Noise Reduction in Controlled Environment using Simulation

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ABSTRACT: *Noise reduction with rotating elements in controlled environment is being studied in our work. To support our work, we have presented the results of simulation application. The results were obtained using the Active Noise Control modules. Initially we have explained the basic system of ANC and further described the results with analysis.*

Keywords: Noise Reduction, ANC System, Acoustic Impulse Response

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1. Introduction

The conventional method of noise reduction using passive acoustic absorbers is slightly efficient at frequencies below 500Hz. It is because the fact that at low frequencies the wave length of acoustic signal becomes higher comparing to acoustic absorber. The Active Noise Control gives good results while minimizing the acoustic disturbance at low frequencies.

The Active Noise Control operates on the principle of destructive interference of sound fields [1]. The destructive interference implies the superposition of two sound waves with phase shift of p. In a specific case, one sound wave is generated by a fan and therefore causes a nuisance, while other sound wave is generated by ANC system using loudspeakers and it is called a anti noise. The resulting sound wave at the place of receipt is considerably weakened or even annulated owing to the phase difference between these two waves. The resulting sound wave at the place of receipt is measured by an error sensor, i.e. by a microphone [1-2].

The basic ideas of the active control were first established by Paul Lueg in his patent published in The United States of America in 1936 [3]. In 1953 Harry Olson and Everet May discussed the active noise control system within both plane cockpits and car cabs [4]. In 1956 William Conover analysed the use of noise reduction active control for distributive transformators [5]. The intensive development in this field is highly contributed by the use of digital techniques for signal processing. By construction of adequate devices in the field of digital signal processing the production of practical systems with the active noise control is enabled.

In this study a simulative model of ANC system for noise reduction generated by a element rotating fan is applied in the meeting room. The acoustic impulse response is determined at the point where a microphone is set (where an error is minimised) and at some points of the room which are set at the level of the microphones. The image method is used for determination of acoustic impusle response in meeting premises [6]. The implementation of this method is described in [7]. The simulation is carried out in case that the above decribed system is applied in a room using a sinusoidal arrousal.

The organization of this study is as follows. In the section 2 the working principle of ANC system for noise reduction of fans and the adaptive algorithm of ANC system are described. In the section III the simulative results are shown along with the analysis of the results. The conclusion is in the section 4.

2. ANC Systems

Three requirements can be achieved using the ANC system: a) reaching a minimum of total acoustic power of a sound source, b) forming “silence zone” and c) realization of a system which would operate as a side signal absorber. Two basic control strategies are used: a) Feedback ANC, where ANC system generates signal based on the electric signal obtained by a microphone at the point where the noise elimination is wanted and b) Feedforward ANC where a coherent electric signal of a noise is generated from acoustic noise before the acoustic noise reaches the loudspeakers for the elimination.

2.1. Operating Principle and Adaptive Algorithm

The acoustic-electric equivalent scheme of ANC system for noise reduction of fan is shown in Figure 1 [8].

The sinusoidal signal with frequency which is equal to fundamental frequency of noise is generated in block OSC. The information about fundamental frequency of noise is given by a tachometer. The parameters of ANC system are changed by the adaptive algorithm LMS which also processes both sinusoidal signal and a signal from a microphone. An electric signal for the loudspeaker arousal is generated on the outlet of the ANC system.

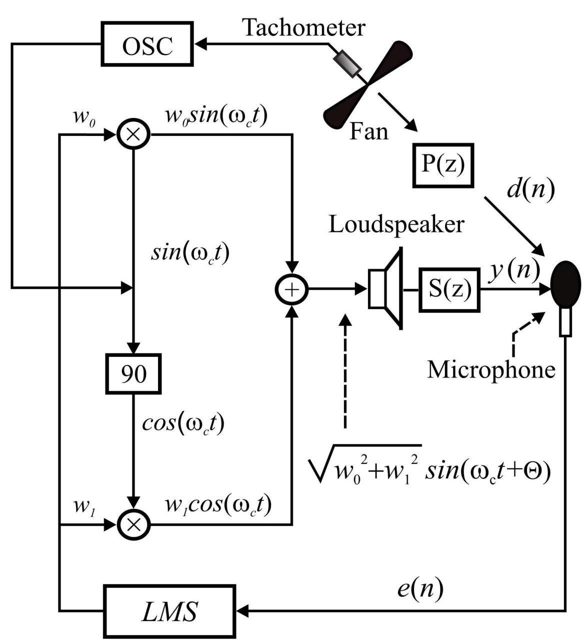

Figure 1. Acoustic-electric block scheme of the system for noise reduction for the fan

The acoustic signal out of a loudspeaker should have such an amplitude and a phase, so that after superposition with an acoustic noise signal of a fan at the point where a microphone is placed, obtains a minimal acoustic signal which represents the acoustic error signal. The electric signal on the outlet of a microphone is the electric error signal, e(n), and it is led to ANC system.

