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AN ADAPTIVE, ACTIVE NOISE REDUCTION SYSTEM IN CLOSED SPACE

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In this article, an active noise reduction system has been described. The ANR system was made on the basis of an finite impulse response filter and realised by LMS or NLMS algorithms. Those algorithms were implemented on the dSPACE card with a floating-point processor TMS320C31. The researches were performed in an enclosure of dimensions 4.4, 3.05, 3.2 m. White noise filtered by third and octave filters with mid-band frequency 125 Hz were used for the experiments.

The result of experiments was a reduction of the sound pressure level, meanly from 10 to 20 dB for the natural sound field. The investigations performed revealed that the location of reference and error microphones in the acoustic system – in the room, influence significantly the efficiency of the ANR system working. The investigations performed in a selected room (volume $V = 41.6$ m^3 and reverberation time $T = 0.53$ s) revealed that projected and built up system gives very good effects in the active noise reduction for random, white noises of selected ranges.

Key words: active sound control, adaptive algorithm, active noise reduction, digital equalizer.

1. Introduction

The origin of active methods is given in Huygens's works. It was based on the well-known Huygens–Fresnel principle [4]. In 1933 P. LUEG submitted for patenting a project entitled "Verfahren zur Dampfung von Schallschwingungen" in Germany [15]. A lot of scientists admit this fact as beginning of the active noise control methods. H. OLSON [18] suggested the conception and application of an electronic sound absorber in the 1950s. It was the next step in the evolution of active noise control methods.

Polish scientist also took part in the development of active methods. M. Jessel and G.D. Maluziniec appreciated the theoretical works of W. Rubinowicz. The works of W. CZARNECKI [3] and his co-worker M. VOGT made for the determination general conditions of phase compensation in the sound field as well [23]. Nowadays Z. Engel is esteemed as beginner and originator of active methods in Poland. He organised the Polish School of Active Methods of Noise and Vibration Reduction together with J. Nizioł. First prototypes of systems for noise and vibration reduction were made under his direction.

Nowadays at the AGH University of Science and Technology and the Central Institute for Labour Protection are carried out investigations which aim at the constructing of an active noise control system. Similar investigations were made at the Cracow University of Technology and the Silesian University of Technology. First prototypes of noise control systems in ducts, active mufflers and personal hearing protectors were construted, as well as active noise control systems in closed space. One of the ANR systems in closed space is the subject of this article.

2. Short characteristic of active noise reduction systems

Active noise reduction systems are divided into two groups: the analog and digital ones. The analog systems are made most often as ANR feedback systems, whereas the digital ones usually are feed forward systems. Most of scientists and researchers are recently interested in the latter group of systems. The reason is the development of technology, especially electronics, computer science, automatics and the theory of control engineering.

The present digital ANR systems aresignal processors equipped with accessory systems, an analog-to-digital converter AC and a digital-to-analog converter DA. Those systems make possible the active sound control by an accepted criterion. As the correctness criterion of the system operation, for active noise reduction systems, was admitted to minimize the acoustic pressure at the point at which the error microphone (or microphones) was located.

Finite Impulse Response (FIR) filters or Infinite Impulse Response (IIR) filters and their modifications, e.g. a lattice filter structure, are used for creating active noise reduction systems. The latest investigations are aimed at the applying of neural networks [22] for the improvement of the effectiveness of ANR systems.

Besides digital filters and neural networks, an adaptive noise reduction system is made of an adaptive algorithm, making it possible to weight the correction in real time for a selected structure. The algorithms for adaptive noise reduction systems most often used are: the LMS, FLMS, RLS, FRLS, NLMS, SHARF or FXLMS algorithms.

Adaptive noise reduction systems can be classified with respect to the mode of receiving, processing and emitting signals i.e. single-channel and multi-channel systems, as well with respect to the mode of sound control: in the time or in the frequency domain.

The adaptive noise reduction system presented in this work was made on the basis of a finite impulse response filter and realised the algorithms LMS or NLMS.

3. Adaptive algorithms LMS and NLMS

Figure 1 shows the structure of an adaptive control system. The system consists of two basic elements: the finite impulse response filter (FIR) of $N-1$ order and the system that implements the adaptive algorithms LMS or NLMS. In the following figure the transfer function $P(z)$ describes unknown plant-acoustic field.

