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SPATIAL EQUALIZATION OF SELECTED SOUND SOURCE BY DIGITAL INVERSE FILTERING

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The article presents digital inverse equalizers, enabling active sound control by means of directional characteristics equalization of selected sound source. Theoretical grounds of these systems are based on inverse filtration theory. Mathematical model of the equalizer, as well as experimental results confirming effectiveness of its operation have also been described in the study, which presents directional characteristics of Visaton BG17 loudspeaker, and its selected impulse responses prior to, and after equalization.

Keywords: active sound control, digital inverse equalizer.

1. Introduction

Active sound control systems **ASC** are systems where the controlled source of sound energy is applied for modifying the existing acoustic field. They can be divided into two groups: single-channel and multi-channel systems.

Another classification of active sound control systems determines their practical applications. In this case, four basic development trends may be recognized, namely:

- noise reduction systems,
- acoustic loudspeaker parameters control systems,
- acoustic field parameters control systems,
- serving echo removal systems.

There is not many works in Polish literature, undertaking sound control issue. The study of E. HOJAN [11], in which the author describes principles of sound amplification for rooms and for open space, Z. ENGEL and J. KOWAL [7] work, presenting systems and principles of vibroacoustic processes control, as well as multi-author work [10] edited by A. GOŁAŚ, deserve special attention here. Publications concerning issues of electroacoustic transductors, including ŻYSZKOWSKI [22, 23] and DOBRUCKI [4] works, constitute separate group. Works of H. ŁOPACZ [18], W. CIESIELKA [1, 2],

P. DZIECHCIŃSKI [5] can be counted among studies dedicated to digital equalization of sound sources that have been published recently in Polish literature.

Issues of active sound control gained much more attention abroad. The systems being described there, mainly include three initial groups of systems [3, 6, 12-14, 19, 21].

From among many foreign publications regarding issues of sources' parameters as well as acoustic field control with use of inverse filtration theory, the works [8, 9, 16, 17] have been given top priority.

The aim of this study is to prove a thesis, that equalization of directional characteristics of selected sound source is possible with use of digital inverse equalizers.

The study presents mathematical model of inverse equalizer; results of laboratory tests confirming effectiveness of its operation have also been given.

2. Testing of selected loudspeaker properties

Broadband loudspeaker Visaton model BG17 was used for designing of inverse digital equalizer. Selected acoustic and non-acoustic parameters of the loudspeaker are given in Table 1 and Fig. 1.

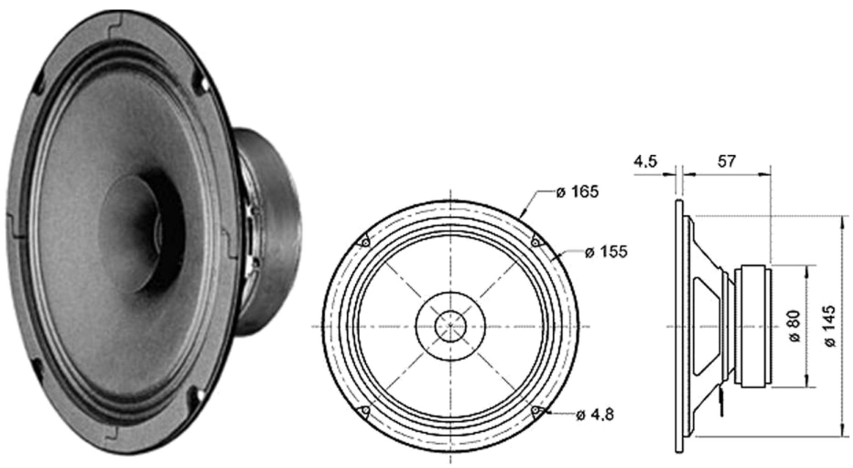


Fig. 1. Visaton BG17 loudspeaker and its dimensional sketch [20].

Table 1. Selected parameters of BG17 loudspeaker.

