Simulation Model of the ANC System for Noise Reduction in the Real Ambient

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***Abstract*—The simulation model of ANC system for noise reduction caused by rotating machines in a room was described in the first part of this paper. This simulation model was presented in an acoustic-electrical diagram. The detailed mathematical analysis of the adaptive algorithm was performed. The second part of the paper presents the simulation results of the application of the ANC system for the noise reduction of fans in a room intended for a classroom. Simulation was performed for sine and real aroused signal. The results are presented both numerically and graphically and the comparative analysis was also done.**

***Index Terms*—active noise control (ANC), adaptive algorithm, fan noise, digital signal processing.**

I. Introduction

Control and elimination of an undesirable acoustic signal, noise, is a scientific field that has been undergoing an intense development in the last two decades [1]-[4]. The noise control may be: a) passive (reducing of noise by various absorbing materials), b) active (setting of a secondary source of sound) or c) combined [5], [6]. The passive method is very efficient for reduction of noise on higher frequencies (over 500 Hz). For reduction where frequencies are lower the absorbing materials are bulky and heavy because of greater wave lengths of the acoustic waves. The active noise control (ANC) applies interference of two acoustic waves with the same amplitudes and frequencies and opposite phases, so that it comes to mutual annulment. Realization of this idea implies application of a controlling system that, on the base of the frequency and phase of an undesirable acoustic signal, i.e. antinoise, forms a new one with the opposite phase and reproduces it by means of a loudspeaker [7]-[10].

Where noises are narrow-banded (noise caused by rotating machines), characteristics are measured not by a microphone but by a tachometer which provides the information about the fundamental frequency of noise. On the base of the fundamental frequency the controlling system can carry out modeling of all the important harmonics of noise. This kind of an active control is suitable for reducing of car motors noise because these acoustic signals are not synchronized with the motor rotation [10].

This paper presents an analysis of possible application of described system for noise reduction of fans where real values of impulse response of a room are taken into consideration. The acoustic impulse response was determined in the point where the microphone was placed (where the error is minimized) and in the points of the room in the level of the microphone. Determination of the acoustic impulse response of the room was performed by the *image* method suggested in [11]. Implementation of this method is described in [12]. The described method uses factors such as reflection degree, dimensions of the room, reflection coefficients of the walls and the ceiling as well as the directness of the microphone [13]. Simulation was done for the case of the application of the described system in the room by means of: a) sine arousal and b) real arousal.

This paper is organized as follows. Types of the ANC systems are described in Section 2. The principle of work of the ANC system for noise reduction of fan, the adaptive algorithm of the ANC system and the proposed simulation model for solving the problem of noise in a room, are described in Section 3. The simulation process, the results and their analysis are presented in Section 4. Section 5 gives the conclusion.

II. ANC systems

ANC systems are based on the following two principles [10]: a) Feedforward ANC where the coherent referent electric noise signal is generated on the base of the acoustic noise before the acoustic noise gets to the loudspeaker for elimination and b) Feedback ANC (with feedback) where ANC generates a signal on the base of an electric signal in the point where noise elimination is wanted.

Feedforward ANC systems are nowdays very widespread because of their efficiency. These systems, according to the frequency range where noise is reduced, can be divided into: a) adaptive wide-ranged ANC systems, where data about the acoustic signal of noise are gained by means of an acoustic sensor (microphone) and b) adaptive narrow-banded ANC systems, where data about the acoustic signal of noise are gained by means of a nonacoustic sensor (tacho generator, optical sensor, etc.). Application of nonacoustic sensors eliminates the undesirable effect of the feedback from the loudspeaker for the noise compensation.

Advantages of using the nonacoustic sensor may be seen in resolving the engineering problems like determination of the reference noise in devices with high temperatures, turbulent tubes (exhaust car tubes) etc. The ANC system in narrow-banded disturbances may control every harmonic separately. It is possible to use FIR (Finite Impulse Response) filter of the lower order which brings to reducing the calculation complexity and increasing the efficiency of the system.

III. Noise reduction by means of the ANC system

A. *Working principle of the ANC system*

The principal block diagram of the system for elimination of a fan noise is shown in Fig. 1 [10]. The information about

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the fundamental frequency of noise is gained by means of a tachometer. On the base of that information a sine signal with the frequency equal to the fundamental frequency of noise is generated in the block OSC. The sine signal and the signal from the microphone are being processed by the adaptive algorithm that changes the parameters of the ANC system.

