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Chronicle

16th International Symposium on Sound Engineering and Tonmeistering Warszawa, Poland, October 8 – 10, 2015

The 16th International Symposium on Sound Engineering and Tonmeistering (ISSET) organized by the Institute of Radioelectronics and Multimedia Technology (Warsaw University of Technology), Department of Sound Engineering (Fryderyk Chopin University of Music) and the Polish Radio, under auspicious of the Polish Section of the Audio Engineering Society was held in Warsaw on October 8 – 10 in 2015. The main topics of the Symposium covered mostly all domains of audio engineering, i.e. musical acoustics, noise control, signal processing, room acoustics, radio and television, multimedia, sound engineering and tonmeistering, perception and quality assessment, and many others. The extra attention has been paid for the problems of loudness of audio programs in radio and TV broadcasting. Over 60 people from different branches of audio technology participated in this Symposium and shared their knowledge and experiences during the paper sessions, technical tours, workshops and special presentations. The selection of abstracts of the papers presented at the ISSET'2015 are inserted below.

Abstracts

Acoustic database of musical signals encoded with chosen techniques

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The rising interest in high quality transmission of speech signal, music and video demands designing of new encoding techniques. The aim of the coding is the optimization of transmission, i.e. obtaining almost imperceptible quality worsening of transmitted signals for the lowest bitrate. In the paper the music signals data-base is presented where chosen coding techniques were applied (AAC, mp2, mp3, ac3, ogg and wma) with various bitrates (from 32 kb/s to 320 kb/s). The realized database can be used not only for the analysis of influence of the coding techniques and bitrates for the quality of music signal but also for the validation of quality of phonic signal transmission systems and amplifying or the verification of new objective quality evaluation techniques.

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Perceptual evaluation of classical guitars in relation to their physical properties

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Finding the relation between measurable physical properties of musical instruments and perceptual evaluation of their tone quality is one of the crucial topics in musical acoustics. Although the research in this field for classical guitars is substantial, the relation between objective and subjective properties is still not clearly specified. The aim of the research presented in this paper was to analyze physical features of different classes of classical guitars in relation to their tonal character and quality. The analysis was based on the results of structural vibrations and sound pressure response measurements. Twenty seven classical guitars of different price range have been examined – both factory-made and top hand-crafted instruments. Tonal quality and characteristics of the instruments were determined based on the results of subjective assessment listening tests performed on a group of experts. Based on the scrupulous statistical analysis the relation between the established objective and subjective classical guitar parameters has been investigated and determined. The strong correlation between the price of instrument and its subjective evaluation has been also confirmed.

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Implementation of the performance rules in sound synthesis of wind instruments using modified sampling method

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The paper presents the method and the effect of implementing the performance rules in sound synthesizer of

wind instruments in symphony orchestra, that is currently under development. The synthesizer is based on a modified sampling method that uses samples containing the whole musical motifs, as well as algorithms that seamlessly connect such samples into a coherent-sounding phrases. The performance rules define variations in regular playback of the material contained in musical notation, that are characteristic of the natural way of playing the instrument. These variations are due, among others, the specific nature of the instrument, the skills of the player, and above all proper shaping of the expression in musical structures, that characterizes live musicians performances. The article presents a selected group of the performance rules with quantitative description of the variations in appropriate sound parameters. Changing sound parameters in the sampling synthesis requires the use of various signal processing methods on the recorded samples. The paper presents methods applied in the modified sampling synthesizer. Finally, the effect of the modifications is discussed.

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The impact of different selective mixing algorithm depths on intelligibility of multiple talkers

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The issue of words intelligibility of multiple talkers has been approached by one of the authors in his previous studies. During the experiments it was proposed to reduce spectral overlapping of words by selecting only the strongest ones in each spectral region (selective mixing), supposedly to increase intelligibility. Some differences were found, but they did not prove statistically significant. In the current study, the authors used the same selective mixing algorithm as in the previous experiments, but varied the depth of selection. Not only one word, but also two and three words were allowed to occupy one spectral region. Regarding the different depths of "selections", there were different distributions of number of distinguished words, with a little and statistically significant difference in favor of the samples without the selective summing algorithm applied. 31 people took part in the experiments, and a dedicated software for listening tests was used, developed by one of the authors. In this paper, the outcome of the listening tests will be presented, along with the analysis of the collected data.

