CHAPTER 1

INTRODUCTION

1.1 INTRODUCTION:

The headsets using the passive noise cancelation used in aircraft provides good noise reduction for medium to high frequency range noise, but does not provide good noise reduction for low frequency noise. To get better noise reduction for low frequency, the weight and size of the cup has to be large. That would have practical limitations. Therefore ANR (Anti- Noise Reduction) is to be used.

1.2 THESIS OVERVIEW:

Acoustic noise problems increased with the increased number of large industrial equipments such as Constant speed drives, Generation, Integrated derive generators, Engines, transformers and compressor.

Propeller driven aircraft to a certain extent have noise range. The broadband random noise (for example noise of the wind) and narrowband noise (for example engine, propeller, compressor, alternators, constant speed drives), these makes the noise boring and tiresome [1].

Active noise cancellation (ANC) is an electro-mechanical or electro-acoustic system used cancels unwante noise. A secondary source (commonly a loudspeaker derived by control algorithm) creates an anti-noise of opposite phase and equal amplitude that is added to primary noise and results in the cancelation of noise. However ANC systems are ideally suited for use in low frequency range below approximately 800 Hz. Although higher frequency ANC systems have been built, a number of technical difficulties, both structural, acoustic and electronic limit their efficiency. At higher frequency, passive system generally becomes more cost effective. Therefore a complete ANC system would usually consists of active control for low frequency and passive control for higher frequency [3].

The two most common technique used for ANC are feed Forward (FF), generally used in environment where correlated reference signal is available to compare with primary signal. An adaptive FF ANC system consists of reference sensors an error sensor, a secondary control source, a control algorithm and electronic controller, Second is Feed Back (FB) technique, generally used in situation where a correlated reference signal is not available. An adaptive FB system consists of an error sensor, a secondary control source, a control algorithm and an electronic controller. A FB controller, the error signal is processed to drive a suitable control signal for control source such that error signal is used to help controller to optimize its performance in terms of minimizing error signal and is reference signal that is processed to generate control signal. Standard Least Mean Square (LMS) algorithm is modified to Filtered-x Least Mean Square (FxLMS) algorithm.

1.3 OBJECTIVE OF THESIS:

The Objective of the research is "*Design of an Integrated Active Noise Cancellation (ANC) System Using FXLMS Algorithm for Periodic Noise"*. For the Far End information signal (Signal received) and the Near End information signal (Signal being sent). By using ANC to reduce near end noise. And a noise canceller to improve near-end information before sending it to far end.

1.4 THESIS OUTLINE:

This thesis report is divided into six chapters. A brief description is as follows:

Chapter 2 – This chapter includes brief introduction of Active Noise Cancellation (ANC), basic terms related to stochastic processes, this Chapter also includes the performance evaluation and system optimization of ANC system location and placement of reference and error microphone location of secondary source noise source and other factors that effects performance of ANC system.

Chapter 3 This Chapter briefly covers the type of adaptive filter and adaptive algorithm. This chapter also covers an FIR and IIR filter. A detailed account of MSE and LMS algorithm has also been discussed.

Chapter 4 – This chapter includes the Secondary Path estimation. Secondary path introduces the anti-noise in the system in an appropriate time to cancel the effect of primary noise. Therefore exact modeling and estimation of secondary path becomes very important. This chapter also cover online techniques of secondary path estimation. Finally FxLMS algorithm has been incorporated to provide an exact modeling of the system. This chapter also discusses the design of an Integrated ANC system to attenuate the near-end periodic noise.

Chapter 5 – This chapter discusses the simulation results of Integrated ANC system using FxLMS algorithm for the periodic noise.

Chapter 6 – This chapter contains a final conclusion of the thesis and future recommendation of Integrated Active Noise Cancellation.

CHAPTER 2

INTRODUCTION OF ACTIVE NOISE CANCELLATION SYSTEM OPTIMIZATION AND PERFORMANCE EVALUTION

The general introduction of noise control and active noise cancellation (ANC) system is discussed in chapter 2. This Chapter also includes brief introduction of Active Noise Cancellation (ANC), basic terms related to stochastic processes, the performance evaluation and system optimization of ANC system location and placement of reference and error microphone location of secondary source noise source and other factors that effects performance of ANC system.

2.1 NOISE CONTROL:

Noise is defined as sound or sound that is loud, unpleasant, unexpected or undesired. Especially random and persistent disturbances, that obscure or reduce clarity of speech. Noise has many effects on people. It creates hearing loss, loss in communication signals, decreases the mental and physical performance, and disturbs sleeping habits. Reducing the noise pollution and its effects on human being is of great interest. The easiest way to decrease the noise pollution is to switch the noise source off [5].

2.2 TYPES OF NOISE:

There are two types of noise. Broadband and Narrowband noise.

2.2.1 NARROW BAND NOISE:

Narrowband noise is actually because of rotating machines,therefor it is nearly periodic and therefore most of its energy is concentrated at particular frequencies. Examples of narrowband noises are internal combustion engines, blowers, compressors, Generators etc. [8, 9]

2.2.2 BROAD BAND NOISE:

Broadband noise is the random and therefore contains its energy uniformly across frequency band, e.g noise of airplane and noise of explosion.

2.3 NOISE CANCELLATION:

Conventionally the passive noise reduction can be achieved with vibration insulators, vibration damping materials, acoustical absorbing materials and enclosures and barriers. The passive noise method needs no power therefore known as passive noise control. At low frequencies the noise reduction is insufficient and control of low frequency is unfeasible and costly [7].

Active noise cancellation provides cost effective, efficient, and lesser in weight solution. In active noise cancellation the anti-noise is generated which is opposite in phase that interferes and cancels the noise level at low frequency. The active noise cancellation is combination of acoustics, signal processing and mechanics.

2.3.1 NARROW BAND NOISE CANCELLATION:

For the narrow band noise cancellation instead of using a reference sensor, periodic ANC system can be applied, as the repetitive noise or harmonics of machine rotational rate is present. The control system will cancel these frequencies [1].

2.3.2 BROAD BAND NOISE CANCELLATION:

In order to cancel the broadband noise the environmental noise must be used in advance as an input signal. The primary noise signal will be cancelled at the secondary path.

2.4 ACTIVE NOISE CANCELLATION TECHNIQUES:

ANC system is a combination of electronic and acoustical transducers interacting with primary acoustic field. In an ANC system the secondary sound field is added into the existing sound field. The core component of ANC system is a control system that controls the transducers generating the secondary field. Now if the control system is self-adjusting then the system is known as ADAPTIVE system. Non-adaptive system have fixed parameters and the small environmental change can make system ineffective [4]. The ANC system is fed with signal containing information about noise. These signals can be either advance signal that holds information about residual noise, known as error signal.

The ANC system can be implemented by using Feed Forward and Feed Back technique.

2.4.1 FEED FORWARD CONTROL:

In feed forward ANC control system, a reference sensor is used to sense a coherent reference noise before sending it to secondary source. The Feed Forward ANC system is further divided into.

2.4.1.1 FEED FORWARD ANC SYSTEM FOR NARROW BAND NOISE:

As we know the primary noise is produced by rotating machinery. Therefore reference sensor can be replaced by non-acoustic or non-vibration sensor, such as tachometer or an optical sensor as shown in fig 2.1 [8]. The acoustic feedback from cancelling loudspeaker to reference sensor is removed [8, 9].

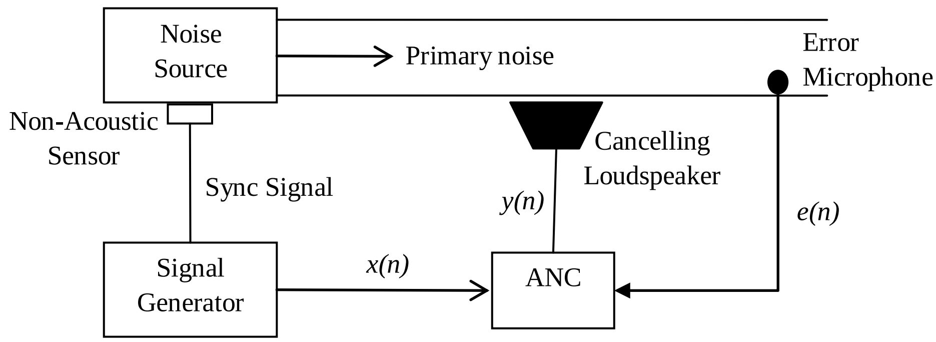


Figure 2.1:
Single Channel Narrowband Feed forward ANC System

2.4.1.2 FEED FORWARD ANC SYSTEM FOR BROAD BAND NOISE:

The Feed Forward ANC Control system for the broadband noise consists of reference sensor, error sensor and a speaker as a secondary source. An undesired signal $x(n)$ is sensed before sending it to the secondary path by using microphone placed near to the noise source as shown in the figure 2.2 The ANC control system uses a reference signal to generate a signal $y(n)$, which produces secondary noise signal in the secondary path which is 180^0 out of phase from unwanted noise present in the primary signal, therefore it cancels the unwanted noise.

In the system the residual error $e(n)$ is measured by using the error microphone, which can be reduced by changing the weights of the adaptive filter [8]. The overall system forms a close loop enhancing the complexities and making system unstable [8, 9].

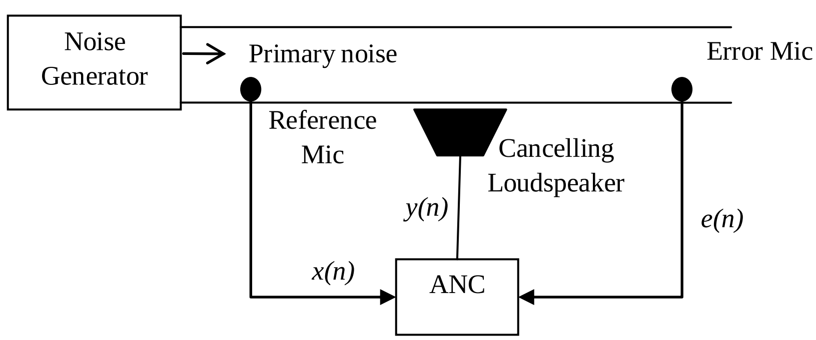


Figure 2.2: Single Channel Broadband FF ANC System

2.5 FEEDBACK ANC SYSTEM:

The feedback ANC system can be used in the applications where the coherent reference signal is not possible. Such application requires spatially incoherent noise generated from many sources and path. The figure 2.3 [9] show a single channel feedback ANC system. This feedback system creates a quiet zone using feedback control of secondary source located in range of an error sensor. The output of sensor is processed by an amplifier that has gain greater than unity and is out of phase to actual noise signal. Only one sensor is required and hence avoid secondary to reference sensor feedback problem. In a feedback ANC system the error sensor is used to pick the primary noise, which is inverted to derive loudspeaker placed near error sensor. In order to minimize the time delay the error sensor is positioned very near to the secondary source. The limited broadband noise attenuation is one of the drawback in feedback ANC system and the system instability caused by positive feedback at higher frequencies.

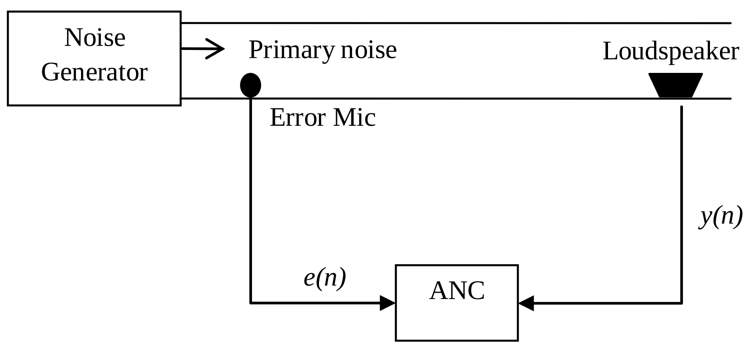


Figure 2.3: Single Channel Feedback ANC System

Another application of feedback ANC system is to control broadband noise in an active headset as shown in fig 2.4. The space between a loudspeaker and microphone minimized the phase shift and signal delay. The Feedback ANC system achieved a 10-db noise cancellation at ear drum for frequency up to 3KHz. Although the error microphone can be placed near to the secondary source, a high frequency limit is still set by the inevitable phase shift which increases with frequency causing instability. [8, 9]

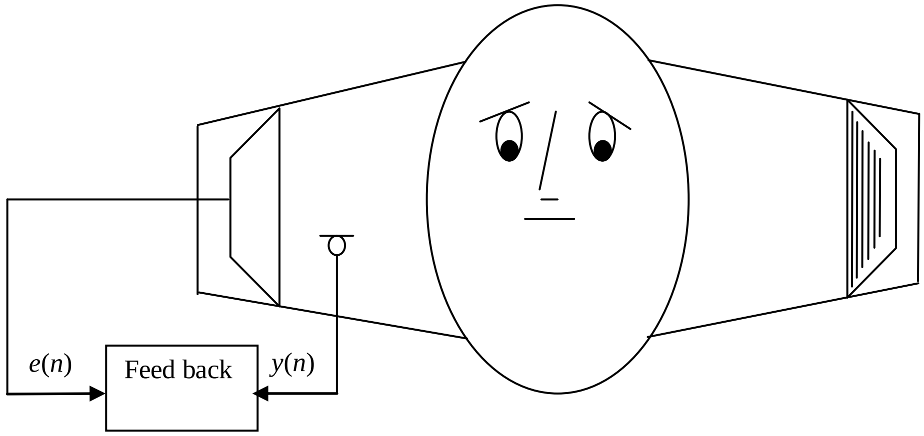


Figure 2.4: Application of Feedback ANC in Active Headset [8]

2.6 BASIC TERMS AND DEFINITIONS:

Following are some of the basic terms and the definitions.