The adaptive algorithm LMS should adjust the parameters of ANC system so that after several iterative procedures the error signal equals zero. There is a fan, a loudspeaker and a microphone in the room where the simulative model is applied.

We are observing: a) the primary path of the acoustic signal of noise from a fan to a microphone and b) the secondary path of the acoustic signal of noise from a loudspeaker to a microphone. The characteristics of these path are described by acoustic impulse response $h(n)$ and subsequently by using Z-transformation of the transfer functions can be determined.

$P(z)$ is the transfer function of a primary path, while $S(z)$ is a transfer function of a secundary path. The effect of a secundary path model on the system characteristics is analysed in [9-12].

Reference signal is $x(n) = cos(\omega_0 n)$. Owing to its simplicity $A = 1$ is taken. For n-th iteracion results as follows [7]:

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

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

where μ is a step size of iteration, while ω_0 is a normalised circular frequency:

$$\omega_0 = 2\pi \frac{f_0}{f_s}, \tag{3}$$

where f_s is a sampling frequency. Having in mind that the objective of this system is minimising the difference $e(n)$ between a primary noise $d(n)$ and a generated noise $y(n)$, a transfer function is here [13]:

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

where is $\beta = \mu A^2 A_S$, A_S i ϕ_S are the amplitude and the phase function $S(z)$.

$$A_e(\omega_0, \omega_1) = \frac{2A_d|\cos\omega_1 - \cos\omega_0|}{\sqrt{[(2-\beta)(\cos\omega_1 - \cos\omega_0)^2 + (\beta\sin\omega_1)^2]}}, \tag{5}$$

Where is ω_1 normalized circular frequency

$$\omega_1 = 2\pi \frac{f_1}{f_s}. \tag{6}$$

The error signal is:

$$e(n) = A_e(\omega_0, \omega_1)\cos(\omega_1 n + \phi), \tag{7}$$

where. ϕ is a phase angle. For the same frequencies is get:

$$A_e(\omega_0, \omega_1)\big|_{\omega_0 = \omega_1} = 0. \tag{8}$$

Accordingly, when the frequency of a referent signal ω_0 is strictly equal to the primary frequency ω_1 a remaining signal equals 0.

3. Simulation and Results Analysis

3.1. Simulation

The ANC system model for noise reduction is defined by the acoustic-electric block scheme shown in the Figure 1.

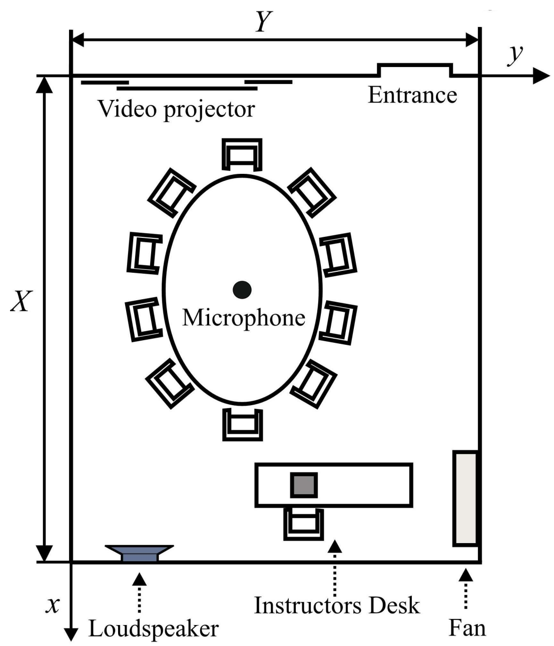

Figure 2. The image of the room

The acoustic impulse responses between the loudspeaker (k_{Ls}, l_{Ls}) and the positions of noise minimization (k_0, l_0) are determined. The characteristics of the transfer function $S(z)$ and the characteristics of transfer functions between the loudspeakers and some points within the room (k, l), $S_{kl}(z)$ are also established. The transfer function from the noise source (k_F, l_F) and point (k_0, l_0), $P(z)$ is determined, as well as the transfer function from the noise source (k_F, l_F and point (k, l), $P_{kl}(z)$.

