ARCHIVES OF ACOUSTICS
34, 4, 559–577 (2009)

A Multi-Channel System for Sound Control in the Open Space

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(*received March 15, 2008; accepted October 19, 2009*)

In the paper, a multi-channel system for sound control is presented. The system enables creation of the sound space through generation of the acoustic signals with the help of sixteen independent controlled sound sources, in accordance with the accepted quality criterion.

The heart of the realized control system is digital filters system and the digital delay lines implemented in ADSP BF-537, the multi-channel card by Analog Device Company, equipped with the extension module. Besides the above-mentioned card with the microprocessor, the system is constructed of a number of subsystems, including the acoustic signal generation system, the multi-channels power amplifier and sixteen sound sources equipped with the special closed box enclosures.

The system design was preceded with the laboratory and model-and-simulation measurements and the verification measurements carried out in the field. The laboratory tests were made in free sound field conditions in the anechoic chamber of the Department of Mechanics and Vibroacoustics, AGH University of Science and Technology. The model-and-simulation research was carried out using the geometrical methods implemented in the Raynoise package. The final stage of the work included the tests confirming the operation effectiveness of the designed and realized system, which were carried out in the open space – in the selected area.

As a part of the model-and-simulate work and the verification research, the A-sound level and Speech Transmission Index (STI) distributions were determined in the control area and in the protected area.

Keywords: sound control, multi-channel systems, digital system, creation of the sound space.

1. Introduction

Along with the computer technique development, a concept of sound control has appeared. This term should be understood in the meaning of an attempt to solve comprehensively the problem of the hearing sensation creation. It should be emphasized that any activity in the field of forming an acoustic signal reaching a person should be based on the psycho-acoustic aspects of sound perception [14, 20].

The sound control should be also treated as a synthesis of some evolutionary changes in acoustic adaptation [10, 22] and signal processing [6, 19]. From that, it appears that the idea of sound control includes control of the sound source parameters, control of the sound signal parameters and control of the listener environment parameters [3, 12].

As the sound control systems we understand the systems, in which the controlled source of sound energy is employed for the modification of the existing acoustic field. They can be divided into two main groups: single-channel systems and multi-channel ones.

Another classification of the active sound controlling ways is determined by their applications. In this case we can distinguish four main directions of their development:

- the noise reduction systems,
- the systems controlling the sound sources parameters,
- the systems controlling the acoustic-field parameters,
- the systems for echo removing in the electro-acoustical and communication systems.

In the Polish literature there are only few published papers undertaking the subject of sound control. The following works: E. Hojan's paper [13], in which the author describes the principles of sound reinforcement system in rooms and open space, the work by Z. Engel and J. Kowal [8], presenting the systems and the principles of the vibroacoustic processes and the joint publications edited by A. Gołaś [11], deserve the special attention.

Outside Poland, the subject of sound control has drawn much more attention. Among the numerous foreign publications [15–17] come to the fore. The systems described in them enable modification of the acoustic field in different rooms by increasing the reverberation energy, i.e. increasing the reverberation time. Among the presented systems the following ones can be numbered: Reverberation with Delay (Rev/Delay), Reverberation on Demand (RODS), Assisted Resonance (AR), Multiple-Channel Amplification Reverberation (MCR), Electronic Reflection Energy System (ERES), Assisted Acoustics System (AAS) and Acoustic Control System (ACS). However, the specialists, solving these problems, emphasized special difficulties in the practical applications of the existing solutions. These systems work (with different results) only in several objects.

One of the more interesting solutions of the sound control systems, capable of being used in the concert halls, is a system of the French company CARMEN [5]. The proposed solution is based on the so-called "virtual walls" effect that enables to modify the acoustic field in the room through increasing the reverberation time depending on the kind of music.

To the wide array of the active systems for sound control belong also the surrounding sound systems. Currently, there exist several competitive multi-channel sound treatment systems. Dolby Surround (DS), Dolby Surround Pro Logic (DSPL), Dolby Digital (DD), Digital Theater Systems (DTS), Tomlinson Holman's eXperiment (THX) and Digital Cinema Sound (DCS) can be numbered among them. In the first stage of their development, these systems had applications in the movies industry as standards for transmission of sound accompanying the film projection. Currently, however, they are very popular with the users of the home theater systems [2].

Apart from the sound control systems described above, there are conventional sound systems available for both closed as well as open areas. They encompass centralized, decentralized and zone systems. The most advanced systems, operating in a centralized mode, use linear balanced sets or so-called line-array systems [24, 25]. All the above-mentioned systems dedicated for open space are only focusing on premium reproduction of sound into a particular area. Other components such as minimization of noise within the protected areas or provision of noise spatiality are not taken into consideration here. This paper presents an original system designed and put together by the author in order to enable not only the reproduction of sound but also to create sounds according to prescribed criteria.

2. Design of the multi-channel sound control system

The work was aimed at design and realization of the Multi-channel System for Sound Control (MSSC) that enables the sound synthesis in the internal area (controlled area), fulfilling the following requirements: good speech intelligibility, a uniform sound amplification, a spatial sound together with simultaneous minimization of the sound level in the external area (protected area).