2.6.1 Stochastic Process:

It is defined as the "time evolution of a statistical phenomenon according to probabilistic laws". It means that the stochastic process is time dependent [11]. Examples are;

- Speech signal
- Noise
- Seismological data
- Digital computer data
- TV signal
- Radar Signal

The sequence $u(n), u(n-1), \cdots, u(n-M)$ is a time series containing the present observation $u(n)$ made at any time *n*, and M is the past observations of the process at time $n-1, n-2, \cdots, n-M$ [11].

2.6.2 Stationary Process:

A random process which is independent of time i.e do not vary with time. For example, a discrete time stochastic process $u(n), u(n-1), \cdots, u(n-M)$ can be treated as strictly stationary, if the joint probability density functions of the random variables represented by the observation times $n-1, n-2, \cdots, n-M$ remain the same no matter what values we assign to *n* for a fixed M [11].

2.6.3 Mean Variance and Autocorrelation Functions:

For a discrete time stochastic process *u*(*n*), we define the mean value function as,

$$\mu(n) = E\left[u(n)\right] \tag{2.1}$$

Where E is the statistical expectation. From equation (2.1), we define the autocorrelation function of the process as,

$$r(n, n-k) = E\left[u(n)u(n-k)\right] \tag{2.2}$$

Where $k = 0, \pm 1, \pm 2, \cdots$

We define the auto covariance function of the process as,

$$c(n, n-k) = E\left[\left(u(n) - \mu(n)\right)\left(u(n-k) - \mu(n-k)\right)\right] \tag{2.3}$$

From equation (2.1) through (2.3),

$$c(n, n-k) = r(n, n-k) - \mu(n)\mu(n-k) \tag{2.4}$$

From equation (2.4), the autocorrelation function and the auto covariance function have the same value for mean $\mu(n) = 0$. If a process is strictly stationary, then mean value function $\mu(n)$ is a constant, i.e., $\mu(n) = \mu$, For all n [11].

Consequently, since the auto-covariance and the autocorrelation function are dependent only on the difference between the time *n* and *n* - *k*, that is they depend on the time lag *k*, therefore,

$$r(n, n-k) = r(k)$$
$$c(n, n-k) = c(k)$$

Where *k* = 0, corresponds to a time difference or lag of zero, $r(0)$ equals the mean square value of *u(n)*. From equation (2.2),

$$r(0) = E\left[\left|u(n)\right|^2\right] \tag{2.5}$$

Also,

$$c(0) = E\left[\left|u(n)\right|^2\right] - \mu^2 = \sigma_u^{\,2} \tag{2.6}$$

i.e., *c*(0) = mean square value – square of the mean value [11].

2.6.4 White Noise / Gaussian Noise:

"The Random variables taken from a fixed distribution is usually referred to as Gaussian with constant variance and zero mean". Than series is called as "White Gaussian noise". Specifically, for *v*(n) to be a White Gaussian noise [11],

$$E\left[v(n)\right]=0 \tag{2.7}$$

and,

$$E[v(n)v(k)]=\begin{cases}\sigma_v^{\ 2} & \forall k=n\\ 0 & \forall k\neq n\end{cases}$$

2.6.5 Mean Square Error:

The estimation error $e(n)$ is the difference between the desired response $d(n)$ and the filter out put $y(n)$, i.e.

$$e(n)=d(n)-y(n) \tag{2.8}$$

For design optimization, we reduce the mean square value of $e(n)$. Therefore the cost function is,

$$J=E\left[\left|e(n)\right|^2\right] \tag{2.9}$$

The requirement is to determine the operating conditions under which *J* attain its minimum value [11].

2.6.6 Power Density Spectrum:

"The frequency domain statistical parameter is the power spectral density also called as power spectrum density". Which is given as,

$$S(\omega)=\sum_{l=-\infty}^{\infty} r(l)e^{-j\omega l} \tag{2.10}$$

Where, $-\pi\leq\omega\leq\pi$ and $S(\omega)=$ Power spectral density and $r(l)=$ Autocorrelation function of a wide sense stationary stochastic process of time series $u(n)$ for lag l and equation (2.10) represents the Discrete Time Fourier Transform of the autocorrelation function $r(l)$ [11].

2.6.7 Wiener Filter:

A discrete time linear filter is called a Wiener filter, if the filter is optimized in the sense of MSE (mean square error) value of error signal as defined by equation (2.9) [11].

2.6.8 Rate Of Convergence:

"It is defined as the number of iterations required for the algorithm in response to stationary inputs, to converge close enough to the optimum Wiener solution in the MSE sense" [11].

2.6.9 Finite Word Length:

It is a hardware problem related with the limit of decimal places allocated to the coefficient of filters when quantized, after binary coding the discrete values, for example, ($2^N = 2^8 = 256$) [8,11].

2.6.10 Impulse Response:

In the case of a time domain signal, we can find the output signal $y(n)$ of an LTI (Linear time invariant) processor by convolving its input signal with a second time domain function representing the processor itself. This second function is the processor's response to the unit impulse function $\delta[n]$, or simply its impulse response [11].

2.6.11 z – Transform:

z-Transform is the frequency analysis technique related with digital signals and processors. The z-transform and Fourier transform are closely related. The z-transform of a digital signal $x(n)$ is defined as [11].

$$X(z) = \sum_{n=0}^{\infty} x(n) z^{-n}$$

2.7 ANC OPTIMIZATION AND PERFORMANCE:

The performance of an Active Noise Cancellation system is ruled by a number of related factors that must be under consideration in the appropriate order during the design process. Air acoustic ANC is useful for low frequency sound where passive technique are less effective, generally below 500 Hz. Narrow band or tonal sounds can be reduced by 30-db or more, while broadband or random disturbances can be reduced by 15-20 db [9, 12].

2.8 PRACTICAL LIMITATIONS IN PERFORMANCE EVALUATION:

In acoustic ANC, some of the practical issues that affect the performance and efficiency of ANC system are as follows [9]:

- Placement of loudspeaker and microphone.
- Adaptation algorithm.
- Increasing the power of the speaker.
- Reducing the flow of noise on microphones.
- Increasing the durability of the microphone and speaker.
- Reducing the cost of the controller.

2.9 SYSTEM OPTIMIZATION:

2.9.1 PLACING THE CONTROL SOURCE:

The maximum levels of attenuation that is possible with a particular active noise cancellation system is determined first by the placing the control source. This means, no matter how well the error sensor have been placed or how good are the electronics, an ANC system will not function efficiently, if the control source is not placed properly [9, 12].

2.9.2 LOCATION OF ERROR SENSOR :

After the control source has been placed efficiently, the maximum performance observed with the active control system will be enhanced by the location of the error sensor. The error sensor must be so located as to effectively sense all parts of both the primary and control signal.

2.9.3 ELECTRONICS CONTROL SYSTEM:

The final performance factor for a feed forward control system is the performance of the electronic control system which is related to its calculation accuracy and dynamic range as well as the control algorithm it uses. The dynamic range is determined by the number of bits characterizing A/D (Analog to digital) converter as well as the adjustment of input amplifiers to ensure that the maximum possible signal level which generated the maximum digital output [9].

2.10 WHY USE ADAPTIVE DIGITAL CONTROLLER:

The goal of ANC may be to attenuate the sound to a maximum value or to achieve certain desired spectral characteristics. Both of these applications involve the reduction of noise and distortion for retrieving information from the signal. The problem increases with changing environmental conditions such as temperature and transducer wear. Therefore the controller must be “self-tuning” or “adaptive” rather than a conventional (fixed one) [13,14]. Adaptive system modifies their characteristics to achieve certain objectives and usually accomplish the modification/cancellation automatically [15].

CHAPTER 3

ADAPTIVE FILTER AND ALGORITHM

This Chapter briefly covers the type of adaptive filter and adaptive algorithm. A comparison between FIR and IIR filter has been made. A detailed account of MSE, steepest descent algorithm, LMS algorithm, NLMS algorithm has also been discussed.

3.1 FILTERS:

In the context of digital signal processing, the term “filtering” refers to the linear process designed to change the contents of an input signal in a defined manner. Conventional filters are linear and time invariant.

3.2 ADAPTIVE FILTERS:

In the case of adaptive filters there is no restriction of time invariance. Adaptive filter is defined as a filter in which the parameters such as frequency and bandwidth are time dependent. This can be done by varying the co-efficient of adaptive filters and automatically adjusting by the algorithm [11].

There are two parts of an adaptive filter: a digital filter used for signal processing, an adaptive algorithm for adjusting the co-efficient of that filter. An adaptive filter is shown in fig 3.1 [11], $d(n)$ is desired signal, the output of a digital filter is $y(n)$ and the error $e(n)$ is difference between $d(n)$ and $y(n)$. The function of the adaptive algorithm is to adjust the coefficients of the digital filter so as to reduce the mean square value of $e(n)$ [8, 11].

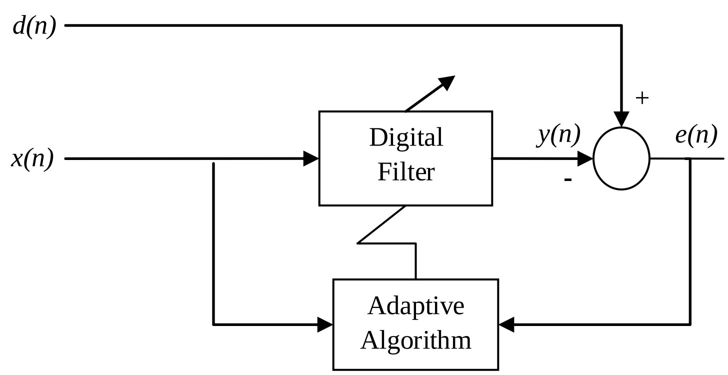


Figure 3.1: Adaptive Filter

3.2.1 IIR FILTER:

In some cases IIR filters are used where the noise to be controlled is broadband in nature. IIR filters are also used in feed forward ANC system where there is feedback from the control source to reference microphone resulting corruption of the reference signal. This problem can be reduced by using directional sound source and microphone arrangements, it cannot be removed completely. IIR filter architecture is shown in fig 3.2 [9] and may be considered of two FIR filters.

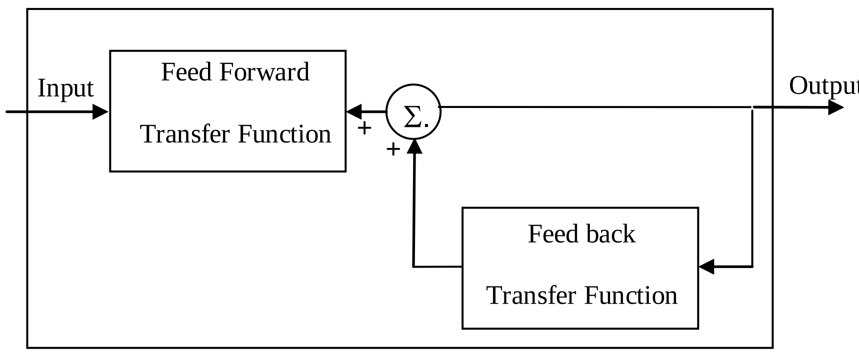


Figure 3.2: IIR Filters

3.2.2 FIR FILTER:

An FIR filters is given in figure 3.3 where, z^{-1} is a delay of one (input) sample and w_i represents the filter weights i. The number of stages in the filter (the number of present and past input samples used in the output derivation) is usually referred as the number of filter "taps". When subject to unit step input, the filter output will eventually decay to zero. FIR filters are usually used for tonal noise problem where the reference signal is one or perhaps a few sinusoids and where the control signal does not corrupt the reference signal [9].

