case exploiting these possibilities, we conceived a notepad application to track rhythmic ideas by recording taps on a surface with a smartphone's built-in microphone, refined by subsequent detection of onsets, clustering, instantana-

neous rearranging of the detected events, and export capabilities.

Convention Paper 8661

13:00

- P18-5 An Optical System to Track Azimuth Head Rotations for Use in Binaural Listening Tests of Automotive Audio Systems**—*Anthony Price*, Bang and Olufsen a/s, Struer, Denmark, presently at University of Surrey, Guildford, Surrey, UK

Binaural technology is used to capture elements of an automotive audio system and reproduce them over headphones. This requires the tracking of azimuth head rotations of listening test participants in order to assist source localization. The parameters and faults of the currently implemented system are discussed and a new method of tracking azimuth head rotations is described. The system is tested, implemented, and found to have an error within 0.26 degrees. The potential for its further development, and the development of the field, is discussed.

Convention Paper 8662

13:00

- P18-6 Investigation of Salient Audio-Features for Pattern-Based Semantic Content Analysis of Radio Productions**—*Rigas Kotsakis*, *George Kalliris*, *Charalampos Dimoulas*, Aristotle University of Thessaloniki, Thessaloniki, Greece

The paper focuses on the investigation of salient audio features for pattern-based semantic analysis of radio programs. Most “news and music” radio programs have many structure similarities with respect to the appearance of different content types. Speech and music are continuously interchanged and overlapped, whereas specific speakers and voice patterns are more important to recognize. Recent research showed that various taxonomies and hierarchical classification schemes can be effectively deployed in combination with supervised and unsupervised training for semantic audio content analysis. Undoubtedly, audio feature extraction and selection is very important for the success of the finally trained expert system. The current paper employs feature ranking algorithms, investigating audio features saliency in various classification taxonomies of radio production content.

Convention Paper 8663

13:00

P18-7 Listeners Who Have Low Hearing Thresholds Do Not Perform Better in Difficult Listening Tasks—

Piotr Kleczkowski, Marek Pluta, Paulina Macura, Elzbieta Paczkowska, AGH University of Science and Technology, Krakow, Poland

The relationship between measures of hearing acuity and performance in listening tasks for normally hearing subjects has not found a solid evidence. In this work six one-parameter measures of hearing acuity, based on audiograms, were used to investigate whether a relationship between those measures and listeners' performance existed. The quantifiable results of several listening tests were analyzed, using speech and non-speech stimuli. The results showed no correlation between hearing acuity and performance thus demonstrating that hearing acuity should not be a critical factor in the choice of listeners.

Convention Paper 8641

[Paper presented by Paulina Macura]

SPATIAL AUDIO: PART 2

Chair: **Antti Kelloniemi**, Nokia Corporation, Espoo, Finland

14:30

- P19-1 Sound Field Reproduction Method in Spatio-Temporal Frequency Domain Considering Directivity of Loudspeakers**—*Shoichi Koyama, Ken'ichi Furuya, Yusuke Hiwasaki, Yoichi Haneda*, NTT Cyber Space Laboratories, NTT Corporation, Musashino-shi, Tokyo, Japan

A method for transforming received signals of a microphone array into driving signals of a loudspeaker array for sound field reproduction is needed to achieve real-time sound field transmission systems from the far-end to the near-end. We recently proposed a transform method using planar or linear microphone and loudspeaker arrays in the spatio-temporal frequency domain, which is more efficient than conventional methods based on a least squares algorithm. In this method, the directivity of loudspeakers in the array is assumed to be omnidirectional to derive the transform filter. However the directivity of common loudspeakers is not always omnidirectional, especially at high frequencies. We therefore propose a transform method that takes into consideration the directivity of loudspeakers in the array, which is derived using analytical and numerical approaches. Numerical simulation results indicated that the accurately reproduced region of the proposed method was larger than that of the method with an omnidirectional assumption.

Convention Paper 8664

15:00

- P19-2 Practical Applications of Chameleon Subwoofer Arrays**—*Adam J. Hill, Malcolm O. J. Hawksford*, University of Essex, Colchester, Essex, UK

Spatiotemporal variations of the low-frequency response in a closed-space are predominantly caused by room-modes. Chameleon subwoofer arrays (CSA) were developed to minimize this variance over a listening area using multiple independently-controllable source components and calibrated with one-time measurements. Although CSAs are ideally implemented using hybrid (multiple source component) subwoofers,

they can alternatively be realized using conventional subwoofers. This capability is exploited in this work where various CSA configurations are tested using commercially-available subwoofers in a small-sized listening room. Spectral and temporal evaluation is performed using tone-burst and maximum length sequence (MLS) measurements. The systems are implemented with practicality in mind, keeping the number of subwoofers and calibration measurements to a minimum while maintaining correction benefits.

Convention Paper 8665

15:30

P19-3 Localization in Binaural Reproduction with Insert Headphones—Marko Hiipakka, Marko Takanen, Symeon Delikaris-Manias, Archontis Politis, Ville Pulkki, Aalto University, Espoo, Finland

Circumaural headphones are commonly used in binaural reproduction and it is well known that individual equalization of the headphones improves the quality of the reproduction. The suitability of insert headphones to binaural reproduction has not been studied partly due the lack of a commonly accepted individual equalization method for insert headphones. Recently, a method to estimate the frequency response evoked by insert headphones has been presented. In this paper the localization accuracy of test subjects is evaluated in binaural listening with insert headphones and high-quality circumaural headphones. The results show that the accuracy with inserts is similar to that with circumaural headphones when the recently proposed method is applied for equalization, which motivates their use in binaural reproduction.

Convention Paper 8666

16:00

P19-4 A Comparative Evaluation between Numerical Techniques for Implementing the Acoustic Diffusion Equation Model—Juan M. Navarro,¹ Juan E. Noriega,¹ Jose Escolano,² Jose J. Lopez³

¹San Antonio's Catholic University of Murcia, Guadalupe, Spain

²University of Jaén, Linares, Spain

³Universidad Politécnica de Valencia, Valencia, Spain

The acoustic diffusion equation model is an energy-based model that is being successfully applied in room acoustics for predicting the late part of the decay, in the past few years. Early researches usually used a finite element method to solve the diffusion equation

model. Recently, an alternative implementation, using finite difference methods has been proposed. A comparison between both numerical techniques could be helpful to clarify the pros and cons of each method. In this paper this evaluation is made by several simulations in a cubic shaped room. Both prediction accuracy and computational performance are compared using different absorption distributions. It is suggested that the finite difference implementation is less computationally intensive than the finite element method. Moreover, the obtained values in the simulations are accurate, at least as well as other geometrical models.
Convention Paper 8667

16:30

P19-5 A Bayesian Framework for Sound Source

Localization—José Escolano,¹ Maximo Cobos,²
Jose M. Pérez-Lorenzo,¹ José J. López,³ Ning Xiang⁴
¹University of Jaén, Linares, Spain
²University of Valencia, Burjassot, Valencia, Spain
³Universidad Politécnica de Valencia, Valencia, Spain
⁴Rensselaer Polytechnic Institute, Troy, NY, USA

The localization of sound sources, and particularly speech, has a numerous number of applications to the industry. This has motivated a continuous effort in developing robust direction-of-arrival detection algorithms. Time difference of arrival-based methods, and particularly, generalized cross-correlation approaches have been widely investigated in acoustic signal processing. Once a probability function distribution is obtained, indicating those directions of arrival with highest probability, the vast majority of methods have to assume a certain number of sound sources in order to process the information conveniently. In this paper a model selection based on a Bayesian framework is proposed in order to determine, in an unsupervised way, how many sound sources are estimated. Real measurements using two microphones are used to corroborate the proposed model.

Convention Paper 8668

17:00

- P19-6 A Comparison of Modal versus Delay-and-Sum Beamforming in the Context of Data-Based Binaural Synthesis**—*Sascha Spors, Hagen Wierstorf, Matthias Geier*, Deutsche Telekom Laboratories, Technische Universität Berlin, Berlin, Germany

Several approaches to data-based binaural synthesis have been published that capture a sound field by means of a spherical microphone array. The captured sound field is decomposed into plane waves that are then auralized using head-related transfer functions (HRTFs). The decomposition into plane waves is often based upon modal beamforming techniques that represent the captured sound field with respect to surface spherical harmonics. An efficient and numerically stable approximation to modal beamforming is the delay-and-sum technique. This paper compares these two beamforming techniques in the context of data-based binaural synthesis. Their frequency- and time-domain properties are investigated, as well as the perceptual properties of the resulting binaural synthesis according to a binaural model.

Convention Paper 8669

TRANSDUCERS

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

09:00

P20-1 Loudspeaker for Low Frequency Signal Driven by Four Piezoelectric Ultrasonic Motors—Juro Ohga,¹ Ryousuke Suzuki,² Keita Ishikawa,² Hirokazu Negishi,³ Ikuo Oohira,⁴ Kazuaki Maeda,⁵ Hajime Kubota²

¹Shibaura Institute of Technology/MIX Corporation,
Kamakura, Japan

²Chiba Institute of Technology, Narashino, Japan

³MIX Corporation, Yokosuka, Japan

⁴I. Oohira and Associates, Yokohama, Japan

⁵TOA Corporation, Takarazuka, Japan

The authors have been developing a completely new direct-radiator loudspeaker construction that is driven by continuous revolution of piezoelectric ultrasonic motors. It converts continuous revolution of ultrasonic motors to reciprocal motion of a cone radiator. This loudspeaker shows almost flat phase frequency characteristics in low frequency region, because it includes no resonance in low frequency region. Therefore it is useful for radiation of the lowest frequency part of the audio signal. At this convention the authors are going to present a practical model of this loudspeaker driven by co-operation of four ultrasonic motors.

Convention Paper 8670

09:30

P20-2 Comprehensive Measurements of Head Influence on a Supercardioid Microphone—Hannes Pomberger,¹ Franz Zotter,¹ Dominik Biba²

¹University of Music and Performing Arts Graz, Graz,
Austria

²AKG Acoustics GmbH, Vienna, Austria

The directional pickup pattern of microphones is designed as to assist the audio engineer in avoiding acoustic feedbacks or interference from other sound sources. If the pattern deviates from the specified one, it is important for the audio engineer to know. This paper presents comprehensive measurements of a

supercardioid microphone directivity under the influence of a dummy head. This dummy models the diffraction of a human talker or singer in front of the microphone. The discussed measurements collect information on a 15x15 degree grid in azimuth and elevation for different distances between the microphone and dummy head. Based on this data, we are able to discuss the influence of a singer's body on the free field directivity of the supercardioid microphone, its directivity index, and its front-to-back random ratio in detail.