Rated power	40	W
Nominal impedance (Z)	8	Ohm
Frequency response	80–20000	Hz (−10 dB)
Effectiveness	93	dB (1 W/1 m)
Resonance frequency (fs)	110	Hz
Total quality factor (Qts)	0.64	
Equivalent volume (Vas)	9.7	l

Designing and production of BG17 loudspeaker closed box enclosure was initial stage of conducted works. The enclosure of 10 litres in volume, together with the loudspeaker mounted in it, is shown in Fig. 2. The shape of closed box enclosure adopted for implementation resulted from compromise between ensuring the best acoustic parameters of sound processing, price, and possibility of combining identical sources into multi-channel electroacoustic systems.

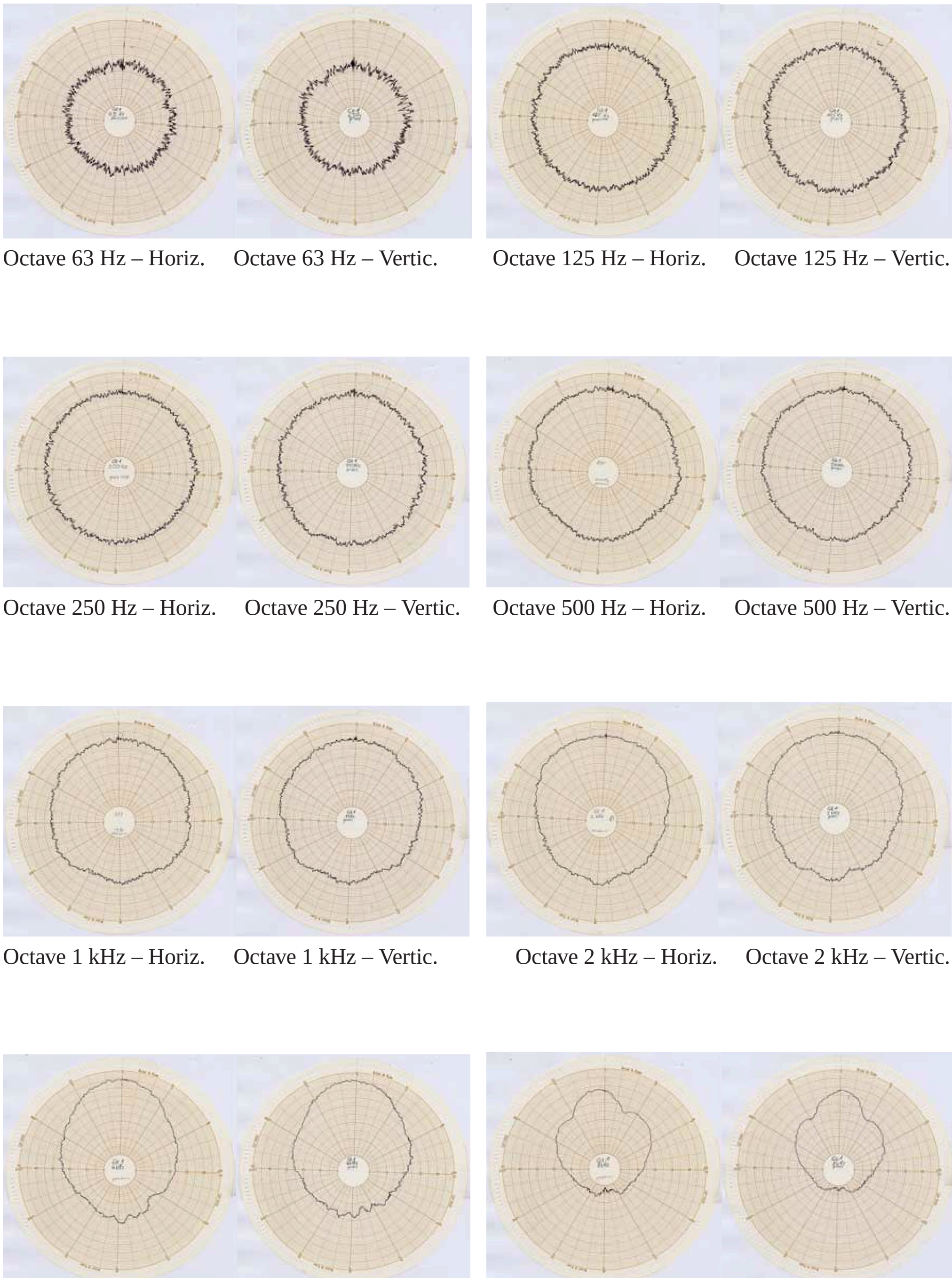
Fig. 2. The station for directional characteristics' measurement of Visaton BG17 loudspeaker in closed box enclosure.

