original article

Active noise cancellation of low frequency noise propagating in a duct

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ABSTRACT

Aims: The object is to find a manner of reduction primary noise from a source by an electro acoustic device that cancels unwanted sound by generating an anti sound (anti noise) of equal amplitude and an opposite phase is described as an active noise control system.

Materials and Methods: In this method, the primary noise is acquired via a microphone, and the anti noise propagates with the same amplitude and the reverse phase at the primary noise path via a speaker to cancel the original noise. The effectiveness of cancellation of the primary noise depends on the accuracy of the amplitude and phase of the generated anti noise.

Results: The results present a noise reduction of 16 dB until 20 dB overall. A change in the system setup and noise power can reach a noise reduction up to 25 dB. Perhaps, this is due to the background noise and primary noise differences that create a powerful anti noise for canceling the original noise.

Conclusions: With regard to the wide range of frequencies of different noise sources, having optimized circumstances in the duct, microphone location on the duct body or even the distance of the speakers may be important in signal processing, noise sampling and anti noise production.

Key words: Active noise control, digital signal processing, duct, low frequency noise, matrix laboratory simulation

INTRODUCTION

Using adaptive filters to control noise is not a new idea. Creating a copy of the noise and using it to cancel the original or active noise cancellation dates back to the early part of this century. In the mid 1970's there was a major step forward with the application of adaptive filters to generate the anti noise. Now digital computer technology has evolved to a point where cost-effective digital signal processing (DSP) microcomputers can perform complex calculations involved in noise cancellation.[1] An electro acoustic device that cancels unwanted sound by generating an anti sound (anti noise) of equal amplitude and opposite phase is described as the active noise control (ANC) system. The original, unwanted sound and the anti noise acoustically combine, resulting in the cancellation of both sounds.

In this article as we are considering low frequency noise (LFN) propagating in a duct as a single directional noise with a low variation in frequency it is taken for analysis. The noise signal propagating in a duct is fed into a computer through a data acquisition card (DAC). This signal is analyzed by the

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computer and generates a signal with the same amplitude but with a 180 degree phase shift. This signal is fed into the duct through the DAC. Two signals will be cancelled and the original noise eliminated based on the sound interference fundamentals. The National Instrument's data acquisition card is used for input of the noise signal from the duct via microphones. The signal is analyzed by the DAC which generates the cancellation signal based on the MATLAB software programs. The cancellation signal is given to the duct via the DAC by an amplifier and a speaker.[2]

MATERIALS AND METHODS

Filter length and delays

The effectiveness of the cancellation of the primary noise depends on the accuracy of the amplitude and phase of the generated anti noise. Some factors have to be considered when designing the active noise control system. The sampling rate, filter length and delays in the microphones, loudspeakers, and electronics must be considered to have an accurate working system. Real time digital signal processing requires that the processing time t be less than the sampling

$$t < T = \frac{1}{f_s}$$

Where f_s is the sampling rate and f_M is the highest frequency of interest.

$f_S \geq 2f_M$

The sampling rate can be expressed in terms of the physical distance and the ability of the system to resolve this distance at room temperature.

Active noise control system design

Two signals have been generated by a computer using MATLAB programming. One signal is inverted at the phase shift, as the anti noise of original signal.[3] Then the two signals are fed to the speakers through a DAC as shown in the Figure 1. Primary noise signal, anti noise and the residual noise are shown in this figure.

Feedforward active noise control system

There can still be some errors due to the time delays at different parts of the system. Therefore, the whole noise will not be canceled. Thus, a feedforward system is used to eliminate the entire noise. The feedforward ANC system is the main technique of noise cancellation for duct and ventilation systems. Figure 2 demonstrates a model of the feedforward ANC system in a duct.

