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Chronicle

14th International Symposium on Sound Engineering and Tonmeistering ISSET 2011 Wrocław, Poland, May 19 – 21, 2011

The 14th International Symposium on Sound Engineering and Tonmeistering will be held on May 19-21, 2011, in Wrocław. The Symposium is organized by the Chair of Acoustics, Institute of Telecommunications, Teleinformatics and Acoustics, Wrocław University of Technology, under auspicious of the Polish Section of the Audio Engineering Society. The organizers cordially invite sound engineers, music producers, acousticians, and specialists in sound reinforcement, scientists who deal with sound engineering, sound recording and related areas, students, and employees of the audio industry to participate in the Symposium. The Symposium programme will include lecture sessions and workshop presentations.

Abstracts

1. A pitch, timbre and time-scale modification techniques for vocal processing

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The paper presents techniques that can be used for implementing a set of audio effects for vocal processing. High quality pitch, timbre and time scale modifications are performed through hybrid parametric representation of the signal based on deterministic/stochastic decomposition. The signal is considered as a combination of sinusoidal and narrow-band noise components that are represented by different parametric models. Instantaneous sinusoidal parameters are extracted from the signal using analysis filters with modulated impulse responses. Pitch contour estimated from the signal is used for initial estimation of harmonic trajectories and modulated bands of the analysis filters. Timbre of the voice is modified by a linear transformation that is trained using parallel sequences of both source and target singers. Some experimental results are provided in the paper.

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2. iSRC – studio-grade quality synchronous sample rate converter and requantizer

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The paper describes the design of a synchronous sample rate converter and requantizer for audio files, employing a unique approach for minimizing conversion errors. The converter/requantizer has been implemented as a PC application and is capable of achieving studio-grade processing quality. Instead of time-invariant models describing such phenomena as the polyfilter mismatch in the process of interpolation or the emergence of numerical and quantization errors which are usually modeled using additive stationary noise sources, more realistic nonlinear and time-dependent models have been used. The proposed models assume the error is directly correlated with the signal transients and it temporarily changes the properties of the process, thus making it time-variant. So, in our approach such changes in converter's transfer function have been reduced as much as possible in the audio frequency band. The subjective tests confirm high quality and fidelity of the conversion compared to solutions available on the market.

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3. Sound quality evaluation of DAB+ musical programs

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Paper presents the results of sound quality evaluation of various musical material transmitted via DAB+ system. The subjective assessment was compared to the psychoacoustical model-based objective evaluation for various bit-rates (from 24 kbit/sec to 136 kbit/sec) and the significant dependence of listener preferences as well as the model-based results on investigated bit-rates has been obtained. It was also found that the type of music influences the subjective assessment but in different manner than the model-based evaluation. As the important conclusion it can be stated that the sampling frequency (32 kHz and 44.1 kHz) does not influence obtained results for only high bit-rate values, and Spectral-Band-Replication mode significantly improves the subjective evaluation.

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4. The digital audio mixer based on universal I/O interface for SHARC processor

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The paper presents a digital audio mixer for digital audio recording studio. The mixer is based on the universal input/output digital interface for an evaluation board ADSP-21364 EZ-KIT with the SHARC-21364 processor. The mixer can be controlled from any MIDI console thru the MIDI2SPI converter. First the detailed assumptions of such a system are show. Then all of the systems modules are describe and proper system configuration is given. Then the DSP software, both for digital signal processing routines and MIDI control procedures, is presented. In particular, a lot of attention is paid to the use of appropriate algorithms, to ensure high sound quality. Finally, the results of objective tests and sound quality assessment of out mixer are presented and compared with other, both software and hardware, commercial solutions.

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5. Application of adsorbtion effect in sound absorbers

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Application an activated carbon in sound absorbers was discussed in the paper. The aim of using activated carbon is lowering resonance frequency of panel and perforated sound absorbers as compare to standard one with the same depth. It can be achieved because of the adsorbtion effect occurred for activated carbon. Sound absorption coefficients of the perforated absorber without damping, with mineral wool and with adsorber were measured in Kundt tube. The relation between the rate of activated carbon volume to absorber volume was investigated and discussed. In the papers it was shown that the application of activated carbon lowers resonance frequency of perforated sound absorber up to 1.8 times as compare to the absorber without damping. The activated carbon is also influences on the sound absorption coefficients of the absorbers. The advantages of application activated carbon in sound absorbers are discussed.

