ACTIVE NOISE CONTROL IN FRONTAGES OF BUILDINGS

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ABSTRACT

Active noise control is an interesting alternative to the problem of frontage insulation in low frequency, where passive solutions are less efficient. The major objective of this research project is to consider, in an integrated way, the contribution of active insulation methods to decrease harmful effects of noise caused by the traffic growth (aircraft, railway, road).

Measurements in the vicinity of Liège Airport have shown that the main weak parts of the building insulation are the windows (including the frame), the roof, the shutter boxes (if any) and of course the air supply system which is intended to bring fresh air, for example in the sleeping rooms. Our project is presently focused on ventilation ducts and applications of active systems in a shutter box. Concerning the air supply system, a prototype was designed with the requirement of being incorporated in a frontage.

Acoustical transducers are used and the involved techniques are based on feedforward algorithms (FXLMS) implemented on a DSPACE system.

This work has been performed in the framework of the ISACBAT project funded by the Walloon Region of Belgium.

1. INTRODUCTION

The growth of transport activities in our industrial and leisure societies generates noise with increasingly harmful effects. Passive methods currently offer interesting solutions for noise insulation at medium and high frequencies. Unfortunately, they are less effective in the lowest frequencies of the audible spectrum.

It is in this frequency range that the active control of noise (ANC) can take over [1]. The principle of ANC stated in 1933 by P. LUEG [2] consists in making interfere in a destructive way two waves of equal amplitudes, but of opposite phases. Limited by technology, this technique took a significant rise only these last years with the advent of increasingly powerful DSP (Digital Signal Processors).

However, ANC is not usually applied in building acoustics. The rare products available on the market are in the field of ventilation ducts. Nowadays, this technique can be applied with minimal equipment, at a reasonable cost, while offering possibilities of low frequency insulation for which a passive solution would require much more significant means (excessive weight and obstruction).

The ISACBAT project, founded by the Walloon Region, wants to benefit from this concept by considering in an integrated way the contribution of ANC in dwelling’s insulation. This project considers some weak elements of the frontage’s insulation: the roofs, the ventilation duct and the double walls. The goal is to develop active systems (prototypes) whose efficiency will be demonstrated on a real site, close to the airport of Liège-Bierset.

2. SETUP

The control system is composed of:
- a development board DSPACE DS1104;
- three analogical filters for anti-alias and reconstruction (Butterworth, order 6, $F_C = 600$Hz);
- two microphones associated with their pre-amplifiers;
- two loudspeakers connected to the same channel of an amplifier and placed on lateral sides of the duct.

The development board offers instantaneous possibilities of control through Simulink/RTW, the algorithms being implemented by means of diagrams blocks instead of direct programming (C or assembler) for rapid prototyping. Moreover, it comprises a significant number of inputs-outputs which are essential for an eventual MIMO control.

The ventilation duct has a rectangular cross-section of about 600cm². It is conceived to be inserted in a frontage wall, with an air input outside and the other end inside the building. The interior surfaces were covered with absorbing materials (foam) which brings passive attenuation at medium and high frequencies, active control taking care of the remainder.

3. CONTROL

The objective of this project is to reach a noise reduction between 50 Hz and 250Hz. Considering broad
The techniques resting on a pure feedback are not sufficient \[3\]. The random nature of noise prevents any use of “a posteriori” predictive algorithm: it’s impossible to predict the instantaneous pressure at the position of the control loudspeakers without preliminary information. Tests confirmed this statement since noise reduction obtained with feedback control is only about 3dB. This is why the algorithm generally used under these conditions is of feedforward type \[4\]. The shape of the ventilation duct lends itself particularly well to this method thanks to its longitudinal character: the reference microphone can be easily placed at the air inlet side of the duct, the control loudspeakers halfway and the error microphone at the air outlet (see Figure 1).

![Figure 1. Feedforward control](image)

The main iteration of the FXLMS algorithm consists in computing equation (1) & (2), where \( W(n) \) is the FIR filter coefficients vector, \( \mu \) the convergence coefficient, \( C(n) \) the secondary path transfer function, \( R(n) \) the reference signal, \( X(n) \) the filtered reference signal and \( E(n) \) the error signal.

\[
W(n+1)=W(n)+2\mu E(n)X(n) \tag{1}
\]
\[
X(n)=C(n) \ast R(n) \tag{2}
\]

where \( \ast \) denotes the convolution symbol.

A restriction for this kind of systems is due to the fact that the time of sound propagation in the duct between the reference microphone and the control loudspeaker must be higher than the processing time inherent to the controller, including the anti-alias filters delays and possibly other components (signal conditioners) delays. A first attempt to control noise with loudspeakers in the middle of the duct showed poor performance. A simple study of delays suggested a displacement of the loudspeakers towards the end of the duct. With this solution:

- a better stability was achieved since the influence of the feedback is reduced by the increase in the distance between the reference microphone and the control loudspeakers;
- a faster convergence was reached thanks to the proximity between the error microphone and the loudspeakers.

