AES 23rd INTERNATIONAL CONFERENCE

SIGNAL PROCESSING IN AUDIO RECORDING AND REPRODUCTION

Copenhagen, Denmark
May 23–25, 2003

Photograph Courtesy of Danish Tourist Board

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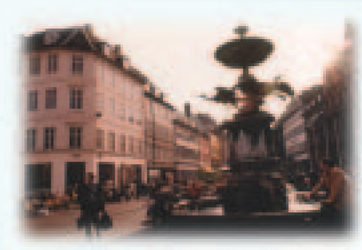

	FRIDAY, MAY 23	SATURDAY, MAY 24	SUNDAY, MAY 25
08:00			
08:30			
09:00	**Registration**	Session 3 **Interfacing Loudspeaker and Room**	Session 5 **DSP in Loudspeakers, Part 1**
09:30			
10:00			
10:30			
11:00			Lunch
11:30			
12:00	Lunch	Lunch	
12:30			
13:00			Session 6 **DSP in Loudspeakers, Part 2**
13:30		Session 4 **Creating Space with DSP**	
14:00			
14:30	**Opening Ceremony and Keynote Address**		
15:00			
15:30	Session 1 **Signal Conversion**		Closing Ceremony
16:00			
16:30	Session 2 **DSP in Recording**		
17:00			
17:30		**Visit to Kronborg Slot (Shakespeare's "Elsinore Castle")**	
18:00			
18:30			
19:00			
19:30			
20:00		**Banquet**	
20:30			
21:00			
22:00			
23:00			

This schedule reflects accurate information as of press time. For up-to-date information see www.aes.org/events/23.

AES 23rd INTERNATIONAL CONFERENCE

Marienlyst Hotel, Helsingør
Copenhagen, Denmark
May 23–25, 2003

The beautiful city of Copenhagen and its surroundings have been a favorite venue for AES conferences. We return this year, May 23–25, to this lovely area of Scandinavia for the AES 23rd International Conference, *Signal Processing in Audio Recording and Reproduction.* The conference will be held at the Marienlyst Hotel and Conference Center in Helsingør, which is 45 km north of Copenhagen. Shakespeare immortalized the local castle, Kronborg Slot, as Elsinore Castle in *Hamlet*.

As signal processing becomes increasingly crucial in audio, the aim of this conference is to focus on signal processing at both ends of the electrical audio signal life cycle, namely the recording and reproduction stages. Signal processing in the recording process depends much on the way the signal is going to be reproduced, and signal processing in the reproduction stage must take into account processing during recording. New techniques and standards in digital audio merely emphasize this point. For this reason it has become necessary to consider the recording and reproduction setups, environments, and pieces of equipment as a single entity when using signal processing. This conference will bring together researchers and developers in all areas of signal processing for audio.

THE TECHNICAL PROGRAM

Jan Abildgaard Pedersen and Lars Gottfried Johansen, papers cochairs, have assembled a high-quality program. The conference will begin on Friday afternoon with a keynote address by Jeff Bier, who will examine the emerging advances in DSP hardware that are revolutionizing audio processing, changing system designs, and creating new product categories. Then the opening session will be *Signal Conversion and Perceptual Effects.* There will be an early evening session, *DSP in Recording*, which will feature two invited papers: "Voice-Specific Signal Processing Tools," by Patrick Bastien, and "Metadata and the Audio Media Lifecycle," by Elizabeth Cohen.

Saturday morning will be devoted entirely to the six-paper session, *Interfacing Loudspeaker and Room.* The afternoon session, *Creating Space with DSP*, will start off with the invited paper "Spatial Postprocessing of Audio Recordings" by Jean-Marc Jot.

Sunday's two-part session is *DSP in Loudspeakers*. Ronald Aarts' invited paper, "Applications of DSP for Sound Reproduction Improvement," starts off the morning session. Wolfgang Klippel's invited paper, "Active Compensation of Transducer Nonlinearities," leads off the afternoon session. The calendar, complete program with abstracts, and registration form follow this page.