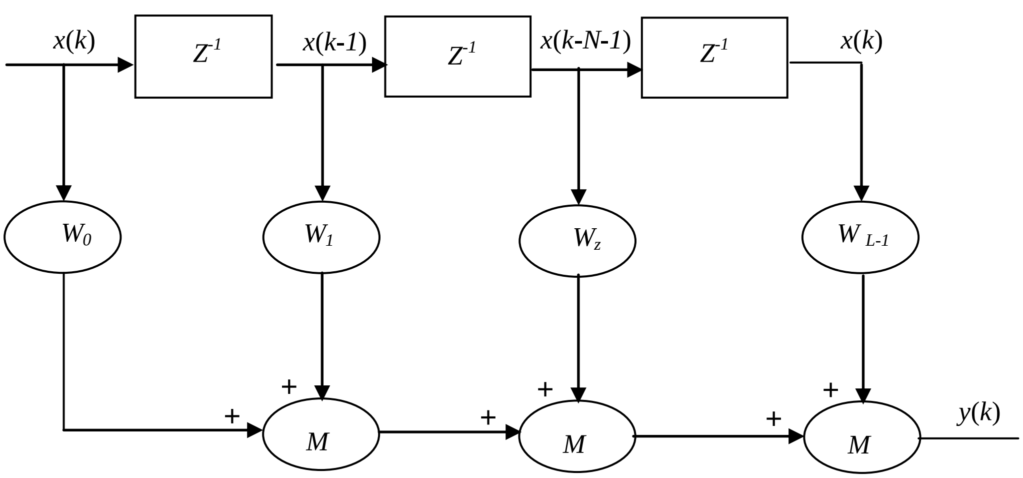


Figure 3.3: FIR Filter

3.3 APPLICATION OF ADAPTIVE FILTERS:

The adaptive filter are usually used for an unfamiliar environment and also to track the variation in time of the input so as to make it a powerful device for signal processing and control application.

An adaptive filter may be interpreted as "adaptive system identification" in feed forward technique and an "adaptive predictor" in feedback scheme.

3.3.1 ADAPTIVE NOISE CANCELLATION:

The basic idea of adaptive noise cancellation is to process signals from two sensors and to reduce the unwanted noise with adaptive filtering techniques. As shown in figure 3.4, the primary microphone is placed next to to the signal source to sense the speech signal. However, the primary sensor also picks some noise from the noise source. The reference microphone is placed near the noise to pick the noise signal only. The Adaptive noise cancellation system correlates the noise signal sensed by the primary sensor as well as by reference sensor [8].

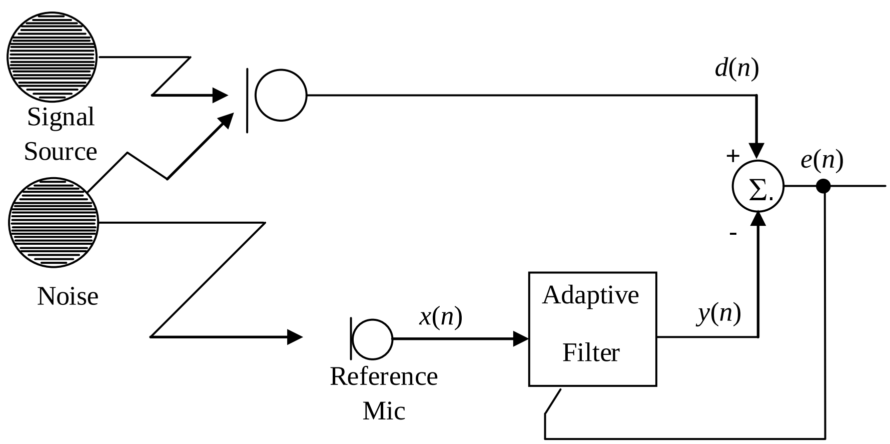


Figure 3.4: Basic model of ANC [8]

In figure 3.5, *P(z)* is the transfer function between the primary sensor and the noise source. The primary input *d(n)* contains signal $y(n)$ and noise $x^{'}(n)$. The reference input consists of noise $x(n)$ only[8]. To reduce the residual error *e(n)*, the output *y(n)* is generated by adaptive filter *W(z)* that is an estimate of $x^{'}(n)$.

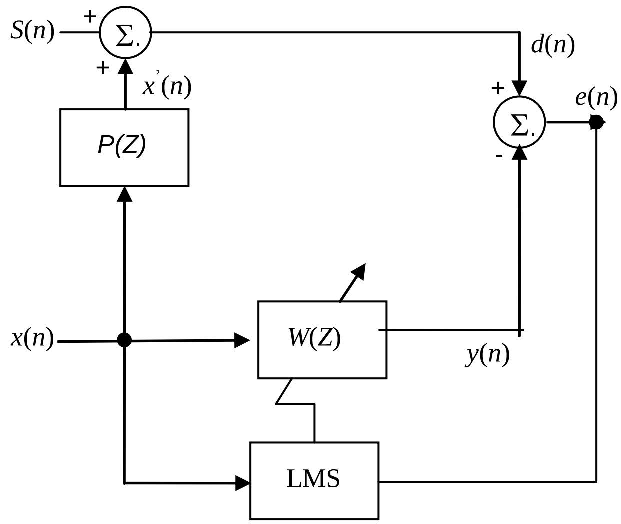


Figure 3.5: Block Diagram of ANC [8]

3.3.2 ADAPTIVE SYSTEM IDENTIFICATION:

"Adaptive System identification is a practical approach to the modeling of a plant". An adaptive filter can be used in modeling, specifically replicating the behavior of physical system dynamics.

Refer to figure 3.6, an input exciting signal *x(n)* is an input to the adaptive filter *W(z)* and an system *P(z)*. Assuming the input signal *x(n)* is providing spectral excitation, the output *y(n)* after convergence will approximate *d*(*n*). And we will achieve this when the co-efficient of the adaptive filter *W(z)* are tuned to the same value as that the unknown system *P(z)* provided that the order of the adaptive filter is same as that of the unknown system.

Referring to figure 3.6 [8], we have,

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

$$e(n) = \sum_{l=0}^{L-1} [p_{l-w_l}] x(n-l)$$

Where p_l is the parameter of the plant. By selecting each $w_l(n)$, close to corresponding p_l the error $e(n)$ will be small. Conversely, minimizing $e(n)$ with an LMS criterion will force $w_l(n)$ to approach p_l. Thus, *W(z)* is said to have identified the system *P(z)* [8].

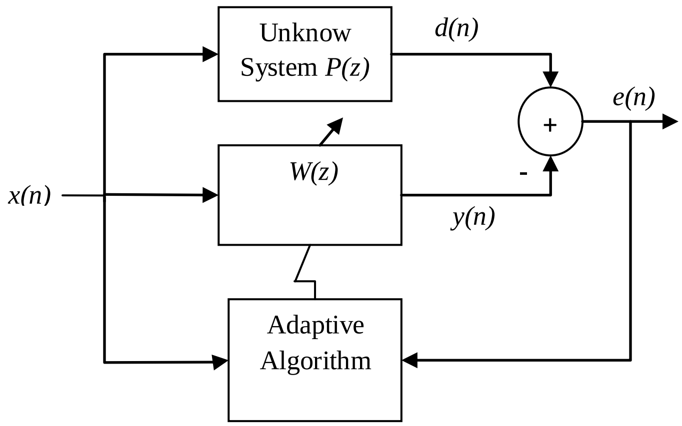


Figure 3.6: Adaptive System Identification

3.4 ADAPTIVE FILTER ALGORITHM:

Adaptive filters are used in many signal processing applications. The Adaptive filters used for ANC applications are based on transversal (Wiener) filter structure which is commonly used with LMS algorithm.

3.5 MEAN SQUARE ERROR (MSE) PERFORMANCE SURFACE:

In figure 3.7, the co-efficient of the digital filter are updated to optimize the performance. Which is based on the MSE,

$$\xi(n) = E\left[e^2(n)\right] \tag{3.1}$$

For an adaptive Finite Impulse response (FIR) filter as shown in figure 3.7, $\xi(n)$ will depend on the 'L' filter weights $w_0(n), w_1(n), \cdots, w_{L-2}(n), w_{L-1}(n)$. If we ensure that adaptive weight vector $w(n)$ is a deterministic sequence, then MSE performance is as follows,

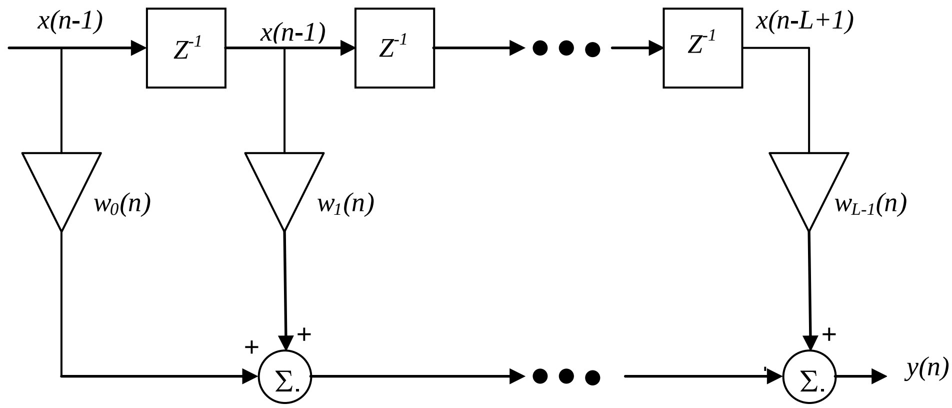


Figure 3.7: Adaptive FIR Transversal (Wiener Filter)

$$\mathbf{x}(n)=\left[x(n)x(n-1)x(n-2)\cdots x(n-L+1)\right]^T$$
$$\mathbf{w}(n)=\left[w_0(n)w_1(n)\cdots w_{L-1}(n)\right]^T$$

$$y(n)=\sum_{l=0}^{L-1} w_l(n)x(n-l) \tag{3.2}$$

Writing equation (3.2) ,

$$\begin{aligned} y(n)&=\mathbf{w}^T(n)\mathbf{x}(n)\\ y(n)&=\mathbf{x}^T(n)\mathbf{w}(n) \end{aligned} \tag{3.3}$$

Generally, the system output $y(n)$, from figure (3.1) is,

$$\begin{aligned} e(n)&=d(n)-y(n)\\ e(n)&=d(n)-\mathbf{w}^T(n)\mathbf{x}(n) \end{aligned} \tag{3.4}$$

From figure (3.1),

$$\xi(n) = E\left[e^2(n)\right]$$
$$\xi(n) = E\left[d(n) - y(n)\right]^2$$
$$\xi(n) = E\left[d^2(n)\right] - 2E\left[d(n)y(n)\right] + E\left[y^2(n)\right]$$
$$\xi(n) = E\left[d(n) - \mathbf{w}^T(n)\mathbf{x}(n)\right]^2$$
$$\xi(n) = E\left[d^2(n)\right] - 2E\left[d(n)\mathbf{w}^T(n)\mathbf{x}(n)\right] + E\left[\mathbf{w}^T(n)\mathbf{w}(n)\mathbf{x}^T(n)\mathbf{x}(n)\right]$$
$$\xi(n) = E\left[d^2(n)\right] - 2E\left[d(n)\mathbf{x}(n)\right]\mathbf{w}^T(n) + \mathbf{w}^T(n)\mathbf{w}(n)E\left[\mathbf{x}(n)\mathbf{x}^T(n)\right]$$
$$\xi(n) = E\left[d^2(n)\right] - 2\mathbf{p}^T\mathbf{w}(n) + \mathbf{w}^T(n)\mathbf{R}\mathbf{w}(n) \tag{3.5}$$

Where,

$$\mathbf{p} = E\left[d(n)x(n)\right]$$
$$\mathbf{p} = [r_{dx}(0)\ r_{dx}(1)\ r_{dx}(2) \ldots .r_{dx}(L-1)]^T$$

and,

$r_{dx}(k) = E\left[d(n)x(n-k)\right]$ is the cross correlation between d(n) and x(n).

$$\mathbf{R} = E\left[x(n)x^T(n)\right]$$

$$R = \begin{bmatrix} r_{xx}(0) & r_{xx}(1) & \cdots & r_{xx}(L-1) \\ r_{xx}(1) & r_{xx}(0) & \cdots & r_{xx}(L-2) \\ . & . & . & . \\ . & . & . & . \\ r_{xx}(L-1) & r_{xx}(L-2) & \cdots & r_{xx}(0) \end{bmatrix}$$

and,

$$r_{xx}(k) = E\left[x(n)x(n-k)\right]$$

Is the autocorrelation function of $x(n)$.

