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AND PROFESSIONAL EVENTS

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AES Conferences and Conventions .... **Back cover**

Ron Streicher

Welcome back to Berlin! It has been more than a decade since the AES last held a convention in this historic city, and much has changed in the intervening years: the audio industry has continued its progression toward totally digital, and in many cases workstation-based creation and production; the Internet has become a primary means of distribution and sales for all types of audio—and video—programming; formal audio education has continued to expand throughout the worldwide academic community; many recording companies have been absorbed into one of a few “corporate giants” while at the same time the individual artist/engineer/producer has assumed a significant role in all genres of the entertainment industry; similarly, a lot of the development, manufacturing, and production of audio equipment has merged toward either the large conglomerate companies or “cottage” entrepreneurs; and economic and political factors have generated new relationships among the peoples, nations, and regions of the world.

The 116th AES Convention addresses all of these issues. The program of technical papers, workshops, and tutorial seminars will offer you the knowledge and insights necessary to succeed in these challenging times. The exhibition will allow you to lay “hands on” the latest production tools and, even more important, the opportunity to meet and talk with the people who have designed this equipment. The education track of events—which should be of interest to all attendees because, after all, we should never stop learning—is the key to securing the industry’s future. And, finally, our social events and other gatherings provide the opportunity for the “networking” with colleagues from all over the world that is so crucial to personal and professional growth.

Please also be sure to stop by the AES Membership and Publications booths. If you are not yet a member, we invite you to join, and we urge you to look over the wealth of information available in the many special publications available for you to add to your personal library.

Ron Streicher
President

Reinhard O. Sahr

At last I can welcome the guests of the 116th AES Convention for a second time to Berlin, Germany's capital city. Following the end of Berlin's unnatural division and isolation, it is a mark of distinction that the leading organization in professional audio worldwide has found its way again to the River Spree.

Berlin is a pulsating city, home to all important fields of the ProAudio industry. The traditional film studios, the great variety of broadcasters, music publishers, recording studios, the marvelous concerts and theater performances, the special event and sound industry is all present in this business nexus.

This variety is reflected in the convention program, which is more extensive than ever before. New ideas and developments have become so comprehensive and voluminous that we decided to concentrate and to structure the program planning around specific topics of interest. The clearly laid out and arranged program offering makes it easier for you to understand and to gather all the information you need.

In the technical program (papers, posters, workshops, tutorials, and exhibitor seminars), you will hear about both high-level research topics and practical information that you can use in your day-to-day work. Many of the 116th's exhibitors have already announced very exciting new product presentations and innovations.

Numerous student activities, including recording competitions and an education forum, are helping to develop future industry leaders. But the past is not forgotten, as the convention will also have many historical presentations with papers and exhibits of vintage audio and film equipment. The historical highlight will be a tour to the old Babelsberg film studios, founded nearly 100 years ago.

Another very important mission of the AES, reviewing new technologies and developing industry standards, is performed at the meetings of many standards and technical council committees. To these meetings, everybody with technical qualifications and interest is invited.

Another good reason to visit Berlin and the AES Convention is the opening of eastern European markets and the expansion of the European Union in 2004. Berlin has increasingly become the focus of international attention. As both an east-west and north-south hub, the city is becoming increasingly attractive for trade shows and conventions. This growing cultural and media capital has become the center of electronic media in Europe. Berlin has always had a resounding name worldwide and now has the chance to be a congress center for East and West.

I wish all visitors a fulfilling and informative convention, and I hope you have time to get to know the city of Berlin and its surroundings.

Reinhard O. Sahr
Convention Chair

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REGISTRATION
Registration Desk Hours:

Friday, May 7	16:00 h – 19:00 h
Saturday, May 8	08:00 h – 17:30 h
Sunday, May 9	08:30 h – 17:30 h
Monday, May 10	08:30 h – 17:30 h
Tuesday, May 11	08:30 h – 15:30 h

REGISTRATION FEES:
AES MEMBERS (all grades)
Full Program (technical sessions, workshops, seminars, special events, and exhibits):

• AES Honorary/Life Members	No Fee
• AES Members & Associates	EUR 200
• AES Student Members	EUR 50

Full Program plus **Symposium**

• AES Honorary/Life Members	EUR 60
• AES Members & Associates	EUR 260
• AES Student Members	EUR 80

Symposium Only (includes exhibition)**:**

• AES Members & Associates	EUR 100
• AES Student Members	EUR 40

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

• AES Members & Associates	EUR 20
• AES Student Members	EUR 10

NONMEMBERS

• Full Program	EUR 280
• Full Program plus Symposium	EUR 340
• Student (with I.D.), Full Program	EUR 90
• Student, Full Program & Symposium	EUR 120

Symposium Only (includes exhibition)**:**

• Nonmembers	EUR 120
• Student Nonmembers	EUR 50

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

• Nonmembers	EUR 30
• Nonmember Students	EUR 15

INDIVIDUAL TICKETS:

• One Full Day (Members)	EUR 65
• One Full Day (Nonmembers)	EUR 90
• One Full Day (Students)	EUR 20
• One Workshop (All)	EUR 30
• One Tutorial Seminar (All)	EUR 20

Individual tickets may be purchased at the Special Events Desk. You must have purchased a 4-day Exhibits Only pass to obtain individual event tickets.

PAYMENTS

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

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

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

Saturday, May 8	09:00 h – 18:00 h
Sunday, May 9	09:00 h – 18:00 h
Monday, May 10	09:00 h – 18:00 h
Tuesday, May 11	09:00 h – 18:00 h

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

EXHIBITS

The **Exhibit Booths** are located in Halls 2 and 4. **Demonstration Rooms** are on the mezzanine between Halls 2 and 4 and just outside Hall 4 on ground level. Please refer to the *116th Convention Exhibitor Directory* for a complete list of exhibitors and their locations.

Exhibit Hours

Saturday, May 8	10:00 h – 18:00 h
Sunday, May 9	10:00 h – 18:00 h
Monday, May 10	10:00 h – 18:00 h
Tuesday, May 11	10:00 h – 17:00 h

MEMBERSHIP

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

AES PUBLICATIONS

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

116th CONVENTION PAPERS

Single Copy	EUR	4
Complete Set (single copies of 185 papers)	EUR	135
Complete Set on CD-ROM (single copies of 185 papers)	EUR	135
Complete Set and CD-ROM (single copies of 185 papers plus disk)	EUR	200

PRESS

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

PRESS CENTER

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

Press Center Opening Hours

Saturday, May 8	10:00 h – 18:00 h
Sunday, May 9	10:00 h – 18:00 h
Monday, May 10	10:00 h – 18:00 h
Tuesday, May 11	10:00 h – 17:00 h

AES DAILY

The editorial office of the AES official Daily “Convention News” is located in Hall 2. Three issues are released, the third one serving days 3 and 4 of the convention.

BUSINESS CENTER

The Messe Berlin is equipped with several pay phones using phone cards and credit cards. A Business Center provided with telephones, copy and fax machines, scanners, and printers, as well as Internet connections (e-mail) is available at the Service Center in Hall 7, 2nd Floor.

SHUTTLE BUS SERVICE

No shuttle bus service is avaliable to convention hotels as Berlin provides a wide choice of transportation.

PUBLIC TRANSPORTATION INFORMATION

The Exhibition Grounds operated by Messe Berlin are easy to reach thanks to excellent road and rail links. The city itself also has an extensive and clearly laid out public transport network. The best way, to reach the Messe Entrance South, is to use the S-Bahn Line S75 or S9 in the direction of Spandau from the main Station “Zoologischer Garten.” The train station “Messe Süd” is directly in front of the Entrance to the AES Convention location in the Messegelände.

Taxis will queue up at the South entrance of Messe Berlin. There will be brochures available on site regarding public transportation options.

INFORMATION ON BERLIN

There will be a booth of the Berlin Tourism office in the South Entrance with detailed information about the city of Berlin. Here you will find city maps; you can make hotel reservations, book sightseeing tours, and purchase tickets for the opera, concerts, and theater. You can also get restaurant recommendations and the “Berlin Welcome card,” which gives discounts on many locations (museums, etc.) in the city.

PARKING

Parking spaces are available in the neighborhood of the Convention Entrance (Messe South); P11 and P13 at the “Avus Nordkurve” with a 5-minute walk to the AES (there are 500 places free of charge), another 300 places at the P14 in front of the “Deutschlandhalle” (free of charge), and behind Gate 25 is parking space P18 with 800 places, which cost Euro 7.50 per day per car.

Please follow the AES - P signs that lead you to the mentioned parking areas from the Autobahn and the main streets in the direction to Messe Berlin.

FIRST-AID SERVICES AT MESSE BERLIN

First-aid is available by calling +49 (0)30 3038 2222.

AES CONVENTION HOTELS

Apart Hotel Hanse	+49 (0)30 211 9052
Alsterhof Berlin	+49 (0)30 212 42-0
Berlin Hotel Excelsior	+49 (0)30 315 5-0
Best Western Hotel Boulevard	+49 (0)30 884 25-0
Comfort-Hotel Fruhling am Zoo	+49 (0)30 889 11-0
Concept	+49 (0)30 884 26-0
Crowne Plaza Berlin City Centre	+49 (0)30 210 07-0
Domicil Berlin	+49 (0)30 329 03-0
Dorint Schweizerhof Berlin	+49 (0)30 2696-0
Golden Tulip Residenz Hotel Berlin	+49 (0)30 884 43-0
Hamburg Ringhotel Berlin	+49 (0)30 264 77-0
Hotel Berlin	+49 (0)30 2605-0
Kudamm 101	+49 (0)30 520 055-0
Marriott Hotel	+49 (0)30 2 000-0
Palace Berlin	+49 (0)30 2502-0
Park Plaza Hotel Berlin	+49 (0)30 884 13-0
Steigenberger Hotel Berlin	+49 (0)30 2127-0
Swissotel Berlin	+49 (0)30 220 10-0
Sylter Hof Berlin	+49 (0)30 2120-0
The Westin Grand Berlin	+49 (0)30 2027-0

Opening Ceremonies

AWARDS PRESENTATION AND KEYNOTE ADDRESS
Saturday, May 8, 11:30 h–12:30 h
Room 7.1a-1

Opening Remarks:

- Executive Director Roger Furness
- President Ron Streicher
- Convention Chair Reinhard Sahr

Program:

- AES Awards Presentation
- Keynote Address by Raina Konstantinova, Director

EBU Radio Department: "The Radio Landscape in Europe and Digital Strategies for the Future"

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 Address

Raina Konstantinova, Radio Director of the EBU/UER in Geneva, is this year's Keynote Speaker. Mrs. Konstaninova was the former Radio Director of Bulgarian Broadcast. The topic of her talk is "The Radio Landscape in Europe and Digital Strategies for the Future." The focus of her presentation includes relevant data about the countries in Central and Eastern Europe.

Technical Tours

A total of 11 Technical Tours is planned. They cater to a wide range of interests. All tours have a limited capacity. Tickets will be allocated on a first-come-first served basis. To participate, please sign up at the AES Tours Desk. The cost of each tour is noted below.

TECHNICAL TOUR 1
Konzerthaus Berlin (Concert House Berlin)
Friday, May 7, 13:00 h –16:00 h

When it was opened in 1817 this building by K. F. v. Schinkel was a theatre. It was partly destroyed during the war and re-opened in 1984 as a concert hall with some 1600 seats. You can attend a rehearsel of the Rundfunk Sinfonieorchester Berlin, Michail Jurowski conducting. Compositions of Prokofiev, Matthus and Rimski-Korsakow will be played. You can listen to the orchestra from different locations in the hall.

Ticket price: EUR 5.

Notes:

Tickets can be purchased on Friday morning at the Convention Center at the registration desk at the South Entrance.

No bus transfer from the Convention Center to the Konzerthaus is provided.

TECHNICAL TOUR 2
Philharmonic Hall Berlin
Saturday, May 8, 14:00 h–17:00 h

This famous “vineyard shaped” Philharmonic Hall designed by H. Scharoun opened in 1963. The hall has 2200 seats and is the home of the Berliner Philharmoniker. You can attend a rehearsal of the Berliner Symphoniker under Lior Shambadal with works of Berg and Bruckner. You will be able to listen to the orchestra from different locations in the hall.

Ticket price: EUR 15.

TECHNICAL TOUR 3
Teldex Studio
Sunday, May 9, 10:00 h–13:00 h

This fifty-year old recording hall, noted for its legendary acoustics, is now equipped with exciting state of the art surround technology. It is the rebirth of one of the largest European studio complexes. Presentations of noteworthy classical and pop surround productions will be made here.

Ticket price: EUR 15.

TECHNICAL TOUR 4
ICC (International Congress Center Berlin)
Sunday, May 9, 10:00 h–12:00 h

Visit the most modern congress center in Europe equipped with the latest venue technology for stage, lighting, sound reinforcement, and simultaneous translation. It can accommodate as many as 20,000 guests in two separate or one combined hall, having a full stage usable for both halls.

Ticket price: EUR 0.

TECHNICAL TOUR 5
Studio Complex Nalepastrasse (former GDR Radio) and Stagetec Company
Sunday, May 9, 13:00 h–17:00 h

This tour takes you to the outstanding Music and Drama Studio Complex of the former GDR (DDR) Radio. The complex was built in 1954 and houses four studios for different music genres and two drama studios. The studios are famous for their acoustics and acoustical design. Barenboim is a regular recording guest here.

A visit to Stagetec will follow. This company, which develops professional top-of-the-line audio equipment, is situated next to the studio-complex. Stagetec manufactures the "True Match 28 Bit-Converter," routing systems, and mixing desks for use in theaters and broadcasting.

Ticket price: EUR 15.

TECHNICAL TOUR 6
BMW Motorcycle Plant
Monday, May 10, 08:30 h–11:30 h

In 1923 the first BMW motorbike was built. Over the years these bikes became famous for performance and reliability as well as their sound. Since 1967 all these motorcycles are manufactured in Berlin. You will be shown the mechanical production facilities, body shop, paint shop, and final assembly area.

Ticket price: EUR 15.

TECHNICAL TOUR 7
Transmitting Station Nauen of the Deutsche Telekom
Monday, May 10, 09:30 h–12:00 h

The Nauen transmitter station dates back to 1906. The historic buildings for the old transmitters are situated in a wonderful area just outside Berlin. Nowadays huge AM transmitters, especially for short wave, are in use with movable antennas of impressive dimensions.

Ticket price: EUR 15.

TECHNICAL TOUR 8
Radio Berlin Brandenburg
Monday, May 10, 14:00 h–17:00 h

The new Broadcasting Center in Potsdam is one part of the Public Radio of Berlin and Brandenburg (rbb). Since 2001 it has its new digital home in the Babelsberg “media city” in Potsdam. With the new home came a completely new studio technology, based on Mandozzi desks and matrixes and D’accord operation software. In Potsdam, rbb produces three of its seven radio programs, called ANTENNE BRANDENBURG, FRITZ and RADIO EINS. You will see the studios, program sites, and technical rooms. This radio broadcasting facility is optimized to cope with the workflow imposed by a digital and fully networked environment.

Ticket price: EUR 15.

TECHNICAL TOUR 9
Babelsberg Film Sound Tour
Monday, May 10, 14:00 h–17:00 h

This tour will take you to the historical as well as the modern sites of the Film City Babelsberg. Depending on the production schedule you will see the “Urhaus”—the oldest film-studio from 1911; the “Tonkreuz”—a fascinating film sound studio from 1929; or the “Marlene Dietrich Halle”— which was built in 1925 for “Metropolis” and other productions. Also included in this tour are visits to mixing-studios (old and new), the dubbing stage, and editing suites.

Ticket price: EUR 15.

TECHNICAL TOUR 10
TV Tower Berlin Alexanderplatz
Tuesday, May 11, 10:00 h–13:00 h

The visit will take you to the 368-m high TV tower right in the heart of Berlin. You will see transmitters and aerials for 23 FM radio stations and 4 DVBT (Terrestrial Digital Video Broadcasting) blocks. In Berlin all analog TV transmitters have been switched off since the introduction of DVBT. The tower is operated by Deutsche Telekom. A special bonus is the magnificent view from the tower of the city of Berlin and its surroundings.

Ticket price: EUR 15.

TECHNICAL TOUR 11
Kammermusiksaal der Philharmonie
Wednesday, May 12, 15:30 h–18:30 h

Visit the Chamber Music Hall, “vineyard shaped” and built like the Philharmonic Hall but half its size. You will attend a rehearsal of the famous RIAS Kammerchor and the Akademie für Alte Musik Berlin, conducted by D. Reuss. Compositions of Schönberg, Haydn and Krenek will be performed. You can listen from different locations in the hall.

Ticket price: EUR 5.

Note: No bus transportation will be provided.

Historical Program

HISTORICAL CORNER	**Hall 4.1, Booth 5619**
Saturday, May 8	10:00 h–18:00 h
Sunday, May 9	10:00 h–18:00 h
Monday, May 10	10:00 h–18:00 h
Tuesday, May 11	10:00 h–17:00 h

Highlights

The History of Sound and Film in Berlin—Babelsberg from 1912 to the present day, will be a highlight of the Historical Room. Babelsberg was one of the first the centers of film-production using gramophone wax plates in playback mode. Later, Babelsberg produced one of the first sound films in Germany starring Marlene Dietrich in “Der blaue Engel” (“The Blue Angel”).

The historical ambience around the Historical Room at the AES Convention will therefore show Babelsberg with the huge old studios and equipment used for early sound recording.

Presentations/Papers

The Historical Room features special contributions related to the Babelsberg Studios, such as by Ingo Kock, Professor at the Hochschule für Film und Fernsehen in Babelsberg (“Sound and Films in Babelsberg, from the History to the Future”), and Ullrich Illing, Sound Engineer at Babelsberg Studios (“92 Years of Sound Movies from Babelsberg”).

The presentations include old films and old equipment and will therefore need some more time for discussions.

For a comprehensive listing see Historical Presentations on the following pages.

Historical Display

There will be in the Historical Room an exhibit of selected artifacts from the early days of audio technology. The motto is “Hands on Vintage Equipment.” Old microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

The historical ambience shows the history of Babelsberg, which, for over 90 years, has been a center for film production.

Historical Committee

The Historical Committee will hold its meeting in the Historical Room on Monday, May 10, 12:30 h–14:00 h.

Historical Tour

A historical tour is scheduled for Monday, May10, 14:00 h–17:00 h, to the Babelsberg film studios to see the many activities of this small Hollywood next to Berlin. The tour can be a supplement to the Babelsberg presentations in the Historical Room. The participants will be shown around in the old studios with the new installations and see for themselves the huge changes that occurred following the unification of Germany.

HISTORICAL PRESENTATION PROGRAM

Saturday, May 8 10:00 h–11:00 h

100TH BIRTHDAY ANNIVERSARY OF EDUARD SCHUELLER, "FATHER OF TAPE RECORDER TECHNOLOGY"

Presenter: **Gerhard Kuper**, Consulting Engineer, Wedel, Germany

January 13, 2004 marks the 100th anniversary of Eduard Schueller's birth. He passed away May 19, 1976. In his lifetime he applied for nearly 100 patents, several of which were fundamental to modern technologies, e.g., his "Ringkopf" (toroid-shaped tape head) of 1933, which is the basis for all magnetic storage technologies from tape recorders and video recorders up to computer hard disk drives. And his "Schrägspur" patent (helical scan recording), applied for in 1953—the basic principle for all video (and some audio/data) tape recorders all over the world.

The paper deals with Eduard Schueller's life, especially the part he played in the development of magnetic tape recording technology. Beginning with his diploma, continuing with his meeting with Fritz Pfleumer, the industrial development by AEG under the protection of Hermann Buecher, the cooperation with IG Farben, civil and military tape recorder development and up to the first low noise stereo tape recordings, made by the German RRG (Reichs Rundfunk Gesellschaft) in 1943/44. After the end of WWII and the acquisition of the AEG patents by the allies, the German Magnetophon development and production was continued; in the beginning at AEG in Hamburg, later at Telefunken in Wedel—but always managed by Eduard Schueller.

The presentation will center on the very successful first tape recorders for private use as well as for audio broadcasting, TV, and film studio applications and, later, for data recording.

The last chapter of the paper will deal with Eduard Schueller's activities after his retirement, when he contributed his know-how to a team developing a TV disk, similar to a phonograph record. For this particular activity he was awarded the German Order of Merit in 1972.

Saturday, May 8 12:30 h–13:30 h

HANDS ON VINTAGE EQUIPMENT

Demonstrated by: **Hans-Otto Hoffmann**, Bayerischer Rundfunk, Munich, Germany

Vintage microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

Saturday, May 8 14:00 h–15:00 h

SOUND AND FILMS—FROM THE PAST TO THE FUTURE

Presenter: **Ingo Kock**, Hochschule für Film und Fernsehen, Babelsberg, Germany

At the 50th anniversary of the establishment of the Film and Television College (Hochschule für Film und Fernsehen - HFF) Prof. Ingo Kock describes the 90-plus-years of history of film sound recording in Berlin, and especially in Babelsberg. The area we now call Studio Babelsberg was founded in 1912. Since that time, film recordings always took place together with sound. Film sound with gramophone play-back, optical sound, and digital sound were developed here. The development of the College for Film and Television is presented.

Saturday, May 8 15:30 h–16:30 h

ON THE ACOUSTICS OF OLD BERLIN STUDIOS FOR FILM AND RADIO

Presenter: **Ernst-Jo. Völker**, Institute for Acoustics and Building Physics, Oberursel, Germany

A certain acoustical environment was always necessary when an adequate sound quality had to reach the audience. That applied both for natural sound and for sound reproduction via loudspeakers using electrical or mechanical amplification. Long before microphones, amplifiers, and loudspeakers were developed and used, studios in the form of “Glasshouses” were built, e.g., in 1911 in the City of Babelsberg near Berlin, using bright sunlight. For sound recordings, huge horns connected to wax-plates or wax-cylinders were employed. Sound had to be absorbed by curtains, carpets, and much plush, which was already well-known since the first stereophonic transmission during the First Electrical Fair in Paris in 1879. Radio started in in Berlin with the Eugin Reiß carbon microphone in an almost over-damped studio on October 29, 1923. Some years later a “Haus des Rundfunks” was opened with many studios for different uses, including a concert hall. Film and radio went their own ways with multichannel reproduction or, for a long time, only with mono transmission. Some acoustical aspects of the first studios will be described.

Saturday, May 8 16:30 h–17:30 h

HANDS ON VINTAGE EQUIPMENT

Demonstrated by: **Hans-Otto Hoffmann**, Bayerischer Rundfunk, Munich, Germany

Vintage microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

Sunday, May 9 10:00 h–11:00 h

HISTORY OF AUDIO EFFECT UNITS

Presenter: **Udo Zölzer**, Helmut Schmidt University, Hamburg, Germany

The presentation will discuss old analog audio effect devices and their specific development over the past century, toward complete digital implementations. Audio effects are based on physical phenomena of sound production and transmission but are also created by musicians with their specific method of playing a musical instrument. The driving forces for different implementations and the use of different technologies will be explained with several sound examples.

Sunday, May 9 11:30 h–12:30 h

VINTAGE GUITAR VALVE AMPLIFIERS

Presenter: **Udo Zölzer**, Helmut Schmidt University, Hamburg, Germany

Guitar tube amplifiers developed in the 1950s and 1960s still enjoy high popularity. The original sound of different amplifiers will be presented on the basis of video and sound clips. The circuit designs and the development of these valve amplifiers will be discussed. A perspective towards complete digital implementations will be demonstrated.

Sunday, May 9 12:30 h–13:30 h

HANDS ON VINTAGE EQUIPMENT

Demonstrated by: **Hans-Otto Hoffmann**, Bayerischer Rundfunk, Munich, Germany

Vintage microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

Sunday, May 9 14:00 h–15:00 h

92 YEARS SOUND MOVIES FROM BABELSBERG

Presenter: **Ulrich Illing**, Studio Babelsberg, Babelsberg, Germany

The Babelsberg Studios have been well-known since 1912, when the first “Glasshouse” was built in order to work under optimal daylight conditions. First productions used hand-cranked cameras and gramophones with horns for sound reproduction. The gramophones served as a play-back system for the actors.

In 1926 huge studios were built in Babelsberg, where among others the silent film “Metropolis” was produced. In these days there was much resistance against sound in films. Therefore, the Triergon sound film shooting of “1925” on the Ufa site was only of little interest. However, impressed by the boom of sound movies in the USA, Ufa built a new complex with halls using room-in-room-construction for higher sound proofing and better acoustical properties. One of the first light/sound films produced here was the very famous “Der blaue Engel” (“The Blue Angel”).

In the following years film production not only increased in Babelsberg, but many important technical developments for film sound recording were introduced. At the end of WWII the

Babelsberg film site was in ruins. East-German moviemakers and technicians created, with much inventiveness, the foundation for the production of nearly 700 DEFA feature films up to 1992. Today the Babelsberg studios are an important film and television center again, both for filming/recording and postproduction.

Sunday, May 9 15:00 h–16:00 h

HANDS ON VINTAGE EQUIPMENT

Demonstrated by: **Klaus Dieter**, Bayerischer Rundfunk, Munich, Germany

Vintage microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

Sunday, May 9 16:00 h–17:00 h

FIRST LOUDSPEAKERS—SOME HISTORICAL ASPECTS

Presenter: **Hans-Otto Hoffmann**, Bayerischer Rundfunk, Munich, Germany

Loudspeakers can be seen as devices that radiate loud sounds. Speech and music were included from the beginning of sound reproduction. In 1881 the first stereo reproduction was provided during the Electrotechnical Worlds Fair in Paris when a transmission took place from the Paris Opera to a demonstration room near the Eiffel Tower. For listening, headphones were installed with left and right information separately for each ear. The door was now open to electromagnetic loudspeaker speaker systems. Meanwhile legendary phonographs invented by Edison and others were used, sometimes in parallel, to reach a larger audience in movie theaters. When Lieben invented the amplification tube, an important step was achieved toward larger and more powerful loudspeakers following the same electromagnetic transmission method. With the beginning of radio in the early 1920s, the first monitoring speakers appeared for controlling the recorded sound simultaneously with a wireless transmission.

The paper will describe some important inventions and developments, which led to our present high standards in monitoring loudspeakers.

Sunday, May 9 17:00 h–18:00 h

GOLDEN MICROPHONES IN THE OLD DAYS OF RECORDING

Presenter: **Norbert Pawera**, Com. AKG, Munich, Germany

Neither the contact microphone of Philip Reis in 1861 nor the carbon microphone of Graham Bell became real recording microphones for transmitting speech or music. When Eugen Reis, in the 1920s, proposed his carbon microphone it became very famous and could be called a golden microphone. A real break through came in 1928 with the first condenser microphone from Neumann. The quality was much better than that of the recording media such as wax plates. Although condenser microphones existed, the Eugen Reis carbon microphone was still in use in the 1930s. Then their time was over. The new high quality “magnetophones” required much better microphones. The standard recording technique used one microphone in front of the orchestra and later a stereophonic microphone was positioned on the same spot. In 1944 Helmut Krüger made tape recordings with condenser microphones suspended above the left and right side of the orchestra to produce a stereophonic sound image. The tapes were captured by the Russians, but were, fortunately, retrieved and could be used for making a CD of the recordings in 1983.

Other golden microphones followed such as high-directivity microphones and wireless microphones. Many old microphones are still very well known today. They will be shown and explained.

Monday, May 10 10:00 h–11:00 h

HISTORY OF THE TONMEISTER TECHNOLOGY IN RUSSIA

Presenter: **Pavel Ignatov**, Student member of Russian Section

The history of sound recording in Russia dates to the end of the 19th century. Due to this fact it is possible to find some wax disks with voices of great Russian writers such as Tolstoy and Chekhov. The creation of the first sound recording studios began in the 1920s and 1930s. Although the technical facilities that were used seemed to be quite primitive, the work of an outstanding tonmeister such as M. G. Khustov, A. B. Grossman, and D. G. Gakhlin made it possible to create wonderful recordings of classical music and live concerts. The main feature of the years between 1950 and 1980 is the great development of the TV-, radio-, and recording studios (292 large television centers and radio studios had been built by the 1980s). Because of the work of the tonmeisters, the masterpieces of Russian and world musical culture were preserved. Today the new digital technologies and surround sound systems are used in tonmeister practice. Masters such as S. G. Shugal, V. V. Vinogradov, P. K. Khondrashin, V. G. Dinov, and many others create new methods of digital sound recording.

Monday, May 10 11:00 h–12:00 h

ACOUSTIC RECONSTRUCTION OF BUILDINGS IN THE ANCIENT CITY OF OLYMPIA

Presenter: **John Mourjopoulos**, University of Patras, Greece

Two famous buildings, which are now in ruins, in the ancient Greek city of Olympia (birthplace of the Olympic Games) are the Temple of Zeus and the Echo Hall. These are reconstructed as 3-D computer models. Their acoustic properties are analyzed via computer-aided prediction and auralization, so that detailed and in-depth conclusions for their acoustic performance are derived and presented, together with audio demonstrations. Such a methodology introduces a form of acoustical archaeology, since it presents novel findings for these ritual buildings’ acoustic behavior, especially with respect to the modes of speech communication and general functionality.

Monday, May 10 14:30 h–15:30 h

WAS THE CD ALREADY THERE? OLDER IDEAS REVIEWED

Presenter: **Werner Hinz**, Retired Chief Engineer of WDR, Bergisch Gladbach, Germany

Sometimes inventions are made twice because the subject is very timely. Several people come up with very similar ideas. However, some inventions are not followed by practical applications. Instead, other scientists achieve the breakthrough and the financial success. This description may well apply to the well-known compact disc, the CD.

At the beginning of the 1980s the Philips Company introduced the complete system. By 1983 the production of the CD exploded. The so-called black disk became unimportant. Meanwhile the CD is already an old product and is replaced by DVD or mini disk.

Recently the work of Jim Russel, an American physicist, came to light. He already had invented CD technology around 1965, long before Philips in the years between 1980 and 1983. Russel invented the optical track of digital signals on thin disks. The bits were in the micrometer range. Optical read-out was part of the system.

At that time Russel worked for the Batelle Institute. Batelle had no interest in this optical CD. That is why the revolutionary invention was not introduced at the time.

In his paper Werner Hinz will describe the work of Russel and will include the first “Optophone” which already was invented in 1931.

Monday, May 10 15:30 h–16:30 h

THE ACOUSTICS OF ANCIENT GREEK ODEA

Presenter: **Christos Goussios**, Aristotle University of Thessaloniki, Greece

Apart from the world famous ancient Greek theaters, whose acoustics often attracted engineers, smaller closed amphitheatric halls—called Odea (plural of the Greek word odeion)—had been constructed and used through the Greek and Roman period. The acoustical characteristics for most of them and information concerning their location, use, history, and architectural elements are presented. An effort for the modeling and estimation of their acoustics was made. Results of measurements that had been also carried out are discussed.

Monday, May 10 16:30 h–17:30 h

HANDS ON VINTAGE EQUIPMENT

Demonstrated by: **Klaus Dieter**, Bayerischer Rundfunk, Munich, Germany

Vintage microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

Tuesday, May 11 10:00 h–11:00 h

HANDS ON VINTAGE EQUIPMENT

Demonstrated by: **Hans-Otto Hoffmann**, Bayerischer Rundfunk, Munich, Germany

Vintage microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

Tuesday, May 11 12:00 h-13:00 h

HANDS ON VINTAGE EQUIPMENT

Demonstrated by: **Klaus Dieter**, Bayerischer Rundfunk, Munich, Germany

Vintage microphones and loudspeakers will be demonstrated and used together with direct cutting on an old record cutting machine.