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6. Column microphone

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This paper presents concept of column microphone for use in speech reinforcement in lecture, conference and theater halls. The main goal is to build an acoustic model of

microphone system with large horizontal angle and very small vertical one. As an input parameters are defined number of microphones in column, space between microphones and distance from sound source. There are presented results of computer simulations made in MATLAB – frequency response and directional characteristics, designing rules of the column, their main properties and helpful equations.

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7. Loudness of audio broadcast signals – objective parameters of subjective loudness perception according to ITU and EBU recommendations

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For many years one have searched for adequate, appropriete and commonly used mesurement method describing sensation of loudness. Particularly the problem of radio and television advertisements' high loudness level has been noticed in Poland, as well as in other european countries. It has been found quite irritating by common listeners exposed to unpleasant and annoying sound level changes.

For some time, there have been efforts to develop an algorithms to measure subjectively perceived hearing impression of loudness. Apart from standards published in some countries, there is noticable lack of widely accepted one. Additionally, nowadays commonly used VU and PPM meters do not properly reflect the essence of the phenomenon. It seems that the appropriate solution is to create a measurement system based on the methodology proposed by ITU and EBU organizations.

Algorithms described in this paper reflects computational methods of obtaining listener's subjective impression of loudness. The latest published standards and recommendations regarding loudness metering and normalization were taken into consideration, which is seen as the way to avoid excessive loudness jumps between content and to counter the practice of extreme audio signal processing. Various measurement time periods were taken into account, particularly, short for describing the dynamics of rapidly changing signals and long-term to set the desirable target level of broadcasted programme. Also there is a brief summary of result-improving methods which has been used: weighting filters (frequency domain), as well as gating procedures with the lenght of measurement period (time domain). Attention has been paid to the choice of measured parameters, descriptors and their corresponding interpretation (especially in the context of the laws in force in Poland).

Methodology described in this paper shows inevitable need of changing the approach to monitoring of tv/radio production focused on reaching the desired target loudness level of broadcast programme. New type of loudness meters provides ways of proper sound control in the studio and in signal distribution system. This paper contains results of measurements performed on material derived from polish broadcasters. All the mesurements and analysis was based on the standard developed by PLOUD group, a part of European Broadcasting Union.

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8. Real-time multichannel convolution processor based on FPGA platform

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In the paper will be described a real-time multichannel convolution processor. The FPGA platform was chosen in order to reduce the latency time. The most important advantage of the FPGA is the fact that the programmable structure makes possible the parallelization of processing of each channel, so we can achieve lower latency time. The convolution is used to achieve realistic reverberation effects or spatial effects. The presented design will be suitable for applying reverberation effects of real concert halls to recordings made in the anechoic studios. In some fields of sound engineering the reference and optimal impulse responses (like The Carnegie Hall's impulse response) are required to create good listener's music sensations. A impulse response got from real concert hall normally is long and the hardware requirements which have to be used are larger than the typical DSP possibilities which should be used in order to get a real-time effects. The presented design has many possible applications in sound engineering. One of the most interesting fields of usage is in the real-time spatial processors which use head-related transfer function recorded in real listening spaces. Providing spatial effects in headphone systems is the way of down-mixing the surround recording and applying reverberation effects to the sound. Low latency time is extremely important in systems coupled with the vision. The lack of synchronization between the vision and the audio is very unpleasant for the spectator who would like to listen to the concert or human speech and in the same time watch it. Using the FPGA in that system is the possibility of using the greatest FPGA capability – concurrency. Using a FPGA concurrent design approach may be a solution in this case. The implementation is done on the Atlys board with Spartan-6 LX45 FPGA. The paper contains the theory of operation, the implementation approach and some subjective and objective performance results.

