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C H R O N I C L E

The 12-th International Symposium on Sound and Vision Engineering and Mastering (ISSVEM'07)

The 12-th International Symposium on Sound and Vision Engineering and Mastering (ISSVEM'07) was organized by the Multimedia Systems Department of the Gdańsk University of Technology under the auspices of the Polish Section of the Audio Engineering Society and the Polish Academy of Sciences. The conference was held on June 15–16, 2007 in Gdańsk, Poland. The conference was hosted by the Multimedia Systems Department (URL address: http://www.multimed.org/issvem07).

The Symposium gathered interdisciplinary mixture of respectful engineering and musicologists scientists and specialists. Consequently, this conference was designed to combine future perspectives along with state-of-the-art research and practice in the interdisciplinary music and technical domains. The Symposium started with two joint sessions co-organized by the Scientific Committees of the KKRRiT'2007 conference (hosted by the Department of Radiocommunication Systems and Networks, GUT and ISSVEM'07. The first session was entitled: "Tools for multimedia data analysis and compression and plenary session". The Symposium was officially opened by Prof. Andrzej Czyzewski during the first plenary session. The plenary speakers were: Kristian Kroschel and Michael Grimm form the University of Karlsruhe, Germany who presented a paper entitled: "Audio-visual emotion detection" and the second paper was given by Wiesław Woszczyk, President of the Audio Engineering Society (McGill University, Canada). The second plenary session included a paper by Andrzej Rakowski, Sylwia Makomaska, and Piotr Rogowski from the Fryderyk Chopin Academy of Music in Warsaw, entitled: "Short-term memory for pitch in absolute-pitch possessors" and a paper presented by Wieslaw Woszczyk on "High-resolution multisensory communication using broadband networks – McGill UltraVideo".

The conference included also regular sessions given by experts in the specific subject areas on the topics such as: Musical Informatics, Sound and Vision Perception and Measurement, and Electroacoustic Devices and Systems. In addition there was a poster session scheduled with papers on sound and video restoration, animation, and a presentation of students' recordings. The Symposium program included also a visit and presentation of the Polish Baltic Philharmonic Hall.

All papers were reviewed by the Scientific Committee. Also, some papers have been awarded with the "best paper" prize. The results were announced during the Symposium Closing Ceremony. The following papers were chosen for this price:

- Piotr Bobiński, Bartosz Bielawski – *MIDI controlled audio-DSP system*
- Joanna Szczepańska–Antosik – *Roughness of two simultaneous harmonic complex tones in various pitch registers*
- Jacek Wołkowicz, Zbigniew Kulka – *N-gram based approach to composer recognition*
- Aleksandra Młyńska – *Output filter implementation for digital waveguide clarinet model*
- Piotr Pruchnicki, Przemysław Plaskota – *Automatic measuring system for head-related transfer function measurement*

Abstracts

1. MIDI controlled audio-DSP system

BOBIŃSKI Piotr, BIELAWSKI Bartosz

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The paper presents a MIDI controlled hardware station for digital audio signal processing. The system consists of an evaluation board ADSP-21364 EZ-KIT with the SHARC-21364 processor, and a MIDI to the SPI converter designed for the communication with a DSP processor. Short information about hardware and protocols is included. The design of the converter device and the implemented software modules that enable to control the applications running on the DSP processor are described. Finally, two applications of the developed systems are introduced: synthesis of music with steering from a MIDI keyboard, and realization of recording with steering from a MIDI console.

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2. The LPCC-based method for automatic identification of a music performer

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The paper describes a method of automatic identification of different music performers playing identical pieces of music on the same instrument. The performer model based on the LPCC features and vector quantization are proposed as methods of classification. The presented approach was verified with a database of experimental samples of Bach's 1st Cello Suite recorded especially for this study and the original audio CD recordings of Bach's 6 Cello Suites performed by six famous cellists.

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3. Conversion of stereo recording to 5.1 format using head-related transfer functions

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The paper presents the conversion of stereo recordings into multi-channel format 5.1 by means of the HRTF filtering. An algorithm of additional channels preparation (a central one and two surround ones)

using various filters created on the base of HRTF was tested. The filter bank was prepared on the base of impulse responses given by users. The algorithm uses the impulse response convolution of the filter with a processed signal. The convolution is realized in frequency domain, i.e. Fourier transforms of the processed signals are multiplied. Recordings created in such way were tested for the features of sound color distortion and spaciousness of the audio-scene in frontal plane.

