ACTIVE CONTROL OF NOISE PROPAGATING IN DUCTS

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Abstract

Active Noise Control belongs to the fields of control Systems Engineering and Signal Processing. This project gives the model of eliminating noise propagation in Ducts. In a duct the signal is limited to a one directional Signal. Noise signal propagating in a duct is taken to a Computer through a Data Acquisition Card (DAC). This signal is analysed by the Computer and generate a signal with the same amplitude but 180 phase shift. (i.e. Inverted Signal). This signal is given to the duct through the DAC Card. By the principle of destructive interference these two signals will be cancelled and eliminate the noise. To input the noise signal use a National Instrument's Data Acquisition Card and the signal is input by a Micro phone. Signal is analysed and generate the cancellation signal by using a software written by using Visual Basic Language. Cancellation signal is given to the duct through the DAC Card with a amplifier and a Speaker.

1. INTRODUCTION

Active Noise Cancellation is not a new idea. Creating a copy of the noise and using it to cancel the original dates back to the early part of this century. The first systems used a simple " delay and invert" approach and showed some promise, but the variability of real world components limited their effectiveness.

In the mid 1970's a major step forward took place with the application of adaptive filters to generate the anti-noise. This greatly enhanced the effectiveness of the systems as they could continuously adapt to changes in their external world as well as changes in their own components.

A second breakthrough in the mid 1970's was the recognition that many noise sources, particularly those produced by man-made machines, exhibit periodic or tonal noise. This tonal noise allows a more effective solution as each repetition of the noise is similar o the last and the predictability of the noise allows creation of an accurate anti-noise signal.

Practical application of this technology still had to wait as the electronic technology available at that time was not sufficient for implementation of Active Noise Cancellation systems. Now digital computer technology has evolved to the point where cost effective Digital Signal Processing (DSP) microcomputers can perform the complex calculations involved in noise cancellation.

This technology advance has made it feasible to apply Active Noise Cancellation to previously insoluble problems in low frequency environmental noise at a reasonable cost. In this project since we are considering a noise propagating in a duct a single directional noise with a low variation of frequency is taken in to analysis.

The problem of attenuating noise in a duct can be best understood by considering the scheme depicted in Fig. 1.The noise w (here generated by a speaker) is measured by the microphone y. The control goal is then to generate a control signal u, in order to minimize the sound pressure level at z. Different controller configurations have been suggested, i.e. feed-forward, feed-back and combinations of both. However, since a minimal distance between u and z has to be kept to fulfil the plane wave assumptions, feedback schemes have little effect in general.

2 DESIGN OF ANC SYSTEM

Some factors have to be considered when designing of an ANC System.

Sampling Rate and Filter Length controller must complete the entire signal the task of the controller is to estimate precisely the delay and any amplitude changes that occur as the unwanted noise travels from the input microphone to the loudspeaker. This includes delays in the microphones, loudspeakers, and electronics. The processing task before the primary noise arrives at the loudspeaker. Real-time digital signal processing requires that the processing time t be less than the sampling period T. i.e.:

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

where fs is the sampling rate, which must be held high enough to satisfy the Nyquist criterion. i.e.:

$$f_s \geq 2f_M$$

where f_M is the highest frequency of interest—approximately 500 Hz for most practical ANC

applications. This yields a minimum sampling rate of 1 kHz and a maximum processing time of 1 ms. 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. The sampling resolution can be expressed as:

$$\Delta_s = \frac{c_0}{f_s}$$

where c_0 is the speed of sound in air, which is 343 meters per second at 75°F. The modeling of the primary plant is done in the time domain using the FXLMS algorithm or the filtered-U RLMS algorithm. The number of direct weights (W(z) in the FXLMS algorithm and A(z) in the filtered-U RLMS algorithm) times the sampling resolution determines the model length in time or an equivalent distance. i.e.:

meters. where l is the length of duct from the input microphone to the canceling loudspeaker that can be modeled by an adaptive filter. For example, N _ 64 and a sampling rate of 2 kHz results in a 32-ms model, which corresponds to a duct length of 10.976

Coherence Function

Input microphone will appear at the loudspeaker after a delay. Unfortunately, both the input and the error microphones detect the primary noise plus the self-generated flow noise of the air passing over the surface of the microphone. Therefore, flow noise and turbulent pressure fluctuations at the microphones can limit cancellation effectiveness. This problem is rather significant for ducts of low sound pressure levels such as those in an air conditioner. velocity, using multiple distributed sensors, and by good fluid-mechanical design to minimize localized turbulent noise.