Convention Paper 8671

10:00

P20-3 Simulation of a 4-Inch Compression Driver Using a Fully Coupled Vibroacoustic Finite Element Analysis including Viscous and Thermal Losses

—René Christensen, Ulrik Skov, iCapture ApS, Gadstrup, Denmark

A 4-inch JBL compression driver is simulated using a finite element analysis, FEA. Compared to a conventional electrodynamic driver a compression driver has a phase plug with slits in front of the diaphragm. The slits are acoustically narrow and the diaphragm is separated from the phase plug only by a thin gap so an accurate model must include viscothermal effects to account for the losses associated with the narrow air gaps. Air domains, structural domains, and viscothermal domains are all fully coupled to ensure the proper continuity of their variables. Simulated results are compared to experimental measurements and it is demonstrated how the viscothermal effects dampen out acoustic and structural modes.

Convention Paper 8672

10:30

P20-4 Design of Vented Boxes Using Current Feedback Filters—Juha Backman,¹ Tim Mellow²

¹Nokia Corporation, Espoo, Finland

²MWT Acoustics, Farnham, UK

A current feedback arrangement for a loudspeaker system that can be tuned to provide a pre-determined frequency-response shape over a fairly wide and continuous range of box volumes is discussed. A conventional high-pass filter only allows the system to be tuned to give a particular frequency-response shape if the box volume is correct. The traditional arrangement is either a flat response amplifier, a passive filter between the amplifier and loudspeaker, or an active fil-

ter before the amplifier. This paper discusses an alternative arrangement where current feedback and filter provide the desired amplifier output impedance and voltage transfer function characteristics. These interact directly with the complex load-impedance of the loudspeaker. Practical realizations of the current feedback implementations are presented.

Convention Paper 8673

11:00

P20-5 Midrange Coloration Caused by Resonant Scattering in Loudspeakers—Juha Backman,
Nokia Corporation, Espoo, Finland

One of the significant sources of midrange coloration in loudspeakers is the resonant scattering of the exterior sound field from ports, recesses, or horns. This paper discusses a computationally efficient model for such scattering, based on waveguide models for the acoustical elements (ports, etc.), and mutual radiation impedance model for their coupling to the sound field generated by the drivers. This allows rapid evaluation of the effect of port placement, suitable for numerical optimization of loudspeaker enclosure layouts.

Convention Paper 8674

POSTERS: AUDIO EQUIPMENT AND INSTRUMENTATION

12:00

- P21-1 Evaluation of Vibrating Sound Transducers with Glass Membrane Based on Measurements and Numerical Simulations—György Wersényi, Széchenyi István University, Győr, Hungary**

In recent years manufacturers introduced so-called “invisible sound” solutions. In-wall, surface mount, or glass mount versions of different vibrating transducers are commercially available. The entire surface becomes a loudspeaker delivering sound, and the frequency response is said to be equivalent to conventional diaphragm speakers. Furthermore, the sound is omnidirectional at nearly all frequencies (60 Hz–15 kHz) while channel separation is maintained. This paper presents measurement results of the SolidDrive SD1g transducer mounted on different glass surfaces, including vibration measurements and acoustic parameters. Furthermore, based on a numerical FEM-model using COMSOL, comparison between measured and simulated results and estimation of transfer function and directional characteristics are presented.
Convention Paper 8675

12:00

- P21-2 Voice Coil Inter-Turn Faults Modeling and Simulation—German Ruiz, J. A. Ortega, J. Hernández, UPC-Universitat Politècnica de Catalunya, Terrassa, Spain**

The purpose of this paper is to present a new model to study inter-turn short circuit faults in a dynamic loudspeaker. The loudspeaker is modeled by using a classical voice coil parametric model attached to a mechanical piston, and the equations are modified to take into account the voice coil inter-turn faults. The loudspeaker model is global and can work in both normal and fault conditions due to a fictitious resistance in the winding circuit. Various simulation results have been presented indicating the fault instant in time and its corresponding effect in power spectral density. The model can serve as a step toward development of fault detection and diagnosis algorithm.
Convention Paper 8676

12:00

- P21-3 Headphone Selection for Binaural Synthesis with Blocked Auditory Canal Recording**—*Florian Völk*, AG Technische Akustik, MMK, Technische Universität München, Munich, Germany

Binaural synthesis aims at eliciting the reference scene hearing sensations by recreating the sound pressures at the eardrums, typically using headphones. If all transfer functions involved are approximated based on eardrum probe microphone or traditional artificial head measurements, the headphones have been shown not to influence the synthesis. It is also possible to achieve correct binaural synthesis with transfer functions measured at the entrances to the blocked auditory canals. Then, the headphones may influence the results. In this paper a blocked auditory canal headphone selection criterion (HPC) for binaural synthesis is proposed. Further, a procedure is derived, which allows to evaluate the (HPC) for specific circum-aural headphones based on four measurements using a specifically designed artificial head.

Convention Paper 8677

12:00

- P21-4 A Low Latency Multichannel Audio Processing Evaluation Platform**—*Yonghao Wang*,¹ *Xiangyu Zhu*,² *Qiang Fu*³

¹Queen Mary University of London, London, UK

²Hebei University of Science and Technology, Shijiazhuang, China

³Shijiazhuang Mechanical Engineering College, Shijiazhuang, China

For live digital audio system with high-resolution multichannel functionalities, it is desirable to have accurate latency control and estimation over all of the stages of the digital audio processing chain. The evaluation system we designed supports 12 channel- 24-bit sigma delta based ADC/DAC, incorporating both a programmable FPGA and digital signal processor. It can be used for testing and evaluation of different ADC/DAC digital filter architectures, audio sample buffer subsystem design, interrupt and scheduling, high level audio processing algorithms, and other system factors that might cause the latency effects. It also can estimate the synchronization and delay of multiple channels.

Convention Paper 8678

QUALITY EVALUATION

Chair: **György Wersényi**, Széchenyi István University, Győr, Hungary

14:00

P22-1 Evaluating Spatial Congruency of 3-D Audio and Video Objects—*Kristina Kunze, Judith Liebetrau, Thomas Korn*, Fraunhofer Institute for Digital Media Technology, IDMT, Ilmenau, Germany

In this paper we demonstrate the evaluation of spatial congruency of object based audio and 3-D video reproduction. With current developments in 3-D video representation we are able to introduce a depth dimension. Furthermore, audio reproduction systems like Wave Field Synthesis are able to reproduce the sound field of virtual sound sources at various positions in the room, also in front of or behind a video screen. When combining these technologies audio objects can be placed at the positions of 3-D video objects. Subjective evaluations are needed to investigate the quality of such combinations. In our experiment we displaced the audio and video objects with respect to certain angles and evaluated the noticeable displacement angle. Displacements of more than 5° are noticeable and become annoying above 10°.

Convention Paper 8679

14:30

P22-2 Some New Evidence that Teenagers and College Students May Prefer Accurate Sound Reproduction—*Sean Olive*, Harman International Industries Inc., Northridge, CA, USA

A group of 18 high school and 40 college students with different expertise in sound evaluation participated in two separate controlled listening tests that measured their preference choices between music reproduced in (1) MP3 (128 kbp/s) and lossless CD-quality file formats, and (2) music reproduced through four different consumer loudspeakers. As a group, the students preferred the CD-quality reproduction in 70% of the trials and preferred music reproduced through the most

accurate, neutral loudspeaker. Critical listening experience was a significant factor in the listeners' performance and preferences. Together, these tests provide some new evidence that both teenagers and college students can discern and appreciate a better quality of reproduced sound when given the opportunity to directly compare it against lower quality options.

Convention Paper 8683

15:00

- P22-3 Evaluation of Cultural Similarity in Playlist Generation**—*Mariusz Kleæ*, Polish-Japanese Institute of Information Technology, Warsaw, Poland

Choosing appropriate songs that satisfy one's needs is often frustrating, tiresome, and ineffective due to the increasing number of music collections and their sizes. Successive songs should fit the situation, our mood, or at least have common features. Consequently, there is a need to develop solutions that would enrich our experience of listening to music. In this paper musical similarity has been studied at the cultural level in playlist generation process. Also, an author's program designed for testing different playlists will be described. It is used to perform an experiment examining quality of playlists created using a cultural similarity model.

Convention Paper 8681

15:30

- P22-4 Toward an Unbiased Standard in Testing Laptop PC Audio Quality**—*Ravi Kondapalli, Ben-Zhen Sung*, CCCRMA, Stanford University, Stanford, CA, USA

For a rapidly growing population, laptop PCs and tablet devices have become a primary means of watching and listening to electronic media. Nonetheless, form factor restrictions and placement of device functionality over audio quality have left a gap in the overall quality of the laptop listening experience. Multiple audio Post-processing solutions exist from audio DSP developers that aim to improve this. However, the quality of audio produced by pairing these post-processing solutions with PCs from a variety of manufacturers varies greatly. To date, there have been no blind and systematic comparisons of audio quality resulting from unique PC/post-processing algorithm pairings. Here we present an unbiased methodological approach for evaluating such combinations, looking at audio quality for three different commercially available post-processing solutions as implemented on laptop PCs from two different manufacturers.

16:00

P22-5 User Evaluation on Loudness Harmonization on the Web—*Gerhard Spikofski, Peter Altendorf, Christian Hartmann*, Institut für Rundfunktechnik, Munich, Germany

The problem of annoying loudness jumps occurring between different TV and radio stations or programs became generally known over the last decades. Since the publication of the recommendations ITU-R BS.1770-1 (in 2006) and EBU-R 128 (in 2010), and the actual version of the ITU Recommendation BS.1770-2 (2011), more and more tools have become available that allow loudness harmonization in broadcasting. The European research project NoTube integrates loudness harmonization in its research concepts in order to investigate the applicability of these tools in web environments where the situation is even worse. The user evaluation of different loudness and loudness range adaptations presented in this paper was carried out online and thus considers real web conditions. In particular the interdependence between parameters of the listening environment and the loudness/loudness range adaptations are highlighted based on the results of nearly 100 participants.