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4. Implementation of dynamic range controller on digital signal processor

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The aim of this work was to build a real time digital range controller which could be used in the radio and television recording and broadcasting studios. The main purpose of the system is to change the dynamic range of an audio signal in a predefined way without introducing perceptible distortions, what guarantees an optimal use of the overall available dynamic range of an audio system. As a processor, the ADSP-21065L digital signal processor manufactured by Analog Devices was used. An input/output digital audio signal interface was designed and constructed using Atmega8L microcontroller manufactured by Atmel and FPGA device. The characteristics of the dynamic range controller were performed.

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5. FPGA-based implementation of digital sigma-delta modulators for high-resolution audio D/A converters

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Sigma-delta modulators (SDMs) are now widely used in high-resolution audio analog-to-digital converters (ADCs) and digital-to-analog converters (DACs). The field programmable gate array (FPGA)-based hardware is particularly suitable for the implementation of basic functional blocks of audio sigma-delta DACs. This paper presents the design, simulation, and the FPGA implementation of the lowpass digital sigma-delta modulators (SDMs) in various configurations. The results can be used for digital SDM designs and its future applications in the audio sigma-delta DACs as well as for educational purposes.

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6. Preparation of an anaglyph movie

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This paper presents problems related to the realization of a 3D movie designed as an anaglyph movie. Some notions concerning human perception of stereoscopy are also described. The investigation of the bit rate influence on the quality of a perceived 3D image is shortly outlined.

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7. Optimization of the total harmonic distortions of the acoustic vacuum tube push-pull amplifier

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The study presents the method of reducing total harmonic distortions of an acoustic vacuum tube amplifier with the 30W output power. The optimization of the amplifier's circuit is achieved through applying the solutions known from transistor technology such as differential amplifiers with a source of a current. The means of how a selection of vacuum tubes of the power and driver stages, as well as its operating points, influence the total harmonic distortion, was studied. By optimizing the amplifier, it was presented how a simple circuit solution assures very good final parameters with relatively low costs. The measurements were conducted using the systems of Audio Precision Company.

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8. Surround mixing in Pro Tools LE

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Today's multimedia use surround sound extensively, and most often in format 5.1. A lot of popular software packages used for sound production, including Pro Tools in LE version, do not have the ability to form mixes in this format. However, thanks to an appropriate configuration of internal audio buses and the cooperation with the multichannel input/output interface, this mode of operation is possible. Unlike other proposals, based only on send channels to individual output channels, the authors present a new configuration that uses the Pro Tools LE software. Thanks to this configuration, placing an apparent source of sound in the surround space becomes more intuitive, as it is accomplished through the use of panning sliders which determine space coordinates. Direct send channels serve exclusively to form the central and low frequency effect channel (LFE). This configuration was applied to make the final mixdown of a multichannel recording of the choir concert with the use of a microphone matrix in the OCT system.

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9. The electromagnetic microwave sensors in improving speech intelligibility after laryngectomy

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The paper contains the recapitulation of analytic, constructional and development investigations of low power electromagnetic microwave sensors (EMS) designed for a laryngectophone system. The general procedure and construction examples are presented. The measurement stand used to research signals produced by the sensors of articulators motion are described, and the analytic method used to examine the usefulness of these signals to identify the vibrant phonemes is reviewed.

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10. Output filter implementation for digital waveguide clarinet model

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The modification of a dynamic physical clarinet model based on a digital waveguide is presented in the paper. In particular, the influence of the output filter parameters on signals generated by the model was investigated. As the output filter, the adaptive filter was designed and implemented with different parameter values for different reference signals. The recorded clarinet sound samples and the signals generated by the clarinet model were compared.

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11. Automatic measuring system for head-related transfer function measurement

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In this paper the Head Related Transfer Function (HRTF) Automatic Measurement Equipment is presented. The system allows to measure HRTF with high spatial resolution within a considerably short time. The system employs many switched sound sources and a swivel chair. HRTF describes a transfer function of the human head and pinna and is unique for each human. Individual measurement of HRTF is necessary for applications where precise simulation of sound source localization is essential. An accurate determination of HRTF requires the position of sound source to be changed in the space surrounding test participants. It necessitates hundreds of measurements with high spatial resolution. The procedure becomes very time consuming and tiring for the participants.

