ARCHIVES OF ACOUSTICS
31, 1, 89–97 (2006)

ACTIVE CONTROL OF SOUND BY MEANS OF DIGITAL EQUALIZERS

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(*received April 30, 2005; accepted October 25, 2005*)

The article describes single-channel digital systems, enabling active sound control according to the set criterion. Theoretical fundamentals of these systems are based on digital filter theory, using which the digital graphic, phase, inverse and adaptive equalizers have been designed and programmed. The described systems have been implemented into two cards with signal processors TMS320C25 and TMS320C31 made by Texas Instruments. The article presents mathematical models of the above mentioned equalizers and results of the experimental tests, which verify their performance.

Key words: active sound control, digital equalizer.

1. Introduction

Active sound control systems **ASC** are the systems, in which controlled source of sound energy is used for modification of the existing acoustic field [4]. ASC systems can be divided into two basic groups: single- and multichannel systems.

Another division of active sound control systems determines their practical applications. In this case we can distinguish three basic trends of development of active sound control systems, namely:

- noise reduction systems,
- control systems of acoustic field characteristics,
- systems for echo elimination in electro-acoustic or communications systems.

Most published works on active sound reduction systems describe single- or two-channel systems. This choice is not made by accident and results from two premises. The first one is the ultimate use of the system (unidimensional systems i.e. hearing protector, wave-guides or multidimensional – 3D space). The second one is directly connected with the costs of designing of the system (signal processor cards, electro-acoustic transducers). Due to large financial outlays resulting from application of the latest technologies, in the world there are only few corporations which have built multichannel systems of active noise reduction. They include the following companies: TechnoFirst, Dassault Electronique – France, Lord Corporation, Digisonix, Active Noise and Vibration Technology – USA, Ultra Electronics Ltd., Noise & Vibration Audio System – Great Britain, Toyota, Nissan – Japan.

Another group of applications are systems of active space sound control; this relates first of all to concert halls, auditoriums and temples, i.e. objects of relatively large dimensions. Among many systems, which have been used for modification of acoustic field indoors, are the following systems: Delay/rev., RODS, AR, MCR, ERES, AAS, ACS [5, 10, 11, 15], or French system made by CARMEN.

A wide range of ACS complete the Dolby Surround systems. Today there are several competitive multichannel sound processing systems. They include Dolby Surround, Dolby Surround Pro Logic, Dolby Digital, DTS, THX or DCS. These systems at their initial stage of development have been used in cinema applications as a format of transfer of the image projection sound and now they are very popular among the Home Theater users.

This article describes four active sound control systems i.e. digital graphic, phase, inverse and adaptive equalizer. These systems can be used as independent systems modifying acoustic climate in the limited space, or be an element of multichannel systems of active sound control.

2. Digital equalizers versus sound control criteria

Sound control is to be understood as intentional intervention in the sound structure – defined by space-time-frequency relations – in order to modify it according to the set criterion.

These criteria can be divided into two basic groups. The first one includes "*objective criterion*" such as: quality of speech, reduction of acoustic pressure level, uniform sound amplification, 3D listening effect etc. The second group includes "*subjective criteria*" equally important, which depend on user's individual liking such as tone quality.

Until recently, to control sound analog systems have been used, today due to dynamic development of electronics and IT sector, the sound can be controlled digitally. Special devices called signal processors have been designed, which allow real-time signal processing (Digital Signal Processing).

Based on these devices many systems have been designed, enabling spatial sound control. The word equalizer has also gained a new meaning, which until recently was regarded as an analog device with all advantages and disadvantages of analog systems. Today when we use the word equalizer, we have in mind such digital systems as *digital graphic equalizers* allowing to control the quality of sound on the amplitude-frequency level, *digital phase equalizers* allowing to create a structure of sound on amplitude-phase level, *digital inverse equalizers* allowing to compensate characteristics of electroacoustic circuits, as well as the latest *digital adaptive equalizers* combining these functions, allowing to control the sound according to the set criterion.

This article presents mathematical models of the above mentioned equalizers and the results of laboratory tests, which verify their performance.

3. Graphic equalizer

Based on the card of the fixed-point signal processor TMS320C25 by Texas Instruments, OPAL language and computer graphics created in C language, a system has been designed allowing the user to control the sound on the amplitude-frequency level – sound quality control [1].