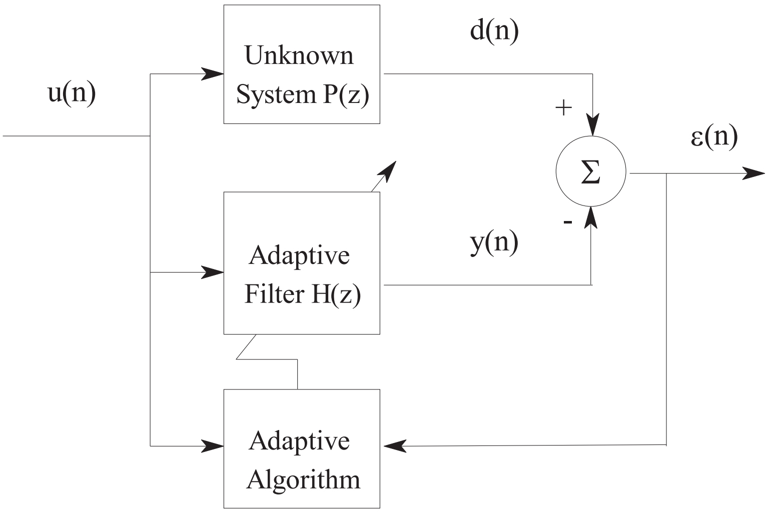

Fig. 1. Block diagram of the adaptive control system.

Equation (1) describes the output filter, which is applied as the estimate reference signal. As the result of comparison of the estimate $y(n)$ to the reference signal $d(n)$ one obtains the estimate error (4).

$$y(n) = \sum_{k=1}^{N} \widehat{h}_k(n)u(n-k+1) = u^{\mathrm{T}}(n)\widehat{h}(n), \tag{1}$$

where

$$\widehat{h}(n) = [\widehat{h}_1(n), ..., \widehat{h}_N(n)]^{\mathrm{T}} - \text{weight vector}, \tag{2}$$

$$u(n) = [u(n), u(n-1), ..., u(n-N-1)]^{\mathrm{T}} - \text{input vector at time } n, \tag{3}$$

$$\varepsilon(n) = d(n) - y(n). \tag{4}$$

Equation (5) describes updating adaptive filter weights making use of the algorithm LMS. The vector $h(n+1)$ contains successive weights computed in real time of the active noise control system.

$$\widehat{h}(n+1) = \widehat{h}(n) + \mu u(n)\varepsilon(n), \tag{5}$$

where μ – adaptation step, $0 < \mu < 1$.

The LMS algorithm presented above is a characteristic at constant adaptation step μ. The slower convergence of this algorithm may be a consequence of this fact. Therefore, the algorithm based on a time-varying step size, given by Eq. (6), was implemented on a card with signal processor. The dependence (7) expresses the adaptation coefficient $\mu(n)$ for the NLMS algorithm.

$$\widehat{h}(n+1) = \widehat{h}(n) + \mu(n)u(n)\varepsilon(n), \tag{6}$$

$$\mu(n) = \frac{\alpha}{u^{\mathrm{T}}(n)u(n)}, \tag{7}$$

where α – positive constant; $0 < \alpha < 2$.

Preliminary research of the ANR system based on the NLMS algorithm shows the considerable impact of the input signal on the working algorithm. Therefore, there is an additional coefficient β in the denominator of equitation (8). This coefficient enables the correct working of the algorithm and the whole system, when the tap-input vector value $u(n)$ is small.

$$\mu(n) = \frac{\alpha}{\beta + u^{\mathrm{T}}(n)u(n)}, \tag{8}$$

$$\widehat{h}(n+1) = \widehat{h}(n) + \frac{\alpha}{\beta + u^{\mathrm{T}}(n)u(n)} u(n)\varepsilon(n), \tag{9}$$

where $0 < \alpha < 2$, $\beta > 0$.

Finally, the active noise control system was built at the base of the constant step size LMS algorithm using the (1), (4) and (5) dependencies. Afterwards, on the card with the signal processor, the program based at the time-varying step size NLMS algorithm has been made. The dependence (9) was used instead of (5) during computations.

4. Structure of the active noise reduction system

The prototype of the active noise reduction system is built on a controlled board with a signal processor equipped with an analog-to-digital converter, a digital-to-analog converter and accessory elements requisite for the regular system working. The block diagram of this system is presented in Fig. 2.

