

## Active Noise Cancellation

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### ABSTRACT

In this work, we develop a new approach to active noise cancellation. Theoretically, we use a 180 degree phase shift of the noise, i.e. generate an anti-noise, to cancel the original signal. Here, we have designed FIR filter which is driven by the input signal and has the noise signal as the feedback, and output of which is the required result. The active noise control system contains an electro-acoustic device that cancels the unwanted audio signal by generating an anti-noise of equal amplitude and opposite phase. The original, unwanted sound and the anti-noise combine acoustically, resulting in the cancellation of both sounds. The effectiveness of cancellation of the unwanted signal depends on the accuracy of the amplitude and phase of the generated anti-noise.

**Keywords:** Active Noise Cancellation, LMS algorithm, FIR filter, Simulink, DSP kit, Silentium.

### I. INTRODUCTION

Communication is an integral part of daily life for conveying information. But, what comes with the information is various types of unwanted signals, called as NOISE. Noise pollution or Noise disturbance is the disturbing or excessive noise that may harm humans as also animals. The source of most outdoor noise worldwide is due to machines and transportation systems i.e. motor vehicles, aircraft, trains etc. (Refer Fig. 1.1). Outdoor noise can often be summarized by the term "Environmental Noise". Noise pollution can also be caused due to inefficient infrastructural facilities in cities, for instance, side-by-side industrial and residential areas. There are various sources of indoor noise such as those caused by machines, building activities, and music performances, especially in some workplaces. Noise-induced hearing loss can be caused by outside (e.g. trains) or inside (e.g. music) noise.

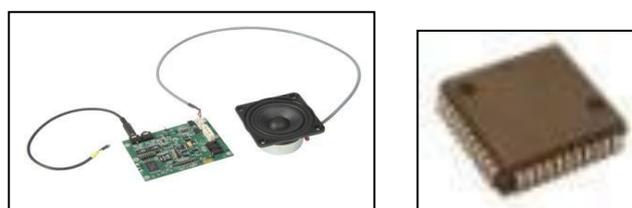


Fig. 1.1 Sources of noise

In this paper, we have used Simulink model to get the desired noise cancelled signal.

### II. REVIEW OF THE RELATED LITERATURE

The idea for this project was inspired from the SILENTIUM products (Ref. fig. 2.1) & 'Noise Cancelling Headphones' by Sony, BOSE, Panasonic etc., which are becoming very popular in recent years. Noise cancellation makes it possible to enjoy music without raising the volume excessively.



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Fig. 2.1 Silentium Products (Ref. [6])

Noise cancellation is of major concern in the acoustic industries also. The noise of blowers in a ventilation system is carried over a few kilometres through a duct. This affects the acoustic environment. Use of passive silencers and baffles alone does not solve this problem as only the high frequency noise components are cancelled out, while the lower frequency components are reverberated inside the ventilation duct. Accompanying the passive silencers, an active noise silencer is required in order to cancel these noise components.

However, in India, only a few of such projects are implemented practically (on hardware). If such products are imported, they cost very high (approx. ₹1 lakh). This definitely does not help the Indian economy. Hence, there is an ardent need of manufacturing such product in India, after consideration of factors such as the effectiveness, reliability and the most importantly, its cost.

### III. BLOCK DIAGRAM

Refer Fig.3.1. for block diagram

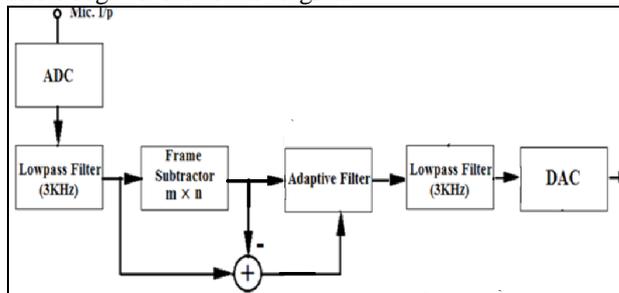


Fig 3.1. System Block Diagram

Fig. 3.1 Represents the block diagram of the system to be implemented. The system input is through a Microphone and is passed to the Analog to Digital Converter (ADC), which converts the analog input to digital samples at a sampling rate of 8 kHz.

