AES 32nd INTERNATIONAL CONFERENCE

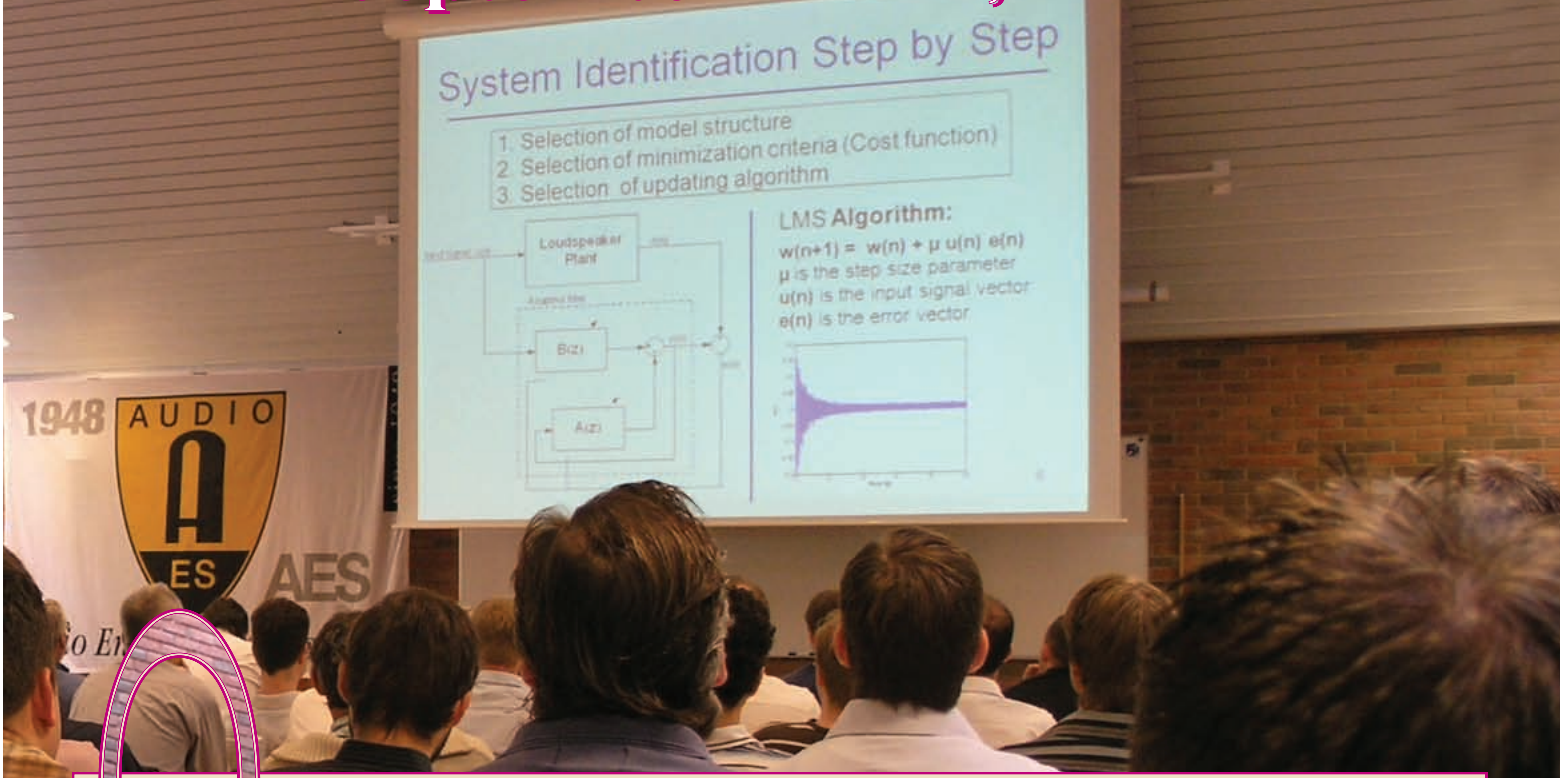
Jan Abildgaard Pedersen
conference chair