This project consist of three parts

1) Cancellation of a known Signal
2) Cancellation of a signal after analysing it
3) Minimising the errors by using a feedforward system.

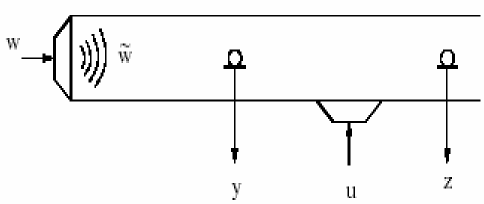


Figure 1: General system set-up.

3. APPROACH

3.1 Cancellation of a known Signal

Here the two signals have generated by a computer using Visual Basic Programming. One signal is the inverting signal of the other. Then two signals are given to speakers through a DAC Card as shown in the Fig2. Two signals will be cancelled.

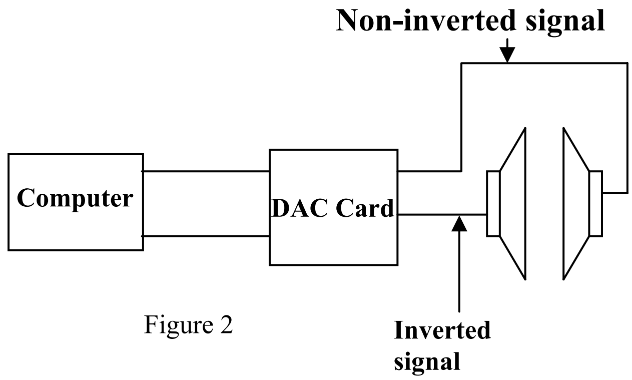


Figure 2

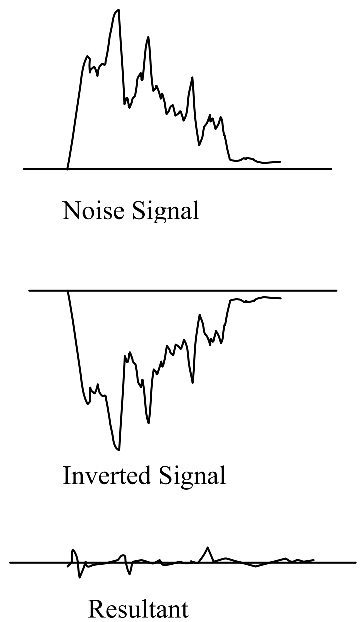


Figure 3

Noise signal, inverted signal and the resultant signal are shown in Fig 3.

3.2 Cancellation of a Signal After Analysing it

Here the noise signal is taken to the Computer from a duck through a DAC card, Analyse it

Noise signal is generated from a repetitive noise generator. AC Plant is a good example. Here in our project we limit the noise signal to less than 1khz. Basic method of capturing this signal is by using a Microphone. But microphone itself does not pick the exact wave form. There is an error. To eliminate this, the best method is to use other type of sensor such as a tachometer to sense the repetitive signal which is a proportional function of a noise signal.

This signal is given to the DAC card and sample it. Sample frequency should be more than 2khz. This sample is put in to an array. Then this array is shifted by time delay which is equal to time taken to propagate sound from source to output speaker.

This array is inverted and multiplied by the gain. Gain is a constant which is varied by the type of duct used. This has to be hand varied. Then the cancellation signal is transferred to the Duct through the DAC out put and the speaker. Since this doesn't consist of a minimise the error total noise is not cancelled. Proper method is needed to minimise this error. Feedback or Feedforward system is needed.