Convention Paper 8680

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At recent AES conventions, authors have had the option of submitting complete 4- to 10-page manuscripts for peer-review by subject-matter experts. The following two papers have been recognized as co-winners of the AES 132nd Convention Peer-Reviewed Paper Award.

**Linear Mixing Models for Active Listening of Music
Productions in Realistic Studio Conditions—**

*Nicolas Sturmel,¹ Antoine Liutkus,² Jonathan Pinel,³
Laurent Girin,³ Sylvain Marchand,⁴ Gaël Richard,²
Roland Badeau,² Laurent Daudet¹*

¹Université Paris Diderot, Paris, France

²Telecom ParisTech, Paris, France

³GIPSA-Lab, Grenoble INP, Grenoble, France

⁴Université de Bretagne Occidentale, Brest, France

Convention Paper 8594

To be presented on Thursday, April 26 in Session P4

—Sound Reinforcement and Studio Technologies

and

**MPEG Unified Speech and Audio Coding—The ISO/MPEG
Standard for High-Efficiency Audio Coding of All
Content Types—**

*Max Neuendorf,¹ Markus Multrus,¹
Nikolaus Rettelbach,¹ Guillaume Fuchs,¹ Julien Robil-
liard,¹ Jérémie Lecomte,¹ Stephan Wilde,¹ Stefan
Bayer,¹ Sascha Disch,¹ Christian Helmrich,¹ Roch
Lefebvre,² Philippe Gournay,² Bruno Bessette,² Jim-
my Lapierre,² Kristofer Kjörting,³ Heiko Purnhagen,³
Lars Villemoes,³ Werner Oomen,⁴ Erik Schuijers,⁴ Kei
Kikuri,⁵ Toru Chinen,⁶ Takeshi Norimatsu,⁷ Chong
Kok Seng,⁷ Eumi Oh,⁸ Miyoung Kim,⁸ Schuyler
Quackenbush,⁹ Bernhard Grill¹*

¹Fraunhofer IIS, Erlangen, Germany

²Université de Sherbrooke, Sherbrooke, Quebec,
Canada

³Dolby Sweden AB, Stockholm, Sweden

⁴Philips Research Laboratories, Eindhoven,
The Netherlands

⁵NTT DOCOMO, INC., Yokosuka, Kanagawa, Japan

⁶Sony Corporation, Shinagawa, Tokyo, Japan

⁷Panasonic Corporation

⁸Samsung Electronics, Suwon, Korea

⁹Audio Research Labs, Scotch Plains, NJ, USA

Convention Paper 8654

To be presented on Saturday, April 28 in Session P16

—High Resolution and Low Bit Rate

Engineering Briefs

EB1	Papers	Friday, April 27 16:45–18:30 Room Liszt
EB2	Posters	Saturday, April 28 15:30–17:00 Foyer

ENGINEERING BRIEFS—PAPERS

Chair: **Josh Reiss**, Queen Mary University of London,
London, UK

17:00

EB1-1 What Is the Worth of Pre-DSP—Justin Paterson,
London College of Music, University of West London,
London, UK

Moorefield's "illusion of reality" was focused on the perception of authenticity in recordings, something that often continues to concern music producers. Editing techniques have increasingly been applied to move beyond this mimetic reproduction and "re-perform" musical elements, and furthermore, DSP commonly offers such extensive manipulation possibilities that all identifiable components of authenticity might be masked, even subverted, offering "virtual timbres" and revised sonic meaning. Here, editing and processing are considered along with their aesthetic and technical implications, placed in a historical perspective and augmented through the synthesis of contributions from a number of professional producers. Several perspectives are presented, and their tensions considered. The fluxive nature of authenticity will be further revealed in the trajectory toward the "reality of illusion."

Engineering Brief 41

17:15

EB1-2 Subjective Differences in Digital Audio Workstation Math—Brett Leonard, Padraig Buttner-Schnirer,
McGill University, Montreal, Quebec, Canada, Centre
for Interdisciplinary Research in Music, Media &
Technology, Montreal, Quebec, Canada

The subject of sound quality often arises when discussing the merits of various digital audio workstations (DAWs). While many engineers argue that one DAW "sounds better" than another, very little quantified data exists on the subject. In order to test these claims, multiple DAWs are fed the same multi-track digitized audio from a single converter. This audio is then processed by lowering all faders in each DAW by a fixed, arbitrary

amount, generating identical mixes, save the internal math performed through the gain change and the internal summing of each DAW. The resulting mixes are then tested for discriminability by trained listeners through the use of ABX testing. While there were differences between mixes, most listeners struggled to discriminate between DAWs.

Engineering Brief 42

17:30

EB1-3 Virtual Microphones: Optimum Real-Time Demodulation of Phase Modulated Ultrasound

—Tom Ritter,¹ Tobias Merkel,¹ Hans-G. Lühmann²

¹Beuth University of Applied Sciences Berlin, Berlin, Germany

²Lütronic Elektroakustik GmbH, Berlin, Germany

The project “Virtual Microphones” aims to replace conventional audio microphones by a system consisting of ultrasonic sensors. The fundamental physical principle utilized is that ultrasonic waves are phase-modulated when superimposed by audio waves. Due to the physical relationships the phase changes of the ultrasonic carrier are very small. The precise demodulation of the audio signal was in the focus of the investigation among other measures, like providing a very stable carrier frequency or the focusing of the ultrasound beam on the receivers. Several methods of demodulation for phase-modulated signals are compared. The comparatively low resulting signal-to-noise ratio after the demodulation required further signal processing in form of noise reduction to be investigated and its performance accessed.

Engineering Brief 43

17:45

EB1-4 The Audio Plugin Generator: Rapid Prototyping of Audio DSP Algorithms—Robert Cerny, Fritz

Menzer, dlab GmbH, Winterthur, Switzerland

This paper introduces the Audio Plugin Generator (APG), a rapid prototyping tool that allows transforming Simulink models into VST plugins with little effort. VST is a well-known audio plugin interface, supported by a variety of host programs, to process audio signals with low latency. On the other hand, Simulink enables DSP engineers to use clean, human-readable descriptions of their algorithms. The generated plugin supports parameter tuning as well as data logging in real-time.

Furthermore, signals can be analyzed inside the plugin with signal scopes. The paper explains the APG workflow and how it bridges the gap between the Simulink Coder and the VST standard. A light version of APG can be obtained free of charge from the APG website: <http://audioplugingenerator.com/>
Engineering Brief 44

18:00

EB1-5 MS Mastering of Stereo Microphone Signals—
Martin Schneider, Georg Neumann GmbH, Berlin, Germany

MS (mid-side) technique is a classic tool for recording and mastering based on coincident microphone setups. For these, localization theory tells us that on reproduction via loudspeakers, the interaural level differences (ILDs) will yield stable and fully mono-compatible results. Yet, when a mastering engineer obtains a mix, he will rarely be told how the specific sources were recorded. Was it as mono signals, level panned to a certain reproduction angle, or with coincident, spaced, or mixed stereo microphone setups. While the first two techniques only involve ILDs, spaced microphone techniques also imply interaural time differences (ITDs). MS matrixing techniques can be applied to change the apparent width of a full stereo mix in post-production. The engineer should be aware that he then cross-feeds signals between the channels. The signal paths from a signal sound source, via the microphones, the matrixing, and the loudspeakers, to finally reach the ears, can now be seen as eight, instead of four, for coincident techniques. An increased sensation of spaciousness or envelopment may arise, but one should be aware of the physics behind the process. Some of the arising comb filters are shown here, for spaced and ORTF setups.
Engineering Brief 45

18:15

EB1-6 A New Visual Paradigm to Surround Sound Mixing
—Thierry Dilger, www.sonabilis.com, Paris, France

Today's tools for surround mixing are like add ons to old style linear software. Their paradigm for surround mixing are almost the same: a sound file (or a track) is routed to a surround panner, with the help of the timeline and automation, sounds are moving in space. But

you cannot prepare several sound movements at once, you cannot have a whole timeline overview of the spatialization process, you cannot compare easily two surround mixes, you can nearly not work on non-traditional set-ups, you cannot create generative surround sounds. The model presented during this presentation is based on a color coding system (RGB) working on three axes which are: time sounds are playing (x), the speakers number and position (y), and the intensity of each sound (z). I discuss its pros (it gives answers to all the flaws previously announced) and cons compared to traditional tools and about its field of application.

Engineering Brief 46

EB1-7 Sounds Not Signals: A Perceptual Audio Format

—*Michael J. Terrell, Andrew J. R. Simpson, Mark Sandler*, Queen Mary University of London, London, UK

This Engineering Brief presents a brief overview of a novel audio transmission and storage format. The main feature of the format is the use of absolute sound pressure within the format. A secondary feature is the use of perceptual features to describe the associated sound content. Error reporting and correction systems are briefly described and a case study used to demonstrate the principles involved.

Engineering Brief 52

ENGINEERING BRIEFS—POSTERS

15:30

EB2-1 An Exact Harmonic Computing Technique for Polynomial Nonlinearities—Nay Oo,¹ Woon-Seng Gan²

¹Fraunhofer Institute for Digital Media Technology, IDMT, Oldenburg, Germany

²Nanyang Technology University, Singapore, Singapore

An exact harmonic computing technique for polynomial nonlinearities is developed from first principles. By applying this technique, when an input sinusoid with arbitrary amplitude, frequency and phase information and a polynomial nonlinearity are given, the exact computation of DC component and output harmonics' amplitudes, frequencies, and phases is computationally possible without discrete Fourier transform (DFT). Two basic mathematical results such as power of cosine and harmonic addition theorem are utilized to develop this technique.

Engineering Brief 47

15:30

EB2-2 Suspension Creep Models for Miniature Loudspeakers—Holger Hiebel, Knowles Electronics Austria GmbH, Vienna, Austria, Graz University of Technology, Graz, Austria

Different models for describing the suspension creep behavior of loudspeakers are available at the moment. They include the "LOG"-model from Knudsen, the model used in the Klippel Analyzer system (same as Knudsen, but neglecting the frequency dependent losses) and the 3-parameter model of Ritter and Agerkvist. In contrast to HiFi speakers the differences between the creep models can be seen more clearly in miniature loudspeakers using compound (multilayer) membranes. Benefits from choosing the right creep model are tabulated in terms of RMS fit errors and evaluation results for miniature loudspeaker samples are shown.