When the system is started and noise is present at the air inlet, the algorithm starts to converge, more or less quickly according to the value of the coefficient \( \mu \), until a minimum of noise is reached at the exit. A first problem appeared for non-stationary signals like aircraft noise. With usual values of \( \mu \), the time of convergence was between one and two minutes, which makes the system unable to control noise instantaneously. A solution consists in "training" the algorithm (offline with white noise) until obtaining \( W_{optimal} \), which will then be assigned as an initial condition \( W(0) \) to the
controller. With this method, an effective control is immediately obtained.

A second problem occurs in the absence of noise. Indeed, the microphones only collect insignificant and uncorrelated background noise. The algorithm then slowly deviates from its optimal position, until having a quasi random $W$. Often, the algorithm diverges and produces a noise while saturating, until it is stopped. The solution, in this case, consists in calculating the noise level at the reference microphone and allowing the adaptation only when significant noise is present. We thus keep $W_{optimal}$ while benefiting from the adaptability, which is necessary to take into account the ageing of some elements or the modification of some system’s characteristics. For example, we’ve noticed that the state of the protective grids (open-closed) at the air inlet and outlet was perfectly compensated by the algorithm. The algorithm was further enhanced by loading the optimal filter after each plane passing. Indeed, in some cases, the algorithm didn’t converge towards the optimal attenuation. It can be due to uncontrollability at some frequencies, probably the lowest ones. After some time, the control was not efficient anymore. Thanks to this resetting of the filter, we keep an optimal attenuation all the time.

A third problem appeared when the prototype was tested under real conditions, i.e. when inserted into a wall. The vibration of the house, when a truck is passing nearby for example, creates a peak at 12Hz in the reference and error signals whose amplitude is much greater than the average signal. This prevents the controller to converge efficiently. A digital high-pass filter was then placed in the error signal path. The ideal solution consists in a high-pass filtering of both reference and error signals but this cannot be done for causality reasons.

Some tests of multi-channel control were carried out. Instead of driving the two loudspeakers with the same source, a LMS controller was assigned to each of them. A second test consisted in placing a second pair of loudspeakers a few centimeters away from the existing ones towards the beginning of the duct and driving each pair with its own controller. These tests didn’t bring further improvements (less than 1dB).

4. RESULTS

The prototype to be tested is placed in a rigid wall between two mechanically and acoustically isolated reverberation rooms.

A broadband noise source generates a diffuse sound field in the emission room. Estimates of the space and time average mean square sound pressure in emission room 1 and in reception room 2 have been performed, as specified in ISO 140-3 [5] (see figure 3).

![Figure 3. Insulation measurement in reverberant rooms](image)

The aim is to determine the gain in insulation obtained by active noise control. For each experiment, two insulation measurements were thus carried out: with and without control.

The efficiency of active noise control was first studied while visualizing coherence function between the reference and error signal [6], and comparing with the sound reduction obtained at the error microphone (see figure 4).

![Figure 4. coherence and error signal](image)

These results show that the limits of performance are reached: the control is efficient in the bandwidth 50-450 Hz, where the coherence is close to its maximum. A global attenuation of 23.3 dB is obtained at the error microphone in the bandwidth [20-1kHz].

We can note that the anti-alias and reconstruction filters limit the operating range of active noise control at the frequencies less than 600 Hz. Beyond that frequency, active control is not significant and does not introduce additional noise at all.

Other tests were performed for two different kinds of noise generated in the source room: broadband white...
noise and aircraft noise. Results are presented in figures 5 & 6, with and without control.

In each case, the global attenuation obtained by the ANC system is about 10 dB. It is interesting to note the stability and the robustness of the algorithm which allowed effective control throughout the whole experiment (in particular in the case of aircraft noise). This result was also confirmed by varying the distance of the measurement microphone to the wall, as illustrated in figure 7.

Finally, the sound reduction index was measured in order to characterize the structure studied (wall + active air duct). These results are reported in figure 8 and compared to the value of sound reduction index before insertion of the prototype.

The decrease of the sound reduction index after insertion of the ventilation duct is unavoidable (a loss of 8 dB in Rw was observed). This loss in insulation is particularly visible at low and high frequencies. However, ANC is able to recover the SRI at low frequencies (Rw = 56 dB). These results are very encouraging and clearly meet the objectives of the project.

5. CONCLUSION

From ANC point of view, the prototype now seems to be mature with very satisfactory performances. The addition of optimized passive insulation elements should still increase these performances in insulation above 2kHz without influencing the behavior of the active controller.

ACKNOWLEDGEMENTS

This project was supported by the Walloon Region of Belgium / convention n° 114890.