3.3 Using a Feedforward or Feedback System

When the signal is cancelled as shown in fig 2 still there can be some errors due to the time delay. i.e. Therefore the whole noise will not be cancelled. Therefore it needs a feedforward or a feedback system to eliminate the entire noise .Feedforward ANC systems are the main techniques used today. Systems for feedforward ANC are further classified into two categories**:**

- Adaptive broadband feedforward control with an acoustic input sensor

- Adaptive narrowband feedforward control with a non-acoustic input sensor

Here in our Project, we use the Narrowband Feedforward System since propagation of a noise is a uni-directional and the frequency of the signal is not highly varied.

3.3.1 The Narrowband Feedforward System :

In applications where the primary noise is periodic (or nearly periodic) and is produced by rotating or reciprocating machines, the input microphone can be replaced by a no acoustic sensor such as a tachometer, an accelerometer, or an optical sensor. This replacement eliminates the problem of acoustic feedback (described in the subsection the block diagram of a narrowband feedforward active noise control system is shown in Figure 4. The no acoustic sensor signal is synchronous with the noise source and is used to simulate an input signal that contains the fundamental frequency and all the harmonics of the primary noise. This type of system controls

Harmonic noises by adaptively filtering the synthesized reference signal to produce a cancelling signal. In many cars, trucks, earth moving vehicles, etc., the revolutions per minute (RPM) signal is available and can be used as the reference signal. An error microphone is still required to measure the residual acoustic noise. This error signal is then used to adjust the coefficients of the adaptive filter.

Here the error is fed in to the Array. And this array is added with the array taken from the Noise signal in previous part. Since this output consist of Discrete

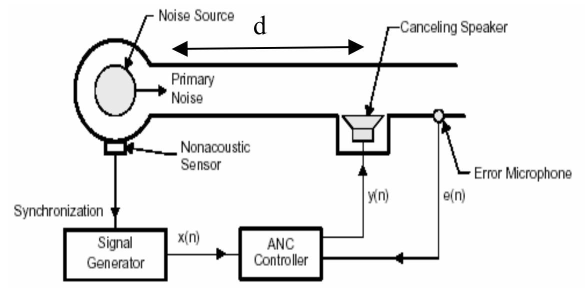


Fig. 4 Narrowband Feedforward ANC System

Voltages, required cancellation signal is not generated. Therefore a Filter is needed. Easiest method is to use a filter function given in the DAC card. But there will be a time delay and inaccurate. Therefore the output is transferred to the Speaker through a Filter. This filter should be a FIR (Finite impulse response) filter.

4 ALGORITHM : The principle of feedback

microphone. The basic idea of this algorithm is to estimate the primary noise and to use this as the reference input for the adaptive filter.

Cancellation Signal is given by

$y(\mathbf{n}) = \mathbf{G}\,[\theta(\mathbf{n}) - x(\mathbf{n}+\tau)]$

Where

$x(n)$: Source Signal

$\theta(n)$: Error Signal

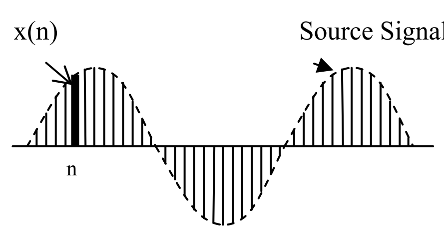
τ : Constant $\tau = v/d$

Where

v : Velocity of Sound

d : Distance between source sensor and the output speaker.

n : pulse index of the waveform at a specific time



5 CONCLUSION

We have successfully completed the stage one and stage two of this project. Two noises were cancelled according to the principle of destructive interference.

In step 3 feedforward system is used to minimise the error. We have almost reached to the target. But there can be some limitations.

1) We use normal speaker. It consists of some error.

2) This system cannot apply to noise signals which its frequency is highly varied.

3) This system can only be applied to uni directional system. But some times sound is reflected by the duct. These reflected signals cannot be analysed from our system since our DAC Card is not a multi channel system.

4) When the environment noise is added during the analysing period of the signal (by the computer), it does not response to it.

5) Sine DAC Card consists of very few input /output channels this cannot be used for multiple duck systems.