Social Tours

Saturday, May 8 14:00 h–18:00 h

ST1 POTSDAM - SANSSOUCI

This tour will take you to the historic sites of the Prussian Kings—especially Frederick the Great. See the castles and gardens of Sanssouci including a visit to Frederick's castle. See the Cecilienhof Castle, where Truman, Churchill, and Stalin negotiated and signed the Agreement of Potsdam in July/ August 1945. And see the City of Potsdam with the Dutch Quarter and the Russian Colony "Alexandrowka." *Ticket price: EUR 35.*

Monday, May 10 14:00 h–18:00 h

ST2 INNER CITY TOUR AND BOAT TRIP

On this tour you will see the heart of Berlin with "Unter den Linden" and its historical buildings, the "Brandenburger Tor," the new Government Quarter including the "Reichstag." A 1-hour boat trip and a visit to one of Berlins most significant museums, the Pergamon Museum, will be included. *Ticket price: EUR 35.*

Wednesday, May 12 08:00 h-19:00 h

ST3 DAY TRIP TO MEISSEN (FAMOUS MEISSNER CHINA) AND DRESDEN

Visit the world-famous Meissen porcelain factory. The white, European hard porcelain manufacturing process was developed here in 1707–1708. Then go on to Dresden to see the old city of Saxon emperor times with renaissance and baroque buildings. Despite vast destruction during the last war, the city has preserved fascinating ensembles. The most famous symbol of reconstruction is the "Dresdner Frauenkirche," which today dominates the city center. By the way, it was trumpet soloist Ludwig Güttler who initiated the reconstruction. *Ticket price:EUR 60*.

Special Events

AES MULTICHANNEL SYMPOSIUM
THE EFFECT OF MULTICHANNEL ON RADIO OPERATION
Friday, May 7, 10:00 h–18:00 h
Room 7.1a-1
Preconvention Special Event; additional fee applies

Chairs: **Jürgen Marchlewitz**, WDR, Köln, Germany
Martin Wöhr, BR, Munich, Germany

Topics and presenters:

EFFECTS ON PRODUCTION, ACTUAL SITUATIONS AND EFFICIENCY
Bosse Ternström, Swedish Radio
Jean-Marie Geijsen, Polyhymnia, The Netherlands
Kimio Hamasaki, NHK, Japan

EFFECTS ON ARCHIVING OF MULTICHANNEL PRODUCTIONS
Udo Appel, Bayerischer Rundfunk, Germany
Yvonne Graf, IBM, Germany

EFFECTS ON BROADCASTING DISTRIBUTION PROCESSES
Heinz-Peter Reykers, Westdeutscher Rundfunk
Gerhard Möller, D.A.V.I.D., Germany
Francis Rumsey, University of Surrey, UK
Günther Theile, IRT, Germany

FUTURE ASPECTS ON THE EBU-B/MCAT WORKING GROUP
Gerhard Stoll, IRT, Germany

WHERE ARE THE SOLUTIONS FOR OPTIMIZING WORKFLOW AND COSTS FOR MULTICHANNEL AUDIO IN BROADCAST?—A panel discussion between broadcasters, developers, and industry leaders.
Ernst Dohlus, Bayerischer Rundfunk, Germany
Yvonne Graf, IBM, Germany
Kimio Hamasaki, NHK, Japan
Rüdiger Malfeld, Westdeutscher Rundfunk, Germany
Gerhard Möller, D.A.V.I.D., Germany
Bosse Ternström, Swedish Radio, Sweden
Günther Theile, IRT, Germany

THREE LOCAL STUDIO TOURS SPONSORED BY RADIO BERLIN BRANDENBURG (RBB)

1. Seminar on Microphone Recording in Practice at Studio 3 of RBB (Radio Berlin Brandenburg) in “Haus des Rundfunks”
This event is of interest to sound engineers. It provides an opportunity to listen to live recordings and sound checks with different microphones. Various types of microphones from different companies will be placed in front of musicians. The microphones can be switched during the recording session. The positions of the microphones can be changed and rearranged by the Tonmeister in charge. This continous recording allows the visitors to be not only passive listeners, but to select the best possible recording.

There will be an additional program of a 5.1 recording with the Count Basie Orchestra. This recording was made with different types of microphones for interesting comparisons. The recording exists as an SACD. The Tonmeister of this recording, Mike Pappas, will be present

Saturday, 8 May	15:00 – 16:00 h (SACD)
	16:30 – 17:30 h (SACD)
Sunday, 9 May	11:00 – 12:30 h (Live recording)
	13:30 – 15:00 h (Live recording)
	16:00 – 17:30 h (Live recording)
Monday, 10 May	11:00 – 12:30 h (Live recording)
	13:30 – 15:00 h (Live recording)
	16:00 – 17:30 h (Live recording)
Tuesday, 11 May	11:00 – 12:00 h (SACD)

The shuttle bus (“RBB”) will depart from the South Entrance 15 minutes before each session.

2. Studio Presentation of Radio Plays and Music Recordings in Stereo and 5.1 at Studio 2 of RBB (Radio Berlin Brandenburg) in “Haus des Rundfunks”
In this presentation, two radio plays (“Das Herz der Tänzer” and “Piratinnen,” direction by Iris Disse, recording by Peter Avar) will be demonstrated. The location shooting (music and word) is done with microphone positions similar to the IRT-cross. With studio postrecording and postproduction all of the advantages of radio dramas in 5.1 can be shown. The variety of possibilities of music recordings is most interesting since no acoustic/aesthetical targets exist for radio plays. Additionally, music recordings are presented in 5.1, which were made at the local broadcaster RBB, 8th Symphony of Gustav Mahler with the Berlin Philharmonic, and a concert with the Belgian group “Musique à Neuf.” This last concert was recorded at the opening of the *Prix Europa 2003* in the large studio hall of RBB, which was Europe’s first satellite radio transmission (DVB-S) in 5.1 DTS, live for Swedish Broadcasting Corporation.

Saturday, 8 May	15:00 – 17:00 h
Sunday, 9 May	11:00 – 13:00 h
	15:00 – 17:00 h
Monday, 10 May	11:00 – 13:00 h
	15:00 – 17:00 h
Tuesday, 11 May	11:00 – 13:00 h

The shuttle bus (“RBB”) will depart from the South Entrance 15 minutes before each session.

3. “In Touch” Daily Studio Tour to Blackbird Studios

We will show you one of the most modern studios in Germany and present nonlinear production for TV/film and postproduction using a high technical standard. Take part in one of the two daily guided visits to the Blackbird Studios. Discussion will follow.

Free shuttle bus “Blackbird Studios” will be available at the South Entrance. In less than 10 minutes it will bring you into a real audio production environment with the newest standard. After 50 minutes, the return shuttle bus will bring you back to the AES Convention where you may continue your visit.

Departure: Saturday, 8 May, Sunday, 9 May, Monday, 10 May at 11:30 h and 14:00 h.

These RBB Special Events are free of charge

MIXER PARTY
Saturday, May 8, 18:00 h-19:00 h
South Entrance Atrium

Social gathering with live music in the new South Entrance Atrium of the Messe Berlin, where you meet all your friends and make new ones.

Cash bar; no entrance fee.

SING-ALONG MOZART—HAUS DES RUNDFUNKS
Sunday, May 9, 10:30 h-17:00 h

The Mozart Requiem KV 626 will be performed by the Rundfunk Sinfonieorchester Berlin, the Rundfunkchor Berlin and its soloists. Take part in a sophisticated sing-along concert with two rehearsals on the same day.

The Haus des Rundfunks is a radio studio built between 1929-31 and is situated next to the fairground of the Messe Berlin.

Note: Advance booking is required. After receipt of your booking you will be sent the score, enabling you to practice before you join the actual rehearsals and sing-along.

Ticket price: EUR 10.

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

Sunday, May 9, 18:00 h–19:30 h
Room 7.1a-1

Lecturer: **Kees Schouhamer Immink**

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. Mr. Heyser was also an AES governor and AES Silver Medal recipient.

The Richard C. Heyser distinguished lecturer for the 116th AES Convention is Kees Schouhamer Immink, president and founder of Turing Machines Inc. During his career, he has contributed to the digital audio, data, and video revolution by developing coding technologies for essentially all consumer optical and magnetic recording formats such as the compact disc, minidisc, DCC, DVD, and BluRay disc.

For his many pioneering contributions to the digital revolution he won wide recognition among his peers. Among the many honors received are an Emmy Award, IEEE Edison Medal, AES Gold and Silver Medal, IEEE Ibuka Consumer Electronics Award, SMPTE Poniatoff Gold Medal, and IEE J.J. Thomson Medal. Immink was named a Knight in the Order of Orange Nassau, elected into the Royal Netherlands Academy of Sciences (KNAW), inducted into the Consumer Electronics Hall of Fame, and named a fellow of the IEEE, AES, SMPTE, and IEE. He served the AES as a governor and VP since 1996, and was president in 2003.

Imminks's lecture is entitled, "From Analog to Digital."

In 1964, the inventor of the pulse code modulation (= digital data transmission) system, Alec H. Reeves, wrote in an IEE article: "Twenty-five years after its invention (in 1937), it can be said that pulse code modulation has little past as yet; the real interest is in its future. This future depends a great deal on how well, or how badly, its main planning problems are tackled during the next decades or so. There is little or no agreed view of the technical and more general points involved in this planning."

Now, another 40 years later, the "planning problems" mentioned were tackled well, and humanity has witnessed the digital audio and video revolution. Digital revolutionary fruit, such as the compact disc, MiniDisc, DAT, DVR, DVD, and so on, is less than 20 years young. New products are now on the market showing features that did not exist in the early 1980s. The DVD, introduced in 1996, has become a high-tech commodity product by now. Solid-state storage brought perfect portable audio players, where crashes resulting from jogging are absent and power consumption is low.

In retrospect one may say that the digital audio revolution is an immense success as everybody is satisfied with the outcome. With reluctance, the speaker acknowledges the fact that there are a few nostalgic exceptions, who cry out in longing to return to the fleshpots of the radio hiss, the warm distortion of the electron tubes, the scratchy sound of the gramophone, and, not to forget, the greener grass.

At this junction, almost seventy years after the invention of pulse code modulation, the audio world is in an evolutionary consolidation phase again, and it seems to be a good idea to appraise how well we fared so far. Immink will address some historical notes on pulse code modulation and will show that the digital revolution rests on the scientific fundamentals laid by researchers at Bell Labs—Nyquist, Shannon, Hamming, to mention just a few giants—in a period of time when the term “research project” was an oxymoron.

Kees Immink will use his glass ball to see what the future might offer.

ORGAN CONCERT BY GRAHAM BLYTH
Sunday, May 9, 20:30 h- 22:15 h
St. Matthias Cathedral

Graham Blyth will perform an organ recital at St. Matthias Cathedral, famous for its excellent acoustics. This will give you an opportunity to relax after the working hours at the convention. Mr. Blyth, known to many for his past AES organ recitals, will play several pieces from his treasure trove. The concert will feature J.S.Bach's "Passacaglia and Fugue in C minor," Guilmant's "1st Sonata," and three movements from Widor's "5th Symphony."

Graham Blyth 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 the Trinity College of Music. In the late 1980s he renewed his studies with Sulemita Aronowsky for piano and with Robert Munns for organ.

Blyth made his international debut with an organ recital at St. Thomas Church, New York, in 1993, and since then has played in San Francisco (Grace Cathedral), Los Angeles, Amsterdam, Copenhagen, Munich, and Paris (Madeleine Church). He gives numerous concerts each year, principally as an organist and a pianist, but also as a conductor and a harpsichord player.

Blyth is founder and technical director of Soundcraft. He divides his time between his main career as a designer of professional audio equipment and organ-related activities. He has lived in Wantage, Oxfordshire, U.K., since 1984, where he is currently artistic director of the Wantage Chamber Concerts and director of the Wantage Festival of Arts. He is also founder and conductor of the Challow Chamber Singers & Players. He is involved with Musicom Ltd., a British company at the leading edge of the pipe organ control system and digital pipe synthesis design. He also acts as tonal consultant to the Saville Organ Company and is recognized as one of the leading voicers of digital pipe modeling systems.

The bus will leave the Convention Center at 19:45 h for those who want to enjoy this performance.

BANQUET
Monday, May 10, 20:00 h- 23:00 h
Deutsche Telecom Telephone Exchange Room

An informal Convention Banquet will take place in the old Telephone Exchange Room in the Berlin headquarters of Deutsche Telecom, which dates from 1880. Don't miss this excellent opportunity to meet with pro-audio friends in a relaxed atmosphere with music, good food, and wine.

Ticket price:	*AES members: EUR 60.*
	Nonmembers: EUR 75.

Student Activities

Student activities are open to all attendees with a full program badge. All attendees with an interest in audio education are welcome to attend. The main program contains a series of 15 entry-level **Tutorial Seminars**, which are of particular interest to students. These are described on pages 153 to 168.

OPENING AND STUDENT DELEGATE ASSEMBLY MEETING – 1
Saturday, May 8, 15:00 h–17:00 h
Room 7.1c-1

Chair: **Natalia Teplova**

Vice Chair: **Martin Berggren**

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. In this session, which will be chaired by the SDA chair and vice-chair elected at last year's European convention, the activities of the SDA and the student sections will be discussed and the student program for the convention is presented. Students and student sections will be given the opportunity to introduce themselves and their activities, in order to stimulate international contacts.

During this session nominations will be made for the new Europe/International Regions SDA vice chair. The AES Regional Vice Presidents of the European and International regions can each nominate a candidate from their region. Election results and Recording Competition and Poster Awards will be given at the Student Delegate Assembly Meeting – 2 on Tuesday, May 11, at 12:00 h.

EDUCATION FAIR
Sunday, May 9, 11:00 h–13:00 h
Corridor 7.1a

The Education Fair is the perfect opportunity for representatives of educational institutions to present themselves to potential new students and to share experiences with people from other schools. In this "tabletop session," information on each school's respective program will be made available through the display of literature and informal conversations with representatives.

For each school a table and a poster board are made available for displaying promotional material. There is no charge for schools to participate. Admission is free and open to everyone.

STUDENT POSTERS
Sunday, May 9, 11:00 h–13:00 h
Corridor 7.1a

The event will display the scholarly/research works from AES student members in the form of a poster presentation. Unlike previous years, the student poster session will now be held in the same space and at the same time as the Education Fair. This will ensure that the posters will reach a large audience, thus providing a great opportunity to display and discuss the presented work with professionals, educators, and other students.

LIVE RECORDING OF ORGAN CONCERT
Sunday, May 9, 16:30 h–22:00 h
St. Matthias Cathedral

A limited number of students can participate in a live recording of the Organ Concert by Graham Blyth. Details will be communicated during the Student Delegate Assembly Meeting – 1 on Saturday, May 8.

RECORDING COMPETITION – PART 1
Monday, May 10, 09:00 h–12:00 h
Room 7.1c-1

09:00 h–09:45 h Classical
10:00 h–10:45 h Pop/Rock
11:00 h–11:45 h Jazz/Folk

Finalists selected by an elite panel of judges will give brief descriptions and play recordings in the different categories. The panel of judges will comment on recordings. One submission per category per school/student section.

Meritorious awards will be presented at the closing Student Delegate Assembly Meeting on Tuesday.

RECORDING COMPETITION – PART 2
Monday, May 10, 13:00 h–16:00 h
Room 7.1c-1

13:00 h–13:45 h Movie Sound
14:00 h–14:45 h Classical Surround
15:00 h–15:45 h Nonclassical Surround

Finalists selected by an elite panel of judges will give brief descriptions and play recordings in the different categories. The panel of judges will comment on recordings. One submission per category per school/student section.

Meritorious awards will be presented at the closing Student Delegate Assembly Meeting on Tuesday.

STUDENT PARTY
Monday, May 10, 20:00 h–
UdK Berlin

A special student party will be organized at the UdK Berlin. Further information will be given during the Student Delegate Assembly Meeting – 1 on Saturday, May 8.

EDUCATION FORUM
Tuesday, May 11, 10:00 h–11:30 h
Room 7.1c-1

This event is a meeting of the AES Education Committee, teachers, authors, students, and members interested in the issues of primary and continuing education of the audio industry. It is an opportunity to discuss the programs of the Education Committee and to provide input for future projects of this committee.

STUDENT DELEGATE ASSEMBLY MEETING – 2
Tuesday, May 11, 12:00 h–13:30 h
Room 7.1c-1

At this meeting the SDA will elect new officers. One vote will be cast by the designated representative from each recognized AES student section in the European and International Regions. Judges' comments and awards will be presented for the Recording Competitions and the Student Poster Session. Plans for future student activities at local, regional, and international levels will be summarized.

AES Meetings

REGIONS AND SECTIONS MEETING

A meeting of the officers of all AES Sections will take place on Sunday, May 9, from 09:00 h–11:00 h, in Room 7.1c-1.

HISTORICAL COMMITTEE MEETING

A meeting of the Historical Committee will take place on Monday, May 10 from 12:30 h–14:00 h, in Hall 4.1, Booth 5619.

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.

SATURDAY, MAY 8

12:30 h	Perception and Subjective Evaluation of Audio (TC Room 1, Hall 7.1b)
12:30 h	Audio for Games (TC Room 2, Hall 7.1b)
14:00 h-	Audio for Telecommunications (TC Room 1, Hall 7.1b)
16:00 h	Archiving, Restoration and Digital Libraries (TC Room 1, Hall 7.1b)
16:00 h	High Resolution Audio (TC Room 2, Hall 7.1b)

SUNDAY, MAY 9

09:00 h	Automotive Audio (TC Room 1, Hall 7.1b)
09:00 h	Semantic Audio Analysis (TC Room 2, Hall 7.1b)
12:00 h	Loudspeakers and Headphones (TC Room 1, Hall 7.1b)
12:00 h	Acoustics and Sound Reinforcement (TC Room 2, Hall 7.1b)
17:00 h	Audio Recording and Storage Systems (TC Room 1, Hall 7.1b)

MONDAY, MAY 10

10:00 h	Network Audio Systems (TC Room 1, Hall 7.1b)
11:30 h	Multichannel and Binaural Audio Technologies (TC Room 1, Hall 7.1b)
14:00 h	Studio Practices and Production (TC Room 1, Hall 7.1b)
17:00 h	Signal Processing (TC Room 1, Hall 7.1b)
17:00 h	Transmission and Broadcasting (TC Room 2, Hall 7.1b)
18:00 h	Coding of Audio Signals (TC Room 1, Hall 7.1b)
18:00 h	Microphones and Applications (TC Room 2, Hall 7.1b)

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

Sunday, May 9 18:00 h-19:30h Room 7.1a-1

Lecturer: **Kees Schouhamer Immink**

For complete details see page 26.

Standards Committee Meetings

STANDARDS COMMITTEE MEETINGS

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

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

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

STANDARDS MEETINGS

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

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

The schedule of meetings follows.

Meetings, including plenary meetings of the Standards Committee, are scheduled to take place in Standards Room 1 (Room Z8) or Room 2 (Room Z9). The schedule is subject to changes and additions. Daily updates may also be obtained in the Standards Facilities Room (Room Z7).

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

Contact: standards@aes.org.

THURSDAY, MAY 6

13:30 h SC-02-02 Digital Input/Output Interfacing (Room 1)

17:00 h SC-02-05 Synchronization (Room 1)

FRIDAY, MAY 7

12:00 h SC-06-04 Internet Audio Delivery System (Room 1)

16:00 h SC-05-05 Grounding and EMC Practices (Room 1)

SATURDAY, MAY 8

09:00 h	SC-03-06 Digital Library and Archive Systems (Room 1)
09:00 h	SC-05-02 Audio Connectors (Room 2)
11:00 h	SC-06-06 Audio Metadata (Room 1)
11:00 h	SC-03-04 Storage and Handling of Media (Room 2)
14:00 h	SC-06-02 Audio Applications Using the High Performance Serial Bus (IEEE 1394) (Room 1)
17:00 h	SC-03-02 Transfer Technologies (Room 1)

SUNDAY, MAY 9

10:30 h	SC-04-07 Listening Tests (Room 2)
12:00 h	SC-06-01 Audio File Transfer and Exchange (Room 1)
13:30 h	SC-02-01 Digital Audio Measurement Techniques (Room 2)
14:00 h	SC-04-04 Microphone Measurement and Characterization (Room 1)
16:00 h	SC-04-03 Loudspeaker Modeling and Measurement (Room 1)

MONDAY, MAY 10

09:00 h	SC-04-01 Acoustics and Sound Source Modeling (Room 2)
10:30 h	SC-05 Subcommittee on Interconnections (Room 1)
11:00 h	SC-03-01 Analog Recording (Room 1)
12:00 h	SC-04 Subcommittee on Acoustics (Room 1)
13:00 h	SC-03-12 Forensic Audio (Room 2)
13:30 h	SC-06 Subcommittee on Network and File Transfer of Audio (Room 1)
15:00 h	SC-02 Subcommittee on Digital Audio (Room 1)
16:30 h	SC-03 Subcommittee on the Preservation and Restoration of Audio Recording (Room 1)

TUESDAY, MAY 11

11:30 h	AESSC Plenary (Room 1)

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

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

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

The information contained in this book is subject to change.

PAPERS

Technical Paper Sessions

Session A	**Audio Networking**	Saturday, May 8 09:30 h–11:30 h Room 7.1b-1
Session B	**Spatial Perception and Processing, Part 1**	Saturday, May 8 09:30 h–11:00 h Room 7.1b-2
Session Z1	**Posters: Automotive Audio and Instrumentation and Measurement**	Saturday, May 8 09:30 h–11:00 h Corridor 7.1b
Session C	**Audio Archiving, Storage, and Restoration; Content Management**	Saturday, May 8 13:00 h–16:00 h Room 7.1b-1
Session D	**Spatial Perception and Processing, Part 2**	Saturday, May 8 13:30 h–16:00 h Room 7.1b-1
Session Z2	**Posters: Audio in Computers and Audio Video Systems**	Saturday, May 8 14:00 h–15:30 h Corridor 7.1b
Session E	**Analysis and Synthesis of Sound, Part 1**	Saturday, May 8 16:00 h–18:00 h Room 7.1b-1
Session Z3	**Posters: Signal Processing and Audio in Broadcasting**	Sunday, May 9 09:30 h–11:00 h Corridor 7.1b
Session F	**Analysis and Synthesis of Sound, Part 2**	Sunday, May 9 10:00 h–13:00 h Room 7.1b-1
Session G	**Low Bit-Rate Audio Coding, Part 1**	Sunday, May 9 10:00 h–12:30 h Room7.1b-2
Session H	**Multichannel Sound**	Sunday, May 9 13:00 h–17:00 h Room 7.1b-1
Session I	**Low Bit-Rate Audio Coding, Part 2**	Sunday, May 9 13:00 h–15:30 h Room 7.1b-2
Session Z4	**Posters: Spatial Perception and Processing and Analysis and Synthesis of Sound**	Sunday, May 9 13:00 h–14:30 h Corridor 7.1b
Session J	**Spatial Audio Coding**	Sunday, May 9 15:30 h–18:00 h Room 7.1b-2
Session Z5	**Posters: Psychoacoustics, Perception, and Listening Tests**	Sunday, May 9 16:00 h–17:50 h Corridor 7.1b
Session K	**Signal Processing, Part 1**	Monday, May 10 09:00 h–12:30 h Room7.1b-1

Session L	**Loudspeakers, Part 1**	Monday, May 10 09:00 h–12:00 h Room 7.1b-2
Session Z6	**Posters: Room and Architectural Acoustics and Musical Acoustics**	Monday, May 10 09:30 h–11:00 h Corridor 7.1b
Session M	**Loudspeakers, Part 2**	Monday, May 10 12:30 h–15:30 h Room 7.1b-2
Session Z7	**Posters: Multichannel Sound and Wave Field Synthesis**	Monday, May 10 12:30 h–14:00 h Corridor 7.1b
Session N	**Signal Processing, Part 2**	Monday, May 10 13:30 h–16:30 h Room 7.1b-1
Session O	**Microphones**	Monday, May 10 15:30 h–18:00 h Room 7.1b-2
Session Z8	**Posters: Audio Recording and Reproduction and Archiving and Content Management**	Monday, May 10 15:30 h–17:00 h Corridor 7.1b
Session P	**Psychoacoustics, Perception, and Listening Tests**	Tuesday, May 11 09:30 h–12:30 h Room 7.1b-1
Session Q	**Audio Recording and Reproduction and High-Resolution Audio**	Tuesday, May 11 09:30 h–12:00 h Room 7.1b-2
Session Z9	**Loudspeakers and Microphones**	Tuesday, May 11 09:30 h–11:00 h Corridor 7.1b
Session R	**Instrumentation and Measurement**	Tuesday, May 11 13:00 h–15:30 h Room 7.1b-1
Session S	**Room and Architectural Acoustics and Sound Reinforcement**	Tuesday, May 11 13:00 h–16:00 h Room7.1b-2
Session Z10	**Posters: Low Bit-Rate Coding**	Tuesday, May 11 13:00 h–14:30 h Corridor 7.1b

Session A **Saturday, May 8** **09:30 h–11:30 h**
Room 7.1b-1

AUDIO NETWORKING

Chair: **Thomas Sporer**, Fraunhofer IIS AEMT, Ilmenau, Germany

09:30 h

A-1 An XML-Based Approach to the Generation and Testing of mLAN Sound Installation Configurations
—*Jun-ichi Fujimori*[1], *Rob Laubscher*[2], *Richard Foss*[3]
[1]Yamaha Corporation, Hamamatsu, Japan
[2]Otic Systems, Cape Town, South Africa
[3]Rhodes University, Grahamstown, South Africa

An application, called the mLAN Installation Designer, has been developed that enables the user to graphically design and validate an mLAN sound installation. This application is built upon a model of mLAN systems that is defined by an Extensible Markup Language (XML) schema, ensuring cross platform portability and future scalability. The XML schema provides sufficient flexibility to form the basis for a standard effort to describe the configuration of IEEE 1394-based sound installation environments. The output from the mLAN Installation Designer application file is an XML document, consistent with the defined schema, which allows a configuration tool to configure the mLAN devices for automatic operation during deployment of the system.
Convention Paper 5994

10:00 h

A-2 Plug and Play? An Investigation into Problems and Solutions of Digital Audio Networks—*Christian Frandsen*[1], *Morten Lave*[2]
[1]TC Electronic A/S, Risskov. Denmark
[2]TC Applied Technologies Ltd., Markham, Ontario, Canada

It is a challenge to predict fault tolerance of the total system using point-to-point digital audio interfaces to build complex routing structures. In real life, digital interfacing is therefore still considered less robust than analog. This paper provides a systematic investigation of factors determining reliability in a number of widely used professional audio and synchronization interfaces such as AES3, SPDIF, ADAT, TDIF, and World Clock. Electrical characteristics, phase-offset and tolerance to offset, intrinsic jitter and tolerance to jitter, and sample rate precision have been tested. Additionally, compliancy with standards has been

evaluated. Finally, a discussion of how these problems can be dealt with followed by specific thoughts about the next generation of interfaces will be presented with examples.
Convention Paper 5995

10:30 h

A-3 Delivering High-Quality Audio over WLANs—*Andreas Floros, Theodore Karoubalis,* ATMEL-Hellas S.A., Multimedia & Communications Group, Patras, Greece

Based on the current version of the forthcoming IEEE802.11e standard, the paper examines the wireless, real-time transmission of high-quality audio streams. The required procedures that provide the necessary Quality of Service (QoS) support are presented and optimized for digital audio applications, and their effect on the achieved playback quality is estimated through a sequence of tests in terms of the achieved wireless bit rate and the end-to-end packet delay. Both two-channel and multichannel audio playback setups are considered in order to accurately simulate typical stereo and home theater wireless applications.
Convention Paper 5996

11:00 h

A-4 Advances in Sinusoidal Analysis/Synthesis-Based Error Concealment in Audio Networking—*Sang-Uk Ryu, Kenneth Rose*, University of California, Santa Barbara, CA, USA

This paper investigates error concealment based on sinusoidal analysis and synthesis. Major shortcomings are identified with focus on the extraction of sinusoidal frequency evolution and sinusoid matching. A new approach to frame loss concealment is proposed. It involves parallel Fourier transformation with long and short windows to accurately extract model parameters and is complemented with two sinusoid matching techniques—sinusoidal pair alignment by dynamic programming and harmonics-based matching. Moreover, due to the incompatibility of sinusoidal representation with broadband, noise-like signals, an alternative "sinusoids plus residual" model is incorporated. The new algorithm was applied to CD-quality audio of various genres and was demonstrated to improve the perceptual quality with considerable gains for nontransient frames.
Convention Paper 5997

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Session B **Saturday, May 8** **09:30 h–11:00 h**
Room 7.1b-2

SPATIAL PERCEPTION AND PROCESSING, PART 1

Chair: **Günther Theile**, Institut für Rundfunktechnik, Munich, Germany

09:30 h

B-1 Unidimensional Simulation of the Spatial Attribute "Ensemble Depth" for Training Purposes—Part 2: Creation and Validation of Reference Stimuli—*Tobias Neher, Tim Brookes, Francis Rumsey*, University of Surrey, Guildford, Surrey, UK

In the context of devising a spatial ear-training system, a study into the perceptual construct “ensemble depth” was executed. Based on the findings of a pilot study into the auditory effects of early reflection (ER) pattern characteristics, exemplary stimuli were created. Changes were highly controlled to allow unidimensional variation of the intended quality. To measure the psychological structure of the stimuli and hence evaluate the success of the simulation, multidimensional scaling (MDS) techniques were employed. Supplementary qualitative data were collected to assist with the analyses of the perceptual (MDS) spaces. Results show (1) that syllabicity of source material (rather than ER design) is crucial to depth hearing and (2) that unidimensionality was achieved, thus suggesting the stimuli to be suitable for training purposes.
Convention Paper 5998

10:00 h

B-2 Audibility Thresholds of Spatial Variations in a Single Acoustic Reflection—*Marinus M. Boone, Hiske W. Helleman*, Technical University of Delft, Delft, The Netherlands

When recording impulse responses of a concert hall for later processing in a spatial audio reproduction system such as Wave Field Synthesis (WFS), the question arises as to how far these impulse responses can be used for different source positions without a loss in spatial perception. A preliminary study has been carried out to find the threshold of audibility of spatial variations in the position of a single reflection. It was found that the minimum audible distance variation of a single reflection is 1 to 2 m, or 5 to 10 degrees, depending on the spatial configuration and whichever is the largest. From that result preliminary conclusions can be drawn about the necessary resolution in recording and synthesis of reflection patterns for WFS

rendering or other spatial reproduction systems.
Convention Paper 5999

10:30 am

B-3 Spatial Perception in Wave Field Synthesis Rendered Sound Fields: Distance of Real and Virtual Nearby Sources—*Helmut Wittek*[1, 2], *Stefan Kerber*[1, 3], *Francis Rumsey*[2], *Günther Theile*[1]
[1]Institut für Rundfunktechnik, Munich, Germany
[2]University of Surrey, Guildford, Surrey, UK
[3]Technical University of Munich, Munich, Germany

In this paper we investigate an alternative to the Gaussian density for modeling signals encountered in audio environments. The observation that sound signals are impulsive in nature, combined with the reverberation effects commonly encountered in audio, motivates the use of the sub-Gaussian density. The new sub-Gaussian statistical model and the separable solution of its maximum likelihood estimator are derived. These are used in an array scenario to demonstrate with both simulations and two different microphone arrays the achievable performance gains. The simulations exhibit the robustness of the sub-Gaussian-based method while the real world experiments reveal a significant performance gain, supporting the claim that the sub-Gaussian model is better suited for sound signals.
Convention Paper 6000

