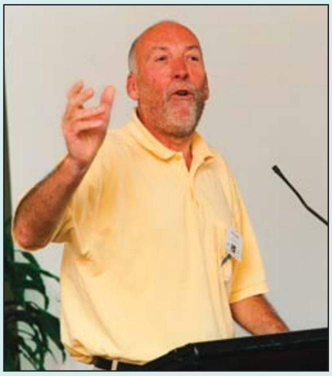
AES 23rd INTERNATIONAL CONFERENCE

Signal Processing in Audio Recording and Reproduction

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

Jeff Bier, keynote speaker



Kees Immink, AES president

Per Rubak, conference chair

The town of Helsingør is situated on the northeast corner of Denmark overlooking the narrowest part of the Øresund, a busy waterway that links the North Sea to the Baltic Sea. The history of the town can be traced back to 70 AD, and the area contains a number of impressive royal castles dating from as early as 1100 AD. Against this historic backdrop, from May 23rd to the 25th, the AES held its 23rd International Conference, *Signal Processing in Audio Recording and Reproduction*, covering some of the very latest techniques in signal processing for audio.

The conference committee worked hard to ensure that the conference was a success. Per Rubak as conference chair and Jan Abildgaard Pedersen and Lars Gottfried Johansen as papers cochairs drew together over 20 excellent papers covering many aspects of signal processing. They were ably assisted by Knud Bank Christensen, conference secretary, Eddy Bøgh Brixen, facilities chair, and Subir Pramanik, treasurer.

The conference was held in the Marienlyst Hotel and Conference Center, overlooking the Øresund and the shore of Sweden across the water. Included in the program was a mix of papers sessions, demonstrations, and social events. In addition to these, there were plenty of opportunities for the delegates to discuss hot topics and to contribute to the global community that is the Audio Engineering Society.

OPENING

Conference Chair Per Rubak opened the proceedings by giving an overview of the wide range of applications that were possible through the increasing use and availability of digital signal processing technology. He reflected on the great amount of progress that has been achieved in this area so far and hoped that the conference would provide inspiration for further progress.

AES President Kees Immink thanked the hard-working committee for producing a successful conference. He stated that it was an odd place to hold an audio event, as the reigning king of Denmark had introduced a tax on the Sound in 1429. He was, of course. talking about the tax levied on ships passing through the Øresund (Sound). He encouraged the delegates to take advantage of the conference format by conferring with each other to gain from the international expertise and knowledge of those present.

SIGNAL PROCESSING HARDWARE

The keynote speech, given by Jeff Bier of Berkeley Design Technology, was an overview of trends in signal pro-

INVITED AUTHORS

Clockwise, from bottom left: Wolfgang Klippel, Jean-Marc Jot, Ronald Aarts, and Patrick Bastien.

cessing hardware. He covered different types of signal processing hardware that may be used for audio purposes, considering not only the processing capability but also the practical and commercial aspects of each option. He explained that one of the main decisions that has to be made when choosing a suitable processor is the compromise between efficiency and flexibility, and that the optimum choice can differ greatly depending on the specific application. Bier summarized a number of important trends in the consumer marketplace—including the convergence of multimedia applications into other devices and the increasing connectivity between devices. "These are interesting times," he stated, because of the increasing ubiquity of digital audio, the change in emphasis from hardware to software, and the capabilities and challenges of increasing connectivity. He forcast that we will see a great increase in the use and development of the techniques discussed at the conference.

SIGNAL CONVERSION

The afternoon session of the first day started with a paper by Søren Nielsen and Thomas Lund of TC Electronics, focusing on the topic of overload in signal conversion. It was an interesting study into the clipping that can occur even when the digital sample peaks are below the full range afforded by the system. The authors suggested a measurement technique to evaluate this problem and showed results from a number of processes and commercial devices. They finished by summarizing that the problem could be avoided by reducing the level either on the recorded media or prior to conversion.

SPEECH PROCESSING AND METADATA

This session started with an invited paper by Patrick Bastien of TC-Helicon on voice-specific signal processing tools. He explained some of the unique characteristics of voice signals that have to be considered when processing and gave a number of entertaining and informative audio demonstrations to illustrate the potential pitfalls and solutions. He concluded by showing a complex voice-modeling system that could alter and enhance vocals, but he conceded that it could not yet turn Joe Cocker into Britney Spears.

Continuing the topic of speech processing, Niels Henrik Pontoppidan and Mads Dyrholm of the Technical University of Denmark covered the problem of separating multiple speech signals captured with one receiver. Demonstrations of the system they developed showed that even though the resulting audio quality was not of a high standard, the speech signals were sufficiently separated to allow further analysis and processing.

The final paper of the day was presented on video tape, because invited author Elizabeth Cohen of UCLA was not able to travel to the conference. In the presentation she discussed the value of metadata and stressed the importance of creating metadata at the same time as the content so that it can accompany the audio signal through the entire chain from creation to product.