Wieslaw Woszczyk
AES president

Stanley Lipshitz
keynote speaker

DSP for Loudspeakers

Hillerød, Copenhagen, Denmark
September 21–23, 2007



Once again Denmark hosted another AES conference, this time on the topic "DSP for Loudspeakers." The conference was a three-day event held at the excellent Pharmakon Conference Center. 125 engineers from more than 20 countries around the world traveled to this lovely area of North Zealand, a short train ride northwest of Copenhagen. The conference offered a great combination of technical papers, workshops, technology demonstrations, informal discussions, and social events. The large number of delegates participating in the conference clearly showed that "DSP for Loudspeakers" is a significant area of interest within the audio engineering community.

Digital signal processing (DSP) offers loudspeaker system designers a very powerful set of tools. It may be as simple as a delay line, which hardly needs DSP anyway. Or it may be a precise and detailed filtering with just the desired magnitude and phase characteristics. Such filtering may be used to construct a single very accurate loudspeaker, or used for room correction, or it may be used to create beam-steering arrays or even wavefield synthesis. In addition to filtering, digital signal processing may be used for distortion compensation or for enhancing the subjective performance by means of nonlinear processing. On the analysis side, DSP can be used to monitor in real time the state of the loudspeaker and model important driver parameters such as voice coil temperature. Throughout the last couple of decades, AES members have been in the forefront of this research.

Meanwhile, the maturity and price/performance ratio of essential system components such as processors, software development tools, AD/DA converters, and power amplifiers have reached a point where it makes commercial sense for a large part of the audio industry to apply loudspeaker-specific DSP in their products. Indeed, there are already classes of products completely dependent on it.

Jan Abildgaard Pedersen, conference chair, and his committee assembled a conference program that presented an overview of the current state-of-the-art in a broad perspective and addressed many of the scientific disciplines involved in this emerging field. Knud Bank Christensen, papers chair, coordinated the technical program of 21 papers in 7 sessions.

DAY 1

Jan Abildgaard Pedersen opened the conference on Friday by welcoming everyone to Denmark and expressing his plea- ➥

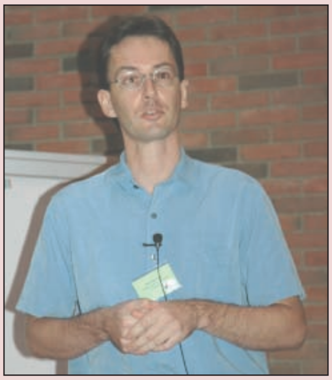
Authors

Andrew Bright

Wolfgang Klippel

Sofus Birkedal Nielsen

Bo Rohde Pedersen



Finn Agerkvist

Fares El-Azm

Thomas Sporer
invited speaker

Ulrich Horbach

Marshall Buck

Jaeyoun Cho

Matti Karjalainen
invited speaker

David Gunness

sure with the attendance of 125 delegates. He pointed out that it takes approximately two years of planning and preparation to organize an AES International Conference. He then introduced the other members of the committee: Knud Bank Christensen, papers chair; Thomas Mørch, secretary; Eddy Bøgh Brixen, facilities; Subir Pramanik, treasurer; Preben Kvist, webmaster; and Ole Juhl Pedersen, assistant. After explaining the schedule for the the next three days, Pedersen thanked AES Headquarters in New York, represented by Roger Furness, executive director, for their valuable and professional help in preparing the conference. Then Wieslaw Woszczyk, AES president, added his words of welcome and pointed out that AES international conferences play an important role for the Society as venues for high-level scientific exchanges in highly focused areas of research.

The keynote address was given by Stanley Lipshitz of the University of Waterloo, who spoke on the topic “The Loudspeaker in the Digital Age.” He started by taking the audience back to the introduction of the electrodynamic loudspeaker in 1877. He moved on to binaural audio, stereo, ambisonics, and wavefield synthesis as examples of different ways in which signals from soundfields generated by loudspeakers reach the ears of listeners. Lipshitz went on talking about the directivity of monopole, dipole, cardioid, and and other types of loudspeakers. He mentioned that room correction, phase response, and nonlinearities as very interesting applications of DSP for loudspeakers.