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Acoustics of educational facilities of the Department of Sound Engineering at the Fryderyk Chopin University of Music

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The main goal of this research was to gather data which may lead to determine if the acoustic properties of rooms used for recording, listening and editing have any impact on the teaching results. Research concerned reverberance, colorization of sound and influence of acoustics on clarity of speech and music in these rooms. Impulse responses were taken in concert halls, recording rooms, control and editing rooms, lecture halls and laboratories. Measures like T30, EDT, C50, C80, TS, STI, IACC, frequency responses, distribution of reflections within time and waterfall plots were derived from collected IRs. The results were compared with international recommendations regarding acoustics of recording studios and listening rooms.

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Automatic music genre recognition applied to virtual bass synthesis algorithm

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In the paper an intelligent algorithm for the synthesis of low frequencies in mobile devices (Smart VBS) is described. The Smart VBS algorithm recognizes musical genre and according to it sets the optimum low frequency synthesis parameters. Low frequency synthesis is carried out using the method of non-linear functions (NLD). Modification is performed according to the type, number and gain of the nonlinear function. The algorithm is prepared in the form of an application written in C++/QT. The application identifies six most popular musical genres: classical, electronic, jazz, pop, rap, rock. On the basis of subjective listening tests sets of predefined settings are prepared, which dynamically adjust the music file currently being processed. The end result of the program is a modified audio file processed in accordance to the recognized genre.

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Virtual conductor

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Virtual Conductor is a project that lets you feel like a conductor and control the virtual orchestra playing one of the six well known classical pieces. Application exploits possibilities of such software as Max MSP, Ableton Live and INScore, and it is all controlled by the gestures performed in front of motion sensing device Kinect. The interactive manual introduces user to all the things he needs to know and that is how the conducting begins. User can control selected musical parameters such as tempo, dynamics or intros of specific instruments. He is leaded by a score which is synchronized with music and contains extra tips. Project's algorithm allows to continue its development by implementing another features. This Project is a tool with educational benefits for both pupils and students who are starting to learn score reading. Virtual Conductor combines

musical skills and motor coordination of users being at the same time great music entertainment.

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Dynamic control of midi sound parameters using built-in smartphone sensors

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In popular music performance there is a common practice to change music parameters like timbre, tempo or pitch in real time. This is especially true for music in MIDI format. These changes may be predefined or manually entered during playback using appropriate input devices. This paper describes the concept of mobile system that allows for real time modifications of sound parameters synchronously with the hand gestures of an operator handling a smartphone with built-in sensors (accelerator, gyroscope). In the proposed solution the task of real time external control of sound parameters of music being played or performed may be transferred to the listener who has the opportunity to actively contribute in shaping a musical performance. This gives him/her the ability to cross the traditional barriers of passive listening and allows for an entirely new dimension of musical experience. The paper also presents an example of the control system implemented in the iOS environment using iPhone 4s and a MIDI hardware controller.

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Acoustics modernization of the student culture zone at Wrocław University of Technology

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The purpose of this article is to present the results of improving acoustical properties in a multipurpose hall at the Student Culture Zone at Wrocław University of Technology. The building was designed according to the newest trends in design and architecture hence concrete and glass are the most widely used materials and therefore it does not meet proper acoustic field requirements. The main focus of this project was to adjust the reverberation time parameters. To reach desired reverberation levels a multi-stage plan was developed. The first step was to measure acoustical parameters of the room. Then several simulations were conducted in order to design appropriate acoustical panels that would also fulfill all hygienic and fire safety requirements. The last step was to measure acoustical parameters after the treatment and comparing the results with those acquired before the treatment.

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Reproduction of phantom sources improves with separation of direct and reflected sounds

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In virtual acoustics or artificial reverberation, impulse responses can be split so that direct and reflected components of the sound field are reproduced via separate loudspeakers. The authors had investigated the perceptual effect of angular separation of those components in commonly used 5.0 and 7.0 multichannel systems, with one and three sound sources respectively [KLECZKOWSKI *et al.* (2015), J. Audio Eng. Soc., **63**, 428–443]. In that work, each of the front channels of the 7.0 system was fed with only one sound source. In this work a similar experiment is reported, but with phantom sound sources between the front loudspeakers. The perceptual advantage of separation was found to be more consistent than in the condition of discrete sound sources.

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Application of dynamic element matching techniques for linearity enhancement of multi-bit audio sigma-delta A/D and D/A converters

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This paper presents properties of multi-bit audio sigma-delta A/D and D/A converters with single-bit and low-bit quantizers. In particular, application of dynamic element matching techniques for linearity enhancement of internal D/A converters using as construction part of multi-bit sigma-delta converters with low-bit quantizers/truncators are briefly described.