THE CASTLE, THE SEA, THE MERMAID

In addition to the challenging technical program, the conference committee, headed by Per Rubak, has planned a visit to Kronborg Slot on Saturday evening followed by a banquet at the Marienlyst Hotel. These events are included in the registration fee, along with the conference proceedings and CD-ROM.

The hotel and conference center (www.marienlyst.dk) is on the shore of the Øresund (the Sound). This is Scandinavia's busiest shipping channel; there is frequent ferry service going to and from Sweden every day. Besides the usual hotel facilities, the Marienlyst boasts a subtropical swimming area, a casino, and a sandy beach on the Sound. Helsingør is just a 55-minute train ride from Copenhagen. Trains run every 20 minutes throughout the day.

Famous for its scenic waterfront and The Little Mermaid statue, Copenhagen is a walkers' paradise: compact and easy to navigate. In addition to many city parks, the gardens and amusement park at Tivoli in the heart of the city are world famous. Strøget, the world's longest pedestrian street, is a shopper's dream. The weather in Denmark in May is generally warm and comfortable. The Danish people are warm and friendly, apparently having vented all their anger in their fearsome Viking era.

"To be or not to be" among those attending this exciting international conference should be an easy choice. Don't procrastinate, go to www.aes.org now for more details and register online and join your colleagues in Denmark in May.

TECHNICAL SESSIONS

Friday, May 23 **2:30 pm–3:30 pm**

KEYNOTE ADDRESS: TRENDS AND DIRECTIONS IN SIGNAL-PROCESSING HARDWARE FOR AUDIO APPLICATIONS

Guest Speaker: **Jeff Bier**, Berkeley Design Technology, Inc. (BDTI), Berkeley, CA, USA

Recent advances in DSP hardware have radically altered audio processing. Audio processing that only a few years ago required a room full of expensive equipment can now be implemented with little more than an ordinary PC. At the same time, advances in DSP hardware have improved the quality of audio processing and enabled new product categories. In this paper, we will examine emerging advances in DSP hardware that will further revolutionize audio processing. We will explore the tradeoffs associated with key audio processing implementation technologies, including DSPs, general-purpose processors, and ASICs. We will also highlight how advances in DSP hardware are changing system designs and creating new product categories.

3:30 pm–4:00 pm

SESSION 1: SIGNAL CONVERSION AND PERCEPTUAL EFFECTS

1-1 **Overload in Signal Conversion—***Søren H. Nielsen, Thomas Lund*, TC Electronic A/S, Risskov, Denmark

Digital audio material is often mastered at such hot levels that frequent clipping in the digital domain occurs. While this in itself can cause listener fatigue, the sound is often severely distorted in the digital-to-analog conversion process at the listener end due to lack of headroom in the reproduction. Purely digital processing such as sample rate conversion may exhibit the same problem. This paper describes measurements to quantify the problem and new listening tests performed on commercial recordings to demonstrate the phenomena. Furthermore, methods to avoid the distortion are discussed and demonstrated.

4:30 pm–7:00 pm

SESSION 2: DSP IN RECORDING

2-1 **Voice-Specific Signal Processing Tools—***Patrick Bastien*, TC-Helicon, Victoria, BC, Canada **(Invited)**

Like virtually any track in a mix, vocals are processed. However, there is more to voice processing than the usual compression, EQ, and reverberation. Over the past decade, some processing tools were designed specifically for the voice, allowing the unique features of this unique musical instrument to be manipulated. These algorithms have found various applications in the music industry, from creating virtual harmony voices for the live performer to perfecting lead vocals in the studio. This paper covers the different types of processing currently available on the market, as well as their applications and the underlying technology.

2-2 **Fast Monaural Separation of Speech—***Niels Henrik Pontoppidan* and *Mads Dyrholm*
[1]Technical University of Denmark, Lyngby, Denmark

We have investigated the possibility of separating signals from a single mixture of sources. This problem is termed the Monaural Separation Problem. Lars Kai Hansen has argued that this problem is topologically tougher than problems with multiple recordings. Roweis has shown that inference from a Factorial Hidden Markov Model, with nonstationary assumptions on the source autocorrelations modeled through the Factorial Hidden Markov Model, leads to separation in the monaural case. By extending Hansens work we find that Roweis' assumptions are necessary for monaural speech separation. Furthermore we develop a hierarchical Viterbi algorithm for the Factorial Hidden Markov Model yielding a significant decrease in complexity of inference.