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9. Oxygen-free copper from DCC-AGH® technologies in highly-advanced audio-video cable applications

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The common uses of electronics and electro-technology have led to the development of new types of audio-video products with highly-advanced properties such as cables, conductors, microcables and connecting elements. An infallible transmission of electrical signal is the vulnerable point of nearly every type of appliance using electrical energy. The basic material which is used for conductors in these types of applications is copper with a specially formed structure produced from high-chemical-purity cathodes. This results

from the above-standard requirements of the product in terms of electrical conductivity which is necessary for technological processes of its manufacturing. At the same time, oxygen which forms copper oxide Cu_2O is limited due to the substantial decrease in tensility of copper for microwires. Thus, the standard for these applications has become OFC (Oxygen Free Copper), OFHC (Oxygen Free High Conductivity Copper) and its variations with minimal amounts of grain LGC (Long Grain Copper) and LCC (Long Crystal Copper), thanks to which the transmission of signal in audio-video constructions has become more infallible and the elimination of connection flaws (skin effect) has become possible. This is called functionally excellent copper (FPC).

Interest in such materials led to the development of laboratory and industrial solutions for the production of oxygen-free copper directly prepared for drawing into microwires dedicated for electrical purposes (audio-video cables, heat resistant conductors, and electrical looms). Among such solutions, the most technically advanced are the DCC-AGH® and the Ohno Continuous Casting® laboratory methods which consist of melting cathodes, reducing oxygen and casting copper with a directional crystallite structure additionally characterised by minimal amounts of grain.

Properties obtained in the aforementioned processes significantly depend on the casting parameters including flow rate, crystalliser cooling water temperature and casting speed. There is a possibility of adjusting these parameters; however, the impact of the change can be visible in the material's structure. It can also be concluded that a change, in casting speed for example, has a significant impact on the electrical and mechanical properties as well as on other technological properties of a material.

This article concentrates on the manufacturing and processing issues of functionally excellent copper. The parameters of cables obtained from this class of copper depend on their metallurgic quality, chemical purity and their structural state formed during plastic processing and heat treatment. A number of these properties are a natural consequence of the applied metallurgic processes, such as the quality of the copper concentrate, the anodes and the cathodes; others depend on crystallisation processes, while the rest result from the technological path of copper processing. On this basis, it is obvious that to understand the parameters of the newly-developed functionally excellent copper, an initial synthetic review of technologies for its manufacturing and processing is required. The results of tests on the formation of qualities of oxygen-free copper with a casting structure from DCC-AGH® technologies, and for comparison from UPCAST® technologies, allowing for its application in modern sound engineering and the power industry have also been presented in this paper.

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10. Research on authentication of compressed audio recordings

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Since digital audio recordings appeared, audio authentication has become more difficult and in most cases impossible. Currently available technologies and free editing software allow forger to cut or paste any single word without audible artifacts. The presentation will address the problem of tampering detection and will discuss one of the methods used for authenticity analysis of digital audio recordings. Presented approach is based on checking frame offsets in audio files compressed by using perceptual audio coding. The influence of compression algorithm parameters and window shape will be shown using the most popular encoders as an example. The usage of this tool in detecting forgeries will be shown and the effectiveness in analyzing human speech will be discussed.

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11. National Forum of Music in Wrocław, Poland

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The idea behind the National Forum of Music was to create a meeting place for most of the artistic initiatives, ideas and projects in Wrocław. The NFM building is bound to become an inherent part of Wrocław and its cultural landscape. We would like the new institution to become a luxury brand recognized all over the world – a unique place, the importance of which will stretch far beyond the city borders. We would like to invite you to participate in the process of materializing the concept of the National Forum of Music and to help us create its history.

In order to ensure development of high art and its full access to the audience, it is necessary to create an adequate space for its exhibition. An inspiring place is more than just a background – it can complement the events taking place in it, engage in dialogue and stimulate the creation of new quality stemming from the interpenetration of various influences. The National Forum of Music in Wrocław will offer ideal conditions for the presentation of remarkable renditions of musical pieces and for the exhibition of the most distinguished works from other branches of art.

The construction works on the National Forum of Music in Wrocław – the largest and the most modern cultural investment in Poland – have been launched. The name of the building and, at the same time, of the institution which will be soon formed through the merge of the Wroclaw Philharmonic and the International Wratislavia Cantans Festival, refers to the plans which date one hundred years back. The plans involved the erection of the Forum of the Arts at the place of state and military celebrations. This area was previously called the Royal Forum (the palace of the King of Prussia was located there). Today we would like the name – the Forum of Music – to be associated with a place of exhibition of works from different periods – a place which provides the arena for a genuine dialogue and an artistic dispute on contemporary art.