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12. Sound quality of reverberation obtained by various auralization methods

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The purpose of the study was to assess sound quality of artificial reverberation obtained with the use of various software packages and a hardware system. The assessment was made using samples of music and speech processed with the room acoustics simulation software (CATT-Acoustic and EASE), a plug-in reverberation software (Audio Ease Altiverb), a hardware reverberation system (Lexicon 960), and recordings of the same samples made in a listening room and in a concert hall. The sound quality was judged by a group of sound engineering students. The listeners' task was to rate the samples for naturalness. The achieved results show that samples processed with hardware reverb are perceived more natural than those obtained with the use of advanced auralization methods.

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13. Roughness of two simultaneous harmonic complex tones in various pitch registers

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The purpose of the study was to determine the dependence of perceived roughness on the frequency ratio of two simultaneous harmonic complex tones. A set of 36 dyads forming musical intervals of various tuning systems was presented in three pitch registers. Twelve sound engineering students judged each dyad for roughness by the method of absolute magnitude estimation. Results show that roughness considerably varies with the frequency ratio of the two complex tones, what is a well-known phenomenon. A new finding, being in contrast to published theories of roughness, is that some of the equally-tempered intervals are perceived less rough than their counterparts based on integer frequency ratios. This effect is attributed to slow beats that arise between the harmonics of two complex tones when the frequency ratio of the tones slightly departs from the integer ratio.

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14. Hybrid vacuum tube – solid state guitar amplifier

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In this study, a design of a guitar amplifier with an innovative "micro vacuum tube power amp" in the preamplifier section is presented. Thanks to such construction, a very good tonal quality was obtained at fairly low final cost. The process of designing is presented. Special attention was put to shaping proper gain characteristics of the preamp section in frequency domain. During this process computer simulations in PSpice of the electronic circuit were performed. The results of subjective tests performed on the prototype are also included. The final amplifier schema is based on them. The results of objective tests obtained with the Audio Precision System Two, are presented as well.

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15. Multimedia presentation of musical instruments

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The problem presented in the paper belongs to a wide subject concerning the use of multimedia technology for the improvement of educational activities in museums and for attracting the audience with objects of art. Several examples of temporal and regular multimedia performances in galleries are known in Poland and abroad, but still the technology is not fully used in traditional museums. In the paper, the case of multimedia presentation devoted to the collection of the historical instruments from the Museum of Musical Instruments in Poznań is described. By their nature, musical instruments are interactive, but for the reasons of security and costs, only on rare occasions may visitors listen to their sound. Multimedia can change this situation. The inspiration for creating the multimedia presentation was the 250th Anniversary of Wolfgang Amadeus Mozart birth. The historical XVIII century instruments from the collection of the museum accompanied by the music of the composer have been virtually exposed. Shown during the "Museums Night" in May 2006, the presentation has gained interest of hundreds of guests and proved useful for future educational actions.

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16. N-gram based approach to composer recognition

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The paper describes how tools provided by Natural Language Processing and Information Retrieval can be applied to music. A method of converting complex musical structure to features (n-grams) corresponding with words of text was introduced. Mutual correspondence between both representations was

shown by demonstrating certain important regularities known from text processing, which may also be found in music. Theoretical aspects of the case were applied to the problem of automatic composer attribution where statistical analysis of n-gram profiles, known from statistical NLP, was used. A MIDI files corpus of piano pieces was chosen as the source of data.

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17. Computational complexity of the algorithm creating hypermetric rhythmic hypotheses

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This paper presents the algorithm creating rhythmic hypotheses worked out by the authors and presented in their previous works, and the analysis of its computational complexity. The algorithm for creating and ranking hypotheses is presented first. Then, the analyses of the algorithm computational complexity proceeds. Three phases of the algorithm: creating periods, creating simplified hypotheses and creating full hypotheses are examined. The analyses of assume that the engineered method is expected to rank rhythmic hypotheses formed of three rhythmic levels above meter.

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18. Automatic singing quality recognition employing Artificial Neural Networks

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The aim of the paper is to determine how the quality of a singing voice can be recognized automatically. For this purpose, a database of singing voice sounds with samples of voices of trained and untrained singers, was created and is presented. The methods of a singing voice parameterization are shortly reviewed and a set of descriptors is outlined. Each of the presented samples is parameterized and judged by experts, and the resulting feature vectors and quality scores are then used to train an artificial neural network. A comparison between experts' judgments and automatic recognition results is performed. Finally, statistical methods are applied to prove that an artificial neural network is able to automatically determine the quality of a singing voice with the accuracy very similar to expert assessments. The paper includes the discussion of results and derived conclusions.

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