Broadband feedforward system

A considerable amount of broadband noise is produced in ducts such as exhaust pipes and ventilation systems. A reference signal x(n) is sensed by an input microphone close to the noise source before it passes to a loudspeaker. The noise canceller uses the reference input signal to generate a signal y(n) of equal amplitude but 180 degrees out of phase. This anti noise signal is used to drive the loudspeaker to produce a canceling sound that attenuates the primary acoustic noise in the duct.[4]

Broadband active noise control can be described in a system identification framework, as shown in Figure 3. Using a digital frequency domain representation of the problem, the ideal active noise control system uses an adaptive filter W(z) to estimate the response of an unknown primary acoustic path P(z) between the reference input microphone and the error microphone [Figure 3]. The z transform of e(n) can be expressed as:

$E(z) = D(z) + Y(z) = X(z)[P(z) + W(z)]$

Where E(z) is the error signal, X(z) is the input signal, and Y(z) is the adaptive filter output. After the adaptive filter W(z) has converged, $E(z) = 0$. The above equation becomes:

$W(z) = -P(z)$

Then it implies that,

$y(n) = -d(n)$

Therefore, the adaptive filter output y(n) has the same amplitude but is 180 degrees out of phase with the primary noise d(n). When d(n) and y(n) are acoustically combined, the residual error becomes zero, resulting in the cancellation of both sounds based on the principle of superposition

RESULTS

The primary noise signal is generated by the MATLAB software and moved from the computer inside the duct through a DAC. The reference microphone takes it again to process and simulate it as a reverse signal [Figure 4]. We

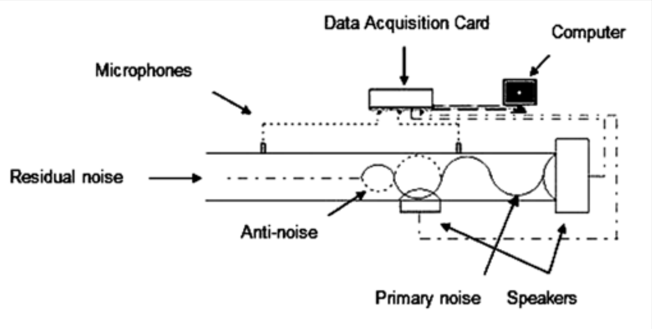

Figure 1: ANC system for canceling a generated signal in a duct

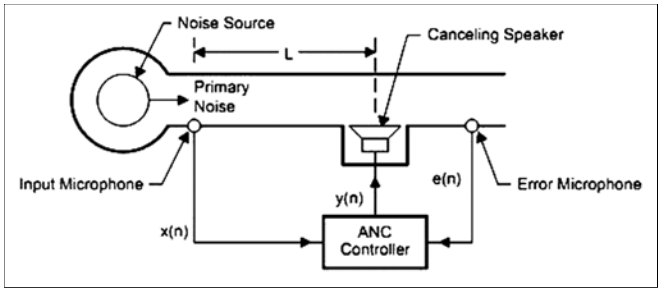

Figure 2: Feedforward ANC system in a duct

limit the LFN signal to 500 Hz frequency. An adapting filter with the least mean square algorithm is used in order to adapt the original noise and simulated anti noise.[5] The error microphone samples the residual signal of the cancellation action on the other side of the duct. The waveforms of noise and anti noise produced by MATLAB simulation in an equal amplitude and reversed phase are presented in Figure 5.

This is an experimental study done by using the active noise control method. Noise is produced by a speaker at the interested frequencies which will then propagate in a Plexiglas duct with the dimensions of 20 × 20 × 150 cm. The noise is processed by the digital signal processing; the anti noise propagates along the duct to cancel the primary noise as a response of DSP. The canceller signal is transferred to the duct by a speaker through the DAC output. The system needs to treat itself in a proper manner to cancel the entire noise and minimize the system error.