The equalizer was built using the structures of low-pass and high-pass filters described by relations (1) and (2). The selection of these filters was dictated by the fact of limited designing capabilities of the available signal processor. For this reason, for further analysis the following model has been adopted. The frequency band has been split into five ranges. Filters with mid frequencies of 250 Hz, 1 kHz, 4 kHz have been designed as combination of two types of filter, low- and high-pass filter, which in the result gave a bandpass filter. The filter limiting the band from below has been designed as a low-pass filter with a cut-off frequency of 100 Hz, and from the top the band has been limited using a filter with a cut-off frequency of 10 kHz. Sampling frequency was 30 kHz.

$$H_{\text{Low}} = \frac{\alpha_L}{[1-(1-\alpha_L)z^{-1}]}, \tag{1}$$

$$H_{\text{High}} = 1 - \frac{\alpha_H}{[1-(1-\alpha_H)z^{-1}]}. \tag{2}$$

The front panel of the digital graphic equalizer *DGE* is shown in Fig. 1.

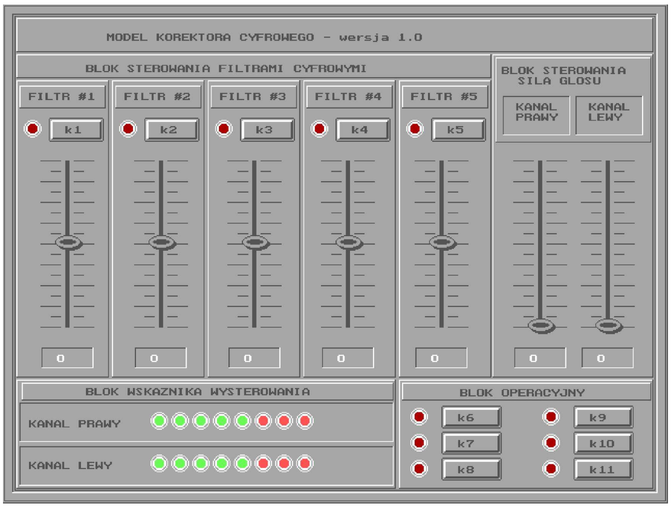

Fig. 1. Front panel of the digital graphic equalizer.

The designed digital graphic equalizer enables the user to freely shape the structure of the sound, by amplification or attenuation of individual subranges of the frequency and at the same time, emphasizing or eliminating particular groups of instruments or performers.

Plugging the devices into the acoustic channel allows to eliminate diversions from ideal frequency characteristics for such equipment as tape recorder or gramophone. It also allows to eliminate diversions of frequency characteristics, introduced into acoustic channel by column loudspeakers.

By designing the digital graphic equalizer we can very quickly find out what the shape of the output signal is. The shape of the amplitude-frequency characteristics is indicated by potentiometer sliders located on the front panel of the graphic equalizer. The great advantage of the digital equalizer is the fact that with little effort you can change the parameters of individual filters.

4. Phase equalizer

In order to speak of sound control in the enclosed space, it is not enough to narrow down to creation of the structure of sound on the amplitude-frequency level provided e.g. by graphic equalizers. Sound control involves also shaping of the sound on the amplitude-phase level, with the systems allowing such modification to be called phase equalizers *DPE* [2].

Digital phase equalizer just like the above mentioned graphic equalizer, has been implemented on card OROS AU22 with signal processor TMS320C25.

The phase equalizer incorporates all-band filters. These filters enable adjustment of signal phase shift without any effect on the change of the signal amplitude. This means that the amplitude of transfer function is equal to 1. Phase control of the processed signal is effected by changing the all-band filter coefficient β. The structure of the all-band filter of the first order is described by relation (3).

$$H_{\mathrm{allpass}} = \frac{\beta + z^{-1}}{1 + \beta z^{-1}}. \tag{3}$$

In order to evaluate the performance of the digital phase equalizer it has been used for active sound reduction. The experiments were performed in the dead room of the Department of Mechanics and Vibroacoustics at the AGH University of Science and Technology in Kraków.