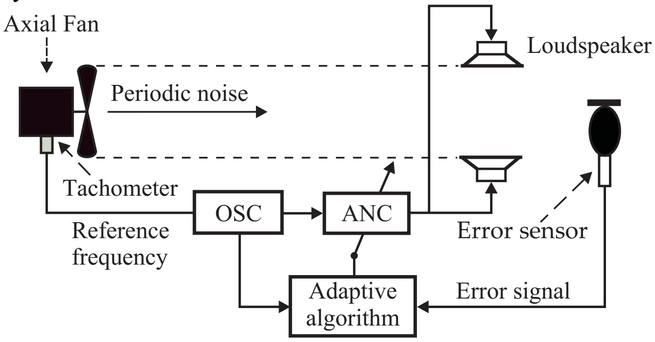


Figure 1. Block diagram of the system for noise reduction of the fan.

An electric signal for the loudspeaker arousal is generated on the outlet of the ANC system. The acoustic signal from the loudspeaker should have such an amplitude and phase that after superimposing with the acoustic signal of noise from the fan in the point of the microphone a minimal acoustic signal is gained. This signal is called the acoustic signal of error. The electric signal on the outlet of the microphone represents an electric signal of error and is led into the ANC system**.**

The adaptive algorithm after a certain number of iterative steps should so adjust the parameters of the ANC system that the signal of error may be equal to zero. Taking into consideration dimensions of the room where the fan, the loudspeaker and the microphone are, we can observe: a) the primary way that the acoustic signal passes from the fan to the microphone and b) the secondary way that the acoustic signal passes from the loudspeaker to the microphone. Characteristics of these ways may be described by means of the acoustic impulse response $h(n)$. On the base of the impulse response the transfer function can be determined when the Z-transformation is applied.

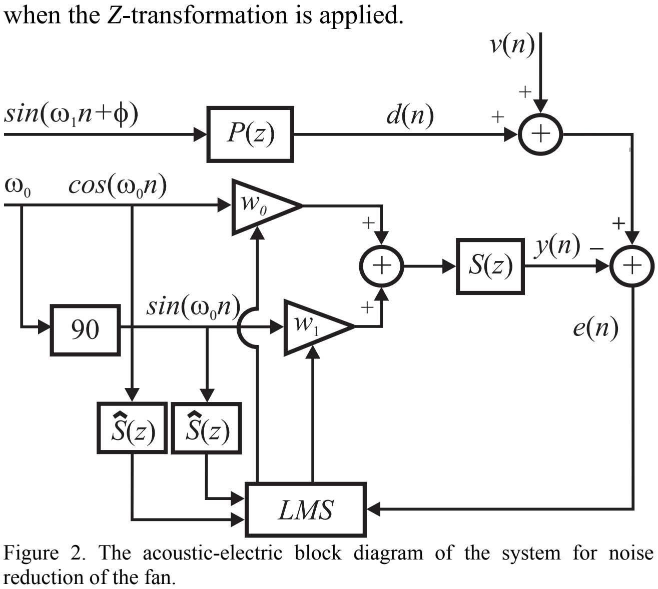


Figure 2. The acoustic-electric block diagram of the system for noise reduction of the fan.

The acoustic-electric equivalent diagram of the ANC system for reduction of the fan noise is presented in Fig. 2 [10]. The designations are: $P(z)$ the transfer function of the primary way, $S(z)$ the transfer function of the secondary way and $\hat{S}(z)$ model of the secondary way. In this paper it is taken that $S(z)$ is equal to $\hat{S}(z)$ as well as that the shift $v(n)=0$. The effect of the secondary way model on the performances of the system are analyzed in [14]-[20]. Strategy shown in Fig. 2 may be spent for the case when the primary noise contains the products of multiplication of harmonics when larger numbers of adaptive filters are used [10].

B. Adaptive algorithm

The reference signal generated on the base of the signal from the tachometer is $x(n)=A\cos(\omega_0 n)$, where the reference frequency is f_0. For the purpose of simplifying the mathematical apparatus without diminishing the generality of the analysis, it may be supposed that A=1. Parameters of the ANC system are determined by the adaptive algorithm. For the nth iteration it will be [10]:

$$w_0(n+1)=w_0(n)+\mu e(n)[S(z)\cos(\omega_0 n)], \quad (1)$$

$$w_1(n+1)=w_1(n)+\mu e(n)[S(z)\sin(\omega_0 n)], \quad (2)$$

where μ is the size of the iteration step and ω_0 the normalized circular frequency:

$$\omega_0 = 2\pi\frac{f_0}{f_s} \quad (3)$$

where f_S is the sampling frequency. The discrete system in Fig. 2 is intended for minimizing the difference $e(n)$ between the primary noise $d(n)$ and the generated noise (antinoise) $y(n)$. Then the transfer frequency is [5]:

$$H(z)=\frac{z^2-2z\cos\omega_0+1}{z^2-2z\cos\omega_0+1+\beta S(z)[z\cos(\omega_0-\phi_s)-\cos\phi_s]}, \quad (4)$$

where A_S and ϕ_S are the amplitude and the phase of the function $S(z)$, and $\beta=\mu A^2 A_s$. If the iteration step μ is small (μ << 1), then it will be:

$$H(z)=\frac{z^2-2z\cos\omega_0+1}{z^2-[(2-\beta)\cos\omega_0]z+1-\beta}. \quad (5)$$

