ARCHIVES OF ACOUSTICS
32, 2, 205–213 (2007)

ACTIVE NOISE REDUCTION SYSTEM

Wojciech CIESIELKA

AGH University of Science and Technology
Department of Mechanics and Vibroacoustics
Al. Mickiewicza 30, 30-059 Kraków, Poland
e-mail: ghciesie@cyf-kr.edu.pl

(*received April 11, 2007; accepted April 24, 2007*)

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

Researches were performed in the anechoic chamber and in the enclosure of dimension $4.4 \times 3.05 \times 3.2$ m and reverberation time $T = 0.53$ s. White noise filtered by third and octave filter with mid-band frequency 125 Hz was used for the experiments.

The ANR system working in free field conditions (in the anechoic chamber) allowed to obtain the average acoustic pressure level reduction ranging from 9.1 to 24.1 dB for octave, and 14.7 to 23.1 dB for third octave.

Measurements carried out in natural acoustics conditions in a selected room allowed to obtain the following values of average acoustic pressure level reduction: for octave from 2.2 to 14.2 dB, and for third octave from 6.4 to 19.8 dB.

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

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

1. Introduction

Active sound reduction systems play very important role among numerous sound control systems. They use the effects of interference or compensation of acoustic field generated by secondary sources with noise emitted by the primary source [3, 7, 9].

We may divide compensation effect into two major parts: natural and forced compensation [10]. Certain technical and physical conditions must be satisfied to obtain phase compensation effect. Technical conditions require necessary analysing-control-measuring equipment.

The equipment should consist of the following devices: card with signal processor allowing to process signals in real time, power amplifiers, filters and electro-acoustic transducers. Next step in building a system for active sound compensation is to select an algorithm carrying out adaptive filtration and proper digital filter structure [4, 8].

Owing to rapid digital processing development and high elasticity, adaptive systems are more and more often used in many fields of science and technology. The following applications of these systems deserve attention as well: their use to identify and control dynamic objects; to suppress echo generated during telecommunication connections (calls) especially at longer distances, or active noise reduction [5, 6].

This paper presents the system of active sound reduction in limited space, using a filter with finite impulse response and the following algorithms: least mean square (LMS) and normalised NLMS for filter weights correction [1, 2].

2. Active noise reduction system

Active Noise Reduction System has been built based on dSPACE card equipped with the TMS320C31 signal processor from Texas Instruments. The system consists of two main subsystems: digital filter with finite impulse response (FIR) and adaptive algorithm allowing to correct filter weights using adaptive methods. Two algorithms (LMS and NLMS) have been implemented in the signal processor card.

Besides signal processor card equipped with analog-to-digital and digital-to-analog converters, the prototype ANR active noise reduction system contains additional elements necessary for proper system functioning. These elements form four electro-acoustic lines specified below. These are:

- ***transmitting line:*** tape recorder, power amplifier and loudspeaker – primary source (1);
- ***analysing-identifying line:*** reference (2) and error (5) microphone, microphone pre-amplifiers, measuring amplifiers (3, 6), antialiasing filters (4, 7), and adaptive system (8);

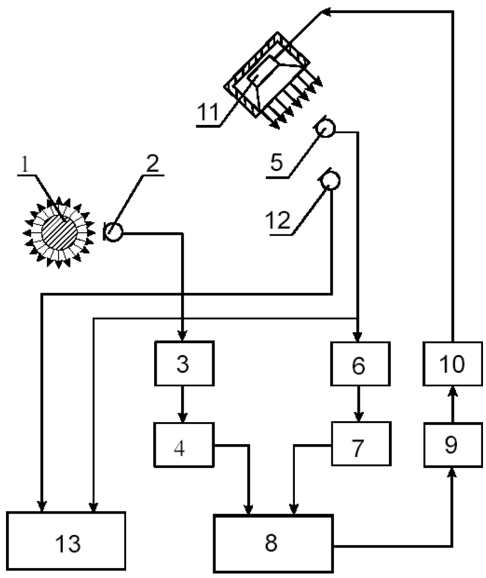

Fig. 1. Block diagram of an active noise reduction system.

- ***control line:*** adaptive system (8), reconstruction filter (9), power amplifier (10) and loudspeaker (11);
- ***measuring line:*** measuring microphone (12), microphone preamplifier and the B&K 2034 two-channel analyser (13).

Figure 1 shows block diagram of the ANR system. Two main lines are distinguishable in the system structure: electrical and acoustic one.

3. The ANR system laboratory tests in free field conditions

Measuring station was prepared in an anechoic chamber at the Department of Mechanics and Vibroacoustics, and the whole system was tested in order to verify correct operation of the designed system.