3.6 LMS ALGORITHM:

LMS algorithm uses the instantaneous values of the squared error $e^2(n)$ to estimate the mean square error given by,

$$\hat{\xi}(n) = e^2(n) \tag{3.6}$$

LMS algorithm simply uses the instantaneous gradient (slope) of a single squared error sample. Mathematically

$$\frac{\partial}{\partial w(n)} e^2(n) = 2e(n) \frac{\partial}{\partial w(n)} e(n)$$
$$\frac{\partial}{\partial w(n)} e^2(n) = 2e(n) \nabla e(n)$$
$$\nabla \hat{\xi}(n) = 2e(n) \nabla e(n) \tag{3.7}$$

From equation (3.4),

$$e(n) = d(n) - y(n)$$
$$e(n) = d(n) - \mathbf{w}^T(n)\mathbf{x}(n)$$
$$\frac{\partial}{\partial w(n)} e(n) = \frac{\partial}{\partial w(n)} d(n) - \frac{\partial}{\partial w(n)} \mathbf{w}^T(n)\mathbf{x}(n)$$
$$\nabla e(n) = 0 - \mathbf{x}(n) = -\mathbf{x}(n) \tag{3.8}$$

Substituting equation (3.8) in equation (3.7),

$$\nabla \hat{\xi}(n) = -2\mathbf{x}(n)e(n) \tag{3.9}$$

Now replacing $\nabla \xi(n)$ by $\nabla \hat{\xi}(n)$,

$$\begin{aligned} \mathbf{w}(n+1) &= \mathbf{w}(n) - \frac{\mu}{2}\left[-2\mathbf{x}(n)e(n)\right] \\ \mathbf{w}(n+1) &= \mathbf{w}(n) + \mu\mathbf{x}(n)e(n) \end{aligned} \tag{3.10}$$

3.6.1 ANALYSIS OF LMS ALGORITHM:

A block diagram in figure 3.8 illustrates LMS algorithm implementation of transversal (Wiener) filter. The summary of the algorithm is as follows [8, 19]:

- Choose parameters L, μ and $w(0)$. Where μ is the step size, L is the order of the filter and $w(0)$ is the initial weight.
- Calculate filter output,

$$y(n)=\sum_{l=0}^{L-1} w_l(n)x(n-\Delta-l)$$

- Compute error signal output,

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

- Update adaptive weight by using LMS algorithm, in equation (3.10),

$$w_l(n+1)=w_l(n)+\mu x(n-l)e(n) \tag{3.11}$$

Where, $l = 0, 1,2,\ldots\ldots.L-1$

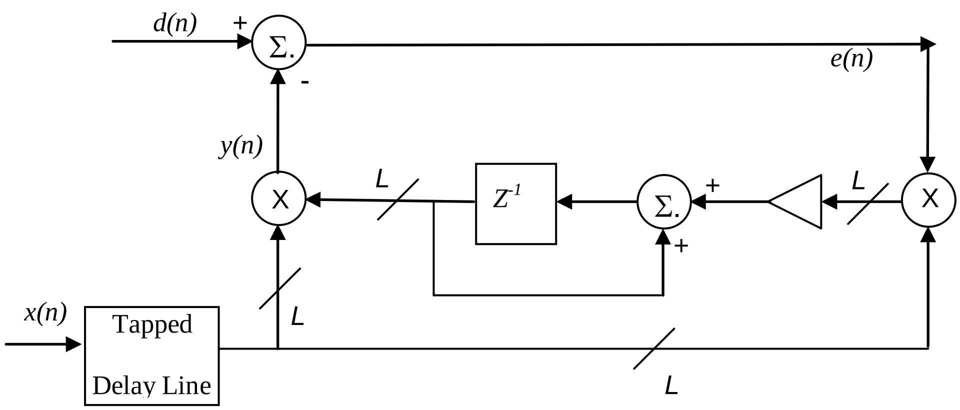


Figure 3.8: Block diagram of LMS adaptive filter [8]

3.6.2 STABILITY AND CONVERGENCE ANALYSIS OF LMS:

The stability and convergence of LMS algorithm depends on the value of λ_l, i.e. the eigenvalues of $\mathbf{R}$ matrix. Following is the limiting condition.

$$0<\mu<\frac{2}{\lambda_{\max}} \tag{3.12}$$

Where $\lambda_{\max}$ is the largest eigenvalue of $\mathbf{R}$ matrix and μ is step size. Since the computation of $\lambda_{\max}$ becomes difficult, specially, where 'L' is very large; therefore, in many practical applications, $\lambda_{\max}$ is estimated as follows,

$$tr[\mathbf{R}] = Lr_{xx}(0) = \sum_{l=o}^{L-1} \lambda_l \tag{3.13}$$

Where, $tr[\mathbf{R}]$ is trace of matrix $\mathbf{R}$. It follows that,

$$\lambda_{\max} \le \sum_{l=0}^{L-1} \lambda_l = Lr_{xx}(0) = Lp_x \tag{3.14}$$

Where,

$$p_x = r_{xx}(0) = E\left[x^2(n)\right] \tag{3.15}$$

denotes the power of $x(n)$.

Substituting equation (3.14) in (3.18), we get,

$$0 < \mu < \frac{2}{Lp_x} \tag{3.16}$$

Equation (3.16) applies only to the convergence of mean weight. Convergence of weight variance or MSE imposes more stringent step size constraints. For Gaussian signals, convergence of the mean square error requires.

$$0 < \mu < \frac{2}{3Lp_x} \tag{3.17}$$

3.6.3 RESULTS:

- Since the μ is inverse relation with L, therefore smaller μ is used for large filter order.
- Since μ has inverse relation with input power p_x; weaker signals uses a large μ and stronger signal have to use a smaller μ. In practice.

$$\frac{0.01}{Lp_x} < \mu < \frac{0.1}{Lp_x} \tag{3.18}$$

CHAPTER 4

SECONDARY PATH ESTIMATION

This chapter includes the Secondary Path estimation. Secondary path introduces the anti-noise in the system in an appropriate time to cancel the effect of primary noise. Therefore exact modeling and estimation of secondary path becomes very important. It also indclude different offline and online techniques of secondary path estimation. Finally FxLMS algorithm has been incorporated to provide an exact modeling of the system.

4.1 BASIC PRINCIPLE:

The basic (broad band) ANC systems consist of an acoustic system, as shown in figure 4.1. The reference microphone is used to measure the undesired noise from a primary noise source, passes from an adaptive filter, and used to derive a secondary sound sources usually a speaker, to remove the unwanted noise in the duct. The input signal from the reference sensor must be correlated with the noise from the primary source. For a broad band active noise control system, the reference signal provides information in advance about the primary noise before it reaches the loudspeaker to cancel unwanted noise. The error microphone picks the residual noise and updates the co-efficient of the adaptive filter to minimize the residual noise (if reference signal is not affected by feedback from the secondary source).

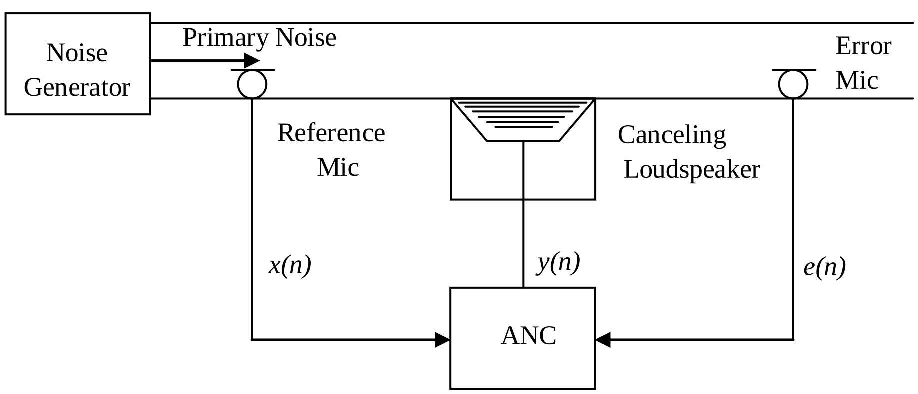


Figure 4.1: Single Channel Broad band FF ANC in a duct [8]

4.2 SYSTEM IDENTIFICATION:

The figure 4.2 provides a system identification scheme in which an unknow plant *P(z)* is estimated by using an adaptive filter *W(z)*. The output of the adaptive filter *y*(n) after convergence, will approximate *d(n)* in an optimum sense so as to reduce the error. The adaptive filter *W(z)* is said to have identified the plant *P(z)*. If the plant is dynamic, the model will be time dependent. The adaptive algorithm then keeps the error small by continuously tracking variation of the plant dynamics. The acoustic path between the reference microphone and the error microphone is known as primary path *P(z)* where the noise attenuation is to be realized. The difference between figure 4.2 [8] and figure 3.6 is that instead of subtraction of the electrical signal the acoustic summing junction is used. From figure 4.2,

$$E(z) = D(z) - Y(z)$$
$$E(z) = P(z)X(z) - W(z)X(z)$$

For convergence,

$$E(z) = 0$$
$$P(z) = W(z) \tag{4.1}$$

where

$$X(z) \neq 0$$

and;

$$y(n) = d(n) \tag{4.2}$$

Equation (4.2) shows that the adaptive filter output is equal to the desired signal *d*(*n*), therefore,

$$e(n) = d(n) - y(n) \tag{4.3}$$

Which depicts a causal system and results in perfect cancellation of sound.

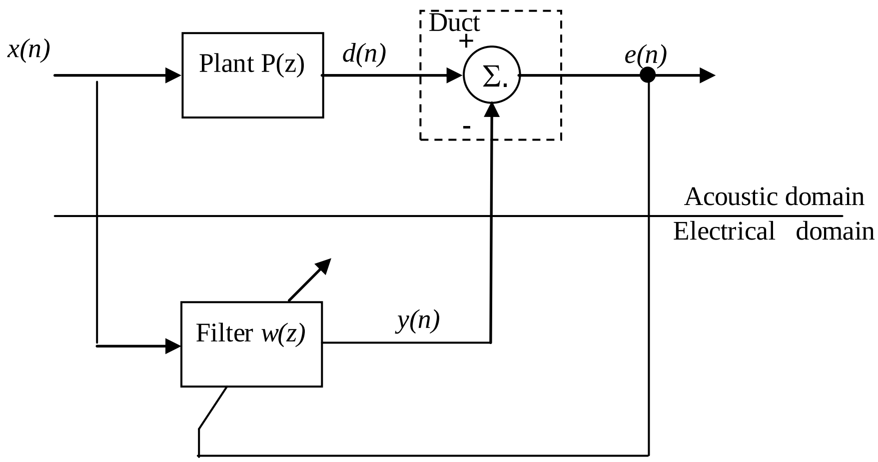


Figure 4.2: System Identification

4.3 ANALYSIS OF SYSTEM IDENTIFICATION SCHEME:

If the model *W(z)* is a good estimation of the plant *P(z)*, the feed forward approach is able to cancel broad band (random) noise very effectively. This, of course, assumes that there is a sufficient time for the system to complete the model calculation, determine the inverse signal and generate the required secondary sound before the unwanted noise passes through the plant. As the signals pass through the model and the plant in at the same time, changes in the primary source do not affect the level of the cancelled sound if the model adequately represents the plant.

4.4 DEFINITIONS:

4.4.1 SECONDARY PATH (CANCELLATION PATH):

"It is the path that "anti-noise" takes from the output loudspeaker to the error sensor within the quiet zone".

4.4.2 ELECTRICAL DELAY:

With reference to figure 4.1 after the reference signal is being sensed up by the reference sensor, the controller will take some time in manipulating the output to derive the canceling speaker. This time is called electrical delay.

4.4.3 ACOUSTIC DELAY:

The time taken by the primary signal from reference microphone up to canceling loudspeaker is called acoustic delay.

4.4.4 CAUSAL SYSTEM:

In terms of ANC, if the generation of anti-noise and primary noise takes place in same time domain, the system is called 'Causal".

4.5 SECONDARY PATH EFFECT:

At the reference microphone, the acoustic pressure is used to produce the electrical signal. Also, at error sensor again an electrical error signal is obtained from the residual acoustic noise. Finally an acoustic output (anti-noise) is generated by the loudspeaker from an electrical output signal. As shown in figure 4.2, the summer represents acoustic superposition from the error microphone to the canceling loudspeaker, where the output of the filter is combined with the primary noise. The effect of electro acoustic path between the electrical input to the control sensor and the electrical output from error sensor is called secondary path effect which has to be compensated by using a secondary path transfer function $S(z)$ as shown in figure 4.3 [8, 16, 17].