Engineering Brief 48

15:30

EB2-3 Smart Microphone Sensor System Platform—*Elias Kokkinis, Konstantinos Drossos,² Nicolas-Alexander Tatlas,³ Andreas Floros,² Alexandros Tsilfidis,¹ Kyriakos Agavanakis³*

¹University of Patras, Patras, Greece

²Ionian University, Corfu, Greece

³BLUEdev Ltd., Patras, Greece

A platform for a flexible, smart microphone system using available hardware components is presented. Three subsystems are employed, specifically: (a) a set of digital MEMs microphones, with a one-bit serial output; (b) a preprocessing/digital-to-digital converter; and (c) a CPU/DSP-based embedded system with I2S connectivity. Basic preprocessing functions, such as noise gating and filtering can be performed in the preprocessing stage, while application-specific algorithms such as word spotting, beam-forming, and reverberation suppression can be handled by the embedded system. Widely used high-level operating systems are supported including drivers for a number of peripheral devices. Finally, an employment scenario for a wireless home automation speech activated front-end sensor system using the platform is analyzed.

Engineering Brief 49

15:30

EB2-4 Sound Field Synthesis Toolbox—*Hagen Wierstorf, Sascha Spors, TU Berlin, Berlin, Germany*

An open source toolbox for Sound Field Synthesis (SFS) is introduced. The toolbox is able to numerically simulate sound fields synthesized by SFS methods like Wave Field Synthesis or higher order Ambisonics. Various loudspeaker driving signals for the mentioned methods are provided for 2-, 2.5-, and 3-dimensional synthesis. The toolbox allows mono-frequent as well as broadband excitation signals. The latter allows generation of snapshots of the spatio-temporal impulse response of a chosen reproduction technique. The toolbox furthermore includes the computation of binaural room impulse responses (BRIR) for a given SFS setup. These can be used to simulate different sound field synthesis methods via binaural resynthesis. The toolbox is provided for Matlab/Octave and comes with an online documentation.

Engineering Brief 50

15:30

EB2-5 Listening to the Large Hadron Collider—

*Daniel Deboy,^{1,2} Ralph W. Abmann,¹ Roderik Bruce,¹
Florian Burkart,¹ Marija Cauchi,¹ Clement Derrez,¹
Alessandro Masi,¹ Stefano Redaelli,¹ Belen
Salvachua,¹ Gianluca Valentino,¹ Daniel Wollmann¹*

¹CERN, Geneva, Switzerland

²University of Music and Performing Arts,
Graz, Austria

The Large Hadron Collider (LHC) at CERN is a high-energy particle accelerator in a 27 km long tunnel located in the underground of the Geneva area, Switzerland. Protons are accelerated to 99.9999991 percent of the speed of light before they collide with a total momentum of up to 14 TeV. It is the largest machine that human mankind has ever built. Over 10.000 sensors monitor the state of the LHC during operation. Recently, microphones have been added as an experimental setup. An acoustic monitoring system to detect and localize beam accidents is under current investigation. Such a system may reduce expensive downtime dramatically in an accident scenario. The acquired signals can also be used for other applications, e.g., sonification, media art installations, etc.

Engineering Brief 51

WORKSHOPS

W-1	What Every Sound Engineer Should Know about the Voice	Thursday, April 26 10:30–12:30 Room Bartók
W-2	Game Audio	Thursday, April 26 11:00–12:30 Room Brahms
W-3	Audio Hardware in Smartphones	Thursday, April 26 14:30–16:30 Room Brahms
W-4	The Replay of Historical Magnetic Tape	Thursday, April 26 17:00–18:00 Room Brahms
W-5	Use Cases for High Performance Audio-over-IP in Broadcast Facilities	Friday, April 27 09:00–10:30 Room Bartók
W-6	Open Questions in Spatial Audio	Friday, April 27 11:00–13:00 Room Brahms
W-7	Distributed Music Panel	Friday, April 27 12:15–13:45 Room Liszt
W-8	Highly Directional Microphones for Sound Recording	Friday, April 27 14:00–15:30 Room Brahms
W-9	Active Acoustics Today: Challenges, Solutions, Applications	Friday, April 27 14:00–16:00 Room Bartók
W-10	Screen-less Navigation for High-Resolution Audio on Blu-ray Disc	Saturday, April 28 09:00–10:00 Room Bartók
W-11	MUSHRA Reloaded	Saturday, April 28 10:30–12:00 Room Brahms
W-12	Spatial Sound Reproduction with Height	Saturday, April 28 12:30–14:30 Room Brahms
W-13	Deploying the Loudness Concept in Europe	Saturday, April 28 14:30–16:30 Room Brahms
W-14	Semantic Audio Analysis in the Real World	Sunday, April 29 09:00–10:00 Room Brahms

**W-15 Synchronization in a
Multichannel Digital Age**

Sunday, April 29
10:30–12:30
Room Brahms

**W-16 Listen Professionally
or Train Your Ear!**

Sunday, April 29
11:00–13:00
Room Bartók

Workshop 1
Thursday, April 26
Room Bartók

10:30 – 12:30

**WHAT EVERY SOUND ENGINEER SHOULD KNOW
ABOUT THE VOICE**

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

Panelists: *Henrik Kjelin*, Complete Vocal Institute, Denmark
Mikkel Nymand, DPA Microphones, Denmark
Cathrine Sadolin, Complete Vocal Institute,
Denmark

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

Workshops

Workshop 2
Thursday, April 26
Room Brahms

11:00 – 12:30

GAME AUDIO

Chair: **Michael Kelly**, DTS, London, UK

Panelists: *Zoltan Nyakacska*, Nemesys Games
Zach Zebrowski, Nemesys Games

This workshop starts with a general introduction to the games industry and goes on to discuss the specific implementation of audio in games.

The workshop, presented by Nemesys Games, looks at the pros and cons of selecting different audio engines for game development in the context of their current title, *Ignite*. The talk shows how the whole audio system for the game is built and discusses the challenging task of implementing car engine sounds; outlining the technical and creative problems and how they are solved.

The workshop concludes by comparing the nonlinear nature of game-sound development to the production of a more linear piece such as a short movie or trailer.

This event is aimed at a general audience who may not be familiar with game development but presents an appropriate level of detail for those who want to dig deeper.

Workshop 3
Thursday, April 26
Room Brahms

14:30 – 16:30

AUDIO HARDWARE IN SMARTPHONES

Chair: **Antti Kelloniemi**, Nokia Corporation,
Espoo, Finland

Panelists: *Juha Backman*, Nokia Corporation, Espoo, Finland
Jörg Rehder, Knowles Electronics, Denmark
Friedrich Reining, Knowles Electronics GmbH,
Vienna, Austria
Chao Wang, AAC Technologies, China
Anders Weckström, GoerTek, Finland
Erick Wiederholtz, Knowles Electronics GmbH,
Vienna, Austria

Comparison between novel smartphones and any other professional or consumer audio equipment reveals that phones actually provide highly sophisticated audio functions for their size and price. High quality noise reduction and echo cancellation for telephony and multichannel audio capture with wide frequency response and high dynamic range are expected, as well as loud and clear sound output for alarm tones and multimedia playback. The quality expectations keep rising, while devices should be kept small and affordable.

To start this workshop, the audio solutions in a smartphone are presented on an introductory level. After that, the panel, who are representatives of major audio component manufacturers, discuss the current audio performance requirements and present examples of state-of-the-art in audio component technology.

Workshops

Workshop 4

Thursday, April 26

17:00 – 18:00

Room Brahms

THE REPLAY OF HISTORICAL MAGNETIC TAPE— MORE THAN PRESSING THE PLAY BUTTON

Chair: **Nadja Wallaszkovits**, Phonogrammarchiv,
Austrian Academy of Sciences, Vienna, Austria

Panelists: *Dietrich Schüller*, Phonogrammarchiv,
Austrian Academy of Sciences, Vienna, Austria
TBA

Since analog magnetic tape technology is no longer a part of the audio production process, specific knowledge is endangered to decline. This workshop discusses the various problems occurring in the transfer process of historical magnetic audio tapes. Starting with a definition of historical tape brands and an overview of early magnetic tape developments, the practical handling of critical tapes is outlined: based on an analysis of the physical and chemical preservation status of the individual tape, the choice and adjustment of the replay equipment and parameters are discussed. Problems of carrier handling and physical as well as chemical restoration are outlined, as well as possible signal enhancement on the playback process only. The workshop focuses on the handling and reproduction of original tapes from the early ages of magnetic recording, stored under irregular conditions.

Workshop 5
Friday, April 27
Room Bartók

09:00 – 10:30

**USE CASES FOR HIGH PERFORMANCE
AUDIO-OVER-IP IN BROADCAST FACILITIES**

Chair: **Stefan Ledergerber**, Lawo Group,
Zurich, Switzerland

Panelists: *Axel Holzinger*, ALC Networkx, Munich, Germany
Lars Jonsson, Swedish Radio, Stockholm, Sweden
Sonja Langhans, IRT, Munich, Germany
Greg Shay, Telos Systems Inc., Cleveland,
OH, USA

In today's broadcasting houses the topic of audio networking is heavily discussed. In the overall discussion the audio network is very often seen as a solution for everything. But what are the real use cases today for an audio-over-IP solution within facilities? The panelists will each make a 15 minute presentation about their views, followed by an open discussion with the audience about the hot topics of the migration from established wires into network land.

Workshops

Workshop 6
Friday, April 27
Room Brahms

11:00 – 13:00

OPEN QUESTIONS IN SPATIAL AUDIO

Chair: **Karlheinz Brandenburg**

Panelists: *Jens Blauert*
David Griesinger
Frank Melchior
Stefan Werner

For more than 60 years spatial audio systems are in use. Just in the last 20 years the number of loudspeakers and transmission channels used have been increased, and with Walkman and MP3, player listening via headphone has become popular, too. Both scenarios provide much better audio quality than the old stereo systems, but very often strange, unexpected effects are limiting the perceived quality. This workshop will list aspects where the knowledge in spatial audio quality is not perfect. Examples are influence of room reflections, mixing of the room acoustical properties of recording and reproduction rooms, influence of room on binaural headphone reproduction (sic!), audio-visual coherence in large rooms, and issues about perception of elevated sound sources.