The first stage involved identification of electro-acoustic line elements and determination of directional characteristics for the GDN16/30 loudspeakers.

Random signals were used in tests – white noise filtered by third octave filter and octave-band filter with mid-band frequency of 125 Hz.

The experiments were carried out for various arrangements of sources and microphones. Figure 2 presents view of the active noise reduction system set up in free field conditions, that is in the anechoic chamber.

Fig. 2. Active noise reduction system in the anechoic chamber.

Acoustic pressure minimisation in the location of so-called error microphone was taken as the criterion of the designed system correct operation.

$$J = \sum_{n=1}^{D_{\min}} e^2(nT), \tag{1}$$

where T – sampling time, $D_{\min}$ – number of samples.

Figure 3 shows measuring system employed in tests carried out in free field conditions and in selected room. It is characteristic of complete asymmetry regarding layout of primary and secondary sources, as well as reference and error microphones.

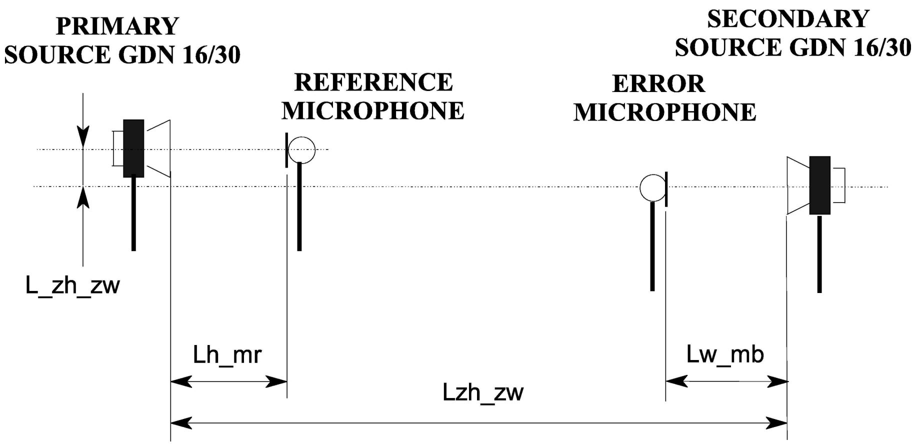

Fig. 3. Measuring system employed in tests of active noise reduction system in free field conditions.

The following parameters values were assumed during the tests: L_zh_zw = 130 [mm] and Lzh_zw = 1050 [mm]. Noise source, so-called primary source, and reference microphone were put at the height of 1280 [mm], while secondary source and error microphone – at the height of 1240 [mm]. Other parameters, as Lh_mr and Lw_mb, were altered during the experiments. The values in [mm] are specified in Table 1.

Table 1. Parameters of digital adaptive equalizer and sound reduction results obtained in free field conditions.

		Lh_mr – 0 Lw_mb – 70		Lh_mr – 70 Lw_mb – 70		Lh_mr – 750 Lw_mb – 70		Lh_mr – 850 Lw_mb – 70		Lh_mr – 875 Lw_mb – 70	
	Algorithm	*LMS*	*NLMS*	*LMS*	*NLMS*	*LMS*	*NLMS*	*LMS*	*NLMS*	*LMS*	*NLMS*
Octave 125 Hz	Filter Order N	16	16	16	16	12	12	10	10	8	8
	Adap. Coeff. μ	0.1	0.01	0.1	0.01	0.1	0.01	0.01	0.005	0.005	0.0025
	Lp_red	**23.7**	**24.1**	**14.4**	**14.8**	**10.7**	**11.1**	**11.7**	**11.9**	**9.1**	**9.2**
Third octave 125 Hz	Filter Order N	16	16	16	16	16	16	12	12	12	12
	Adap. Coeff. μ	0.1	0.1	0.1	0.05	0.1	0.05	0.025	0.01	0.0175	0.005
	Lp_red	**23.1**	**23.0**	**19.9**	**19.2**	**17.2**	**17.0**	**16.2**	**16.6**	**14.7**	**14.8**

Moreover, the table shows parameters of digital adaptive equalizer. The following values are specified: adaptive filter order, and adaptation coefficient μ for the LMS algorithm and α for the NLMS algorithm.