This designated task required a definition of multi-criterion quality index that would include the above-mentioned criteria. The indicator in form of mathematical formulas is defined by Eqs. (1)–(3). In order to obtain the best solution we wish to arrive at a minimized global function of costs J_{system} by simply minimizing the indicators $J_{\text{area\_external}}$ and $J_{\text{area\_internal}}$:

- quality index for sound control system

$$J_{\text{system}} = f(J_{\text{area\_external}}, J_{\text{area\_internal}}), \tag{1}$$

- quality index for external area

$$J_{\text{area\_external}} = \frac{1}{M} \sum_{m=1}^{M} \left| p_{\text{m\_ext}}(r_m) - p_{\text{d\_ext}}(r_m) \right|^2, \tag{2}$$

• quality index for internal area

$$J_{\text{area\_internal}} = \frac{1}{N}\sum_{n=1}^{N}\left|p_{\text{n\_int}}(r_n) - p_{\text{d\_int}}(r_n)\right|^2, \tag{3}$$

where $p_{\text{m\_exl}}$ – pressure within external area model, $p_{\text{d\_ext}}$ – desired pressure within external area, r_m – location of sound receiver within external area, $p_{\text{n\_int}}$ – pressure within internal area model, $p_{\text{d\_int}}$ – desired pressure within internal area, r_n – location of sound receiver within internal area, M, N – number of receivers located in external and internal areas, respectively

To realize such a goal in practice, the control system has to generate the sufficient amount of acoustic energy to the main controlled area, together with minimization of energy generated to the acoustic insulated area. The key aim is surrounding the listeners with the created sound space. Only the multi-channel control system, with numerous sound sources, the power amplifiers and the acoustic signal generation system, is capable to create such a space. The hearth of such a system will be a signal processor with the software providing a very accurate real time processing of the signals, with a full simultaneous synchronization of the whole system work.

In order to design and build a multi-channel sound control system based on the above-mentioned criteria, it is necessary to go through a series of steps in accordance with the algorithm presented below:

1. Assume quality index.
2. Define control area.
3. Develop control system algorithm.
4. Design and build sound sources.
5. Design and build digital equalization filters.
6. Design and build digital audio delay lines.
7. Perform laboratory testing with provision of so-called free field for sound control subsystems.
8. Develop acoustic field model simulation tests for a single channel as well as multi-channel systems.
9. Define a sensitivity of selected acoustic field parameters for change of location, change of source power and directionality.
10. Construct a multi-channel sound control system.
11. Test the designed and constructed system in real conditions.

3. Laboratory testing of sound control subsystems

During the realization of the sound control system, the tests of individual subsystems were carried out subsequently. The laboratory tests were made in free sound field conditions in the anechoic chamber of the Department of Mechanics and Vibroacoustics, AGH University of Science and Technology.

A number of 16 identical wide-range speakers of Visaton Model BG17 type equipped with 10 litre volume closed enclosures were used for a design of the multi-channel sound control system. The shape of enclosures accepted for this project was a result of a compromise between assurance of the best sound processing acoustic parameters, price and a possibility of setting the identical sound sources into multi-channel electroacoustic systems. Therefore it was possible to assemble multi-channel sound sources from partial single sources; incorporating speaker columns, speaker matrices and other combination of sound sources.

Measurements of designed speaker enclosure directionality characteristics and impulse responses were performed during the preparatory steps.

Directionality characteristics were determined in two planes: horizontal and vertical (the sound source was placed on its left-hand side when facing its speaker). A pink noise filtered in octave bands with filters having frequency band midpoints at 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz and 8 kHz, were used during these tests.

Figures 1 and 2 show selected directionality responses determined in vertical and horizontal planes for octave frequency 125 Hz and 8 kHz of the build sound sources described in more detail in Ref. [4, 23].

Following steps included the design of digital filters and digital delay lines. As a result of analyses performed for various filter structures reviewed to build the sound control system, filters with a finite impulse response (FIR) were selected for a direct structure with a length of 1024 coefficients. These filters were built based on inverse filtration theory (unbraided) [19] and used for correction of the speaker responses [3, 4]. Figures 3–5 show amplitude-frequency responses of the enclosed loudspeaker, digital equalization filter and electro-acoustic system after equalization, respectively.

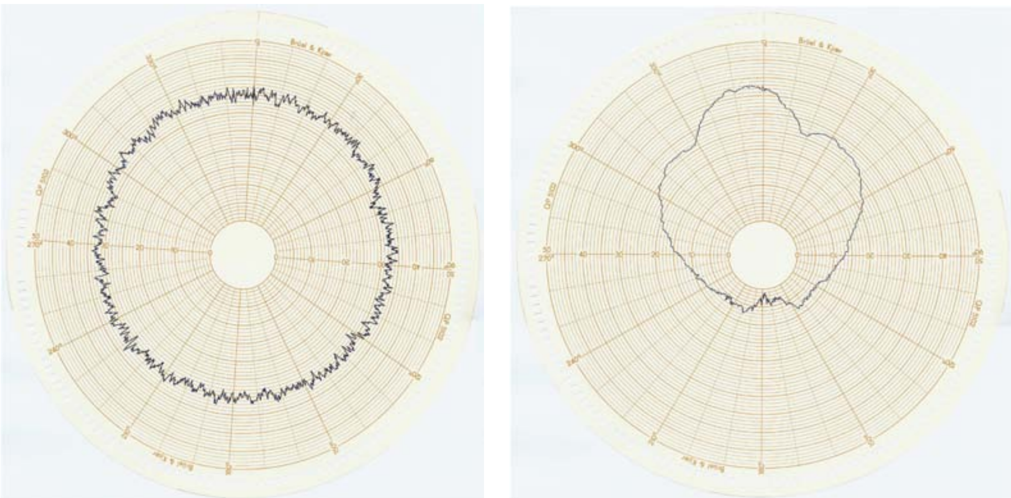

Fig. 1. Directional characteristics of BG17 loudspeaker in enclosure, determined within horizontal plane for octave bands 125 Hz and 8 kHz.