2-3 **Metadata and the Audio Media Lifecycle—**
Elizabeth Cohen, UCLA Department of Information Studies, Los Angeles, CA, USA **(Invited)**

As multimedia offers a range of physical and cyberspaces for audio materials, metadata becomes increasingly crucial to the longevity of audio materials. The aim of this paper is to focus on inclusion of metadata as part of the signal at the creative end of the audio signal life cycle, specifically as part of the recording stage. The inclusion of metadata at this stage enables efficiencies of production, repurposing, and audio media information retrieval. The integration of metadata with object creation will radically and sensibly expand the creative palette. We will show how Including both descriptive and technical metadata increases the creative reach of the artist. The environments that call for the repurposing and reuse of audio media include recording studios, dubbing stages, music archives, music libraries, the home desktop, or the "Garage Cinema."

Saturday, May 24 **9:00 am–12:00 noon**

SESSION 3: INTERFACING LOUDSPEAKER AND ROOM

3-1 **Adaptive Bass Control—The ABC Room Adaptation System—***Jan Abildgaard Pedersen*, Bang & Olufsen A/S, Struer, Denmark

This paper presents a system for adapting a loudspeaker to its position and to the acoustic properties of the listening room: the ABC room adaptation system. Adaptive Bass Control (ABC) measures the acoustic radiation resistance seen by the bass drive unit and calculates the coefficients of a digital IIR filter, which is inserted in the signal path before the power amplifier. This enables ABC to provide a room adaptation, which is globally valid throughout the listening room, i.e., all listening positions benefit from this system. DSP is used to measure the radiation resistance, calculate the filter target, and filter coefficients and to implement the IIR filters.

3-2 **Listening room compensation for Wave Field Synthesis. What can be done?—***Etienne Corteel*[1], *Rozenn Nicol*[2]

[1]IRCAM, Paris, France
[2]France Telecom, Lannion, France

Interaction of a reproduction system with the listening room introduces distortions of the audio content. In this paper we present the key points that have to be addressed for compensating the listening room effect. The paper is focused on the particular situation of sound reproduction using Wave Field Synthesis (WFS). As a volume solution, WFS reproduces a given sound field in an extended area covering the whole listening room. Therefore, room compensation cannot be considered anymore for limited and discrete positions. The paper proposes solutions for achieving efficient treatment in an extended zone.

3-3 Statistical Analysis of an Automated In-Situ Frequency Response Optimisation Algorithm for Active Loudspeakers—*Andrew Goldberg, Aki Mäkivirta*, Genelec Oy, Iislami, Finland

This paper presents a novel method for automatically selecting the optimal in-situ acoustical frequency response of active loudspeakers within a discrete-valued set of responses offered by room response controls on active loudspeakers. An overview of optimization techniques is given, the resulting optimization algorithm described, as is the rationale of the room response controls for the active loudspeakers. The frequency response, calculated from the acquired impulse response, is used as the input for the optimization algorithm to select the most favorable combination of room response controls. Examples are given and the performance of the algorithm is analyzed and discussed. This system has been implemented and is currently in active use by specialists who set up and tune studios and listening rooms.

3-4 The Loudspeaker-Room Interface—Controlling Excitation of Room Modes—*Rhonda J. Wilson, Michael D. Capp, J. Robert Stuart,* Meridian Audio Limited, Huntingdon, UK

Room modes occur due to the superposition of standing waves. In the low frequency range modal overlap tends to be negligible, and some modes can be particularly dominant. When listening to audio material in a room with strong room modes unwanted characteristics are observed, such as significant reverberation, boominess at particular frequencies, and apparent pitch changes as a tone at one frequency excites and is then dominated by a strong resonance at a slightly different frequency. These undesirable audible effects can be considerably reduced by prefiltering the signal. This paper derives the pre-filter required and investigates the psychoacoustic criteria required to optimize the pre-filter.