Figure 4 present active sound reduction results obtained in the anechoic chamber as a result of using the LMS and NLMS adaptive algorithms. These results were obtained

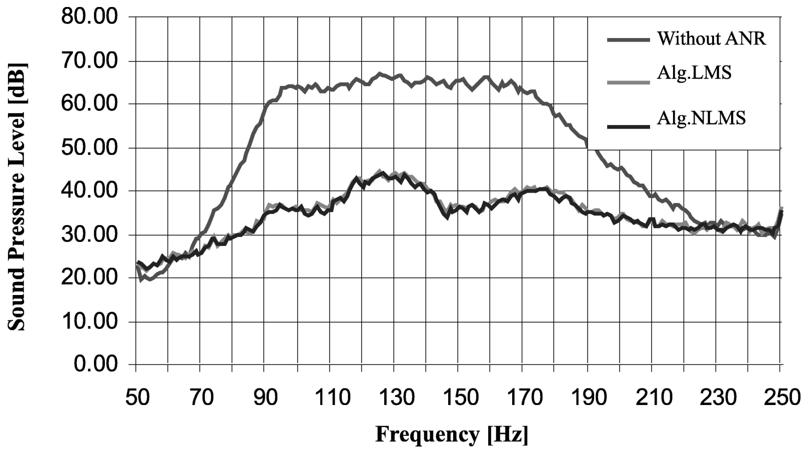

Fig. 4. Active sound reduction – white noise filtered by octave filter with mid-band frequency $f_0 = 125$ Hz.

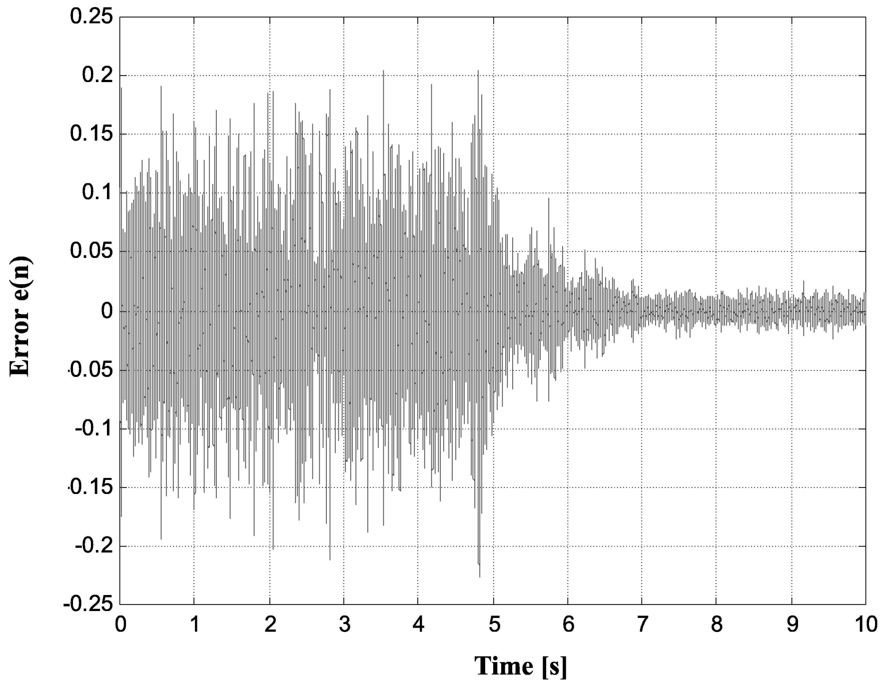

Fig. 5. Sound reduction process in the anechoic chamber, recorded with error microphone using the LMS algorithm – the 125 Hz octave.

for the system with removed reference microphone $\mathrm{Lh\_mr} = 0$. Figure 5 show sound reduction process registered while digital adaptive equalizer was in operation.

4. The ANR system laboratory tests in selected room

The experiments were performed in previously described room with cubic capacity of 41.6 [m^3] and reverberation time – 0.53 [s]. White noise filtered by third octave filter and octave-band filter with mid-band frequency of 125 Hz was used in the tests.

During the experiments, researchers used the same measuring system as in the case of free field – the anechoic chamber.

Same as in free field conditions, electro-acoustic line elements were identified first. The identification involved determination of amplitude-frequency and phase-frequency characteristics and coherence function.

Performed tests were divided into two stages. The first one concerned carrying out of sound reduction measurements. The second stage involved determination of so-called *silence zone*, occurring around the error microphone. The experiments were carried out for various arrangements of sources and microphones.

Measurements carried out in natural acoustics conditions in a selected room allowed to obtain the following values of average acoustic pressure level reduction: for octave from 2.2 to 14.2 dB, and for third octave from 6.4 to 19.8 dB.