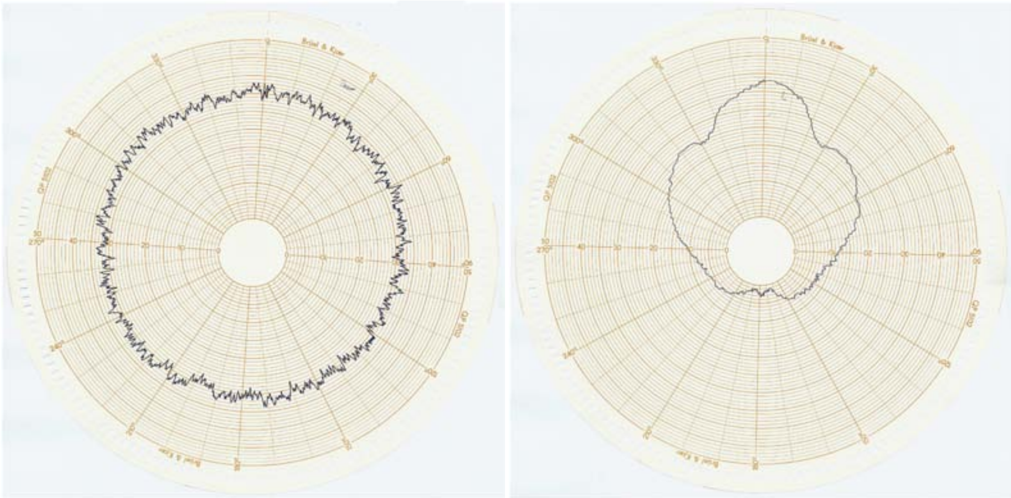

Fig. 2. Directional characteristics of BG17 loudspeaker in enclosure, determined within vertical plane for octave bands 125 Hz and 8 kHz.

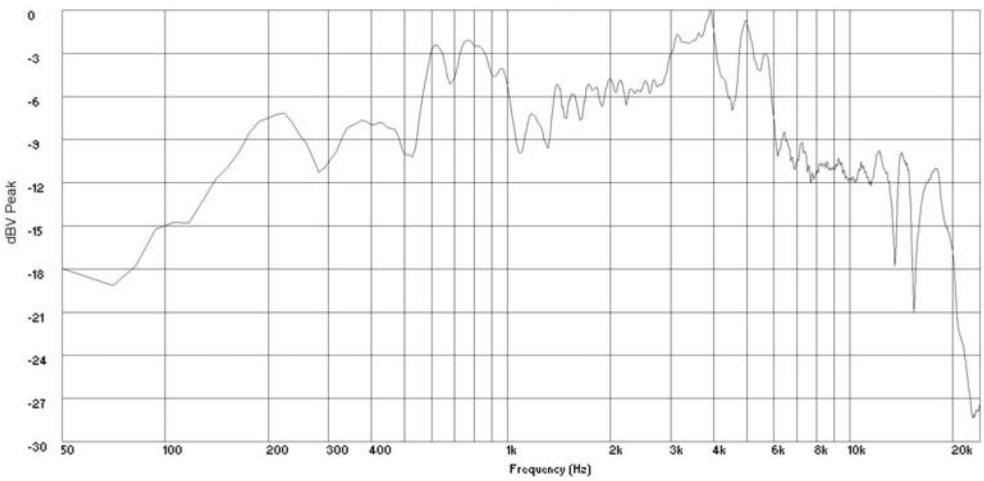

Fig. 3. Amplitude-frequency response of BG17 loudspeakers.

Also filters of infinite impulse response (IIR) of second degree were implemented within the system. These filters serve as parametric equalizers enabling (through a change of midrange frequency and quality coefficient) a modification of tone according to the user's preference. Digital equalization filters (FIR) were implemented for each input channel, while infinite impulse (IIR) filters were applied for the output channels. During the field testing in open space, the sound tone of processed signals were not modified with IIR filters.

Last step of the laboratory work phase encompassed the design and verification of digital delay line operation correctness. Those lines were implemented for

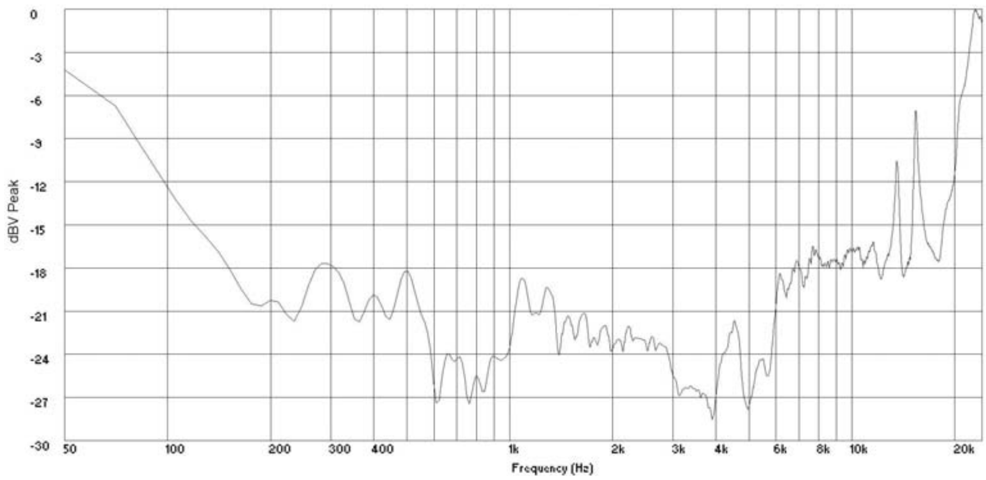

Fig. 4. Amplitude-frequency response of digital equalization filter.

