OBJECTIVE AND SUBJECTIVE ASSESSMENT OF COMMERCIAL SOUND ENHANCEMENT SYSTEMS

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Abstract
Sound enhancement systems introduced in 50's of the XX century, are becoming more and more popular even in big and reputable concert halls. Musicians and concertgoers have accepted this kind of solution in places with acoustic faults, where the renovation is impossible, or would be very expensive. Acousticians recommend nowadays solutions, as a natural sounding way of correction reverberation time and improvement of the stage acoustics. There are several commercial sound enhancement systems on the market offering similar acoustical parameters. For this reason the proper choice depends on the place of usage and calibration competences of the installation team.
In the paper, the evaluation of two commercial sound enhancement systems installed in the same room is presented. Energetic acoustic parameters (like reverberation times EDT and T20) were supplemented by the directional analysis of the space impulse responses. What is more, subjective listening tests were conducted, where discrimination of the systems were analyzed.

1 Introduction

1.1 Sound enhancement systems

In recent years sound enhancements systems becomes more and more popular due to greater stability and accessibility for reasonable money. An increasing number of manufacturers of such systems caused that they have been differed from each other, both in terms of architecture and principles of operation. Nowadays there are three types sound enhancements systems: regenerative, in-line and hybrid. Each of them provides the ability to increase a value of the reverberation time in a room, increase the sound pressure level and improve the support of early reflections on the stage. Some limitations of systems still could be supplemented only by acoustic structures, especially sound diffusers [1] and overhead reflective panels [2].

Because of the lack of comparative studies between similar systems it is difficult to determine which of them is most suitable for a given interior.

The selection of an appropriate sound enhancement system is a complex task because there is no direct connection between technical parameters of a system and auditory impression of a listener. Also the objective room acoustic parameters are not always sufficient for a fully assess of the interior [3], but they could be a first stage of it.

To carry out a subjective evaluation of a sound enhancement system it is recommended to perform measurement and psychoacoustic procedures based on modern auralization techniques. Such tests take into consideration the subtle differences between the systems and indicate their significant features [4]. Recently, also the ambisonic tests have become fairly popular [5-7]. In this type of study it is essential to select which aspects of sound quality should be included in the questions dedicated for test participant.

In Stanislaw Moniuszko Auditorium in Teatr Wielki – Polish National Opera there was a need to install one of the sound enhancement systems. It was associated mainly with hindered communication between orchestra placed in the orchestra pit and artists playing on the stage [8]. The choice of the most suitable one was linked to the problem of correct assessment of submitted tenders.

The comparative measurements of two commercial systems were conducted in the Orchestra Rehearsal Room in Teatr Wielki – Polish National Opera. Parameters of the system were set in a way to obtain an increased value of the reverberation time by the 20% in comparison to the natural reverberation time Tm = 1.2 s.

An equivalent of the sound source was a directional loudspeaker placed at heights 1 m and 1.6 m in relation to the typical altitudes of a violin and a head of a singer or a lecturer, respectively. A signal generation was preceded by a calibration procedure to ensure that the same acoustic power source was emitted in the room, as in an anechoic chamber. A microphone which allows to record signal from three directions along the main axes XYZ was placed at height of 1.2 m.

2   Methodology

2.1   Sound samples recording

Sound samples used as a sound sources were recorded in the anechoic chamber in University of Science and Technology in Krakow. The anechoic chamber is a cube of the internal volume of 465 m^3. The volume between wedges is 342 m^3, and the accessible floor dimensions are 6.7 x 7.1 m. It is formed with a steel grid and is placed 0.5 m above the wedges covering actual floor. The SPL of the background noise is -2 dB(A) and the cut-off frequency is 80 Hz [9].