The first lowpass filter designed based on FIR algorithm is used to limit data frequency below cut-off frequency. Frame Subtractor block is used to find the difference between successive frames, i.e.,  $Z[n+1]-Z[n]$ . The frame generated is a vector of size "m x n". This gives us the reference signal. The original and the reference signal are added out-of-phase to obtain the noise signal, which is then used as feedback for the adaptive algorithm. The adaptive filter block adapts the filtered weights as per the feedback it receives.

The second lowpass filter is used as a precautionary measure to avoid any addition of internal noise due to hardware limitations. The Digital to Analog Converter (DAC) converts the digital samples of output to analog output at a reconstruction sampling rate of 8 kHz.

### IV. LIST OF COMPONENTS

#### A. HARDWARE REQUIREMENTS

- DSP kit – TMS320C6713
- Microphone
- Speaker
- Power supply

#### B. SOFTWARE REQUIREMENTS

- Matlab R2014a
- Simulink
- Code Composer Studio v3.1

### V. SYSTEM DESIGN & IMPLEMENTATION

Refer Fig.5.1 for System Design diagram

Fig. 5.1, representing the system design, consists of acoustic environment selector switch. It allows to provide the internal as well as external input. All the signals are stored in workspace and they can be played by clicking the buttons provided.

The adaptive algorithm block (Normalized Least Mean Square) consists of five ports namely input, desired, stepsize, adapt and reset.

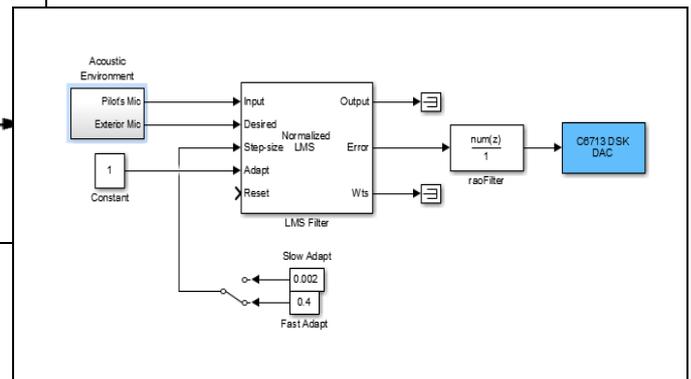


Fig.5.1. Simulink model for Active Noise Cancellation using Least Mean Square Algorithm (Ref [5])

Input: The original input of the system is through an external microphone connected to the system at this port.

Desired: The reference signal is fed through this port.

Step-size: It defines the rate of adaptation of the adaptive filter, generally denoted by  $\mu$ .

Adapt: It is used to make the adaptive algorithm ON or OFF. It is an optional port.

Reset: It resets the entire system to take the original input again and start the process afresh.

In most of the practical applications, Adaptive filters are used and are preferred over fixed digital filters because adaptive filters have the property of adjusting its frequency response and allowing the filter to adapt to the response as the input signal characteristics vary. The general configuration of an Adaptive Filter is shown in Fig. 5.2.

It has two inputs: the primary input  $d(n)$ , which represents the desired signal plus noise, and the reference signal  $x(n)$ , which is the noise to be filtered out of the system.