The opening of one of the most prestigious concert halls in this part of Europe – the National Forum of Music – will take place in 2013. The building will house: a concert hall with around 1800 seats, three chamber halls, rehearsal rooms, office and conference space, a recording studio, exhibition space and an underground parking lot.

The National Music Forum is based on a state-of-the-art architectural design. The contract for the design of the building was awarded to Kuryłowicz & Associates Architecture Studio. The world-class acoustic conditions will be ensured by Artec Consultants Inc – a New York company specializing in the design of concert halls.

The NFM will be the home venue of numerous festivals and ensembles of the Wrocław Philharmonic.

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12. Research on correlation of sound and vision in 3D domain

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Research on impact of vision on sound perception has been initiated over a century ago. Experiments proving this phenomenon involved only subjective evaluation and could be difficult to repeat in other conditions. The eye-gaze tracking system engineered at the Multimedia Systems Department (MSD) enables to objectivize experiment results devoted to audio-visual correlation investigations. The eye-gaze tracking system determines the user's fixation point on the computer screen. The current line of sight or point of fixation allow for better understanding the influence of the observed image on the perceived sound. The user's visual activity is presented as a dynamic heat map overlayed on the original video stream. The research conducted in the gaze tracking domain allowed for extending application of eye-gaze tracking to analyze visual activity in 3D movie samples. Analysis of gazing on objects in 3D scene is based on video content indexing. Audio-video samples are prepared in two configurations: 3D movie and stereo sound, 3D movie and surround sound. The results obtained confirm that the proposed approach for investigating 3D audio-visual correlation is very promising.

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13. Comparison of different speech time-scale modification methods

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The objective of this work is to investigate the influence of the different time-scale modification (TSM) methods on the quality of the speech stretched using the designed non-uniform real-time speech time-scale modification algorithm (Nu-RtTsm). Nu-RtTsm algorithm is a combination of the typical TSM algorithm with the vowels, consonants, stutter, transients and silence detectors. Based on the information about the content and the estimated value of the rate of speech (ROS), the algorithm adapts the scaling factor value, and removes the redundant signal i.e. silence, stutter and transients. TSM algorithms named: SOLA, PSOLA and WSOLA were examined in order to assess the quality of the stretched speech, the complexity of the calculation and the possibility of their usage in the Nu-RtTsm algorithm. Subjective tests were performed in order to compare the quality of the different time-scaling methods.

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14. Music mixing process controlled by hand gestures

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In the paper a system enabling to control sound mixing process by hand gestures is presented. The hardware part of the system composes of a multimedia projector with a webcam attached and a screen for projected image placed between front studio monitors of surround sound system. An interface that runs on a standard PC, recognizes gestures and controls music production system such as Protools or Cubase by sending MIDI messages, has been engineered. Dynamic hand gestures, i.e. the ones based on hand motion trajectories, and the static gestures, i.e. based on forming a particular hand shape, are recognized by the system. A user is situated in a sweet spot between the screen and the projector and by hand gestures changes sound characteristics. It is possible to change two parameters simultaneously since the user can assign one audio parameter to his/her right hand and the other one to the left hand. Such a feature is not available when handling modern sound mixing software by mouse and keyboard only. Moreover, due to the fact that there is no equipment spaced between the speakers and the user, audio reflections colouring sound (such as the ones occurring from a mixing desk) can be eliminated. For static hand gesture recognition SVM classifiers have been used. The recognition of dynamic gestures is based on motion trajectories modelled by fuzzy logic.

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15. Influence of the force factor variation in electrodynamic loudspeakers

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The non linearities in the motor of a classical electrodynamic loudspeaker are still an issue. This paper studies the influence of the shape of the force factor variation on the harmonic and intermodulation distorsion. The real variation is described at least by a linear and a quadratic term. The effect of each term is studied separately, as they don't influence the same kind of frequencies, harmonics or intermodulation. Both terms together result in enhanced effects. The dissymmetry of the Bl variation with regard to the coil centered position has also peculiar effects. Therefore, the power of each harmonic and intermodulation frequency is calculated and the terms are compared.