The system was tested for two settings – symmetrical and asymmetrical secondary source, against the test microphone. For the analysis, monoharmonic signals with mid frequencies of the third order from 50 to 1000 Hz were used.

The test results are presented in a graphical form in Fig. 2.

By using the phase equalizer, reduction of the acoustic pressure level was achieved accordingly: for the symmetrical system from 9.1 to 35 dB, and for the asymmetrical system from 6.4 to 36.3 dB.

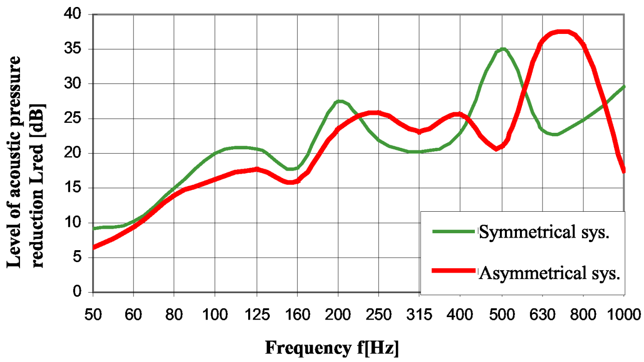

Fig. 2. Level of acoustic pressure reduction.

The designed and built equipment can be used not only as an active sound reduction system but also as a delay line or reverberation unit – i.e., digital reverberator.

5. Inverse equalizer

Inverse filtration (unbraiding) theory has been known for many years, however its practical application has become possible along with the development of digital signal processing (DSP), with commonly available fast signal processors, which enable real-time braiding and unbraiding of signals.

Based on this theory, a digital inverse equalizer *DIE* has been designed and built.

In order to design an inverse equalizer we must find inverse transmittance of the digital filter described by transmittance $H_{\text{inv}}(z)$, in such a way, that relation (4) is fulfilled.

$$H(z) \cdot H_{\text{inv}}(z) = 1. \tag{4}$$

In the case of minimal-phase systems this does not pose major problems. However electro-acoustic systems are very rarely described by minimal-phase systems, in most cases these are maximal-phase or mixed-phase systems.

If the system is physically executed, then the transmittances of the mixed-phase system can be formulated as a product of two transmittances $H_{\text{eq}}(z)$ (5) and $H_{\text{ap}}(z)$ (6).

$$H_{\text{eq}}(z) = \prod_{i=1}^{k}(1 - a_i z^{-1}) \prod_{i=k+1}^{N}(z^{-1} - a_i), \tag{5}$$

$$H_{\text{ap}}(z) = \frac{\displaystyle\prod_{i=k+1}^{N}(1 - a_i z^{-1})}{\displaystyle\prod_{i=k+1}^{N}(z^{-1} - a_i)}. \tag{6}$$

Transmittance (7) describes electro-acoustic system after equalization, using digital inverse equalizer.

$$H_{i\_\mathrm{mix}}(z) = H_{\mathrm{eq}}(z) H_{\mathrm{ap}} \frac{1}{H_{\mathrm{eq}}(z)} = H_{\mathrm{ap}}(z). \tag{7}$$

Performance tests of digital inverse equalizer were carried out in the free field – DMaV dead room. For experiments loud speaker GDN16/30 made by TONSIL was used. For design of the digital inverse equalizer, FIR filters of 84 order and all-band IIR filters of 18 order were used.

Digital inverse equalizer has been implemented on the dSPACE card with signal floating-point processor TMS320C31 made by Texas Instruments.

Figure 3 shows amplitude-frequency and amplitude-phase characteristics of the system before and after equalization by digital inverse equalizer.