Workshop 7
Friday, April 27
Room Liszt

12:15 – 13:45

DISTRIBUTED MUSIC PANEL

Chair: **Alex Carôt**, Hochschule Anhalt,
Köthen/Anhalt, Germany

Panelists: *Alvaro Barbosa*, UCP Porto, Porto, Portugal
Nathan Brock, University of California at San
Diego, San Diego, CA, USA
Karl Steinberg, Digitalmusician, Hamburg
Andrea Szigetvári, Ferenc Liszt Academy
of Music, Budapest, Hungary
David Willyard, Musicianlink, San Jose, CA, USA

This workshop covers the emerging field of music intended for performance over networks, including both advanced research networks and the public Internet. The discussion will cover experimentally determined limits for delay between locations, and methods for performance both above and below these thresholds. Representatives of several different dedicated systems and projects will be present, each with many years' experience in the networked music scene. Soundjack and the Jamlink hardware interface will be discussed as examples of realistic interaction over networks, while JackTrip and the Digital Musician Link will show methods for overcoming long latencies on long-distance network links. There will also be a discussion of artistic strategies for overcoming excessive delay in performance.

Workshops

Workshop 8
Friday, April 27
Room Brahms

14:00 – 15:30

HIGHLY DIRECTIONAL MICROPHONES FOR SOUND RECORDING

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

Panelists: *Christof Faller*, Illusonic LLC
Michael Millitzer, Microtech Gefell

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

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

Workshop 9
Friday, April 27
Room Bartók

14:00 – 16:00

**ACTIVE ACOUSTICS TODAY:
CHALLENGES, SOLUTIONS, APPLICATIONS**

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

Panelists *Leo de Klerk*, Bloomline Acoustics b.v.,
's-Gravendeel, The Netherlands
Doyuen Ko, McGill University,
Montreal, Quebec, Canada
Karoly Molnar, Meyer Sound Labs, Inc.,
Berkeley, CA, USA

Active acoustics uses digital signal processing and electroacoustics to deliver adjustable correction and enhancement of the response of acoustic enclosure used for music and speech communication. Using variable virtual architecture, correction and adaptation of existing acoustics can be done quickly and efficiently for each specific application, something not possible using passive variable acoustics. Recent advances in active acoustics allow us to create superb stage and auditorium acoustics for performing musicians and listening audiences, as well as for recordings. The panelists will discuss about the state of the art in active acoustics, the challenges and solutions encountered in the field, and brainstorm about future directions.

Workshops

Workshop 10
Saturday, April 28
Room Bartók

09:00 – 10:00

SCREEN-LESS NAVIGATION FOR HIGH-RESOLUTION AUDIO ON BLU-RAY DISC

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

Panelists: *Jim Anderson*, New York University,
New York, NY, USA
Tomi Pietilä, Producer/Engineer, Tampere, Finland
Darcy Proper, Wisseloord Studios, Hilversum,
The Netherlands
David Walstra, Consultant AV Entertainment
Industry

High-resolution audio, presented as uncompressed LPCM, has been waiting for a suitable transport format for some time. The Blu-ray Disc (BD) format offers such a transport and supports the necessary linear and lossless codecs as part of its basic specification. While many BD players can be found in home theater and games environments, there are some issues that need to be addressed before they can be introduced into a hi-fi environment that does not have a screen to present visual menus for audio stream setup and track selection. This recommended method specifies a structure for authoring a BD ROM to enable playback in screen-less consumer systems, and to provide simple track selection from the remote control.

Workshop 11
Saturday, April 28
Room Brahms

10:30 – 12:00

MUSHRA RELOADED

Chair: **Judith Liebetrau**, Technical University Ilmenau,
Ilmenau, Germany

Panelists: *Poppy Crum*, Dolby Laboratories,
San Francisco, CA, USA
Frederik Nagel, Fraunhofer Institute
for Integrated Circuits IIS/International
Audio Laboratories, Erlangen, Germany
Thomas Sporer, Fraunhofer Institute
for Integrated Circuits IIS, Ilmenau, Germany

Since its finalization in 2001 Recommendation ITU-R BS.1534, nicknamed MUSHRA, has become very popular. MUSHRA is an acronym for “multi stimulus with hidden reference and anchors.” Many researchers created variants of the original design and called it still MUSHRA, and very often test reports are missing important information about test parameters and statistical analysis. Very often the mandatory anchor is not used.

ITU-R WP 6C is currently working on a revision of MUSHRA. This revision will specify in more detail how to perform the test and the statistical analysis, will add some of the most popular variants as official alternatives, and will offer new, more powerful anchors.

This workshop will present current discussions and results from the spring 2012 meeting of ITU-R.

Workshops

Workshop 12

Saturday, April 28

12:30 – 14:30

Room Brahms

SPATIAL SOUND REPRODUCTION WITH HEIGHT: WHY, WHERE, HOW, AND WHEN?

Co-Chairs: **Frank Melchior**, BBC R&D, Salford, UK
Florian Völk, Technical University Munich,
Munich, Germany

Panelists: *Jan-Mark Batke*, Technicolor, Research
and Innovation, Hannover, Germany
Kimio Hamasaki, NHK Science and Technical
Research Laboratories, Tokyo, Japan
Michael Kelly, DTS, Inc., London, UK
Stephan Mauer, IOSONO GmbH, Erfurt, Germany
Norbert Niemczyk, Daimler AG, Sindelfingen,
Germany
Jan Plogsties, Fraunhofer Institute for Integrated
Circuits IIS, Erlangen, Germany
Nicolas Tsingos, Dolby Laboratories, Inc.,
San Francisco, CA, USA

At the present time, expanding spatial audio reproduction to the vertical plane seems to be the next logical step from an engineering point of view. The consequence is that new transmission formats, reproduction and production techniques, and processing methods are required. Before starting the discussion from a technical point of view, we propose to determine applications and environments for such new reproduction systems. The aim of this workshop is to ask and discuss the six W's of information gathering in terms of spatial audio with height: Who is it about? What will happen? Where will it take place? When will it take place? Why will it happen? How will it happen? The workshop will bring together experts of different approaches and use cases where spatial audio reproduction with height is considered being important now and in the future.

Workshop 13
Saturday, April 28
Room Brahms

14:30 – 16:30

**DEPLOYING THE LOUDNESS CONCEPT IN EUROPE—
HOW, WHEN AND WHERE**

Chair: **Florian Camerer**, ORF, Vienna, Austria

Panelists: *Alfio DiFazio*, tpc Switzerland
Matthieu Parmentier, France Television
Askan Siegfried, NDR
Alessandro Travaglini, Fox International
Channels Italy
Richard Van Everdingen, Dutch Loudness
Committee

European broadcasters are gearing up fast putting the concept of loudness normalization into practice on the basis of the EBUB recommendation R 128. The panelists are representatives of broadcasters who are on the forefront of this development and will shed a light on the process, the intricacies and solutions.

Workshops

Workshop 14
Sunday, April 29
Room Brahms

09:00 – 10:00

SEMANTIC AUDIO ANALYSIS IN THE REAL WORLD

Chair: **Andrew Mason**, BBC Research & Development,
London, UK

Panelists: *Andy Hill*, I Like Music, London, UK
Mark Sandler, Queen Mary University
of London, London, UK
Nigel Smith, BBC Audio and Music Interactive,
London, UK
Christian Uhle, Fraunhofer Institute for Integrated
Circuits IIS, Erlangen, Germany

Semantic audio analysis presents many opportunities for research, and a huge amount of academic effort is expended. Naturally, a significant proportion of the effort is directed toward solving problems of interest to those doing the research. As a result there is a perception that many of the emergent applications are of limited commercial interest: a musicologist might be interested in minutiae of a composer's style, or in creating a play-list from a collection of 10,000 MP3 files. However, in the wider world there are other potential users: archivists with hundreds of thousands of hours of material to manage, and huge value to be realized from their archives; program makers searching for sound bites; and so on. This workshop aims to present some industry needs and to seek out new opportunities for developments in semantic audio analysis.

Workshop 15
Sunday, April 29
Room Brahms

10:30 – 12:30

SYNCHRONIZATION IN A MULTICHANNEL DIGITAL AGE

Chair: **John Grant**

Panelists: *Steve Harris*
Stefan Heinzmann
Thomas Sporer, Fraunhofer Institute for Digital
Media Technology, IDMT, Ilmenau, Germany

Once upon a time, not so long ago, in any system built up from separate items of digital audio equipment everything would be locked to house sync. Now, systems are increasingly built from equipment that is connected to an asynchronous network without a separate sync input; indeed, in many environments (such as in the home) there is no house sync available. And, increasingly, sound fields are produced by independent digitally-interfaced loudspeakers. This workshop examines requirements for synchronization in these scenarios and techniques for achieving it.

Workshops

Workshop 16

Sunday, April 29

11:00 – 13:00

Room Bartók

LISTEN PROFESSIONALLY OR TRAIN YOUR EAR!

Chair: **Sungyoung Kim**, Yamaha Corporation,
Hamamatsu, Japan

Panelists: *Kazuhiko Kawahara*, University of Kyusyu,
Fukuoka, Japan
Doyuen Ko, McGill University, Montreal,
Quebec, Canada
Søren Vase Legarth, Delta SenseLab,
Hørsholm, Denmark
Atsushi Marui, Tokyo University of the Arts,
Tokyo, Japan
Mark McKinnon-Bassett, University
of Sydney, Sydney, NSW, Australia
Sean Olive, Harman International,
Northridge, CA, USA

It has been generally accepted that critical listening ability is essential for audio engineers. Recent training programs provide multiple trainees with fast acquisition of such listening ability through a systematic curriculum optimized for the required task. Considering the interests and growth of ear training in the audio communities, it is timely and important to have a chance to share and discuss the opinions from the experts about necessary features and methods that assist trainees in acquiring the critical listening ability with efficiency, both for personal and group training. For this purpose, a workshop at the previous AES 131st convention invited panelists from all around the world who shared their in-depth experience, know-how, and insights in ear training. The current workshop is an extension of the previous one that aims to let workshop attendees experience and compare the characteristic functions of various ear training programs through hands-on demonstrations by the panelists. While the workshop locally aims to provide the attendees with a chance to experience theoretical and empirical matters of ear training programs around the world, it also globally aims to consider the importance of “listening” in the current video-oriented society.