The conclusions show an attenuation of 20 dB overall and a peak of 25 dB for the interested frequency. The original signal amplitude at the error microphone is presented as a function of time in seconds before cancellation by the anti noise produced at the active noise control system. An experiment is constructed to demonstrate reduction in pure tones of noise and the normalized least mean squares (NLMS) algorithm has been implemented to build the anti noise in this study. The least mean squares (LMS) algorithm is a feedforward algorithm that uses an independent reference signal related to the tone of the primary source. Figure 6 shows the ANC system application in a duct to cancel a pure tone LFN of 500 Hz frequency.

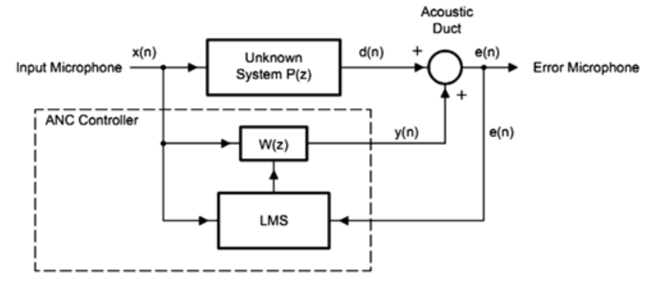

Figure 3: Block diagram of broadband feedforward ANC system in an acoustic duct

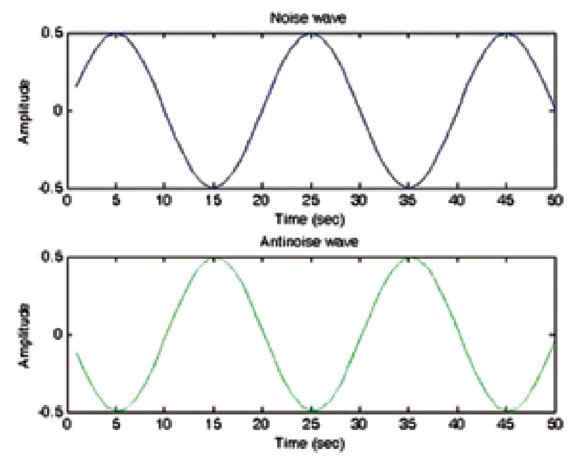

Figure 5: Spectra of waveforms noise and anti noise

The typical power level spectra for the error sensor microphone with original and attenuated signals are presented in Figure 7. From the figure, a reduction of 20 dB can be seen for this frequency in the duct.

DISCUSSION

As it was explained, the effectiveness of the cancellation of the primary noise depends on the accuracy of the amplitude and phase of the generated anti noise. The basic principle of the broadband feedforward approach is that the propagation time delay between the upstream noise sensor (input microphone) and the active control source (speaker) offers the opportunity to electrically reintroduce the noise