The designed system consists of four basic lines. The sending line contains a digital tape recorder, acoustic mixer, power amplifier, and a primary source (1). The second line analyses and identifies one of the contained reference (2) and error (5) microphones, the voltage amplifiers (3, 6), filters (4, 7) and the adaptive system (8). There is the secondary source (11); the power amplifier (10), filter (9) and the adaptive system (8) are in the control line. The last line is a test one, which contains the microphone (12) and analyser B&K2034 (13).

In order to achieve a regular work of the active noise reduction system, adequate conditions should be performed. These conditions depend on the electric and acoustic parameters of several system channels.

The electric parameters are the frequency and phase response as well as the transient response of several elements of the electro-acoustic channels. Nowadays, thanks to the dynamic development of electronics and signal processing, one can select these parameters at discretion.

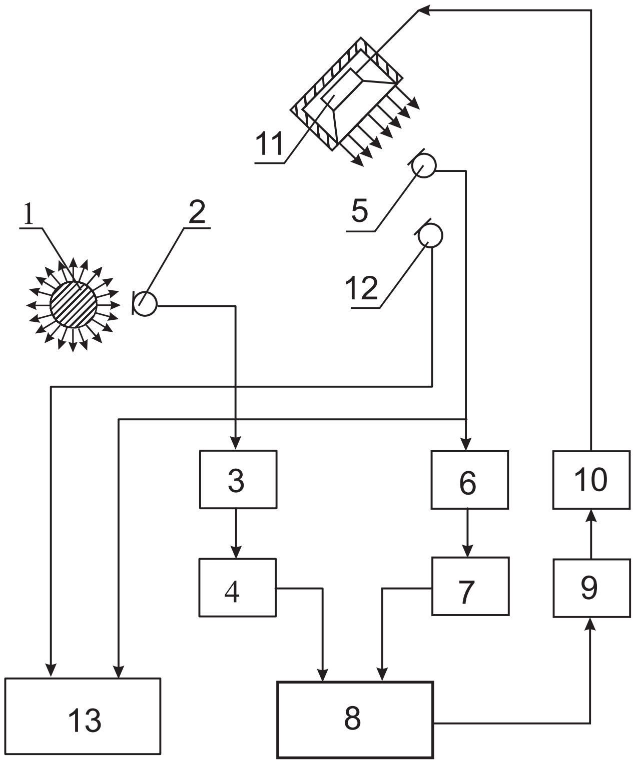

Fig. 2. Block diagram of the active noise reduction system.

Acoustic parameters include the modal density and modal dumping of the enclosure. They follow directly from the physical properties of the plant-acoustic field. They will be described in a further part of this article.

5. Influence of the room geometry on operation of the ANR system – simulation researches

The effectivness of the ANR system operation does not depend only on the electric parameters of the control system. It also depends on the sound field, which is under control. In general, one can discern two strategies of sound control in enclosed space: the global and local controls. This subdivision follows from the dimension of the sound field in which the sound control is carried on.

The dependence (10) terms the minimal frequency, under which one can acquire the global sound reduction in closed space [12].

$$f_g = 700\sqrt{\frac{T}{V}}, \tag{10}$$

where f_g – frequency of the global sound reduction [Hz], T – reverberation time [s], V – volume [m^3].

When the frequency is greater than the value resulting from dependence (10), one can carry out local sound control and acquire local reduction of the sound pressure level – zones of quiet. In this case, the dimensions and shape of the room, the element location inside the room, but not the volume are dominant elements, which influence the acoustic field distribution.

One can reveal, applying the wave theory of the sound field analysis, that the above mentioned elements influence the acoustic field distribution in the room, and thereby the efficiency of the active noise control system.

Corresponding to the wave theory, the room may be regarded as complex of acoustic resonators. The resonance frequency of each resonator is definite dependent on the excitation sound frequency.

The vibrations of air in the room are excited at adequate frequencies dependent on the excitation sound frequency. Proper vibrations are observed as standing waves, which may be excited in the acoustic system. They are axial, tangential and oblique.

The eigen frequencies f_r [Hz] of a rectangular room may be expressed by the equation:

$$f_r = \frac{c}{2}\sqrt{\left(\frac{k}{l}\right)^2 + \left(\frac{m}{b}\right)^2 + \left(\frac{n}{h}\right)^2}, \tag{11}$$

where c – speed of sound $[\mathrm{ms}^{-1}]$, l, b, h – length, width and height of the closed space [m], j, m, n – natural numbers 0, 1, 2, 3, ...