Three different soundtracks were selected to be recorded, because of the most significant applications of systems. In order to let the listeners to analyze the speech intelligibility and the naturalness of the systems, an excerpt from Polish text was also used. Intelligibility of sung text could be rated on the basis of soprano excerpt from Hanna's Aria From The Haunted Manor opera composed by Stanisław Moniuszko. This masterpiece could be called Polish national epic, and its premiere took place in the very place, where analyzed systems were designed. As an example of instrumental music, the excerpt of the First Violin Concerto written by Karol Szymanowski was chosen. Both pieces (Aria and Violin Concerto) have rich and complex structure. In both excerpts, there were long legato notes as well as pizzicato by violin and wide, masterly passages in soprano.

Recordings were made with the use of three microphones: Schoeps Mk2, a measurement microphone GRAS 46AE and low noise GRAS 40HL. The microphones were placed on the height of the sound

source at a distance ca. 1.5 m. Subjective analysis of recorded signals revealed that for sound levels of selected sources, Schoeps gave the most natural results. Apart from natural signals, pink noise with the fixed sound pressure level was recorded. That allowed to calibrate playback systems and play sounds with the same loudness.

2.2   Impulse response measurements

In order to obtain acoustic parameters, especially related to spaciousness in the Orchestra Rehearsal Room of Teatr Wielki - Polish National Opera, and the influence of tested sound enhancement systems, spatial impulse responses (SIR) were measured. Measured SIRs were used for auralizations as well. The SoundField Portable ST350 system was employed - the first order ambisonic microphone allowed to obtain B-format signals both for analysis and auralizations. SIRs were measured using EASERA 1.2.8 software. As a test signal sweep sine of length 2.73 s and 48 kHz sampling frequency was emitted by Genelec 8030B loudspeaker. Scheme of the setup used for the SIRs measurements is presented in the Figure 1.
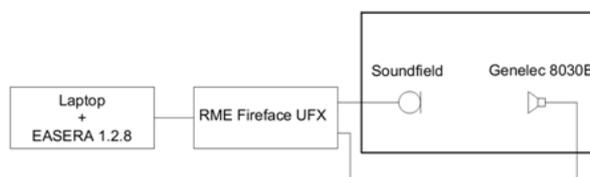

Figure 1 - Scheme of the setup for the impulse response measurements

SIRs in the Rehearsal Room were obtained for two tested sound systems with three different settings for each enhancement system. First, when the system was off, then with early reflection simulation only (ER) and finally with the reverberation time and early reflection simulation on (ALL). The recordings were made at 7 different geometrical setups of the source and the receiver each located in the audience (Figure 2).

2.3   Listening tests stand

Prepared sound samples (par. 2.1) were convolved with SIRs in order to obtain B-format recordings. The point was to achieve signals with identical musical content, but for three different sound enhancement system setups (shortcuts in bracket are used in further figures or tables):
- sound enhancement system turned off (off)
- first sound enhancement system (A)
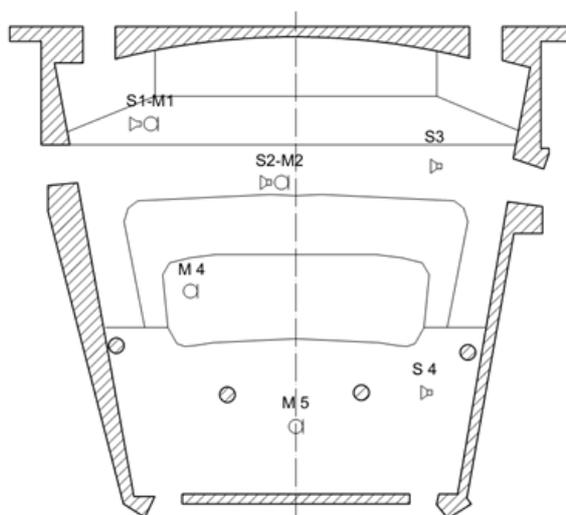- second sound enhancement system (B)

Figure 2 - Scheme of sound source and receivers locations in Rehearsal Room

The 16-channel setup was prepared using the RME ADA converters and Genelec 6010 monitors. The selected loudspeakers are quite small but their sensitivity is 93 dB SPL with flat frequency response from 74 Hz to 18 kHz (±2,5 dB). The loudspeakers are spherically placed around the listener, whose configuration is shown in Figure 3.
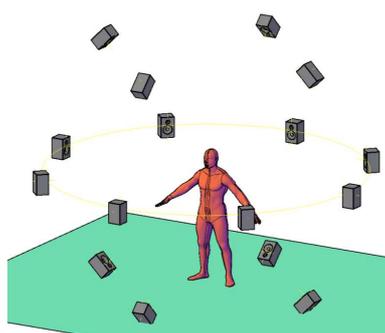