PAPERS

Session Z1 **Saturday, May 8** **09:30 h–11:00 h**
Corridor 7.1b

P A P E R S

POSTERS: AUTOMOTIVE AUDIO AND INSTRUMENTATION AND MEASUREMENT

09:30 h

Z1-1 TANDEM Digital Audio Amplifier—*Giovanni Franceschini*[1], *Alberto Bellini*[1], *Antonio De Benedetti*[1], *Michele Burlenghi*[1], *Francisco Violi*[2]
[1]University of Parma, Parma, Italy
[2]ASK Industries, Reggio Emilia, Italy

State-of-the art audio amplifiers can be classified into two major classes: linear amplifiers and switching amplifiers. The former class features low distortion but poor efficiency, while the latter features high efficiency coupled with high distortion and low bandwidth. In this paper a hybrid architecture is presented that combines linear and switching topology in order to obtain an audio amplifier featuring high efficiency, low distortion, and high bandwidth. The intrinsic structure of the switching stage allows an automatic spreading of the switching frequency, reducing EMI issues. A prototype amplifier was realized and tailored for automotive applications. The proposed architecture is patent pending.
Convention Paper 6001

09:30 h

Z1-2 Update to Automotive Doors as Loudspeaker Enclosures—*Roger Shively, Josh King*, Harman Becker Automotive Systems, Martinsville, IN, USA

This paper is an update to a previous study (Convention Paper 5752, presented at the AES 114th Convention), which used mechanical dynamic behavior data, impedance, and distortion measurements of several automotive doors to compare low-frequency performance and low-frequency sound quality. The updated information further investigates a methodology for quantifying door enclosures and refines the criteria for qualifying automotive doors as loudspeaker enclosures.
Convention Paper 6002

09:30 h

Z1-3 Evaluating Different Vehicle Audio Environments through a Novel Software-Based System—*Stefano Squartini*[1], *Francesco Piazza*[1], *Romolo Toppi*[2], *Massimo Navarri*[2], *Walter Lori*[2], *Ferruccio Bettarelli*[3], *Emanuele Ciavattini*[3], *Ariano Lattanzi*[3]

[1]Universitá Politecnica delle Marche, Ancona, Italy
[2]Faital S.p.A, Milan, Italy
[3]Leaff Engineering S.r.l., Jesi, Ancona, Italy

An original software-based system, featuring two different tools, is proposed for vehicle audio quality assessment. The first one performs the acquisition of relevant data for system modeling and canceling the undesired effects of the acquisition chain. The second offers a user-friendly interface for real-time simulation of different car audio systems and consequent subjective evaluation, where the listening procedure is directly experienced at a PC workstation. The validity of this approach has been examined through a subjective listening test set (more than 50 participants and 3 cars involved), developed by means of a dedicated software environment and based on appropriate ITU recommendations. Experimental results have shown that the quality rating delivered by conventional in-car procedure is confirmed when the software-based approach is used.
Convention Paper 6003

09:30 h

Z1-4 Measurement of Active Speech Level Inside Cars Using a Throat-Activated Microphone—*Fabio Bozzoli, Angelo Farina*, University of Parma, Parma, Italy

One of the most used intelligibility parameters is the Speech Transmission Index (STI). The technique for determining it uses an artificial speaker and listener. When signal-to-noise ratio is particularly low, for example inside cars, the value of STI is mainly influenced by this ratio. Determining the sound power of actual speakers is the only way to correctly determine the artificial mouth. We have implemented a technique that is based on a throat-activated microphone, which is able to find the level of a real speaker's voice inside the noisy spaces in effective conditions. We have particularly studied the speech inside cars and discovered how the value defined by norms may be extremely different from the real one. In this way, we have been able to produce more reliable excitation signals.
Convention Paper 6004

09:30 h

Z1-5 The Use of Continuous Phase for Interpolation, Smoothing, and Forming Mean Values of Complex Frequency Response Curves—*Joerg Panzer*[1], *Lampos Ferekidis*[2]

[1]Consultant, Salgen, Germany
[2]Consultant, Barsinghhausen, Germany

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The direct application of interpolation, smoothing or mean-value algorithms to complex-valued frequency response data may cause interference patterns and, because to this, does not yield the expected result. This paper demonstrates the effect of the use of continuous phase in a variety of applications such as interpolation between two frequency response curves, complex smoothing with down-sampling using a logarithmic grid, and forming mean values of a set of complex frequency response curves. The continuous phase-approach takes into account the multivalued property of the exponential function of the phase term.
Convention Paper 6005

09:30 h

Z1-6 Web-Based Acoustic Noise Measurement System—*Andrzej Czyzewski, Jozef Kotus*, Gdansk University of Technology, Gdansk, Poland

The concept and implementation of a multimedia computer system for the monitoring of environmental noise threats is presented. The principal aim of the project is to improve the effectiveness of prophylaxis of hearing diseases. This system makes it possible to receive, store, analyze, and visualize noise data coming from noise measurement equipment and from electronic questionnaires accessible through the Internet. A new concept of the USB noise meter with GPS is also presented.
Convention Paper 6006

09:30 h

Z1-7 Software Application for Electroacoustic Measurements Using the Time-Delay Spectrometry (TDS) Method—*Evaggelos Parlantzas, Charalampos Dimoulas, George Kalliris, George Papanikolaou, Christos Sevastiadis*, Aristotle University of Thessaloniki, Thessaloniki, Greece

This paper presents a software application that conducts electroacoustic measurements using a digital approach to time-delay spectrometry. Development is focused on simplified hardware requirements such as a personal desktop or laptop computer. A friendly and flexible user interface has been designed. Linear and logarithmic sweep test signals are generated and reproduced. System under test (e.g., room) response is recorded and stored in the hard disk. Energy time curve (ETC) and frequency domain analysis procedures are guided efficiently. Reverberation time in the case of a room is estimated very quickly. All task data may be restored later for further analysis. Finally, the results of comparison measurements using our appli-

cation to measurements with a widely accepted TDS analyzer are presented.
Convention Paper 6007

09:30 h

Z1-8 Triode Simulator—*Dimitri Danyuk*, Digital Research Labs, haverhill, MA, USA

The design for a low-noise amplifier is presented. The amplifier has a tube-like transfer characteristic and produces harmonic distortion components that are similar to triode preamplifiers.
Convention Paper 6008

PAPERS

Session C **Saturday, May 8** **13:00 h–16:00 h**
Room 7.1b-1

AUDIO ARCHIVING, STORAGE, AND RESTORATION; CONTENT MANAGEMENT

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

13:00 h

C-1 Taking Care of Tomorrow Before it Is Too Late—A Pragmatic Archiving Strategy—*Nicolas Hans*[1], *Johan de Koster*[2]
[1]Dalet Digital Media Systems, Paris, France
[2]Radio Netherlands, Hilversum, The Netherlands

An increasing number of broadcasters and organizations are considering the digitization of their media archives. Implementing digital media libraries so as to ensure the proper preservation of legacy archives has been recognized as a priority. Yet, many organizations are faced with a paradox: although strategic, these digitization projects are postponed because of budgetary constraints. This paper discusses several case studies and suggests a new approach to implementing a successful digital archiving strategy—one that will get approval and support from management.
Convention Paper 6009

13:30 h

C-2 Archiving of Radio Broadcast Data Using Automatic Metadata Generation Methods within MediaFabric Framework—*Jobst Löffler*[1], *Joachim Köhler*[1], *Helge Blohmer*[2], *Kai-Uwe Kaup*[2]
[1]Fraunhofer Institute for Media Communication, Sankt Augustin, Germany
[2]VCS Aktiengesellschaft, Bochum, Germany

This paper describes methods for automatic extraction of descriptive metadata for audio material and the workflow of archiving. These new algorithms and archiving tools developed at Fraunhofer IMK are to be directly integrated into MediaFabric, a commercially available radio broadcasting framework. Processing steps are based on pattern recognition algorithms and include speech/nonspeech detection, loudspeaker change detection and classification, jingle and advertising recognition. The extracted audio structure is described as a hierarchical representation of segment nodes annotated with suitable metadata. The extended retrieval application allows interactive display and navigation of the audio structure. A novel

approach to keyword search based on a syllable representation of audio material is used for effective retrieval within the digital radio archive.
Convention Paper 6010

14:00 h

C-3 EBU Tests of Commercial Audio Watermarking Systems—*Andrew Mason*, BBC Research and Development, Tadworth, Surrey, UK

Audio watermarking has recently had a resurgence of interest, spurred on by the desire for copyright protection of digital audio recordings. Several audio watermarking techniques, some dating back more than 30 years, are described briefly here. The uses to which watermarking might be put are also summarized. Attention is then focussed on the requirements identified by the EBU applicable to distribution over the Eurovision and Euroradio networks. The EBU issued a call for systems to meet its requirements. Subjective and objective tests were done on the systems supplied for testing. Audibility and robustness of the watermarks were measured. The results are encouraging for those considering using audio watermarking in broadcast applications.
Convention Paper 6011

14:30 h

C-4 Morphological Sound Description: Computational Model and Usability Evaluation—*Julien Ricard, Perfecto Herrera*, Pompeu Fabra University, Barcelona, Spain

Sound samples of metadata are usually limited to a source label and several related textual labels. In the context of sound retrieval this makes the retrieval of sounds having no identifiable source ("abstract sounds") a hard task. We propose a description framework focusing on intrinsic perceptual sound qualities, based on Schaeffer's research on sound objects, which could be used to represent and retrieve abstract sounds and to refine traditional search by source for non-abstract sounds. We show that some perceptual labels can be automatically extracted with good performance, avoiding the time-consuming manual labeling task, and that the resulting representation is evaluated as useful and usable by a pool of users.
Convention Paper 6012

15:00 h

C-5 A Nonlinear Rhythm-Based Style Classification for Broadcast Speech-Music Discrimination—*Enric Guaus, Eloi Batlle***,** Pompeu Fabra University, Barcelona, Spain

Speech-music discriminators are usually designed under some rigid constraints. This paper deals with a more general speech-music discriminator designed for the AIDA project. The system is based on a Hidden Markov Model (HMM) style classification process in which the styles are grouped into two major categories: speech or music. The goals of this subsystem are: (1) the expandable possibilities with the addition of some new styles (like "phone female voice"); (2) the use of new rhythmical descriptors in combination with other typical ones; and (3) the robustness of our speech/music discriminator in many different environments by using some mathematical morphology and nonlinear postprocessing techniques. The techniques used in our system allow a fast track in changes between styles and, thus, typical confusions in commercials can be easily cleaned. The accuracy of this system can be up to a 94.3 percent in broadcast radio environment.
Convention Paper 6013

15:30 h

C-6 Audio Patch Method in Audio Decoders—MP3 and AAC—*Han-Wen Hsu, Chi-Min Liu, Wen-Chieh Lee*, National Chiao Tung University, Hsin-Chu, Taiwan

Current audio encoders like MP3 or AAC leads to some artifacts due to the bit-rate constraint. This paper considers two artifacts. The first artifact is the unusual spectral valley which is perceptually heard as fishy noise. The second one is the spectrum clipping which leads to the muffling audio. This paper proposes the spectrum patch method to handle the two artifacts in the decoders. The technique can be included in MPEG1—Layer3 and MPEG4—AAC (Advanced Audio Coding) decoders to conceal the artifacts without prior information on the original audio tracks. Intensive experiments have been conducted on various encoders and audio tracks to check the quality improvement and the possible risks in degrading the quality. The objective test measures used is the recommendation system by ITU-R Task Group 10/4.
Convention Paper 6014

16:00 h

Technical Committee Meeting on Archiving, Restoration, and Digital Libraries (TC Room 1, Hall 7.1b)

Session D **Saturday, May 8** **13:30 h–16:00 h**
Room 7.1b-2

PAPERS

SPATIAL PERCEPTION AND PROCESSING—PART 2

Chair: **Francis Rumsey**, University of Surrey, Guildford, Surrey, UK

13:30 h

D-1 Motion-Tracked Binaural Sound—*V. Ralph Algazi, Richard Duda, Dennis Thompson*, University of California, Davis, CA, USA

A new method is presented for capturing, recording, and reproducing spatial sound. The method generalizes binaural recording, preserving the information needed for dynamic head-motion cues. These dynamic cues stabilize the perceived sound field, largely eliminate front/back confusion, and greatly reduce the need for customization to the listener. During either capture or recording, the sound field in the vicinity of the head is sampled with a microphone array. During reproduction, a head tracker is used to determine the microphones that are closest to the positions of the listener's ears. Interpolation procedures are used to produce the headphone signals. The properties of different methods for interpolating the microphone signals are presented and analyzed.
Convention Paper 6015

14:00 h

D-2 IKA-SIM: A System to Generate Auditory Virtual Environments—*Andreas Silzle*[1], *Pedro Novo*[1], *Holger Strauss*[2]
[1]Ruhr-Universität Bochum, Bochum, Germany
[2]VCS Aktiengesellschaft, Bochum, Germany

The basic requirements for an Auditory Virtual Environment (AVE) are presented and a system based on a physical approach (IKA-SIM), employing the mirror-image model to generate the early reflections, is described. The static and dynamic structure of the IKA-SIM software (written in C++) is shown in diagrams and the computational requirements for real-time performance are delineated. IKA-SIM is able to render rooms of arbitrary shape, to account for frequency-dependent absorption factors, and to calculate high-order reflections in real-time on a standard PC. The different interfaces for real-time interaction are presented. IKA-SIM supports headphone and loudspeaker reproduction. A new elevation panning algorithm for loudspeaker reproduction is introduced. Design aspects relevant to a real-time AVE system are presented.
Convention Paper 6016

PAPERS

14:30 h

D-3 Further Study of Sound Field Coding with Higher Order Ambisonics—*Jérôme Daniel, Sébastien Moreau*, France Telecom R&D, Lannion, France

Higher Order Ambisonics (HOA) is a spatialization technology based on the spherical harmonic decomposition of a sound field. This technology provides a flexible way to represent and render 3-D sound scenes. Nevertheless, it is only recently that the problem of representing near field sources and recording natural sound fields (infinite bass boost) has been addressed and partially solved. This paper proposes a further study on the frequency-dependent amplitude of the spherical harmonic components for finite distance source encoding, by connecting it with several parameters: the source distance, the microphone array size (case of natural recording), the size of the targeted reproduction area, and the distance of the reproduction loudspeakers. A solution is investigated to limit excessive low- frequency amplification of high order ambisonic components while still achieving a correct reproduction of wave fronts. As a particular result, it leads to improved distance coding tools for virtual sources, especially when these are simulated inside the listening area.
Convention Paper 6017

15:00 h

D-4 Sound-Source Radiation Synthesis: From Stage Performance to Domestic Rendering—*Olivier Warusfel, Nicolas Misdariis*, IRCAM, Paris, France

A diffusion device based on a digitally controlled 3-D array of loudspeakers—La Timée—was developed in order to synthesize a given radiation pattern from the combination of a set of elementary directivities. This radiation synthesis method, designed for musical and performance constraints (real-time control, musical vocabulary associated to different directivity patterns, etc.), has been used for stage performances and sound installations. In order to translate the sound experience for domestic reproduction, the paper addresses the postproduction step where the spatial image associated with the radiation synthesis is transcoded on conventional formats like transaural, ambisonic or 5.1 formats. The method is based on the characterization of the performance room with the different elementary directivities, which are then superimposed according to the musical score.
Convention Paper 6018

15:30 h

D-5 Surround Sound: Relations of Listening and Viewing Configurations—The Useful Assignment of Loudspeaker Basis Width to Video Picture Dimension
—*Gerhard Steinke*, Audio Consultant, Berlin Germany

The growing penetration of the DVD into today's marketplace also simulates a more intimate association of sophisticated multichannel sound and larger high-quality images with the "ideal" TV format 16:9 (1.78:1). Nevertheless, different geometrical assignments may exist between image size and loudspeaker basis width in production studios, multimedia rooms, and home living rooms—besides varying room-acoustical and qualitative conditions. For best possible use of program essences, the exact locations of sound and picture sources should be assigned as near as possible, i.e., with corresponding horizontal listening angles and viewing angles for avoiding disturbing discrepancies between acoustical and optical perspective. Essential connections are considered, and the recommendation is derived to adjust the optimum viewing distance 2H with regard to appropriate large loudspeaker basis width and image size for high-quality home theater experiences.
Convention Paper 6019

16:00 h

Technical Committee Meeting on High-Resolution Audio (TC Room 2, Hall 7.1b)

Session Z2 **Saturday, May 8** **14:00 h–15:30 h**
Corridor 7.1b

POSTERS: AUDIO IN COMPUTERS AND AUDIO VIDEO SYSTEMS

14:00 h

Z2-1 Film Music Recording Using Technology—*Robert Ellis-Geiger*, Hong Kong Polytechnic University, Hong Kong

This paper represents a new approach to recording acoustic music for film and has the potential to dramatically improve the performance of an orchestra, small ensemble or solo performer for highly emotional scenes. Additionally, this approach to film music production could allow for sudden changes to be made during the scoring session, such as last minute film edits that mostly result in changes to the final score. This paper will also reveal some of the processes in film music composition and the use of technology as integral to understanding this new method of music production for film.
Convention Paper 6020

14:00 h

Z2-2 Development of a Multimedia Learning Module Covering the Field of Perceptual Audio Coding
—*Daniel Pape*[1], *Gerrit Kalkbrenner*[2], *Jan Maihorn*[3]
[1]ZAS Berlin, Germany
[2]Universität Dortmund, Dortmund, Germany
[3]Rotterdam Conservatory for Music and Dance, Rotterdam, The Netherlands

An electronic learning module covering the field "perceptual audio coding" (famous representative: MP3) was specified, designed, and implemented by means of the multimedia software Macromedia Director. The presented program is split into different modules. These include: (1) an auralization of the filterbank implemented in MP3; (2) simulations of various classic psychoacoustic experiments (mainly masking thresholds) for three different music styles—other audio examples exhibit (1) a comparison of the sound quality of a Fraunhofer MP3 codec at different bit rates and (2) a comparison of today's most important audio and speech codecs (like Windows MediaEncoder and Real9) at different bit rates; and (3) audio examples and explanation of typical error signals introduced by perceptual audio coding. Finally, a structured explanation of the mode-of-operation of an MP3 encoder and technical papers with further references to publications on perceptive coding were included in the presented software.
Convention Paper 6021

14:00 h

Z2-3 Controlling the Quality of Audio Services in the Internet—*Bernhard Feiten, Ingo Wolf, Andreas Graffunder*, Media Solutions, Berlin, Germany

Future services in the Internet have to support heterogeneous networks and end-devices. Audio and video services have to support a flexible adaptation of bit rate. MPEG-21 provides a multimedia framework that supports the "digital item adaptation" in various ways. Adaptation of the quality of a service is supported by the bitstream syntax description language (BSDL). Additionally, utilities exist to describe the relation between the scaling of the bitstream and the related perceived quality. The brightness, the cleanness, and the wideness are proposed as dimensions to assess the quality and to derive parameters for controlling the audio transmission. A mapping of these features on the model output values (MOV's) of the ITU assessment method PEAQ is proposed.
Convention Paper 6022

14:00 h

Z2-4 Design and Implementation of a Commodity Audio System—*Men Muheim*, Swiss Federal Institute of Technology, Zurich, Switzerland

This paper presents a Ph.D. thesis that envisions a distributed audio system based on commodity computer components. It examines to what extent the "real-time" attributes of mainstream operating systems lead to audio dropouts and therefore to quality loss. It studies extrapolation methods to prevent loss of quality and shows that quality improvement to a nonannoying level is possible. A synchronization mechanism is implemented on application layer in order to facilitate the use of Ethernet as the only communication network. Thereby the thesis shows that a synchronization accuracy of 10µs between separated loudspeakers is feasible. Furthermore the thesis proposes a novel software framework, which makes the development of distributed audio services easier.
Convention Paper 6023

14:00 h

Z2-5 Advanced 3-D Audio Algorithms by a Flexible, Low Level Application Programming Interface—*Aleksandar Simeonov, Giorgio Zoia, Robert Lluis Garcia, Daniel Mlynek*, EPFL, Lausanne, Switzerland

The constantly increasing demand for a better quality in sound and video for multimedia content and virtual reality

compels the implementation of more and more sophisticated 3-D audio models in authoring and playback tools. A very careful and systematic analysis of the best available development libraries in this area was carried out, considering different application programming interfaces, their features, extensibility, and portability among each other. The results show that it is often difficult to find a tradeoff between flexibility, efficiency, quality, and speed. In this paper we propose a low level, modular DSP library, which can be used to implement advanced 3-D audio models; it is based on reconfigurable primitive methods required by most 3-D algorithms and it provides fast development and good flexibility.
Convention Paper 6024

14:00 h

Z2-6 Real-Time Internet MPEG-4 SA Player and the Streaming Engine—*Alvin Su, Yi-Song Shao*, National Cheng-Kung University, Tainan, Taiwan

MPEG-4 structure audio is an algorithmic-based coding standard designed for low bit-rate high-quality audio. With this standard, the desired sound can be identical on both the encoder side and the decoder side by using Structured Audio Orchestra Language (SAOL) to generate sound samples. It requires a player and a streaming engine when real-time interactive Internet presentations are necessary. In this paper we present such a system implemented and applied over IBM PC-based computers. The proposed streaming engine follows ISMA specification and its implementation is closely related to Apple's Darwin Server. After the streaming SA player receives the bitstream from the server, it converts SAOL data stream to JAVA codes and links to a proposed scheduler program generated from SASL data stream for direct execution such that one can hear the sound in real time. Unlike sfront ?, no intermediate C codes and C compilers are necessary. In order to improve the performance, optimized software modules such as the core opcodes and the core wavetable engine have been embedded. Significant speedup is achieved compared to the reference SAOLC decoder. Real-time demonstration of the system will be made during the presentation. Discussion of the possible future algorithmic coding method using JAVA is also given.
Convention Paper 6025

14:00 h

Z2-7 Application Scenarios of Wearable- and Mobile-Augmented Reality Audio—*Tapio Lokki, Heli Nironen, Sampo Vesa, Lauri Savioja, Aki Härmä, Matti Karjalainen,* Helsinki University of Technology, Espoo, Finland

Several applications for wearable and mobile reality audio are presented. All applications exploit a headset where microphones are integrated into small headphone elements. The proposed system allows us to implement applications where virtual sound events are superimposed to the user's auditory environment to produce an augmented audio display. In addition, binaural audio-over-IP connections, wired or wireless, are discussed. Finally, some future application scenarios are sketched.
Convention Paper 6026

14:00 h

Z2-8 ITC Clean Audio Project—*Ben Shirley, Paul Kendrick*, University of Salford, Salford, UK

The Clean Audio project involves the assessment of a number of processes on perception of Dolby Digital 5.1 audio for TV. Specifically, the research aims to assess the effect on the enjoyment and clarity of television sound for hard-of-hearing viewers. The preliminary study presented here used subjective listening tests to assess the level of left and right front surround channels required to enhance the enjoyment of the audio without detracting from the clarity of the dialog. The findings provide useful guidelines on the benefits and use of surround sound for hearing impaired viewers.
Convention Paper 6027

14:00 h

Z2-9 Audiovisual Virtual Environments: Enabling Real-Time Rendering of Early Reflections by Scene Graph Simplification—*Andreas Dantele, Ulrich Reiter, Mathias Schwark*, Technical University of Ilmenau, Ilmenau, Germany

In an audiovisual virtual 3-D environment the conformance of visual and auditory impression is important to provide a high level of immersion. Restrictions of processing power for the auralization (including early and late reverberation) are usually high due to the demanding visual rendering. For the audio part a trade-off between high accuracy and speeding up the rendering process has to be found, especially for real-time user interaction. We show how the rendering process of early reflections can be done in real time by reducing the scene representation to auditory relevant elements. A suitable scene simplification algorithm and corresponding audio rendering issues are discussed.
Convention Paper 6028

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Session E **Saturday, May 8** **16:00 h–18:00 h**
Room 7.1b-1

ANALYSIS AND SYNTHESIS OF SOUND—PART 1

Chair: **Oliver Hellmuth**, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

16:00 h

E-1 A Methodology for Detection of Melody in Polyphonic Musical Signals—*Rui Pedro Paiva, Teresa Mendes, Amílcar Cardoso*, University of Coimbra, Coimbra, Portugal

In this paper we present a bottom-up method for melody detection in polyphonic music signals. Our approach is based on the assumption that the melodic line is often salient in terms of note intensity (energy). First, trajectories of the most intense harmonic groups are constructed. Then, note candidates are obtained by trajectory segmentation (in terms of frequency and energy variations). Too short, low-energy, and octave-related notes are then eliminated. Finally, the melody is extracted by selecting the most important notes at each time, based on their intensity. We tested our method with excerpts from 12 songs encompassing several genres. In the songs where the sole stands out clearly, most of the melody notes were successfully deleted. However, for songs where the melody is not that salient, the algorithm performed poorly. Nevertheless, we could say that the results are encouraging.
Convention Paper 6029

16:30 h

E-2 Octave-Error Proof Timbre-Independent Estimation of Fundamental Frequency of Musical Sounds—*Alicja Wieczorkowska, Jakub Wróblewski*, Polish-Japanese Institute of Information Technology, Warsaw, Poland

Estimation of fundamental frequency (so called pitch tracking) can be performed using various methods. However, all these algorithms are susceptible to errors, especially octave ones. In order to avoid these errors, pitch-trackers are usually adjusted to particular musical instruments. Therefore problems arise when one wants to extract fundamental frequency independent on the timbre. Our goal is to elaborate a method of fundamental frequency extraction, which works correctly for any timbre. We propose a multi-algorithm approach, where fundamental frequency estimation is based on results coming both from a range of frequency tracking methods and additional parameters of sound. We also propose frequency tracking methods based on direct analysis of a signal and its spec-

trum. We explain the structure of our estimator and the obtained results for various musical instruments and sound articulation techniques.
Convention Paper 6030

17:00 h

E-3 Further Steps towards Drum Transcription of Polyphonic Music—*Christian Dittmar, Christian Uhle*, Fraunhofer Institute for Digital Media Technology, Ilmenau, Germany

This paper presents a new method for the detection and classification of unpitched percussive instruments in real-world musical signals. The derived information is an important prerequisite for the creation of a musical score, i.e., automatic transcription, and for the automatic extraction of semantic meaningful metadata, e.g., tempo and musical meter. The proposed method applies independent subspace analysis using non-negative independent component analysis and principles of prior subspace analysis. An important extension of prior subspace analysis is the identification of frequency subspaces of percussive instruments from the signal itself. The frequency subspaces serve as information for the detection of the percussive events and the subsequent classification of the occurring instruments. Results are reported on 40 manually transcribed test items.
Convention Paper 6031

17:30 h

E-4 Generation of Musical Scores of Percussive Unpitched Instruments from Automatically Detected Events—*Christian Uhle, Christian Dittmar*, Fraunhofer Institute for Digital Media Technology, Ilmenau, Germany

This paper addresses the generation of a musical score of percussive unpitched instruments. A musical event is defined as the occurrence of a sound of a musical instrument. The presented method is restricted to events of percussive instruments without determinate pitch. Events are detected in the audio signal and classified into instrument classes, the temporal positions of the events are quantized on a tatum grid, musical meter is estimated, and preparatory beats are identified. The identification of rhythmic patterns on the basis of the frequency of their occurrence enables a robust identification of the tempo and gives valuable cues for the positioning of the bar lines using musical knowledge.
Convention Paper 6032

PAPERS

Session Z3 **Sunday, May 9** **09:30 h–11:00 h**
Corridor 7.1b

POSTERS: SIGNAL PROCESSING AND AUDIO IN BROADCASTING

09:30 h

Z3-1 Efficient Arbitrary Sample Rate Conversion Using Zero Phase IIR Filters—*Seyed Ali Azizi*, Harman/Becker Automotive Systems, Ittersbach, Germany

Modern asynchronous sample rate converters (ASRCs) are composed of an interpolation filter to increase the sample rate by an integer factor, followed by a polynomial interpolator that produces the desired output samples at arbitrary output sampling time instants. A crucial feature determining the precision of the ASRCs is the phase linearity of the interpolation filter in use. That is the main reason why traditionally easily realizable linear phase FIR filters, but not IIR filters suffering from inherent phase nonlinearity, have been employed as interpolation filters, although IIR filters are more economical. This paper introduces a novel ASRC design approach which uses the zero phase IIR filtering concept to produce highly efficient, linear phase IIR interpolation filters to be used in ASRCs. The basic concept is explained and the functions of the involved units are investigated.
Convention Paper 6033

09:30 h

Z3-2 A Study on Implementing Switching Transfer Functions Focusing on Wave Discontinuity—
Akihiro Kudo, Haruhide Hokari, Shoji Shimada, Nagaoka University of Technology, Niigata, Japan

Many papers have described moving sound image localization schemes that use loudspeakers or headphones. Most of these schemes are based on switching spatial transfer functions, so wave discontinuity occurs at the moment of switching, which degrades the sound quality. While the characteristics of the wave discontinuity depend on the moving sound image localization schemes, no paper appears to have considered the relationship between the wave discontinuity and the scheme used. To rectify this omission, this paper examines three approaches: simple switching approach, overlap-add approach, and fade-in-fade-out approach, We assess the sound degradation caused by wave discontinuity and use the objective measure of spectrum distortion width to quantify the wave discontinuity. We also carry out paired comparison tests as subjective assessments. Both assessments verify that the third approach is the best of the three.
Convention Paper 6034

PAPERS

09:30 h

Z3-3 Warped DFT Based Perceptual Noise Reduction System—*Alexander Petrovsky, Marek Parfieniuk, Adam Borowicz*, Bialystok Technical University, Bialystok, Poland

This paper considers a novel application of the Warped Discrete Fourier Transform in a single channel noise reduction system. Namely, the WDFT is simultaneously the basis for the spectral weighting and psychoacoustic model, thus allowing the overall system to operate strictly in a critical band domain. The warped transform allows nonuniform allocation of the z-transform frequency samples in accord with the Bark scale. Thus, the psychoacoustic modeling is more accurate than in the DFT-based solutions, and the subjective quality of enhanced speech increases. The noise suppression algorithm utilizes the majority of currently most advanced ideas in perceptually motivated spectral weighting. Its advantage is in the fact that the masking threshold is directly involved in the weighting rule.
Convention Paper 6035

09:30 h

Z3-4 Digital Loudspeaker Arrays Driven by 1-Bit Signals —*Nicolas-Alexander Tatlas, John Mourjopoulos*, University of Patras, Patras, Greece

Loudspeaker arrays driven by digital bit streams are direct digital-signal to acoustic transducers, usually comprising a digital signal processing module with driving actuators. Current research efforts are focusing on topologies directly driven by multibit digital bit streams. In this paper the above investigations are extended to the case of using 1-bit signals such as sigma-delta for driving such topologies using time and frequency domain analysis. Simulation results will be presented for ideal actuators. Finally, an optimized architecture for such a loudspeaker will be proposed, based on this analysis.
Convention Paper 6036

09:30 h

Z3-5 Unsupervised Classification Techniques for Multipitch Estimation—*Julie Rosier, Yves Grenier*, ENST, Paris, France

In this paper we present a fast and efficient technique for multipitch estimation of musical signals. We deal with mixtures where several instruments are present in a monophonic recording. The approach consists in clustering the spectral peaks of the mixture to obtain a spectral repre-

sentation of each musical note. These spectra are then used to estimate the fundamental frequencies. We compare two techniques for the classification of the spectral peaks: a K-means procedure and a simpler aggregation technique associated with a criterion that represents the closeness to harmonicity for any couple of frequency peaks. This comparison is made on complex mixtures holding various musical instruments and piano chord mixtures. The effectiveness of the two estimation methods is presented using computation of pitch recognition rates and mean source number estimate.
Convention Paper 6037

09:30 h

Z3-6 Speaker Array Calibration Using Inter-Speaker Range Measurements—*Jeffrey Walters*[1], *Scott Wilson*[1], *Jonathan Abel*[2]
[1]Stanford University, Stanford, CA, USA
[2]Universal Audio, Inc., Santa Cruz, CA, USA

Given an array of speakers and a set of noisy inter-speaker range estimates, we consider the problem of estimating the relative positions of the array elements. A closed-form position estimator that minimizes an equation error norm is presented and shown to be related to a multidimensional scaling analysis. The information inequality is used to bound position estimate mean square error and to gauge the accuracy of the closed-form estimator. A geometric interpretation of the bound variance is given and used in examining our simulation results.
Convention Paper 6038

09:30 h

Z3-7 Loudness in TV Sound—*Jean Paul Moerman*, VRT, Belgian National Broadcasters for the Flemish Community, Brussels, Belgium

Nowadays, in a world of super-audio formats, the loudness problem is one of the most important elements for an audience to get an informative and relaxed experience. When zapping through the channels, loudness-differences are quite the usual thing. But also within one broadcaster, levels are not consistent from one program switch to another. Viewers are extremely annoyed and complaints are to be expected, but no major enhancement has been undertaken in the broadcast world. Surprisingly enough the transition from analog to digital did not improve matters—on the contrary, it became much worse!