Measurements of directional characteristics, as well as impulse responses of designed loudspeaker and closed box enclosure unit, were accomplished after completion of preparatory works.

Measuring station was set up in anechoic chamber of Mechanics and Vibroacoustics Department in AGH University of Science and Technology. The object under tests was Visaton BG17 loudspeaker.

Measuring station, constructed from two channels: transmitting one and receiving one, was used for directional characteristics testing.

Transmitting channel was set up from the following devices: pink noise generator B&K 2133 analyser, B&K 1614 filter, ASHLY FET1500 power amplifier and BG17 loudspeaker in enclosure. Receiving channel consisted of B&K 4134 measuring microphone, B&K 2619 microphone preamplifier, B&K2610 measuring amplifier, and B&K 2305 recorder. Rotary table B&K 3921, on which the loudspeaker in closed box enclosure was located, has been also employed in testing. Both during testing of directional



Octave 63 Hz – Horiz. Octave 63 Hz – Vertic. Octave 125 Hz – Horiz. Octave 125 Hz – Vertic.

Octave 250 Hz – Horiz. Octave 250 Hz – Vertic. Octave 500 Hz – Horiz. Octave 500 Hz – Vertic.

Octave 1 kHz – Horiz. Octave 1 kHz – Vertic. Octave 2 kHz – Horiz. Octave 2 kHz – Vertic.

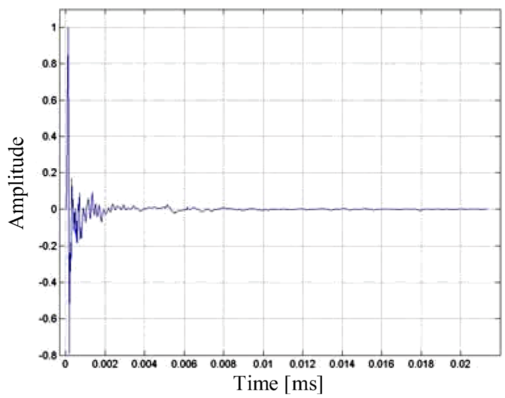
Octave 4 kHz – Horiz. Octave 4 kHz – Vertic. Octave 8 kHz – Horiz. Octave 8 kHz – Vertic.

Fig. 3. Directional characteristics of BG17 loudspeaker in enclosure, determined within horizontal and vertical planes for octave bands.

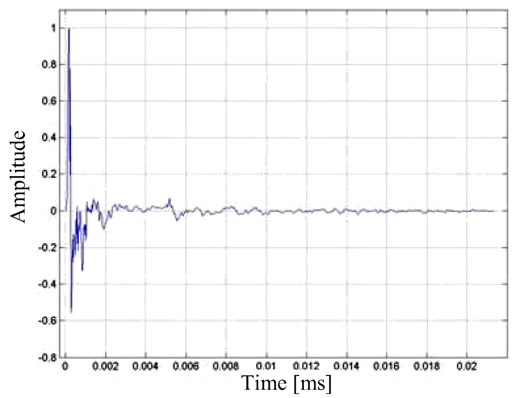
characteristics and impulse responses, the microphone were placed in 1 m distance from the loudspeaker.