TUTORIALS

T-1	Mix Kitchen: Mixing Techniques of Electronic Music	Thursday, April 26 16:30–18:30 Room Bartók
T-2	Ear Training for Mastering Engineers	Friday, April 27 11:00–13:00 Room Bartók
T-3	Listening Tests Part 1: Experimental Design	Saturday, April 28 09:00–10:30 Room Brahms
T-4	Loudness Leveling and Normalization	Saturday, April 28 10:15–12:15 Room Bartók
T-5	Reality Is Not a Recording/ A Recording Is Not Reality	Saturday, April 28 12:30–14:30 Room Bartók
T-6	Drum and Percussion Programming	Saturday, April 28 14:30–16:30 Room Liszt
T-7	How Does It Sound Now? The Evolution of Audio	Saturday, April 28 15:00–16:30 Room Bartók
T-8	Semantic Web and Semantic Audio Technologies	Saturday, April 28 16:30–18:00 Room Liszt
T-9	Listening Tests Part 2: Statistical Analysis	Saturday, April 28 16:30–18:00 Room Brahms
T-10	Noise on the Brain—Hearing Damage on the Other Side	Sunday, April 29 09:00–10:30 Room Liszt
T-11	Design of a Dynamic Range Compressor	Sunday, April 29 09:00–10:30 Room Bartók
T-12	Large Room Acoustics: Basics and New Developments	Sunday, April 29 11:00–13:00 Room Liszt
T-13	Binaural Auditory Models	Sunday, April 29 12:00–13:00 Room Lehar
T-14	Small Room Acoustics	Sunday, April 29 14:00–15:30 Room Liszt
T-15	The Making of the Beach Boys <i>Smile</i> Sessions	Sunday, April 29 15:00–16:30 Room Brahms

Tutorial 1**Thursday, April 26****Room Bartók****16:30 – 18:30****MIX KITCHEN: MIXING TECHNIQUES
OF ELECTRONIC MUSIC**

Presenter: **Marek Walaszek**, Addicted to Music Media Group,
Jozefow, Poland

This tutorial will be based on already-done mixes that will show techniques of modern mixing in electronic and semi-electronic modern music. Marek Walaszek will show a few mixes and talk through the approach to the mixes, effects used, and mix philosophy. He will talk about main production errors, how to prevent them, and how to cure them in the mixing stage.

This tutorial will be supported by a power point presentation and an extensive Q&A.

Tutorials

Tutorial 2

Friday, April 27

11:00 – 13:00

Room Bartók

EAR TRAINING FOR MASTERING ENGINEERS

Presenters: **Andrés Mayo**, Andres Mayo Mastering,
Buenos Aires, Argentina
Darcy Proper, Wisseloord Studios,
Hilversum, The Netherlands

This tutorial includes a comprehensive set of tools to improve ear training, focused on what the Mastering Engineer needs to do his job. Dynamic EQ, de-essing, de-woofing, and many other techniques will be shown, aiming to help audience to recognize different ranges of frequency.

Tutorial 3
Saturday, April 28
Room Brahms

09:00 – 10:30

LISTENING TESTS PART 1: EXPERIMENTAL DESIGN

Presenters: **Søren Bech**, Bang & Olufsen a/s,
Struer, Denmark
Nick Zacharov, Genelec OY, Finland

This tutorial aims at discussing current practices of listening test preparation and execution. It will cover the choice of test items, the selection of participants, and, last, the means of test design.

Questions such as "What are naive or expert listeners?" "What is a convenience sample and what can we learn from the results?" "What is the difference between small and intermediate artifacts?" "Might other test methods be more adequate?" "Which characteristics should the items under test have?" exist but are hardly topics of publication in audio research.

This tutorial aims at providing researchers with a better understanding what happens during listening tests and giving guidelines for a better listening test practice.

Tutorials

Tutorial 4

Saturday, April 28

Room Bartók

10:15 – 12:15

LOUDNESS LEVELING AND NORMALIZATION—BASICS, MISUNDERSTANDINGS, DANGERS, AND SOLUTIONS

Presenter: **Florian Camerer**, ORF, Vienna, Austria,
Chairman EBU-group PLOUD

The switch from peak to loudness normalization is in full swing not only in Europe, but worldwide. There is an international measurement algorithm (ITU-R BS.1770) as well as a few recommendations, all based on that algorithm. This is arguably the biggest change in the audio leveling world in decades. This tutorial will cover the basics of loudness work and look into some details of where potential pitfalls lie, where there might be or are misunderstandings or misconceptions, and give a brief overview of the current status of loudness implementations in Europe and the world. Examples will be played to illustrate the presentation..

Tutorial 5
Saturday, April 28
Room Bartók

12:30 – 14:30

**REALITY IS NOT A RECORDING/
A RECORDING IS NOT REALITY**

Presenter: **Jim Anderson**, New York University,
New York, NY, USA

The former *New York Times* film critic, Vincent Canby, wrote “all of us have different thresholds at which we suspend disbelief, and then gladly follow fictions to conclusions that we find logical.” Any recording is a “fiction,” a falsity, even in its most pure form. It is the responsibility, if not the duty, of the recording engineer and producer, to create a universe so compelling and transparent that the listener isn't aware of any manipulation. Using basic recording techniques, and standard manipulation of audio, a recording is made, giving the listener an experience that is not merely logical but better than reality. How does this occur? What techniques can be applied? How does an engineer create a convincing loudspeaker illusion that a listener will perceive as a plausible reality? Recordings will be played.

Tutorials

Tutorial 6

Saturday, April 28

Room Liszt

14:30 – 16:30

DRUM AND PERCUSSION PROGRAMMING

Presenter: **Justin Paterson**, London College of Music,
University of West London, London, UK

Drum programming has been evolving at the heart of many studio productions for some 30 years. Over this period, technological opportunities for enhanced creativity have multiplied in numerous directions, from sequenced MIDI one-shots to sample loops, and from DAW cut and stretch techniques to deterministic beat-slicer plug-ins, etc. The palette of sounds available today ranges from ever more realistic to ever more synthetic/exotic. This tutorial will embrace all of these techniques and more—and include numerous live demonstrations. Although introducing all key concepts from scratch, its range and hybridization should provide inspiration even for experienced practitioners, leading up to the state-of-the-art. A number of genres will be covered from the pseudo-realism of jazz and funk, to the exploitation of synthetic textures in “intelligent dance music.”

Tutorial 7
Saturday, April 28
Room Bartók

15:00 – 16:30

HOW DOES IT SOUND NOW? THE EVOLUTION OF AUDIO

Presenters: **Gary Gottlieb (Moderator)**
Jim Anderson
Andres Mayo

One day Chet Atkins was playing guitar when a woman approached him. She said, “That guitar sounds beautiful.” Chet immediately quit playing. He asked, “How does it sound now?” The quality of sound in Chet’s case clearly rested with the player, not the instrument, and the technical and aesthetic quality of our product lies with our engineers and producers, not solely the equipment. The dual significance of this question, “How does it sound now,” informed my research for the last three years and will inform our discussion, since it addresses both the engineer as the driver and the changes we have seen and heard as our methodology evolved through the decades. The book that resulted from this research, *How Does It Sound Now?* received the 2010 ARSC Award for Excellence in Historical Recorded Sound Research for the Best Research in General History of Recorded Sound. One of the most interesting facets of the research, comprised of interviews with top engineers and producers, was the way the conversation kept returning to the thread of quality. They loved to talk about how they strived for quality then, and still do. Let’s talk about how engineers and producers retain quality and create a product that conforms to their own high standards. This may lead to other conversations about musicians, consumers, and the differences and similarities between their standards and our own. How high should our standards be? How did it sound then? How does it sound now? How should it sound tomorrow?

Tutorials

Tutorial 8

Saturday, April 28

Room Liszt

16:30 – 18:00

SEMANTIC WEB AND SEMANTIC AUDIO TECHNOLOGIES

Presenters: **Gyorgy Fazekas**, Queen Mary University
of London, London, UK
Thomas Wilmering

The emerging Semantic Web provides a powerful framework for the expression and reuse of structured data. Recent efforts have brought this framework to bear on the field of Semantic Audio, as well as information management in audio applications. This tutorial will provide an introduction to Semantic Web concepts and how they can be used in the context of music-related studies. We will outline the use of the Resource Description Framework (RDF) and related ontology and query languages. Using practical examples, we will demonstrate the use of the Music and Studio Ontologies, and show how they facilitate interoperability between audio applications and linked data sets on the Web. We will explore how signal processing tools and results can be described as structured data and utilized in audio production.

Tutorial 9
Saturday, April 28
Room Brahms

16:30 – 18:00

LISTENING TESTS PART 2: STATISTICAL ANALYSIS

Presenters: **Poppy Crum**, Dolby Laboratories,
San Francisco, CA, USA
Frederik Nagel, Fraunhofer Institute
for Integrated Circuits IIS/International
Audio Laboratories, Erlangen, Germany
Thomas Sporer, Fraunhofer Institute for Digital
Media Technology, IDMT, Ilmenau, Germany

This tutorial aims at discussing current practices of listening test evaluation. Topics to be covered are post-screening of tests results enabling to reject data from unreliable subjects and analysis of the results. This will cover parametric and non-parametric statistics, and the decision when which of these is necessary. Special emphasis will be set on which conclusions from the analysis are adequate from a mathematical perspective. This tutorial aims at providing researchers with a better understanding of the results of listening tests.

Tutorials

Tutorial 10

Sunday, April 29

Room Liszt

09:00 – 10:30

NOISE ON THE BRAIN—HEARING DAMAGE ON THE OTHER SIDE

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

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

Tutorial 11
Sunday, April 29
Room Bartók

09:00 – 10:30

DESIGN OF A DYNAMIC RANGE COMPRESSOR

Presenter: **Josh Reiss**, Queen Mary University of London,
London, UK

Despite being one of the most widely used audio effects, dynamic range compression, remains poorly understood. And there is little formal knowledge and analysis of compressor design techniques. This tutorial will describe several approaches to digital dynamic range compressor design. It will demonstrate how to build a compressor from the ground up, and provide audio examples showcasing differences between designs. It will explain why the designs sound different, and provide distortion-based metrics to analyze their quality. It will also provide recommendations for high performance compressor design.