INTERFACING LOUDSPEAKER AND ROOM

One of the major emerging trends in signal processing for sound reproduction that was discussed at the conference was compensation for a less than ideal interaction between the loudspeaker and the room. Saturday morning's session was devoted entirely to this topic, covering a range from modal equalization to full room compensation. One of the common themes throughout these presentations was the importance of the spatial robustness of the processing, to ensure that any attempts to improve the sound at a single listening position do not degrade the sound anywhere else in the room.

In the first presentation of the session, Jan Abildgaard Pedersen of Bang & Olufsen noted that loudspeaker manufacturers go to great trouble to optimize a large number of parameters. However, the final reproduction room of the customer has a large effect on the sound. But since the properties of the room can vary widely and are unknown by the manufacturer, they cannot be compensated for in a static design. A solution to this dilemma is the use of active compensation, where the processing is adapted to take into account the effect of an individual room and even a specific loudspeaker position within that room. Pedersen described one method that involves measurement and compensation for each loudspeaker. He explained that this can be done by including a microphone in the loudspeaker cabinet to make measurements of the loudspeaker radiation resistance for two receiver positions. This information can then be used to correct the frequency response from 20 to 500 Hz.

A Bang & Olufsen loudspeaker with this technology was shown in a small demonstration room during the confer-

Attendees commented that the quality of the papers presentations was consistently high throughout the entire conference; questions and comments from the audience enhanced the exchange of technical information.

ence. The loudspeaker was used for replay in more than one position in the room, exhibiting the problems of the position-dependent interaction between the loudspeaker and the room. The active compensation was then demonstrated by aligning the loudspeaker in each position using the method described in the presentation. This resulted in the pair of loudspeakers in different positions having a more similar timbre.

Two further papers on the equalization of room modes were given by Rhonda Wilson and Michael Capp of Meridian Audio and Matti Karjalainen, Poju Antsalo, and Aki Mäkivirta of Helsinki University of Technology and Genelec. The common aim of these papers was to reduce the decay time of low-frequency room modes, though each used different approaches to analyze and filter the most prominent modes in a given reproduction room.

The application-based papers on the subject of modal equalization were supported by a theoretical paper presented by Jean-Dominique Polack of the University of Paris and coauthored by Jan Abidgaard Pedersen. They used semiclassical theory to explain and predict the creation of room ➥

Coffee breaks and meals gave attendees time to relax, catch up with old friends, and make new ones.

modes and the correction required for loudspeakers at different positions. Comparison of measured and predicted results showed that the approximation was reasonable, but that further refinements could be used to improve the accuracy of the predicted results.

Other papers in the session covered a range of topics related to the interaction between a loudspeaker and the room. Andrew Goldberg and Aki Mäkivirta of Genelec presented a paper that described an automated method to determine the correct equalization settings on an active loudspeaker in a given room based on the measured frequency response. By the use of heuristic analysis developed from knowledge gained from manual alignment, the complexity of the process was greatly reduced, meaning that the system could compute the optimum settings in a short time.

A paper by Etienne Corteel of IRCAM and Rozenn Nicol of France Telecom focused on the problems that the reproduction room can cause for wavefield synthesis (WFS) and how the properties of the WFS system can be used to compensate for this. By the use of simulations they showed that ➥

Jan Pedersen, right, demonstrates new loudspeaker technology of Bang & Olufsen.

Wolfgang Klippel, left, demonstrates practical systems for loudspeaker measurement and active compensation.

Some of the attendees congregated at the conference center entrance before the short walk to Helsingør Castle.

Saturday evening attendees were treated to a guided tour of historic Kronborg Castle, immortalized by Shakespeare in *Hamlet*.

WFS can be used to cancel early reflections in the horizontal plane over a wide listening area, though only up to the spatial aliasing frequency of the reproduction system.

CREATING SPACE WITH DSP

The Saturday afternoon session focused on using signal processing to simulate and reproduce spatial properties. An invited paper by Jean-Marc Jot and Carlos Avendano of the Creative Advanced Technology Center considered the problem of combining sources and reproduction systems with a wide range of spatial characteristics (mono, 2-channel, and 5.1 surround). A number of conversion techniques were discussed, including headphone virtualization, stereo widening, upmixing, and downmixing. To support the presentation a separate demonstration was given that compared a number of different techniques for creating a 5-channel surround

Subir Pramanik (far left), conference treasurer, and Roger Furness, AES executive director, welcomed everyone at Saturday night's banquet.

ing digital signal processing. He finished by suggesting a method for simulating the distance of the sound source together with a means of transmitting the parameters to ensure that the sound is rendered correctly.

KRONBORG TOUR AND BANQUET

Following the final session of the day, the delegates were treated to a tour of Kronborg Slot, a large castle built in the Dutch Renaissance style. It is one of the largest and most extravagant castles of the period. The tour guides explained that the castle was built to defend the Øresund and to show off the wealth of Denmark. It is better known in English-speaking countries as the Elsinore Castle immortalized in Shakespeare's *Hamlet*. It is not known whether Shakespeare ever visited the castle, though he could have heard descriptions of the castle and the traditional Danish tale on which *Hamlet* is based from the links between the royalty of Denmark and England.