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16. Developing digital version of a musical instrument

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Modern digital instruments are far more advanced in design and capabilities than their electronic ancestors. They incorporate real time tone generation based on sampling or synthesis technologies, DSP and FPGA technologies allow the designer to implement complicated algorithms and audio effects in one device and they are smaller and cheaper. This paper presents a digital instrument based on sampling technology. Each of the instrument's key plays one sample (full multisampling) in a combination with layering (multi-layering). Recorded samples are formed in files of 61 samples, stored on the SD card, mixed together, processed and played back with a program running on FPGA in real time.

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17. Monitoring of audience employing vector sound intensity sensors

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A system automatic audio-visual monitoring of activity of audience is presented. The system uses cameras and acoustic sensors to analyze the activity of listeners in a lecture or conference hall. The aim is to detect such activities as raising a hand, asking questions or unwanted activities such as disturbing the lecturer or speaker. The processing is performed employing two modalities – sound and vision. The sound processing algorithms use the signals from a multichannel sound intensity probe. Vocal activity is detected, using sound recognition algorithms, and localized in the auditorium. In the video domain, two cameras are employed. The cameras form a master-slave setup. The master camera with large resolution is used to detect the faces of listeners. The PTZ camera can then zoom in on the chosen region in the audience. In the video stream, optical flow is used to detect movement. Thus the activity of listeners can be detected using two modalities. The results of multimodal sound and image processing are synchronized to reinforce the decision. The proposed system can be used in lecture halls to monitor the activity of students or as a part of conference setup. In the latter case, the system can detect the person who takes part in the discussion and point the camera to them. The image of the speaking person can then be automatically displayed on the main screen.

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18. Measurements of active studio monitors with room correction

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An important direction of development in active studio monitors is making them "intelligent" by including DSP correction of room characteristics. There is a question about the subjective perceptual effect of these corrections. Two stereo systems of this new generation of monitors were put under test: Genelec 8250 and Equator Audio Q 10. They both use a proprietary software for semi-automatic room correction and both have a networking capability. The test consisted in the estimation of the system's impulse response in the same position in a chosen control room. The impulse response was measured with sine-sweep method, utilised in both software systems. Responses with and without room correction were compared. The objective measurements were complemented by a subjective listening test, in which several sound engineers evaluated the Equator system in two modes of operation: with and without the room correction.

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19. Comparison of auditory and performance pitch intonation skills in a group of violinists

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In a musical practice a concept of pitch intonation refers to a skill to precisely reproduce a pitch of a given sound, or more generally, to precisely produce a desired pitch. However, pitch reproduction is at least a two-stage process, the first of which is an accurate evaluation of a pitch heard, and the second – its precise performance. The research presented focuses on a problem of impact of each of those stages on a pitch intonation phenomenon. A group of a violin class students from the Academy of Music in Krakow was tested. Such a group was selected due to general opinion about violinists among the musicians, that they have particularly good hearing abilities, and because pitch intonation is an important aspect of a violin performance technique. As the first stage of the experiment the frequency discrimination was examined, since this parameter is often used to estimate hearing abilities in a frequency domain. Tests of pitch tuning on a dedicated software were also carried out, in order to isolate pure intonation skills from the instrument performance problems. During the second stage of the experiment a process of tuning a sound played on an instrument to a reference sound reproduced through a loudspeaker was recorded. Different playing techniques were examined. The signal recorded was then analysed in a frequency domain basing on autocorrelation, time, and spectral algorithms.

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20. Sound design in open-air opera spectacle in multichannel technology

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Proposed presentation will disscuss complexity of sound design in open-air opera spectacle based on Roger Waters "Ca Ira" done in Poznań. During preproduction stage, according to composer directions, I decide to choose surround, 8-channel system for 12000 audience at International Poznań Fair square. My goal was to cover audience with sound effects coming from different localisation not exactly from loudspeaker direction. Huge dimension of 3 stages (20 m wide/25 deep) and audience square 80 m by 160 m, made major problems with sound localisation and forced me to build unusual sound system to manage with it.

Opera score has 14 soloist, 6 kids solo, symphonic orchestra, 80 pcs mixed choir, 22 pcs kids choir, 20 pcs gospel choir plus about 100 sound effects cue marked in score. We have almost +200 sound source channels to mix. Additionaly we also need to syncing video playback with live orchestra cue. Basic estetic decision was how to fill audience with music, surround ambience and keep focus on main performing stage.