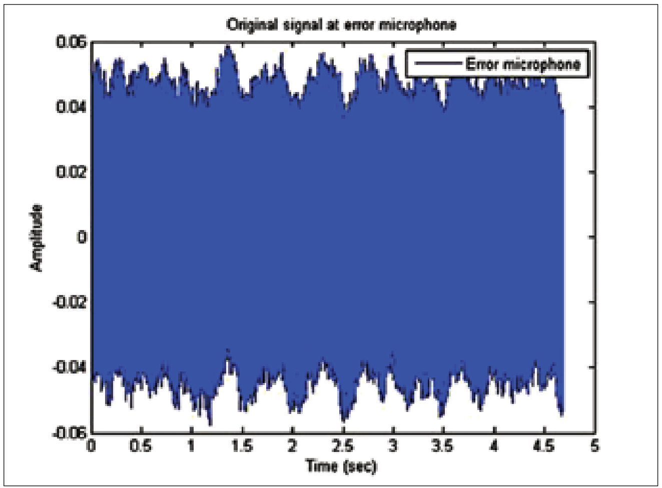

Figure 4: Spectrum of original noise before cancellation by ANC system

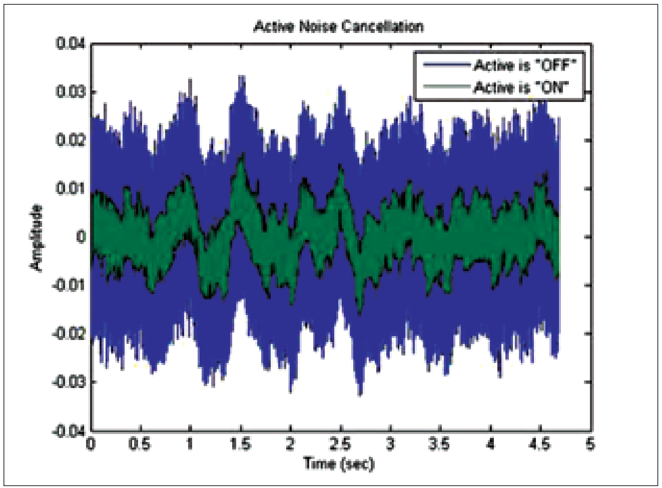

Figure 6: Reduction of noise as a function of time in seconds using ANC procedures

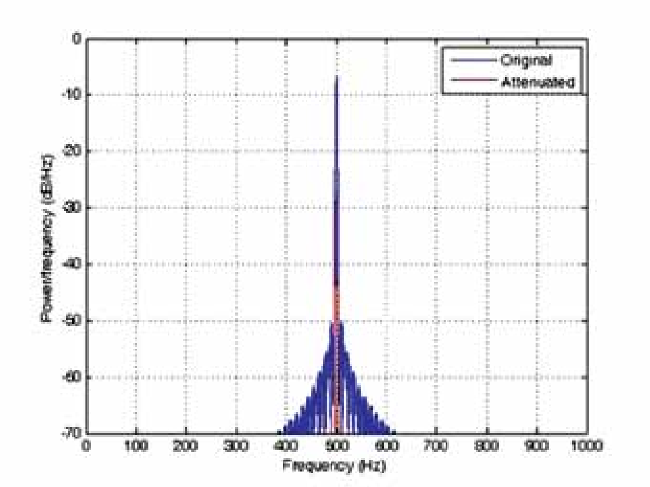

Figure 7: Power level reduction of 500 Hz frequency pure tone at a duct

at a position in the field where it will cause a cancellation. The spacing between the microphone and the loudspeaker must satisfy the principles of causality and high coherence, meaning that the reference must be measured early enough so that the anti noise signal can be generated by the time the noise signal reaches the speaker. Also, the noise signal at the speaker must be very similar to the measured noise at the input microphone, meaning the acoustic channel cannot significantly change the noise. The noise canceller uses the input signal to generate a signal y(n) which is of equal amplitude and is 180 degrees out of phase with x(n). This noise forms the output to the loudspeaker and is used to cancel the unwanted noise.

With regard to a wide range of frequencies of different noise sources, providing optimized circumstances in the duct design and the microphone's location on the duct's body or even the distance of speakers maybe important in signal processing, noise sampling and producing anti noise. In the present study a gap of 0.90 m between the two microphones and a distance of 0.30 m from the speaker to the reference microphone can be the reasons for eliminating LFN in the area of 500 Hz frequencies. It is clear that increasing the gap which means increasing the duct's length is an opportunity for optimizing frequency sampling at a rate of 8000 Hz with more reduction than 25 dB, It is different from what has been resulted for power density in this study, as shown at Figure 7. It means that the best situation for the reference microphone is to increase the distance from the source's speaker to a minimum of 30 cm. However, optimized cancellation for frequencies at the area of 250 Hz and less will occur with a duct at a length of more than 3 m and proper location of the microphones on the duct. It is concluded that the active noise cancellation system setup for producing anti noise has equal amplitude and 180 degrees out of phase to cancel the primary generated noise produced by MATLAB function generator. The two noises were cancelled according to the principle of destructive interference.

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