If it is $S(z)$=1, the amplitude characteristic of the discrete system is:

$$T(\omega)=\left|H\left(e^{j\omega}\right)\right|=\frac{2|\cos\omega-\cos\omega_0|}{\sqrt{[(2-\beta)(\cos\omega-\cos\omega_0)]^2+(\beta\sin\omega)^2}}. \quad (6)$$

The amplitude of the error signal is:

$$A_e(\omega_0,\omega_1)=\frac{2A_d|\cos\omega_1-\cos\omega_0|}{\sqrt{\left[(2-\beta)(\cos\omega_1-\cos\omega_0)^2+(\beta\sin\omega_1)^2\right]}}, \quad (7)$$

where it is:

$$\omega_1=2\pi\frac{f_1}{f_s}, \quad (8)$$

normalized circular frequency. The error signal can be written down as:

$$e(n)=A_e(\omega_0,\omega_1)\cos(\omega_1 n+\phi), \quad (9)$$

where ϕ represents the phase angle. When the frequencies are mutually equal, then it will be:

$$A_e(\omega_0,\omega_1)\big|_{\omega_0=\omega_1}=0. \quad (10)$$

If the difference among the circular frequencies is small, the error signal is:

$$A_e(\omega_0,\omega_1) \approx A_e(\omega_0,\omega_1)\big|_{\omega_0=\omega_1} + \frac{\partial}{\partial \omega_0}\left[A_e(\omega_0,\omega_1)\right]\Big|_{\omega_0=\omega_1}(\omega_0-\omega_1) = \frac{2}{\beta}\left|\omega_0-\omega_1\right| \quad (11)$$

From the Eq. (10) it can be seen that the remaining noise equals 0, if the frequency of the referent signal ω_0 is strictly equal to the primary frequency ω_1. From the equation (11) it can be seen that the ANC system is exceptionally sensitive toward the frequency error ω_0-ω_1, so that even small errors can cause great degradations of the system.

C. Simulation model

The simulation model for noise reduction proposed in this paper is determined on the base of the acoustic-electric block diagram shown in Fig. 2 and the acoustic-electric block diagram shown in Fig. 3 [10]. The signal $e_{k_0 l_0}(n)$, detected on the location (k_0,l_0), represents the error signal that is minimized by the LMS algorithm. In some points of the room (k,l), where $0 \le k \le K\text{-}1$ and $0 \le l \le L\text{-}1$, the noise signal e_{kl} is calculated which was created as a consequence of activity of the noise source and the ANC system.

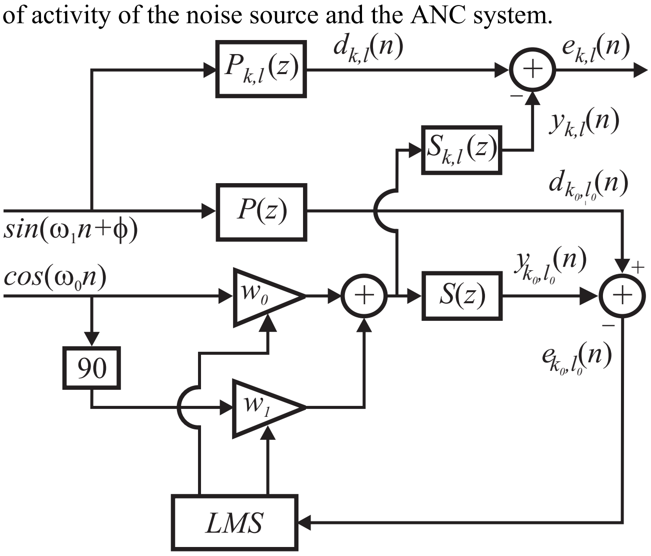


Figure 3. The acoustic-electric block diagram of the system for noise reduction in points in the level of the microphone.

