Analysis and Implementation of Active Noise Control Strategies using Piezo Actuators

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ABSTRACT
Ventilation systems are used in order to provide an appropriate temperature inside buildings. For this reason, convection heat transfer is achieved by fans and extractors to produce wind flows. Nevertheless, fans are loud in operation, therefore passive or active mechanisms are used to reduce the noise level inside buildings. However, passive mechanisms are costly and need big space to be fixed around the ventilation system to absorb noise compared to Active Noise Control (ANC).

ANC systems are being employed to avoid big costs and space requirements. Accordingly, strategies and algorithms for active noise reduction were analyzed, implemented and tested experimentally in this paper for applications in duct ventilation systems. Algorithms implemented were simulated by Matlab software. Piezo actuator was utilized as an active element of ANC system during experimental testings.

Keywords: Active Noise Control, Filter X Least Mean Square, Hybrid Algorithm.

1. INTRODUCTION
Ventilation systems are needed in many offices and buildings so as to achieve a pleasurable environment for people who work or live inside. Notwithstanding, noise pollution, which is produced by fans motors, is the disadvantage of using ventilation systems [2]. For this reason, loud sound wave is canceled by a wave with same frequency and opposite amplitude, this property is applied in many transport systems [9], ANC is applied to prevent external noise signals affecting the performance of the driver and crew. This objective is also found in several intelligent headphones [1]. The mining industry also requires ANC [9] for hearing health care of workers, who work with noisy motors. Currently, ANC is widely investigated in smartphone applications, in order to increase their functionality in noise control [7], [2].

Simulated and experimental tests of an adaptive controller designed for a prototype ventilation system are shown. The characteristics of the adaptive controller achieves attenuation of sound signals produced by the ventilation system and external disturbances which affect the control system.

The system designed has a quickly response in finding appropriate weights for the filter representing the system behavior. A ventilation system prototype was described by author [2] for testing of designed algorithms.

2. ACTIVE NOISE CONTROL ANC

Fig. 1 shows the system model identification proposed by [3]. The estimation of an unknown plant \( P(z) \) is achieved using an adaptive filter \( W(z) \) was proposed by [3]. The adaptive filter is flexible according to the change of the weights to be assigned. After system identification, it is possible to design some kind of adaptive controller. In order to get the attenuation of the undesired noise signal, Least Mean Square (LMS) is defined as a procedure to obtain the error between the output and reference signal by a successive correction of the filter weights [2].

![Figure 1. Simple ANC scheme proposed by authors [3].](image_url)

The error is given by the equation 1. That equation is achieved from the figure (1), which means a feedforward analysis in the system with the plant transfer function \( P(z) \) through the secondary path \( S(z) \) and the filter \( W(z) \). The system is analyzed in Z transform, because it will be processed by a computer. The primary path is the acoustic response calculated from the reference sensor to the error...
sensor, that means transfer function $P(z)$. On the other hand, the secondary path $S(z)$ is the transfer function calculated from the filter output to the error sensor.

$$E(z) = [P(z) - S(z)W(z)]X(z)$$  \hspace{1cm} (1)

**FXLMS algorithm**

The filter that properly adjust the error with the reference signal $X(Z)$, is known as Filter X Least Mean Square (FXLMS). This filter avoids instabilities caused by the presence of the transfer function $S(Z)$ of the secondary path FXLMS filter [3] and [5]. Equation 2 shows the error signal through the FXLMS algorithm in which $d(n)$ is the desired signal, $S(n)$ is the impulse response of the secondary path transfer function applied at time instant $n$, $w(n)$ and $X(n)$ are the coefficient and signal vectors of the filter $W(z)$, $\xi(n)$ is the mean square cost function, * denotes linear convolution [3].

$$e(n) = d(n) - S(n) * [w^T(n)X(n)]$$ \hspace{1cm} (2)

Then the error output of adaptive filter is given by equation 3 [3].

$$\hat{\xi}(n) = e^2(n)$$ \hspace{1cm} (3)

It is usual that the secondary path signal is distorted when high noise level is introduced in low frequencies. The solution is to modify the cost function in order to constrain the adaptive filter weights, as was proposed by the authors in [3] and shown in equation 4 that represents a leaky FXLMS algorithm, $\gamma$ is weight in the control, $\omega(n)$ and $X(n)$ are the coefficient and signal vectors of the filter $W(Z)$, $\xi(n)$ is the mean square cost function, $e(n)$ is the error, $\mu$ is step size, as is analyzed by authors [3].

$$\hat{\varepsilon}(n) = e^2(n) + \gamma \omega^T(n)\omega(n)$$ \hspace{1cm} (4)

in which

$$\omega(n + 1) = v\omega(n) + \mu x'(n)e(n)$$ \hspace{1cm} (5)

$$v = 1 - \mu \gamma$$ \hspace{1cm} (6)

and the Leakage Factor is inside the range $0 < v < 1$ . Leakage factor has information of stabilizing effect of the system.