The trap to be the loudest is very tempting. The use of heavily compression techniques and the development of new signal processors have fed a culture of rivaling

loudness. Louder attracts attention, but in the end the viewer will turn down the volume and discover a beaten, compressed, and uninteresting sound. A common solution to the loudness-problem is to try to correct the level at the end of the production chain. Inserting just one peace of equipment right before transmission cannot solve this: a processor, which solves all of the problems. This results in a sound that even causes listening fatigue. It should be clear that a more extensive solution is necessary.

Our solution was the installation of a broadcast processor in every facility unit within the VRT. The program will also be processed just before transmission and pro format: mono, nicam-stereo, and recently audio for DVB-T. Most important was not to forget the training of all technicians from every unit as postproduction, studio, OB-facility, continuity, and transmission. Even the (non-sound-minded) editors who fill in all the production aspects in an off-line video facility, do need some facts on how to judge loudness. The external production units of advertising trailers and programs should also be given the necessary information.
Convention Paper 6039

09:30 h

Z3-8 Audio Processing for Digital Broadcast Mediums
—*Frank Foti*, Omnia Audio, Cleveland, OH, USA

Over the past few years, as development, testing, and rollout progressed regarding the HD-radio (IBOC), DAB, and DRM transmission systems, audio processing has been one of the key components to augment this new technology. It became apparent that dynamics processing would figure in both the aural and technical performance aspects of these new systems. It has been successfully proven that signal processing improved other bit-rate-reduced audio services such as Internet audio streaming, especially at low bit rates. This paper will offer examples of proven methods that demonstrate the benefits of audio processing in the digital broadcast system. There are some important issues that must be considered, or digital radio's benefits will not be fully realized.
Convention Paper 6040

12:00 h

Technical Committee Meeting on Loudspeakers and Headphones (TC Room 1, Hall 7.1b)

Technical Committee Meeting on Acoustics and Sound Reinforcement (TC Room 2, Hall 7.1b)

PAPERS

Session F **Sunday, May 9** **10:00 h–13:00 h**
Room 7.1b-1

ANALYSIS AND SYNTHESIS OF SOUND—PART 2

Chair: **Matti Karjalainen**, Helsinki University of Technology, Espoo, Finland

10:00 h

F-1 Some Clues to Build a Sound Analysis Relevant to Hearing—*Laurent Millot*, ENS Louis Lumiere, Noisy le Grand, France

Analysis tools used in research laboratories for sound synthesis by musicians or sound engineers can be rather different. Discussion of the assumptions and of the limitations of these tools allows us to propose a tool as relevant and versatile as possible for all the sound actors with a major aim: one must be able to listen to each element of the analysis because hearing is the final reference tool. This tool should also be used, in the future, to reinvestigate the definition of sound (or acoustics) on the basis of some recent works on musical instrument modeling, speech production, and loudspeaker design. Audio illustrations will be given.
Convention Paper 6041

10:30 h

F-2 Synthesizing Coupled-String Musical Instruments by a Multichannel Recurrent Network—*Wei-Chen Chang, Alvin W. Y. Su*, National Cheng Kung University, Tainan, Taiwan

Struck string instruments such as pianos usually have groups of strings terminated at some common bridges. Because of the strong coupling phenomenon, the produced tones exhibit highly complex amplitude modulation patterns. Therefore, it is difficult to determine the synthesis model parameters such that the synthesized tones can match the recorded tones. In this paper a multichannel recurrent network is proposed based on three previous works: the coupled-string model, the commuted piano synthesis method, and the IIR synthesis method. This paper attempts to automatically extract the synthesis parameters by using a neural-network training algorithm without the knowledge of physical properties of the instruments. Encouraging results are shown in the computer simulations.
Convention Paper 6042

11:00 h

F-3 Nonlinearity Modeling for Spectral Pattern Recognition in Piano Chords—*Luis Ortiz-Berenguer,*

Javier Casajús-Quirós, Universidad Politécnica de Madrid, Madrid, Spain

The nonlinear behavior of the piano strings is a very important issue when the chords have to be recognized using spectral patterns. In order to calculate the spectral patterns and masks used in the recognition algorithm it is necessary to model the effects of nonlinearity. A model using intermodulation products have proved to give good results. For validation of the model we recorded 11 pianos and analyzed the "A" note of the octaves 1 to 7, using 4 different forces. The basis of this model are presented in this paper.
Convention Paper 6043

11:30 h

F-4 Waveform Synthesis Using Bezier Curves with Control Point Modulation—*Bob Lang*, University of the West of England, Bristol, UK

Bezier curves are frequently used in graphical applications and drawing packages. In this paper the author presents a technique of direct sound wave synthesis using Bezier curves. The technique is further expanded by modulating the position of the Bezier control points as synthesis takes place to create waveforms with complex harmonic structures. The paper also outlines how the technique can be used to create a musical instrument (synthesizer).
Convention Paper 6044

12:00 h

F-5 A Highly Optimized Nonlinear Least Squares Technique for Sinusoidal Signal Analysis: From $O(K^2N)$ to $O(N \log(N))$—*Wim D'haes*, University of Antwerp, Antwerp, Belgium

In the field of sinusoidal modeling, two types of least squares amplitude estimation methods are distinguished. A first group of methods estimate the complex amplitude of each sinusoid in an iterative manner. Although their main disadvantage is that they are unable to resolve overlapping frequency responses, they are used frequently because of their computational complexity being $O(N \log(N))$. By contrast, methods that compute all amplitudes simultaneously can resolve overlapping frequency responses but their computational complexity scales with a power of three in function of the number of sinusoidal components. In this work a method is proposed which allows to compute all amplitudes simultaneously and still has an $O(N \log(N))$ complexity. This is realized by explicitly including a window with a band-limited frequency

response in the least squares derivation resulting in a band diagonal system of equations which can be solved in linear time. Since overlapping frequency responses are allowed, an iterative method must be used to optimize the frequencies resulting in a nonlinear least squares technique. A commonly used technique is Newton optimization which requires the computation of the gradient and the Hessian matrix. Also here, the same computational gain is realized by applying the same methodology.
Convention Paper 6045

12:30 h

F-6 Partial Tracking Based on Future Trajectories Exploration—*Mathieu Lagrange*[1], *Sylvain Marchand*[2], *Jean-Bernard Rault*[1]

[1]France Telecom R&D, Cesson Sevigne cedex, France
[2]University of Bordeaux, Bordeaux, France

This paper introduces a partial-tracking algorithm suitable for the sinusoidal modeling of polyphonic sounds. A new method, based on the backward exploration of possible extensions of the partials in future frames, is proposed to cope with the lack or the corruption of spectral data. The allocation of spectral peaks to a partial is done by considering possible trajectories in future frames where frame hopping is allowed. A suitable transition probability that takes into account missing or rejected peaks is proposed. The trajectory that exhibits the higher probability is searched for and the corresponding peak for the current frame is chosen to extend the partial.
Convention Paper 6046

09:00 h

Technical Committee Meeting on Semantic Audio Analysis (TC Room 2, Hall 7.1b)

Session G **Sunday, May 9** **10:00 h–12:30 h**
Room 7.1b-2

LOW BIT-RATE AUDIO CODING—PART 1

Chair: **Jürgen Herre**, Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany

10:00 h

G-1 MPEG-4 Audio Lossless Coding—*Tilman Liebchen*[1], *Yuriy Reznik*[2], *Takehiro Moriya*[3], *Dai Tracy Yang*[4]
[1]Technical University of Berlin, Berlin, Germany
[2]RealNetworks, Inc., Seattle, WA, USA
[3]NTT Human and Information Science Lab, Atsugi, Japan
[4]University of Southern California, Los Angeles, CA, USA

Lossless coding will become the latest extension of the MPEG-4 audio standard. The lossless audio codec of the Technical University of Berlin was chosen as the reference model for "MPEG-4 Audio Lossless Coding (ALS)." The MPEG-4 ALS encoder is based on linear prediction, which enables high compression even with moderate complexity, while the corresponding decoder is straightforward. The paper describes the basic elements of the codec as well as some additional features, gives compression results, and points out envisaged applications.
Convention Paper 6047

10:30 h

G-2 A Low Power SBR Algorithm for the MPEG-4 Audio Standard and Its DSP Implementation—*Osamu Shimada*[1], *Toshiyuki Nomura*[1], *Yuichiro Takamizawa*[1], *Masahiro Serizawa*[1], *Naoya Tanaka*[2], *Mineo Tsushima*[2], *Takeshi Norimatsu*[2], *Chong Kok Seng*[3], *Kuah Kim Hann*[3], *Neo Sua Hong*[3]
[1]NEC Corporation, Kanagawa, Japan
[2]Matsushita Electric Industrial Co., Ltd., Osaka, Japan
[3]Panasonic Singapore Laboratories Pte. Ltd., Singapore

This paper proposes a Low Power Spectral Band Replication algorithm (LP-SBR) adopted in the MPEG-4 audio standard. It operates with low computational complexity compared to the conventional SBR algorithm called the High Quality SBR algorithm (HQ-SBR). LP-SBR utilizes real-valued processing instead of complex-valued processing used in HQ-SBR for complexity reduction. To minimize the sound quality degradation caused by this reduction, LP-SBR employs aliasing reduction techniques and a gain compensation technique. Subjective quality test results show that there is no statistical difference between LP-SBR and HQ-SBR when they are incorporated into

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AAC decoders. A complexity comparison of both SBR decoders implemented on 16-bit fixed-point DSPs shows that an AAC decoder with LP-SBR requires 30 percent less computational complexity than that with HQ-SBR.
Convention Paper 6048

11:00 h

G-3 MP3 Surround: Efficient and Compatible Coding of Multichannel Audio—*Jürgen Herre*[1], *Christof Faller*[2], *Christian Ertel*[1], *Johannes Hilpert*[1], *Andreas Hoelzer*[1], *Claus Spenger*[1]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]Agere Systems, Allentown, PA, USA, USA

Finalized in 1992, the MP3 compression format has become a synonym for personalized music enjoyment for millions of users. The paper presents a novel extension of this popular format which adds support for the coding of multichannel signals, including the widely used 5.1 surround sound. As a prominent feature of the extended format, complete backward compatibility with existing stereo MP3 decoders is retained, i.e., standard decoders reproduce a full stereo downmix of the multichannel sound image. The paper discusses the underlying advanced technology enabling the representation of multichannel sound at bit rates that are comparable to what is currently used to encode stereo material. Results for subjective sound quality are presented; related activities of the MPEG standardization group are reported.
Convention Paper 6049

11:30 h

G-4 Reduction of Artifacts in MPEG-AAC with MDCT Spectrum Regularization—*Olivier Derrien*[1], *Laurent Daudet*[2]
[1]Université de Toulon et du Var, La Garde, France
[2]Université Pierre et Marie Curie, Paris, France

In the context of lossy audio coding, the power spectral density of stationary tones can be over/underestimated in some windows due to the time-shift sensitivity of the Modified Discrete Cosine Transform (MDCT), which leads to potentially audible coding artifacts. This paper discusses the advantages of using a nearly time-shift invariant regularized MDCT spectrum for the bit allocation in MPEG-AAC coders. We show how this modification applies to the standard iterative algorithm, as well as to a more efficient model-based framework. Objective and subjective results indicate that the overall quality is significantly improved

when rich stationary sounds are encoded at low bit-rates or when the coder operates in a variable bit-rate mode.
Convention Paper 6050

12:00 h

G-5 The Efficient Temporal Noise Shaping Method—
Chi-Min Liu, Wen-Chieh Lee, Tzu-Wen Chang, National Chiao Tung University, Hsinchu, Taiwan

Temporal noise shaping has been defined in MPEG-4 AAC to control the pre-echo noise in attack signals. The module, which is especially important for the MPEG-4 Low Delay AAC due to the absence of a window switching mechanism, can shape and control quantization noise spread to improve the quality under bit rate constraint. However, this paper illustrates that the TNS will introduce three artifacts. The first artifact is similar to the Gibbs phenomenon which has high noise level occurring at the edge of the attack signal. The second effect is the time-domain aliasing noise which has unusual noise at a distance from the attack time frame. The third is the noise spreading with the TNS filter orders. This paper will propose the efficient TNS method which shapes noise with good concerns on the above three artifacts. Also, we provide an efficient computing method to activate the TNS. Both subjective and objective tests are conducted to illustrate the improvement over existing TNS methods.
Convention Paper 6051

09:00 h

Technical Committee Meeting on Automotive Audio (TC Room 1, Hall 7.1b)

P A P E R S

Session H **Sunday, May 9** **13:00 h–17:00 h**
Room 7.1b-1

MULTICHANNEL SOUND

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

13:00 h

H-1 Multiactuator Panel (MAP) Loudspeakers: How to Compensate for Their Mutual Reflections—*Rik van Zon*[1], *Etienne Corteel*[2], *Diemer de Vries*[1], *Olivier Warusfel*[2]
[1]Technical University of Delft, Delft, The Netherlands
[2]IRCAM, Paris, France

Wave Field Synthesis (WFS) allows reproduction of spatial and temporal properties of a target sound field over a large listening area. Thanks to their screen shape, Multi-Actuator Panels (MAP) represent a good alternative for WFS reproduction in multimedia installations. However, MAP loudspeakers act as reflectors for acoustic waves that disturb the perception of the target sound field. A general listening room compensation technique is proposed, based on multichannel inversion, that allows attenuating early reflections caused by a reflector using loudspeakers integrated into this reflector (e.g., MAP loudspeakers). After an analysis of the geometrical arrangement of the panels, the method processes separately the free field equalization of the loudspeaker array and the reflection compensation. Simulation and measurements show that the attenuation is effective over the entire listening area.
Convention Paper 6052

13:30 h

H-2 Advanced Multichannel Audio Systems with Better Impression of Presence and Reality—*Kimio Hamasaki, Koichiro Hiyama, Toshiyuki Nishiguchi, Kazuho Ono*, NHK, Science & Technical Research Laboratories, Tokyo, Japan

Various sound systems have been studied in NHK with the objective of developing the next-generation broadcasting system. This paper introduces the ultimate 22.2 multichannel audio system for ultrahigh definition video with 4000 scanning lines, and an advanced multichannel sound system with frontal loudspeakers placed in several rows for reproducing a live sound field. The former system has 3 vertical layers of loudspeakers with 2 LEFs. The latter system consists of frontal loudspeaker-ranks and rear loudspeaker-arrays for reproducing a natural impression of depth and ambience. This paper describes the principal

advantages of the newly proposed multichannel audio system over ordinary multichannel sound systems such as 5.1.
Convention Paper 6053

14:00 h

H-3 Visualizing Spatial Sound Imagery of Multichannel Audio—*John Usher, Wieslaw Woszczyk*, McGill University, Montreal, Quebec, Canada

To describe a multichannel audio experience in terms of its spatial features requires us to consider separately how we hear both the direct and indirect sound. We have developed and tested a Graphical User Interface (GUI) to allow a listener to describe where they hear both of these acoustic parts in an audio scene. The GUI has previously been used as a tool for describing where we hear the direct sound in an audio sound field, and we now extend the experimental paradigm to measure where we hear the indirect sound. We map the spatial extent of the reflected sound and describe a category system for describing a spatial sound attribute called "definition." We tested the GUI using 5 loudspeakers arranged according to BS-775 to replay "live" multichannel sound recordings of three different musical pieces (of which two were duets and one solo). Graduate Tonmeister students used the GUI to describe these sound scenes, and a variety of statistical analyses are presented which show how data from the GUI can be used to represent perceived spatial sound imagery.
Convention Paper 6054

14:30 h

H-4 Wave Field Synthesis in the Real World: Part 2—In the Movie Theater—*Thomas Sporer, Beate Klehs*, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

In anechoic rooms the concept of Wave Field Synthesis (WFS) has already proven to provide superior spatial sound over a large part of the room. In anechoic space WFS needs a huge number of loudspeakers. In "normal" listening conditions simulated and real acoustics interfere with each other making the generated wave field less exact. This paper describes listening tests conducted to evaluate WFS in a movie theater with about 100 seats. Parameters being tested are the number of loudspeakers, the distance between loudspeakers, the position of the simulated source, and the position of listeners relative to the loudspeakers. In an additional test the audio-visual coherence has been investigated.
Convention Paper 6055

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15:00 h

H-5 **Wave Field Synthesis 3-D Simulator Based on Finite-Difference Time-Domain Method—***Jose Escolano*[1], *Sergio Bleda*[1], *Basilio Pueo*[1], *José Javier López*[2]
[1]University of Alicante, Alicante, Spain
[2]Technical University of Valencia, Valencia, Spain

The Finite-Difference Time-Domain (FDTD) method was successfully developed to model electromagnetic systems. This technique has been also used in several disciplines such as optics and acoustics. A new approach for Wave Field Synthesis (WFS) simulation using FDTD instead of the finite difference classic method is presented. This software allows the precise evaluation and behavior monitoring of different WFS configurations in time domain and thus in a particular frequency band. Moreover, simulations can be analyzed inside a room or in a free space.
Convention Paper 6056

15:30 h

H-6 **Reproduction of Reverberation with Spatial Impulse Response Rendering—***Ville Pulkki*[1], *Juha Merimaa*[1,2], *Tapio Lokki*[1]
[1]Helsinki University of Technology, Espoo, Finland
[2]Ruhr-Universitaet Bochum, Bochum, Germany

A technique for spatial reproduction of room acoustics, Spatial Impulse Response Rendering (SIRR), has been recently proposed. In the method, a multichannel impulse response of a room is measured, and responses for loudspeakers in an arbitrary multichannel listening setup are computed. When the responses are loaded to a convolving reverberator, they will create a perception of space corresponding to the measured room. The method is based on measuring with a sound field microphone or a comparable system, and on analyzing direction-of-arrival and diffuseness at frequency bands. An omnidirectional response is then positioned to a loudspeaker system according to analyzed directions and diffuseness. In this paper the SIRR method is reviewed and refined. The reproduction quality of SIRR and some other systems is evaluated with listening tests, and it is found that SIRR yields a natural spatial reproduction of the acoustics of a measured room.
Convention Paper 6057

16:00 h

H-7 **New and Advanced Features for Audio Rendering in the MPEG-4 Standard—***Jürgen Schmidt, Ernst F.*

Schröder, Thomson Corporate Research, Hannover, Germany

Since the early days of audio stereophony, we tend to think of audio transmission and audio presentation in terms of loudspeaker feeds or "channels." This seemed to be appropriate for as few channels as two and still reasonable for five, but is rapidly losing its meaning with the advent of technologies like, e.g., wave field synthesis. A key part of MPEG-4 is the introduction of object-oriented thinking for the description, generation, transport, and rendering of audio scenes. Binary Information for Scenes (BIFS) is that part of the MPEG-4 standard that enables transmission of scene descriptions together with the audio signals to facilitate the final rendering. The latest version of BIFS (Version 3) now has a number of improvements and new concepts including: presentations of sound fields (inclusion of Ambisonics and Wave Field Synthesis); presentation of "shaped sounds"; and the possibility of combining 3-D audio with 2-D video. The concepts and achievements by MPEG with audio BIFS V3 will be explained in detail.
Convention Paper 6058

16:30 h

H-8 The Quick Reference Guide to Multichannel Microphone Arrays Design Part II: Using Supercardioid and Hypocardioid Microphones—
Michael Williams[1], *Guillaume Le Du*[2]
[1]Sounds of Scotland, Le Perreux sur Marne, France
[2]Radio France, Paris, France

This paper is the second part of a paper presented at the 110th AES Convention in Amsterdam. A selection of different multichannel microphone arrays is again presented but this time using Supercardioid and Hypocardioid microphones. Five-channel array configurations are described with respect to their particular characteristic: microphone directivity, specific segment coverage, segment offset values where necessary, microphone coordinates and orientations. Arrays have been chosen so as to assist the sound engineer in the search for the optimum microphone array for a given recording situation.
Convention Paper 6059

17:00 h

Technical Committee Meeting on Audio Recording and Storage Systems (TC Room 1, Hall 7.1b)

PAPERS

Session I **Sunday, May 9** **13:00 h–15:30 h**
Room 7.1b-2

LOW BIT-RATE AUDIO CODING—PART 2

Chair: **Markus Erne**, Scopein Research, Aarau, Switzerland

13:00 h

I-1 Audio Coder Enhancement Using Scalable Binaural Cue Coding with Equalized Mixing—*Frank Baumgarte, Christof Faller, Peter Kroon*, Agere Systems, Allentown, PA, USA

A major application for Binaural Cue Coding (BCC) is multichannel audio coding. A previously proposed system combines a full-band BCC coder for spatial parameters with an audio coder for a down-mixed representation of the multichannel input. This paper presents a scalable hybrid coder combining a partial-band BCC as preprocessor and postprocessor with a subband coder. The hybrid system supports a gradual tradeoff of bit rate and spatial image ranging from transparent multichannel and stereo to full-band BCC. To avoid coloration from the required up- and downmixing within BCC, an equalized mixing scheme based on a binaural loudness model is proposed. Subjective tests and bit rate simulations confirm the expected benefits of the hybrid coder in the transition range from full-band BCC to stereo.
Convention Paper 6060

13:30 h

I-2 Spatial Decomposition of Time-Frequency Regions: Subbands or Sinusoids—*Aki Härmä*[1], *Christof Faller*[2]
[1]Helsinki University of Technology, Espoo, Finland
[2]Agere Systems, Allentown, PA, USA

Techniques where a stereo or a multichannel signal is decomposed into spatial source-labeled time-frequency slots by level, time-difference, and coherence metrics have become popular in recent years. Good examples are binaural cue coding and up/downmixing techniques. In this paper we will provide an overview and discuss parallel approaches in the field of array processing and blind source separation. Typically, time-frequency slots are formed from subband representations of signals. However, it is also possible to produce a similar spatial decomposition for a parametric representation (sinusoids, transients, and noise) of a stereo or multichannel audio signal. Advantages and disadvantages of the two approaches in audio coding applications are discussed in this paper.
Convention Paper 6061

14:00 h

I-3 A Guideline to Audio Codec Delay—*Manfred Lutzky*[1], *Gerald Schuller*[2], *Marc Gayer*[1], *Ulrich Krämer*[2], *Stefan Wabnik*[2]
[1]Fraunhofer Institute for Integrated Circuits IIS, Erlangen, Germany
[2]Fraunhofer Institute for Digital Media Technology IDMT, Imenau, Germany

Digital audio processing has been revolutionized by perceptual audio coding in the past decade. The main parameter to benchmark different codecs is the audio quality at a certain bit rate. For many applications, however, delay is another key parameter that varies between only a few and hundreds of milliseconds depending on the algorithmic properties of the codec. Latest research results in low-delay audio coding can significantly improve the performance of applications such as communications, digital microphones, and wireless loudspeakers with lip synchronicity to a video signal. This paper describes the delay sources and magnitude of the most common audio codecs and thus provides a guideline for the choice of the most suitable codec for a given application.
Convention Paper 6062

14:30 h

I-4 Parametric Audio Coding Based Wavetable Synthesis —*Marek Szczerba, Werner Oomen, Marc Klein Middelink*, Philips Digital Systems Laboratories, Eindhoven, The Netherlands

For mobile applications memory and computational complexity requirements are very strict. Therefore, traditional wavetable/FM synthesis methods have to compromise between the number and the quality of instruments in the soundbank. This paper presents a wavetable synthesizer employing a parametric representation of the soundbank samples, sharing the advantages of both wavetable and parametric synthesis methods. The soundbank is compact and thus easy to store and transmit, and the sound quality can match that of traditional wavetable synthesis. Moreover, postprocessing of samples in a parametric representation—such as pitch change, filtering, and envelope—can be performed directly in the parametric domain, effectively reducing synthesizer complexity.
Convention Paper 6063

15:00 h

I-5 Removal of Birdie Artifact in Perceptual Audio Coders—*Vinod Prakash, Anil Kumar, Preethi Konda*,

PAPERS

Sarat Chandra Vadapalli, Ittiam Systems Pvt. Ltd., Bangalore, India

The birdie artifact is the predominant factor affecting audio quality of perceptual coders operating at very low bit rates. Conventional approaches to overcome the birdie artifact involve use of low-pass filters to reduce the amount of signal to quantize. This approach does not eliminate the birdie artifact if the effect is seen in the in-band components. This paper proposes a new algorithm to overcome the birdie artifact and hence improve the audio quality. The proposed algorithm modifies the bit allocation strategy such that the critical bands are preserved, while still maintaining the perceptual distortion criteria. Results of spectrogram analysis are presented.
Convention Paper 6064

Session Z4 **Sunday, May 9** **13:00 h–14:30 h**
Corridor 7.1b

P A P E R S

POSTERS: SPATIAL PERCEPTION AND PROCESSING AND ANALYSIS AND SYNTHESIS OF SOUND

13:00 h

Z4-1 Objective Measurements of Sound-Source Localization in a Multichannel Transmission System for Videoconferencing—*Juan José Gómez-Alfageme, Elena Blanco-Martin, S Torres-Guijarro, F. Javier Casajús-Quirós*, Universidad Politécnica de Madrid, Madrid, Spain

In videoconference systems formed by microphone and loudspeaker arrays, the sound field reproduced in the receiving room must be as similar as possible to the sampled field by the microphone array (according to the wave field synthesis). Different measurements of objective and subjective quality can be made. A measurement method has been developed based on spatial localization in the horizontal plane. In order to do it, two different situations have been compared: first, a real source placed at different azimuth angles in front of the listener; second, the virtual source created by the loudspeaker array. Interpolated HRTFs have been calculated according to several methods and in order to determine the azimuth angle, the cross-correlation function (IACC) and the interaural time difference (ITD) have been evaluated.
Convention Paper 6065

13:00 h

Z4-2 Plane-Wave Decomposition of Volume Element Mesh Data Simulations—*Bård Støfringsdal, U. Peter Svensson*, Norwegian University of Science and Technology, Trondheim, Norway

Sound-field simulations at low frequencies usually employ finite elements or other mesh-based methods. For auralization, output data from these methods need to be converted to a format compatible with auralization methods such as Wave Field Synthesis (WFS), Higher Order Ambisonics (HOA) or binaural reproduction. A method is proposed for converting the mesh data to plane wave components using a circular array of virtual sources centered around the listening position. The method is based on solving sets of linear propagation equations in the frequency domain. Results are presented for two-dimensional examples and numerical issues are discussed.
Convention Paper 6066

13:00 h

Z4-3 Headphone Processor Based on Individualized Head-Related Transfer Functions Measured in a Listening Room—*Witold Mickiewicz, Jerzy Sawicki*, Technical University of Szczecin, Szczecin, Poland

Listening via headphones as opposed to loudspeakers introduces changes in perception of an acoustic atmosphere and spaciousness (lateralization effect). This can be improved using Head-Related Transfer Function (HRTF) technology. In contrast to previous works, we propose a method based on individualized HRTFs measured simply by the end-user in acoustical conditions of a listening room using his own hi-fi set. It gives even better subjective results using standard equipment and a proper postprocessing (equalization) than available market products based on non-individual filters. We present an idea based on individualized head-and room-related transfer function, the algorithm, and technical details of individualized headphone processors. All necessary processing can be done in DSP or FPGA to create a PC-independent consumer-electronics unit.
Convention Paper 6067

13:00 h

Z4-4 A Lateral Angle Tool for Spatial Auditory Analysis—
Ben Supper, Tim Brookes, Francis Rumsey, University of Surrey, Guildford, Surrey, UK

A new method is presented for examining the spatial attributes of a sound recorded within a room. A binaural recording is converted into a running representation of an instantaneous lateral angle. This conversion is performed in a way that is influenced strongly by the workings of the human auditory system. Auditory onset detection takes place alongside the lateral angle conversion. These routines are combined to form a powerful analytical tool for examining the spatial features of the binaural recording. Exemplary signals are processed and discussed in this paper. Further work will be required to validate the system against existing auditory analysis techniques.
Convention Paper 6068

13:00 h

Z4-5 A Layered Data Model for Information Management in Sound Coding Architectures—*Enrique Alexandre, Antonio Pena*, Universidade de Vigo, Vigo, Spain

This paper presents some ideas for the appropriate management of every information source present in a generic speech or audio coder. This task becomes more neces-

sary as coding structures become more complex. An appropriate organization and processing of this information is a key point for an efficient implementation, in terms of complexity and quality. First, a data structure will be proposed, inspired by classic comprehension theories, which sorts the information into three different hierarchical levels. Based on this structure, a global sound encoder block diagram will be described. This model is based on blackboard models, commonly applied in speech recognition applications. Finally, it will be shown how an MPEG-2/4 AAC-LC coder can be considered as a particular case of the proposed model.
Convention Paper 6069

13:00 h

Z4-6 Real-Time Room Equalization Based on Complex Smoothing: Robustness Results—*Panagiotis Hatziantoniou, John Mourjopoulos*, University of Patras, Patras, Greece

This paper investigates the robustness of room acoustics real-time equalization using inverse filters derived from the complex smoothing of the transfer function using perceptual criteria. The robustness of the method is assessed by real-time tests that compare the performance of complex smoothing-based equalization (for different filter lengths) with the traditional, ideal inverse filtering, over a range of room locations, which differ from the ones where response measurements were taken. Objective measurements and audio examples will show that the complex smoothing-based equalization performance is largely immune to position changes and does not introduce processing artifacts, problems affecting the traditional ideal inversion.
Convention Paper 6070

13:00 h

Z4-7 Personalized Mobile Ring Tone Generator Using Madelbrot Music—*Suthikshn Kumar*, Larsen & Toubro Infotech Ltd., Bangalore, India

Mandelbrot equations are very popular for generating images and music. We propose to use them for generating mobile ring tones. These mandelbrot ring tones are both entertaining and melodious. As the computations required for generating melodious mandelbrot tones are simple iterations, the ring tone generator can be integrated with the mobile handset. The Fuzzy Mandelbrot sets are proposed for extending the usefulness of the ring tone generator. This ring tone generator is personalized by using the audiogram. People with hearing impairments will benefit by the personalized ring tone generator. A PC-based

mobile phone ring tone generator demonstration is being developed based on the Nokia series 60 SDK for Symbian OS mobile handsets. This will be used for demonstrating the concepts proposed in this paper.
Convention Paper 6071