For the proposed simulation model the acoustic impulse response between the loudspeaker (k_{Ls},l_{Ls}) and the position of the noise minimization (k_0,l_0) are determined and on the base of them the characteristics of the transfer function $S(z)$ are determined. Besides, the characteristics of the transfer function between the loudspeaker and the points (k,l) of the room, $S_{kl}(z)$, the transfer function of the noise source (k_F,l_F) and the points (k_0,l_0), $P(z)$, as well as of the noise source and the points (k,l), $P_{kl}(z)$ are also determined.

IV. Simulation and Analysis of the Results

A. Simulation

The described ANC simulation model (Eq. (1)–(13)) is used for noise reduction in the room meant for a classroom (Fig. 4), dimensions: X, Y, H. Simulation was done in $K \times L$ points, where:

$$K = \left\lfloor \frac{X}{d_x} \right\rfloor,\ L = \left\lfloor \frac{Y}{d_y} \right\rfloor. \quad (12)$$

d_x and d_y are distances on the x-axis and y-axis between the adjacent points of the grid. Indices within the grid are: $0 \le k \le K\text{-}1$ i $0 \le l \le L\text{-}1$. The ANC system for noise reduction of the fan (k_F,l_F) consists of the loudspeaker and the microphone that are in points (k_{Ls},l_{Ls}) and (k_0,l_0), respectively, where it is:

$$k_F = \left\lfloor \frac{x_F}{d_x} \right\rfloor,\ l_F = \left\lfloor \frac{y_F}{d_y} \right\rfloor,\ k_{Ls} = \left\lfloor \frac{x_{Ls}}{d_x} \right\rfloor,\ l_{ls} = \left\lfloor \frac{y_{Ls}}{d_y} \right\rfloor,$$
$$k_0 = \left\lfloor \frac{x_0}{d_x} \right\rfloor,\ l_0 = \left\lfloor \frac{y_0}{d_y} \right\rfloor, \quad (13)$$

where are $0 \le x_0$, $x_{Ls,}$ $x_F \le X$; $0 \le y_0$, $y_{Ls,}$ $y_F \le Y$. As a measure of efficiency of ANC system for reduction of the fan noise the mean absolute error is used. The mean absolute error is calculated for the signal of N samples for the last T samples (n=N-T,...,N). According to the notation from the Fig.3 the mean absolute value is calculated for any location (k,l) in the room:

$$\overline{E_{k,l}} = \frac{1}{T}\sum_{n=N-T}^{N}\left|e_{k,l}(n)\right|, \quad (14)$$

where $e_{k,l}$ is the simulation signal in points (k,l). The absolute mean value for the referent point (k_0,l_0) where the microphone is placed is:

$$\overline{E_{k_0,l_0}} = \frac{1}{T}\sum_{n=N-T}^{N}\left|e_{k_0,l_0}(n)\right|. \quad (15)$$

The mean absolute value for the referent point (k_0,l_0) caused by the action of the fan is:

$$\overline{E_{F_{k_0,l_0}}} = \frac{1}{T}\sum_{n=N-T}^{N}\left|d_{k_0,l_0}(n)\right|. \quad (16)$$

The mean absolute value for the referent point (k_0,l_0) caused by the action of loudspeaker (anti noise) is:

$$\overline{E_{LS_{k_0,l_0}}} = \frac{1}{T}\sum_{n=N-T}^{N}\left|y_{k_0,l_0}(n)\right|. \quad (17)$$

The efficiency of the noise reduction for the referent point (k_0,l_0) caused by the action of ANC is:

$$\eta_{k_0,l_0} = \frac{\overline{E_{F_{k_0,l_0}}}}{\overline{E_{k_0,l_0}}}, \quad (18)$$

and

$$\varepsilon_{k_0,l_0} = 20\log\frac{\overline{E_{F_{k_0,l_0}}}}{\overline{E_{k_0,l_0}}}\ [dB]. \quad (19)$$

Due to the action of the fan the mean absolute value in the entire room is:

$$\overline{E_F} = \frac{1}{KLT}\sum_{k=0}^{K-1}\sum_{l=0}^{L-1}\sum_{n=N-T}^{T}\left|d_{k,l}(n)\right|, \quad (20)$$

while due to the action of ANC system (loudspeaker) is:

$$\overline{E_{LS}} = \frac{1}{KLT}\sum_{k=0}^{K-1}\sum_{l=0}^{L-1}\sum_{n=N-T}^{T}\left|y_{k,l}(n)\right|. \quad (21)$$

The total mean absolute value in the entire room is:

$$\overline{E} = \frac{1}{KLT}\sum_{k=0}^{K-1}\sum_{l=0}^{L-1}\sum_{n=N-T}^{T}\left|e_{k,l}(n)\right|, \quad (22)$$