Figure 3 - Multichannel sound sources system in anechoic chamber

This setup was installed using microphone stands with sphere diameter of 3.2 m. The system was positioned with a laser angular meters and then calibrated and phase checked. Loudspeakers of the system are situated on a sphere around the listener in three planes – above, under and at the level of listener's head. The exact angular coordination with respect to the listener is presented in the Table 1. B-format signals were distributed using digital audio workstation Reaper [10] with Harpex parametric ambisonic decoding [11, 12]. Harpex VST plugin allows to decode B-format signal to multichannel setup, by configuring directional signals according to physical positions of the sound sources. Parametric decoding of B-format signals allows to obtain lower spatial resolution than directly from the ambisonic theory [13]. This procedures are meant to reconstruct acoustic field for previously described acoustic conditions and to avoid spatial aliasing.

3   Listening test

The recorded dry samples, convolved with SIRs, were prepared for listening tests to evaluate their subjective quality. In order to obtain some reference results, apart from signals auralized with the sound enhancement system on the signals without system were also used as stimuli.

Listening group consisted of 15 people with normal hearing with at least 10 years experience in music or acoustics engineering. Six women and 9 men at the age of 24 to 60 were tested.

Listening procedure was performed individually. In the anechoic chamber only one listener and the operator were present. Before listening evaluation, there was a training session to prepare each subject to listening procedure. The listening procedure lasted no longer than 45 minutes (including training).

Test procedure was pair comparison 3IFC (3 intervals, forced choice) with hidden reference. For 3 elements set, there are 6 permutations with repetitions. Every triad was presented 5 times, which gave 30 randomly compiled test series. As a sound sample, the speech signal (13 second long) was chosen. Single session consisted of 3 pieces, which gave 40 second, (the whole test took around 20 minutes).

4   Results

4.1   Objective evaluation

As it was stated above, sound systems were set to increase reverberation time by approximately 20% for operational mode ALL. In Figure 4 it could be observed, that EDT parameter increased quite evenly for all frequency, while T20 gain is highly irregular. For low frequency, the original reverberation time is quite small due to plate resonators in the room. Both systems, increased that value even more than 70%. On the other hand, for high frequency, both systems increased reverberation time by less than 20 %. There was also a problem with increasing early decay time EDT. Only for low frequency there was a 20 % gain, while for mid and high frequency range, the increase was much smaller. Even worse than for T20, EDT at 4000 Hz for System A was the same as without it.
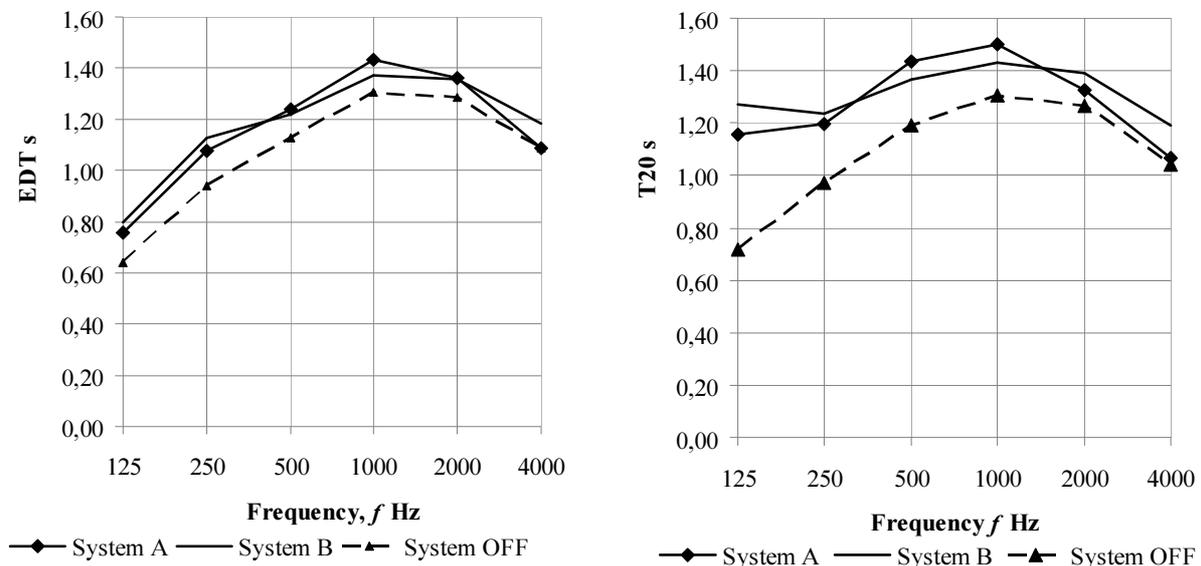