In Fig. 3 directional characteristics are presented, determined in two planes: horizontal and vertical one (the source being located on the left side, when viewed in front of the loudspeaker). Pink noise was employed in testing, which was filtered within octave bands by means of filters having mid-band frequencies respectively 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz and 8 kHz.

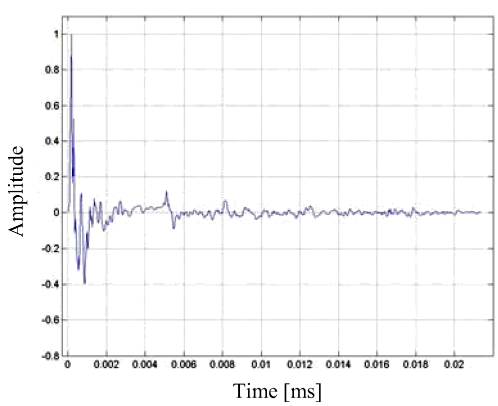
Impulse responses of BG17 loudspeaker in enclosure were determined during second stage of research. Measuring station constructed from two electroacoustic channels: transmitting one and receiving one was used for testing purposes. Transmitting channel



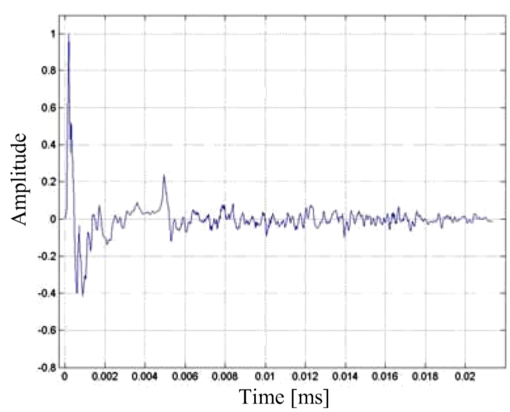
Impulse response along main axis of loudspeaker – 0° angle



Impulse response for 30°C deflection angle from main axis of loudspeaker



Impulse response for 60°C deflection angle from main axis of loudspeaker



Impulse response for 90°C deflection angle from main axis of loudspeaker

Fig. 4. Impulse responses of BG17 loudspeaker in enclosure, determined for various angles within horizontal and vertical planes.

is composed of the following equipment: PC equipped with Sound Blaster Audigy 2 sound card and card with ADSP-BF537 signal processor, ASHLY FET1500 power amplifier and BG17 loudspeaker. Receiving channel is composed of N-1220 microphone, N-1201 preamplifier, GRAS measuring amplifier and PC.

During research Matlab software package was used – in which inverse equalizers were designed, as well as CoolEdit, Aurora and Sample Champion software. The testing process consisted in determination of impulse response with use of MLS (Maximum Length Sequence) technique, followed by finding of inverse digital filter, which utilization would enable compensation of BG17 loudspeaker.

For purposes of MLS sequences generation, recording, as well as determination of impulse response, CoolEdit, Aurora and Sample Champion software packages were used. Experimental data were recorded on PC hard disk in digital "wav" format with 16 bit resolution and 48 kHz sampling frequency.

Figure 4 contains selected time characteristics – impulse responses of BG17 loudspeaker in closed box enclosure, determined each 30 degrees around the loudspeaker within vertical and horizontal plane.

3. Inverse equalizer – mathematical model and experimental testing

The inverse filtration theory has been known for a number of years; however its practical application has been made possible with the development of the digital signal processing systems, accompanied by the common availability of rapid signal processors, allowing the real-time signal convolution and deconvolution.

This theory has served the designing and construction of the digital inverse equalizer.