ANC system is so conceived to minimize the noise level in the referent point (k_0,l_0). In this point the fan noise and the anti noise are in the phase shift of $\phi=\pi$ and they mutually annul themselves. In other points (k,l) the phase shift of $\phi\neq\pi$ was not gained so that the effect of ANC is smaller in relation to the referent point. There are points where the effect of ANC system is such that the noise level becomes higher because of the small phase difference of the acoustic signals of the fan (k_F,l_F) and the loudspeaker (k_{Ls},l_{Ls}). It is possible to make notation locations where the effect of ANC system causes decreasing the noise level according to the criterion:

$$L_S(k,l)=\begin{cases}1, & \text{for } \overline{E_{F_{k,l}}} > \overline{E_{k,l}} \\ 0, & \text{otherwise}\end{cases}, \qquad (23)$$

Simulation was carried out for two kinds of disturbances:
a) the sine acoustic noise (simulation 1) and
b) the real acoustic fan noise (simulation 2).

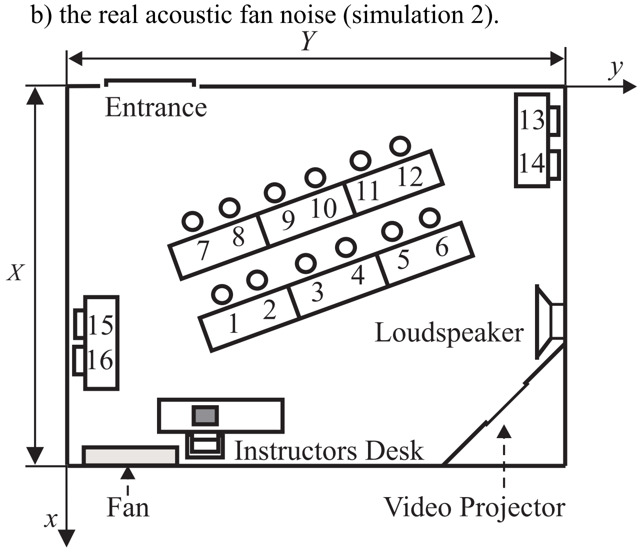


Figure 4. Presentation of the room where the impulse response was observed.

The real acoustic fan noise was gained by recording the noise of the fan in the real ambient conditions.

Simulation of the work of the ANC system was done for points: $M(k_0,l_0,h_0)$, where the microphone is situated (where the error is minimized) and in the points in the level of the microphone (point $M(k,l,h_0)$) by means of the presented acoustic electric block diagram (Fig. 3).

The acoustic impulse response of the room are determined by the algorithm described in the paper [12].

Dimensions of the room are: X=8m, Y=10m, H=3,1m. Positions of the fan, the microphone and the loudspeaker are: (k_F, l_F, h_F)=(8, 1.5, 2.8), (k_0, l_0, h_0)=(2.5, 4.5, 1.15) i (k_{Ls}, l_{Ls}, h_{Ls})=(5, 10, 1) , respectively. In the whole room the level of noise is calculated in $K\times L$=33$\times$41=1353 points, where d_x=d_y=0,25m. In calculation of the impulse response of the room [11]-[13] the real reflection coefficients of the walls, the floor and the ceiling of the classroom for which the simulation was done are taken into consideration {0.88, 0.88, 0.88, 0.88, 0.85, 0.58}. The parameters of the algorithm are: μ=0.1, f_0=15.097Hz, f_s=4kHz, A=1, A_s=2.5, β=0.25, N=4000, T=400.

B. Simulation results

1) Simulation 1

When the process of simulation for the sine arousal was applied, the following results are gained: $\overline{E_{F_{k_0,l_0}}}=3.8\cdot10^{-3}$, $\overline{E_{LS_{k_0,l_0}}}=3.9\cdot10^{-3}$, $\overline{E_{k_0,l_0}}=0.05\cdot10^{-3}$, $\eta_{k_0,l_0}=67.7$, $e_{k_0,l_0}=36.623dB$, $\overline{E_F}=5.1\cdot10^{-3}$, $\overline{E_{LS}}=5.6\cdot10^{-3}$, $\overline{E}=6.6\cdot10^{-3}$. Time diagrams are presented in Fig. 5 (noise signal $d(n)$ and the compensation signal $y(n)$) and Fig. 6 (error signal).

Noise distribution in the room is presented in Fig. 7 (signal of the level of the fan noise E_F), Fig. 8 (signal of the level of the loudspeaker noise E_{Ls}), Fig. 9 (the noise level of the superimposed signal E) and Fig. 10.a (presentation of isophonic lines of the room). The surface of the room with the compensated noise is presented in Fig. 10.b.