Session J **Sunday, May 9** **15:30 h–18:00 h**
Room 7.1b-2

SPATIAL AUDIO CODING

Chair: **Erik Schuijers**, Philips Digital Systems Laboratories, Eindhoven, The Netherlands

P A P E R S

15:30 h

J-1 High-Quality Parametric Spatial Audio Coding at Low Bit Rates—*Jeroen Breebaart*[1], *Steven van de Par*[1], *Armin Kohlrausch*[1,2], *Erik Schuijers*[3]

[1]Philips Research Laboratories, Eindhoven, The Netherlands
[2]Eindhoven University of Technology, Eindhoven, The Netherlands
[3]Philips Digital Systems Labs, Eindhoven, The Netherlands

Recently, so-called binaural cue coding schemes have been introduced. These audio coding schemes transmit two perceptually relevant sound localization cues (i.e., level and time differences between the input channels), combined with a mono audio signal. Although these schemes are able to reconstruct the locations of various sound sources quite effectively, other aspects of the spatial ambience (such as the spatial diffuseness of reverberation) cannot be captured in this way. In this paper we present an extension to these spatial coding schemes, which comprises, in addition, a spatial sound-field parameter that is able to capture ambience properties. Experiments show that the combination of three spatial parameters enables highly efficient, high-quality audio representations.
Convention Paper 6072

16:00 h

J-2 Low-Complexity Parametric Stereo Coding—*Erik Schuijers*[1], *Jeroen Breebaart*[2], *Heiko Purnhagen*[3], *Jonas Engdegård*[3]

[1]Philips Digital Systems Laboratories, Eindhoven, The Netherlands
[2]Philips Research Laboratories, Eindhoven, The Netherlands
[3]Coding Technologies, Stockholm, Sweden

Parametric stereo coding is a technique to efficiently code a stereo audio signal as a monaural signal plus a small amount of stereo parameters. The monaural signal can be encoded using any audio coder. The stereo parameters can be embedded in the ancillary part of the mono bitstream creating backwards mono compatibility. In the

decoder, first the monaural signal is decoded after which the stereo signal is reconstructed from the stereo parameters. In this paper a low- complexity decoder solution is described based on complex-modulated filter banks. Combinations of the parametric stereo decoder with both a parametric coding scheme and with aacPlus will be elucidated.
Convention Paper 6073

16:30 h

J-3 Synthetic Ambience in Parametric Stereo Coding—*Jonas Engdegård, Heiko Purnhagen, Jonas Rödén, Lars Liljeryd*, Coding Technologies, Stockholm, Sweden

Parametric stereo coding in combination with an efficient coder for the underlying monaural audio signal results in the most efficient coding scheme for stereo signals at very low bit rates available today. While techniques for lateral localization have been studied since early intensity stereo coding tools, synthesis of stereophonic ambience was only recently applied in parametric stereo coding systems. This paper studies different techniques for synthetic ambience generation in the context of parametric stereo coding systems and discusses their mono-compatibility. Implementations of these techniques in combination with mp3PRO and aacPlus are presented together with experimental results.
Convention Paper 6074

17:00 h

J-4 Efficient Bit Distribution Strategy for Stereophonic Audio Coders—*Sarat Chandra Vadapalli, Vinod Prakash*, Ittiam Systems Pvt. Ltd., Bangalore, India

Maintenance of audio quality under the resource constraints on embedded platforms is very crucial. One of the major factors affecting the quality of stereophonic audio coders is the method of distribution of bits across channels of a stereo pair. Conventional approaches use perceptual entropy, a computationally intensive metric, to distribute bits across channels. Improper computation or absence of this metric can severely degrade the audio quality. This paper presents an efficient and robust scheme to distribute the bits across channels, without using perceptual entropy, while still maintaining the audio quality. In the proposed scheme, the bit allocation for both channels is performed simultaneously, by allocating bits from a common bit pool. A detailed example illustrating this scheme is presented.
Convention Paper 6075

17:30 h

J-5 Backward Linear Prediction for Lossless Coding of Stereo Audio—*Jean-Luc Garcia, Philippe Gournay, Roch Lefebvre*, University of Sherbrooke, Sherbrooke, Quebec, Canada

Lossless audio coding aims at achieving the lowest possible bit rate for transmission or storage of audio without any loss of information. This is usually done by first removing redundancy from the audio signal, and then applying entropy coding to the residual signal. Linear prediction (LP), when applied to monophonic signals, is a very effective way to remove redundancy. It produces minimum-phase predictors that are efficiently compressed by combining vector quantization with a meaningful representation of the LP coefficients (such as the LSFs). When applied to stereo signals, however, joint channel prediction often produces nonminimum-phase predictors, whose quantization requires a high bit rate and poses stability problems. In this paper we show that backward estimation of the LP coefficients (where those are estimated at the decoder, on the past decoded signal) solves most of the problems associated with the use of joint channel prediction in a lossless audio coder.
Convention Paper 6076

Session Z5 **Sunday, May 9** **16:00 h–17:30 h**
Corridor 7.1b

PAPERS

POSTERS: PSYCHOACOUSTICS, PERCEPTION, AND LISTENING TESTS

16:00 h

Z5-1 The Causal Relationship of Headphone Tone-Coloration Variations Related to the Human Pinna Influence—*Florian M. König*, Ultrasone AG, Penzberg, Germany

Head-related sound reproduction devices vary in transducer characteristics: the basic acoustic principles such as open/closed or circum- or supra-aural systems. Furthermore, the transducers de-centered placement inside the ear cup influences the tone quality. These headphone techniques were evaluated many times. One creation with a spatial reproduction of sound was much more conspicuous statistically because of a higher quantity recommended sound quality judgment as “too much” and “less high frequency range.” This forced investigations to find the reason for those strange review accumulations. Four different headphone types were measured by seven testers by inserting probe microphones in the auditory canal. The research result shows an electro-acoustic cause for perceived tone coloration of headphones by transducer positioning and the human pinna filtering efficiency.
Convention Paper 6077

16:00 h

Z5-2 Auditory Cues in the Perception of Self-Motion—*Bill Kapralos, Daniel Zikovitz, Michael Jenkin, Laurence R. Harris*, York University, Toronto, Canada and Centre for Research in Space Technology, Toronto, Ontario, Canada

Despite its potential importance, few studies have methodically examined the role of auditory cues in the perception of self-motion. Here we describe a series of experiments that investigate the relative roles of various combinations of physical motion and decreasing sound source intensity cues to the perception of linear self-motion. Subjects were blindfolded and physically moved toward a target in the presence or absence of a fixed-intensity stationary sound source, remained stationary while presented with a sound stimulus whose intensity was decreased or remained stationary while the sound stimulus was physically moved away from them. In all conditions an over-estimation of self-motion resulted that systematically varied with acceleration. Performance was most veridical with both auditory and physical cues. With-

out physical motion, auditory cues resulted in the greatest over-estimation, however, accuracy improved with increasing acceleration.
Convention Paper 6078

16:00 h

Z5-3 A New Mathematical Approach to Describe Localization —*Philip Mackensen*, T-Systems, Berlin, Germany

The localization of a single sound source can be described mathematically by a new formalism to be presented here. Commonly, the HRTF (head-related transfer function) is described as a function of variables related to the sound source position and of variables related to the spectrum. In this new approach the multivariable representation of the HRTF is replaced by introducing two independent transfer functions, one only regarding the position and the other regarding solely the source's spectral attributes. Therefore, it can be separated between the three dimensional local space and the "spectral space." This offers a localization independent of the Gestalt of the sound source.
Convention Paper 6079

16:00 h

Z5-4 Strategies to Increase the Applicability of Methods for Objective Assessment of Audio Quality—*Jayme Garcia Arnal Barbedo, Amauri Lopes*, State University of Campinas, Campinas, Brazil

The current ITU standard for objective assessment of audio quality, Perceptual Evaluation of Audio Quality (PEAQ), has some shortcomings that prevent its reliable use for a number of codification conditions and some kinds of signals. This paper aims to improve the PEAQ performance through the following proposals: (1) modifications in the manner the signals are submitted for the assessment; (2) improvement of existing Model Output Variables (MOVs); (3) creation of new MOVs; (4) determination of a better architecture for the neural network that maps the MOVs into a single estimate for the subjective score. The results are compared to those ones achieved by PEAQ.
Convention Paper 6080

16:00 h

Z5-5 Subjective Evaluation of an Equalization Method for Loudspeakers Based on Random Parametric Optimization of IIR Filters—*Germán Ramos, José Javier López*, Universidad Politecnica de Valencia, Valencia, Spain

In this paper a subjective evaluation of a novel method for loudspeaker equalization is presented. The equalization is performed using a direct method with random parametric optimization for the design of a bank of second-order peak filters, RaPOSOS. The subjective evaluation has been carried out using a preselected jury composed of lecturers, research staff, and university students related to the audio engineering field. For evaluating its performance, it has been compared with other well known equalization methods using the ABX test. In particular, our method with different levels of approximation, has been compared with long FIR filters obtained by minimum square error criteria. The results show that with relatively low-order filters, the perceived difference is anecdotic or nonexistent, requiring less computational cost.
Convention Paper 6081

16:00 h

Z5-6 A Composite Physiological Model of the Inner Ear for Audio Coding—*Alexei V. Ivanov*[1], *Alexander A. Petrovsky*[2]
[1]Belorussian State University of Informatics and Radio Electronics, Minsk, Belarus
[2]Bialystok Technical University, Bialystok, Poland

The alternative approach to psychoacoustical masking modeling is to model such phenomena as suppression, spread-of-excitation, and IHC adaptation, which are among the underlying physiological phenomena for psychoacoustically observed masking. This paper proposes a physiologically grounded model for threshold estimation. It includes a reconfigurable nonuniform filterbank to simulate the "cochlear amplifier" and associated suppression effect; a digital compartmental IHC model to account for their adaptive responses; a spiking neuron auditory nerve model to simulate the spread-of-excitation. It allows designing coders, which retain enough information to create the identical excitation pattern in the auditory nerve compared to that of the original signal. Since our model is based on the masking physiology, its application is justified in the complex audio signals case.
Convention Paper 6082

16:00 h

Z5-7 Perception of Temporal Decay of Low-Frequency Room Modes—*Matti Karjalainen*[1], *Poju Antsalo*[1], *Aki Mäkivirta*[2], *Vesa Välimäki*[1]
[1]Helsinki University of Technology, Espoo, Finland
[2]Genelec Oy, Iisalmi, Finland

Modal equalization has recently been of research interest in order to improve sound reproduction in rooms that have

excessively strong modes at low frequencies. Instead of acoustic treatment by expensive and space-reserving absorbing structures, modal equalization is based on DSP affecting the electric-to-acoustic reproduction chain. Several DSP-based techniques for modal equalization have been proposed recently and tested in performance. From a perceptual point of view, however, no clear picture of the importance of controlled temporal decay has been shown, although it is known that toward lowest frequencies human hearing becomes increasingly insensitive to temporal details. In the present paper we conducted listening tests where only a single synthetic mode with increased decay time but magnitude-equalized response was used to find the JND threshold of increased decay time. The main conclusion is that at typical listening levels, downward to 100 Hz, the modal decay time (T60) is allowed to increase from 0.3 seconds by 0.1 to 0.4 seconds, while at 50 Hz even decay times of up to 2 seconds do not make a noticeable difference.
Convention Paper 6083

16:00 h

Z5-8 Psychoacoustic Cues in Room-Size Perception—
Sharaf Hameed, Jyri Pakarinen, Kari Valde, Ville Pulkki, Helsinki University of Technology, Espoo, Finland

The ability of human listeners to estimate the size of a room from the acoustical response of that room is an interesting and not yet thoroughly examined phenomenon. This paper uses simulated multichannel room impulse responses convolved with speech signals as stimuli in listening tests to explore the perception of room size. The synthetic room impulse responses contained two adjustable parameters, and our goal was to study how these parameters affect the perceived size of this virtual room. Listening tests were conducted to test the effect of reverberation time and the direct to reverberant energy ratio (D/R ratio). Sound samples with different parameter settings were presented as stimuli in a paired comparison test procedure. The results reveal that reverberation time is unequivocally the most important parameter. It appears that D/R ratio is not used in room size perception.
Convention Paper 6084

16:00 h

Z5-9 Proposed Changes to the Methods of Objective, Perceptual -Based Evaluation of Compressed Speech and Audio Signals—*Piotr Kozlowski, Andrzej Dobrucki*, Wroclaw University of Technology, Wroclaw, Poland

PAPERS

This paper discusses research about objective methods, which use psychoacoustic knowledge to estimate the quality of audio signals. The software written especially for this research is presented. This program allows for implementation of the different published methods for evaluation of the quality of perceptually coded audio signals. Protocols such as PAQM, PSQM, NMR, PEAQ, and PESQ are ready to use. All of these algorithms are used for simulation of the auditory system. The software is open for addition to the next protocols as the plug-ins. There is a possibility to change and improve earlier published protocols. Suggested changes, which improve results of objective evaluation, are presented. The criterion of optimization is the difference between results of subjective and objective evaluation.
Convention Paper 6085

Session K **Monday, May 10** **09:00 h–12:30 h**
Room 7.1b-1

SIGNAL PROCESSING—PART 1

Chair: **Stanley Lipshitz**, University of Waterloo, Waterloo, Ontario, Canada

09:00 h

K-1 Signal Processing Techniques for Robust Multichannel Sound Equalization—*John Sarris, Nick Stefanakis, George Cambourakis*, National Technical University of Athens, Athens, Greece

Multichannel equalization is generally accomplished by designing inverse filters to remove the distortion associated with the transmission paths between a set of sources and receivers. The filters are estimated by minimizing a cost function based on the least squares error criterion. However, under certain conditions this least squares error-based formulation fails to provide a solution or provides a solution that lacks robustness. These conditions are investigated and modifications are introduced in the definition of the cost function so that the problem always has a solution with increased robustness. Moreover, the multiple error LMS algorithm is employed to adapt the filter coefficients to their optimum values. Issues like convergence speed and stability are discussed, and simulation results are presented.
Convention Paper 6087

09:30 h

K-2 Equalization Methods with True Response Using Discrete Filters—*Ray Miller*, Rane Corp., Mukilteo, WA, USA

Equalizers with fixed frequency filter bands, although successful, have historically had a combined frequency response that at best only roughly matches the band amplitude settings. This situation is explored in practical terms with regard to equalization methods, filter band interference, and desirable frequency resolution. Fixed band equalizers generally use second-order discrete filters. Equalizer band interference can be better understood by analyzing the complex frequency response of these filters and the characteristics of combining topologies. Response correction methods may avoid additional audio processing by adjusting the existing filter settings in order to optimize the response. A method is described which closely approximates a linear band interaction by varying bandwidth, in order to efficiently correct the response.
Convention Paper 6088

PAPERS

10:00 h

K-3 Direct Method with Random Optimization for Parametric IIR Audio Equalization. Applications to One Way and Multiway Systems—*Germán Ramos, José Javier López*, Universidad Politécnica de Valencia, Valencia, Spain

This paper presents a novel method for audio equalization using IIR (Infinite Impulse Response) filters. The algorithm is based on a direct method with a random parametric optimization process using second- order sections (RaPOSOS). Given a loudspeaker response, and the definition of the desired electroacoustical target response, an optimized filter is obtained. For full band loudspeakers, a bank of peak filters is designed to perform the equalization. For multiway systems, the process is repeated for each way with bandpass targets using lowpass, highpass, and peak filters computing the combined response and performing time-align correction. The final result provides the parameters that define each filter (frequency, gain, Q) in correct order of importance; first the ones that perform deepest improvement, so that scalable solutions with different degrees of correction could be derived.
Convention Paper 6089

10:30 h

K-4 Performance Improvements for Audio Algorithms that Use Nonsequential Memory Accesses on Digital Signal Processors—*Matthew Watson*[1], *Vineet Ganju*[1], *Gaganjot Maur*[2]
[1]Texas Instruments, Inc., Stafford, TX, USA
[2]Texas Instruments (India) Pvt. Ltd., Bangalore, India

Many audio algorithms, such as room simulators and reverberators, operating on digital signal processors access large delay buffers in a nonsequential fashion. Generally, these delay buffers are too large to reside in the on-chip memory of the processor, so they must be placed in external, slow memories. Furthermore, the nonsequential accesses present a problem for maintaining high performance. This paper presents a number of methods that may be employed to improve the performance of the memory accesses of such algorithms. Methods examined include the use of direct CPU memory access, hardware data cache, and dedicated direct memory access (DMA) controllers. Additionally, the type of algorithm, delay taps, and sample block size will be examined and performance results will be presented.
Convention Paper 6090

PAPERS

11:00 h

K-5 Method for Estimating Magnitude and Phase in the MDCT Domain—*Corey Cheng*, Dolby Laboratories, San Francisco, CA, USA

This paper introduces a method for estimating the magnitude and phase responses in audio coders that employ the Modified Discrete Cosine Transform (MDCT). This technique computes magnitude and phase estimates at the decoder using two pieces of information: (1) MDCT coefficients transmitted by the encoder; (2) an estimate of the Modified Discrete Sine Transform (MDST) computed from the transmitted MDCT coefficients. In this manner, approximate magnitude and phase estimates suitable for use with some decoder-oriented signal processing techniques can be constructed entirely from MDCT coefficients available at the decoder. We show that these approximate methods are less computationally intensive than exact methods, and we compare the performance of the approximate methods to exact methods.
Convention Paper 6091

11:30 h

K-6 The Harmonic Content of a Limit Cycle in a DSD Bitstream—*Joshua Reiss, Mark Sandler*, Queen Mary, University of London, London, UK

This paper explores the effects of limit cycles on the frequency content in the DSD bitstream. We show how any periodic bitstream can be expressed as a sum of square waves of various phases with width equal to the sampling period. A Fourier expansion may be used to exactly determine the phases and amplitudes of all spectral content. We thus determine all harmonics that appear in the output, and through the comparison with psychoacoustic models, determine the audibility of limit cycles. These results are verified through the simulation of realistic high-order sigma-delta modulators, and put into the context of recent advances in the theory of limit cycles and idle tones in sigma delta modulators.
Convention Paper 6092

12:00 h

K-7 Toward a Better Understanding of 1-Bit Sigma-Delta Modulators—Part 4—*John Vanderkooy, Stanley Lipshitz*, University of Waterloo, Waterloo, Ontario, Canada

This is Part 4 of an ongoing investigation into the behavior of 1-bit sigma-delta modulators. In this paper we question the usual concept of the "average quantizer gain" as it

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applies to the quantizer transfer characteristic and the stability of a 1-bit modulator. We show that the concept is very poorly defined and of little use for understanding the operation of the 1-bit modulator. We also investigate a number of possible alternative definitions of the gain, and their significance.
Convention Paper 6093

Session L **Monday, May 10** **09:00 h–12:00 h**
Room7.1b-2

PAPERS

LOUDSPEAKERS—PART 1

Chair: **Juha Backman**, Nokia Mobile Phones, Espoo, Finland; Helsinki University of Technology, Espoo, Finland

09:00 h

L-1 Bit Expansion in Digital Loudspeakers with Oversampling and Noise Shaping—*Haihua Zhang*[1], *Simon Busbridge*[1], *Peter Fryer*[2],
[1]University of Brighton, Brighton, East Sussex, UK
[2]B&W Loudspeakers Ltd., Steyning, West Sussex, UK

The resolution of true digital loudspeakers is currently limited by their physical construction. Transducer arrays require 2N-1 speaklets and multiple voice coil topologies require N coils (N = the number of bits). Oversampling and noise shaping have been used to maintain resolution with fewer bits. Results are presented where the oversampled signal falls both within and outside the bandwidth of the radiator. A linear model is being developed to understand the observations. The radiator displacement shows little difference between the original and oversampled cases. It is concluded that the limited bandwidth of existing acoustical radiators is advantageous in acting as the reintegration filter. In circumstances where this is not possible the auditory system may perform this task.
Convention Paper 6094

09:30 h

L-2 Geometrical Stiffness of Loudspeaker Cones—*Peter Larsen*, Loudsoft, Horsholm, Denmark

The frequency response of a loudspeaker cone is affected by two main factors: material parameters and geometry. While the first may be generally understood, the inherent stiffness due to the basic geometry is the subject of this paper. Using Finite Element Modeling (FEM), first a flat cone disk is analyzed followed by shallow and deep conical cones plus curved concave and convex cones. The results are extended to include softer and high damping cone materials. The cone break-up behavior and frequency response is shown to be strongly dependent on the geometrical stiffness of the cone, which should therefore be considered a very important design parameter.
Convention Paper 6095

PAPERS

10:00 h

L-3 A Circuit Approach to Short Circuit Ring Design for High Power Woofers—*Lorenzo Fontanesi, Alessandro Salvini*, University of Rome, Rome, Italy

Demodulation ring solutions can offer many advantages in terms of harmonic distortion reduction in high power 18-inch woofers. In this paper we show a circuit approach to evaluate the effects of aluminum short circuit rings properly shaped to improve woofer performances. To find the Laplacian force that acts on the voice coil, the proposed approach allows the partial inductance calculation method to evaluate the distribution of eddy currents into the massive ring aluminum conductors. By partitioning the conductor into cells and modeling each cell by an equivalent circuit this method can give results showing a maximum error equal to 6 percent by comparing measurements to simulations.
Convention Paper 6096

10:30 h

L-4 On the Velocity Distribution at the Interface of Horn Driver and Horn—*Gottfried Behler, Michael Makarski*, Aachen University, Aachen, Germany

For the numerical simulation (BEM) of horns, the sound velocity distribution at the horn throat is required as one boundary condition. It is common to use plane wave excitation even at high frequencies since the shape of the real wave front in general is unknown. The error in the simulation result (directivity / frequency response) is difficult to predict and can only be judged by measurement of the real system. To achieve accurate simulation results the specific velocity distribution of each driver is required, which must be measured at the interface between horn driver and horn. A more general approach for simulation techniques is created using modal composition. Measurements and simulations of different systems are compared to verify this method.
Convention Paper 6097

11:00 h

L-5 Determining Two-Port Parameters of Horn Drivers Using Only Electrical Measurements—*Michael Makarski*, Aachen University, Aachen, Germany

The basic theory and a measurement procedure for the two-port description of horn drivers and horns was presented at the 111th AES Convention in New York, 2001 (Convention Paper 5409, Behler and Makarski, "Two-Port

Representation of the Connection between Horn Driver and Horn," [*JAES*, Vol. 51, No. 10, 2003]). It was shown that this method is a powerful tool for the development of loudspeakers, but it suffered from the restricted frequency range of the necessary acoustical impedance measurements with the Kundt's tube. A new method of measuring the driver's two-port parameters is presented here, using only electrical measurements and an acoustical reference impedance. The frequency range of the two-port parameters could be extended using this method. The theoretical approach and first results are presented.
Convention Paper 6098

11:30 h

L-6 Analysis and Minimization of Unwanted Resonances in Loudspeaker Systems via FEM Techniques—

Mario Di Cola[1], *Davide Doldi*[1], *Marco Mocellin*[1], *Rinaldo Grifoni*[2], *Paolo Antinori*[2], *Remo Orsoni*[2], *Giogio Santarelli*[2]
[1]Audio Labs Systems—LiSA Design Workgroup, Milan, Italy
[2]Proel S.p.A., Sant'Omero (TE), Italy

High output loudspeaker systems, particularly horn-loaded loudspeaker systems, are often severely affected by unwanted structural resonances due to the high sound pressure locally generated. Modern high power transducers, in fact, are capable of generating very high sound pressure. This sound pressure turns out to be a great stimulus for a cabinet's structural modes. An experimental procedure aimed at resonance minimization is shown. This method is based on FEM structural analysis techniques validated by microphone and accelerometer measurements.
Convention Paper 6099

Session Z6 **Monday, May 10** **09:30 h–11:00 h**
Corridor 7.1b

POSTERS: ROOM AND ARCHITECTURAL ACOUSTICS AND MUSICAL ACOUSTICS

09:30h

Z6-1 Adaptive Room Equalization in the Frequency Domain—*Jorge Leitao*[1], *Gabriel Fernandes*[2], *Aníbal Ferreira*[1,3]
[1]INESC Porto, Porto, Portugal
[2]DEEC, FCTUC, Coimbra, Portugal
[3]FEUP, Porto, Portugal

This paper addresses the implementation of a real-time 20-band adaptive digital audio equalizer for room equalization. The system has been implemented on a TMS320C6711 DSP platform and performs adaptive filtering using techniques of fast filtering in the frequency domain that include an adaptation procedure. The paper explains how the structure of a previously designed graphic equalizer has been improved to support adaptivity, describes its operation as well as its functionality based on a graphical user interface, and presents the results of tests that have been conducted to optimize its performance.
Convention Paper 6100

09:30 h

Z6-2 Acoustic Reconstruction of Buildings in the Ancient City of Olympia—*Stamatis Vassilantonopoulos, John Mourjopoulos*, Univeristy of Patras, Patras, Greece

Virtual acoustics can assist the aural exploration and the study of the acoustic properties of famous buildings of antiquity. Here, examples of such reconstruction of ritual and public buildings of the ancient Greek city of Olympia are presented, and findings of their acoustic behavior are introduced, especially with respect to the modes of speech communication and general functionality. Examples of these auralizations are presented and are made available in an electronic address.
Convention Paper 6101

09:30 h

Z6-3 A Software Application for Estimation of Room Acoustic Behavior by Multisource Excitation—
Athanassios Fouloulis, Christos Goussios, Charalambos Dimoulas, George Kalliris, George Papanikolaou, Aristotle University of Thessaloniki, Thessaloniki, Greece

The purpose of this paper is the design and implementation of a software application for the estimation of the acoustic behavior of a rectangular room when a number of sound sources are activated. The room dimensions, the number, and positions of the sources can be selected. Materials are chosen from a library. Sound level distribution is calculated for a desired section of the room, using the image source method. Room modes are calculated for studying the standing waves. Reverberation times are also calculated using statistical formulas. Work has been done for the use of this software in nonrectangular rooms, based on different estimation methods.
Convention Paper 6102

09:30 h

Z6-4 The Acoustics of Ancient Greek Odea—*Christos Goussios, Christos Sevastiadis, George Kalliris, George Papanikolaou*, Aristotle University of Thessaloniki, Thessaloniki, Greece

Apart from the world famous ancient Greek theaters, whose acoustics often attract engineers, smaller closed amphitheatric halls—called odea (plural of the Greek word odeion)—had been constructed and used through the Greek and Roman periods. The acoustical characteristics for most of them and information concerning their location, use, history, and architectural elements are presented. An attempt was made for the modeling and estimation of their acoustics. Results of measurements also carried out are discussed.
Convention Paper 6103

09:30 h

Z6-5 On the Acoustics of Old Berlin Studios for Film and Radio—*Ernst-Jo. Völker*, Institute for Acoustic and Building Physics

A certain acoustical environment was always necessary when sound of adequate quality had to reach the audience. That applied both for natural sound and for sound reproduction via loudspeakers by using electrical or mechanical amplification. Long before microphones, amplifiers, and loudspeakers were developed and used; studios in the form of "Glasshouses" were built (e.g., in 1911 in the City of Babelsberg near Berlin), using bright and wide sunlight. For sound recordings, huge horns were connected with wax-plates or wax-cylinders. Sound had to be absorbed by curtains, carpets, and much plush, which was already well known since the first stereophonic transmission during the First Electrical Fair in Paris in 1879. Radio started in the twentieth century, in Berlin, with the

Eugin Reiß carbone microphone in an almost overdamped studio on October 29, 1923. Some years later a "Haus des Rundfunks" was opened with many studios of different use and quality including a concert hall. Film and radio went their own ways with multichannel reproduction or, for long time, only with mono transmission. Some acoustical aspects of the first studios will be described.
Convention Paper 6104

09:30 h

Z6-6 MPEG-7-Based Low-Level Descriptor Effectiveness in the Automatic Musical Sound Classification—*Piotr Szczuko, Piotr Dalka, Marcin Dabrowski, Bozena Kostek*, Gdansk University of Technology, Gdansk, Poland

The objective of this paper is to determine which of the MPEG-7 standard low-level sound descriptors are most significant in the process of automatic classification of musical instrument sounds. First, pitch detection is performed. Then, the parametrization stage of musical sounds based on descriptors contained in the MPEG-7 standard is carried out. Next, a thorough statistical analysis of the feature vectors obtained is performed. For the purpose of automatic classification two decision systems, based on artificial neural networks (ANNs) and rough sets, are used. Both decision systems are trained with feature vectors consisting mostly of parameters contained in the MPEG-7 standard, however, their content is reduced after statistical analysis. In addition, a comparison of results obtained by these decision systems with the results derived from the nearest neighbor algorithm is made.
Convention Paper 6105

09:30 h

Z6-7 Scale Degree Profiles from Audio Investigated with Machine Learning Techniques—*Hendrik Purwins*[1], *Benjamin Blankertz*[2], *Guido Dornhege*[2], *Klaus Obermayer*[1]
[1]Berlin University of Technology, Berlin, Germany
[2]Fraunhofer FIRST (IDA), Berlin, Germany;

In this paper we introduce and explore a method for extracting low dimensional features from digitized recordings of music performance. The so-called constant Q scale degree profiles are 12-dimensional vectors that reflect the prominence of the 12-scale degrees in respective analyzed parts of music. Here we study the type and amount of information that is captured in those profiles when calculated from whole short pieces of piano music. The analyzed data set includes pieces from Bach's Well-Tempered Clavier (WTC), part I and II, the sets of pre-

ludes that encompass a piece in every key by Chopin (op. 28), Alkan (op. 31), Scriabin (op. 11), Shostakovich (op. 34), and the fugues of Hindemith's "ludus tonalis" (one fugue for each pitch class, neither major nor minor). For the purpose of investigation we employ supervised and unsupervised machine learning techniques. In a supervised approach we investigated the ability of classifiers to recognize composers from profiles. As unsupervised methods we performed (1) cluster analysis which resulted in one major and one minor cluster, and (2) a visualization technique called Isomap which reveals in its 2-dimensional representation some additional structure apart from the major-minor duality. In summary it is astonishing how much information on a music piece is contained in the 12-dimensional profiles that can be calculated in a straight-forward manner from any digitized music recording.
Convention Paper 6106

09:30 h

Z6-8 Automatic Estimation of Reverberation Time—*José Vieira*, Universidade de Aveiro, Aveiro, Portugal

The correct estimation of the reverberation time of the room acoustics can be an important task for several systems such as sound localizers, hearing-aids, and telephony. These systems are affected by reverberation and need an estimate of this acoustic parameter in order to adapt the algorithms to different environments. This paper presents a method to estimate the reverberation time of a room without using test signals. From the captured signals in the room, the system is able to estimate the reverberation time without any prior knowledge of the sound sources or room geometry. The estimates are obtained from the "tails" of the sounds, and we use a run-length energy integral followed by an algorithm that estimates the decay of the sound energy.
Convention Paper 6107

10:00 h

Technical Committee Meeting on Network Audio Systems (TC Room 1, Hall 7.1b)

Session M **Monday, May 10** **12:30 h–15:30 h**
Room7.1b-2

LOUDSPEAKERS—PART 2

Chair: **Ilpo Martikainen**, Genelec Oy, Iisalmi, Finland

12:30 h

M-1 Performance Comparison of Graphic Equalization and Active Loudspeaker Room Response Controls—
Andrew Goldberg, Aki Mäkivirta, Genelec Oy, Iisalmi, Finland