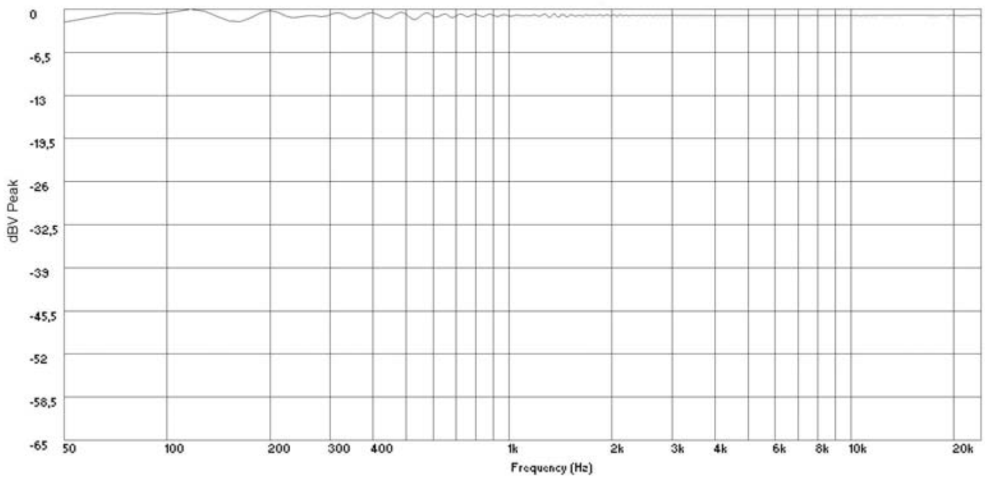

Fig. 5. Amplitude-frequency response of electro-acoustic system after equalization.

each of the particular sound sources presented in Fig. 9. Therefore, the sound sources S1 through S4 were operated without a digital delay line, i.e. the delay was 0 ms. Subsequent sound sources were equipped with the following delay lines: sources S5 and S8 – 9 ms delay, sources S9 and S12 – 18 ms, sources S13 and S16 – 27 ms, sources S17 and S2 – 35 ms, and finally sources S21 and S24 – 47 ms. The delays of those lines were determined based on knowledge of each sound source location in the system as well as sound velocity in the air for normal conditions.

After finishing the laboratory work stage, the model-and-simulation research was carried out using the engineering environment – the Raynoise package [21, 23].

4. Construction of the multi-channel system for sound control

The prototype of the Multi-channel System for Sound Control, MSSC, consists of a card with the processor, equipped with the analog-digital and digital-analog transducers, and of the additional components, necessary for correct operation of the system. These elements form two electro-acoustic lines, which are listed below. These are:

- ***The control line:*** the system for acoustic signal generation – the CD device, PC equipped with the card with the processor ADSP-BF537 and Analog Devices extension card, the multi-channel power amplifiers MPA6-150, the signal distribution system and the loudspeaker set;

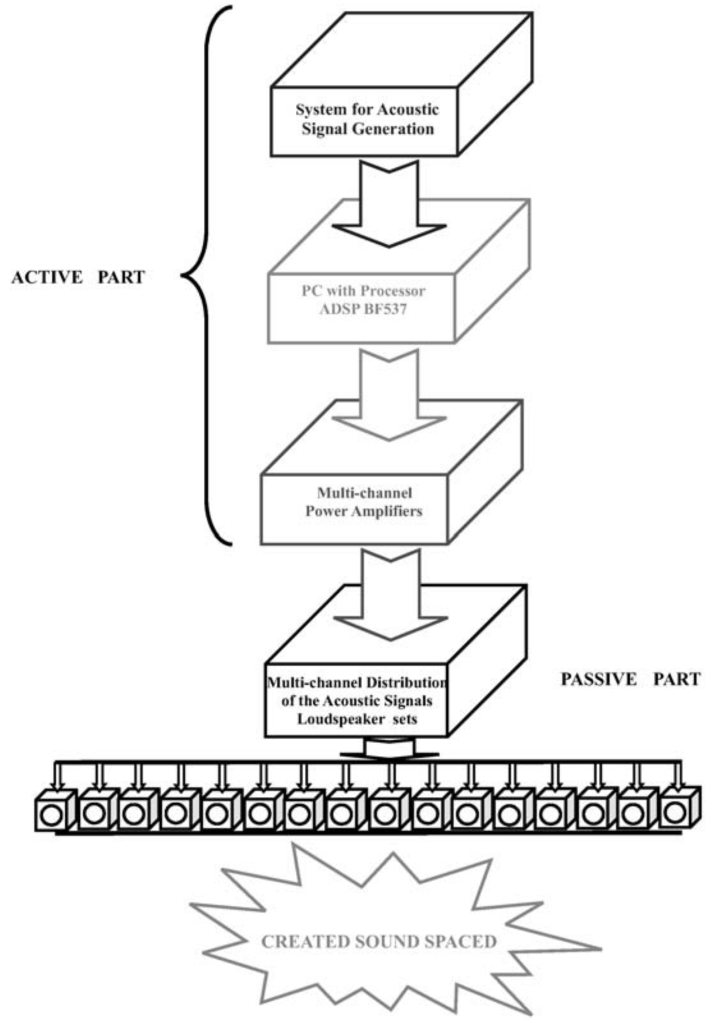

Fig. 6. The diagram of the Multi-channel System for Sound Control.

- ***The measurement line:*** the measurement microphone with preamplifier, the measurement amplifier and PC with software, sound meter Svan945A, the binaural microphones and the professional 2-channel mobile digital recorder.

The block diagram of the Multi-channel System for Sound Control is presented in Fig. 6. The algorithm for multi-channel sound control system is shown in Fig. 7.

