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Simple Feedback Structure of Active Noise Control in a Duct

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An active noise control is usually constructed with the use of electronic filters. For sufficient noise attenuation, electronic filters are not always needed. In this case, the electronic controller and appropriate software is not required so the system can be much easier. This paper deals with the use of a feedback structure of active noise control in an experimental ventilation duct. Simulation was performed to investigate the efficiency of a simple analoguous system for active noise control without incorporating electronic filters. The transfer function of the entire analogous system can be used to predict the maximum attenuation level. Tests were made to verify the simulation and to show what noise attenuation level can be achieved in an experimental duct. It has been shown that in a specific frequency range this kind of a system is efficient enough for use in some ventilation ducts.

Keywords: active noise control, feedback control, ANC simulation, phase shift

0 INTRODUCTION

The beginning of active noise control (ANC) is in year 1936, when Lueg patented his idea about realization of the active noise control [1]. It uses the principle of interference and absorption. Because electronics in those years was not enough advanced to meet the needs of a controller, no real system was produced.

Today, two basic methods are used for active noise control; feed-back and feed-forward. They are using all capabilities of contemporary computers and other electronics. These systems are used mainly in conjunction with adaptive filters, which make them capable to cope with bad system response. But there is a question, if it is possible (in some cases) to use a simpler system, which is efficient enough and at the same time cheaper and more reliable because of less electronic components. This is the purpose of the experimental ventilation duct that was made in this research.

The characteristic of ventilation ducts' fans is a constant rotating speed. This means that the emitting noise is constant and tonal in lower frequency band. The intention of this test is to achieve good noise attenuation with feedback structure of an ANC system without incorporating adaptive or other electronic filters. These filters are capable to compensate bad system response, and at the same time, they contribute to considerable phase shift of the original signal. This part of phase shift can be eliminated if the filter is not included in the feedback loop. It should be aware of the fact that, for example, an amplifier itself also contains filters. In this research, no additional filter was used.

1 FEEDBACK STRUCTURE OF ACTIVE NOISE CONTROL

Feedback structure was found in 1953, when feedback loop was constructed of a microphone, an amplifier and a loudspeaker [2]. Authors have named this scheme as the »sound absorber«. In this scheme, microphone is used as an error microphone, where error means deviation from the theoretical attenuation. Signal is travelling through the amplifier and the controller, where the amplitude is adjusted and phase is shifted for 180°. Modified signal goes to the loudspeaker, where the attenuation of noise occurs. Schematic representation of a feedback structure is shown in Figure 1.

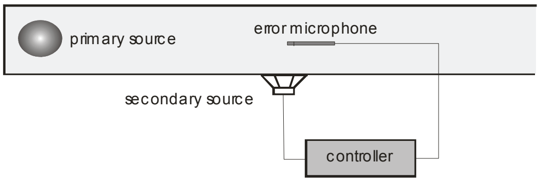

Fig. 1. *Schematic representation of a feedback structure of ANC*

Today, a feedback structure is sometimes used as a way to avoid acoustic feedback in feed-

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forward structure. A feedback ANC configuration is also very suitable when there is no available reference signal [3].

To achieve maximum level of attenuation, signal from the microphone should be amplified as much as possible, but should not exceed the stability range. Block diagram, corresponding to a feedback control system in the complex frequency s-plane, is shown in Figure 2, where *E(s)* is a transfer function of acoustic path between the primary source and the microphone, *F(s)* is a transfer function of acoustic path between the secondary source and the microphone, *M(s)* a transfer function of the microphone, *C(s)* a transfer function of the controller, *N(s)* a transfer function of the electronics (except the controller) and *L(s)* a transfer function of the secondary source.

Typical characteristics of this method is a feedback loop, which leads signal *S(s)* from the secondary source back to the microphone through the acoustic path. There it is added to the reference signal *R(s)* and the error signal *D(s)* is therefore increased. Signal from the error microphone *D(s)* can be written as:

$$D(s) = R(s) + F(s)S(s) \tag{1}$$

or

$$D(s) = E(s)P(s) + \\ + F(s)M(s)C(s)N(s)L(s)D(s) \tag{2}$$

It leads to a simplified form of a transfer function between a microphone signal *D(s)* and a signal of primary source *P(s)*:

$$\frac{D(s)}{P(s)} = \frac{E(s)}{1 - M(s)C(s)N(s)L(s)F(s)} \tag{3}$$

Because the error signal should be as low as possible (theoretically vanish), it is $D(s) = 0$. Further derivation shows that for the best attenuation, the transfer function of the controller should have an infinite amplitude. Because of the stability problem, this is not possible. When a feedback loop is amplified too much, other elements may cause the system to become unstable. Therefore an optimum operational point should be found, which is normally near the stability limit.