We compare the room response controls available in active loudspeakers to a third-octave graphical equalizer. The room response controls are set using an automated optimization method presented in earlier AES publications. A third-octave ISO frequency constant-Q graphic equalizer is set to minimize the least squares deviation from linear? within the passband in a smoothed acoustical response. The resulting equalization performance of the two methods is compared using objective metrics, to show how these standard room response equalizing methods perform. For all loudspeaker models pooled together, the room response controls improve the RMS deviation from a linear response from 6.1 dB to 4.7 dB (improvement 22 percent), whereas graphic equalization improves the RMS deviation to 1.8 dB (improvement 70 percent). Both equalization techniques achieve a similar improvement in the broadband balance, which has been shown to affect a subjective lack of coloration in sound systems. The optimization time for a graphic equalizer is up to 48 times longer compared to that of active loudspeaker room response controls.
Convention Paper 6108

13:00 h

M-2 Spatially Consistent Reproduction of the Reverberant Sound Field—*Graeme Huon, Zeljko Velican*, HuonLabs, Victoria, Australia

A new apparatus for reproducing the reverberant field is described. A model is presented for accurately reproducing direct sound, early reflections, and the reverberant field. The requirements for spatially correct reverberant sound field reproduction are considered and some prior approaches reported on. The authors' two recently reported studies are reviewed and assessed against the requirements for sound reproduction, namely the Depth Render (DR) human acoustic perception model and its implications for direct sound reproduction and the Wave Focus

(WF) model for control of low-frequency room modes. The recent extension of WF for mode-controlled coverage of large audiences is also reviewed. Tests of the new reverberant field apparatus are reported for stand-alone, equidistant surround and DR configurations, both with and without wave focus for low frequencies. Patent applications apply.
Convention Paper 6109

13:30 h

M-3 The Beneficial Coupling of Cardioid Low-Frequency Sources to the Acoustics of Small Rooms—*Lampos Ferekidis*, *Uwe Kempe*, wvier, Lemgo, Germany

Most low-frequency sources radiate energy in an omnidirectional manner. This often leads to unsatisfying results regarding the reproduction of low frequencies in small listening rooms. The influence of different radiation characteristics is investigated concerning the reproduction of low frequencies in a sparsely modal environment. In this paper the room transfer function characteristic of a monopole, a dipole, and a cardioid are compared. The different room mode excitation mechanisms are explained using comparative measurements taken in a reverberation chamber. Furthermore the effect of a single reflective boundary on the low-frequency response is simulated. The cardioid turns out to be the more preferable low-frequency source for the three types investigated.
Convention Paper 6110

14:00 h

M-4 Polar Pattern and Energy Response of Transients in Multiway Loudspeakers—*Juha Backman*, Nokia, Espoo, Finland

The one-cycle time offset between the high-pass and low-pass sections typical to symmetrical constant-amplitude crossover networks implies that the polar pattern is controlled by a single driver (or driver group) during the onset and end of a sharp transient. This implies that the ratio of overall radiated energy to the input energy near the crossover frequency depends on the duration of the transient, which again affects the sound pressure in a reverberant field.
Convention Paper 6111

14:30 h

M-5 Near-Field Beam Forming in Security Relevant Work Spaces Using a Set of Linear Loudspeaker Arrays—*Roman Beigelbeck*[1], *Heinrich Pichler*[2]
[1]Vienna University of Technology, Vienna, Austria
[2]Consultant, Vienna, Austria

In security-relevant work spaces, such as air traffic control rooms, near-field beam forming in small spaces is an important task. In this paper a sound design based on a set of n-linear loudspeaker arrays where each consists of m-elliptic loudspeakers is investigated from a mathematical point of view. Based on these results, optimized array parameters are determined and useful approximations are developed. Three-dimensional near-field directional diagrams of the sound pressure in front of the arrays are shown to visualize the sound field. These diagrams are plotted and evaluated for different frequencies and distances of the field point, in addition to variations in the control signal phases and amplitudes. Finally, these theoretical values are compared with practical results.
Convention Paper 6112

15:00 h

M-6 A Multiple Regression Model for Predicting Loudspeaker Preference Using Objective Measurements: Part I—Listening Test Results—*Sean Olive*, Harman International Industries, Inc., Northridge, CA, USA

Part I of this paper presents the objective measurements and listening test results on 13 loudspeakers rated according to preference, spectral balance, and distortion. In Part II the data provides the framework for the development and verification of a multiple regression model that predicts listeners' preferences based on objective measurements. We review relevant predictive models and test one model currently used by Consumers Union (CU), a consumer product testing organization in the United States. There is no correlation between listeners' loudspeaker preference ratings and CU's predicted accuracy scores ($r = 0.05$; $p = .81$). As the CU model is based largely on the loudspeaker's 1/3-octave sound power response, we conclude that measured sound power, alone, cannot accurately predict its perceived sound quality.
Convention Paper 6113

Session Z7 **Monday, May 10** **12:30 h–14:00 h**
Corridor 7.1b

PAPERS

POSTERS: MULTICHANNEL SOUND AND WAVE FIELD SYNTHESIS

12:30 h

Z7-1 Coding Strategies and Quality Measure for Multichannel Audio—*Soledad Torres-Guijarro*[1], *Jon Ander Beracoechea-Álava*[1], *Isidoro Pérez-García*[2], *F. Javier Casajús-Quirós*[1]
[1]Universidad Politécnica de Madrid, Madrid, Spain
[2]European University of Madrid, Madrid, Spain

The Karhunen-Loeve Transform (KLT) has proven to be an efficient method of decorrelating multichannel signals prior to coding. Careful bit-rate distribution among decorrelated channels reduce the overall bit rate. In order to explore how bits are distributed in the coding process, a new quality measure of the reconstructed sound field is proposed; the binaural signal that the listener would obtain in a real environment is synthesized and evaluated by means of the standard Perceptual Audio Quality Measure (PEAQ). Results on codification via AAC with different kind of audio signals, bit allocations, and multichannel arrangements are reported.
Convention Paper 6114

12:30 h

Z7-2 An Improvement in Sound Quality of LFE Flattening Group Delay—*Shintaro Hosoi*[1], *Hiroyuki Hamada*[1], *Nobuo Kameyama*[2]
[1]Pioneer Corporation, Tokorozawa, Saitama, Japan
[2]NRP Ltd., Tokorozawa, Saitama, Japan

In this paper we raise the issue of bass reproduction of surround music, when using LFE. We show that this issue originates from the method of creating an LFE. Therefore, we propose the practicable method of “LFE phase sync," that improves the quality of bass by applying the proper amount of delay. The optimum delay is calculated for using this method for various cutoffs and order of filters. We introduce the manner in which this method can be used for actual recording projects, and mention the method for monitoring when an encoder is used.
Convention Paper 6115

12:30 h

Z7-3 High Spatial Resolution Multichannel Recording—
Arnaud Laborie, Rémy Bruno, Sébastien Montoya,
Trinnov Audio, Paris, France

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Multichannel recording is certainly one of the most important remaining issues concerning today's sound techniques. A good surround recording is extremely difficult to obtain because it must fulfill a number of conditions including envelopment feeling, accurate localization, and a large sweet spot without compromising the timbres. Advanced signal processing allows one to obtain directivities designed from panning laws that have been designed to optimally drive any multichannel layout. This paper presents the underlying concept of High Spatial Resolution, the spatial equivalent for High Fidelity, and points out why this is a key point to achieve high spatial quality. Actual performances of such a High Spatial Resolution 5.0 microphone featuring a small array of 8 omnidirectional capsules are fully simulated and measured.
Convention Paper 6116

12:30 h

Z7-4 Wave Field Synthesis: Mixing and Mastering Tools for Digital Audio Workstations—*Renato Pellegrini, Clemens Kuhn*, sonicEmotion AG, Dielsdorf Switzerland

Wave Field Synthesis (WFS) provides holographic sound reproduction for a large listening area. Fundamentals of WFS recording and reproduction techniques have been developed in the past few years; however there is a lack of intuitive tools for WFS mixing and mastering. In this paper the authors propose a WFS user interface compatible with available and accepted digital audio workstations. These WFS-plug-ins are based on a novel audio network technology. They open new possibilities for creative audio production in WFS.
Convention Paper 6117

12:30 h

Z7-5 Generation of Highly Immersive Atmospheres for Wave Field Synthesis Reproduction—*Andreas Wagner*[1,2], *Andreas Walther*[1,2], *Frank Melchior*[2], *Michael Strauss*[2]
[1]Technical University Ilmenau, Ilmenau, Germany
[2]Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Wave Field Synthesis (WFS) permits the reproduction of a sound field, which fills nearly the whole reproduction room with correct localization and spatial impression. This technology enables a correct spatial sound reproduction with a proper localization over a wide listening area. So far, this technique has been mainly used and demonstrated for music reproduction. Because of its properties, WFS is ideal for the creation of sound for motion picture or virtual

reality applications. In both cases the creation of highly immersive atmospheres is important to give the auditorium the illusion of being a part of the auditory scene. In this paper a new approach in designing immersive atmospheres (e.g., rain) using wave field synthesis reproduction is presented. New tools and techniques to control and generate these atmospheres have been developed and investigated in listening tests.
Convention Paper 6118

12:30 h

Z7-6 Efficient Active Listening Room Compensation for Wave Field Synthesis—*Sascha Spors, Herbert Buchner, Rudolf Rabenstein*, University of Erlangen-Nuremberg, Erlangen, Germany

Wave field synthesis is an auralization technique which allows control of the entire wave field within the entire listening area. However, reflections in the listening room interfere with the auralized wave field and may impair the spatial reproduction. Active listening room compensation aims at reducing these impairments by using the playback system. Due to the high number of playback channels used for wave field synthesis, the existing approaches to room compensation are not applicable. A novel approach to active room compensation overcomes these problems by a transformation from the space-time to the wave domain and application of wave-domain adaptive filtering.
Convention Paper 6119

12:30 h

Z7-7 Full-Duplex Systems for Sound Field Recording and Auralization Based on Wave Field Synthesis—*Herbert Buchner, Sascha Spors, Walter Kellermann*, University of Erlangen-Nuremberg, Erlangen, Germany

For high-quality multimedia communication systems such as telecollaboration or virtual reality applications, both multichannel sound reproduction and full duplex capability are highly desirable. Full 3-D sound spatialization over a large listening area is offered by wave field synthesis, where arrays of loudspeakers generate a prespecified sound field. However, before this new technique can be utilized for full-duplex systems with microphone arrays and loudspeaker arrays, an efficient solution to the problem of multichannel acoustic echo cancellation (MCAEC) has to be found in order to avoid acoustic feedback. This paper presents a novel approach that extends the current state of the art of MCAEC and transform domain adaptive filtering by reconciling the flexibility of adaptive filtering and the underlying physics of acoustic waves in a systematic and

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efficient way. Our new framework of wave-domain adaptive filtering (WDAF) explicitly takes into account the spatial dimensions of the closely spaced loudspeaker and microphone arrays. Experimental results with a 32-channel AEC verify the concept for both simulated and actually measured room acoustics.
Convention Paper 6120

12:30 h

Z7-8 Equalization of Wave Field Synthesis Systems—
Andreas Apel, Thomas Röder, Sandra Brix, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

Wave Field Synthesis allows the reproduction of arbitrary wave fields in a large listening area. The theoretical driving function for the loudspeaker states that a correction filter must be implemented to get a flat frequency response of the system. Practical implementations require an adaptation of the filter to the current source position. In the current paper measurements of frequency responses for different source positions are compared. Based on those measurements a method for a proper equalization of the system is proposed. Finally, results of listening tests are shown, which compare the quality of a position-dependent filtering with a position-independent filtering.
Convention Paper 6121

11:30 h

Technical Committee Meeting on Multichannel and Binaural Audio Technologies (TC Room 1, Hall 7.1b)

14:00 h

Technical Committee Meeting on Studio Practices and Production (TC Room 1, Hall 7.1b)

Session N **Monday, May 10** **13:30 h–16:30 h**
Room 7.1b-1

SIGNAL PROCESSING—PART 2

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

13:30 h

N-1 Effects of Jitter on AD/DA Conversion; Specification of Clock Jitter Performance—*Bruno Putzeys, Renaud de Saint Moulin*, Philips Digital System Labs, Heverlee, Belgium

The impact of clock jitter on AD/DA conversion performance is detailed for several conversion methods. Account is taken of the spectral distribution of both the jitter and of the converted waveform. The inadequacy of a single "picosecond" performance figure is shown, and the use of a dBc/sqrt(Hz) specification is proposed instead.
Convention Paper 6122

14:00 h

N-2 Nonuniform Sampling Theory in Audio Signal Processing—*Patrick Wolfe*[1], *Jamie Howarth*[2]
[1]University of Cambridge, Cambridge, UK
[2]Plangent Processes, Nantucket, MA, USA

The goal of most sampling schemes is to sample the analog signal of interest at a regular rate sufficiently high to ensure a perfect reconstruction principle in theory. Indeed, analysis and subsequent signal processing is almost always predicated on this requirement. However, the assumption of uniformly spaced samples is often invalidated in practice. Here, we describe nonuniform sampling theory, which provides a framework for the investigation and analysis of such cases. We review aspects of the theory and describe how it may be applied to practical problems of interest in audio signal processing, including those of wow and flutter in the analog domain as well as jitter in the digital domain.
Convention Paper 6123

14:30 h

N-3 Acoustic Positioning and Head Tracking Based on Binaural Signals—*Miikka Tikander, Aki Härmä, Matti Karjalainen*, Helsinki University of Technology, Espoo, Finland

Tracking a user's movement and orientation is essential for providing realistic mobile augmented reality audio (MARA)

services. For mobile use the tracking system needs to be light-weight, wearable, and wireless. Binaural microphones offer a convenient and practical solution for tracking user movement and orientation. These sensors can be easily integrated with portable headphones. In addition to tracking, the microphones also offer several possibilities to control the user's acoustic environment. This paper reviews the latest results in binaural head-tracking with known anchor sources and also discusses the case where there are no known anchor (reference) sources available. Some transducer issues are also discussed.
Convention Paper 6124

15:00 h

N-4 Feature Extractors for Music Information Retreival: Noise Robustness—*Adebunmi Paul-Taiwo, Mark Sandler, Mike Davies*, Queen Mary University of London, London, UK

The challenge in the field of music information retrieval is to discover a set of features that has minimal dimensionality and is also very robust to the variations in the channel and environment. This paper provides an overview of several feature extraction algorithms that have been used for music information retrieval; Mel Frequency Cepstral coefficient (MFCC), Linear Prediction coefficient (LPC), Perceptual Linear Prediction coefficient (PLP), and delta coefficient. This paper also emphasizes a biologically inspired feature extractor (The Human Factor Cepstral Coefficient) which was initially introduced in speech recognition. Its performance compares favorably with the other modeling algorithms. It also reports the findings of experiments that compare the effectiveness of these feature extractors, in the presence of noise in the context of a simple but complete music information retrieval system.
Convention Paper 6125

15:30 h

N-5 A System for Multitask Noisy Speech Enhancement —*Andrzej Czyzewski*[1], *Andrzej Kaczmarek*[1], *Jozef Kotus*[1], *Arkadiusz Pawlik*[2], *Andrzej Rypulak*[2], *Pawel Zwan*[1]
[1]Gdansk University of Technology, Gdansk, Poland
[2]Air Force Academy, Deblin, Poland

A general characteristic of the engineered speech signal registration and restoration system is presented in this paper. It contains a concise description of specific components of the system: the system being a set of advanced tools for registration, analysis, and reconstruction of speech existing in the form of computer software. The

tools included allow for prompt search of desired fragments of recordings and for the improvement of their quality through noise, distortion, and interference reduction. Brief information concerning selected speech reconstruction algorithms is presented also, the use of which allowed for an especially significant increase of processed speech comprehension.
Convention Paper 6126

16:00 h

N-6 Automatic Extraction of High-Level Music Descriptors from Acoustic Signals Using EDS—*Aymeric Zils, Francois Pachet*, Sony CSL, Paris, France

High-level music descriptors are key ingredients for music information retrieval systems. Although there is a long tradition in extracting information from acoustic signals, the field of music information extraction is largely heuristic in nature. We present here a heuristic-based generic approach for extracting automatically high-level music descriptors from acoustic signals. This approach is based on genetic programming, used to build relevant features as functions of mathematical and signal processing operators. The search for relevant features is guided by specialized heuristics that embody knowledge about the signal processing functions built by the system. Signal processing patterns are used in order to control the general processing methods. In addition, rewriting rules are introduced to simplify overly complex expressions, and a caching system further reduces the computing cost of each cycle. Finally, the features built by the system are combined into an optimized machine learning descriptor model, and an executable program is generated to compute the model on any audio signal. In this paper, we describe the overall system and compare its results against traditional approaches in musical feature extraction à la MPEG-7.
Convention Paper 6127

17:00 h

Technical Committee Meeting on Signal Processing (TC Room 1, Hall 7.1b)

18:00 h

Technical Committee Meeting on Coding of Audio Signals (TC Room 1, Hall 7.1b)

Session O **Monday, May 10** **15:30 h–18:00 h**
Room 7.1b-2

PAPERS

MICROPHONES

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

15:30 h

O-1 An Improved Method of Noise Cancellation—*David Herman*[1], *Dudley Haestler*[1], *Simon Busbridge*[2]
[1]AudioGravity Ltd., Hove, UK
[2]University of Brighton, Brighton, East Sussex, UK

The effectiveness of conventional noise cancellation techniques is limited by tolerances between the signal and noise channels. A system is described in which the ambient noise error signal is fed back for further cancellation (Advanced Ambient Noise Rejection Technology, ANRT). Small physically displaced microphones differentiate near-field signals from high-level ambient noise. Band limiting filters further reduce high-frequency phase distortion. The effectiveness is increased such that an unintelligible signal produced by normal speech can result in an SNR improvement of 40 dB in an ambient noise field of 98 dBA. The technology can be integrated into a single, small, low-power CMOS analog integrated circuit; it is also ideally suited for MEMS (Si-Mic).
Convention Paper 6128

16:00 h

O-2 Close-Talking Autodirective Dual Microphone—
Alexander Goldin, Alango Ltd., Haifa, Israel

The paper presents close-talking mode of Autodirective Dual Microphone (ADM) technology developed by Alango Ltd. ADM is an adaptive beamforming technology having two operational modes. In far-talk mode ADM provides optimal directivity for every frequency region such that sounds coming from the back plane are cancelled. In close talk mode all sounds originating outside a close proximity to the microphone are (theoretically) completely cancelled. ADM fast adaptation time leads to excellent noise cancellation in changing noisy environments. ADM technology has a low demand for placing, matching, and distance between individual sensors. This simplifies its integration into mobile and other devices. ADM operational mode is defined by DSP algorithm, easily switching according to situation.
Convention Paper 6129

16:30 h

O-3 About a Digital RF-Condenser Microphone—*Roland Müller; Peter Holstein*, SINUS Messtechnik GmbH, Leipzig, Germany

Digital microphones are commonly based on an LF-condenser with an ADC in the same housing. However, this concept has some disadvantages, such as the inherent problems of LF-condenser microphones with respect to the influence of humidity on sensitivity, distortion, and low cut-off frequency. Therefore, another approach for digital microphones is proposed, whereby the capacity of the microphone capsule controls the frequency of an LC-type generator. The resulting nonlinear distortion is of second order and similar to those of classical microphones with vacuum tube preamplifiers. A negative capacitance can be added to reduce the distortion. There are several ways to implement demodulation and digitalization; simulations show that a sufficient dynamic range can be achieved by using a special kind of delta-sigma-FM-discriminator.
Convention Paper 6130

17:00 h

O-4 Modern Acoustic and Electronic Design of Studio Condenser Microphones—*Stephan Peus*, Georg Neumann GmbH, Berlin, Germany

Condenser microphones have been used for more than 70 years in professional audio recording applications due to their good frequency response, extended frequency range, and wide dynamic range. The basic design of studio microphone capsules today dates back several decades. Some capsules have been in production unchanged for 50 years or more. Nevertheless, the technical performance of microphones has been improved step by step by continued refinement of the associated electronic circuitry (e.g., tubes versus semiconductors, FET technology improvements, circuitry design aspects, etc.). Not until a few years ago did the quality of the electronics finally match that of the capsule in terms of self-noise level and dynamic range. However, the capsule design has also been improved by making use of technological advances and modern materials. Studio microphones developed recently for high-resolution applications are capable of sensitivity corresponding to the noise level of air particles hitting the diaphragm surface due to thermal molecular movement, and at the same time have a dynamic range of 130 dB or more. This is true for both microphones using analog electronics and microphones using the most recent ADC technology. This paper gives an overview of recent

advances in the acoustic and electronic design of studio condenser microphones.
Convention Paper 6131

17:30 h

O-5 Fiber-Coupled Optical Microphones—*Peter Schreiber*[1], *Sergey Kudaev*[1], *Vladimir Gorelik*[2], *Jürgen Peissig*[2]
[1]Fraunhofer Institute for Applied Optics and Precision Engineering, Jena, Germany
[2]Sennheiser Electronic GmbH & Co. KG, Wedemark, Germany

Motivated by the advantages of optical sensors, like immunity with respect to EMI/RFI and electrically isolated realization, today's fiber- and micro-optics technology enables the manufacturing of sensitive optical microphones. In the first part of this paper a short review of applicable sensing principles is given and pros and cons for realization are discussed. In the second part, design, manufacturing, and characterization for different fiber-coupled optical microphones employing optical sampling of a membrane are presented.
Convention Paper 6132

18:00 h

Technical Committee Meeting on Microphones and Applications (TC Room 2, Hall 7.1b)

Session Z8 **Monday, May 10** **15:30 h–17:00 h**
Corridor 7.1b

POSTERS: AUDIO RECORDING AND REPRODUCTION AND ARCHIVING AND CONTENT MANAGEMENT

15:30 h

Z8-1 The History of the Tonmeister Recording Technique in Russia—*Pavel Ignatov*, St. Petersburg, Russia

The history of sound recording in Russia dates back to the end of the 19th century. The creation of the first sound recording studios began in the 1920s and 1930s. Although the technical facilities that were used seemed to be quite primitive, the work of such outstanding tonmeisters as Khustov, Grossman, and Gakhlin made outstanding recordings of classical music and live concerts. The main feature of the second half of the 20th century (1950-1980s) was the important development of TV, RB, and recording studios (292 large television centers and radio studios had been built by the 1980s). Today's new digital technologies and surround sound systems are used in tonmeister practice. Such masters as Shugal, Vinogradov, Khondrashin, Dinov, and many others are creating new methods of digital sound recording. The main periods of the development of tonmeister technology are investigated in this paper.
Convention Paper 6133

15:30 h

Z8-2 Optimization of Microphone Setup for Symphonic Orchestra Recordings During Rehearsal—*Witold Mickiewicz*, Technical University of Szczecin, Szczecin, Poland

Many symphonic orchestras use a nonoptimal 2-microphone setup during rehearsal recordings. These recordings are used for archiving purposes and to evaluate and improve artistic skills of a whole orchestra and its members. For that purpose, good resolution of stereo image during reproduction is needed. The process of choosing the right microphone setup can be based on the geometric parameters of the orchestra podium and acoustical properties of a rehearsal hall. Some theoretical considerations presented in this paper are supported by real recordings made in the hall of the Philharmonic of Szczecin, Poland, and listening tests made by orchestra members.
Convention Paper 6134

15:30 h

Z8-3 3-D Audio Acquisition and Reproduction System Using Multiple Microphones on a Rigid Sphere—*Taejin*

P A P E R S

Lee[1], *Daeyoung Jang*[1], *Kyeongok Kang*[1], *Jinwoong Kim*[1], *Daegwon Jeong*[2], *Hareo Hamada*[3]
[1]Electronics and Telecommunications Research Institute, Daejeon, Korea;
[2]Hankuk Aviation University, Goyang-city, Korea
[3]Tokyo Denki University, Tokyo, Japan

Generally, a dummy-head microphone is used for 3-D audio acquisition. Because of its human-like shape, we can get good spatial images. However, its shape and size are also the restriction of its public use. In this paper we propose a 3-D audio acquisition and reproduction method using multiple microphones on a rigid sphere. We place the 5 microphones on a rigid sphere's special points and generate various audio signals for the reproduction of headphone, stereo, stereo dipole, 4-channel and 5-channel reproduction environments. Subjective reproduction experiments of 4-channel and 5-channel loudspeaker configurations show that the front/back confusion, which is a common limitation of a 3-D audio reproduction system using dummy-head microphone, can be reduced dramatically.
Convention Paper 6135

15:30 h

Z8-4 BeatBank: An MPEG-7 Compliant Query by Tapping System—*Gunnar Eisenberg, Jan-Mark Batke, Thomas Sikora*, Technical University of Berlin, Berlin, Germany

A Query by Tapping System is a multimedia database containing rhythmic metadata descriptions of songs. This paper presents a Query by Tapping system called BeatBank, which allows the formulation of queries by tapping the melody line's rhythm of a song requested on a MIDI keyboard or an e-drum. The query entered is converted into an MPEG-7 compliant representation. The actual search process takes only rhythmic aspects of the melodies into account by comparing the values of the MPEG-7 Beat Description Scheme. An efficiently computable similarity measure is presented, which enables the comparison of two database entries. This system works in real-time and computes the search process online. It computes and presents a new search result list after every tap made by the user.
Convention Paper 6136

15:30h

Z8-5 A Query by Humming System Using MPEG-7 Descriptors—*Jan-Mark Batke, Gunnar Eisenberg, Philipp Weishaupt, Thomas Sikora*, Technical University of Berlin, Berlin, Germany

Query by Humming (QBH) is a method for searching in a

PAPERS

multimedia database system containing metadata descriptions of songs. The database can be searched by hummed queries; this means that a user can hum a melody into a microphone that is connected to the computer hosting the system. The QBH system searches the database for songs that are similar to the input query and presents the result to the user as a list of matching songs. This paper presents a modular QBH system using MPEG-7 descriptors in all processing stages. Due to the modular design all components can easily be substituted. The system is evaluated by changing parameters defined by the MPEG-7 descriptors.
Convention Paper 6137

15:30 h

Z8-6 Music Archive Metadata Processing Based on Flow Graphs—*Bozena Kostek, Andrzej Czyzewski,* Gdansk University of Technology, Gdansk, Poland

The paper addresses the capabilities that should be expected from intelligent Web search tools in order to respond properly to a user's music information retrieval needs. An advanced query algorithm was engineered employing a concept of inference rule derivation from flow graphs with regard to semantic data processing. This concept, introduced recently by Pawlak, is used for mining knowledge in databases. The created database searching engine utilizes knowledge acquired in advance and stored in flow graphs in order to enable searching in musical repositories. Results obtained show that employing the implemented method the resulting search matches are ranked optimally, thus metadata related to recorded sound can be retrieved efficiently with the use of this algorithm.
Convention Paper 6138

15:30 h

Z8-7 Nearest-Neighbor Generic Sound Classification with a WordNet-Based Taxonomy—*Pedro Cano, Markus Koppenberger, Sylvain Le Groux, Julien Ricard, Nicolas Wack, Perfecto Herrera*, Universitat Pompeu Fabra, Barcelona, Spain

Audio classification methods work well when fine-tuned to reduced domains, such as musical instrument classification or simplified sound effects taxonomies. Classification methods cannot currently offer the detail needed in general sound recognition. A real-world-sound recognition tool would require thousands of classifiers, each specialized in distinguishing little details and a taxonomy that represents the real world. We describe the use of WordNet, a semantic network that organizes real world knowledge as the

taxonomy backbone. In order to overcome the huge number of classifiers to distinguish an ever growing number of sounds, the recognition engine uses a nearest-neighbor classifier with a database of isolated sounds unambiguously linked to WordNet concepts.
Convention Paper 6139

17:00 h

Technical Committee Meeting on Transmission and Broadcasting (TC Room 2, Hall 7.1b)

Session P **Tuesday, May 11** **09:30 h–12:30 h**
Room 7.1b-1

PAPERS

PSYCHOACOUSTICS, PERCEPTION, AND LISTENING TESTS

Chair: **Søren Bech**, Bang & Olufsen a/s, Struer, Denmark

09:30 h

P-1 Quality Adviser: A Multichannel Audio Quality Expert System—*Slawomir Zielinski*[1], *Francis Rumsey*[1], *Rafael Kassier*[1], *Søren Bech*[2]
[1]University of Surrey, Guildford, Surrey, UK
[2]Bang & Olufsen a/s, Streuer, Denmark

The basic audio quality of 5.1 multichannel audio reproduction was evaluated under different technical conditions. The obtained database of subjective responses was used to develop a multichannel audio quality expert system. There are three aims of this development: (1) to predict audio quality as a function of individual channel bandwidth, (2) to predict audio quality as a function of a downmix algorithm, (3) to predict the optimum technical trade-off between these factors for a given overall bandwidth of a multichannel audio signal. Obtained results indicate a close correspondence between the predicted and actual quality ratings. It is intended that the final version of the Quality Adviser will be suitable as a decision making aid for broadcasters and codec designers.
Convention Paper 6140

10:00 h

P-2 Subjective Evaluation of Virtual Home Theater Sound Systems for Loudspeakers and Headphones—*Gaëtan Lorho, Nick Zacharov*, Nokia Research Center, Tampere, Finland

A subjective evaluation of Virtual Home Theater systems (VHT) for loudspeaker and headphone reproduction is presented in this paper. Several algorithms for loudspeakers and headphones were selected and applied to six different multichannel audio programs. A subjective experiment was performed for each configuration using screened listeners to assess the performance of these VHT algorithms in terms of overall sound reproduction quality. A paired comparison method was chosen, with the discrete 5-channel reproduction (3/2) system as a reference in the loudspeaker test, and the stereo downmix of the 5-channel material in the headphone test. The stereo downmix was also compared to the 5-channel reference in

the case of the loudspeaker reproduction. The experimental design and the detailed analysis of results are presented in this paper.
Convention Paper 6141

10:30 h

P-3 Elicitation and Grading of Subjective Attributes of 2-Channel Phantom Images—*Hyun-Kook Lee, Francis Rumsey*, University of Surrey, Guildford, Surrey, UK

The subjective attributes of 2-channel phantom images of transient piano, continuous trumpet, and male speech sources were elicited using pair-wise comparison between reference mono images and their phantom images. The attributes elicited included image focus, image width, image distance, brightness, hardness, and fullness. The effect of interchannel time and intensity differences on the perceived difference between the real image and its phantom image was investigated for each sound source with respect to the elicited subjective attributes. Results show that the type of panning method (pure time, pure intensity, and combination of the two) had a statistically significant effect on image focus and image width attributes. It was also found that the type of sound source had a significant effect on all the attributes.
Convention Paper 6142

11:00 h

P-4 Loudness Assessment of Music and Speech—*Esben Skovenborg*[1,3], *René Quesnel*[2], *Søren H. Nielsen*[1]
[1]TC Electronic A/S, Risskov, Denmark
[2]McGill University, Montreal, Quebec, Canada
[3]University of Aarhus, Aarhus, Denmark

An experiment was performed to investigate the assessment of loudness of music and speech using a general linear model. Eight expert listeners participated in the experiment. The method of adjustment was used for loudness matching of stimuli. Both stimuli of each pair was selected from a collection of 147 homogeneous audio segments including representative samples of speech, jazz, rock/pop, and classical music, together with pink noise and a 1-kHz tone. For each segment, a reliable estimate of the loudness level was obtained from the model. Both the uncertainty and the subjectivity factors were shown to depend on the category of the stimuli. An alternative categorization based on four MPEG-7 audio descriptors was also used for the analysis.
Convention Paper 6143