Figure 4 - Reverberation times EDT and T20 for two sound enhancement systems compared with the original value.

Interesting is also the analysis of differences between ER and ALL operational mode (Figure 5). For System B, the was a noticeable decrease of EDT value for low frequency (δEDT>0.1 where Just Noticeable Difference (JND) is 0.05), while for System A, there was an increase of EDT for all

analyzed frequency range. Similar changes were observed for sound clarity (C80) parameter, but only for 250 Hz, System B, the difference was above 1 JND (1 dB). In the case of T20, much bigger changes for low and mid frequency range were observed for System A.
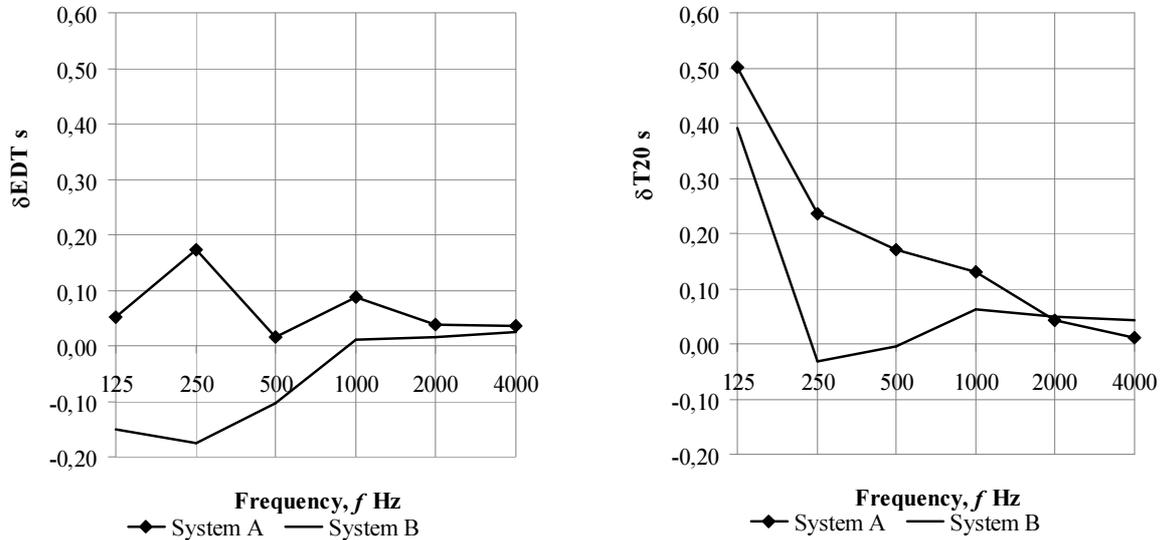
Figure 5 - Differences in EDT and T20 between ER and ALL operational mode for both systems.

Such a big changes are connected with non diffused acoustic field inside the room. In a perfect case, EDT and T20 are the same, as a result of linear decay curve. Sound enhancement systems increase reverberation time by adding some artificial reflections. If they are not evenly distributed in time and space, decay curve becomes not linear, what could be audible in room.