**Feedforward and feedback FXLMS algorithm**

Feedforward algorithm works with the estimated signal in order to get a good compensation with the secondary path signal, Fig. 2. The feedback FXLMS ANC algorithm works with the error signal directly to achieve a better estimated signal. On other hand, the hybrid feedforward/feedback FXLMS ANC algorithm has the advantage of a quick response by the filter adaptation.

**Feedback FXLMS ANC algorithm**

Feedback FXLMS ANC algorithm works with the error signal directly to achieve a better estimated signal, as can be seen in Figure 3. On other hand, the hybrid feedforward/feedback FXLMS ANC algorithm has the advantage of a quick response by the filter adaptation.

**Hybrid controller**

The use of feedback and feedforward controller (hybrid algorithm, which is represented in Figure 4), for ANC, achieves to keep the system controlled in presence of external disturbances. However, the time it takes to generate the estimated signal to attenuate the primary signal (noisy wave) will be longer than the time needed by the feedforward controller, but shorter compared to the time needed by feedback controller.
3. SIMULATIONS

It is necessary to identify the system in order to design control algorithms for ANC, because of System identification provides coefficients to design adaptive control algorithm using the FXLMS filter. These algorithms, for simulations in this paper, were designed analyzing the codes of the authors Mr. Chernukhin and [7].

Simulation identification error is shown in figure 5, which was obtained by FXLMS algorithm execution, in order to generate adaptive coefficients.

Feedforward, feedback, and hybrid (feedforward/feedback) algorithms were designed and simulated in order to analyze their properties for ANC (reference algorithms designed by authors Mr. Chernukhin and [7]). Analysis for feedback and hybrid algorithms were described in [2].

Figure 3 shows execution results of ANC feedforward algorithm, the settling time was 0.07 s, which was analyzed from the noise residue curve. The simulation algorithm tries to create a control signal, that should be more similar possible to the input signal, as it was shown in Figure 3. Finally the output signal decreased its SPL in 23.22 dB [2].
Execution results, from ANC feedback algorithm, are shown in figure 5. The settling time was 0.5 s, which is analyzed from the noise residue curve; the output signal decrease its SPL in 10.39 dB.

Figure 5. ANC using feedback algorithm.

Figure 6 shows execution results of ANC using hybrid algorithm, the settling time was 0.4 s, that is analyzed from the noise residue curve; the output signal decrease its SPL in 16.02 dB.

Figure 6. ANC using hybrid algorithm.

Simulation results related to the noise signal attenuation for feedforward, feedback and hybrid algorithms were summarized in table 1 by [2], in which is shown that feedforward algorithm had shorter settling time in comparison to other algorithms as well as bigger SPL attenuation. However, hybrid algorithm had not short settling time, but feedback algorithm had longer settling time; nonetheless, hybrid algorithm was more stable to signal disturbances [2].

Table 1. Algorithms comparison

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>Decrement SPL (dB)</th>
<th>Settling time (s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Feedforward</td>
<td>23.22</td>
<td>0.07</td>
</tr>
<tr>
<td>Hybrid</td>
<td>16.02</td>
<td>0.4</td>
</tr>
<tr>
<td>Feedback</td>
<td>10.39</td>
<td>0.5</td>
</tr>
</tbody>
</table>

4. EXPERIMENTS

Experimental tests were accomplished by using adequate instrumentation and implementation of algorithms for feedforward control designed by Mr. Chernukhin. Nevertheless, characteristic curves response of the system was obtained by sinusoidal input signals, before the initial experimental tests. The input signals let to recognize the ranges of work where the control algorithms were operating properly, for some amplitude and frequency values [2].

In [2] is explained that for input signal amplitude 0.03V had a different behavior from the other two cases (0.02V and 0.01V as amplitude), which may not provide useful system information to generate an appropriate control signal. For this reason, the signal amplitude 0.02V maintained a similar behavior to changes in supply voltage, it increased in SPL when increased the supply voltages. Therefore, the value of 0.02V amplitude excitation signal at a frequency of 3 kHz was chosen from tested frequency range [2kHz to 5Khz], and voltage providing from power supply should be 2V. That is described by curves (characteristic curves) shown in figure 7.

Figure 7. Characteristic curves with 0.02V amplitude and 2V, 3V and 4V of Power Supply.