The simulation model of the ANC system for noise reduction is applied on the meeting room, with dimensions $X = 8$m, $Y = 3$m, H = 3.5m. The image of the room is given in the Figure 2.

The noise level in whole room is calculated in $K \times L = 33 \times 13 = 429$ points. The indexes $k = (0\text{-}32)$ *i* $l = (0\text{-}12)$ are chosen within the grid. The distances along the x-axis and yaxis between adjacent points of a grid are $dx = dy = 0.25$m. The ANC system of noise reduction from a fan (k_F, l_F) is consisted of a microphone which is set in the points (k_{Ls}, l_{Ls}) and (k0, l0), respectively. The valid values are: $0 = x_0, x_{Ls}, k_F = X$; $0 = y_0, y_{Ls}, y_F = Y$. The following locations are included: $(k_F, l_F, h_F) = (7.25, 3, 3)$, $(k_{Ls}, l_{Ls}, h_{Ls}) = (8, 0.5, 3)$ *i* $(k_0, l_0, h_0) = (3.25, 1.5, 0.75)$. The simulation is implemented for a sinusoidal arousal at point M (k_0, l_0, k_0), where a micophone is location and also at points which are set in the plane of the microphone (points (k, l, h_0)). For the calculation of an impulse response of the room [7], the real reflection coefficients of the walls, the floor and the ceiling are as follows: 0.95, 0.95, 0.95, 0.85, 0.85 i 0.88. The algorithm includes the following parametres: a number of iterative steps $N = 2000$, a number of steps during the analysed $T = 400$, a size of iteration steps $\mu = 0.1$, referent frequency f_0=15.097Hz, sampling frequency f_S=4kHz, amplitude $A = 1$, amplitude of the fuction $S(z)$ $A_S = 2.5$, and $\beta = 0.25$.

The effectiveness of ANC system effect on noise reduction within the meeting room is characterized by a mean absolute value of the signal at a specific point of the room $\overline{E_{k,l}}$ and a mean absolute value of the signal for the whole room E. At a referent point in the room (k_0, l_0), the effectiveness of the effect is characterized by two values: the relation between a mean absolute value for noise level of the fan before the effect of the ANC system and a mean absolute value for noise level at the measured point during the ANC operation, η_{k_0,l_0}, and its logarithmic function, ε_{k_0,l_0} (dB).

3.2. Simulation Results

The results obtained using the simulative model of ANC system for the sinusoidal arousal are as follows: $\overline{E_{F_{k_0,l_0}}} = 1.8\times 10^{-3}$, $\overline{E_{Ls_{k_0,l_0}}} = 1.8\times 10^{-3}$, $\overline{E_{k_0,l_0}} = 0.37\times 10^{-7}$, $\overline{E_F} = 2.1\cdot 10^{-3}$, $\overline{E_{Ls}} = 2.1\cdot 10^{-3}$, $\overline{E} = 0.08\cdot 10^{-3}$.

In the Figure 3 the time diagram of noise signal $d(n)$ and compensational signal $y(n)$ is shown. In the Figure 4. the time diagram of an error signal is shown. In the Figure 5 and Figure 6 the spatial distribution of noise level signal for the fan E_F and the loudspeaker E_{Ls} is shown. In the Figure 7 the distribution of a noise level for the superposed signal E is shown. The Figure 8 shows the equal-loudness contours of the room. The room area with compensational noise is shown in the Figure 9.