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Overview of applications of audio temporal envelope in sound engineering

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The domain of audio signals classification and recognition is based to a large extent on their representation in the frequency domain, or in a combined time-frequency domain. However, in some applications a signal temporal envelope is analyzed. For sound synthesis an ADSR model (Attack, Decay, Sustain, Release) is used. MPEG 7 standard includes metadata reflecting a shape of a temporal

envelope of audio signal (LAT – log attack time and TC – time centroid). The shape the temporal envelope is used, among others, to analyze audio transients, to study the intelligibility of speech, determining the dynamic range and compression of audio signals. In the luthier's practice the shape of the envelope of the string instrument response to the impulse stimulus determines the responsiveness of the instrument. Neurophysiological experiments proved that a human brain has an independent specialized centers enabling the detection of the envelopes separately in the time domain and in the frequency domain, which confirms the soundness of the temporal envelope perception. The paper reviews various aspects of audio temporal envelope (e.g. its generation, interpretation, descriptors or human perception) also from the perspective of an instrumentalist, sound engineer and tonmeister. We present methods to determine and analyze audio temporal envelope and to consider possible future applications.

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Reverberation in choral music

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Choral music has rich timbre and sophisticated texture. Room response is an integral part of this kind of music. Reverberation influence both performers and listeners. The article presents research about perception of choral music in a variety of reverberant condition. The most significant and preferred by listeners factors of room impulse response were investigated. Ambisonic system was used in the listening test. Multidimensional scaling was used in statistical analysis.

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Designation of the noise map with the use of clouds computing

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The paper presents Noise Maps, the Grid-based service, which employs the noise prediction algorithm and source model developed at the Multimedia Systems Department, Gdansk University of Technology. The web application enables the user to create acoustic maps for road noise estimation without utilizing additional commercial software. A short review concerning noise modeling and sound propagation in urban spaces is presented. The results obtained with the application are compared against those collected employing commercial software with two road noise prediction models, i.e. NMPB-Routes-96, recommended by the European Union and the International RLS-90. In the paper guidelines concerning the proposed application usage for users who have small experience in the subject are presented. The developed application is also been tested for interface clarity, proper operation and computing performance.

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Mastering procedure in mid-side domain based on objective parameters derived from existing phonograms

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In the paper the automatized mastering procedure is presented, which can be applied to semiprofessional and archive recordings of classical music. To obtain in the same time potential improvement in tonal balance and stereophonic spread, the proposed algorithm performs in mid-side domain. In first step the existing and under-mastering phonograms of the same music piece are analyzed and a set of objective parameters is derived. The results of the parameter comparisons control the processing step – equalization in mid-side domain – of input recording. In the paper the details of the proposed procedure is presented and subjective mastering effects are assessed.

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Interactive sound design

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Interactive multimedia installation, which enables process of the sound creation performed by participants, based on the previously prepared sound structures. The whole process of sound creation is performed simultaneously with playback of the short animation (which was equally as sound samples prepared previously). How it works: Each participant can use a piano keypad with 61 keys, divided into 5 octaves. Each octave contains a different set of various matched audio samples, divided in the ratio 1: 1 (1 sample – 1 key), arranged by following categories: 1 oct. (the lowest octave) – a set of 12 sound samples with dark and obscure nature, which behaves as low ambients, representing the lowest frequency part of a sound composition; 2 oct. – a set of 12 spatial, light sound samples, which should fill whole composition; 3 oct. – a set of 12 various audio samples, which mainly work in the medium frequency band; 4 oct. – a set of 12 high, intense drones, which cause an impression of intensification of the whole composition; 5 oct. (the highest octave) – a set of 13 various sound structures, corresponding directly with picture example; main part of the composition, include direct interaction process ("image – sound"). Each octave (without 5) has possibility of different audio processing: 1 oct. – Changing frequency parameter of the frequency shifter 2 oct. – Changing frequencies

parameters of the comb filters 3 oct. – Changing various parameters of the granulator 4 oct. – Changing frequency parameter of the comb filters.

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Psychoacoustic context of multichannel theatrical sound mix

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Cinema sound is mixed in an environment that in theory is acoustically extremely similar to the one within which is later enjoyed by the audience. So why does the average 5.1 cinema theatre remain the bottleneck for the quality of the sound design? In this paper the constraints of 5.1 for-mat are discussed by comparison to the new 3D formats: Dolby Atmos and Auro 3D. Then the basic psychoacoustic principles are presented as the framework of a proper theatrical 5.1 mix. The effects of well-known perceptual principles and chosen cognition-based rules will be presented as the fundaments of proposed efficient mixing tricks that help to 'hack' the hearing system of cinemagoers in order to bring them clear and more powerful experience. The practical application of equal loudness contour, the ear's acoustic reflex, temporal masking, simultaneous masking as well as Cherry's effect, audio-visual correspondence, auditory scene analysis, head related loss of brightness and Hass effect are discussed and illustrated by the multiple film excerpts of existing films and the author's works in 5.1 format.