2 SYSTEM STABILITY

When constructing the ANC system with a feedback loop, the stability problem should always be observed. Position of the microphone and the secondary source has strong influence on stability. In theory, it is considered that both elements are infinitely close to each other. But this is not possible to reach because of the following reasons. First, it should be taken into account that both elements can not be placed close to each other because a near-field influence would be too high to get good accuracy. Second, dimensions of a loudspeaker and a microphone are preventing to define their optimum placement. For this reason, there is always some gap between the microphone and the loudspeaker, which affects the efficiency of the whole system.

If the microphone and the loudspeaker would be so close to each other that the sound waves around a microphone would be defined solely by the movement of the loudspeaker, the microphone would get the same signal as the loudspeaker produces (no phase shift). This is not possible and the gap causes some time delay in the signal, which results in phase shift. If the wavelength of the signal is significantly greater than a gap between the microphone and the loudspeaker (intermediary length, *l*), phase shift is relatively small.

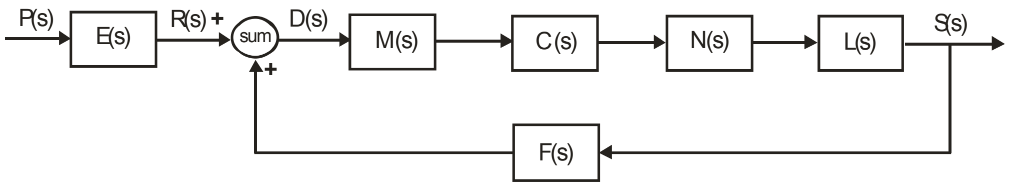

Fig. 2. *Block diagram of a feedback control system of ANC*

But if the wavelength is similar to the intermediary length (this occurs at the higher frequencies), phase shift becomes relatively large and efficiency of the system drastically decreases. If the frequency is increasing (at some fixed intermediary length), the phase shift is therefore also increasing, while the efficiency of the system is decreasing. Besides, it should be also taken into account the property of the real acoustical and other components, that each of them causes some phase shift in the signal. This is more significant for loudspeakers and filters.

When (for a specific sound frequency) the intermediary length corresponds to the equation (4), the phase shift between the sound, produced by the loudspeaker, and the sound, captured by the microphone, is equal to 180°:

$$l = \frac{\lambda}{2} \tag{4}$$

This means that the work of the controller, which reverses the phase for 180°, is totally ineffective and instead of attenuation it causes amplification of the primary noise.

Real noise, which is wanted to be attenuated, consists of many narrow frequency bands. Therefore, at every single intermediary length, one narrow frequency band exists, which causes the noise at this frequency to be amplified, and not attenuated. When the amplification is increasing, the amplitude is also increasing and at the specific point it exceeds the stability limit. Unstable working conditions occur and the system is not working appropiately anymore. The longer the intermediary length, the lower the limit frequency for stable operation.

Many systems of the ANC are constructed with the use of electronic adaptive filters [4] and [5]. They are able to partly compensate non-ideal response of each component and other causes of reduced efficiency. One of the filter problems is, that it causes phase shift in the signal, which results in lower efficiency. Therefore it is interesting to find out how these filters can be avoided.

3 MEASUREMENTS

The purpose of this experiment was to construct an ANC system with a feedback loop in an experimental ventilation duct without any additional compensation filter, and to get the noise attenuation, which is good enough. The set-up was constructed of plywood with thickness of 2 cm (Figure 3). For simulating a fan, the loudspeaker with generated broadband pink noise was used. Narrowband or tonal noise seems to be a better choice for simulating a fan, but the experiment was intended to examine the efficiency of the system through the wider frequency spectrum.