3-5 Modal Equalization by Temporal Shaping of Room Response—*Matti Karjalainen*[1], *Poju Antsalo*[1], *Vesa Välimäki*[1], *Aki Mäkivirta*[2]
[1]Helsinki University of Technology, Espoo, Finland
[2]Genelec Oy, Iislami, Finland

The low-frequency behavior in listening rooms is often problematic due to long-ringing modes that are difficult and expensive to control by acoustic means. Modal equalization has been proposed recently to correct the low-frequency reproduction problems by shortening the decay times of problematic modes through modification of transfer function poles. While the previous methods were based on the estimation of isolated modes and their parameters, the new method proposed here is a technique to change the time-domain response more directly. It is an advanced windowing technique where the temporal shaping of a given impulse response can be done in a frequency-dependent manner. The method is compared with previous modal equalization techniques.

3-6 Semi-classical Approximation of Loudspeaker Positioning in Room—*Jean-Dominique Polack*[1], *Jan Abildgaard Pedersen*[2]
[1]Laboratoire d'Acoustique Musicale, Paris, France
[2]Bang & Olufsen A/S, Struer, Denmark

The field radiated by a loudspeaker is sensitive to the loudspeaker placement, specially at low frequencies, because the room exerts an acoustical load on the membrane. One of the authors has based on it a compensation system called ABC. The paper focuses on the simulation of the acoustical load seen by the source. The method used is the semi-classical theory; that is, a variant of ray tracing that takes phases into account. It builds up the coherent part of the point transfer function—or acoustical load—for different source positions and the ratios give the compensation filters. The computed filters are compared to measurements obtained by the ABC system. The same overall shapes are obtained. Discrepancies subsist due to the finite size of the loudspeaker and phase shifts at the reflections on the walls not accounted for at present in the definition of absorption and reflection coefficients. So far, the simulation has been developed for rectangular rooms and point sources only.

1:30 pm–4:30 pm

SESSION 4: CREATING SPACE WITH DSP

4-1 Spatial Post-Processing of Audio Recordings—*Jean-Marc Jot*, Creative Advanced Technology Center, Scotts Valley, CA, USA **(Invited)**

This paper reviews signal processing methods for improving the presentation of two-channel and multichannel recordings by spatial processing in consideration of the playback system. The goal is to enhance the listener's experience by attempting to address limitations of the playback system or recording format, while preserving the original audio content as faithfully as possible. The approaches reviewed here include: (1) artificial reverberation, (2) virtualization of recordings for playback over headphones or two loudspeakers, and (3) up-mix or stereo widening of two-channel recordings for playback over two-channel or multichannel loudspeaker systems.

4-2 3-D Audio Communications Services for Future Mobile Networks—*Yasuyo Yasuda*[1], *Tomoyuki Ohya*[1], *David McGrath*[2], *Patrick Flanagan*[2]
[1]Multimedia Laboratories, NTT DoCoMo, Inc., Yokosuka, Kanagawa, Japan
[2]Lake Technology Ltd., Ultimo, NSW, Australia

3-D audio has been suggested as a way to augment communication and provide unique services in future mobile networks. Concepts of the services 3-D audio can provide are presented. For concept demonstration, a prototype wireless 3-D audio system, the Tohya system, has been built and evaluated. Tohya realizes a wireless multiuser virtual 3-D audio communication space, overlaid on the actual space the user inhabits. Each user of Tohya is equipped with a microphone, headphone, and a PDA with wireless networking. The Tohya system tracks the user's position and head orientation. Users communicate with other remote users in an overlaid (augmented) 3-D audio space. 3-D audio processing should allow mobile communications systems to evolve from voice communications to rich multimedia-multidimensional services.