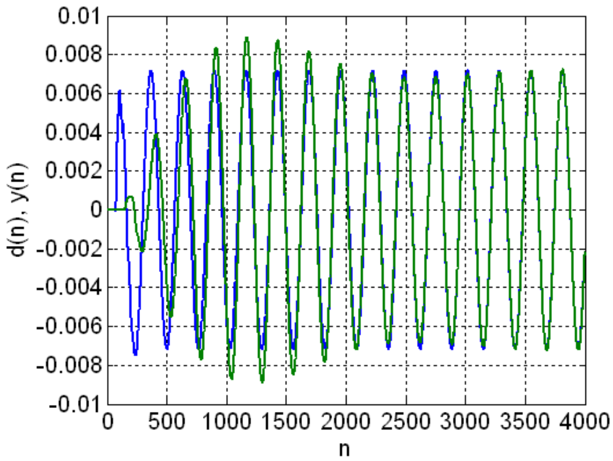


Figure 5. The noise signal d(n) and the compensation signal y(n).

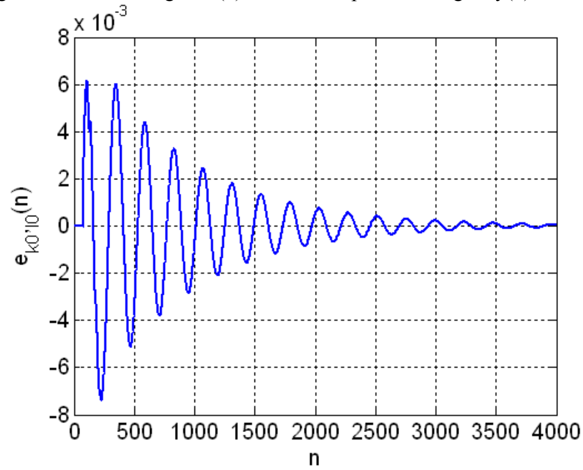


Figure 6. The error signal.

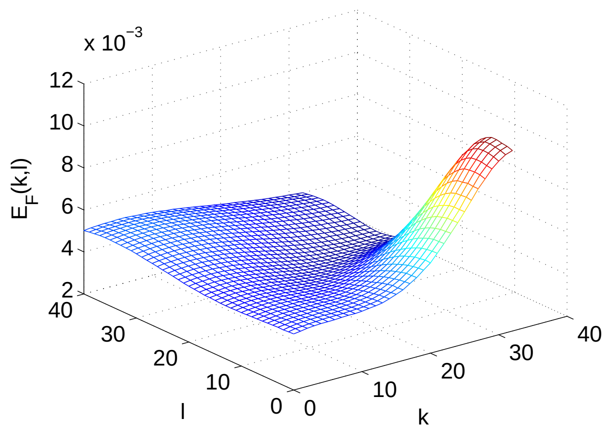


Figure 7. Signal of the level of the noise of the fan.

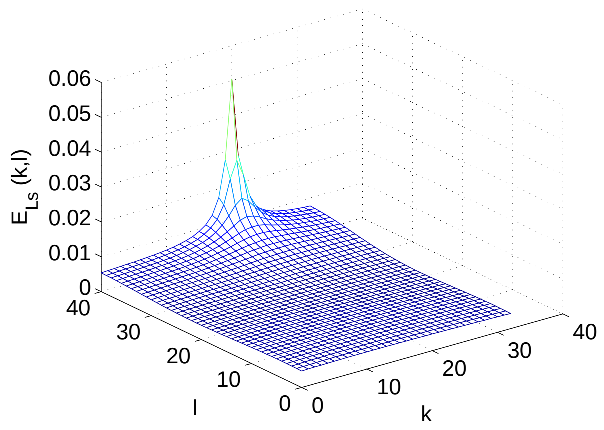


Figure 8. Signal of the noise level of the loudspeaker.

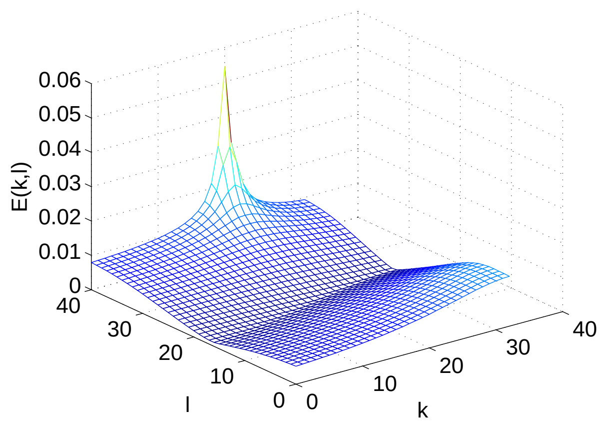
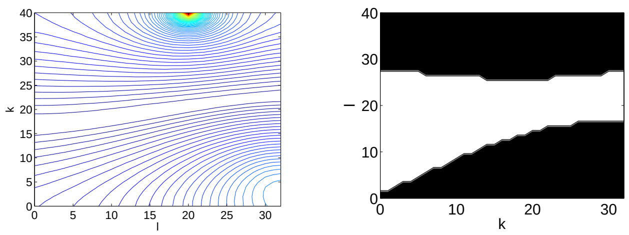