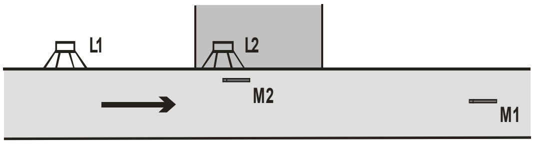

Fig. 3. *Experimental ventilation duct scheme*

A feedback loop, used for generating anti-noise, was constructed of a microphone Bruel&Kjaer (BK) 4165 (M2 in Fig.3), a measuring amplifier BK 2636 and a loudspeaker Visaton W170S8 with reversed polarity (L2 in Fig. 3). For primary noise generation (pink noise) the following equipment was used: a sine random generator BK 1027, a power amplifier BK 2706 and a loudspeaker JVC CS-HX621 (L1 in Figure 3).

3.1. Frequency Response of a Loudspeaker

The loudspeaker for anti-noise generation was built in an open box. According to its casing, theoretical response of the loudspeaker was calculated by the program WinISD 0.44. From the theoretical frequency response in Figure 4 it can be evident that because of resonant frequency at around 30 Hz, the loudspeaker is suitable for the use in a frequency range above 40 Hz.

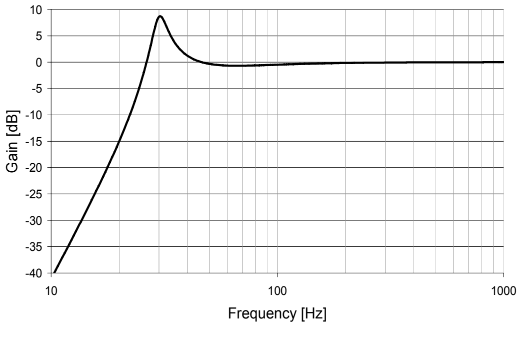

Fig. 4. *Theoretical frequency response of the loudspeaker*

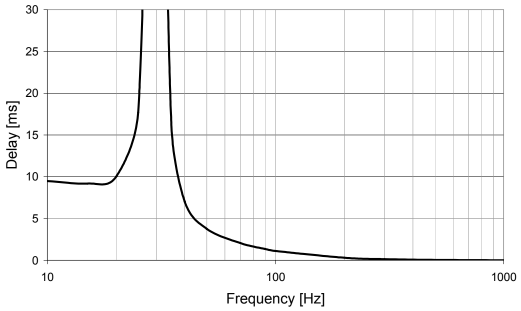

Fig. 5. *Theoretical group delay of a loudspeaker*

Frequency range, appropriate for noise attenuation, is seen in Figure 5, which shows theoretical group delay of the loudspeaker. At 60 Hz and lower, a group delay is too long for the system to work properly.

3.2. Frequency Response of a System

Many components of the ANC system have influence on the effectiveness of noise attenuation and each of them has its own transfer function. The main reason for delayed and modified signal in a simple ANC system (without electronic filters) is the loudspeaker, other components (microphone, amplifier, cables, ect) contribute less.

Measurement of the input and output was performed in the open loop (Figures 6 and 7), from which the transfer function can be calculated. It is obvious that the signal is very modified at the end of the loop, after it passes all the components of the ANC system. Such a simple system is very hard to compensate, because some frequencies are filtered out by the transfer function of the system. For appropriate compensation, the amplification of these particular frequencies with zero amplitude should theoretically be infinite, which is not possible.

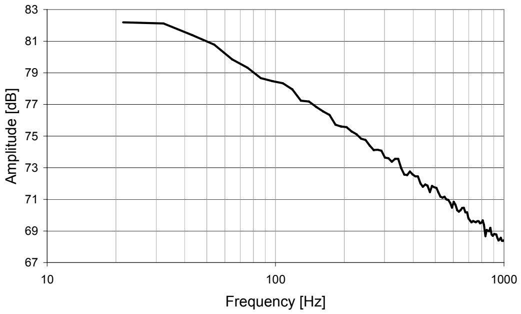

Fig. 6. *Input signal into the system*

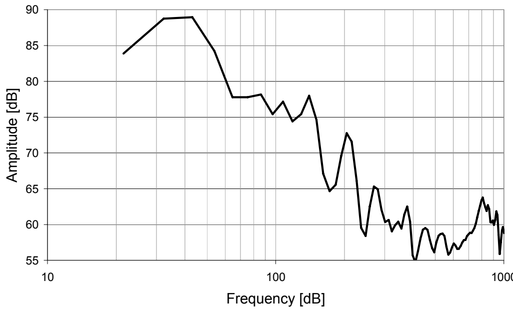

Fig. 7. *Output signal from the open loop*

3.3. Noise Attenuation Measurement

Then the noise attenuation with the ANC system was measured. The microphone, which is used for measurement of the attenuation level (M1 in Figure 3), was placed near the exit of the experimental duct, 45 cm from the end. Noise spectrum with and without use of an ANC, measured with the microphone M1, is seen in Figure 8 and the attenuation level in Figure 9. It is obvious that the useful frequency range of the system is approximately between 40 and 140 Hz.