Following the keynote address, the first papers session was devoted to adaptive systems. Jan Abildgaard Pedersen, Lyngdorf Audio, presented the initial paper, “Fully Auto-

32nd Conference Committee:
from left,
Jan Abildgaard Pedersen, chair;
Knud Band Christensen, papers chair;
Eddy Bøgh Brixen, facilities chair;
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and Subir Pramanik, treasurer

matic Loudspeaker-Room Adaptation—The RoomPerfect System." This system for DSP-based room correction is based on measuring the sound pressure at the listening position and in at least three random positions scattered across the entire listening room. It enables a fully automatic calculation of a target function, which preserves the basic characteristics of the loudspeaker being used.

Fares El-Azm, Lyngdorf Audio, continued with the paper "Natural Timbre in Room Correction Systems (Part II)," in which he describes how a certain part of the increased level at low frequencies due to the influence of a room is natural to the human ear and should not be removed by room-correction systems.

Next Sofus Birkedal Nielsen, Aalborg University, presented "Time Based Room Correction System for Low Frequencies Using Multiple Loudspeakers." In this system 4 low-frequency loudspeakers are placed in specific positions 2 at the front end wall and 2 at the rear end wall. The time domain is used to illustrate how the reflections at the rear end wall are cancelled, leaving a much more even sound pressure distribution in the room.

"Adaptive IIR Filters for Loudspeaker Parameter Tracking," by Andrew Bright, Nokia, concluded the session on adaptive systems. He explained how IIR filters are used in a simple method for in-situ tracking of loudspeaker parameters as they change over time. In particular he investigated the convergence properties and showed a method to optimize convergence time and residual error.

Session 2 focused on nonlinear processing. In the first paper "Modelling Loudspeaker Non-Linearities," Finn Agerkvist, Technical University of Denmark, compared different techniques for modeling the nonlinear parameters of the electrodynamic loudspeaker. He showed that inverse polynomial expansion and localized fitting functions are superior to polynomial expansion.

The second and final paper in this session, "Optimal Design of Loudspeakers with Nonlinear Control," was presented by Wolfgang Klippel, Klippel GmbH. He showed how nonlinear control opens up new degrees of freedom in passive driver design by exploiting the gained efficiency of loudspeaker drivers with nonlinear motors, such as by using a short voice coil overhang. Klippel presented the results of an investigation of the behavior of real loudspeakers under control and the obtained design criteria for active loudspeaker systems.

The first day of the conference was concluded by a combined poster session, product demonstrations, and tabletop demos. Anthony McMahon, Alba Centre, presented the paper "Design and Implementation of a DSP Enhanced Portable Speaker System" in the poster session. McMahon described how DSP enables an enhanced performance of low-cost portable loudspeaker systems and how it can be designed and implemented.

The first of the three demo rooms was used by Lyngdorf Audio to present and demonstrate its RoomPerfect room-correction system together with true-digital amplifiers and compact dipole loudspeakers combined with boundary woofers. Listeners had a chance to evaluate both a global filter and a focused filter. The focused filter is optimal for a specific position, but still uses the information of the entire 3-D soundfield so that everyone in the demo room was able enjoy the same sound from the focused filter. The global filter enabled people to walk around the demo room and hear virtually the same sound quality in all positions within the room.

WaveCapture used another demo room to show its great variety of software products: Room-Capture for room installations, Live-Capture for FOH measurements, EQ-Capture for parametric EQ, Audio-Capture for audio component measurement, and RT-Capture for SPL management.

Sonion used the third demo room to show its newest inventions in microphones and receivers. Sonion works with

DEMOS

Alba Campus, Danville Signal Processing, Four-Audio, Klippel GmbH, Lyngdorf Audio, Oxford Digital, Sonion, and WaveCapture offered product demonstrations during the conference.

miniature components to manufacture and market electroacoustical and electromechanical solutions within hearing instruments, mobile terminals, headsets, and medical devices.