Figure 9. Noise level of the superimposed signal.



a) b)

Figure 10. a) Isophonic lines of the room. b) The surface of the room with the compensated noise.

2) Simulation 2

When the simulation process for the real signal is applied, the following results are gained: $\overline{E_{F_{k_0,l_0}}} = 1.6 \cdot 10^{-3}$, $\overline{E_{LS_{k_0,l_0}}} = 1.5 \cdot 10^{-3}$, $\overline{E_{k_0,l_0}} = 0.47 \cdot 10^{-3}$, $\eta_{k_0,l_0} = 3.47$, $e_{k_0,l_0} = 10.82 dB$, $\overline{E_F} = 2.3 \cdot 10^{-3}$, $\overline{E_{Ls}} = 2.1 \cdot 10^{-3}$, $\overline{E} = 2.8 \cdot 10^{-3}$. Time diagrams are presented in Fig. 11 (the noise signal $d(n)$ and the compensation signal $y(n)$ and Fig. 12 (the error signal). Noise distribution in the room is presented in Fig. 13 (signal of the noise level of the fan E_{kl}), Fig. 14 (the noise level of the loudspeaker E_{Ls}), Fig. 15 (noise level of the superimposed signal E) and Fig. 16.a (presentation of the isophonic lines of the room). The surface of the room with the compensated noise is given in Fig. 16.b.

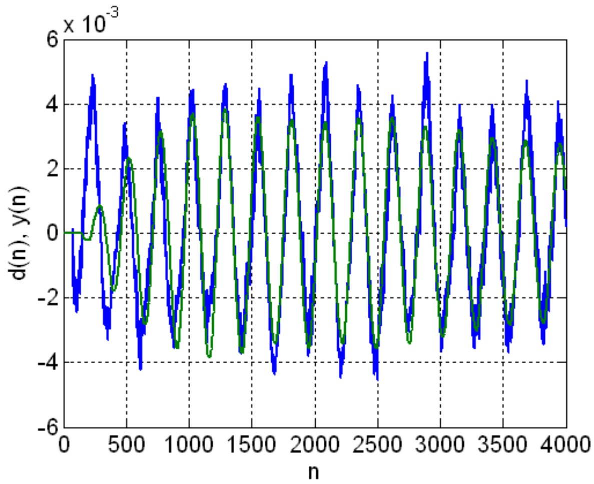


Figure 11. The noise signal d(n) and the compensation signal y(n).

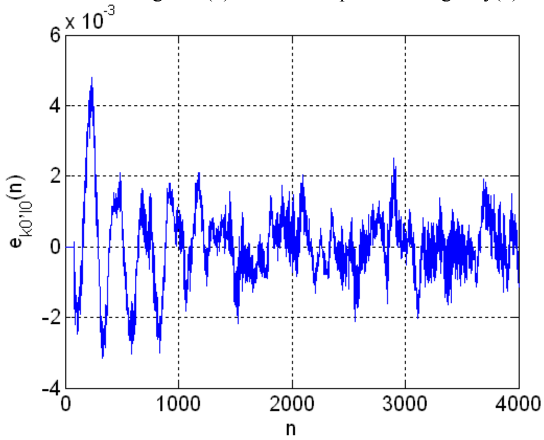


Figure 12. The error signal.

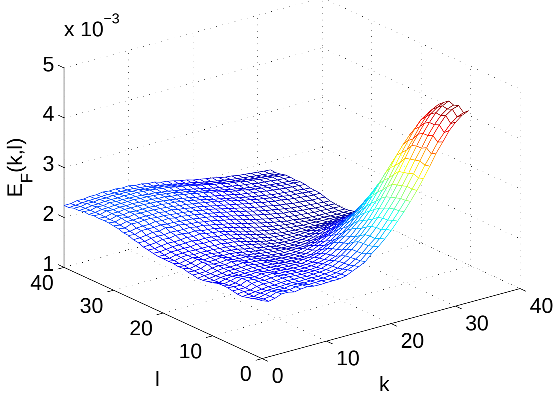


Figure 13. Signal of the noise level of the fan.