Choosen 8-channel Timax system was used as basic localisation matrix and whole sound efx playback source. Each cue had different localisation and finaly audience fill sound space as multidirectional stereo somewhere overhead, avoid direct localisation in sound source as it happen in regular system. All spectacle was realised against score which was marked with sound and video cue. Quantity of cues and short time for rehersal was additional adrenaline for all sound engineers involved directly in process.

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21. Development of timbre discrimination skills in students of sound engineering

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The paper presents results of a test used to assess the progress of sound engineering students in timbre discrimination tasks, at various stages of the Timbre Solfege course. Timbre Solfege is a course in perceptual analysis and evaluation of sound taught to the students of sound engineering at the Fryderyk Chopin University of Music in Warsaw. Results of the test, collected from successive groups of students over a period of 10 years show that sensitivity to changes in timbre may be considerably improved by systematic training. The test proved to be a useful tool for evaluation of the students' progress throughout the course.

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22. Variance in level preference of balance engineers

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Limited research has been conducted that quantifies how much expert listeners vary over time. A task-based testing method is employed to discern the range of variance an expert listener display over both short and long periods of time. Mixing engineers are presented with a basic mixing task, performed over loudspeakers and headphones, comprised of one stereo backing track and a solo instrument or voice. By tracking the range in level in which the mixing engineers place a soloist into an accompanying track over a number of trials, trends are observed. Distributions are calculated for three genres of music and variance is calculated over time. The results show that in fact the variance is relatively low, and even lower for the more experienced subjects. These results provide a baseline for future testing.

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23. Automatic tagging of musical files

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The purpose of this paper is to examine the possibility of automatic tagging of musical files employing the eye-gaze tracking system. The study was conducted with the participation of twenty students having different musical experience. One of the tasks of test participants was to fill in the website survey that allowed for gathering information about the music experience of the subjects. The next task was to assign characteristics of music such as tempo, dynamics, genre, etc. The analysis of the results obtained aimed at determining whether it is possible to automatically tag music characteristics (e.g. tempo) of a music excerpt employing eye-gaze tracking system. In addition, the correlation between music experience of the participants and time needed to fulfill test tasks was examined. This paper presents a detailed description of our experiment with the presentation of selected results and conclusions derived from the analysis.

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24. Active (virtual) acoustics in service of music performance and recording

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Virtual or active acoustics refers to the generation of a simulated room response by means of electroacoustics and digital signal processing. An artificial room response may

include sound reflections and reverberation as well as other acoustic features mimicking the actual room. They will cause the listener to have an impression of being immersed in virtual acoustics of another simulated room that coexists with the actual physical room. Using low-latency broadband multichnnnel convolution and carefully measured room data, optimized transducers for rendering of sound fields, and an intuitive touch control user interface, it is possible to achieve a very high perceived quality of active acoustics. The electroacoustically coupled room resulting from such optimization does not merely produce an equivalent of a back-door reverberation chamber, but rather a fully functional complete room superimposed on the physical room, yet with highly selectable and adjustable acoustic response. The utility of such active system for music recording and performance is presented and supported with examples.

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25. System for automatic generation of music with determined emotional expression

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Music has always played a vital role in human life. People, when creating music, are able to express a variety of determined emotions, and vice versa, when listening to the music, they experience emotions expressed or invoked by a composition. Music is a symbol of human creativity in a pure form and format. But since the times of ancient Greece up to nowadays the pursuit to find out the way for modeling the process of music composition in an algorithmic way has been observed. The paper presents a simple system for automatic generation of music with determined emotional expression. First the impact of musical elements like tempo, loudness, key etc, on the emotional expression of music has been presented. Then the system has been described. It creates music in eight emotional states (octets of emotion representation) in line with the categorization of emotions used in psychology. Music can be automatically generated in two modes. First one is a "free flow" – it produces new notes and chords in every step. Second organizes the individual notes and chords in motifs that are periodically repeated. The system has been written in Java using Java sound API and MIDI synthesis.

Music generated by the system was presented to the group of people (30 computer science students) to check how relevant are the emotions evoked by the generated music to the intended ones. The answers to the short survey mostly agreed with the assumed set of basic emotions. The improved method could be used to automatically compose music in film, games and other environments that demand determined emotional expression by music.

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