During this evening part of the program, there were also tabletop demos in which the authors were able to discuss in more detail aspects of their papers. These tabletop demos sparked many in-depth discussions among the attendees. The relaxed, informal nature of the event fostered networking and allowed the attendees to make new professional connections.

Oxford Digital discussed its products, research and development work, and the consultancy services it offers in a tabletop demo. Alba Campus, during its demo, provided further information from its paper presentation on how DSP enables an enhanced performance of low-cost portable loudspeaker systems.

Wolfgang Klippel, Klippel GmbH, showed his tools for designing, assessing, and diagnosing passive transducers. He also demonstrated the DSP used in active loudspeaker systems as described in his paper presentation on the same topic. Four-Audio presented information on its loudspeaker management system on a multirate platform with FIR and IIR filters. The system also includes dual-range AD conversion, look-ahead peak-limiters, and RMS limiters.

The tabletop demo of Danville Signal Processing and DSP Concepts demonstrated a new DSP crossover platform based on SHARC audio modules and a graphical design tool. The platform enables loudspeaker designers to implement a variety of crossover designs combined with optimized audio processing functions, such as time delays and parametric EQs.

DAY 2

The second day of the conference started with a session on measurement techniques. The first paper, "Musical Transducer-Less Identification of Linear Loudspeaker Parameters," was presented by Bo Rohde Pedersen, Aalborg University. He described how the voice-coil current was used as feedback from the loudspeaker for system identification in a transducer-less measurement system. In this way he generated information on the voice-coil resistance, resonance frequency, and damping factor.

Ho Young Sung, Samsung Electronics, presented the paper "A Method for Objective Sound Quality Evaluation of Headphones," in which he describes how the characteristics of headphones are different from loudspeakers and how objective factors that play a key role in sound quality are carefully selected.

There were four papers in the next session on filter structures and design algorithms. The session began with the invited paper "Special Digital Filters for Audio Reproduction" presented by Matti Karjalainen, Helsinki University of Technology. He gave a great overview of digital-filter categories that have been found important in audio reproduction, covering in particular fractional delays, frequency-warping techniques including Laguerre and Kautz filters, as well as filters used for physics-based modeling,

including wave filters and digital waveguides. He also talked about the applications of these filter categories in audio reproduction.

David Gunness, Eastern Acoustic Works, presented the paper "Optimizing the Magnitude Response of Matched Z-Transform Filters (MZTI) for Loudspeaker Equalization." He explained how MZT filters deviate from analog filters and how these filters are optimized in a way displaying excellent agreement with the analog target filters, both in magnitude and phase response.

D. Sookcharoenphol, KMITL, presented "Realization of Linear Phase Crossover Loudspeaker Network Using IIR Filters," in which he described how a group of IIR filters have their impulse responses truncated to finite length to achieve linear time-invariance. The system provides an overall flat magnitude and group delay responses and also higher slope and stopband attenuation than prototype elliptic IIR filters.

What DSP Can Do for Loudspeakers workshop: from left, Knud Bank Christensen, David Gunness, Wolfgang Klippel, Jan Abildgaard Pedersen, and Thomas Sporer

In the final paper of the session on filters, Rainer Thaden, Four Audio, presented "Loudspeaker Management System with FIR/IIR Filtering." He described how a combination of FIR and IIR filters can be freely combined in each output channel. The system also includes dual-range AD conversion, look-ahead peak-limiters, and RMS limiters.

The final paper session on Day 2 of the conference was entitled "System Design with DSP." Marshall Buck, Psychotechnology Inc., presented "Acoustic and Transducer Considerations for DSP Loudspeakers." He described how a number of acoustic and transducer errors must be minimized to maintain high accuracy in DSP loudspeakers. A two-way DSP-calibrated studio monitor was used as an example to illustrate and discuss these error sources and their correction.