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Timbre solfege training
– development of sensitivity to the sound timbre

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The purpose of this study was to determine whether persons with no formal musical training possess the ability of timbre discrimination similar to sound engineers and musicians. Then discover if such abilities may be effectively developed by training. A group of five students with no musical training and a group of five music students participated in a special ear-training course called Timbre Solfege aimed at the development of auditory sensitivity to timbre. A group of sound engineering undergraduate students who have completed the Timbre Solfege course as a part of their academic curriculum was chosen as a control group for the assessment of the progress obtained by the two experimental groups. The results of training were verified by testing timbre discrimination ability administered at the beginning and at the end of the course. The results of the test show the ability of timbre discrimination had improved in the two experimental groups.

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Auditory-visual interactions:
the influence of color on loudness judgments

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The paper reports the results of an experiment carried out to explore the influence of the color of pictures displayed on a large TV screen on the perceived loudness of sounds simultaneously played back through loudspeakers. A group of sound engineers and a group of ordinary subjects estimated the loudness of the sounds of three short audio-video clips containing the following material: (1) a color patch combined with pink noise, (2) an animated picture of an explosion combined with a synthesized sound effect of an explosion, (3) a realistic film of a kettle combined with the sound of a kettle whistle. The clips were presented to the subjects in combinations of five different colors and seven audio signal levels. The results of the experiment were similar for both groups of subjects and indicated that the color of pictures had practically no effect on the loudness of simultaneously heard sounds. The present results do not agree with the previous findings of the authors who reported that the picture's color has a slight, yet noticeable influence on the loudness of accompanying sounds.

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Film sound in 'calling'. Cinematic space shaping through the workshop of sound engineer

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'Calling' is a full length feature film directed by Marcin Dudziak. It is based on a novel 'Kingfisher' written by Kazimierz Orłoś. Calling lasts eighty minutes and except for a few scenes, it is extracted of the dialogues. It is told through the long shots that present silent characters and a slow action. Making sound for the picture with low-level narrative is an unconventional task, which needed using extraordinary solutions in film sound designing. The audile layer in 'Calling' makes the onsreen space richer and brings lots of added values to the picture. It modifies, in an unconventional way, the space of the world presented in a frame. The sound, thanks to the use of atmos and foley, creates the specific sense of a place of a particular scene. The exceptional use of foley gives the ambience the feeling of nearness and make the nature presented on screen alive. Audile layer and musical structures influences the diegesis, making it subjective and gains the oneiric soundscape features. 'Calling' gets the additional narrative and interpretive layer because of the essential cinematic space shaping through the sound. The audile layer modifies it or even gives the meaning to what can be seen in frame.

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Evaluation of noise induced permanent threshold shift (NIPTS) among music students

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Sound exposure among music students of Fryderyk Chopin University of Music in Warsaw was measured to assess the noise induced permanent threshold shift – NIPTS, according to ISO 1999 standard. For this assessment, measures were taken during 7 rehearsals, 3 concerts and during individual practices. The equivalent continuous A-weighted sound pressure level (L_{Aeq}) was measured, a questionnaire inquiry was conducted and the daily noise exposure level ($L_{EX,8\mathrm{h}}$) was calculated. The music students were usually exposed to sound at an equivalent continuous A-weighted sound pressure level of 72.1–92.0 dB, for 22–47 hours weekly. Assuming 45 years of similar exposure in terms of level and duration, the calculated noise induced permanent threshold shift – NIPTS was predicted to assume median values between 2.2 and 6.7 dB. The highest threshold shift (NIPTS) was predicted for students playing the flute (6.6 dB) and the French horn (6.7 dB).

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Graphical representation of music set based on mood of music

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One of the features for music recommendation, which is useful and intuitive for music listeners, is "mood". The paper presents an approach to graphical representation of mood of music pieces. Subjective evaluation based on listening tests is performed for assigning mood labels of 150 pieces of music and placing them on the 2D mood plane. As a result, a map of songs is created, where music excerpts with similar mood are organized next to each other on the two-dimensional graphical representation. In addition, automatic mapping is performed based on Self-Organized Maps. Comparison and evaluation of the methods and results are then executed. All of the tests and analyses are based on the mood model proposed by authors derived from previous studies and experiments.