11:30 h

P-5 Imperfections at Low Frequencies—How Audible or Annoying Are They?—*Tomas Salava*, ETOS acoustics, Prague, Czech Republic

This paper deals with some open problems of low-frequency sound reproduction, particularly in medium and small listening rooms. First, the basic facts concerning sound fields and transfer functions in bounded spaces are briefly recalled. Specifics of sound quality perception at low frequencies are then outlined. Opinion differences in this field are discussed too. Strong influence of the test signals properties is stressed, and using both musical, and artificial test signals for low-frequency listening tests is recommended. Several examples of different artificial low-frequency test signals are described and compared with musical signals.
Convention Paper 6144

12:00 h

P-6 New Intrusive Method for the Objective Quality Evaluation of Acoustic Noise Suppression in Mobile Communications—*Juha Salmela, Ville-Veikko Mattila*, Nokia Research Center, Tampere Finland

A new intrusive method, combining several independent objective metrics, has been developed for the evaluation of the quality of acoustic noise suppression in mobile communications. Extensive subjective data, including simulations of several noise suppression solutions in various noise environments, was gathered to serve as the benchmark for the metrics. Partial least-square regression and full cross-validation were used to establish the applicability of 26 metrics, which were making use of different measurement procedures, to predict the perceived quality. A Phase IV, vector-based preference model was optimized to predict quality with a correlation of 0.95, resulting in an average prediction error of 8 percent. Different measurement procedures appeared to contribute to a similar extent to the prediction ability of the optimized model.
Convention Paper 6145

PAPERS

Session Q **Tuesday, May 11** **09:30 h–12:00 h**
Room 7.1b-2

AUDIO RECORDING AND REPRODUCTION AND HIGH-RESOLUTION AUDIO

Chair: **Malcolm Hawksford**, University of Essex, UK

09:30 h

Q-1 Respecting the Sound—From Aural Event to Ear Stimulus—*George Brock-Nannestad*, Patent Tactics, Gentofte, Denmark

Graphical or real-time interactive analysis of recorded sound occurred at least 20 years before the invention of reproducible sound in 1877. Scientifically reproduced sound quickly found its way into phonetics and musicology. Early commercial sound recording for entertainment retained an aura of reproduction of a real sound event and prescribed certain calibration features. After commercial success was ensured around 1913, manipulation techniques were developed and refined. The later analog years demonstrated imaginative thinking that came to a climax when fast digital technology enabled satisfactory signals that only contained what the ear requires and no more. The dissociation from the total real sound was complete. This paper provides a balanced, well-documented historical overview of the techniques and their consequences.
Convention Paper 6146

10:00 h

Q-2 A New Approach to Effective Dither in Delta-Sigma Modulation Systems—*James Angus*, The University of Salford, Greater Manchester, UK

This paper presents a new approach to dither in Sigma-Delta Modulation (SDM) systems. In particular it clarifies the position of the overload point in 1-bit SDM systems and presents several overload control methods with comparisons of their efficacy. It then goes on to examine the problem of applying dither to 1-bit systems and describes a new approach to applying high levels of dither. It presents results, which show that such dither can be effective in SDM systems.
Convention Paper 6147

10:30 h

Q-3 Ultra High-Resolution Audio Formats for Mastering Applications—*Malcolm Hawksford*, University of Essex, Essex, UK

To process audio signals prior to DSD and LPCM delivery, an audio data format is required that possesses a resolu-

tion substantially greater than the final release form. A number of strategies are presented capable of enhanced resolution. Techniques using the step-back algorithm are extended to include a multilevel quantizer but where the amplitude range is finite. An earlier scheme based upon multilevel SDM and multistage lossless differential coding is enhanced by incorporating more aggressive noise shaping implemented by means of parametric noise shaping previously used for binary SDM.
Convention Paper 6148

11:00 h

Q-4 Voided Space-Charge Electrets—Piezoelectric Transducer Materials for Electro-Acoustic Applications—*Michael Wegener, Steffen Bergweiler, Werner Wirges, Andreas Pucher, Reimund Gerhard-Multhaupt*, University of Potsdam, Potsdam, Germany

Voided space-charge electrets, such as cellular polypropylene, have recently been developed as piezoelectric materials that exhibit strong electromechanical thickness oscillations corresponding to high piezoelectric coefficients of around 500 pC/N and very good acoustical matching to air (low density of typically around 0.5 g/cm3 and low sound speed). Here, we discuss different aspects of the manufacture and the applicability of cellular polypropylene films as transducer materials at high frequencies and for ultrasound. The frequency response up to 90 kHz and the directivity patterns for several transducer geometries were investigated. Second- and third- order harmonic distortions and the power consumption of cellular polypropylene films in acoustic transducers are also described. Our results demonstrate that the relatively new ferroelectret films are very attractive for a range of device applications.
Convention Paper 6149

11:30 h

Q-5 Wind & Weather—*Martin Schneider*, Georg Neumann GmbH, Berlin, Germany

Microphones are used in all environments, especially for outdoor locations, but also in studio surroundings, wind and humidity characteristics of microphones and their relevant accessories are of interest. This paper presents acoustic and noise measurements plus audio examples of different types of microphones under climatically adverse circumstances with diverse protective accessories like foam windshields, wind baskets, etc. Application guidelines for recording engineers are deduced.
Convention Paper 6150

PAPERS

Session Z9 **Tuesday, May 11** **09:30 h–11:00 h**
Corridor 7.1b

POSTERS: LOUDSPEAKERS AND MICROPHONES

09:30 h

Z9-1 Frequency Domain Experiences in Loudspeaker's Suspensions—*Fernando Bolaños*, Acústica Beyma S.A., Valencia, Spain

This paper proposes to individually analyze each component of a loudspeaker, specifically the diaphragm-surround set. Experiments were performed on low- and medium-amplitude displacement ranges. The paper uses traditional experimental methods in seeking the surround and diaphragm's spectral signatures in the main eigen-value region. Our method consisted in exciting the diaphragm-surround set by a reluctance transducer that was fed by an electric impulse. We then analyzed its response with an Eddy Current Displacement Transducer in the Frequency Domain. The most typical experimental spectral signatures of the nonlinear systems in free response were reviewed. This paper presents the results that were obtained after examining six samples, finding only one sample completely free of nonlinearities.
Convention Paper 6151

09:30 h

Z9-2 Nonuniform Voice-Coil Winding for Electrodynamic Loudspeaker—*Victor Mazin, Yong-Sang Lee*, Samsung BlueTek Co. Ltd., Suwon City, Korea

In electrodynamic loudspeakers the force factor Bl is an irregular and asymmetrical function of voice-coil displacement. This results in diverse distortion during voice-coil oscillation. In this paper a method of artifact reduction is suggested. This method is based on application of nonuniform voice-coil winding, i.e., number of layers varies along the vice-coil axis. The voice coil proposed allows a more regular and symmetrical B1 factor than a conventional voice coil. Theoretical background of the method is given. Effects of the nonuniform voice coil on loudspeaker performance have been investigated using the Klippel Distortion Analyzer.
Convention Paper 6152

09:30 h

Z9-3 An Active Biquadratic Filter for Equalizing Overdamped Loudspeakers—*Neville Thiele*, Consultant, Epping, New South Wales, Australia

When a bridged-T network is inserted into the feedback path of a voltage follower, it can produce an inexpensive biquadratic filter whose transfer function has first-order coefficients as low as 2.5 (Q = 0.4), often approaching 2 (Q = 0.5), in the numerator when those in the denominator lie in the very useful range between 0.5 and 2. Among its applications, it is peculiarly suited to equalizing "over-damped" loudspeakers, i.e., with exceptionally low QT's, that are typical of robust, sensitive drivers with large magnets. The wide range of applications is possible through selection of the more suitable of the two possible configurations of a bridged-T network described as CRRC or RCCR. The work is the subject of intellectual property claims.
Convention Paper 6153

09:30 h

Z9-4 Radiation of an Enclosed Loudspeaker in a Large Baffle: Loudspeaker Simulation Model—*Elena Prokofieva*, Linn Products Ltd., Glasgow, Scotland, UK

A theoretical step-by-step investigation of the conventional loudspeaker, placed into a sealed cabinet and then installed within a rigid wall has been conducted. The loudspeaker diaphragm was simulated by a rigid circle piston and then by a number of concentric rings inserted into a large but finite-sized baffle and enclosure. The acoustic pressure and dynamic displacement expressions were formulated using a quasi-dynamic approach to loading force representation. This simulation allows for the withdrawal of some standard assumptions commonly used in the traditional theory of plates. A block-scheme of a proposed computer simulation using the developed quasi-dynamic model is also presented.
Convention Paper 6154

09:30 h

Z9-5 Practical Considerations for Integrating Switch Mode Audio Amplifiers and Loudspeakers for a Higher Power Efficiency—*Søren Poulsen, Michael A. E. Andersen*, Technical University of Denmark, Lingby, Denmark

An integration of electrodynamic loudspeakers and switch mode amplifiers has earlier been proposed in Karsten Nielsen, Lars Michael Fenger, "The Active Pulse Modulated Transducer (AT), A Novel Audio Power Conversion System Architecture," AES 115th Convention Paper, October 2003. The work presented in this paper is related to the practical aspects of integration of switch mode audio amplifiers and electro dynamic loudspeak-

ers, using the speaker's voice coil as output filter, and the magnetic structure as heat sink for the amplifier.
Convention Paper 6155

09:30 h

Z9-6 Sound Radiation from a Dual Microflim Piezoelectric Loudspeaker in Free Space—*Tim Mellow*, Nokia Product Technology Platforms, Farnborough, UK

Radiation characteristics of a concept loudspeaker are calculated. It comprises two closely-spaced stretched piezoelectric membranes pushed apart by a pressurized gas. A drive voltage applied across conductive coatings on both membranes causes their tensions to vary in opposite phase. Consequently, the membranes are displaced in the same direction. Driven by a class D amplifier, this transducer offers higher efficiency than conventional moving coil technology but with the smooth response and light weight of electrostatic devices. However, the voltage requirement is lower and the potential SPL higher than the latter. Also, if the conductive coatings were transparent, there is the tantalizing possibility of combining it with a display. The only remaining question is whether it can be manufactured economically.
Convention Paper 6156

09:30 h

Z9-7 Sound Source Design in the Very Low-Frequency Domain—*Guillaume Pellerin, Jean-Dominique Polack, Jean-Pierre Morkerken,* Laboratoire d'Acoustique Musicale, Paris, France

Whereas the aerodynamic effects take a significant place in the behavior of sound sources in the low-frequency domain and for signals containing a high specific energy, new complex fluid parameters have to be implemented to take into account possible causes of sound distortion such as the stalling phenomenon in the boundary layer around the mechanical structure. For the design of vented boxes, we show that the choice of a nozzle profile for the resonator ensures a better dynamical stability of the airflow and thus authorizes extreme low cutoff frequencies in "dipole" configurations. We also describe some experimental and computed results based on fluid FEM about the radiating output for this kind of source under 40 Hz.
Convention Paper 6157

09:30 h

Z9-8 Microphone Response in a Closed-Loop System —*Michael Pincus*, Acentech, Inc., Cambridge, MA, USA

A closed-loop audio system can be defined as one in which the loudspeaker is in the same space as the microphone. As such, some sound from the loudspeaker will mix with the source creating an interference pattern. The interference is dependent on the path length from the loudspeaker back to the microphone, the amplitude of the interfering signal, and the latency of the forward-fed signal. This paper investigates this interference and its effect on the output response of the microphone.
Convention Paper 6158

09:30 h

Z9-9 Space Characteristics of Directed Single Gradient Microphones—*Emil Milanov, Elena Milanova*, NEC, Sophia, Bulgaria

In this paper we examine a single gradient microphone with two acoustical entrances. A formula is proposed for defining the space characteristics of the microphone in its whole sound frequency range. The defined formulas are valid when the microphone is in a sphere sound wave and in a plane sound wave. We also explain the reasons that lead to the change of the space and frequency characteristics in the area of the high frequencies when no diffraction events are present.
Convention Paper 6159

09:30 h

Z9-10 Performance Study of a Compact 2-Sensor Noise Canceling System—*Kok Soon Phua, Jian Feng Chen, Louis Shue, Han Wu Sun*, Institute for InfoComm Research, Singapore

In this paper we propose a compact directional noise-canceling device, which consists of a differential microphone formed by two omnidirectional microphones connected in an end-fire orientation. By making use of adaptive beamforming for improved directionality, and spectral shaping, a form of nonlinear speech enhancement, the proposed device is positioned to tackle noise found in real environments, which is typically a mixture of directional, stationary, and nonstationary interferences. Performance evaluation of our real-time implementation is based on the following criteria: (1) directionality, (2) distortions, and (3) speech quality as measured by the Mean-Opinion-Score (MOS), through subjective listening tests and using the ITU-T P.862 Perceptual Evaluation of Speech Quality tool. Our experimental results indicate an average interference suppression of as much as 22 dB, and consistent improvement in speech quality.
Convention Paper 6160

PAPERS

Session R **Tuesday, May 11** **13:00 h–15:30 h**
Room 7.1b-1

INSTRUMENTATION AND MEASUREMENT

Session Chair: **Ian Dennis**, Prism Sound, Cambridge, UK

13:00 h

R-1 Evaluation of Objective Loudness Meters—*Gilbert Soulodre*, Communications Research Centre, Ottawa, Ontario, Canada

There are many applications where it is desirable to objectively measure the perceived loudness of typical audio signals. The ITU-R is investigating suitable objective measures (meters) that would allow the perceived loudness of various program materials to be equalized for broadcast applications. Ten objective loudness meters were submitted for formal evaluation by several private companies and research organizations. The loudness meters were evaluated for their ability to predict the results of an extensive database derived from a series of formal subjective tests conducted at five test sites around the world. The performance of the various loudness meters is compared and rated using several newly proposed metrics. Several basic objective loudness measures were also evaluated.
Convention Paper 6161

13:30 h

R-2 Simulation of the IEC 60711 Occluded Ear Simulator —*Søren Jønsson, Bin Liu, Andreas Schuhmacher, Lars Nielsen*, Brüel & Kjaer, Skodsborgvej, Denmark

Ear simulators are standardized devices used for calibration of, e.g., earphones and telecommunications equipment. In this paper the ear simulator B&K Type 4157 is investigated using a combined boundary/finite element model (BEM/FEM) of the air inside. Traditionally lumped parameter models have been used to create an electrical equivalent diagram for simulating acoustic impedances. However, these lumped parameter models have some built-in limitations and may not be valid for higher frequencies where the acoustic wavelength is in the range of the ear simulator dimensions. A more accurate acoustic model can be derived using well-established techniques like BEM and FEM. Here we present a combined BEM/FEM model, taking into account the thermo-viscous effects which are shown to be required for obtaining realistic results. Comparisons between simulation and measurements are given.
Convention Paper 6162

PAPERS

14:00 h

R-3 High-Performance Wideband Ultrasonic "Sell"-Transducer—*Jürgen Peissig, Vladimir Gorelik, Rainer Wiggers*, Sennheiser Electronic, Wedemark, Germany

The ultrasonic (US) transducer based on Sell's principle is well known to work invertibly as microphone and loudspeaker with a broadband frequency response. US transducers are used for movement and distance sensors, flowmeters, and in parametric transducers where it is important to have a high US sound level in air and good directivity. Driven by these applications we developed several versions of Sell transducers with optimized backplate structures for high sound pressure levels, minimum loss due to the membrane suspension, optimal drive of the membrane surface, and high directivity. Different membrane materials and vent openings result in different frequency responses. The transducer design, its acoustical performance, and the applications will be discussed.
Convention Paper 6163

14:30 h

R-4 Enhancements for Loose Particle Detection in Loudspeakers—*Pascal Brunet, Steve Temme*, Listen, Inc., Boston, MA, USA

During loudspeaker production, particles may become trapped in the loudspeaker motor and voice coil vicinity, resulting in a distinctive defect that is easily heard but difficult to detect by traditional test and measurements. We found that a sine sweep stimulus followed by a high pass filter and RMS envelope analysis efficiently detected loose particles and rub-and-buzz defects. The remaining problem is how to reduce the effect of background noise and get more reliable results. Statistical descriptors such as Crest Factor, Skewness, and Kurtosis are first investigated. Experimental results are given and the different tools are compared. New enhancements are described that effectively increase the overall immunity to background noise and discrimination of the method.
Convention Paper 6164

15:00 h

R-5 Merging Room-Acoustic and Electro-Acoustic Measurement Methods—*Wolfgang Ahnert, Stefan Feistel, Waldemar Richert*, Software Design Ahnert GmbH, Berlin, Germany

Today various acoustic measurement methods are used to investigate rooms or devices. For room-acoustic measurements MLS routines are often applied to obtain the

detailed data according to ISO standard 3382. Instead of MLS, nowadays the dual-channel FFT method based on a sweep stimulus is also commonly accepted. On the other hand, excitation by continuous noise or shot noise is used to obtain a good overview in a short time. For loudspeaker data acquisition or commissioning tests in noisy environments a TDS sweep measurement is performed to achieve results of high accuracy. Here a new measurement tool will be presented, incorporating all of these widely known methods. The advantages and disadvantages as well as the limitations will be discussed for each technique by means of specific examples and measuring applications. A detailed comparison will be provided and recommendations for the practical use under selected acoustic environmental conditions will be given.
Convention Paper 6165

Session S **Tuesday, May 11** **13:00 h–16:00 h**
Room 7.1b-2

ROOM AND ARCHITECTURAL ACOUSTICS AND SOUND REINFORCEMENT

Chair: **Ernst-Joachim Völker**, Institute for Acoustics and Building Physics, Oberursel and Zweihausen, Germany

13:00 h

S-1 Sound Conditioning in Open-Plan Offices—40 Years under Stress?—*Wolfgang Teuber, Ernst-Joachim Völker*, Institute for Acoustics and Building Physics, Oberursel and Zweihausen, Germany

Masking effects are well known and are increasingly used to cover disturbing noise. Data reduction cuts out useless signals that are not audible. In an office environment masking means privacy. But masking sound must follow strict rules. The Acoustical Field of Confidence describes the parameters, such as interfering noise, distance to next working places, intelligibility of speech, acoustical conditions, and the level of masking noise, e.g., around 45 dB(A). "The Steps of Privacy" include the different types of office work. The masking sound must be of special shape to fulfill the purposes, above all not to disturb. The paper deals with open-plane offices that have had a constant background noise for the past 40 years. Measurements have been carried out once a year to check the levels and acoustical properties. Since the beginning of testing, there have been no complaints and no stress for the people working there.
Paper Presented but No Convention Paper Available

13:30 h

S-2 Finite-Difference Time-Domain Acoustic Analysis of Fibrous Sound-Absorbing Materials—*José Escolano, Basilio Pueo, Sergio Bleda*, University of Alicante, Alicante, Spain

A Finite-Difference Time-Domain (FDTD) method was successfully developed to model electromagnetic systems. Since acoustics and electromagnetism share certain undulatory properties, a natural adaptation of this technique has been developed as well. Several acoustics problems require the use of fibrous tangles to attenuate the propagation speed of sound waves, such as room acoustics. Notwithstanding, although free air acoustic propagation is known, FDTD technique is not developed yet to model fibrous materials. To characterize this behav-

ior only a few and measurable set of parameters must be considered. In this paper a new approach for modeling fibrous materials analysis using FDTD is presented and validated. A set of simulations covering various different materials is performed, including some real fiberglass cases.
Convention Paper 6167

14:00 h

S-3 Reverberation Control in an Auditorium Using Loudspeaker Array—*Kazuho Ono*[1], *Kimio Hamasaki*[1], *Setsu Komiyama*[1], *Sumi Sakumoto*[2], *Juro Ohga*[3]
[1]NHK Science and Technical Research Laboratories, Tokyo, Japan
[2]Cosmo Space
[3]Shibaura Institute of Technology, Tokyo, Japan

An electroacoustic reverberation control system is used mainly for multipurpose auditoriums or concert halls whose acoustical designs are not ideal for music performance. The present paper discusses the use of loudspeaker arrays for electroacoustical reverberation control in an auditorium, especially the effect of using multiple loudspeakers on a listening area. The experiment was conducted in our new auditorium equipped with 7 vertical pillar-type loudspeaker arrays for each sidewall of the auditorium. Subjective evaluation tests for lateral balance was conducted with various loudspeaker setups and listening points, including off-center ones. The results were compared with sound pressure distribution created by corresponding loudspeaker setups, based on the criteria of setting loudspeakers to large listening areas.
Convention Paper 6168

14:30 h

S-4 Room Acoustics and Equalization of Loudspeaker Systems for Multipurpose Mixing Theaters—*Andrew Munro*, Munro Acoustics Ltd. And Dynaudio Acoustics, London, UK

For many years a series of equations have been used to design and predict the performance of sound systems and acoustic environments based on statistically diffused sound fields and idealized directivity patterns. Although these equations have been modified for semi-reverberant spaces, there is a significant error produced by the strength of both early reflections and room modes. A comparison of theory and measurement applied to film mixing theaters leads to some interesting conclusions.
Convention Paper 6169

15:00 h

S-5 Implementation of a Nonlinear Room Impulse Response Estimation Algorithm—*Tim Collins*, University of Birmingham, Birmingham, UK

Most techniques for estimating the transfer function (or impulse response) of an acoustical space with a high signal-to-noise ratio operate along similar principles. A known, broadband signal is transmitted at one point in the room while being simultaneously recorded at another. A matched-filter is then used to compress the transmission waveform into an approximate impulse, and equalization filtering is used to remove any coloration caused by the nonuniform energy-spectrum of the transmission and/or the nonideal response of the loudspeaker/microphone combination. In this paper the limitations of this conventional technique will be highlighted, especially when using low-cost equipment. An alternative, nonlinear deconvolution technique is proposed, which will be shown to give superior performance using both synthetic waveforms and practical room measurements.
Convention Paper 6170

15:30 h

S-6 Influence of Ray Angle of Incidence and Complex Reflection Factor on Acoustical Simulation Results
—*Emad El-Saghir*[1], *Stefan Feistel*[2]
[1]Acoustic Design Ahnert Limited, Cairo, Egypt
[2]SDA Software Design Ahnert GmbH, Berlin, Germany

Many ray tracing algorithms make use of the single-valued diffuse-field absorption coefficient to simulate the sound field in a given room computer model. They consider, however, neither the effect of the angle of incidence nor the fact that the reflection factor is complex. If characteristic impedance and wave number, which are measured in an impedance tube, are known, we can expect reflectograms, which look different from those generated by current simulators, and look different for different thicknesses. This paper investigates how much the angle-dependent reflectograms, which consider phase shift due to complex reflection factors, look different from the angle-independent ones respectively, and whether the statistical nature of reflectograms leads to the cancellation of such effects.
Convention Paper 6171

Session Z10 **Monday, May 10** **13:00 h–14:30 h**
Corridor 7.1b

P A P E R S

POSTERS: LOW BIT-RATE AUDIO CODING

13:00 h

Z10-1 Optimal Bit Allocation Strategy for Perceptual Audio Coders Employing Uniform Quantization Schemes
—*Preethi Konda, Vinod Prakash*, Ittiam Systems Pvt. Ltd., Bangalore, India

Using the perceptual distortion metric returned by the psychoacoustic module, conventional bit allocation schemes operate iteratively to maintain equal perceptual distortion in all critical bands. For codecs employing uniform quantization schemes, this paper proposes a new approach to determine the optimal MNR (Mask-to-Noise Ratio) levels for the critical bands. The scheme exploits the fact that the quantizer used is uniform in nature and all critical bands are equally distorted, to arrive at a noniterative solution. Additionally, this method is independent of the target bit-rate. The proposed scheme achieves a 2- to 3-times reduction in the complexity of the quantization block. An example application for this scheme is given with reference to the MPEG-2 Layer 1 and 2 encoder.
Convention Paper 6172

13:00 h

Z10-2 Embedded Speech Codec Based on Speex—*Md. Kamaruzzaman, Hervé Taddei*, Siemens AG, Munich, Germany

Embedded speech coding technique is of interest for many applications like VoIP, multimedia broadcasting, and video conferencing. We propose a CELP-based embedded speech codec that is operable for both narrowband and wideband speech signals. Our three-layered embedded codec offers three bit-rates. This embedded codec is based on the Speex codec. In our embedded speech codec, innovation vectors of the higher layers are embedded in the innovation vector of the lowest layer. All speech coding parameters but the innovation vector are shared between the lowest layer and higher layers. In our algorithm, higher bit rates are rewarded with better quality, penalizing the lowest bit rate.
Convention Paper 6173

13:00 h

Z10-3 A Memory and Computationally Efficient Synthesis Sub-band Filter for MPEG Audio Decoding—

Mahabaleswara Bhatt, India Product Development Center, Analog Devices, Bangalore, India

This paper proposes a novel method for memory and computationally efficient implementation of a sub-band synthesis filter for MPEG audio decoding. In contrast to the conventional approach, this derived approach proposes to compute 64 sets of windowing operations in the beginning, each with eight input samples and four re-arranged window coefficients. Subsequently, these windowed sequences are used for two matrixing operations. The proposed fast algorithm exploits not only the DCT relationship for matrixing operations but also procedure pruning for required DCT coefficient computations. Moreover, the windowing operations make use of the symmetry that exists in the window coefficient array. Additionally, the derived approach eliminates the intermediate arrays and explicit filtering operation by appropriately merging these into the windowing and matrixing operations itself. This yields a benefit in reducing the memory requirement and also involves data transfers while computing.
Convention Paper 6174

13:00 h

Z10-4 Transient Detection for Transform Domain Coders—
Venkata Suresh Babuu, Ashish Kumar Malot, Vijayachandran V. M., Vinay M. K., Emuzed India Pvt. Ltd., Bangalore, India

State-of-the art audio encoders are based on transform-domain coding algorithms. Due to time-frequency uncertainty, transform domain coders suffer from "pre-echo" and "diffusion" artifacts during transient portions of the signal. These artifacts occur because of large transform lengths used to achieve higher coding gains. Audio encoders employ various tools such as adaptive transform lengths, TNS, etc., to efficiently code transient portions of the audio signal. Typically audio signals consist of time domain transients (e.g., castanets), frequency domain transients (e.g., flute, clarinet), and transients observed in speech signals during consonant to vowel transitions, etc. Identification of these transients in an audio signal is vital to achieve perceptual quality at low bit-rates. This paper discusses the various transient classes present in audio signals, apart from describing a transient detector employed for efficient modeling of all classes of transients. The proposed transient detector has been incorporated in MPEG-4 AAC encoder, independent of the psychoacoustic analysis methodology used. Listening tests as well as OPERA scores indicate substantial improvement in audio quality over the baseline encoder.
Convention Paper 6175

PAPERS

13:00 h

Z10-5 Signal-Adaptive Parametric Modeling for High Quality Low Bit-Rate Audio Coding—*Pedro Vera-Candeas*[1], *Nicolas Ruiz-Reyes*[1], *Manuel Rosa-Zurera*[2], *Jose Curpián-Alonso*[1], *Pedro Jesús Reche-López*[1]

[1]University of Jaén, Linares, Spain
[2]University of Alcalá, , Alcalá de Henares, Madrid, Spain

In this paper, signal-adaptive parametric models based on over-complete dictionaries of time-frequency atoms are considered for high-quality low bit-rate parametric audio coding. There are a variety of frameworks for deriving over-complete signal expansions, which differ in the structure of the dictionary and the manner in which dictionary atoms are selected for the expansion. Psychoacoustic-adapted matching pursuits are accomplished for extracting sinusoidal components using an harmonic dictionary, while energy-adapted matching pursuits are carried out for transients modeling with a wavelet-based dictionary. First, transients are detected, modeled (with wavelet functions), and removed from the original audio signal, leaving a residue. Then, sinusoids are modeled using complex exponential functions and removed from the initial residue, leaving a noise-like signal. This final residue is modeled taking advantage of the good time-frequency location of the wavelet transform and considering psychoacoustic principles. An M-depth Wavelet Transform is first applied to the residue. Energy of each wavelet sub-band is then computed, and finally a Time Noise Shaping (TNS) process is applied to each one, which involves a parametric model for the noise-like signal. The resulting multipart model (Sines + Transients + Noise) is efficiently applied by taking into account psychoacoustical information for audio coding purposes. The combination of all these ideas results in nearly transparent parametric audio coding at binary rates close to 16 kbps for most of the CD-quality one-channel audio signals considered for testing. Listening tests allow us to say that our coder achieves better results than MPEG-4 AAC at very low bit rates (close to 16 kbps).

Convention Paper 6176

09:30 h

Z10-6 Decoder-Based Approach to Enhance Low Bit-Rate Audio—*Evelyn Kurniawati*[1], *Chiew Tong Lau*[1], *Benjamin Premkumar*[1], *Javed Absar*[2], *Sapna George*[2]

[1]Nanyang Technological University, Singapore
[2]ST Microelectronics Asia Pacific Pte. Ltd., Singapore

A method to improve the PSNR of a perceptual audio coder is presented. It is based on the use of a noise esti-

mator at the decoder side to relate the quantization parameters and the quantization error. The sp? quartic equation established contains two real roots, of which one is the desired spectral value. This value contains lesser quantization error compared to the dequantized spectral value of a normal decoder. This leads to an improvement of up to 12 dB in SNR without significant increase in the decoder complexity.
Convention Paper 6177

09:30 h

Z10-7 Efficient Intraframe Coding of Monophonic Audio—
Aníbal Ferreira, University of Porto/INESC, Porto, Porto, Portugal

This paper describes the design of an Advanced Audio Spectral Coder (ASC) that seeks: coding efficiency by combining source and perceptual audio coding techniques; bitstream semantic scalability by segmenting the audio signal into transients, sinusoids and noise; low delay coding by using a moderate transform size and no bit stream buffer; and embedded error robustness by not using interframe coding. The operation of ASC is explained, its performance is assessed using a few test results, and potential application areas are also addressed.
Convention Paper 6166

WORKSHOPS

W-1	**Auralization—Tool or Toy**	Saturday, May 8 09:00 h-11:00 h Room 7.1a-1
W-2	**The Do's and Don'ts of Microphones**	Saturday, May 8 12:30 h-14:00 h Room 7.1a-1
W-3	**Multichannel in Automobiles**	Saturday, May 8 14:00 h-16:00 h Room 7.1a-1
W-4	**Perception of Loudspeaker Nonlinear Distortion: An Open Discussion**	Saturday, May 8 16:00 h-18:00 h Room 7.1a-1
W-5	**Wave Field Synthesis: Basics and Authoring Considerations**	Sunday, May 9 09:00 h-11:00 h Room 7.1a-1
W-6	**Subjective Microphone Evaluations**	Sunday, May 9 11:00 h -12:30 h Room 7.1a-1
W-7	**Comparison of Existing Archiving Tools**	Sunday, May 9 13:00 h-15:30 h Room 7.1a-1
W-8	**Interfacing Loudspeaker and Room**	Sunday, May 9 15:30 h-18:00 h Room 7.1a-1
W-9	**Digital Radio Mondiale**	Monday, May 10 09:00 h-11:30 h Room 7.1a-1
W-10	**High Speed Audio Networking**	Monday, May 10 11:30 h-13:30 h Room 7.1a-1
W-11	**Sound Systems for Hearing Impaired People**	Monday, May 10 14:00 h-16:00 h Room 7.1a-1
W-12	**Touring Sound Systems: DoCurrent Speaker Concepts Meet User's Requirements: An Open Discussion**	Monday, May 10 16:00 h-18:30 h Room 7.1a-1
W-13	**Forensic Audio**	Tuesday, May 11 09:00 h-11:00 h Room 7.1a-1
W-14	**The Role of Multiple Low-Frequency Signals in the Perception of Reproduced Sound**	Tuesday, May 11 11:00 h-13:00 h Room 7.1a-1

W-15	**Advanced Recording and Reproduction Paradigms Compatible with 5.1 Media**	Tuesday, May 11 13:30 h–16:00 h Room 7.1a-1
W-16	**Measuring and Verifying the Speech Intelligibility Performance of Voice Alarm and Emergency Sound Systems**	Tuesday, May 11 16:00 h–18:00 h Room 7.1a-1
W-17	**Sound Design in Film Postproduction**	Tuesday, May 11 16:00 h–17:30 h Room 7.1b-1

Workshop 1
Saturday, May 8 **09:00 h-11:00 h**
Room 7.1a-1

AURALIZATION—TOOL OR TOY

Chair: **Jan Voetman**, DELTA Acoustics and Vibration, Lyngby, Denmark

Panelists: *Ingolf Bork*, PTB, Braunschweig, Germany
Dorte Hammershoi, Aalborg University, Aalborg, Denmark
Christoph Moldrzyk, Technical University of Berlin, Berlin, Germany
Jens Holger Rindel, Technical University of Denmark, Lyngby, Denmark
Lise-Lotte Tjellesen, DELTA Acoustics and Vibration, Lyngby, Denmark

Auralization—as you might remember—is the technique used in room acoustic computer modeling, that enables you to “listen” to the orchestra in the simulated, nonexisting room. The technique has fascinated acoustic consultants, their clients, architects, researchers, etc., for years. But quite frankly how realistic or how close does this technique simulate the real world?