Numerical methods were used to analyse the sound field distribution in the case when the room is non-rectangular. The Finite Element Method (FEM) was used for the calculation of eigen frequencies. SYSNOISE is one of the simulation programs basing on FEM. It was used for digital simulation.

Figure 3 shows a model of room used for researches and simulations. Its dimensions were: $4.2 \times 3.05 \times 3.2$ m. The room is rectangular with a niche ($0.4 \times 1.05 \times 3.2$ m). There were floor finish and metal panels on the ceiling of the room. The walls were painted. There were a cabinet, bookcases and other elements of furnishing. The room show in Fig. 3 was digitized using HEXA8 elements. The digital simulation was made for this prepared model and 100 of the first eigen frequencies were set down. The first twenty eigen frequencies are presented below. Figures 4 and 5 show the distribution of the sound pressure at the mode frequency of 79.582 Hz and 123.552 Hz.

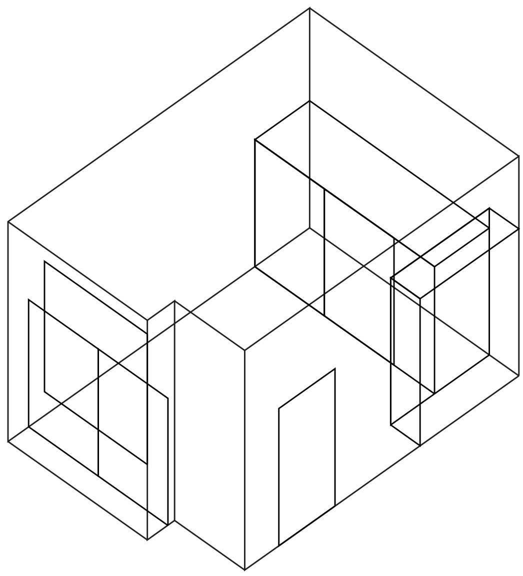

Fig. 3. Model of the room used for researches and simulations.

EIGENVALUES SHIFTED AND RESCALED
EIGENVECTORS NORMALIZED TO MASS MATRIX
SAVING ACOUSTIC MODES INTO DATABASE
CPU Stats — Delta: 2:50:26 (10226.00) Total: 2:50:35 (10235.00)

ACOUSTIC EIGEN FREQUENCIES	
MODE 1 FREQUENCY (Hz) 1.09749227E-06	PHASE .00
MODE 2 FREQUENCY (Hz) 42.953534	PHASE .00
MODE 3 FREQUENCY (Hz) 53.614737	PHASE .00
MODE 4 FREQUENCY (Hz) 57.439463	PHASE .00
MODE 5 FREQUENCY (Hz) 67.603003	PHASE .00
MODE 6 FREQUENCY (Hz) 72.092389	PHASE .00
MODE 7 FREQUENCY (Hz) 79.582275	PHASE .00
MODE 8 FREQUENCY (Hz) 89.043859	PHASE .00
MODE 9 FREQUENCY (Hz) 90.532540	PHASE .00
MODE 10 FREQUENCY (Hz) 106.68133	PHASE .00
MODE 11 FREQUENCY (Hz) 108.22062	PHASE .00
MODE 12 FREQUENCY (Hz) 109.01618	PHASE .00
MODE 13 FREQUENCY (Hz) 113.70396	PHASE .00
MODE 14 FREQUENCY (Hz) 119.03557	PHASE .00
MODE 15 FREQUENCY (Hz) 121.26073	PHASE .00
MODE 16 FREQUENCY (Hz) 122.71750	PHASE .00
MODE 17 FREQUENCY (Hz) 123.55159	PHASE .00
MODE 18 FREQUENCY (Hz) 127.80047	PHASE .00
MODE 19 FREQUENCY (Hz) 130.26392	PHASE .00
MODE 20 FREQUENCY (Hz) 133.12482	PHASE .00