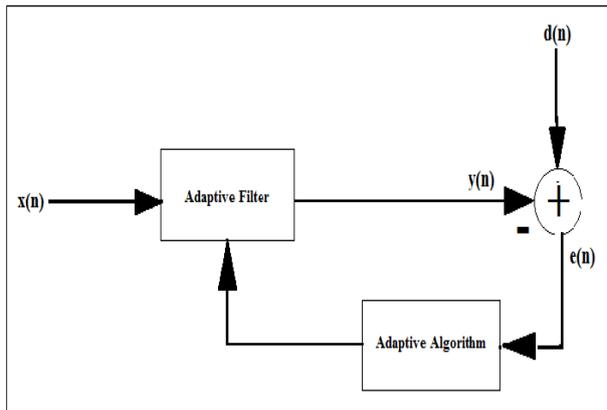


Fig. 5.2 Adaptive Filter Algorithm Block Diagram

The goal of adaptive filtering systems is to reduce the noise portion, and to obtain the uncorrupted signal which is desired. In order to achieve this, a reference of the noise signal, called reference signal  $x(n)$ , is needed. However, the reference signal is typically not the same as the noise present in the primary signal; it may be different in amplitude, phase or frequency. Therefore, the reference signal cannot be simply subtracted from the primary signal to obtain the desired signal at the output.

### A. Least Mean Square algorithm

The LMS algorithm is a gradient based algorithm that uses gradient vectors of the filter tap weights to converge to the optimal weigner solution. With every iteration, the weights of the adaptive filter get updated, which can be calculated by the following equation:

$$w(n+1) = w(n) + 2 \mu e(n) x(n)$$

where,  $x(n)$  is the input vector,

$w(n)$  represents the co-efficients of the FIR filter,

$\mu$  is the stepsize that decides the time of convergence.

If the value of  $\mu$  is too small, the time required for the convergence will be too long; if the value is too large, the filter may become unstable and the output can diverge.

### B. Normalised Least Mean Square Algorithm

To overcome the drawback of Least Mean Square algorithm in case of the large stepsize, normalized LMS algorithm is used in which the stepsize can be calculated as follows:

$$\mu(n) = \frac{\alpha}{c + |x(n)|^2}$$

where,  $\alpha$  is the NLMS adaption constant, that is used to optimize the convergence rate of the algorithm and should satisfy the condition  $0 < \alpha < 2$ . 'c' is a constant term used for normalization, which is always less than 1.

The filter weights using NLMS algorithm are updated by the equation:

$$w(n+1) = w(n) + \frac{\alpha}{c + |x(n)|^2} e(n) x(n)$$

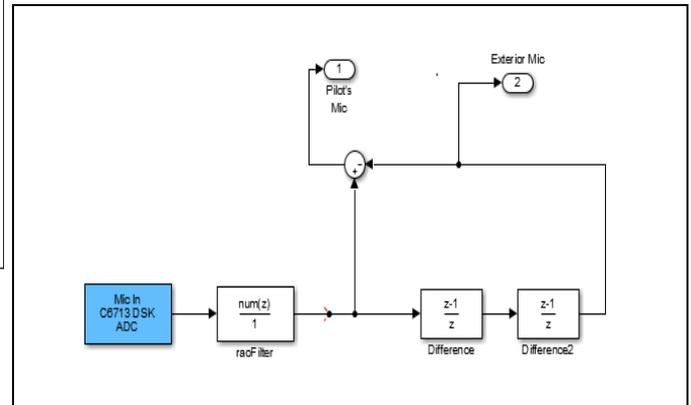


Fig 5.3 Simulink Model of filter used in Fig 5.1 (Ref. [5])

In fig 5.3, the input is given to system from the external microphone. This audio input is then filtered out to allow only frequencies below 3 kHz. This is then given to frame subtractors to get the reference signal which is the noise signal.

The reference signal and the original signal are then added out of phase to give us the noise signal which is given to digital filter and further to the adaptive algorithm block.

The filter selector switch is used to select from two filter options available:

- 1) Bandpass Filter
- 2) Lowpass Filter

Both the filter designs are based on FIR principle of order 39. The Bandpass filter is designed so as to pass frequencies in between 0.3 & 0.5 (normalised). The lowpass filter is designed so as to pass frequencies below 0.25 (normalised).

The numerator coefficients found out are passed to Digital Filter block along with the signal to be filtered out. The output of this filter is given to LMS algorithm block for further processing.

## V. FUTURE SCOPE

This system can be incorporated at workplaces such as schools and colleges, industrial workshops as well as residential areas near airports, market places etc.

Other adaptive algorithms may be used to enhance the efficiency of the system. Combination of existing technology and our system may bring out a new invention which is efficient and cost effective system in the near future.

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