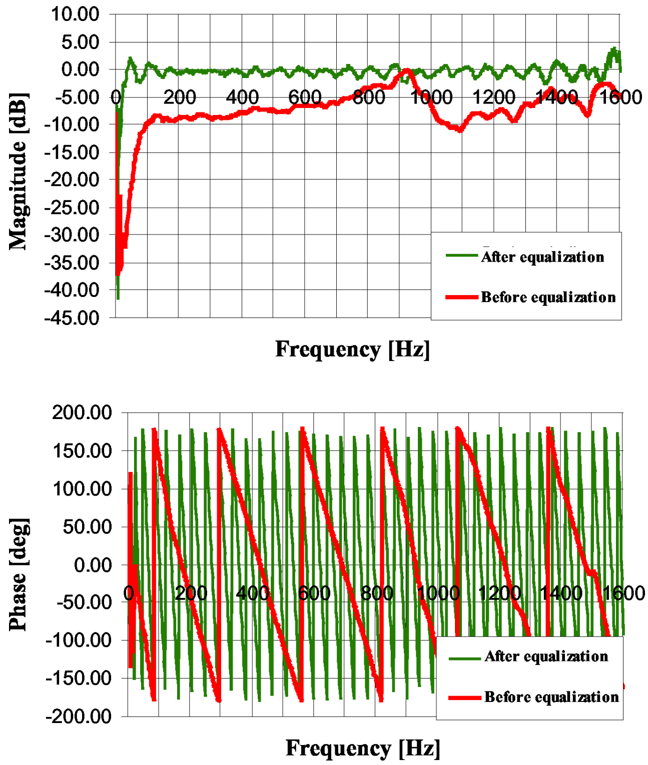

Fig. 3. Amplitude-frequency and amplitude-phase characteristics.

The digital inverse equalizer described above can be used for modification of acoustic climate of a given room or compensation of characteristics of electro-acoustic converters: loudspeakers. The system can be also used as an active noise reduction system, for removing reverberation from acoustic signals or for equalization.

6. Adaptive equalizer

Digital Adaptive Equalizer *DAE* [3, 6–8] enables active noise reduction in the enclosed space. The essence of the built system is the filter of finite pulse response executing adaptive algorithm LMS or NLMS. These algorithms have been implemented on a German card by dSPACE.

Tests aimed at verification of proper operation of the system were performed in the dead room at the Mechanics and Vibroacoustics Department and in the selected room with the dimensions $4.4 \times 3.05 \times 3.2$ [m] and reverberation time $T = 0.53$ [s].

For experiments the white noise was used, which was filtered by octave and third-order filter with a mid frequency of 125 Hz and white noise filtered by a low-pass filter with a cut-off frequency of 355 Hz, and then high-pass filter with a cut-off frequency of 63 Hz.

As a result of the experiments performed for the free field acoustic medium, pressure level was reduced down to 26.9 dB for the noise filtered by the third-order filter, 24.0 dB for noise filtered by octave filter and 11.5 dB filtered in the band from 63 to 355 Hz.

The experiment performed in the room, using random signals – white noise allowed to reduce the level of acoustic medium pressure down to 19.7 dB for the third order and to 14.2 for filter octave of 125 Hz. Reduction of the acoustic pressure level was measured in the location where the error microphone was placed.

Figure 4 shows the sound reduction process – white noise filtered by third-order filter 125 Hz – recorded by the error microphone in the test room, before and after turning on the active noise reduction system.

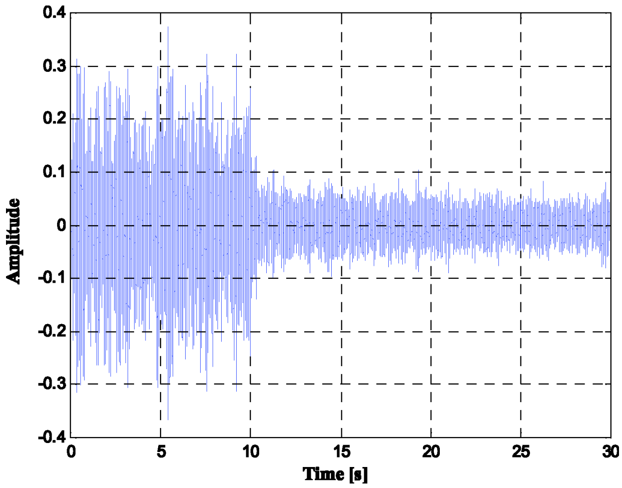

Fig. 4. Noise reduction process in the room recorded using the error microphone.

Tests carried out in order to set the SILENT ZONE, Figs. 5 and 6, performed for the random signal – white noise, allowed, thanks to application of the adaptive system, to reduce the level of acoustic medium pressure from 6.3 to 13.2 dB with a radius of

50 [mm] within a range of 50 to 250 Hz for the signal filtered by octave filter 125 Hz and from 6.0 to 17.7 dB for the signal filtered by the third-order filter 125 Hz.