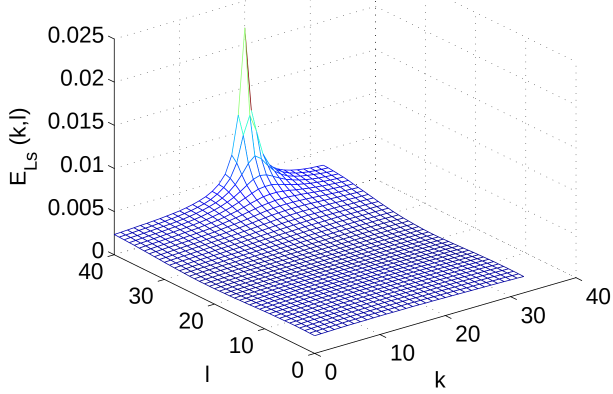


Figure 14. Signal of the noise level of the loudspeaker.

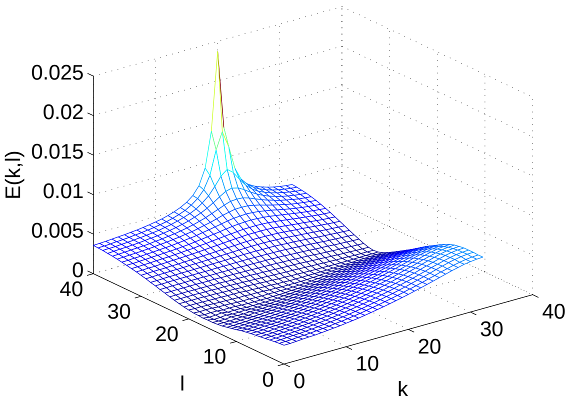
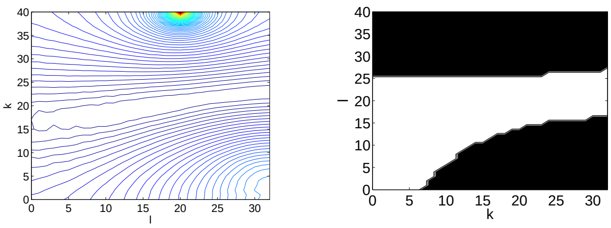


Figure 15. Noise level of the superimposed signal.



a) b)

Figure 16. a) Isophonic lines of the room. b) The surface of the room with the compensated noise.

C. Comparative analysis

The application of the ANC algorithm brings to the noise compensation of the fan for 36.623 dB (sine arousal) and 10.823 dB (real arousal) in the referent point (k_0,l_0) where the microphone is placed. After the analysis of the mean value of noise for the whole room before the action of the ANC ($\overline{E_{F_1}} = 5.1\cdot10^{-3}$, $\overline{E_{F_2}} = 2.3\cdot10^{-3}$) and during the action of the ANC ($\overline{E_1}$ =$6.6\cdot10^{-3}$, $\overline{E_2}$ =$2.8\cdot10^{-3}$) it can be evidently concluded that the mean level of noise is increased. According to the Eq. 23 it is possible to locate the points in the room where, in addition to the referent point, the noise level was decreased. In Figs. 11 and 18 the surfaces of the room, where the ANC system decreased the noise effect (p_1=62.15% and p_2=59.7% of the room surface), are presented. The efficiency of the ANC system is considerably higher in sine arousal because of the possible efficient adaptation to the parameters of arousal. On the other side, the real signal expresses some time uncertainty of parameters, which makes the work of the ANC system difficult and brings to the increase of the prediction error.

V. Conclusion

This paper presents the simulation model of ANC system for fan noise reduction in a room. This model is based on the application of the antinoise source (loudspeaker) and minimization of the noise level in a referent point (microphone). The level of the noise caused by a fan is determined in the whole room in discrete measuring points arranged like a grid. The effect of the antinoise source in the measuring points was determined by superimposing. ANC adaptive algorthm demanded determination of the acoustic impulse response of the room in the measuring points. The detailed analysis was carried out in the case of the fan noise in a sine form as well as the real fan noise. The effect of ANC system is such that the noise level is decreased in the referent point for 36.623 dB for the sine noise and 10.823 dB for the real noise. The analysis of the noise level for each point before and after the effect of ANC showed that in 62.15% of points of the sine arousal in relation to the real arousal (59.7%) the noise level was decreased. When the proposed simulation model is applied, it is possible to carry out detailed analyses of the effects of positions of fan, loudspeakers and microphones, room dimensions and absorbing characteristics of the walls, floors and ceilings and to choose the optimal values.

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