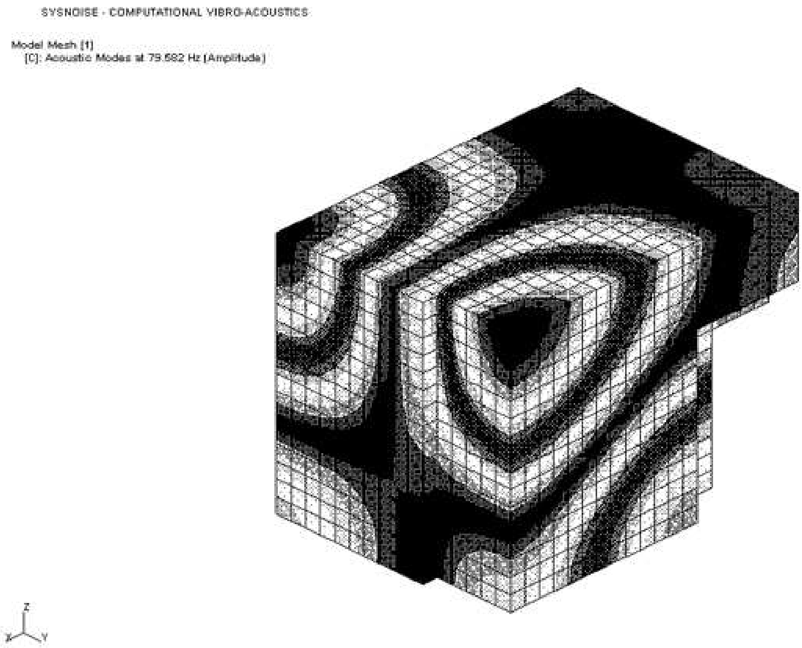

Fig. 4. Graphical representation of the sound pressure distribution at the mode frequency 79.582 Hz.

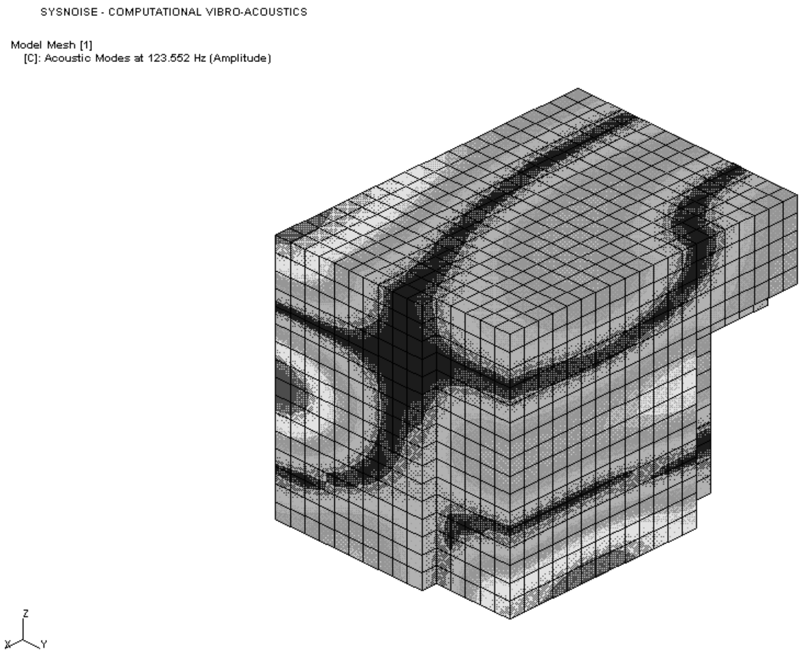

Fig. 5. Graphical representation of the sound pressure distribution at mode frequency 123.552 Hz.

The modal frequency analyse of the room and the knowledge of materials applied for adaptation showed a complex character of the acoustic field distribution. Therefore, in order to estimate the influence the acoustic field distribution on the efficiency of the ANR system, beside simulations, the coherence function was used. This function is termed by the Eq. (12):

$$C_{du}(\varpi) = |\gamma_{du}(\varpi)|^2 = \frac{|S_{du}(\varpi)|^2}{S_{dd}(\varpi)\, S_{uu}(\varpi)}. \tag{12}$$

The coherence assumes a value according to $0 \le |\gamma_{du}(\varpi)|^2 \le 1$. The effectiveness of the active noise reduction system for a single-channel was set up in [dB] by Eq. (13).

$$L_{\text{red}} = -10 \log_{10} [1 - C_{du}(\varpi)]. \tag{13}$$

6. Laboratory researches of ANR systems

In order to verify the operation correctness, the built up active control system was tested in an anechoic chamber and in reverberate three-dimensional space at the Department of Mechanics and Vibroacoustics – AGH University of Science and Technology.