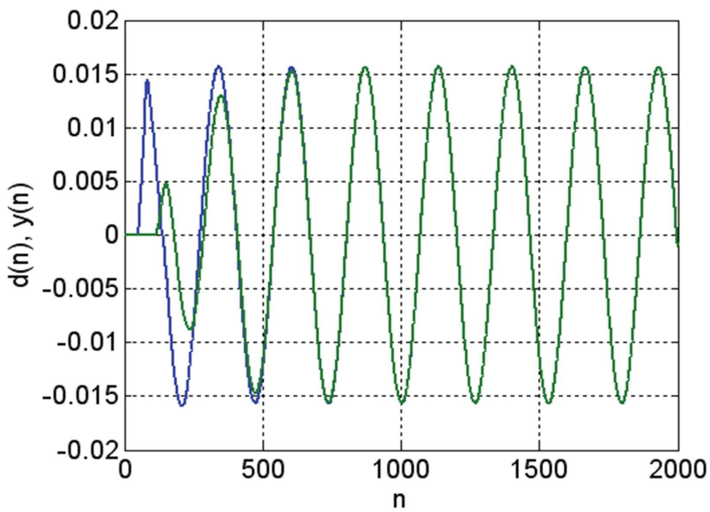

Figure 3. Noise signal $d(n)$ and compensational signal $y(n)$image of the room

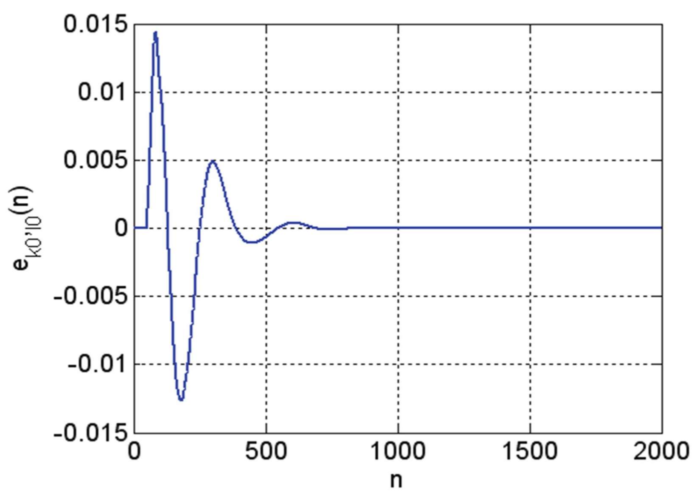

Figure 4. The compensation of noise error signal image of the room

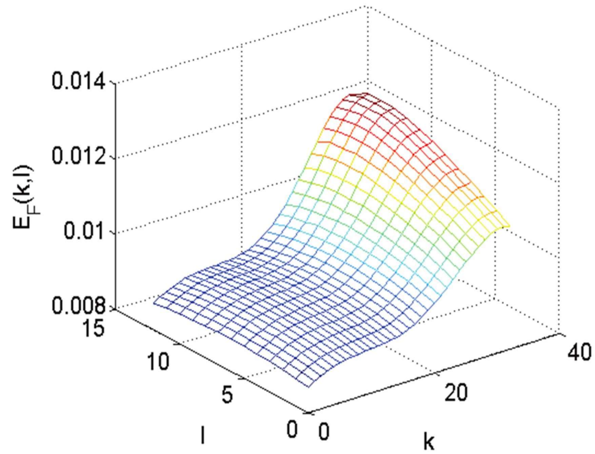

Figure 5. Noise level signal for the fan image of the room

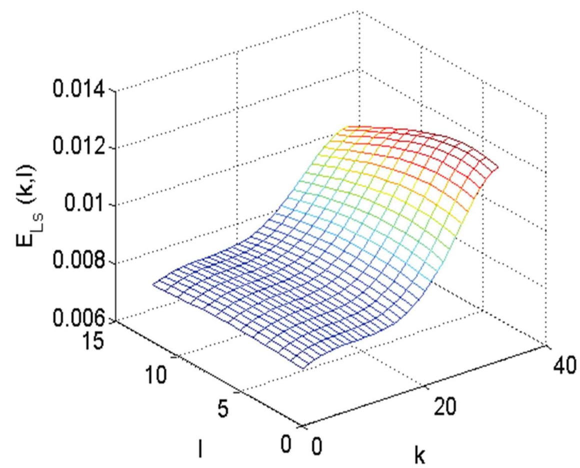

Figure 6. Signal of loudspeaker noise level example

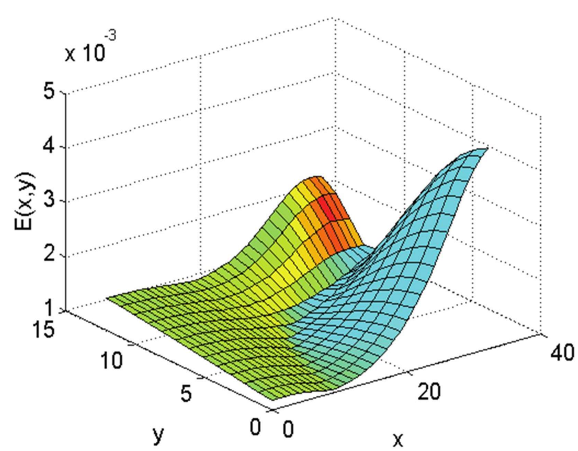

Figure 7. Noise level of the superimposed signal example.