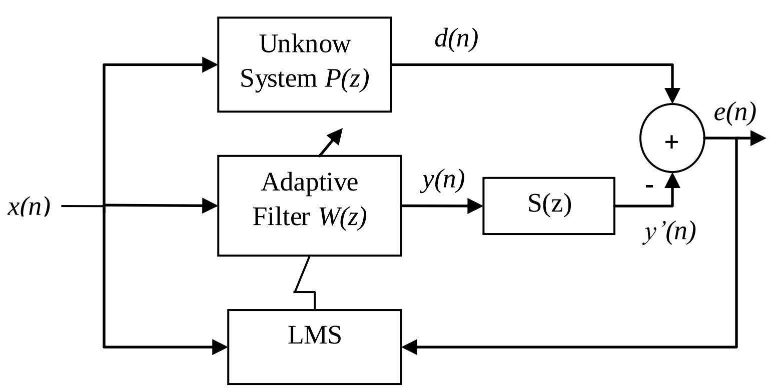


Figure 4.3: Simplified block diagram of ANC System

In figure 4.4, the secondary path transfer function can be divided into two transfer functions, i.e.

$$S(z) = R(z)S'(z) \tag{4.4}$$

Where $S'(z)$ represents the secondary transfer function from the adaptive filter to summer and R(z) represents the residual transfer function from summing junction to the error signal.

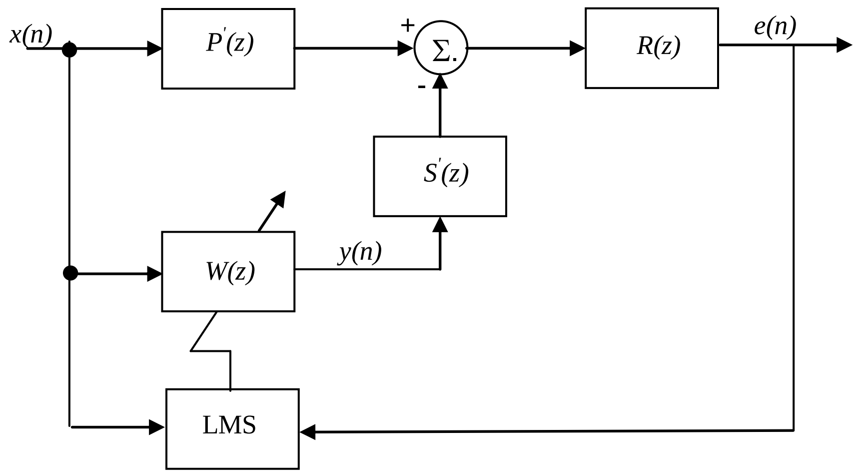


Figure 4.4: Block diagram of an ANC System

Similarly, the primary transfer function from input sensor to error sensor is,

$$P(z) = R(z)P'(z) \tag{4.5}$$

Where $P'(z)$ is the transfer function of this unknown acoustic plant from the reference microphone to the summing junction.

From figure 4.4,

$$\begin{aligned} E(z) &= D(z) - Y(z) \\ E(z) &= R(z)P'(z)X(z) - R(z)S'(z)W(z)X(z) \\ E(z) &= R(z)\left[P'(z) - S'(z)W(z)\right]X(z) \end{aligned} \tag{4.6}$$

In case of convergence (for optimum filter solution),

$$\begin{aligned} E(z) &= 0 \\ W(z) &= W^0(z) = \frac{P'(z)}{S'(z)} \end{aligned} \tag{4.7}$$

Therefore, the adaptive filter has to invert the secondary path transfer function $S'(z)$.

4.6 FXLMS ALGORITHM:

4.6.1 GENERAL:

The filtered xLMS algorithm provides the solution when there is a transfer function in the secondary path after the filter. To provide convergence of the algorithm, the controller actually needs an FIR filter, inserted between the reference signal and the controller algorithm to produce a "filtered version" of the reference signal before it is used by the algorithm as shown in figure 4.5 [8,18];

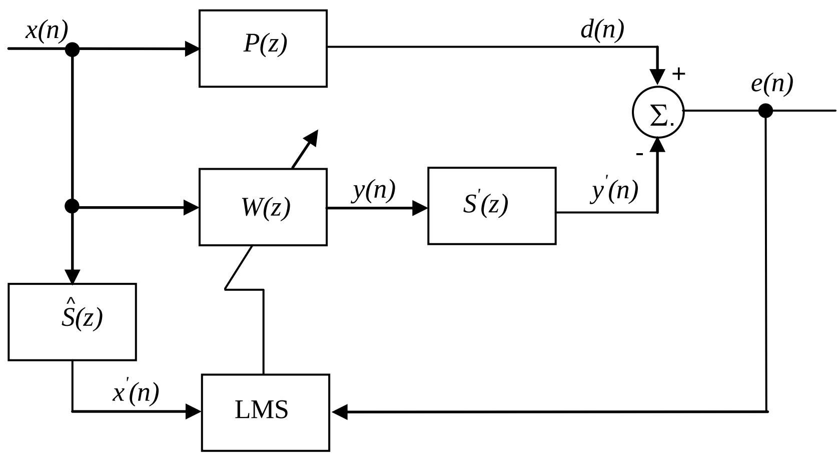


Figure 4.5: ANC System using FXLMS algorithm

4.6.2 DERIVATION OF THE ALGORITHM:

Considering figure 4.5, as $S(z)$ = FIR filter used for modeling of the secondary path and $\hat{S}(z)$ is Estimate of $S(z)$

The residual noise is,

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

$$e(n)=s(n)*y(n)$$

$$e(n)=d(n)-s(n)*\left[\mathbf{w}^T(n)\mathbf{x}(n)\right] \tag{4.8}$$

Where,

$s(n)$ = Impulse response functions of the cancellation path (time domain equivalent of the transfer function in frequency domain). $*=$ Convolution operator

$$\mathbf{w}(n)=\begin{bmatrix} w_0(n) & w_1(n) & \cdots & w_{L-2}(n) & w_{L-1}(n)\end{bmatrix}^T$$

is the coefficient vector of W(z) at time n.

$$\mathbf{x}(n) = [\quad x(n) \quad x(n-1) \quad \cdots \quad x(n-L+2) \quad x(n-L+1)]^T$$

L = Order of the filter *W(n)*.
The objective of the adaptive filter is to reduce the squared error, as discussed in Chapter 3 and

$$\hat{\xi}(n) = e^2(n)$$

The LMS algorithm which updates the weights of the adaptive filter in the negative gradient direction with step size μ is.

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \frac{\mu}{2}\nabla\hat{\xi}(n) \tag{4.9}$$

Where, $\hat{\xi}(n) =$ instantaneous estimate of mean square error at time *n*.

$$\nabla\hat{\xi}(n) = \nabla e^2(n)$$
$$\nabla\hat{\xi}(n) = 2[\nabla e(n)]e(n) \tag{4.10}$$

But from equation (4.8),

$$e(n) = d(n) - s(n) * [\mathbf{w}^T(n)\mathbf{x}(n)]$$
$$\frac{\partial e(n)}{\partial w(n)} = 0 - s(n) * \mathbf{x}(n)$$
$$\nabla e(n) = -\mathbf{x}'(n) \tag{4.11}$$

Where,

$$\mathbf{x}'(n) = [\quad x'(n) \quad x'(n-1) \quad \cdot\cdot \quad \cdot\cdot \quad x'(n-L+1)]^T$$
$$\mathbf{x}'(n) = s(n) * x(n) \tag{4.12}$$

is the "filtered reference signal" obtained by passing the reference signal $x(n)$ through an FIR filter $\hat{S}(z)$, that models the impulse response of the cancellation path. Substituting equation (4.11) in (4.10), we get,

$$\nabla\hat{\xi}(n) = -2\mathbf{x}'(n)e(n) \tag{4.13}$$

Substituting equation (4.13), we get,

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\mathbf{x}'(n)e(n) \tag{4.14}$$

Equation (4.14) represents the filtered-xLMS algorithm. In practical ANC applications, $S(z)$ is unknown and must be estimated by an additional filter $\hat{S}(z)$. Therefore, by passing the reference signal through this estimate of the secondary path the filter reference can be obtained as shown in equation (4.15) i.e.

$$\mathbf{x}'(n) = \hat{s}(n) * x(n) \tag{4.15}$$

Where $\hat{s}(n)$ is the estimated" impulse response" of the secondary path filter $\hat{S}(z)$.

4.7 SECONDARY PATH ESTIMATION (MODELING) TECHNIQUES:

The Secondary path can be estimated using the following modeling technique.

4.7.1 ON-LINE SECONDARY PATH MODELING:

In ANC application, the primary noise is cancelled by a secondary noise that is generated by an adaptive filter through a secondary path. The secondary path can be estimated off-line, prior to the operation of the ANC system. However, in many practical applications, the secondary path may be time varying. In that case, it is desirable to estimate the secondary path on-line when the ANC is in operation.

4.8 WORKING OF ON-LINE MODELING:

A ANC system with adaptive on line modeling (estimation) is shown in figure 4.6. The filter $\hat{S}(z)$ compensates for *S(z)* for standard application of FXLMS algorithm. Here, the coefficients of $\hat{S}(z)$ are continuously adjusted on-line to model the secondary path *S*(*z*) during the operation of ANC filter W(z). A direct on-line secondary path modeling technique which is based on a system identification method is shown in figure 4.7. The secondary signal *y*(*n*) generated by *W(z)* also serves as an excitation signal for modeling of the secondary path. The adaptive filter $\hat{S}(z)$ is adapted by the LMS algorithm to reduce the *f*(*n*), which is the difference between signed inverted residual error *e*(*n*) and the output signal of $\hat{S}(z)$, i.e. the system identification error. The reference signal *x*(*n*) is then filtered by a copy of $\hat{S}(z)$ to update the coefficients of the adaptive filter W(z) using the FXLMS algorithm [8,19,17].

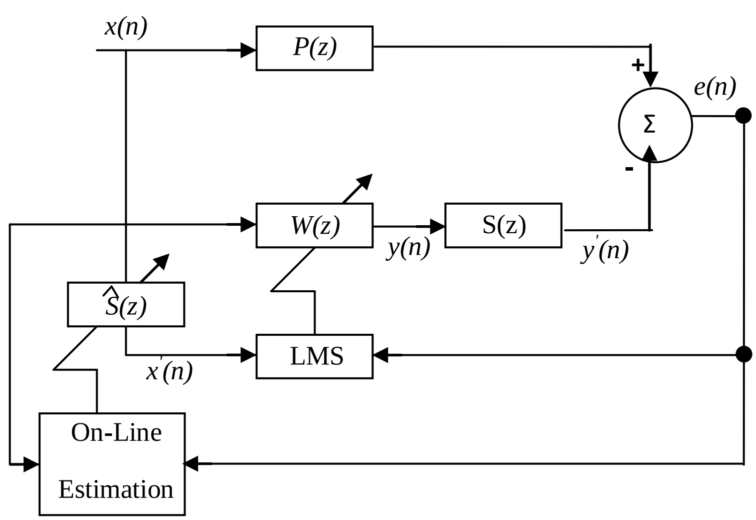


Figure 4.6: ANC System with FxLMS Algorithm and Online Modeling

The error signal $e(n)$ as shown in figure 4.7 contains the signals from primary path $P(z)$ and secondary path $S(z)$. From the system identification point of view, the primary noise $d(n)$, which is highly correlated with the excitation signal $y(n)$, becomes an interference when using the adaptive filter $\hat{S}(z)$ to identify S(z). If we take z-transform of the system identification error f(n) as in fig: 4.7

$$F(z) = -E(z) - \hat{S}(z)Y(z)$$

$$F(z) = -\left[P(z)X(z) - S(z)Y(z)\right] - \hat{S}(z)Y(z)$$

$$F(z) = -\left[P(z)X(z) - S(z)W(z)X(z)\right] - \hat{S}(z)W(z)X(z)$$

$$F(z) = \left[S(z)W(z) - P(z) - \hat{S}(z)W(z)\right]X(z) \tag{4.16}$$

In case of perfect modeling, the identification error f(n) converges to zero i.e.

$F(z) = 0$ and $\hat{S}(z) = \hat{S}^0(z)$

$$S(z)W(z) - P(z) - \hat{S}(z)W(z) = 0$$

$$\Rightarrow \hat{S}^0(z) = S(z) - \frac{P(z)}{W(z)} \tag{4.17}$$

Equation (4.17) shows the estimate $\hat{S}^0(z)$ is biased by a factor $P(z)/W(z)$. This is possible only when the primary noise is completely removed by some other means.