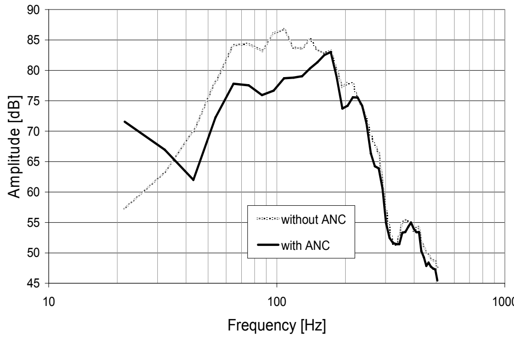

Fig. 8. *Attenuated (black line) and non-attenuated (grey line) noise in the duct*

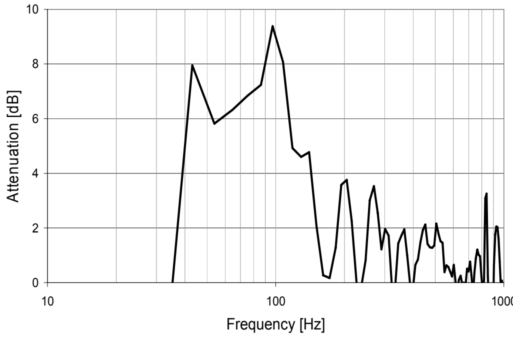

Fig. 9. *Attenuation level in the duct*

With regard to the fact that some ventilation fans produce noise mainly in lower frequency ranges, this ANC system could be very useful in that cases, despite of its narrow working range. Better attenuation could be measured if the primary noise would be narrowband or tonal noise, which is in fact more significant for ventilation fans.

4 SIMULATION

When constructing an ANC system, it is important to know its working limits. Simulation of the ANC was performed, using Matlab 7.1, to predict the effectiveness of the system. It shows the possibilities of a particular ANC system in a sense of how much the noise can be attenuated.

For a simulation to perform, the input and output of the system with the open loop were measured (as mentioned before). These two signals were used to calculate the coefficients of the impulse response, with the help of an LMS filter (Figure 10). From the impulse response coefficients, the response of a closed loop can be calculated.

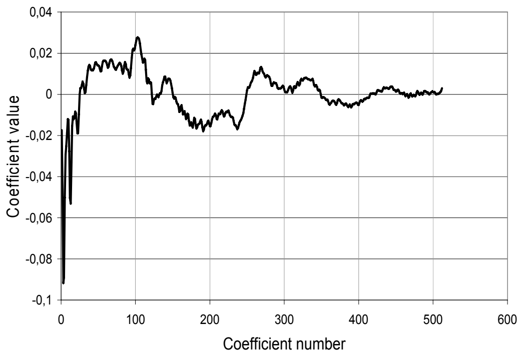

Fig. 10. *Impulse response coefficients*

According to Figure 11, *p(m)* is measured input and *x(m)* is output of a closed loop. Filter coefficients are marked as a_i, where index i means its instantaneous number.

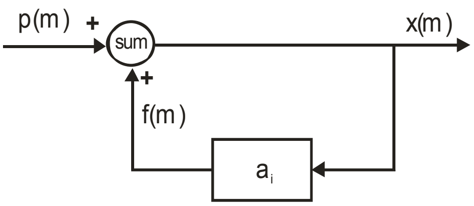

Fig. 11. *Block diagram of a feedback loop*

For a closed loop, the following equation can be written:

$$x(m) = p(m) + f(m), \tag{5}$$

where

$$f(m) = K\left[x(m)a_1 + x(m-1)a_2 + \dots \right. \\ \left. \dots + x(m-(L-1))a_L\right] \tag{6}$$