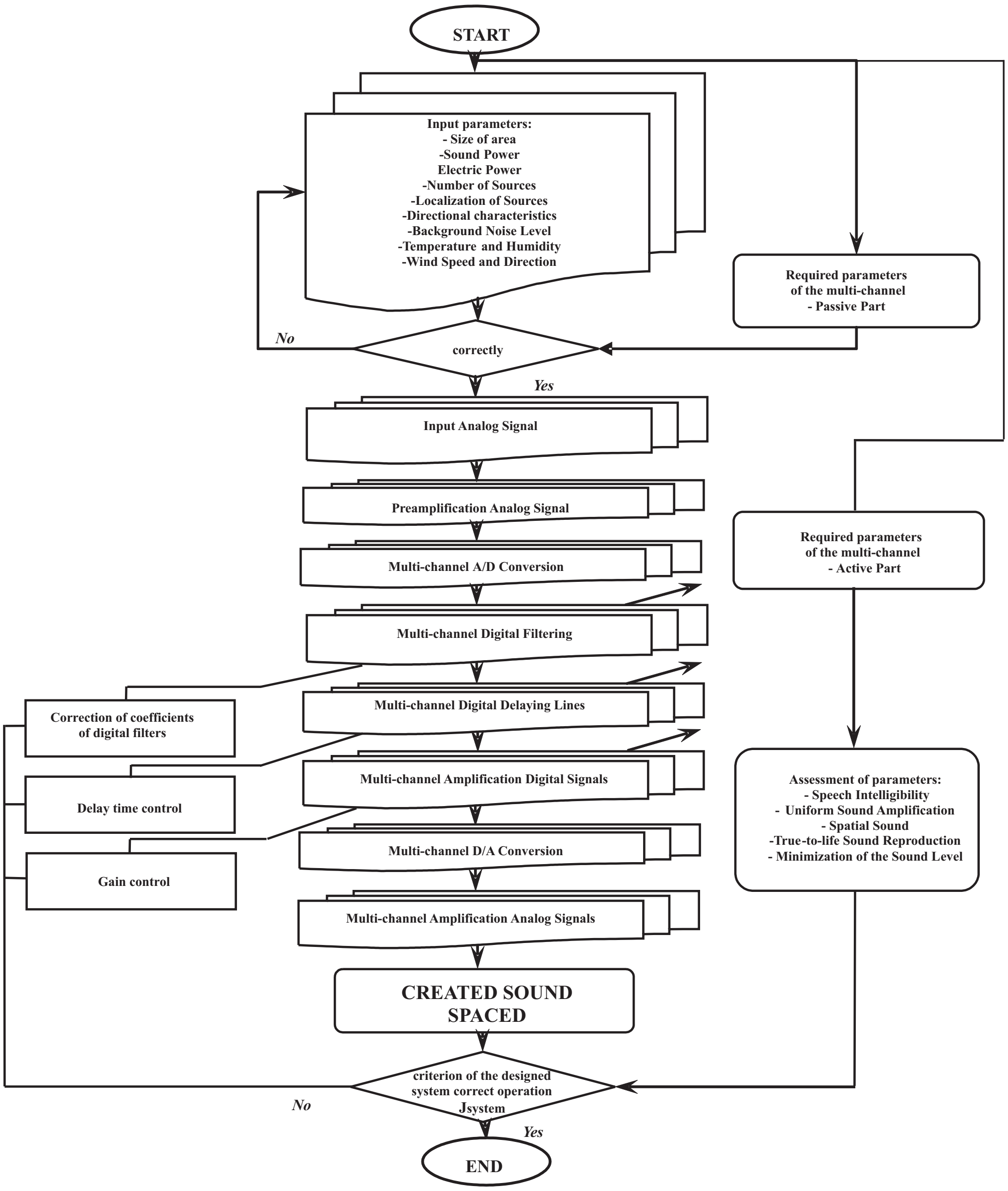

Fig. 7. The algorithm of the Multi-channel System for Sound Control.

The heart of the realized control system is a digital filters system, implemented in the multi-channel Analog Device card ADSP BF-537 with the extension module [1]. In the control systems being designed, the FIR (Finite Impulse Response) filters, the IIR (Infinite Impulse Response) filters and digital delay lines, were used. They are implemented on the above mentioned card with the use of the VisualDSP ++4.5 environment.

5. Computer modelling for sound control systems

Computer modelling process for the above-mentioned sound control system is presented below.

An AutoCad program was used during the first phase modelling, in order to draw a digital model of an area 19 meters wide and 25 meters long.

During the next phase of work, a model in the *.dxf file format was imported to the Raynoise software. Then 1900 virtual microphones were installed in the above-mentioned area – it was a network of receivers spaced at 0.5 meters from each other. This created a regular network of 1900 virtual microphones for which both the sound A level distributions and the speech transmission indices STI were determined. All field distributions were worked out at the height of 1.65 meters above the ground surface.

Sound sources were entered to the object model during the next phase of work, upon entering the geometric data. The sound source coordinates were subsequently determined for the selected model and their directional features as well as acoustic power levels set together with the time delays – delay forming lines. Simulation research was conducted for various sound source parameters, testing their sensitivity to the obtained results. Finally, due to practicality and for the coherence of model tests with subsequent work in real conditions, directional characteristics were entered for the designed and built broadband sound source, using the BG17 speaker made by Visaton.

Model and simulation projects were done for two fundamental systems: single channel SCSCS and multi-channel sound control systems MCSCS. Area models for the above-mentioned systems are presented in Figs. 8 and 9. Conical and triangular methods were used for model testing.

Acoustic field distributions for the open space single channel system were determined during the first phase of research work. A Raynoise computer software package was used in testing. The simulation researches were carried out for 40 reflections and 40 thousands of rays from the source. Visualization in form of maps was included in the final stage of testing.