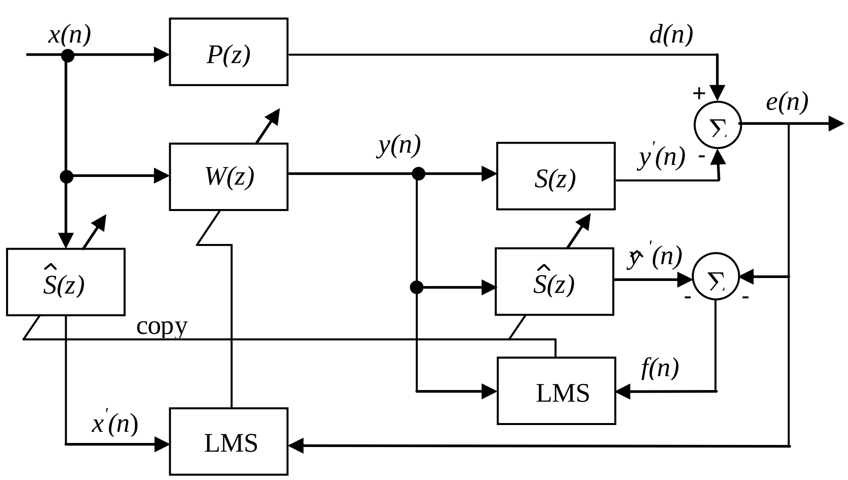


Figure 4.7: On Line Secondary Path Modeling

4.9 CANCELLATION OF NOISE FOR NEAR-END SIGNAL:

In the communication the near-end noise has to be removed before transmitting signal to the far end. In fig 4.9 in order to remove the near-end noise a simple noise cancelling filter *H(z)* is used. The microphone which is used to sense the desired near-end speech will also sense the near-end unwanted noise. Whereas *P(z)* is the "primary path from noise source to the microphone". The adaptive LMS algorithm is given as [20],

$$h_l(n+1) = h_l(n) + \mu y(n-1)e(n), \qquad l = 0,1,2,\ldots\ldots,L-1$$

Where,

"*h(n)* is the output of filter *H(z)* and *y(n)* is correlated with the reference near-end noise".

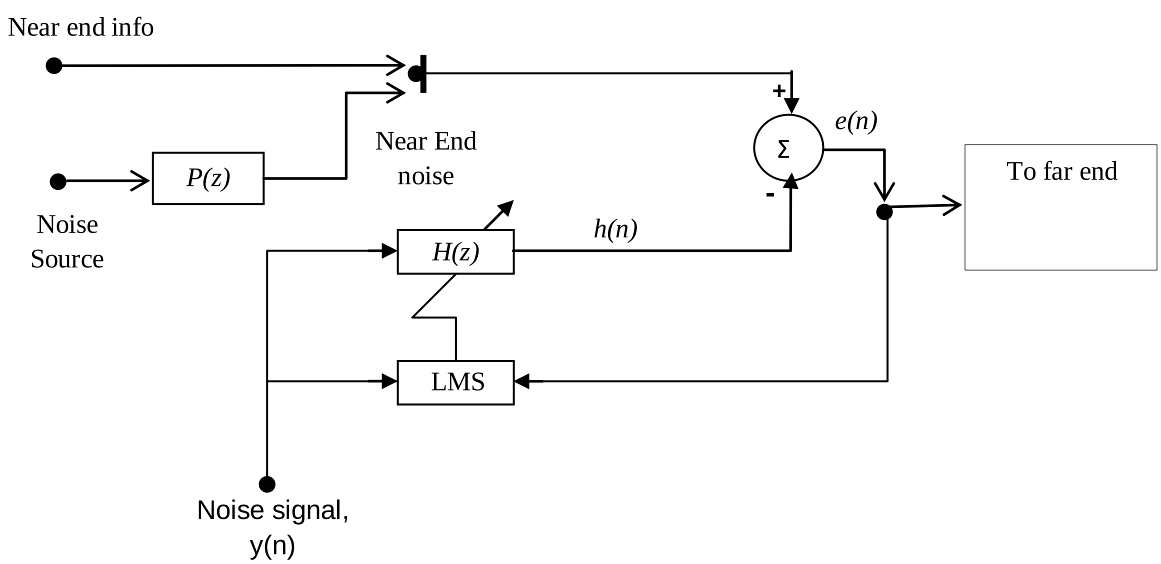


Fig 4.8: Active Noise Cancellation filter for Near-end transmission

4.10 INTEGRATED ANC-COMMUNICATION SYSTEMS:

This system uses three microphones as shown in fig: 4.9. One microphone is located inside headphones ear-cup, this microphone is actually an error microphone to sense the residual noise for updating the adaptive filter. One microphone is placed near to the noise source to sense the unwanted noise. And a microphone is placed outside to ear-cup, which is used to sense the near-end information. [20].

The integrated adaptive feedback ANC is shown in fig 4.9.The error sensor signal *e(n)* consists of the original information and the residual noise. The far-end speech signal *a(n)* is used as a reference signal by the speech interference cancellation filter $\hat{S}(z)$ in order to remove the speech component *e(n)*.

Now the difference error signal $x'(n)$contains the residual noise.

The filter *H(z)* can also be incorporated into the headphones in order to reduce the undesired noise *d(n)* from the near-end speech signal [20].

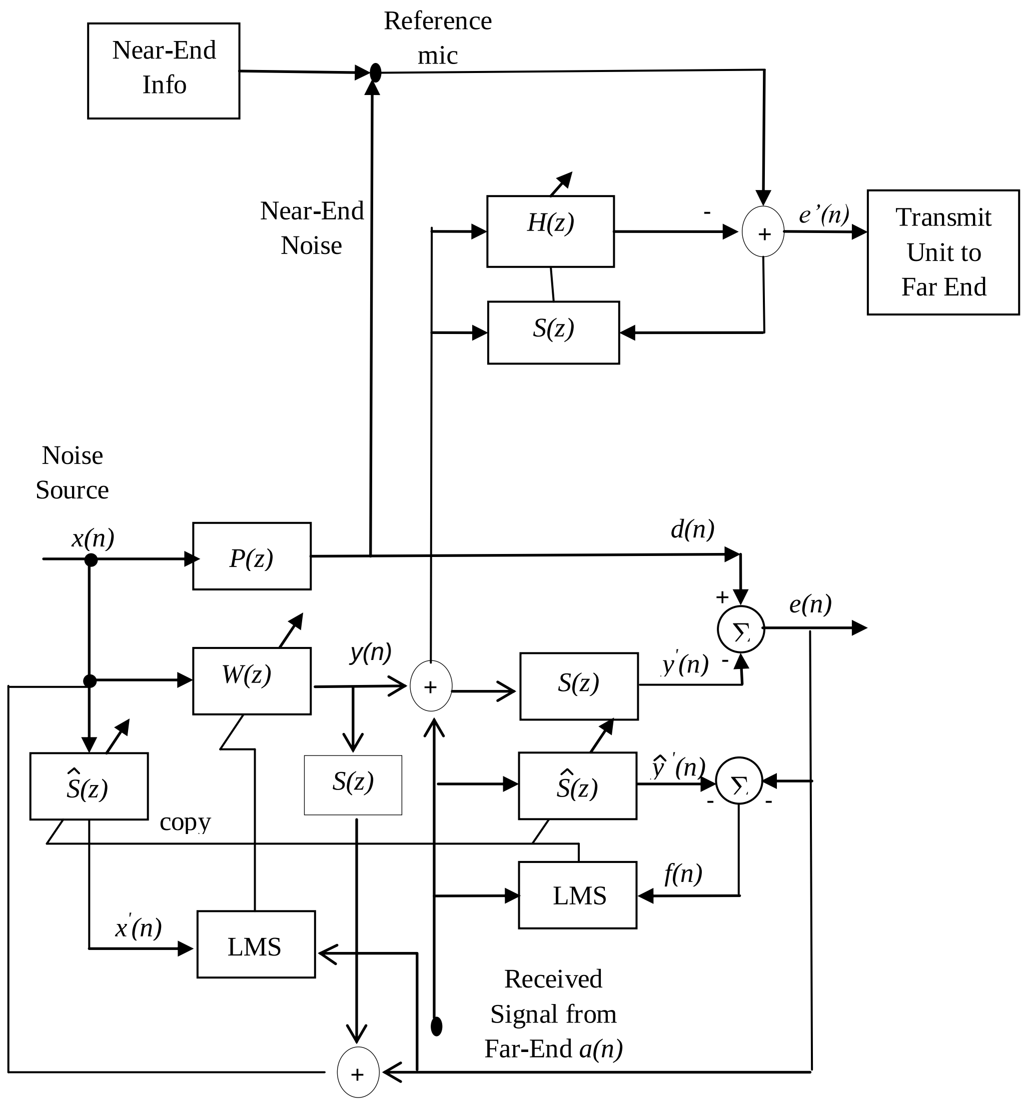


Fig 4.9: Integrated Active Noise Cancellation using FxLMS Algorithm

CHAPTER 5

SILMULATION AND RESULTS

5.1 INTRODUCTION:

Analysis and design of control system has been made easy with the development and enhancement of simulation tool like MATLAB.

This Chapter includes the simulations and results of Integrated ANC system for the far-end information and the near end speech. In this chapter we will show the performance and results of the ANC filter, the secondary path modeling and estimation and the adaptive noise cancelation filter. We will also discuss the performance of the system and analyze the adaptation and the robustness of the ANC system by varying the disturbances and parameter of the primary path and the secondary path. MATLAB has been adapted in analysis and simulation design of the system for its simplicity and comprehensiveness by researchers. Step by step results are observed by running Simulink blocks.

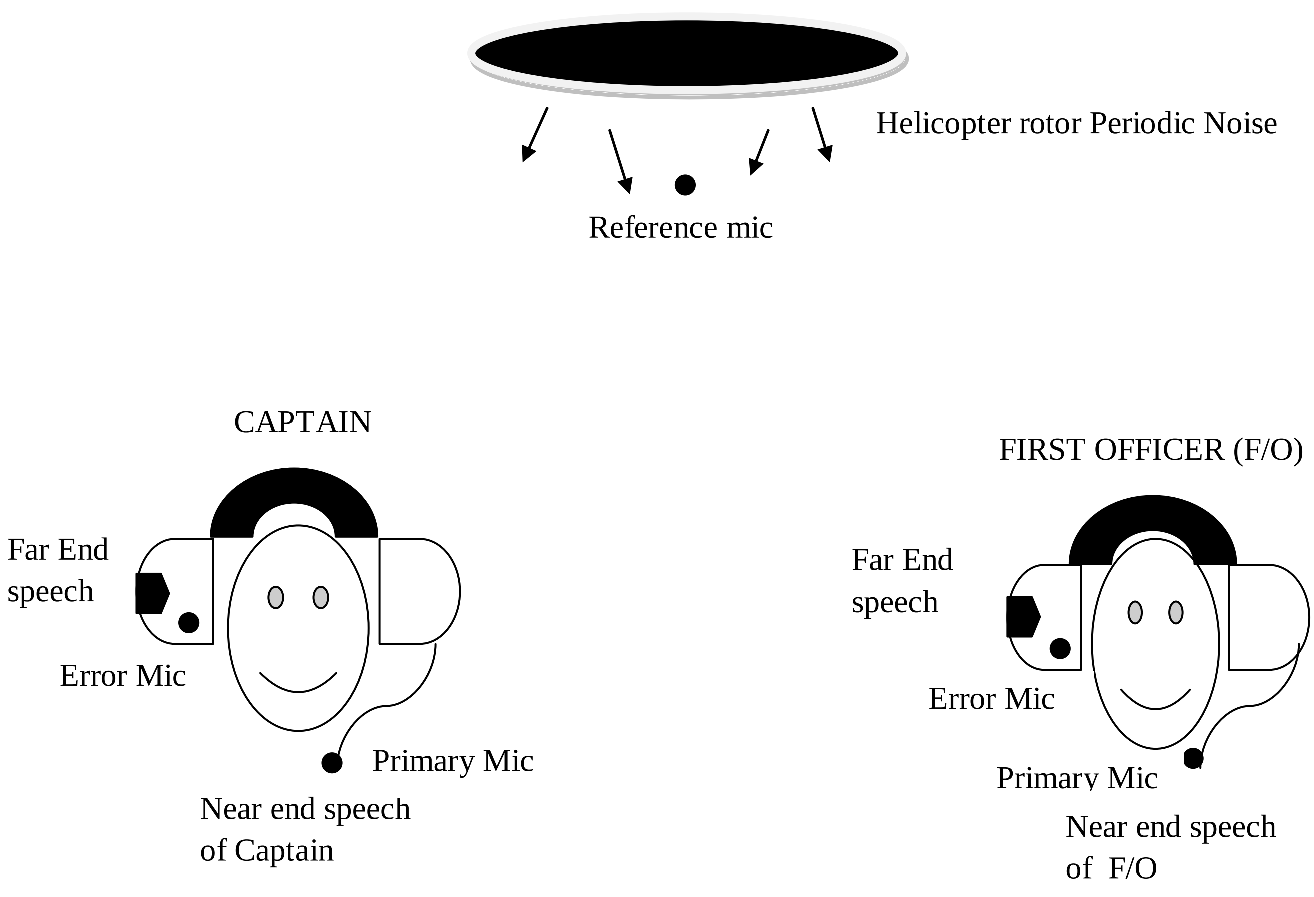


Fig: 5.1 Application of Integrated ANC using FxLMS Algorithm.