K is gain of a feedback loop, m goes from L to the end of the signal and L is the number of filter coefficients. Extended equation is:

$$x(m) = p(m) + K\left[x(m)a_1 + \right. \\ \left. + x(m-1)a_2 + \dots + x(m-(L-1))a_L\right] \tag{7}$$

The final equation of the output (attenuated) signal can be written as:

$$x(m) = \frac{1}{1-Ka_1}\left[p(m) + Kx(m-1)a_2 + \right. \\ \left. + Kx(m-2)a_3 + \dots + Kx(m-(L-1))a_L\right] \tag{8}$$

In this way the transfer function of the whole system was simulated by the computer program. To represent the efficiency of the system, the input signal is compared with the output signal of the closed loop, which comes from the computer algorithm.

The results of the simulation are represented in Figure 12 (attenuated and original signal) and in Figure 13 (attenuation level). As expected, the simulated noise attenuation level is higher than the real measurement, because this is the best possible (ideal) attenuation for this specific ANC system and the equipment used.

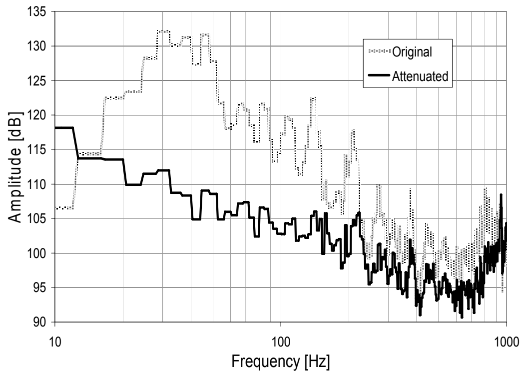

Fig. 12. *Original input (grey line) and simulated attenuated output signal (black line)*

But the useful frequency range is more similar to that in measurement, which confirms that the simulation is correct.

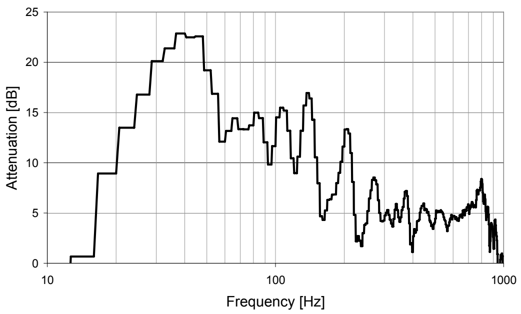

Fig. 13. *Attenuation level of the simulation*

When evaluating the results, some other influences should also be taken into consideration. The result is in strong correlation with the intermediary length in a feedback loop and also with the position of the microphone with regard to the loudspeaker of the secondary source. This is because sound reflections are present in the duct and the sound is travelling in different directions. The more the microphone is far from the near-field, the more the reflections and other phenomena influence on the signal, consequently lower the efficiency of the system is.

Besides, the result also depends on the place, where the attenuation is measured (microphone M1 in Figure 3). Because of the impedance mismatch between the duct and the surroundings, a part of the sound waves is reflected back to the primary source, which leads to the standing waves phenomena [6] and [7]. On the specific points of the duct at the specific frequency, nodes are formed, which mean that at that points the measured noise level would be very low.

5 CONCLUSIONS

The experimental ventilation duct with a feedback method of an ANC was constructed. The possibilities of the ANC without additional electronic or other filters were investigated, because an important part of time delay of the signal may be caused by the filter. The LMS filter and the measured input and output signal were used to do the simulation of an ANC system. It shows the maximum possible attenuation for a specific ANC system. Then the measurement of the attenuation level was performed. The experiment showed that the additional filters can be avoided, if only a specific frequency range of noise (about 40 to 140 Hz) must be attenuated. Some ventilation fans can meet this requirement. In this case, the ANC system becomes very simple and robust, because the feedback loop is constructed of fewer components. It contains just a microphone, an amplifier and a loudspeaker. During a design process, an engineer should be aware of the fact that the efficiency of the system depends on the position of the error microphone with regard to the loudspeaker for the anti-noise generation. They should be located close to each other, but not too close to become influenced too much by the near-field.

The noise, used for measurements and simulations, was broadband pink noise. Better attenuation is expected in case of using narrowband or tonal noise, which are more similar to the real ventilation fan noise.

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