Topology for ANC in the ventilation system is represented in Figure 8. Feedforward control algorithm was executed by ADWIN system. Also the personal computer was involved in the ANC process for analyzing and visualization of every signal [2].
Information received from the reference microphone is processed by ADWIN system, it executes the control algorithm in order to generate an estimated signal, that is as similar as possible to the measured signal. The estimated signal is emitted by the loud speaker, which seeks to mitigate the primary signal, depending on a correct location of its phases. The difference of both signals (error) is measured by the error microphone, this signal is sent to ADWIN system to optimize the generation of the estimated signal [2]. Loudspeaker chosen were piezoelectric, by cause of their frequency range contains the available frequencies analyzed from characteristic curves identification [2].

FXLMS was used to identify the system and to find adaptive coefficients, that were used to execute the control algorithm. The best range of values chosen, in order to get ANC of the system described by [2], were: Sinusoidal input signal with 0.02V of amplitude at 3kHz, 2V proportioned by power supply to energize the system.

The identification algorithm to be executed by ADWIN was designed by the author [2], this code was executed with different frequencies values, in order to analyze the performance of the identification algorithm; Notwithstanding, the primary signal and estimated signal were displaced in 90 degrees approximately. After the system identification a feedforward control algorithm was adopted from algorithms designed by Mr. Chernukhin which was implemented for using with ADWIN. Settling time in the system was 0.45 s, when ANC by feedforward algorithm was executed in frequency range value identified, also the sound signal decrement 8.15 dB as is shown in Figure 9 and Figure 10 [2].

Table 2 summarize the range of values analyzed from the characteristic curves in which is shown the best settling time and noise reduction (in dB).

<table>
<thead>
<tr>
<th>Frequency (kHz)</th>
<th>decrement SPL (dB)</th>
<th>Settling time (s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>4.07</td>
<td>0.43</td>
</tr>
<tr>
<td>2.5</td>
<td>5.83</td>
<td>0.53</td>
</tr>
<tr>
<td>3</td>
<td>5.84</td>
<td>0.040</td>
</tr>
<tr>
<td>4</td>
<td>4.95</td>
<td>0.037</td>
</tr>
<tr>
<td>5</td>
<td>9.58</td>
<td>0.018</td>
</tr>
</tbody>
</table>

Table 3. Identification results

Table 3 shows this range at 3 kHz (in this frequency value was identified the char-acteristic curves, showed in figure 7), settling time in the system was 0.45 s, when ANC by feedforward algorithm was executed, also the sound signal decrement 8.15 dB as is shown in figures 9 and 10.
<table>
<thead>
<tr>
<th>Frequency (kHz)</th>
<th>Decrement SPL (dB)</th>
<th>Settling time (s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>1.6</td>
<td>11.3</td>
</tr>
<tr>
<td>2.5</td>
<td>7.66</td>
<td>3.67</td>
</tr>
<tr>
<td>3</td>
<td>8.15</td>
<td>0.45</td>
</tr>
<tr>
<td>4</td>
<td>10.47</td>
<td>0.33</td>
</tr>
<tr>
<td>5</td>
<td>2.10</td>
<td>9.33</td>
</tr>
</tbody>
</table>

Table 3. Control algorithm comparison at 2kHz, 2.5kHz, 3kHz, 4kHz, 5kHz of primary signal frequency

5. SUGGESTIONS

It is suggested to design an online model in order to identify the system by FXLMS algorithm, as an internal function of the main algorithm. The error signal may be defined as an output condition with dependence of changes in the step size $\mu$, which is explained by [2].

The adaptive coefficients could be obtained, by the algorithm proposed, before to control loop. The criterion to optimize this process depends of the time assigned to execute the LMS as a function inside the main of the algorithm. The objective of this algorithm is to calculate the adaptive weights online, periodically repeating the system identification process inside of the main algorithm. The algorithm is executed till the condition, predefined by user, is not satisfied; this condition could be determined by desired error as a function of the changes in step size [2].

6. CONCLUSIONS

Characteristic curves were found to define the best range of work to estimate the antinoise signal, for which reason the primary signal was measured by the system.

Piezoelectric loudspeakers were used as anti noise actuator, because of their range of work contains the available frequency range obtained in characteristic curves identification, which is [2 kHz to 5 kHz].

A ventilation system prototype described by [2] was used in order to design ANC algorithms, that were simulated by Matlab and compared with experimental responses of prototype designed.

Feedforward algorithms and strategies for ANC were designed, implemented and experimentally tested for the prototype of the duct ventilation system based on the algorithms implemented by Mr. Chernukhin, [2] and [7].

Feedforward, feedback and hybrid algorithms were designed with regard to simulate ANC [7] and [2].

7. ACKNOWLEDGMENT

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8. REFERENCES