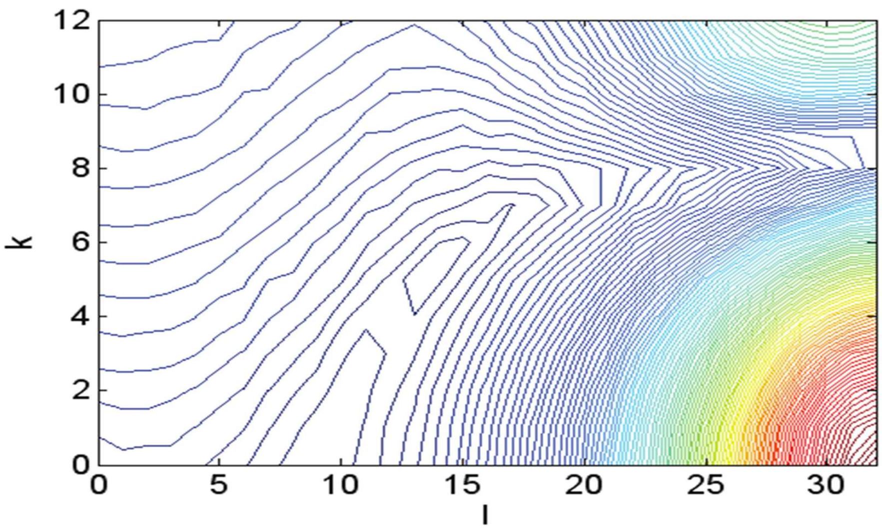

Figure 8. Equal-Loudness contours of the room

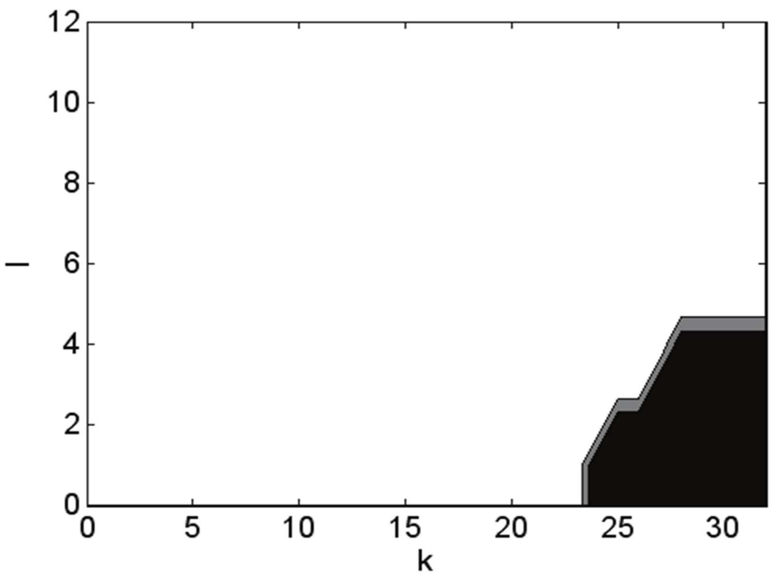

Figure 9. The room area with compensated noise

3.3. Results Analysis

The average noise level is reduced, which can be seen according to both numeric and graphic results. The mean noise level for the whole room before the effect of the ANC system was $\overline{E}_F = 2.1\cdot 10^{-3}$, but during the effect of the ANC system $\overline{E} = 0.08\cdot 10^{-3}$. The compensation of noise for the fan at the microphone location is 93.84dB. The compensation success rate is 91.37% (Figure 9).

4. Conclusion

The results obtained by applying simulative model of the ANC system for noise reduction, which comes from the rotating element, on the meeting premises, show that the noise reduction is not achieved for only 8.63% of the room area. Such results indicate a possibility to investigate the use of this model on an operating system in real time.

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