In order to design the digital inverse equalizer, the transfer function $H_{\text{inv}}(z)$ needs to be arrived at so as to meet the relation (1).

$$H(z) \cdot H_{\text{inv}}(z) = 1. \tag{1}$$

Where the minimum-phase systems are considered, this should not pose significant problems. However, the electroacoustic systems are rarely described by the minimum-phase systems, as these are mostly maximum-phase or mixed-phase systems.

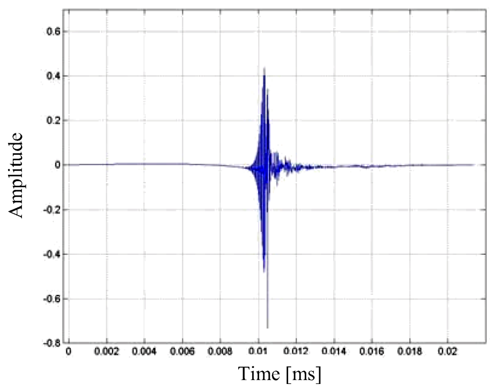
There are many methods of inverse equalizers designing. One of them is Mourjopoulos method, and that is the method employed for designing of above-mentioned equalizers.

The method allows for obtaining impulse response of inverse filter, by means of least squares method. The issue in this case for problem solution is finding unknown impulse response of inverse filter g, while having at disposal original impulse response and expected result of these two values convolution, i.e. Dirac delta function.

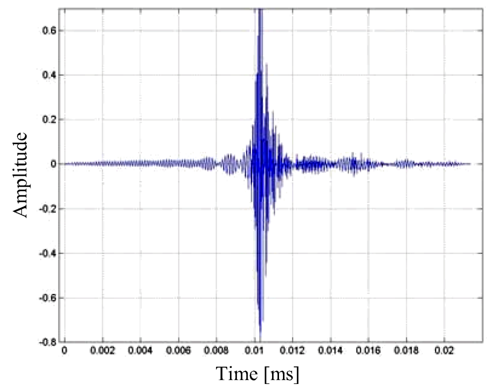
$$[R] * \{g\} = \{d\}. \tag{2}$$

This exercise results in obtaining inversed copy of original impulse response h. $[R]$ matrix is Toeplitz matrix, for which individual elements in the first row are values of original signal autocorrelation function. This method is at present most widely disseminated, because of correct results obtained when using it, simple calculations and no necessity of system stability considerations with any change of filter order. An algorithm was developed for reversing of original impulse response, which has been implemented in Matlab engineering software package.

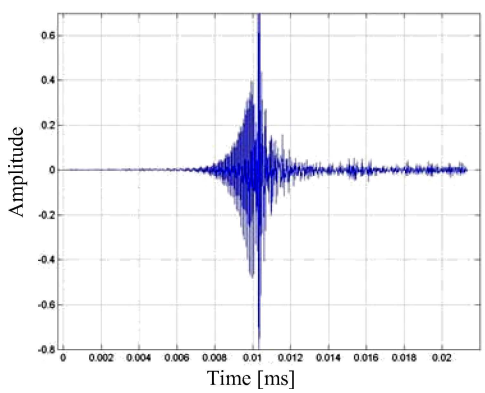
Digital inverse equalizer has been implemented on ADSP-BF537 card with Analog Devices signal processing unit. Selected inverse equalizers are presented in Fig. 5.