Both systems consists of loudspeakers mounted above listeners heads. According to calculations, apparent sound source positions, did not move noticeable up. There could be only observed some enlargement of the sound source width (Figure 6). Sound received by directional microphone, were wider especially for System A.
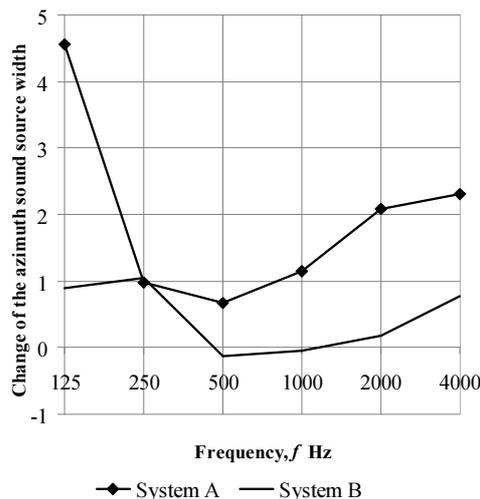
Figure 6 - Change of the azimuth sound source width for System A and System B.

4.2   Subjective assessment

For every compared pair, there were 240 answers (10 answers for single listener, 24 subjects). Correct discriminations are shown in Figure 7. Obtained results are of the binomial distribution. The null hypothesis assumed that the probability of giving the correct answer was 50%, that is it was assumed that the listeners didn't hear the differences between the stimuli. The hypothesis was rejected in all cases with standard level of significance p=0.05. The critical value from cumulative distribution function which doesn't allow the hypothesis to be accepted is about 58%.
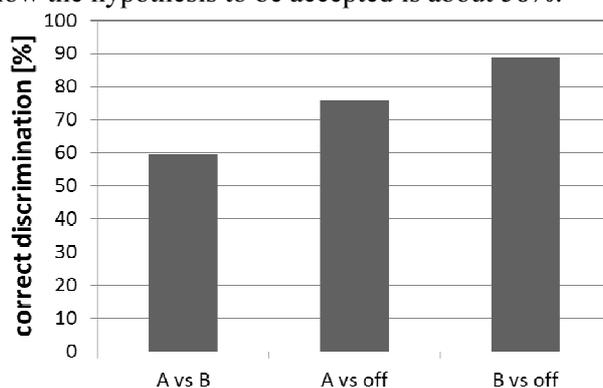
Figure 7 - Correct discrimination of sound enhancement systems (A vs B), and difference between the systems on and off (A vs off, B vs off).

On the basis of the results and calculations, some important conclusions can be stated. The listeners were able to discriminate if the sound enhancement systems were turned on. But the discrimination between particular systems, was not so obvious. In that case, there were only slightly more correct recognitions than statistically significant value. Special attention should be paid to the correct recognition of discrimination if system A or B were working or not. System B vs off was more than 10% more often correctly pointed than system A vs off. Knowing that the reverberation times with working system A and B were almost the same, it could be concluded that system B probably, apart from increasing reverberation time, adds some changes to the frequency, or spatial features of stimuli.

5   Conclusions

As part of work the authors carried out a comparison of the high-quality sound enhancement systems with similar technical parameters. To imitate as closely as possible the sound field of the room in which the systems were installed, the parametric ambisonic decoding was conducted.

Comparing reverberation times EDT and T20 for two sound enhancements systems it could be observed that EDT parameter increased quite evenly for all frequencies, while T20 gain is highly irregular. On the one hand for 125 Hz both systems increased the value more than 65% but for the other hand for 4000 Hz they did not exceed 20% gain. Both systems enlarged the azimuth sound source width, but only for System A that were significant values.

Listening test showed that the subjective differences between the test signals are statistically significant. Furthermore, the differences indirectly indicated which of the compared sound enhancement systems is characterized by higher quality. Further work will be focused on the analysis of acoustic parameters obtained for the tested systems and an attempt to determine their correlation with subjective evaluation.

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