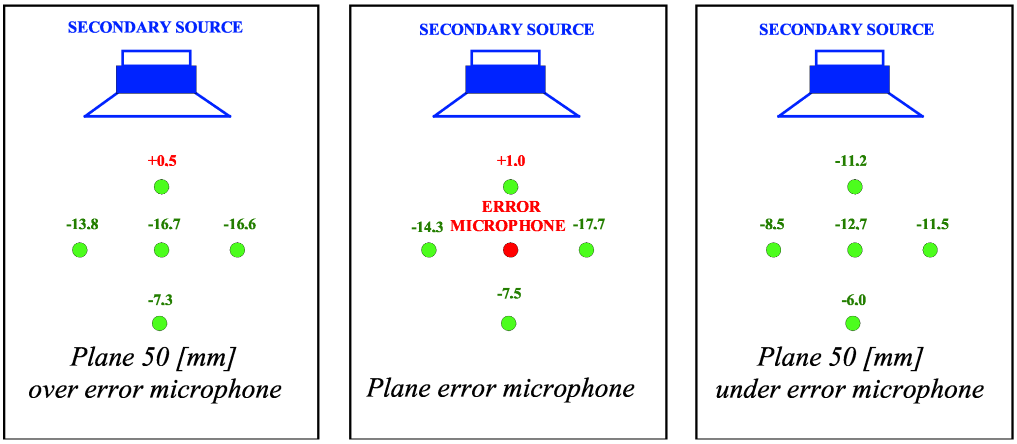

Fig. 5. The silent zone obtained for the white noise filtered by the third-order filter 125 Hz.

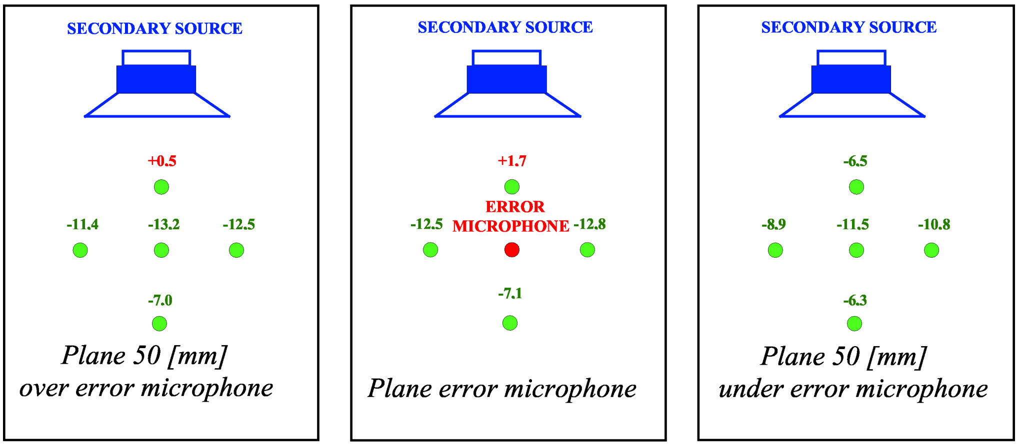

Fig. 6. The silent zone obtained for the white noise filtered by the octave filter 125 Hz.

7. Conclusions

The paper describes single-channel digital systems, enabling active sound control according to the set criterion. The criteria have been divided into two groups: objective and subjective criteria.

The theoretical fundamentals of the designed and built ASC systems are based on the digital filter theory, based on which digital graphic, phase, inverse and adaptive equalizers were designed and programmed.

The described systems have been implemented on two cards with fixed-point signal processor TMS320C25 and floating-point signal processor TMS320C31 by the Texas Instruments. The article presents mathematical models of the above mentioned equalizers and results of the experimental tests.

The designed and built systems ensure good sound control performance and may be used as independent systems for modification of the acoustic climate in the limited space or constitute an element of multichannel systems of active sound control.

Acknowledgment

The paper was presented at the Conference "Structure-Waves-Human Health 2005" hold in Zakopane.

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