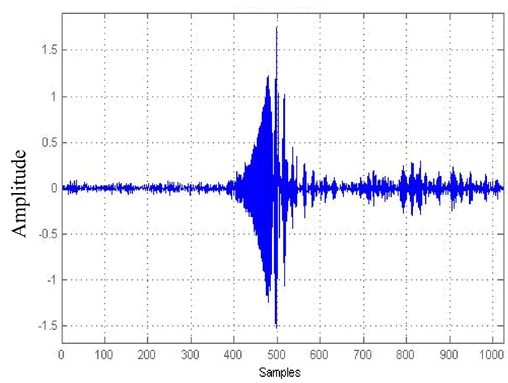
Main axis of loudspeaker – 0° angle



Deflection 30° from main axis of loudspeaker



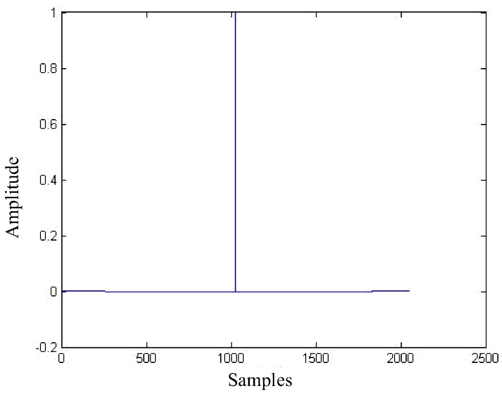
Deflection 60° from main axis of loudspeaker



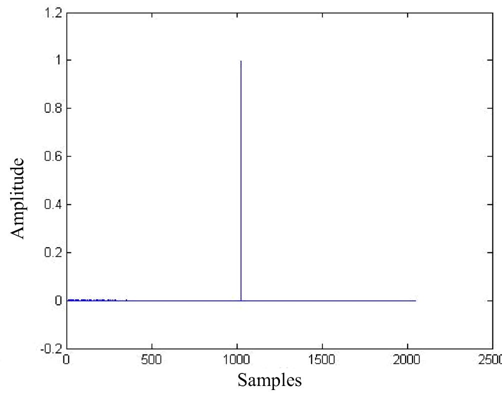
Deflection 90° from main axis of loudspeaker

Fig. 5. Inverse equalizers designed for purposes of BG17 loudspeaker in enclosure correction, for various angles of source radiation.

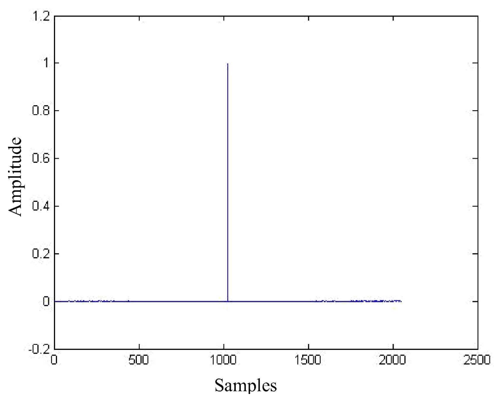
Figure 6 presents obtained impulse responses of BG17 loudspeaker in closed box enclosure, after correction with digital equalizer.



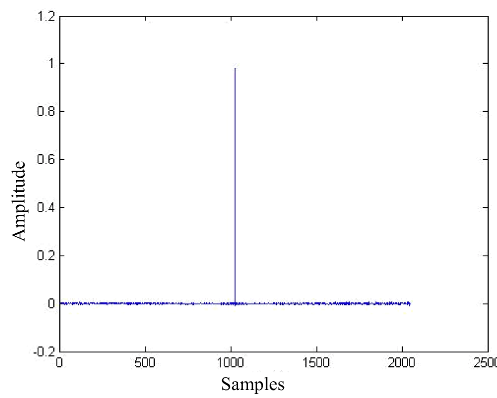
Main axis of loudspeaker – 0° angle



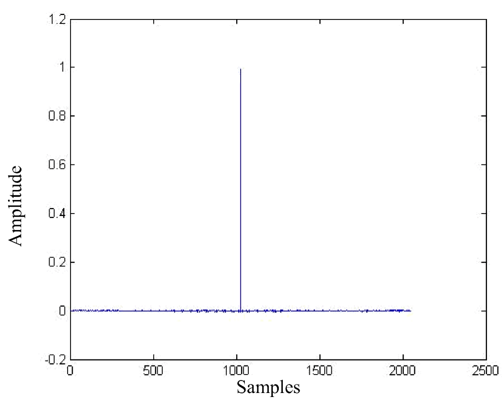
Deflection 30° from main axis of loudspeaker