Tutorials

Tutorial 12

Sunday, April 29

Room Liszt

11:00 – 13:00

LARGE ROOM ACOUSTICS: BASICS AND NEW DEVELOPMENTS

Presenter: **Diemer de Vries**, Amsterdam, The Netherlands

In the sound recording and reproduction chain, the audio engineer has to deal with the acoustics of the environment—usually an enclosed space, a “room”—he or she is working in. The physical properties of the acoustics of a room can be studied by advanced measurements or simulations, but what counts at the end is the perceptual quality of the acoustics. The challenge of the modern room acoustics researcher is to find an unambiguous relation between physics and perception. In the tutorial—which is focused on large rooms—it will be discussed how, long ago, only “reverberation time” was considered as the important parameter. Later, and until now, parameters were derived from impulse responses. Recent developments show that also these responses may have had their longest time as the scientific foundation of room acoustics.

Tutorial 13
Sunday, April 29
Lehar

12:00 – 13:00

BINAURAL AUDITORY MODELS

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

The working principles of brain mechanisms of binaural hearing have been debated during the last decade. In 1990s the common thinking was that human binaural decoding is based on delay lines and coincidence counters, as proposed by the Jeffress model. Later, some neurophysiological findings questioned the existence of such delay lines, and some evidence was found bolstering the count-comparison model proposed by Bekesy. In count-comparison model, the binaural differences are rate-coded between the left and brain right hemispheres. This tutorial will introduce the basic principles of most common binaural auditory models, and review some latest improvements in the models.

Tutorials

Tutorial 14

Sunday, April 29

14:00 – 15:30

Room Liszt

SMALL ROOM ACOUSTICS

Presenter: **Ben Kok**, BEN KOK – acoustic consulting,
Uden, The Netherlands

Acoustic basics of small rooms will be discussed. Specific issues related to the size of the room (room-modes) will be addressed. Absorption, reflection, diffraction, diffusion and how to use it, as well as specific aspects regarding low frequency treatment will be discussed.

Although this will not be a studio design class, specifics and differences of recording rooms and control rooms will be identified, including considerations for loudspeaker and microphone placement.

Tutorial 15
Sunday, April 29
Room Brahms

15:00 – 16:30

THE MAKING OF THE BEACH BOYS *SMILE* SESSIONS

Presenters: **Mark Linett**
Barry Marshall

Arguably the greatest “lost” album of all time, The Beach Boys *Smile* album sessions were finally released last November, in both a 2 CD and 5 CD version, along with vinyl and other digital configurations. The co-producer of the project, Mark Linett, will be interviewed by producer and educator Barry Marshall about the legendary 1966–67 sessions. The presentation will provide historical context on the role of producer Brian Wilson, as well as a focus on the technical and logistical challenges (like mixing and mastering from 45-year-old tapes at different configurations, speeds and sizes!) faced by co-producer and mix engineer Mark Linett in compiling this landmark release.

Cumulative Index

Key: Special Events (SE)/Students (S/C)/ Paper Numbers/eBriefs (EB)/Workshops (W)/Tutorials (T)

Participant	Event / Paper #	Page
Agavanakis, Kyriakos	EB49	98
Aggrawal, Hari Om	8652	70
Agkavanakis, Kyriakos	8615	48
Aichmann, Horst	8657	73
Alemanno, Andrea	8612	45
Alemanno, Andrea	8613	46
Altendorf, Peter	8680	89
Andersen, Michael A. E.	8629	57
Anderson, Jim	8592	33
Anderson, Jim	T5	125
Anderson, Jim	T7	127
Anderson, Jim	W10	112
Ando, Akio	8647	67
Aßmann, Ralph W.	EB51	99
Backman, Juha	8674	84
Backman, Juha	8673	83
Backman, Juha	P9 Chair	48
Badeau, Roland	8594	35
Bank, Balázs	8591	34
Bank, Balázs	8593	35
Barbose, Alvaro	W7	109
Barbour, James L.	8598	38
Batke, Jan-Mark	W12	114
Batke, Johann-Markus	8659	74
Bayer, Stefan	8654	71
Bech, Søren	8639	63
Bech, Søren	T3	123
Bergen, Sakari	8609	44
Bessette, Bruno	8654	71
Biba, Dominik	8671	82
Birch, Thomas Haagen	8631	58
Biswas, Arijit	8637	61
Bitzer, Jörg	8638	62
Blauert, Jens	W6	108
Bock, Stefan	W10 Chair	112
Bolaños, Javier Gómez	8604	41
Brandenburg, Karlheinz	W6 Chair	108
Brandt, Matthias	8638	62
Brixen, Eddy B.	8620	50
Brixen, Eddy B.	W1 Chair	103
Brock, Nathan	W7	109
Bruce, Roderik	EB51	99
Buchholz, Jörg	8645	66

INDEX

Participant	Event / Paper #	Page
Buhl, Niels Christian	8629	57
Bullock, Jamie	8622	51
Burkart, Florian	EB51	99
Buttner-Schnirer, Pdraig	EB42	93
Cabrera, Densil A.	8614	47
Cabrera, Densil A.	8642	64
Camerer, Florian	T4	124
Camerer, Florian	W13 Chair	115
Carôt, Alexander	8657	73
Carôt, Alexander	W7 Chair	109
Case, Alex U.	8592	33
Cauchi, Marija	EB51	99
Cengarle, Giulio	8599	38
Cerny, Robert	EB44	94
Chang, Ji-ho	8597	37
Chen, Zewei	8643	65
Chinen, Toru	8654	71
Cho, Jaeyong	8649	68
Choi, Jung-Woo	8602	40
Choi, Jung-Woo	8618	49
Choi, Jungmin	8649	68
Chon, Sonh Hui	8603	40
Christensen, René	8672	83
Cichowski, Janusz	8605	41
Cobos, Maximo	8635	60
Cobos, Maximo	8668	80
Crone, Lasse	8630	57
Crum, Poppy	T9	129
Crum, Poppy	T10	130
Crum, Poppy	W11	113
Czyzewski, Andrzej	8605	41
Czyzewski, Andrzej	8636	61
Dash, Ian	8614	47
Date, Toshihiko	8600	39
Daudet, Laurent	8594	35
De Klerk, Leo	W9	111
de Vries, Diemer	P4 Chair	35
de Vries, Diemer	T12	132
Deboy, Daniel	EB51	99
Delikaris-Manias, Symeon	8666	79
Derrez, Clement	EB51	99
Dewhirst, Martin	8639	63
DiFazio, Alfio	W13	115
Dilger, Thierry	EB46	95
Dimoulas, Charalampos	8663	76
Disch, Sascha	8654	71
Drosos, Konstantinos	8615	48
Drossos, Konstantinos	EB49	98
Eleftheria Georganti,	8619	50
Enatsu, Masaaki	8600	39
Escolano, Jose	8667	79

Participant	Event / Paper #	Page
Escolano, José	8668	80
Evangelista, Gianpaolo	8590	33
Faller, Christof	P15 Chair	65
Faller, Christof	P17 Chair	72
Faller, Christof	W8	110
Favrot, Sylvain	8645	66
Fazekas, Gyorgy	T8	128
Fenton, Steven	8626	54
Fink, Marco	8637	61
Finn, Saoirse	8588	31
Floros, Andreas	8596	37
Floros, Andreas	8615	48
Floros, Andreas	EB49	98
Francombe, Jon	8639	63
Frank, Matthias	8586	30
Fu, Qiang	8678	86
Fuchs, Guillaume	8654	71
Furuya, Ken'ichi	8664	78
Gan, Woon-Seng	EB47	97
Geier, Matthias	8669	81
Geluso, Paul	8595	36
Georganti, Eleftheria	8617	49
Gerazov, Branislav	8634	60
Ghelfi, Gabriele	8632	58
Girin, Laurent	8594	35
Görne, Thomas	8659	74
Gottlieb, Gary	T7	127
Gournay, Philippe	8654	71
Grand, Aneline	8633	59
Grant, John	W15 Chair	117
Griesinger, David	8610	44
Griesinger, David	P20 Chair	82
Griesinger, David	W6	108
Grill, Bernhard	8654	71
Hamasaki, Kimio	W12	114
Haneda, Yoichi	8664	78
Härmä, Aki	8644	65
Harris, Steve	W15	117
Hartmann, Christian	8680	89
Hatziantoniou, Panagiotis D.	8658	74
Hawksford, Malcolm O. J.	8665	78
Heinzmann, Stefan	W15	117
Helmrich, Christian	8654	71
Hernández, J.	8676	85
Hiebel, Holger	EB48	97
Hiipakka, Marko	8666	79
Hill, Adam J.	8665	78
Hill, Andy	W14	116
Hiwasaki, Yusuke	8664	78
Holzinger, Axel	W5	107
Hsu, Han-Wen	8653	70

INDEX

Participant	Event / Paper #	Page
Hughes, Braham	8660	75
Ishikawa, Keita	8670	82
Ivanovski, Zoran	8634	60
Iversen, Niels Elkjær	8631	58
Jacobsen, Finn	8597	37
Jacobsen, Finn	8627	55
Jaksche, Michael	8661	75
Jonsson, Lars	W5	107
Juha Backman	W3	105
Kaliakatsos-Papakostas, M.	8596	37
Kalliris, George	8663	76
Kamekawa, Toru	8600	39
Kanellopoulos, N-G	8615	48
Kang, Dong-Soo	8618	49
Kawahara, Kazuhiko	W16	118
Kawanaka, Aiko	8600	39
Kellermann, Walter	8637	61
Kelloniemi, Antti	P19 Chair	78
Kelloniemi, Antti	W3 Chair	105
Kelly, Michael	P13 Chair	60
Kelly, Michael	W2 Chair	104
Kelly, Michael	W12	114
Kikuri, Kei	8654	71
Kim, Dohyung	8656	72
Kim, Miyoung	8654	71
Kim, Sungyoung	W16 Chair	118
Kim, Yang-Hann	8602	40
Kim, Yang-Hann	8618	49
Kjelin, Henrik	8620	50
Kjelin, Henrik	W1	103
Kjörling, Kristofer	8654	71
Kleæ, Mariusz	8681	88
Kleczkowski, Piotr	8641	77
Knott, Arnold	8629	57
Knott, Arnold	8630	57
Knott, Arnold	8631	58
Ko, Doyuen	8603	40
Ko, Doyuen	W9	111
Ko, Doyuen	W16	118
Kok, Ben	T14	134
Kokkinis, Elias	EB49	98
Kokkinis, Elias K.	8617	49
Kondapalli, Ravi	8682	88
Koppens, Jeroen	8644	65
Korn, Thomas	8679	87
Kostek, Bozena	8607	42
Kostek, Bozena	P6 Chair	41
Kostek, Bozena	P7 Chair	44
Kotsakis, Rigas	8663	76
Kowalski, Damian	8608	43
Koyama, Shoichi	8664	78