Figure 6 present active sound reduction results obtained in a selected room. The following average acoustic pressure level reduction values were measured for the 125 Hz third octave: $\mathrm{Lp\_red_{LMS}} = 19.7$ [dB] and $\mathrm{Lp\_red_{NLMS}} = 19.8$ [dB]. The following values were recorded for octave: $\mathrm{Lp\_red_{LMS}} = 14.1$ [dB] and $\mathrm{Lp\_red_{NLMS}} = 14.2$ [dB].

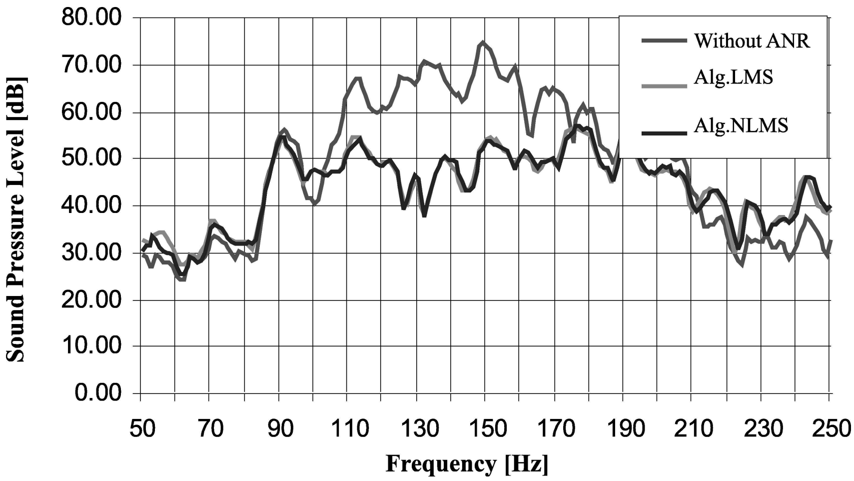

Fig. 6. Active sound reduction – white noise filtered by octave filter with mid-band frequency of $f_0 - 125$ Hz.

Figure 7 show sound reduction process registered while digital adaptive equalizer was in operation, for the case as described above.

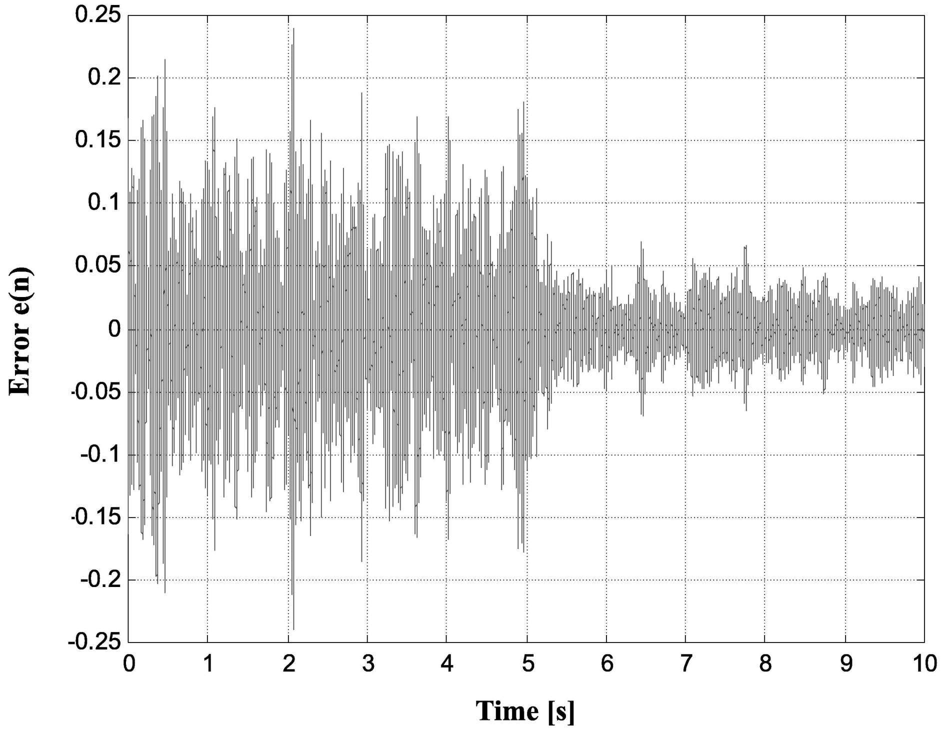

Fig. 7. Sound reduction process in room, recorded with the error microphone when using the NLMS algorithm – the 125 Hz third octave.