4-3 **What Shall We Do with the Center Channel?—***Attila Kiss, Istvan Matok*, Digital Pro Studio, Budapest, Hungary

The first stereophonic recording systems used three channels: left, right and center. The stereophonic LP and FM broadcast systems couldn't be able to handle the center channel. The surround movie soundtracks regenerated the center channel. Now, the multichannel sound is more than movie soundtracks, because the new high-resolution systems (SACD, DVD-Audio) are give new possibilities to these formats. In the big Movie Theater, the center channel playing great part in the perfect localization. But what part plays in the acoustical music recordings, and in the home cinema practice? How can fit the mono center channel into the stereophonic image? What will happen after the downmixing? (3/2 to 3/1 Pro Logic or 2/0) We tested some microphone setups and chose one for our recordings. We recorded an church choir in multichannel format. We made some versions of 3/2 mixing We asked the listeners test group, to chose the most natural version. We made downmixes with several coefficients and asked the listeners to describe the changes in the sound, and choose the perfect downmix.

4-4 **Coloration in Natural and Artificial Room Impulse Responses—***Per Rubak, Lars Johansen*, Aalborg University, Aalborg, Denmark

A literature review concerning perception of coloration is presented. In the paper we also present a review of general psychoacoustic background material concerning auditory modeling including perception of timbre and amplitude modulation. Also some classical room acoustic measures for echo perception are considered. Finally some of the simple engineering measures to rate coloration is evaluated, and an auditory signal processing model including a combined time-frequency detection criteria is presented.

4-5 **Spatial Sound Encoding Including Near Field Effect: Introducing Distance Coding Filters and a Viable, New Ambisonic Format—***Jérôme Daniel*, France Telecom, Lannion, France

Higher order ambisonics have been increasingly investigated in the past years and found promising as a rational, scalable, and flexible way to encode, transmit, and render 3-D sound fields. Nevertheless, studies concerning virtual source imaging or natural 3-D sound encoding mainly focussed on the directional encoding of plane waves and neglected the near field effect of finite distance sources though its presence in any ordinary sound field. This paper highlights that with near field, the infinite bass-boost affecting ambisonic components makes the currently accepted format unviable. By introducing from the encoding stage a near field compensation of reproduction loudspeakers, a viable, modified ambisonic format is defined, distance-coding filters are designed, and higher order ambisonic recording and synthesis become practicable.

Sunday, May 25 — 9:00 am–11:00 am

SESSION 5: DSP IN LOUDSPEAKERS—PART 1

5-1 **Applications of DSP for Sound Reproduction Improvement—***Ronald Aarts*, Philips Research, Eindhoven, The Netherlands **(Invited)**

Today and tomorrow's audio and video, portable audio, and multimedia applications put increasing demands on sound reproduction techniques. On one hand there is a need for reductions in both cost and size, on the other hand we wish to enhance the experience of the user beyond today's possibilities. A good sound reproduction system is, in general, in conflict with the boundary conditions for consumer products both by size as well as by price requirements. A possible way to ease these conflicts is to enhance the reproduction and perception of sound for listeners by exploiting the combination of psychoacoustics, loudspeaker configurations, and digital signal processing. Various examples will be given, such as virtually increasing the distance between loudspeakers (stereo base widening), increasing the perceived bass response of loudspeakers, and increasing the number of loudspeaker channels (converting stereo to multichannel sound). When multichannel reproduction through loudspeakers is not a viable option, the same precept can be simulated over headphones. This method, using active noise control principles, is discussed as well.

5-2 **Towards the All-Digital Audio/Acoustic Chain: Challenges and Solutions—***Nicolas-Alexander Tatlas, Andreas Floros, Panagiotis Hatziantoniou, John Mourjopoulos*, University of Patras, Patras, Greece

The obvious advantages of digital audio technology have up to now being manifested mainly in media storage and processing sub-components, witch are parts of a more elaborate audio / acoustic analog reproduction chain. It is envisaged that the remaining components such as cables, amplifiers and transducers will soon be also implemented in digital form, potentially leading to networked, integrated and highly optimized solutions. The paper examines theoretical and implementation aspects related to each of the modules that could constitute such an all-digital audio / acoustic transducer, namely: (a) the digital wireless receiver, via the Bluetooth and IEEE 802.11 protocols, (b) audio decoding and format adaptation (c) DSP for acoustic compensation, based on measured results for smoothed response equalization, (d) digital amplification, and, (e) all-digital transduction.