The carried out experiments were divided on three stages. The first of them included experiments carried out in a free field [7]. In the next stage of the researches, a sound

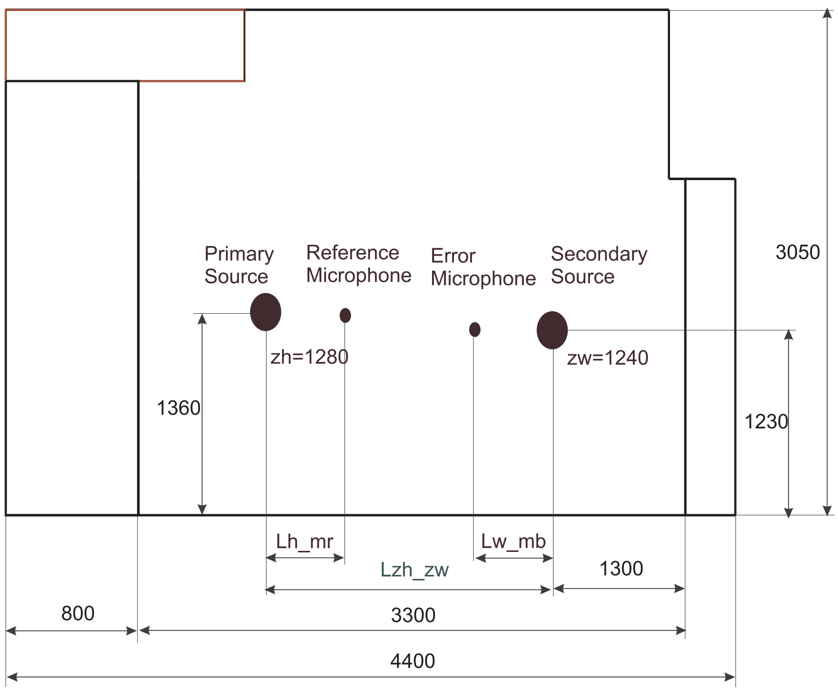

Fig. 6. Location of microphones and sources in the room.

reflection surface was used and the measurement system was located upon it [8, 9]. The last stage was conducted in a room of a volume of 41.6 m^3, and a reverberation time of 0.53 s. White noise filtered by third and octave filters with a mid-band frequency of 125 Hz were used for the experiments.

The carried out researches in closed space [10] were divided into two stages. The first one was regarded with the carried out measurements of the sound pressure level reduction and the coherence function. In the second stage, the silent zone arising around the error microphone was measured.

The experiments were performed for various locations of the source and microphones. Figure 6 presents the location of microphones and sources in the room. All dimensions are in [mm]. The system presented is characterised by a location asymmetry of the primary and secondary sources, the reference and error microphones.

The results for ANR system researches are shown in Figs. 7, 8 and 9. In Figs. 8 and 9, the dashed curve presents the sound pressure level spectrum at turned on the adaptive system.