The main purpose of the designed system was an adequate uniform sound amplification within an open space. A proper level of the acoustic power had to be entered into the system in order to meet this condition. Experience and knowledge of the author and numerous publications indicate that a proper dis-

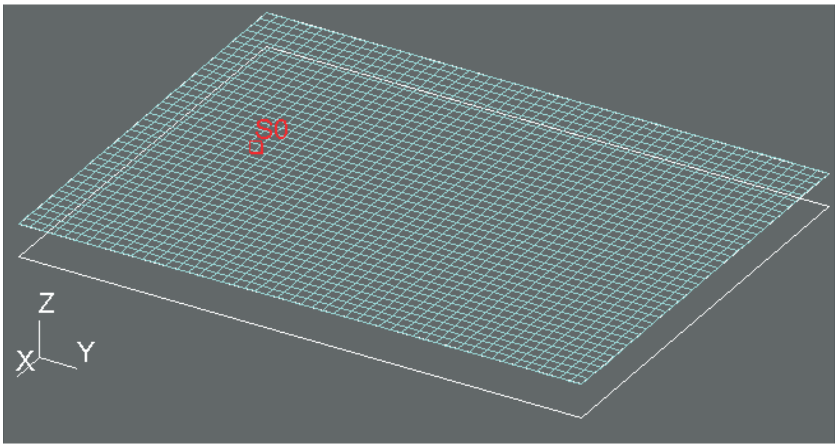

Fig. 8. Digital model of selected plot for single channel system sound control system SCSCS.

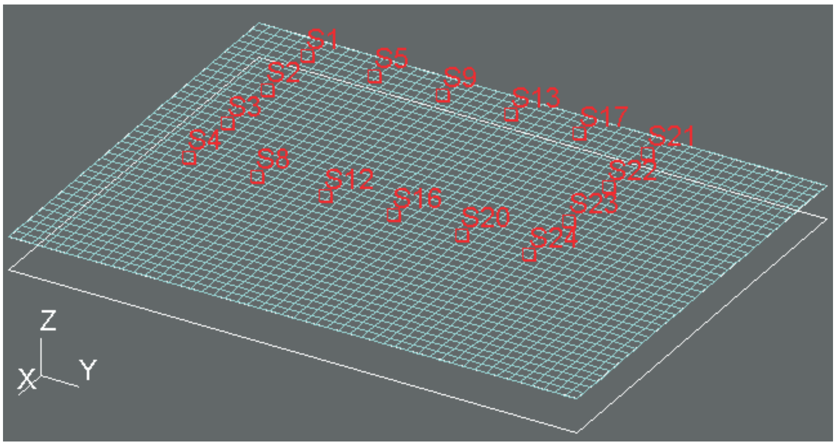

Fig. 9. Digital model of selected plot for multi-channel sound control system MCSCS.

tance of a useful signal from a noise background has to be ensured in order to provide uniform sound amplification within the set space. Therefore, as a result of knowledge gathered *a priori* and *a posteriori* for a single channel as well as multi-channel systems, it was assumed that the sound A level will be equal to 75 dB ±3 dB on the control area boundary.

In order to ensure the mentioned sound A level, the source acoustic power should be equal to 101 dB. This level of acoustic power was assumed for the simulation tests in all octave bands from 63 Hz to 8 kHz.

The models for a multi-channel system were developed during the subsequent phase of research. The source acoustic power level should be equal to 83 dB in order to guarantee a sound A level of 75 dB±3 dB within the analyzed area. That acoustic power level was assumed in simulation tests for all octave bands from 63 Hz to 8 kHz, for all sixteen sound sources. The model sound sources were distributed within a rectangular footprint 9 meters by 15 meters at the height of 1.7 meters.

Next, the time delays in a form of digital delay lines were entered for each source. Digital equalization filters were implemented for each input channel [4].

The acoustic field distribution maps for selected parameters were depicted in subsequent Figs. 10–13. Those maps show sound A level distribution as well as speech intelligibility distribution for single channel and multi-channel sound control systems.

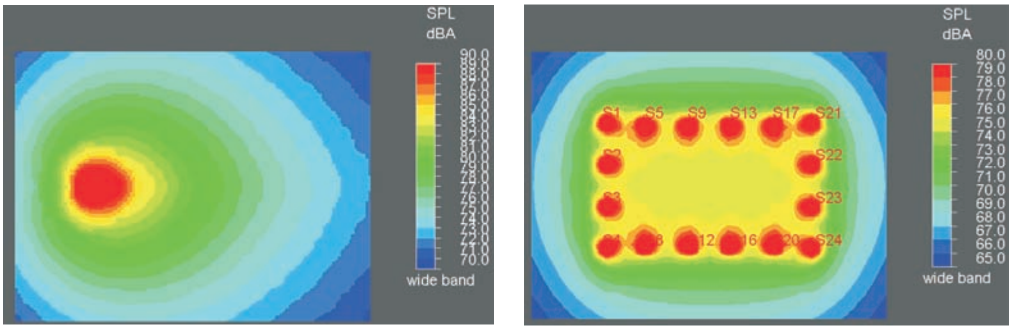

Fig. 10. Results of sound A level distribution for SCSCS and MCSCS for conical beam method.

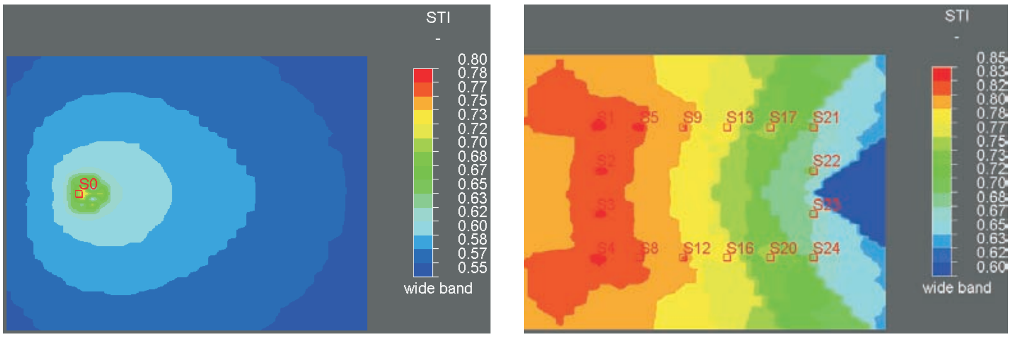

Fig. 11. STI index distribution results for SCSCS and MCSCS for conical beam method.