The next step involved determination of the "silence zone" formed around the error microphone [1]. Due to the introduced adaptive system, tests aimed to determine the SILENCE ZONE and carried out for random signal – filtered white noise, allowed to reduce average acoustic pressure level by 6.3 to 13.2 dB at 50 mm radius within frequency range from 50 to 250 Hz for signal filtered by 125 Hz octave-band filter, and from 6.0 to 17.7 dB for signal filtered by 125 Hz third octave filter.

5. Conclusions

This article presents noise reduction carried out using an adaptive digital equalizer. The system is based on a filter with finite impulse response, which coefficients are determined in real time using the LMS or NLMS adaptive algorithms. These algorithms were implemented in dSPACE card equipped with the TMS320C31 floating-point signal processor.

Efficient operation of the designed system was tested in free field conditions – in the anechoic chamber and in a selected room sized $4.4 \times 3.05 \times 3.2$ [m].

White noise filtered by octave-band filter and third octave filter with mid-band frequency of 125 Hz was used in the tests.

Average acoustic pressure level reduction was observed and analysed within band range from 50 to 250 Hz in the place, where the so-called error microphone was put.

When determining the so-called *silence zones*, also the third microphone known as measuring microphone was used in the experiments.

Digital adaptive equalizer working in free field conditions (in the anechoic chamber) allowed to obtain the average acoustic pressure level reduction ranging from 9.1 to 24.1 dB for octave, and 14.7 to 23.1 dB for third octave.

Measurements carried out in natural acoustics conditions in a selected room allowed to obtain the following values of average acoustic pressure level reduction: for octave from 2.2 to 14.2 dB, and for third octave from 6.4 to 19.8 dB.

In spite of the fact that two different algorithms were used in digital adaptive equalizer, the researchers did not manage to obtain considerable difference in average acoustic pressure level reduction within selected band. On the other hand, they observed much faster NLMS algorithm adaptation process compared with the LMS algorithm.

Moreover, experimental tests proved that so-called correction step considerably affects the adaptive system operation. In digital adaptive equalizer, correction step values were selected experimentally.

The experiments carried out in anechoic chamber allowed to formulate the following conclusion. An element, which considerably affects active sound reduction system operation, is the arrangement of microphones, and thus time required to generate proper adaptive system response. Besides the distance between individual loudspeakers and microphones, another element affecting sound reduction in room is the distribution of own frequency values and reverberation time value.

Due to the introduced adaptive system, tests aimed to determine the SILENCE ZONE and carried out for random signal – filtered white noise, allowed to reduce average acoustic pressure level by 6.3 to 13.2 dB at 50 mm radius within frequency range from 50 to 250 Hz for signal filtered by 125 Hz octave-band filter, and from 6.0 to 17.7 dB for signal filtered by 125 Hz third octave filter.

Tests performed in anechoic chamber and in selected room with cubic capacity of $V = 41.6$ [m^3] and reverberation time $Tp = 0.53$ [s] proved very high efficiency of the designed and built system as regards active reduction of random-type sound, occurring within specified range in form of white noise.

References

[1] CIESIELKA W., *Application of digital equalizers for sound synthesis*, PhD thesis, AGH University of Science and Technology, Kraków 2002.

[2] CIESIELKA W., GOŁAŚ A., *Układ do aktywnej redukcji hałasu*, Urząd Patentowy Rzeczpospolitej Polskiej przyznał patent P – 323959 dnia 10.08.2004 r, zgłoszenie wykonane do UP RP dnia 19.12.1997.

[3] CZARNECKI S., *Investigation of sound transmission properties of a medium resulting from sound wave compensation caused by other sound sources*, J. Sound Vib., **11,** 225–233 (1970).

[4] ENGEL Z., *Aktywne metody redukcji drgań i hałasu (rys historyczny, zagadnienia, problemy)*, Mat. I Szkoły Metody Aktywne Redukcji Drgań i Hałasu, Kraków–Rabka 1993.

[5] HAYKIN S., *Adaptive filter theory*, Third Edition, Prentice Hall, Upper Saddle River, New Jersey 1996.

[6] KUO M. S., MORGAN R. D., *Active noise control systems*, Algorithms and DSP Implementations, John Wiley & Sons. Inc. New York, NY, 1996.

[7] LUEG P., *Process silencing sound oscillations*, US Patent No 2043416 Appl. 1934, Patented 1936. Priority Germany 1933.

[8] NELSON P. A., ELLIOTT S. J., *Active control of sound*, Academic Press, San Diego 1992.

[9] OLSON H. F., *Electronic sound absorber*, E.G. May, JASA, **25,** 130–136 (1953).

[10] VOGT M., *Ogólne warunki kompensacji fazowej sygnałów w polu akustycznym*, Archiwum Akustyki, **11,** 141–157 (1976).