The tour of the castle was followed by a banquet that included an excellent musical performance by two members of the Danish National Symphony Orchestra, Klaus Tönshoff on clarinet and Per Salo on piano. During the banquet Roger Furness, AES executive director, again thanked the organizing committee for the hard work that went into making the conference so successful. After the banquet the delegates had coffee in a lounge overlooking the Sound where they viewed a spectacular display of lightning, for which Facilities Chair Eddy Bøgh Brixen would have liked to claim credit, along with the rest of the faultless organization.

sound signal from a 2-channel stereo original.

The topic of downmixing was continued in a paper by Attila Kiss and István Matók of Digital Pro Studio. This explored the use of the center channel when recording in 5-channel surround sound, considering the effect on any potential downmixing to 2-channel stereo. Yasuyo Yasuda of NTT DoCoMo presented a paper on 3-D audio for mobile communications, which highlighted the wide range of ways in which this technology can be applied. The results of a number of subjective tests that attempted to evaluate the performance of such systems with varying levels of complexity were also discussed.

A paper by Per Rubak and Lars Gottfried Johansen of Aalborg University that considered the perception of coloration in room impulse responses was presented by Per Rubak. This included a detailed literature review on the topic, which resulted in the proposal of a new method of measuring this effect.

The final paper of the day was presented by Jérôme Daniel of France Telecom. He discussed the problem of coding distance in spatial recording and reproduction formats such as high-order Ambisonics. He explained that Ambisonics assumes that the virtual sources and reproduction loudspeakers are in the far field, meaning that plane waves reach the listener. He showed that this is not the case in a practical situation, but that it can be compensated for by us-

DSP IN LOUDSPEAKERS

The final day of the conference consisted of two sessions on signal processing in loudspeaker systems. Ronald Aarts of Philips Research Laboratories presented an invited paper with an overview of a number of digital signal processing techniques that can improve the end-user experience. One such technique improves the perceived low-frequency performance of a loudspeaker by taking advantage of the psychoacoustic effect of virtual pitch, where a fundamental frequency can be perceived even when it is absent. He summarized by stating that the combination of DSP and ➥

Conference committee: from left, seated, Per Rubak, Ole Moesmann, and Martin Rune Andersen; standing, Subir Pramanik, Eddy Bøgh Brixen, Lars Gottfried Johansen, Knud Bank Christensen, Jan Abildgaard Pedersen, and Crilles Bak Rasmussen. Russell Mason, conference webmaster, missed the photo.

psychoacoustics can be a very powerful tool.

This was followed by a presentation given by John Mourjopolous and Nicholas-Alexander Tatlas of the University of Patras. They considered the benefits and potential problems of an all-digital audio signal path from the source to the loudspeaker, including the possible applications of wireless networks and the current limitations in digital loudspeaker designs. Peter Mapp of Peter Mapp Associates presented a paper that focused on the range of signal processing employed in sound reinforcement. He outlined the kinds of problems that can be alleviated by signal processing and those that require a physical solution.

Continuing the theme of using digital signal processing to compensate for poor acoustical performance, two papers discussed active compensation of nonlinear distortion in loudspeaker transducers. In an invited paper Wolfgang Klippel discussed the need for small, inexpensive, and lightweight loudspeakers with a high power output. Klippel explained that it can be difficult to achieve high sound quality with such loudspeakers through transducer design alone. He suggested active loudspeaker control as a method of alleviating some of the problems, and he reviewed the relative advantages and disadvantages of a number of techniques that could be used for compensation.

Throughout the conference Klippel also gave demonstrations of practical systems for loudspeaker measurement and active compensation. He showed examples of the causes of nonlinear distortions in loudspeakers and demonstrated the audible effects on test signals. A short set of measurements were made of the loudspeaker, from which the required compensation was derived. The application of this compensation to the input signal showed that the problems were greatly reduced.

Andrew Bright of Nokia also discussed compensation of nonlinear distortion of loudspeakers in his presentation. He explained his use of a simplified algorithm that was obtained by a discrete-time model of the loudspeaker nonlinearity. He also presented experimental results that demonstrated the improvements to be gained by using this model.

The final paper of the conference was by Steen Munk and Kennet Skov Andersen of Bang & Olufsen ICEpower. This described the use of class D amplifiers, how their performance may be improved by compensation in the modulator and in the analog stage of the amplifier, and methods to implement control of the amplifier gain. They also compared the performance of the class D amplifier with state-of-the-art analog amplifier designs. They found that while the digital amplifier is currently inferior in some respects, it is expected that with further development the performance can be improved to match or exceed the analog designs.

CONFERENCE CLOSE

After three days of informative papers that were conducive to discussion and inspiration, Chair Per Rubak closed the conference by thanking all those involved and hoping that many would return for future international conferences in Denmark. Delegates were impressed with the high technical content of the presentations and the relaxed and friendly atmosphere of the conference. All look forward to similar AES events in the future.