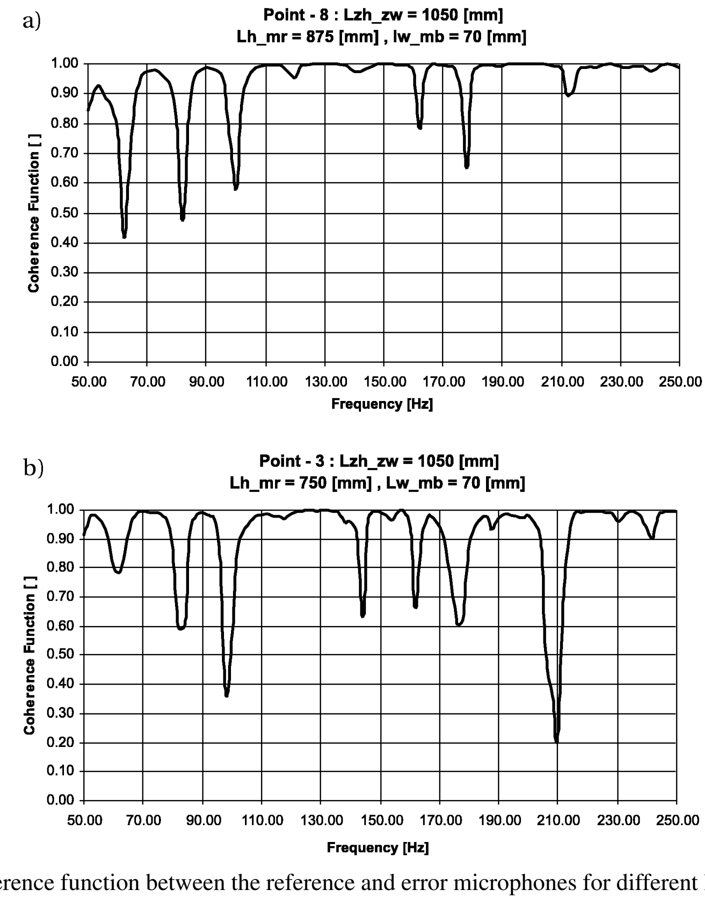

Fig. 7. Coherence function between the reference and error microphones for different locations of sensors in the room.

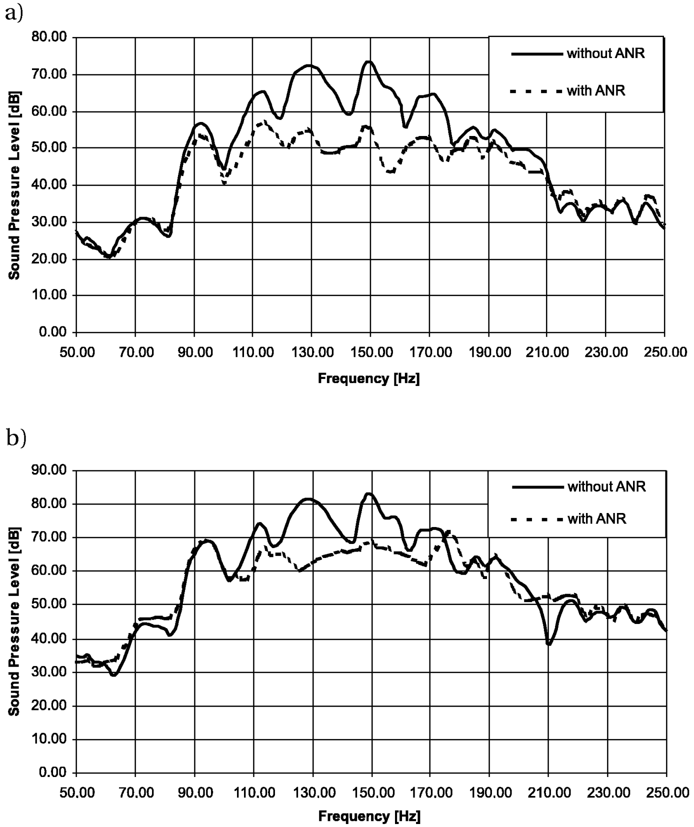

Fig. 8. Reduction of the sound pressure level for white noise filtered by an octave filter $f_0 - 125$ Hz for different locations of sensors in the room: a) average reduction of the sound pressure level within the range 50–250 Hz – 13.8 dB; b) average reduction of the sound pressure level within the range 50–250 Hz – 9.4 dB.