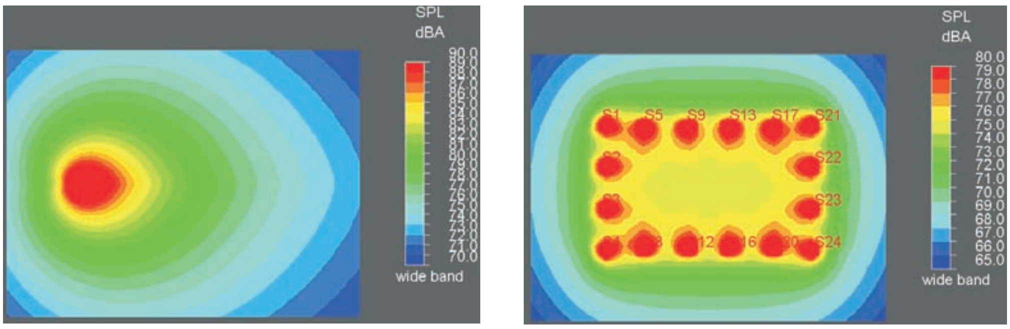

Fig. 12. Results of sound A level distribution for SCSCS and MCSCS for triangular beam method.

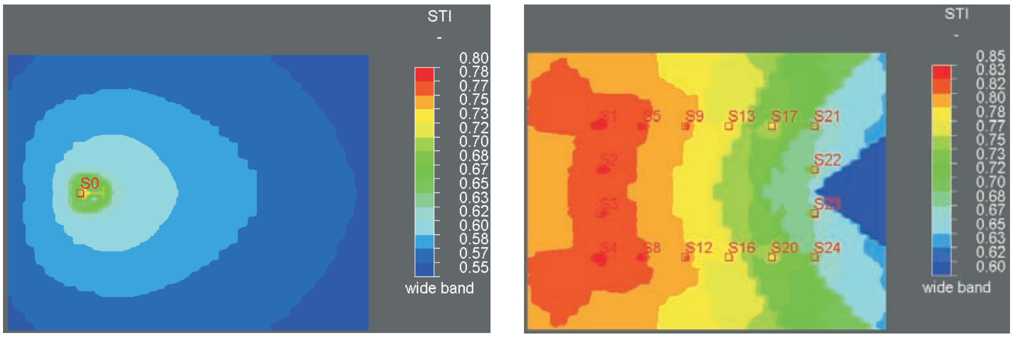

Fig. 13. STI index distribution results for SCSCS and MCSCS for triangular beam method.

6. Studies of the designed and realized systems in real conditions

In order to verify the operation correctness of the designed multi-channel system for sound control, the setup for the measurements in the real conditions in the selected area was built and the measurements of the designed system were carried out.

Arrangement of the individual sound sources in the selected area is presented in Fig. 14.

The experiments made were divided into three main stages. The first stage included the A-sound level measurements in the protected area in four points and

Fig. 14. The area with the Multi-channel System for Sound Control placed in it.

in five points in the area, where the sound was controlled. In the second stage, the speech intelligibility, measured by STI, was determined.

The final stage was audio monitoring carried out by the author team and independent listeners. Effects of these tests will be presented in the next article.

All the sound sources were designed in such a way, that they can match the column stands and the extending elements, i.e. connectors. This offers many possibilities for the arrangements of the sound sources at the different height. During the system planning, adequate lengths of the connecting cables and capability of their universal connecting in the longer lines were anticipated. However, in the final stage of construction and tests, all the cables have the same length, equal to 25 meters. In this way, the electrical "coherence" of the system was ensured.

In the experiment, the sound sources were placed at the distance of 3 meters from each other on the projection of the rectangle 9×15 meters. The principal axes of the loudspeakers were directed towards the controlled area. All the verification measurements were carried out on the height of the adult listener, equal to 1.65 meters.

The MLS (Maximum Length Sequence) signal was used both in the equivalent A-sound level measurements and in the measurements of STI.

The research procedure was as follows: the MLS noise was emitted to the system and the measurements of the equivalent A-sound level were carried out subsequently in the points from 1 to 9. The points from 1 to 4 were localized in the external area, while the points from 5 to 9 – in the internal area. The arrangement of the measurement points is shown in Fig. 15. The results of the first stage research are tabulated in Table 1.

The STI parameters were determined for 9 points, being inside the internal area, by means of the determined impulse responses, with the use of MLS sequence, too. The arrangement of the above-mentioned measurement points are shown in Fig. 16. The results of this stage of research are presented in Table 2.

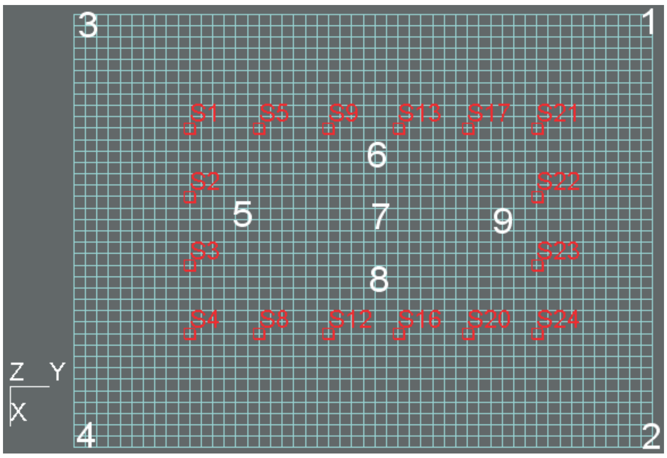

Fig. 15. The arrangement of the points in the measurements of the equivalent A-sound level.