In the figure 5.1 we have considered an helicopter, the noise generated by the rotor of helicopter is creating trouble for the Captain and First officer (F/O) while communicating with each other. Whatever Captain speaks is near-end signal for the Captain himself but his speech is far-end info for the F/O, similarly whatever F/O speaks is near-end signal for himself but his speech is far-end signal for the Captain. It consists of three microphone, one is an error microphone placed close to the noise source so as to sense the noise only, the other is an error microphone which is used to pick the residual noise and the third microphone is the primary mic used to pick the speech of the Captain or the F/O.

5.2 NOISE PROFILE:

A periodic noise of the helicopter rotor was recorded by using MATLAB code with 11.025 KHz, 8 bits is introduced into the system with the peak amplitude of approximately -0.8 and + 0.8. And the most of random noise level due to the vibration of the blade bearing and the leading edge wind load is concentrated to about -0.1 and +0.1 . The periodicity of the rotor noise is approximately 0.09 with changing amplitude because of the pitch and speed changing of the blade.

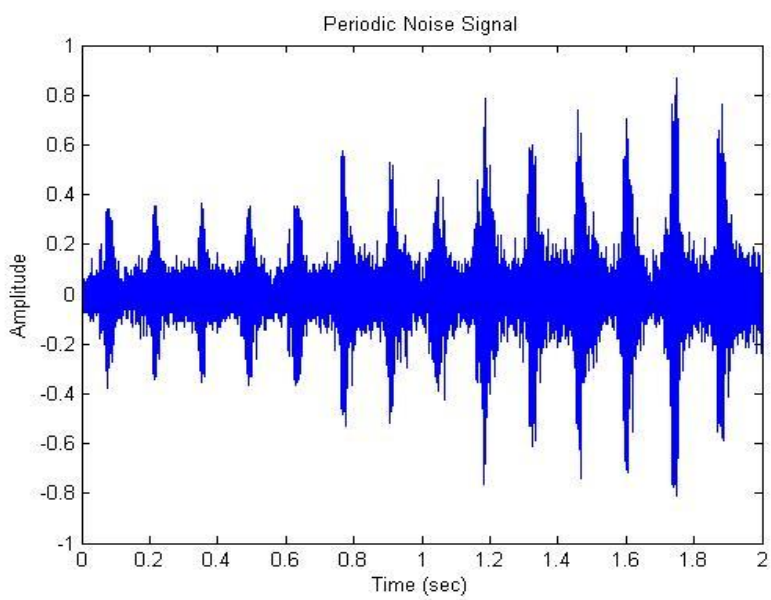


Figure 5.2: Periodic Noise

5.3 INFORMATION SIGNALS:

The two recorded signals are used as a far-end information signal and a near-end information signal with 11.025 KHz and 8bits by using MATLAB.

The signal used for the near-end is actually uttered during designing as "Active noise cancellation" which is shown in figure 5.3, and the far-end speech used in the system is uttered during designing as "Far-end speech signal" as shown figure 5.4 respectively.

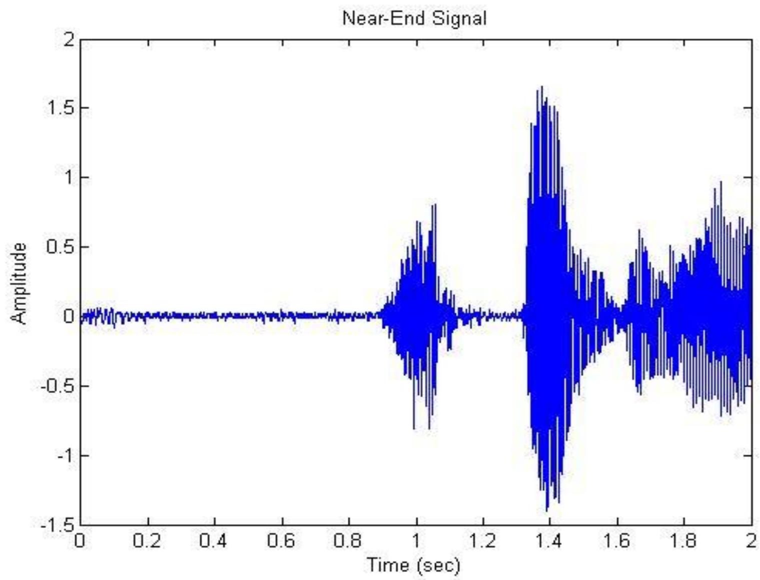


Figure 5.3: Near-End Signal

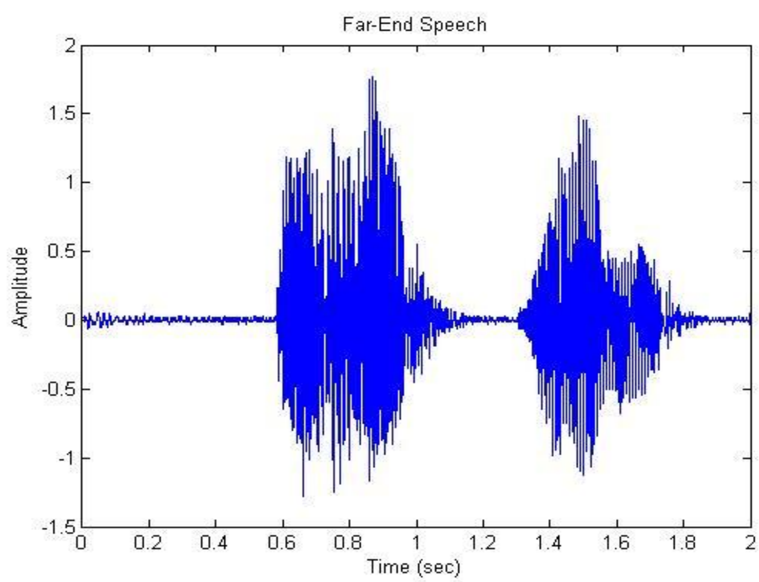


Figure 5.4: Far-End Speech Signal

5.4 SIMULINK MODEL:

In this Chapter we present some experimental results of Integrated Active Noise Cancellation using FxLMS Algorithm for the periodic noise. The *P(z)* and *S(z)* pairs were used in all the simulations [21].

Where, $P(z) = z^{-5} - 0.3z^{-6} + 0.2z^{-7}$ and

$S(z) = z^{-2} + 1.5z^{-6} - z^{-4}$

The unwanted noise from the near-end speech signal is removed by using the *H(z)* filter whose weights are being updated by using the LMS filter as shown in figure 4.9 which is implemented in the simulink model by using the LMS Filter block containing built-in block of LMS and the *H(z)* filter, generating *h(n)* as an output.

$$h_l(n+1) = h_l(n) + \mu y(n-1)e(n)$$

Where, $l = 0,1,2,\dots\dots,L-1$

h(n) is the output of filter *H(z)* and "*y(n)* is correlated with the reference near-end noise".

In order to remove the unwanted noise from the far-end speech signal, at first the estimation of the secondary path $\hat{S}(z)$ is carried out by using the LMS Filter1 as shown in figure 5.4 which consists of LMS filter and $\hat{S}(z)$ as shown in figure 4.9

The output from the LMS Filter1 serves as the desired signal for the LMS Filter3 and LMS Filter4 as shown in figure 5.5, these filters are used to generated the co-related noise signal which is used to remove the unwanted noise from the original far-end speech signal. The *W(z)* filter shown in figure 4.9 is used to generate the replica of unwanted periodic noise which is implemented in simulink by using the LMS Filter 4 in figure 5.5.

The Compensate of the secondary path $\hat{S}(z)$ is added to the LMS filter updating the weights of the filter *W(z)* as shown in figure 4.9 which is implemented by LMS Filter3 in figure 5.5, output of which is added to the output by LMS Filter4 in figure 5.5. Finally the unwanted noise is subtracted from the far end speech signal containing noise by using Add1 in figure 5.5.

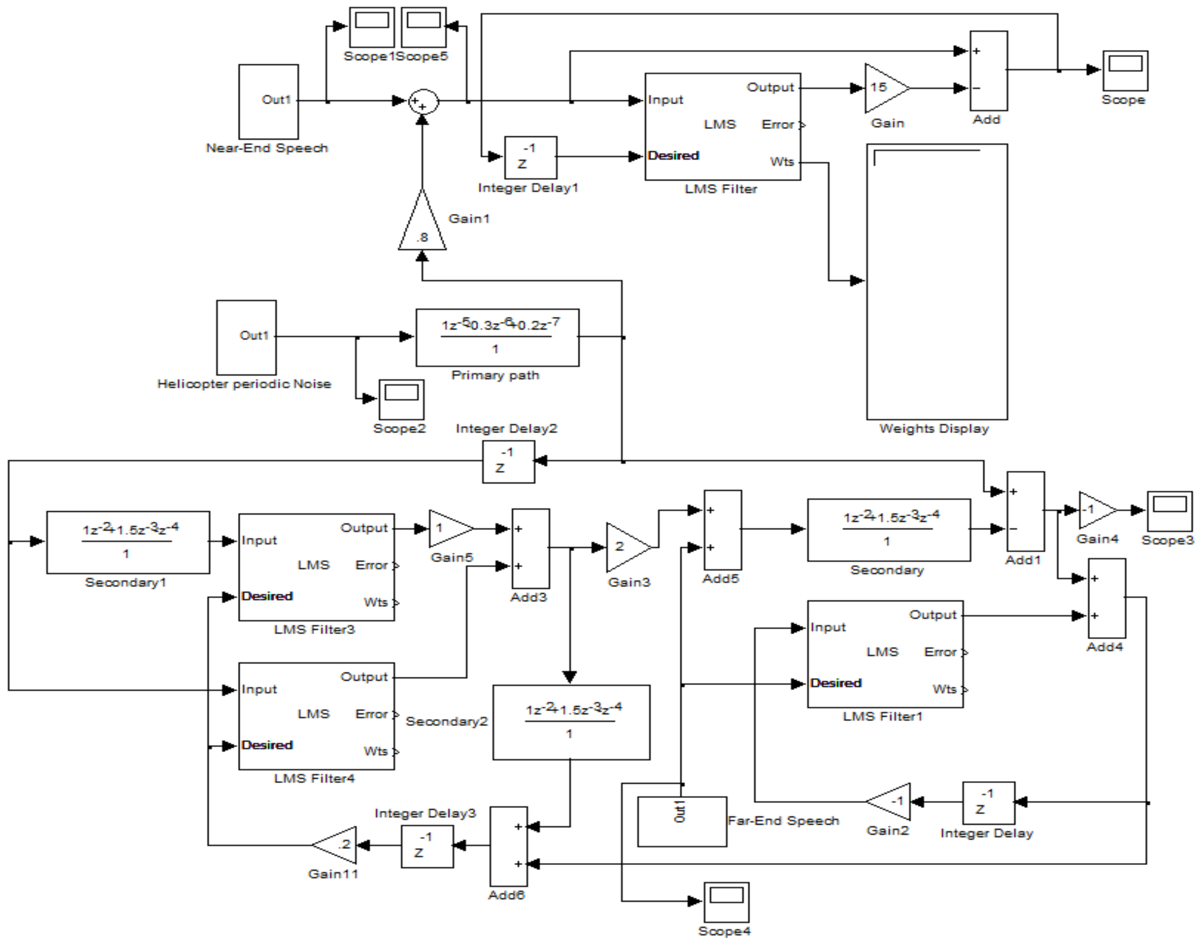


Figure 5.5 Integrated ANC using FxLMS Algorithm for periodic Noise

5.5 SIMULATION RESULTS:

Simulation was carried out in order to evaluate the capabilities and response of the integrated system. A periodic noise of rotor with 11.025 KHz, 8 bit was introduced into the system. The adaptive LMS filter, LMS Filter1 and LMS Filter3 used in the simulation were of different tap weights, and step sizes respectively.

The LMS Filter block in the simulink as shown in fig: 5.5 is used to remove the unwanted rotor periodic noise from the near-end speech signal. The selection of LMS Filter order and the step size are the major concern while using the LMS block. The order and the step size is selected by hit and trial method.

The initial step size and the value selected for LMS block is 0.05 and order of 8-tap. It was observed that the inappropriate selection of these values results in the system instability and the noise cancellation was not efficient and the error was high though the system response was fast.

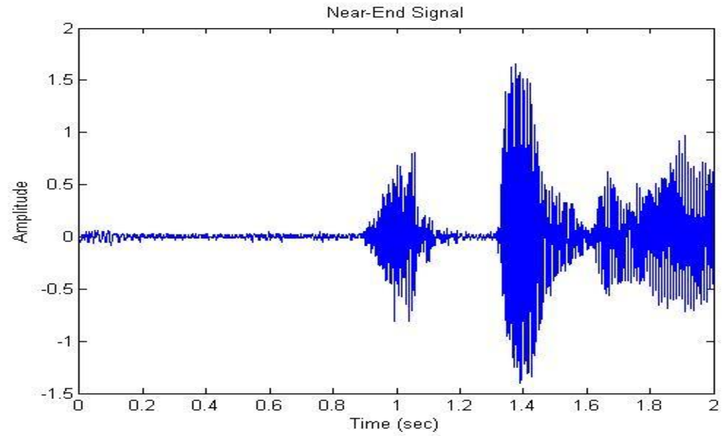
The magnitude of the noise reduction and the convergence of the LMS filter increases by selecting the 0.0005 step size and the 12 tap filter order is the optimum value.