5-3 **Signal Processing in Sound Reinforcement Engineering—***Peter Mapp*, Peter Mapp Associates, Colchester, Essex, UK

Traditionally, there are typically four main areas in which signal processing plays a part in a typical sound reinforcement or PA system. These are : (1) Input signal processing eg pre-amplification and equalization (2) Routing and control (including insert of effects processing) (3) System response tuning, including equalization and signal delay (4) Loudspeaker Processing, eg crossovers, response equalization/alignment and overload drive protection. Only a few years ago, each of the above processes employed a dedicated processor, however, current DSP capabilities will reduce the box count considerably and indeed it is possible to contain all the above, and even include the mixer all in one powerful DSP unit. As it is anticipated that other papers at this conference will be addressing vocal processing and use of signal processing in studios etc, this paper is more directed towards overall system response tuning and processing. However, processing related to intelligibility enhancement is also addressed. Although the paper primarily discusses permanent sound reinforcement and public address installations, many of the issues also directly relate to temporary sound systems and concert sound systems.

1:00 pm–3:00 pm

SESSION 6: DSP IN LOUDSPEAKERS—PART 2

6-1 Active Compensation of Transducer Nonlinearities—*Wolfgang Klippel*, Klippel GmbH, Dresden, Germany **(Invited)**

Nonlinearities inherent in electromechanical and electroacoustical transducers produce signal distortion and limit the maximal amplitude of the output signal. Assessing the large signal performance has been a subject of acoustical research for many years providing nonlinear models and new methods for the measurement of the large signal parameters. The identified model allows prediction and simulation of the nonlinear behavior and direct comparison with measured symptoms. The good agreement between model and reality is the basis for developing novel digital controllers dedicated to transducers that compensate actively for nonlinear distortion by inverse preprocessing of the electrical input signal. This paper gives a summary on the activities in the last 15 years and new challenges of the future.

6-2 Simplified Loudspeaker Distortion Compensation by DSP—*Andrew Bright*, Nokia Group, Helsinki, Finland

A discrete-time algorithm for compensation of nonlinear distortion in loudspeakers is presented. Discrete-time nonlinear loudspeaker models and their use in developing a compensation algorithm are presented. The advantages of these models, particularly with respect to updating from system identification, are discussed. It is shown that this algorithm can be used to compensate for distortion caused by shortened-height voice coil loudspeakers, resulting in higher overall electroacoustic transduction sensitivity.

6-3 State of the Art Digital Pulse Modulated Amplifier System—*Steen Munk*, Bang & Olufsen A/S, Struer, Denmark

Pulse modulated amplifiers (PMS's or Class D amplifiers) have won great interest during the past years. This is mainly because technology is now so advanced that such amplifiers simply have excellent performance! This paper describes advances made by Bang & Olufsen ICEpower a/s within PMA's having pulse code modulated interface. ICEpower amplifiers use feedback loops to compensate for the nonideal characteristics of the power stage. This compensation, the PEDEC (pulse edge detection and error correction), is motivated. A new approach WPWM (Weighted PWM) gives an efficient approximation to NPWM. The paper gives a comparison of WPWM versus other modulation techniques. Last, the intelligent volume control (IVC) is introduced.

TRANSPORTATION

As the major hub in Northern Europe, Copenhagen can be easily reached by air, in most cases non-stop from the major cities in Europe, and from many cities around the world. It is also easily reached by road, and new road and railway bridges connect the island to mainland Europe and Sweden. Excellent fast train services connect most major European cities to Copenhagen. Helsingør is situated 45 km north of Copenhagen. Direct trains to Helsingør run every 20 minutes from the station below Copenhagen International Airport and from the central railway station. Taxis are available round the clock. Car rental is available through all major rental companies. For further details see www.aes.org/events/23.

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