Table 1. The results of the equivalent A-sound level measurements.

Measurement point	Equivalent A-sound level
1	64.7
2	65.3
3	65.4
4	65.5
5	77.1
6	76.6
7	76.4
8	76.8
9	74.7

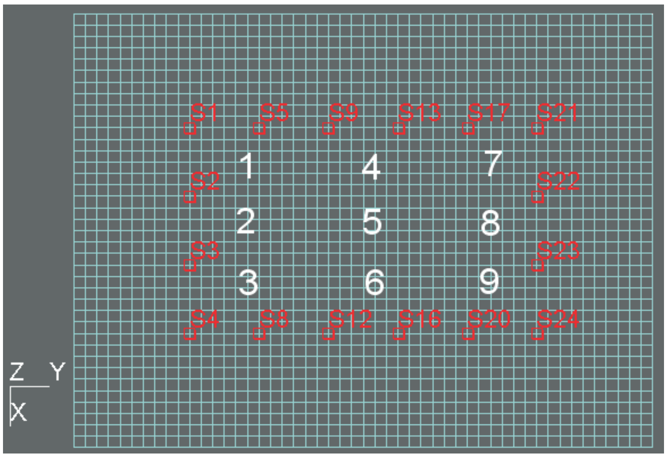

Fig. 16. The arrangement of the points in the STI measurements.

Table 2. The results of the STI measurements.

Measurement point	STI
1	0.73
2	0.73
3	0.79
4	0.73
5	0.75
6	0.77
7	0.76
8	0.76
9	0.81

The most interesting tests were carried out using the music recording on the CDs and introducing it to the system from the designed and realized system for the acoustic signals generation. They demonstrated the whole range of the designed system capabilities. The special sound effect was achieved by introducing of music through two stereo-channels and next, by passing them into each channel independently. In the channels, the signals were delayed by the digital delay lines. The first input signal was being passed into sources S1, S2, S5, S9, S13, S17, S21 and S22. The second input signal was being passed into sources S3, S4, S8, S12, S16, S20, S23 and S24 (Fig. 15 and 16).

7. Summary

In the paper, a multiple-channel system for sound control is presented. The system enables creation of the sound space through the acoustic signals generation with the help of sixteen independent controlled sound sources, according to the accepted quality criterion.

Before the sound control system was created, the simulation tests for the single- and multi-channel systems were performed. The analysis of the obtained distributions of the single-channel system of distributions proved a very non-uniform distribution of the acoustic field featured by the A-weighted sound level and the STI. A difference of 15 dB within the A-weighted sound level between the start area located in close proximity of the source and the boundary of the control area was determined. The STI for most part of the area was below 0.6. This value sets only an acceptable level of the speech quality.

It was also found that introduction of delay line in a single channel system does not produce changes in the acoustic field distribution. This is caused only by a time shift of a sound generated from a single sound source.

The obtained simulation test results depict how a difficult task may be the attempt to control the acoustic field parameters using a single channel sound-control system. For this reason it is necessary to design and use a multi-channel sound control system, and adequate selection of control parameters for a particular set area.

The analysis of the obtained distributions of the multi-channel system proved a very uniform distribution of the acoustic field featured by the A-weighted sound level and the STI speech transmission index. The distributions of the A-weighted sound level within the control area were maximally 3 dB for the conical and triangular methods.

At the same time it was observed in the corners of the analyzed area, within the distance of 6 meters, that the sound level A dropped by 10 dB. Through application of the multi-channel system, very good values of the speech quality of above 0.75 were obtained for most part of the area, whereas for the remaining part the values were above 0.69.

The model and simulation tests for both the conical as triangular method provided almost the same results of distribution of the A-weighted sound level and the STI speech quality. The obtained results of the model and simulation tests and verification tests performed in the real conditions were very similar and proved the operation efficiency of the multi-channel sound control system in relation to the design assumptions.

According to the measuring point accepted as the reference from the internal area, the decrease of the A-weighted sound level within the range from 10 to 12.4 dB was already recorded at the distance of 6 meters for the external (i.e. protected) area. The equability of the sound amplification within the controlled (internal) area was of 2.4 dB order.

The results of the second stage of the study are very satisfactory as well. The STI values from 0.73 to 0.81 were obtained for the points within the controlled area. Thereby, it can be concluded that a good and a very good speech quality within the internal (controlled) area were guaranteed thanks to the MCSCS system. Sixteen identical sound sources with the digital equalizing filters and the delay lines made it possible to obtain a very good sound surround effect.

It was also observed during the tests that application of a greater number of sound sources of set directivity patterns located around the controlled area induced a significant decrease of demand for acoustic power required for synthesis of the sound space.

Currently, the designed Multi-channel System for Sound Control, MSSC, enables sound controlling according to the set criterion in the open space. In the future it will be used to create the sound structure in the interiors.

Besides the low price, the solution proposed by the author, is characterized by: large flexibility and universality of its applications and a fully open architecture of the system. In practice, this means the possibility to interfere in the hardware and software parts of the individual subsystems of the sound control system.

In the presented multi-channel sound control system it is possible to implement the previously designed selected structures of the FIR and IIR digital filters, the digital delay lines, and the elements for amplification and suppression of digital and analog signals. The great advantage of the designed and realized system is its capability to create the spatial sound around the listeners, both according to the objective criteria and the subjective ones.

To sum up, the designed Multi-channel System for Sound Control enables creation of the sound space in the controlled area (i.e. internal one) with simultaneous minimization of the noise in the external area.

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