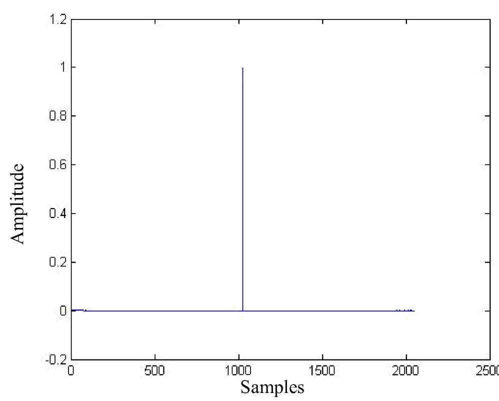
Deflection 60° from main axis of loudspeaker



Deflection 270° from main axis of loudspeaker



Deflection 300° from main axis of loudspeaker



Deflection 330° from main axis of loudspeaker

Fig. 6. Impulse responses of BG17 loudspeaker in closed box enclosure after correction with digital inverse equalizer.

4. Summary and conclusions

The results presented in this study are partial results of works aimed at designing of multi-channel system of sound control. The system would be ultimately built of several wide-band sources. Therefore, considering economic and geometrical issues, utilisation of BG17 loudspeaker has been decided.

The work was divided into three basic parts: analytical, designing and experimental ones.

Analyses of characteristics specified by manufacturer resulted in finding that BG17 loudspeaker under testing had inhomogeneous amplitude-frequency characteristics, of approximately ± 10 dB within 80 Hz up to 20 kHz band. Designing of 10 litres closed box enclosure did not basically change these characteristics, either, what should be expected.

During first stage of works, closed box enclosure of BG17 loudspeaker was designed and its directional characteristics within horizontal and vertical planes were determined for octave bands in range from 63 Hz up to 8 kHz. It has been found in result of analyses, that directional characteristic for octaves from 63 Hz up to 500 Hz, both vertically and horizontally, is close to omnidirectional characteristics, with 10 dB inhomogeneity within above-mentioned band. Within frequency range from 1 up to 2 kHz, yet significantly higher differences, reaching 20 dB, has been observed. For 4 kHz frequency it was 30 dB, and for 8 kHz up to 40 dB.

Having *a priori* and *a posteriori* knowledge of existing methods of inverse equalizers designing methods, the method based on theory being developed by Mourjopouls was chosen for designing inverse digital equalizers.

On the grounds of impulse responses analysis, lengths of inverse digital equalizers were determined. The decision was taken to design FIR filters of 1024 length.

For purposes of digital inverse equalizers designing, programs were developed and programmed with Matlab engineering software package.

After completion of designing stage, digital inverse equalizers were implemented on Analog Devices ADSP-BF537 card, and then audio monitoring tests have been done, which confirmed effectiveness of adopted solutions.

Thanks to use of inverse equalizers almost ideal compensation of amplitude-frequency characteristics was obtained within frequency interval of 80 Hz up to 20 kHz ± 0.5 dB, for angles in range of -90 up to 90 degrees. Thereby effective range of transducer operation was significantly widened, as well as possibility of directional characteristics correction of above-mentioned loudspeaker demonstrated both in vertical, and in horizontal planes.

Attempts of the loudspeaker characteristics correction within extended range, for radiation angles within 120 up to 240 degrees range, were undertaken in course of works realization. For this range of transducer radiation, satisfactory effects of equalization have not been, however, obtained. The reason for this was very high inhomogeneity of amplitude-frequency characteristics of the loudspeaker.

One ought to sum up, that a spatial equalization of characteristics is possible for the loudspeaker in closed box enclosure for angles presented above in the vertical as well as horizontal plain. Designed inverse equalizers after implementing them on the ADSP537 card can be e.g. sequentially switch depending on the direction of the listener localization with the relation of the sound source.

Designed and produced instrument can be utilised in active sound control systems, as one element of multi-channel system.

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