Participant	Event / Paper #	Page
Kozlowski, Piotr Z.	8608	43
Kubota, Hajime	8670	82
Kunze, Kristina	8679	87
Laitinen, Mikko-Ville	8623	53
Langhans, Sonja	W5	107
Lantini, Fabrizio	8613	46
Lapierre, Jimmy	8654	71
Lavoie, Michel	8606	42
Lecomte, Jérémie	8654	71
Ledergerber, Stefan	W5 Chair	107
Lee, Hyunkook	8625	54
Lee, Jung-Min	8618	49
Lee, Kangeun	8656	72
Lee, Shihwa	8656	72
Lefebvre, Roch	8654	71
Legarth, Søren Vase	W16	118
Leonard, Brett	EB42	93
Liebetau, Judith	8679	87
Liebetau, Judith	W11 Chair	113
Lindenbauer, Thomas	8586	30
Linett, Mark	T15	135
Liu, Chi-Min	8653	70
Liu, Yu	8643	65
Liutkus, Antoine	8594	35
Lopatka, Kuba	8636	61
Lopez, Jose J.	8628	55
Lopez, Jose J.	8635	60
Lopez, Jose J.	8667	79
López, José J.	8668	80
Lösler, Stefan	8623	53
Lowis, Christopher	8589	32
Lühmann, Hans-G.	8567	31
Lühmann, Hans-G.	EB43	94
Lukasik, Ewa	8621	51
Macura, Paulina	8641	77
Madden, Andrew	8592	33
Maeda, Kazuaki	8670	82
Malecki, Pawel	8624	53
Mansbridge, Stuart	8588	31
Marchand, Sylvain	8594	35
Marschall, Marton	8645	66
Marshall, Barry	T15	135
Martens, William	8642	64
Marti, Amparo	8635	60
Marui, Atsushi	8600	39
Marui, Atsushi	W16	118
Masi, Alessandro	EB51	99
Mason, Andrew	W14 Chair	116
Mason, Russell	8639	63
Mateos, Toni	8599	38
Mauer, Stephan	W12	114

INDEX

Participant	Event / Paper #	Page
May, Tobias	8619	50
Mayo, Andres	T2	122
Mayo, Andres	T7	127
McGarry, Gregory	8650	69
McKinnon-Bassett, Mark	8642	64
McKinnon-Bassett, Mark	W16	118
Melchior, Frank	W6	108
Melchior, Frank	W12 Chair	114
Mellow, Tim	8673	83
Menzer, Fritz	8584	29
Menzer, Fritz	EB44	94
Merkel, Tobias	8567	31
Merkel, Tobias	EB43	94
Millitzer, Michael	W8	110
Møller, Martin Bo	8627	55
Molnar, Karoly	W9	111
Mønster, Jakob Døllner	8630	57
Morfi, Ezequiel	S/C	22
Mossman, Miles	8614	47
Mourjopoulos, John	8617	49
Mourjopoulos, John	8619	50
Multrus, Markus	8654	71
Nagel, Frederik	T9	129
Nagel, Frederik	W11	113
Navarro, Juan M.	8667	79
Negishi, Hirokazu	8670	82
Neuendorf, Max	8654	71
Niemczyk, Norbert	W12	114
Norcross, Scott	8606	42
Noriega, Juan E.	8667	79
Norimatsu, Takeshi	8654	71
Nyakacska, Zoltan	W2	104
Nymand, Mikkel	W1	103
Oh, Eumi	8654	71
Ohga, Juro	8670	82
Olive, Sean	8683	87
Olive, Sean	W16	118
Olsen, Martin	8627	55
Oo, Nay	EB47	97
Oohira, Ikuo	8670	82
Oomen, Werner	8654	71
Opitz, Martin	8586	30
Ortega, J. A.	8676	85
Paczkowska, Elzbieta	8641	77
Park, Haekwang	8649	68
Park, Munhum	8644	65
Parker, Julian	8609	44
Parmentier, Matthieu	W13	115
Paterson, Justin	EB41	93
Paterson, Justin	T6	126
Pedersen, Jeppe Arnsdorf	8630	57

Participant	Event / Paper #	Page
Pérez-Lorenzo, Jose M.	8668	80
Pfund, Colin	S/C	22
Pietilä, Tomi	W10	112
Pihlajamäki, Tapani	8623	53
Pike, Christopher	8589	32
Pinel, Jonathan	8594	35
Piotrowski, Szymon	8651	69
Plewa, Magdalena	8607	42
Plewa, Magdalena	8651	69
Plewa, Magdalena	S/C	22
Plogsties, Jan	P8 Chair	45
Plogsties, Jan	P10 Chair	51
Plogsties, Jan	W12	114
Ploug, Rasmus	8631	58
Pluta, Marek	8624	53
Pluta, Marek	8641	77
Politis, Archontis	8666	79
Pomberger, Hannes	8671	82
Price, Anthony	8662	76
Proper, Darcy	T2	122
Proper, Darcy	W10	112
Pueo, Basilio	8628	55
Pulkki, Ville	8604	41
Pulkki, Ville	8609	44
Pulkki, Ville	8623	53
Pulkki, Ville	8666	79
Pulkki, Ville	P1 Chair	29
Pulkki, Ville	P3 Chair	33
Pulkki, Ville	T13	133
Purnhagen, Heiko	8654	71
Quackenbush, Schuyler	8654	71
Raimond, Yves	8589	32
Rämö, Jussi	8609	44
Rämö, Jussi	8640	63
Redaelli, Stefano	EB51	99
Rehder, Jörg	W3	105
Reining, Friedrich	W3	105
Reiss, Joshua D.	8588	31
Reiss, Joshua D.	8648	68
Reiss, Joshua D.	8655	72
Reiss, Joshua D.	EB1 Chair	93
Reiss, Josh	T11	131
Répás, József	8611	45
Rettelbach, Nikolaus	8654	71
Richard, Gaël	8594	35
Ritter, Tom	8567	31
Ritter, Tom	EB43	94
Rizzi, Lorenzo	8632	58
Robilliard, Julien	8654	71
Robinson, Martin	8622	51
Roginska, Agnieszka	8592	33

INDEX

Participant	Event / Paper #	Page
Rubisch, Julian	8661	75
Ruiz, German	8676	85
Rumsey, Francis	P2 Chair	31
Sadolin, Cathrine	8620	50
Sadolin, Cathrine	W1	103
Salvachua, Belen	EB51	99
Sandler, Mark	P12 Chair	57
Sandler, Mark	W14	116
Sandler, Mark	EB52	99
Schneider, Andreas	8650	69
Schneider, Martin	EB45	95
Schuijers, Erik	8654	71
Schüller, Dietrich	W4	106
Seetharaman, Prem	8585	29
Seng, Chong Kok	8654	71
Shay, Greg	W5	107
Siegfried, Askan	W13	115
Simpson, Andrew	EB52	99
Sinha, Deepen	8652	70
Skov, Ulrik	8672	83
Skovenborg, Esben	8616	48
Smith, Nigen	W14	116
Son, Changyong	8656	72
Sontacchi, Alois	8586	30
Spikofski, Gerhard	8680	89
Sporer, Thomas	P14 Chair	63
Sporer, Thomas	T9	129
Sporer, Thomas	W11	113
Sporer, Thomas	W15	117
Spors, Sascha	8669	81
Spors, Sascha	EB50	98
Steinberg, Karl	W7	109
Stockman, Tony	8655	72
Sturmel, Nicolas	8594	35
Sung, Ben-Zhen	8682	88
Suzuki, Ryouyuke	8670	82
Svedström, Thomas	8644	65
Szigetvári, Andrea	W7	109
Takanen, Marko	8666	79
Tarzia, Stephen P.	8585	29
Tassart, Stephan	8633	59
Tatlas, Nicolas-Alexander	8615	48
Tatlas, Nicolas-Alexander	8658	74
Tatlas, Nicolas-Alexander	EB49	98
Terrell, Michael	EB52	96
Thesing, Robin	8650	69
Torres, Ana	8628	55
Travaglini, Alessandro	8612	45
Travaglini, Alessandro	8613	46
Travaglini, Alessandro	W13	115
Tsilfidis, Alexandros	EB49	98

Participant	Event / Paper #	Page
Tsingos, Nicolas	W12	114
Uhle, Christian	W14	116
Uncini, Aurelio	8612	45
Valentino, Gianluca	EB51	99
Välimäki, Vesa	8640	63
Välimäki, Vesa	P16 Chair	68
van de Par, Steven	8619	50
van Dinther, Ralph	8644	65
Van Everdingen, Richard	W13	115
Villemoes, Lars	8654	71
Völk, Florian	8677	86
Völk, Florian	W12 Chair	114
Vrahatis, Michael N.	8596	37
Wakefield, Jonathan	8626	54
Wakefield, Jonathan	8660	75
Walaszek, Marek	T1	121
Waldenberger, Philip	S/C	22
Wallaszkovits, Nadja	W4 Chair	106
Walstra, David	W10	112
Wang, Chao	W3	105
Wang, Yonghao	8648	68
Wang, Yonghao	8678	86
Weckström, Anders	W3	105
Weigand, Stefan	8659	74
Werner, Stefan	W6	108
Wersényi, György	8611	45
Wersényi, György	8675	85
Wersényi, György	P22 Chair	87
Wiederholtz, Erik	W3	105
Wierstorf, Hagen	8669	81
Wierstorf, Hagen	EB50	98
Wilde, Stephan	8654	71
Wilkie, Sonia	8655	72
Williams, Michael	8601	39
Willyard, David	W7	109
Wilmering, Thomas	T8	128
Wittek, Helmut	W8 Chair	110
Wollmann, Daniel	EB51	99
Wolters, Martin	8650	69
Woszczyk, Wieslaw	8603	40
Woszczyk, Wieslaw	W9 Chair	111
Xiang, Ning	8668	80
Xie, Bosun	8643	65
Xie, Bosun	8646	66
Yli-Kätkä, Veli-Matti	8609	44
Yu, Guangzheng	8643	65
Zacharov, Nick	T3	123
Zebrowski, Zach	W2	104
Zhang, Chengyun	8646	66
Zhu, Xiangyu	8678	86
Zotter, Franz	8671	82