Ulrich Horbach, Harman Consumer Group, presented the paper "Application of Linear-Phase Digital Crossover Filters to Pair-Wise Symmetric Multi-Way Loudspeakers—Part 1: Control of Off-Axis Frequency Response." He explained a simple, noniterative linear-phase crossover-filter design technique that provides uniform frequency responses vertically off-axis for a given multiway loudspeaker. Part 1 of this paper presents a general method that emphasizes the flatness of arbitrary off-axis frequency responses. Horbach continued with the presentation of "Part 2: Control of Beamwidth and Polar Shape" of this paper. He discussed an alternative design procedure based on specifying the total shape and coverage angle (vertical beamwidth) of the polar patterns generated by pairs of separated point sources.

After a full day of paper presentations the delegates were invited to attend a social event—an excursion to two Danish microbreweries, at which the delegates were introduced to a different part of Danish cultural heritage. Known as the "Champagne of the North," beer has long been an important food group in Denmark, and local microbreweries are experiencing a robust and remarkable renaissance based on the high quality of the product. The visits started with technical tours of the facilities and ended with the subjective evaluation of their products!

Later in the evening the delegates enjoyed a banquet dinner of gourmet foods with fine wines. Music was provided by the Danish band Blast Sisters.

DAY 3

The final papers session of the conference was devoted to spatial processing. The day started with an invited paper, "SHAPES—A Scalable Parallel HW/SW Architecture Applied to Wave Field Synthesis," presented by Thomas Sporer, Fraunhofer IDMT. He discussed a new parallel processor architecture and the first steps toward an adequate optimization of wavefield synthesis. Sporer also discussed a software development environment that assists in creating scalable programs for highly parallel hardware.

Andreas Franck, Fraunhofer IDMT, continued with the paper "Reproduction of Moving Sound Sources by Wave Field Synthesis: An Analysis of Artifacts." He described how wavefield synthesis can offer spatial audio reproduction over an extended listening area, which, however, is also characterized by a number of distinct, audible artifacts. Franck classified and described these artifacts, explained their causes, and discussed means to reduce audible deviations.

Paolo Martignon, University of Parma, presented "A Digitally Controlled Two Dimensional Loudspeaker Array." He described a "Sonic chandelier" system, which is a 64-channel, dome-shaped array of 228 speakers. Martignon covered the physical structure and algorithm design as well as some listening tests.

The final paper of the conference, "A Sound Sources and Reflection Localization Method for Reverberant Rooms Using Arrays of Microphones," was presented by Simon Roper, University of Birmingham. He described the application of compensation of stereo or multichannel sound reproduction systems for the variation in room acoustics and nonideal placement of loudspeakers, which normally requi ➥

AES president Wieslaw Woszyczk offered a toast at the conference banquet, and the Blast Sisters provide the musical entertainment.

On Sunday evening attendees toured two Danish microbreweries, learning about the production techniques and tasting the product.

knowledge of the room geometry and the acoustic impedance of the boundaries. The results were obtained by estimating the image locations and amplitudes of a source.

The workshop *What DSP Can and Cannot Do for Loudspeakers* served as the technical conclusion of the conference. The panelists were David Gunness, Eastern Acoustic Works, Wolfgang Klippel, Klippel GmbH, Jan Abildgaard Pedersen, Lyngdorf Audio, and Thomas Sporer, Fraunhofer IDMT. Each of the panelists gave his view on good and bad applications of DSP for loudspeakers, and the audience participated by asking many good questions and offering additional comments for the panel.

The conference was brought to a close by Jan Abildgaard Pedersen, who thanked his organizing committee, the invited speakers, all the authors, the sponsors, the AES Headquarters, the exhibitors, and all the delegates for contributing to the conference. Roger Furness, AES executive director, thanked Pedersen and his hardworking team for producing such a highly successful international conference.

Editor's note: The conference proceedings book can be purchased at <www.aes.org/publications/conf.cfm>; a PDF of the book can also be purchased.