Very little work has been done in order to compare real situations with simulated, simply because it is quite complicated to do this kind of comparison. For instance, how do you simulate the directional characteristics of an orchestra on a stage?

This workshop will take you through the basics of auralization, discuss the difficulties in doing this kind of comparison, and show you the latest step forward in the technique.

Workshop 2
Saturday, May 8 **12:30 h-14:00 h**
Room 7.1a-1

THE DO'S AND DON'TS OF MICROPHONES

Chair: **Martin Schneider**

This workshop will give practical examples of microphone behavior in standard and non-standard circumstances. Demonstrations will be given of wind, humidity & weather effects, occurrences with phantom power, gain settings and distortion, interference problems with RF and mobile phones; finally, hints for detecting defects and evaluating used equipment, as well as safety & longevity issues. The topics will be covered with short theoretical introductions and extensive audio examples.

14:00 h

Technical Committee Meeting on Audio for Telecommunications (Hall 7.1b)

Workshop 3
Saturday, May 8 **14:00 h-16:00 h**
Room 7.1a-1

MULTICHANNEL IN AUTOMOBILES

Chair: **Tim Nind**, Harman/Becker Automotive Systems, Martinsville, IL, USA

Panelists: *David Griesinger*
Martin Lindsay
others TBA

The adoption of multichannel surround systems for the reproduction of music and cinema sound in the domestic market is finding it's way into the automotive world. A number of the luxury car makers already offer surround systems based on 2-channel source material and the first genuine 5.1 discrete systems are just emerging. This will be followed rapidly by a great number of both 2-channel and discrete systems not only in luxury cars but also those covering the wider market. This poses interesting questions in terms of the way these systems are engineered and evaluated. This workshop will explore some of these issues and will include panellists from both OEM suppliers and the recording industry.

Workshop 4
Saturday, May 8 **16:00 h-18:00 h**
Room 7.1a-1

PERCEPTION OF LOUDSPEAKER NONLINEAR DISTORTION: AN OPEN DISCUSSION

Chair: **John Stewart**, Harman/Becker Automotive Systems, Inc., Martinsville, IN, USA

Panelists: *Michael Keyhl,* Opticom, Erlangen, Germany (Perception and Perception Models)
Wolfgang Klippel, Klippel GmbH, Dresden, Germany (Live Listening Test)
Steve Temme, Listen Inc., Boston, MA USA (Live Buzz and Rub Detection)

The Standards Committee SC-04-03, Loudspeaker Modeling and Measurement, has been struggling with distortion specifications and their relationship to listener perception. The Technical Committee on Loudspeakers and Headphones presents a window into this issue with a workshop on our perception of reproduced sound.

Workshop attendees will have a chance to participate in a distortion threshold test. They will gain insight to the properties of the human hearing mechanism and how it can be modeled mathematically. The detection of what might be called extremely audible nonlinear distortion will be presented. A guide to appropriate signals for audible and measurable nonlinearities will be offered.

Workshop 5
Sunday, May 9 **09:00 h-11:00 h**
Room 7.1a-1

WAVE FIELD SYNTHESIS: BASICS AND AUTHORING CONSIDERATIONS

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

Panelists: *Frank Melchior*, Fraunhofer Institute for Digital Media Technology, Ilmenau, Germany
Renato Pellegrini, sonicEmotion, Dielsdorf Switzerland
Guenther Theile, Institut für Rundfunktechnik, Munich, Germany
Diemer de Vries, Delft Technical University, Delft, The Netherlands

Wave Field Synthesis is on its way to real world applications. This workshop will introduce the current state of the art and focus on authoring: What tools are available today; what needs to be done; what are the new effects available?

Workshop 6
Sunday, May 9 **11:00 h -12:30 h**
Room 7.1a-1

SUBJECTIVE MICROPHONE EVALUATIONS

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

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

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

Workshop 7
Sunday, May 9 **13:00 h-15:30 h**
Room 7.1a-1

COMPARISON OF EXISTING ARCHIVING TOOLS

Chair: **Klaus M. Heidrich**, VCS Nachrichtentechnik GmbH, Bochum, Germany

Panelists: Bayrischer Rundfunk, Bavarian Broadcasting Corporation
W. Grieger, Norddeutscher Rundfunk-North German Radio
Rainer A. Kellerhals, Tecmath AG, Kaiserslautern. Germany
Karl W. Pieper, VCS AG, Nürnberg, Germany
Niko Waesche / *Yvonne Graf*, IBM Business Consulting Services

During the past few years, digital audio archiving solutions, also labeled “Media Asset Management” or “Content Management,” have evolved steadily. Different system concepts and tools have become part of daily operations and have proved successful. However, continuous improvement is a must, in terms of technology, organization and processes, and cost-benefit ratio. This workshop will focus on fundamental issues such as return on investment, standards and interfaces, and workflow integration, rather than on details of dedicated products and tools. The panel is excellently suited to highlight both the broadcaster’s and the industry”s points of view.

Workshop 8
Sunday, May 9 **15:30 h-18:00 h**
Room 7.1a-2

INTERFACING LOUDSPEAKER AND ROOM

Chair: **Jan Abildgaard Pedersen**, Bang & Olufsen a/s, Streuer, Denmark

Panelists: *Andrew Goldberg*, Genelec Oy, Ilsami, Finland
John Mourjopoulos, University of Patras, Patras, Greece
Todd S. Welti, Harman International Industries, Inc., Northridge, CA, USA
Rhonda J. Wilson, Meridian Audio Ltd., Huntingdon, Cambridgeshire, UK

The interfacing of loudspeakers and room has gained more and more focus, and it has proven to be an essential element in optimizing the sound quality in an audio reproduction system. Several different approaches have been presented during the last 2 years, and this workshop will combine short presentations of several of the principal systems by experts within the research field. There will be an open floor discussion where the experts form a panel. In different ways the new systems are addressing some of the problems known from the traditional room equalization systems. Both the use of DSP and different acoustic strategies have enabled this.

Workshop 9
Monday, May 10 **09:00 h-11:30 h**
Room 7.1a-1

DIGITAL RADIO MONDIALE

Chair: **Peter Senger**, DRM, Deutsche Welle, Bonn, Germany

Panelists: *Martin Dietz*, Coding Technologies, Nürnberg, Germany
Heinz-Peter Friedrich, Deutsche Welle, Bonn, Germany
Olaf Korte, Fraunhofer Gesellschaft, Munich, Germany

DRM/Digital Radio Mondiale is a new digital broadcasting system for frequency bands below 30 MHz. It uses the most sophisticated audio encoding system MPEG-4 AAC+ and offers high audio quality in 9 or 10 kHz rf channels. Audio encoding experts from DRM will explain the new system and production experts will explain the new possibilities for radio program producers. After an introduction experts will answer and discuss questions from the participants.

Workshop 10
Monday, May 10 **11:30 h-13:30 h**
Room 7.1a-2

HIGH SPEED AUDIO NETWORKING

Chair: **Peter Henkel**

Panelists: *Markus Berg*, Institut für Rundfunktechnik, Munich, Germany
Martin Pistor, MCI, Germany
Henrik Svantesson, Net Insight, Stockholm, Sweden

In the last few years, transmission standards in audio production networks converged and moved from FDDI and ATM toward Ethernet and IP protocol. This workshop presents a survey of the typical broadcast environment and the data formats used in audio production networks today. Besides the TCP/IP performance parameters which have a great impact on network throughput there will be presented a new technology called dynamic transfer mode (DTM). In wide area networks, DTM can be a cost-effective alternative to carrier technologies like SONET and ATM.

Workshop 11
Monday, May 10 **14:00 h-16:00 h**
Room 7.1a-1

SOUND SYSTEMS FOR HEARING IMPAIRED PEOPLE

Chair: **Birger Kollmeier**, Universität Oldenburg and Kompetenzzentrum HörTech, Germany

Panelists: *Inga Holube*, Fachhochschule Oldenburg /Ostfriesland /Wilhelmshaven, Germany
Stefan Launer, Phonak AG, CH-Stäfa, Switzerland
Torsten Niederdränk, Siemens Audiologische Technik, Erlangen, Germany

What aspects of sounds are not perceived by hearing-impaired listeners? Why does the impaired ear produce more distortion even though it is less nonlinear than the normal ear? And what principles are used in modern hearing instruments to overcome these problems? This workshop will cover signal processing techniques used in state-of-the-art digital hearing instruments as well as the limitations and future developments in hearing aid hardware components. The roadmap towards a "HiFi-Hearing" aid will be covered as well as recent developments in assistive listening devices, such as, e.g., remote directional FM-microphones and telecoil systems. The panelists are recruited from leading experts in fundamental and applied university research as well as from R&D departments of the leading hearing aid manufacturers.

Presentations include:

"Acoustical Requirements for Hearing-Impaired Listeners,“ presented by Birger Kollmeier.

"Design Principles for Modern Hearing Instruments," presented by Inga Holube.

"Hi-Fi-Hearing Aid!?" presented by Torsten Niederdränk.

"Assistive Listening Devices," presented by Stefan Launer.

Workshop 12
Monday, May 10 **16:00 h-18:30 h**
Room 7.1a-2

TOURING SOUND SYSTEMS: DO CURRENT SPEAKER CONCEPTS MEET THE USER'S REQUIREMENTS: AN OPEN DISCUSSION

Chair: **Uli Mall**

Panelists: *Tony Andrews*, Funktion one, Beare Green, Dorking, UK
Christian Heil, L-Acoustics, Marcoussis Cedex, France
Evert Start, DURAN Audio, Zaltbommel, Netherlands

Are current "flavor of the year" touring sound systems hype or technological advancements? What are the pros and cons of different system solutions? What do different types of users really need, and are their requirements met by currently available systems and tool sets? And what are the future perspectives—wishes versus realities. Viewpoints from and an open discussion together with both experienced users from different touring application backgrounds and leading experts from some of the key players in the industry.

Workshop 13
Tuesday, May 11 **09:00 h-11:00 h**
Room 7.1a-1

FORENSIC AUDIO

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

Panelists: *Durand Begault*, Audio Forensic Center & NASA, Mountain View, CA, USA
Werner A. Deutsch, Institut für Schallforschung, der Österreischischen Akademie der Wissenschaften
Wes Dooley, Audio Engineering Associates, Pasadena, CA, USA

Forensic audio covers all types of audio analyses from which the results are evaluated for presentation in court. This includes voice comparison and voice identification, acoustical crime scene analysis, authentication of audio/video recordings, examination of sound incidents on sound recordings, etc. In many cases audio recordings can be the most important evidence in a case. Precision and control for error are very important due to possible legal consequences for the client. In this workshop a number of audio forensic specialists from the AES will present some of the procedures and techniques used and discuss possibilities and limitations in the field.

Workshop 14
Tuesday, May 11 **11:00 h-13:00 h**
Room 7.1a-1

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

Chair: **William Martens**, McGill University, Montreal, Quebec, Canada

Panelists: *Jonas Braasch*, McGill University, Montreal, Quebec, Canada
David Greisinger, Lexicon, Bedford, MA, USA
Geoff Martin, Tonmeister, Bang & Olufsen a/s, Struer, Denmark
Robin Miller, Filmakers Inc, Bethlehem, PA, USA
Gunther Theile, Institut für Rundfunktechnik, Munich, Germany
Todd Welti, Harman International Industries, Inc., Northridge, CA, USA

This workshop will examine the relative value of reproducing more than a single channel low-frequency (i.e., subwoofer) signal in two-channel and multichannel stereophonic sound reproduction. As this workshop is sponsored by the AES Technical Committee on Perception and Subjective Evaluation of Audio Signals, the emphasis of the workshop will be on the differences that people can hear when presented with two or more subwoofer signals, rather than on optimizing bass management schemes for conventional 5.1 channel surround sound. Two of the questions raised are: What is best to do with two LFE signals? What is best when there are none?

Workshop 15
Tuesday, May 11 **13:30 h-16:00 h**
Room 7.1a-1

ADVANCED RECORDING AND REPRODUCTION PARADIGMS COMPATIBLE WITH 5.1 MEDIA

Chair: **Ralph Glasgal**

Panelists: *Angelo Farina*, University of Parma, Parma, Italy
Dave Malham, University of York, Heslington, York, UK
Robin Miller, Filmakers Inc., Bethlehem, PA, USA
Itai Neoran, ks Waves Ltd., Tel-Aviv, Israel
Diemer de Vries, Delft University of Technology, Delft, The Netherlands

Coding schemes such as DTS, Dolby, MLP, etc., and media such as SACD and DVD (and eventually blue laser) may be used to deliver virtual reality, surround cinema, or the 3-D concert-hall experience, via such psychoacoustically valid paradigms as Ambiophonics, Ambisonics, Wave Field Synthesis, 10.2, and even novel 2.0. This panel of experts in both recording and reproduction methodologies, discusses these and new research in related areas of psychoacoustic verisimilitude, including hall ambience convolution and capturing height.

The seminar panel will explore such topics as recording live 360-degree sound fields, using impulse responses instead of microphones to record hall ambience, combining Ambisonic B format for surround with 5.1 LCR speakers, 3-D playback of existing stereo recordings by eliminating crosstalk, ways of achieving 360-degree direct sound via 5.1 speakers, center speaker versus stereo dipole, monitoring 5.1 recording sessions via Ambiophonics, advanced microphone designs, measuring hall impulse responses, etc.

Depending on the interests of those in attendance and their questions, some demonstrations of psychoacoustic phenomena will be staged.

Workshop 16
Tuesday, May 11 **16:00 h-18:00 h**
Room 7.1a-1

MEASURING AND VERIFYING THE SPEECH INTELLIGIBILITY PERFORMANCE OF VOICE ALARM AND EMERGENCY SOUND SYSTEMS

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

Panelists: *Wolfgang Ahnert*, ADA Acoustic Design Ahnert, Berlin, Germany
Thomas Steinbrecher, Bose, Germany
Peter Swarte, P.A.S., Eindhoven, Netherlands

As more and more reliance is placed on voice announcements for emergency alerts and alarms instead of traditional warning tones and sirens, the intelligibility performance of such systems has never been so important. Verification of the intelligibility of voice alarm (VA) systems and "sound systems for emergency purposes" is therefore becoming an increasingly important and topical issue. The workshop will review the currently available techniques and highlight the practical shortfalls and difficulties associated not only with the methods themselves but also with testing emergency sound systems in practice. The workshop will show that although some forms of modern signal processing can be used to enhance intelligibility, the current measurement techniques and metrics do not always indicate the improvement. The errors and accuracy of such measurements will also be discussed and comparisons made with computer aided design and prediction programs. The workshop is a must for anyone involved with the design or testing of public address, voice alarm, and emergency paging systems. It is planned to carry out a number of live demonstrations and measurements during the workshop using the latest state of the art equipment and programs.

Workshop 17
Tuesday, May 11 **16:00 h–17:30 h**
Room 7.1b-1

SOUND DESIGN IN FILM POSTPRODUCTION

Chair: **Robin Pohle**, ATMO Audio Produktion, Berlin, Germany

Panelist: *Jörg Hönle*, ATMO Audio Produktion, Berlin, Germany

Topics for discussion in this workshop include:

- Difference between sound design and sound editing
- Comparison of the approach to sound design in the USA and in Germany, in terms of work flow, team structures, etc.
- The basic demands from a modern Digital Audio Workstation (DAW) and its user, their development and influence on the work flow (with audio examples)
- Features of various DAW systems
- Different recording formats and resulting compatibility problems
- Present state of the art of DAW systems and an outlook to future developments.
- Sound engineer training and his/her job market in Europe (with focus on Germany)

TUTORIAL SEMINARS

TS-1	**The Basics of Digital Audio: A Seminar with Demonstrations**	Saturday, May 8 09:00 h–11:30 h Room 7.1a-2
TS-2	**All About Compressors**	Saturday, May 8 12:30 h–14:00 h Room 7.1a-2
TS-3	**Basics of Sound Reinforcement by Using Different Loudspeaker Types**	Saturday, May 8 14:00 h–16:00 h Room 7.1a-2
TS-4	**Practical Aspects of Wireless Microphones**	Saturday, May 8 16:00h–18:00 h Room 7.1a-2
TS-5	**Working with Microphones: A Practical Review**	Sunday, May 9 09:00 h–11:00 h Room 7.1a-2
TS-6	**WEB-TV—Multimedia through the Internet**	Sunday, May 9 11:00 h–13:00 h Room 7.1a-2
TS-7	**Loudspeakers**	Sunday, May 9 13:00 h–15:30 h Room 7.1a-2
TS-8	**Surround Sound Design in TV**	Sunday, May 9 16:00 h–18:00 h Room 7.1a-1
TS-9	**All About Microphone Preamplifiers**	Monday, May 10 09:00 h–10:30 h Room 7.1a-2
TS-10	**The Center Channel Challenge**	Monday, May 10 11:30 h–13:30 h Room 7.1a-1
TS-11	**Grounding and Shielding**	Monday, May 10 13:30 h–16:00 h Room 7.1a-2
TS-12	**How to Set-Up 5.1 Surround**	Monday, May 10 16:00 h–18:30 h Room 7.1a-1
TS-13	**All About Audio Data Reduction**	Tuesday, May 11 09:00 h–11:00 h Room 7.1a-2
TS-14	**Spectral Processing—Fundamentals and Digital Audio Effects**	Tuesday, May 11 11:00 h–12:30 h Room 7.1a-2
TS-15	**Listening Tests in Practice**	Tuesday, May 11 13:30 h–18:00 h Room 7.1a-2

Tutorial Seminar 1
Saturday, May 8 **09:00 h–11:30 h**
Room 7.1a-2

THE BASICS OF DIGITAL AUDIO: A SEMINAR WITH DEMONSTRATIONS

Presenters: **Stanley Lipshitz, John Vanderkooy**, University of Waterloo, Waterloo, Ontario, Canada

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

Topics to be covered will include:

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

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

12:30 h

Technical Committee Meeting on Perception and Subjective Evaluation of Audio (TC Room 1, Hall 7.1b)

12:30 h

Technical Committee Meeting on Audio for Games (TC Room 2, Hall 7.1b)

Tutorial Seminar 2
Saturday, May 8 **12:30 h–14:00 h**
Room 7.1a-2

ALL ABOUT COMPRESSORS

Chair: **Ed Simeone,** TC Electronic, Westlake Village, CA, USA

Panelists: *Ben Georgiades***,** Engineer, UK
Tobias Lehmann, Teldex (ex-Teldec) Studios, Berlin, Germany
Günther Pauler, Mastering Engineer, Germany
NN

Ed Simeon will present a brief historical overview of compression and the different types of compression in use today. Topics covered include: What do all compressors have in common? What are the various types of compression and when were they introduced: optical compression, tube compression, VCA compression, FET compression, multiband compression (analog and digital).

Guest panelists from the European recording and mastering community will field questions during an extended question-and-answer period.

Tutorial Seminar 3
Saturday, May 8 **14:00 h–16:00 h**
Room 7.1a-2

BASICS OF SOUND REINFORCEMENT BY USING DIFFERENT LOUDSPEAKER TYPES

Presenter: **Wolfgang Ahnert**, ADA Acoustic Design Ahnert, Berlin, Germany

The different types of loudspeakers will be explained:

- Point sources
- Clusters
- Line arrays
- Loudspeaker arrays in general

The physical background of sound radiation will be explained. By means of EASE4.0 the different directivity patterns are shown. In this context different applications will show which loudspeaker type is most suitable for any particular application. The interaction between sound system and room or open-air site will be derived.

This seminar will make clear why it is that one type of loudspeaker cannot be used for every purpose, but that a choice must be made. By understanding the reason why we use different types of speakers for different situations you will avoid complaints and claims from clients or contractors you are working for.

Tutorial Seminar 4
Saturday, May 8 **16:00 h–18:00 h**
Room 7.1a-2

PRACTICAL ASPECTS OF WIRELESS MICROPHONES: 20 Rules of Thumb for the Operation of Multichannel Wireless Microphone Systems and Ear Monitoring

Presenter: **Peter Arasin**, Sennheiser Electronic, Wedemark, Germany

During recent years wireless microphones have attained a high level of operational safety. If trouble comes up however, fast identification and problem fixing is essential to keep the show going on. In more than 95% of all problems no soldering iron was needed, but systematic analysis of the situation was the way to success. The important rules for safe operation of multichannel wireless microphone systems in simultaneous use with wireless ear monitoring will be presented and explained with practical tests.

Tutorial Seminar 5
Sunday, May 9 **09:00 h–11:00 h**
Room 7.1a-2

WORKING WITH MICROPHONES: A PRACTICAL REVIEW

Presenter: **Ron Streicher,** Pacific AV Enterprises, Pasadena, CA, USA

Ron Streicher will present a "hands-on" tutorial seminar covering the fundamental use of microphones. This is not a seminar on “where” to put a microphone to obtain the best pickup, but "how to put it there" to obtain the best performance from it.

A freelance tonmeister for more than forty years and audio production manager of the Aspen Music Festival and School for more than a decade, Mr. Streicher has developed an extensive practical knowledge of microphone mounting and rigging techniques which he will demonstrate using "live" microphones and numerous photographs.

Tutorial Seminar 6
Sunday, May 9 **11:00 h- 13:00 h**
Room 7.1a-2

WEB TV – MULTIMEDIA THROUGH THE INTERNET

Presenter: **Eckhard Meyer,** T-Systems Media & Broadcast, Bonn, Germany

Being a relatively new medium Web-TV has gained a considerable foothold in the dissemination of multimedia content through the Internet over the past years. Although not yet as established as terrestrial radio or television broadcasting the remarkable growth of broadband connections for end users has led to streaming entering the mainstream of media distribution.

This seminar will explain the technological and practical features of the technology as well as the requirements that are needed to set up a successful streaming operation. Together with examples of what this technology can offer the Do's and Don'ts of streaming will also be highlighted. Furthermore the tutorial will cover those characteristics that go beyond a "simple" streaming configuration such as digital rights management and streaming to mobile devices.

Last but not least, commercial aspects that are needed to make the distribution of streaming content a viable and profitable operation will also be discussed.

Tutorial Seminar 7
Sunday, May 9 **13:00 h–15:30 h**
Room 7.1a-2

LOUDSPEAKERS

Chair: **Neil Harris**

Presenters: *Juha Backman*, Nokia Mobile Phones, Espoo, Finland
Wolfgang Klippel, Klippel GmbH, Dresden, Germany
Neville Thiele, Consultant, Epping, New South Wales, Australia
John Vanderkooy, University of Waterloo, Waterloo, Ontario, Canada

This tutorial is aimed at technically-minded people who have an interest in developing a deeper understanding of how loudspeakers work. There are four participants, each of whom is recognized as expert in his respective field. There will be time between presentations, and at the end of the session, for questions from the floor.

Presentations:

Basic Acoustics of Loudspeakers by John Vanderkooy. This tutorial outlines the essential acoustics needed to understand direct-radiator loudspeakers. Topics range from the gas law to the diffraction of a loudspeaker cabinet. Acoustic pressure and particle velocity concepts for plane waves and spreading 3-D waves are explained, leading to the concepts of acoustic impedance. Acoustic output is related to the acceleration of the diaphragm for a baffled system. The splitting of the bands by a crossover allows different drivers to properly disperse the sound radiation.

Electrical Equivalent Circuit by Neville Thiele. When electrical equivalences are applied to its acoustical circuit, the parameters of a loudspeaker may be measured and its performance analyzed as if it were an electrical filter. The special properties of these filters, some problems in measuring them and procedures for coping with them will be presented.

Nonlinear Behavior by Wolfgang Klippel. This tutorial gives an overview of the dominant nonlinearities inherent in loudspeaker systems. The basics of large signal modeling are developed, and different methods for measuring the thermal and nonlinear parameters are compared. Finally, the relationship between physical causes and signal distortion, instabilities, amplitude compression and other nonlinear symptoms is explained.

Practical Devices on a Small Scale by Juha Backman. Making it work on a small scale.

Tutorial Seminar 8
Sunday, May 9 **16:00 h–18:00 h**
Room 7.1a-1

SURROUND SOUND DESIGN IN TV
Dramaturgical Goals, Tools, and Concepts

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

During 2003, public broadcasters in Europe started multichannel audio transmission via digital satellite DVB-S. Notably, the Austrian Broadcasting Corporation was the first to produce, as well as postproduce, surround sound live (New Year's Concert). The aesthetics of the latter will be the focus of this tutorial, where many different aspects will be presented. Multichannel location recording, workflow of a documentary production, physical and dramaturgical tools, as well as key examples from the author's work will provide an insight into advanced soundtrack crafting techniques for 5.1 surround sound.

Tutorial Seminar 9
Monday, May 10 **09:00 h–10:30 h**
Room 7.1a-2

ALL ABOUT MICROPHONE PREAMPLIFIERS

Chair: **John La Grou**, Millenia; Placeville, CA, USA

Panelists: *Geoff Daking*, Geoffrey Daking & Co., Wilmington, Delaware
George Massenburg, George Massenburg Labs, Franklin, TN, USA
Crispin Taylor

Microphone preamplifiers have become a critical component in both the live and recording worlds. Few audio products have a wider cost spread with such similar specifications. This tutorial addresses key issues in microphone preamplifier design, selection, and use. A few of the issues to be reviewed are: The use of transformers, self-noise, impedance, distortion and perceived sonic differences. Plenty of question-and-answer time will be available.

Tutorial Seminar 10
Monday, May 10 **11:30 h–13:30 h**
Room 7.1a-1

THE CENTER CHANNEL CHALLENGE

Presenter: **Jeff Levinson**, DTS Entertainment, Agoura Hills, CA, USA

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

Tutorial Seminar 11
Monday, May 10 **13:30 h–16:00 h**
Room 7.1a-2

GROUNDING AND SHIELDING

Presenters: **Jim Brown**, Audio Systems Group, Chicago, IL, USA
Bill Whitlock, Jensen Transformers, Van Nuys, CA, USA
John Woodgate, J.M. Woodgate & Associates, Essex, England

Grounding and shielding techniques, at both the equipment and system level, have profound effects on immunity to interference. High-performance professional audio systems routinely encounter interference ranging in frequency from 50- to 60-Hz utility-power up to several GHz. A tutorial overview will explain basic interference coupling mechanisms as well as widely used grounding and shielding strategies. Expert panelists will discuss tradeoffs involved in these strategies, results of various equipment and cable tests, and recommendations for equipment and system designers. A question-and-answer session will follow.

Tutorial Seminar 12
Monday, May 10 **16:00 h–18:30 h**
Room 7.1a-1

HOW TO SET UP 5.1 SURROUND

Presenter: **Chistophe Anet**, Genelec Oy, Iisalmi, Finland

A modern audio production facility must be able to serve productions in a large number of different formats. The change from mono and stereo to multichannel reproduction has produced many problems, both in converting existing production facilities to multichannel format and in new installations.

The audio formats that must be handled by a modern production facility include currently:

- Mono, stereo
- Matrix four channel format
- Five channels (5.0 systems)
- Five channels with a separate low frequency enhancement channel (5.1 systems)
- Advanced multichannel formats such as 6.1, 7.1, and more

This seminar discusses multiple practical questions about the monitoring loudspeakers, their set-up, and possible sources of problems that should be avoided. A brief overview of the current multichannel formats and a dedicated section on Bass Management is also included.

This presentation does not seek to explain monitoring loudspeaker design and technology.

Tutorial Seminar 13
Tuesday, May 11 **09:00 h–11:00 h**
Room 7.1a-2

ALL ABOUT AUDIO DATA REDUCTION

Presenters: **Karlheinz Brandenburg**, Fraunhofer IIS/AEMT, Ilmenau, Germany
N.N.

Audio compression has found its way into mainstream consumer electronics and all computers. Still, there is more technical progress and more standardization going on. The tutorial will focus on:

- The basics of audio coding
- MP3 technology: how does it work, what are the limitations
- Newer standards: AAC, MPEG-4, AC-3, and others
- Ongoing research work
- Parametric coding
- Bandwidth extension work (e.g. HE-AAC)
- Lossless, scalable-to-lossless coding

Tutorial Seminar 14
Tuesday, May 11 **11:00 h–12:30 h**
Room 7.1a-2

SPECTRAL PROCESSING—FUNDAMENTALS AND DIGITAL AUDIO EFFECTS

Presenters: **Xavier Jerra**, University Pompeu Fabra, Barcelona, Spain
Udo Zoelzer, Helmut Schmidt University Hamburg, Germany

The goal of this tutorial is to describe digital audio effects with regard to physical and acoustical effects, digital signal processing, and musical applications with acoustical demonstrations. The tutorial will cover the fundamental signal processing algorithms for creating digital audio effects based on a time-frequency representation of the audio signal called spectral processing. The tutorial is based on the book *DAFX-Digital Audio Effects*.

Tutorial Seminar 15
Tuesday, May 11 **13:30 h–18:00 h**
Room 7.1a-2

LISTENING TESTS IN PRACTICE

Chair: **Nick Zacharov**, Nokia Research Center, Audio-Visual Systems Laboratory, Tampare, Finland

Panelists: *Søren Bech*, Bang and Olufsen a/s, Struer, Denmark
Durand Begault, NASA Ames Research Center, Mountain View, CA, USA
William L. Martens, McGill University, Montreal, Quebec, Canada
Sean Olive, Harman International Industries, Inc., Northridge, CA, USA
Gilbert Soulodre, Communications Research Centre, Ottawa, Ontario, Canada
Thomas Sporer, Fraunhofer Institute for Digital Media Technology IDMT, Ilmenau, Germany

This seminar presents a short but effective guide to preparing, performing, and analyzing data for listening tests. The first part of the seminar will provide a general overview of experimental design methods that are generically applicable to all types of listening tests. The second part of the seminar will specifically consider three main types of listening test categories, providing examples of how they are correctly performed/analyzed and what is their scope of applicability.

EXHIBITOR SEMINARS

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

These presentations will take place in Hall 4.1, Room 5730 and Rooms Z4 and Z5. Please refer to the separate Exhibitor Seminar booklet for participating companies and their seminar descriptions.

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