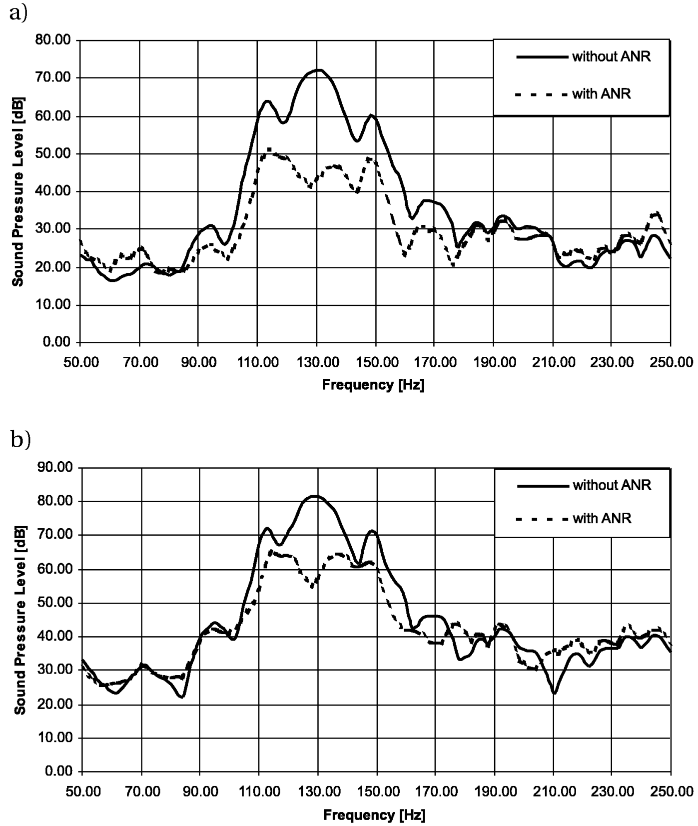

Fig. 9. Reduction of the sound pressure level for white noise filtered by the third filter $f_0 - 125$ Hz for different locations of sensors in the room: a) average reduction of the sound pressure level within the range 50–250 Hz – 19.7 dB, b) average reduction of the sound pressure level within the range 50–250 Hz – 14.0 dB.

Figure 10 shows "silent zones" obtained in the room as the result of the active noise reduction proceeding. The negative values presented define the reduction of the average sound pressure level at points of the control-measurement microphone locations. Distances between the points were 50 [mm].

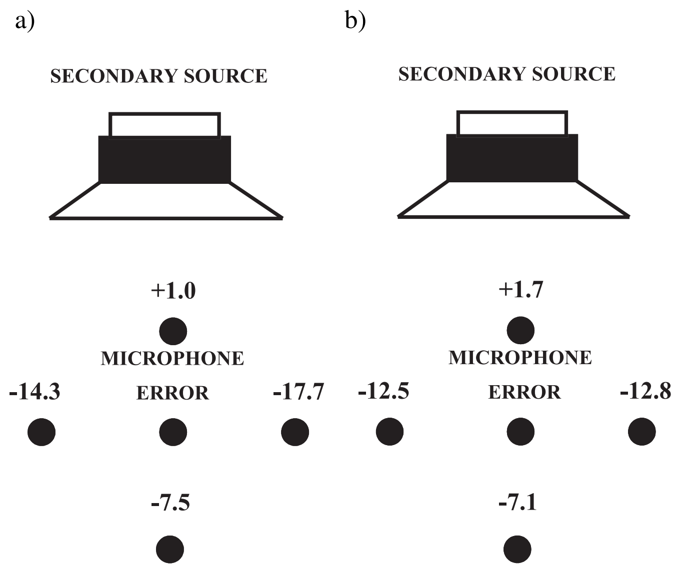

Fig. 8. Silence area created by the ANR systems: a) white noise filtered by the third filter $f_0 - 125$ Hz, b) white noise filtered by the octave filter $f_0 - 125$ Hz.

7. Conclusions

An active noise reduction system has been described in this article. The ANR system was made on the basis of a finite impulse response filter and the realised algorithms LMS or NLMS. Those algorithms were implemented in the dSPACE card with the floating-point processor TMS320C31.

The researches were performed in an enclosure of dimensions $4.4 \times 3.05 \times 3.2$ m. White noise filtered by the third and octave filters with mid-band frequency 125 Hz were used for the experiments.

The result of the experiments was a reduction of the sound pressure level, meanly from 10 to 20 dB for the natural sound field.

The investigations performed revealed that the location of the reference and error microphones in the acoustic system – in the room, influence significantly the efficiency of the ANR system working.

The aim of carried out investigation was to set down a "silent zone" for the random signal – white noise. In these investigations, an adaptive system was used. It caused an average reduction of the sound pressure level on a radius of 50 mm within the range 50–

250 Hz from 6.3 to 12.8 dB for the white noise filtered by the octave filter f_0 – 125 Hz and from 6.0 to 17.7 dB for the white noise filtered by the third filter f_0 – 125 Hz.

The results of the experiments proved that the convergence time of the NLMS algorithm was several times shorter than that of the LMS algorithm.

The investigations performed in a selected room (volume $V = 41.6$ m^3 and reverberation time $T = 0.53$ s) revealed that the projected and build up system gives very good effects in the active noise reduction for random, white noises of selected ranges.

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