By increasing the step size and the order of the filter, the error and noise is reduced to great extend but the system acts slow and the fine results are obtained after some delay.

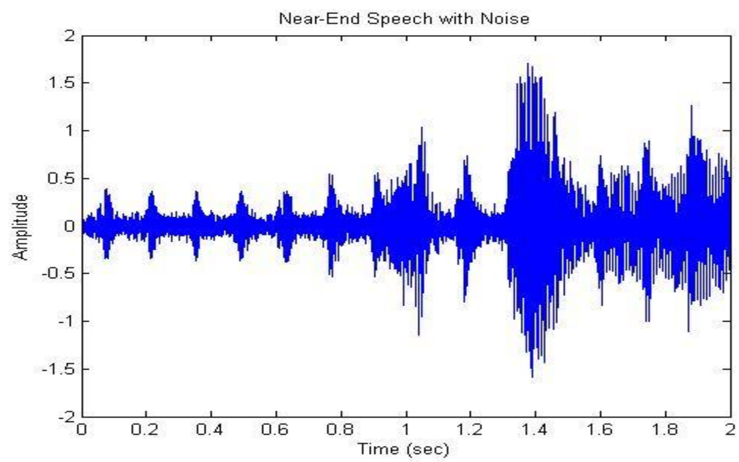
In order to remove the noise from the far-end speech signal, at first the estimation of the unknown secondary path is carried out by using the LMS filter 1, by hit and trial method the optimum value of the step size is 0.005 and the 12 tap order filter is used. The output from the LMS filter 1 serves as the desired signal for LMS filter 3 and LMS filter 4, these filters are used for the compensation of the secondary path estimated by LMS filter 1 and generated the co-related noise signal which is used to remove the unwanted noise from the actual far-end speech signal. The step size and the order of the LMS filter 3 and LMS filter 4 are selected as 0.0005, 0.0005 and 12 tap, 12 tap respectively.

The Near-End speech recorded signal ‘Active noise cancellation’ is mixed with the unwanted periodic noise of rotor as shown in fig 5.6(b) By using the LMS filter the periodic noise is removed from the corrupted information in order to get clear information which is free from the unwanted noise as shown in fig 5.6(c).

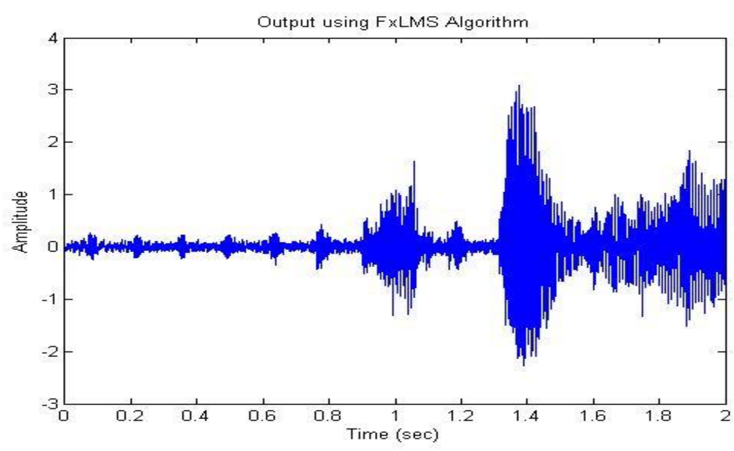
The Figure 5.3 shows the original Near-End information recorded as ‘Active noise cancellation’ which was interrupted by the periodic noise of the rotor. The periodic noise is removed from the actual information by using the LMS algorithm.



(a)

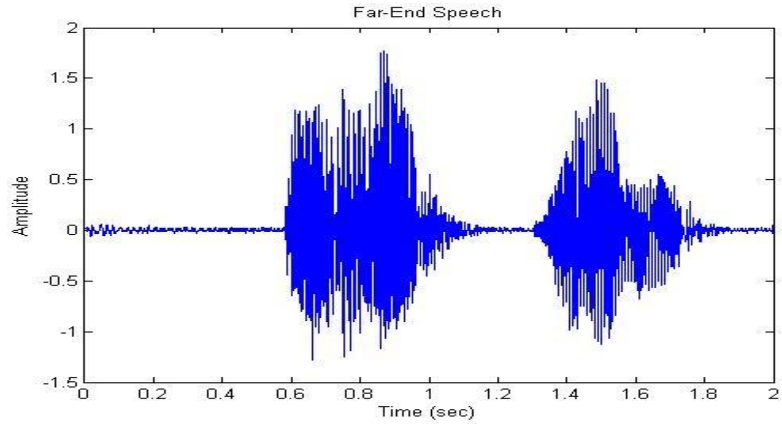


(b)

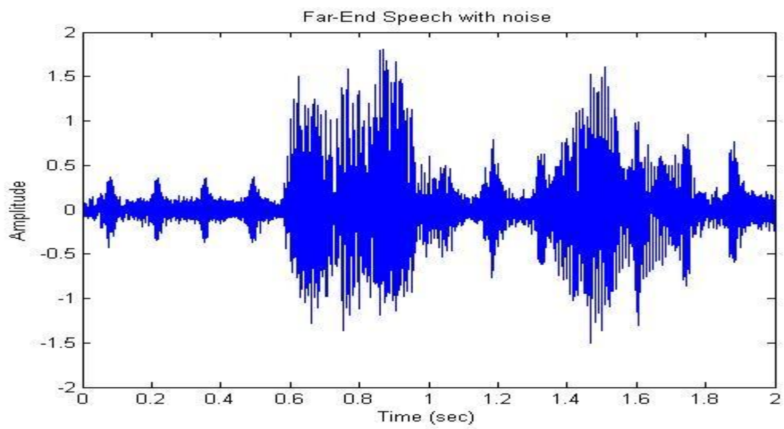


(c)

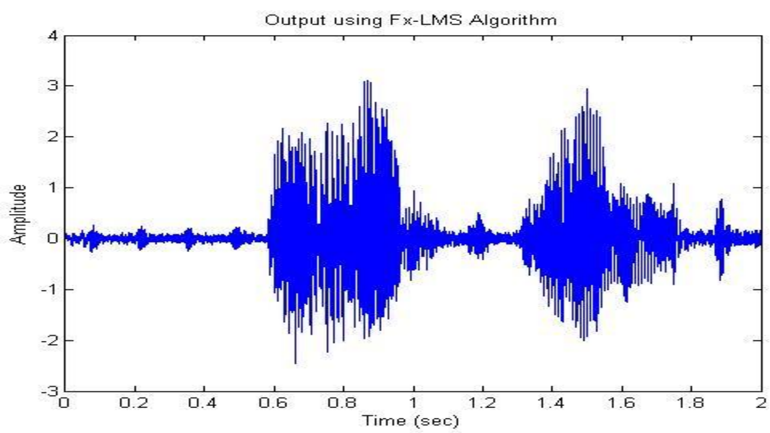
Figure 5.6: (a) Original near-end Speech (b) Near-end Speech with noise (c) Output using FxLMS Algorithm



(a)



(b)



(c)

Figure 5.7: (a) Original far-end Speech (b) Far-End Speech with Noise (c) Output using FxLMS Algorithm

Figure 5.4 shows the Far-End information signal which becomes unclear after corrupting with the periodic noise signal. The Far-End speech signal with noise is shown in figure 5.7 (b), while comparing it with the output using FxLMS algorithm as shown in figure 5.7 (c), it is clear that the system has reduced the unwanted interference without any cancelation of the received far-end information signal. At the same time the system has enhanced the near-end speech signal before transmitting it to the far-end.

CHAPTER 6

CONCLUSION AND FUTURE WORK

6.1 CONCLUSION:

An integrated Active Noise Cancellation using FxLMS algorithm for periodic noise was presented in this thesis. It is used to reduce the unwanted rotor periodic noise without canceling the far-end information signal, it also removes the noise from the near-end speech signal before sending it to the far end. The results of simulation have shown that unwanted noise cancellation using FxLMS Algorithm is achieved and the system is able to give high quality performance and noise reduction in the presence of external noise for both near-end and far-end information signal This provides a practical, cost effective, and lighter in weight solution.

The system provides the robust and fast adaptation to track the changes in the primary path and the secondary path of the system, also adapts the slight changes in the unwanted noise.

6.2 FUTURE WORK:

The successful application of ANC systems is highly dependent on the design and interface of subsystems: (1) the physical system, consisting the reference sensors, secondary sources, and error sensors; and (2) the electronic system that contains the input signals to generate the secondary control signals. The design of the physical system, including the arrangement of secondary sources and sensors, effects the maximum noise reduction that can be achieved by an ideal electronic controller. On the other hand, the quality of the electronic system, including the ANC algorithm and the digital word length of the DSP hardware, will also limit the performance.

The Future work of ANC may include:

- Implementation in mobile phones headsets, the voice signal from microphone can be filtered and improved, creating a high quality transmit link between persons talking.
- Controller can be designed for automobile industry for comfortable and quite vehicles.
- Transmission of engine induced vibrations through engine and power-train mounts into chassis can also be reduced.

BIBLIOGRAPHY:

[1] Arbetsmiljoverket Forfattningssamling Buller AFS 2005:16 Publikotionsservice, Solna, Sweden, 2005, ISBN 91-7930-455-9

[2] O.L.Angevine, "Active cancellation of the Hum of large Electrical Transformer", in proc Inter noise 1992, pp 313-316

[3] M.C.Atlie, C.D.Bremigan and L.J.Erickson, " Hardware and Software considerations for active noise control", in proc. ICASSP, 1998, pp 2598-2601

[4] Hansen C.H, " Understanding Active Noise Cancellation" London spon press, 2001

[5] Rossing T.D, Moore F.R, and Wheeler P.A. "The Science of sound", 3rd Edition, San Francisco, Addison Wesley 2002.

[6] Crocker M.J "Noise Control" in Crocker, M.J (editor) Encyclopedia of acoustic Vol: II, N.Y John Wiley and Sons, Inc. 1997.

[7] Hansen C.H and Synder S.D "Active Control of Noise and Vibration". London E and FN Spon 1997

[8] Sen M. Kuo and Dennis R. Morgan – Active Noise Control System, Algorithm and DSP Implementation; John Wiley and Sons, New York USA.

[9] Colin H. Hansen - Understanding Active Noise Cancellation.

[10] Shing Chow Chan, " A recursive least M – estimate algorithm for robust adaptive filtering in impulsive noise: Fast algorithm and convergence performance performance," IEEE Transaction on Signal Processing, vol 2,No-4, April 2004.

[11] Simon Haykin – Adaptive Filter Theory, 3rd ed, Englewood Cliffs,N.J.Prentice Hall, 1996.

[12] J.F. Abbot, " Acoustic design criteria for active noise control system," in proc. ICASSP, vol v, 1993, pp. 630-633.

[13] M.C.Atlie, C.D.Bremigan and L.J.Erickson, " Hardware and Software considerations for active noise control", in proc. ICASSP, 1998, pp 2598-2601

[14] A.Barbosa, Adaptive technique for active cancellation of periodic sounds, MS Thesis, Northern Illinois University, Aug 1991.

[15] M.Bellenger, Adaptive Digital Filter and Signal analysis,New York: Marcel Dekkar 1987.

[16] C.C.Boucher, S.J.Elliot, and P.A.Nelson, " Effect of errors in the plant model on the performance of algorithm for adaptive feed forward control," IEEE proc. Aug 1991.

[17] C.C.Boucher, S.J.Elliot, and P.A.Nelson, “The effect of error on the performance and stability of active noise control system,” in proc. Recent advances in active control of sound vibration, 1991, pp.290-301.

[18] D.R.Browning and D.R.Morgan, “Apseudo cascade FXLMS approach to active vibration cancellation,” in proc. Nooise Control, 1991 pp. 353-360.

[19] K.Char and S.M.Kuo, “ Performance evaluation of various active noise control algorithm,” In proc. Noise Control, 1994, pp. 331-363.

[20] Woon S. Gan and Sen M. Kuo, “Integrated Active Noise Control Communication Headsets,” Dept: of Electrical Engineering, N.Illinois University, IL, 60115, USA.

[21] Cheng-Yaun Chang and Deng-Rui Chen, “Active noise cancellation without Secondary path identification by using an adaptive Genetic algorithm,” IEEE transactions on instrumentation and measurement, vol.59, No. 9, September 2010.