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Perceptual calibration of narrow-band noise signals for audio applications

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Narrow-band noises are frequently used as test signals in audio system evaluation, psychoacoustics, noise control, and technical listening skills development programs. In perceptual audio assessment and technical listening skills development studies it is crucial that the hearing level thresholds and equal loudness contours for such signals are known in order to ensure proper calibration of the test signals. However, the data regarding human sensitivity to narrow-band noises that are available in the psychoacoustic literature are very limited. Except for threshold values for one-third octave noises presented in a diffuse sound field no other data have been normalized, and even for the threshold data there is still some disagreement in the literature. This paper presents a critical analysis of the existing literature on the human sensitivity to narrow-band signals with a focus on hearing thresholds for one-third octave noises under both loudspeaker and earphone listening conditions. On the basis of this analysis recommendations are made for the perceptual calibration of narrow-band noise signals and their use in auditory training and audio assessment studies.

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Are we able to discriminate compression?

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The aim of the conducted study was to evaluate differences in the audibility of different instruments by three commonly used lossy codecs. Seven instrument tones were compressed using MP3-LAME, Vorbis and Opus to determine how the detection of compressed sounds varies with bit rate, instrument and compression format. Audibility of lossy compression was examined on six naïve subjects during 60 hours of listening. At the bit rate of 32 kbps the compressed signals were easily discriminable with significant differences between subjects. With magnifying the bit rate audibility decreased, the signal becoming inaudible at 64–96 kbps. Discrimination varied significantly from instrument to instrument. Additionally the "learning effect" was examined.

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Analysis of guitar string vibration using fast cameras image processing

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The paper presents guitar string vibration visualization results. The movement was captured by a dedicated experimental setup of fast cameras, able to register deformations along and across the string. The recorded data was processed to precisely measure the vibration along the string,

to enhance spatial resolution and derive acoustic signal from the video. The signal obtained by image analysis was compared with a reference sound recorded with a professional measurement microphone. Presented research aims at studying phenomena of energy transport from the string to the instrument body.

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Research of the differences in MAA for phantom sources in musical recordings with stereo image made with the ITD and IID methods

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Paper presents the results of research on the Minimum Audible Angle (MAA) in musical recordings. The real musical signals (bass guitar and guitar solo) and their panoramic position were the subjects to subjective measurement with group of 12 listeners. For positioning of phantom source the standard methods (ITD and IID) have been used. It turned out that two methods of positioning in stereophonic image featured different effectiveness for loudspeaker listening as well as the headphone exposure. It has been also found that the way of recording exposure (via loudspeakers or headphones) does not influence the results of MAA for particular (IID or ITD) method of stereo panning.

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Preferred loudness of radio programs

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The study was conducted to explore loudness preferences in listening over loudspeakers to various types of radio programs. Three groups of listeners – sound engineers, musicians, and ordinary listeners set seven samples of various types of radio programs at the preferred loudness level, with the use of a volume controller. The samples were played back through a pair of tower loudspeakers and through a boombox radio. Results showed that the three groups of listeners considerably differed as to the preferred loudness settings. The group medians of loudness levels set for different program samples were within a range of 79–88 phons (sound engineers), 72–92 phons (musicians), and 59–75 phons (ordinary listeners). For a given group of listeners the preferred loudness level varied depending on the type of the radio program and, to a lesser degree, on the loudspeakers used for playing back the program samples. This finding indicates that the loudness of different types of radio programs should not be equalized to the same level.

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Free reeds in suction reed organ

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The reed organ is a musical instrument classified by polish authors as an idiophone, which contains a sound source a set of free reeds made of metal vibrating in the presence of an airstream. After it had gained popularity very quickly in the second half of the 19th century as a home and liturgical instrument, it became almost forgotten at the end of the 20th century, as the result of the spread of electronic instruments. Since all producers have been finally forced to change the profile of their business activity, foregoing continuity in the reed organ development has been broken. In recent years the increasing interest in antiques encompassed also preserved items of this instrument. The article contains a general analysis of the sound generation system of the suction reed organ, which is interesting from the standpoint of physical acoustics. The possibility of obtaining varied timbres of certain stops by choice of the scale and the construction of the reed cell and also by the air flow limitation, interference effects and the modulation of emitted sound wave has been investigated. Conclusions may be useful during renovations, to gain sound as close to the original as possible.

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