2. LITERATURE REVIEW

Kuo S M and Morgan D.R (1999) in his review the Acoustic Noise Control traditionally involves passive methods such as enclosures, barriers and silencers to attenuate noise. These techniques use either the concept of impedance change or the energy loss due to sound absorbing materials. These methods are however not effective for low frequency noise. A technique to overcome this problem is Active Noise Cancellation (ANC), which is sound field modification by electracoustic means. ANC is an electro acoustic system that cancels the primary unwanted noise by introducing a canceling “antinoise” of equal amplitude but opposite phase, thus resulting in an attenuated residual noise signal as shown in Figure 2.1.

![Fig. 2.1. Physical concept of Active Noise Control.](image)

The design of ANC systems was first conceived in the 1930’s by Lueg. In the ensuing years, ANC has been the focus of a lot of research. An overview can be found in the tutorial paper by Kuo and Morgan and also in the book by the same authors. ANC systems are based either on feedforward control where a coherent reference noise input is sensed or feedback control where the controller does not have the benefit of a reference signal. Further, ANC systems are classified based on the type of noise they attempt to cancel as either broadband or narrowband. A brief overview of the various approaches to ANC follows next.
2.1 Broadband feed-forward Active Noise Control

Diniz P.S.R. (2002) in his book, the systems that have a single secondary source, a single reference sensor and a single error sensor. The single channel duct acoustic ANC system shown in Figure 2.2 is an example of such a system.

![Diagram of Single channel broadband feedforward Active Noise Control](image)

**Fig. 2.2. Single channel broadband feedforward Active Noise Control.**

This basic broadband ANC system can be described as an adaptive system identification framework as shown in Figure 2.3. Essentially, an adaptive filter $W(z)$ is used to estimate an unknown plant $P(z)$ which consists of the acoustic response from the reference sensor to the error sensor. The objective of the adaptive filter $W(z)$ is to minimize the residual error signal $e(n)$. However, the main difference from the traditional system identification scheme is the use of an acoustic summing junction instead of the subtraction of electrical signals. Therefore it is necessary to compensate for the secondary path transfer function $S(z)$ from the output of the adaptive filter till the point where the error signal gets recorded.
From Figure 2.3, we see that the $z$ transform of the error signal is given by
\[ E(z) = X(z) [P(z) - S(z)W(z)] \]  
Assuming that after convergence of the adaptive filter, the error signal is zero, $W(z)$ is required to realize the optimal transfer function
\[ W(z) = \frac{P(z)}{S(z)} \]  

The introduction of the secondary path transfer function in a system using the standard LMS algorithm leads to instability. This is because, it is impossible to compensate for the inherent delay due to $S(z)$ if the primary path $P(z)$ does not contain a delay of equal length. Also, a very large FIR filter would be required to effectively model $1/S(z)$. This can be solved by placing an identical filter in the reference signal path to the weight update of the LMS equation. This is known as the filtered-X LMS algorithm. The block diagram of an ANC system using the FXLMS algorithm is shown in Figure 2.4.
Fig. 2.4. ANC system using the FXLMS algorithm

A rudimentary explanation of the FXLMS algorithm is presented below. In figure 2.4, the residual error signal can be expressed as

$$E(n) = d(n) - s(n) \ast [w^T(n)x(n)]$$  \hspace{0.5cm} (2.3)

Where $s(n)$ is the impulse response of the secondary path $S(z)$ at time $n$. Assuming a mean square cost function $\bar{\xi}(n) = E[e^2(n)]$, the adaptive filter minimizes the instantaneous square error $\dot{\bar{\xi}}(n) = \dot{e}^2(n)$ according to

$$w(n+1) = w(n) - \frac{\mu}{2} \nabla \dot{\bar{\xi}}(n)$$  \hspace{0.5cm} (2.4)

Since

$$\nabla \dot{\bar{\xi}}(n) = -2x'(n)e(n)$$  \hspace{0.5cm} (2.5)

The weight update equation reduces to

$$W(n+1) = W(n) + \mu X'(n)e(n)$$  \hspace{0.5cm} (2.6)

In practical applications, the secondary path transfer function $S(z)$ is unknown and must be estimated by an additional filter $\hat{S}(z)$. Therefore, $x'(n) = \hat{S}(n) \ast x(n)$, where $\hat{S}(n)$ is the impulse response of $\hat{S}(z)$. As shown by Morgan, the FXLMS algorithm seems to be remarkably tolerant to errors in the estimation of $S(z)$ by the filter $S(z)$ and within the limit of slow adaptation, the algorithm will converge with nearly $90^\circ$ of phase error between $S(z)$ and $S(z)$. Therefore, offline modeling
techniques can be used to model $S(z)$. Nelson and Elliot showed that the maximum step size that can be used with the FXLMS algorithm is given by

$$\mu_{\text{max}} = \frac{1}{P_x'(L + \Delta)}$$

(2.7)

Where $P_x' = E[x'^2(n)]$ is the power of the filtered reference signal and $\Delta$ is the number of samples corresponding to the overall delay in the secondary path. However, errors in estimating the secondary path transfer function will alter the stability bounds on $\omega$. A detailed analysis of the stability criterion is available in the literature.

In the feed-forward ANC system shown in Figure 2.3, the anti-noise output of the speaker also radiates upstream to the reference microphone resulting in acoustic feedback and hence a corrupted reference signal $x(n)$. Instability will occur if the open loop phase lag reaches $180^\circ$ and the gain is greater than unity. This can be solved by using a separate offline adaptive feedback cancellation filter within the ANC system. Feedback can also be solved by using an adaptive IIR filter in place of the FIR filter in the ANC system. However, IIR filters are not unconditionally stable, as adaptation may converge to a local minimum and can have relatively slow convergence rates. A detailed analysis of adaptive IIR filters is available in the literature.

2.2 Narrowband Feed-forward ANC

Boucher C, Elliott S J and Nelson P A, (1990), proceeding article discussed, many noise sources are periodic in nature such as engines, compressors, motors, fans, etc. In such cases, direct observation of the mechanical motion using an appropriate sensor is used to provide an electrical reference signal which consists of the primary frequency and all the harmonics of the generated noise. The basic block diagram is as shown in Figure 2.5.
This technique avoids the undesired acoustic feedback to the reference sensor, as well as nonlinearities and aging problems with acoustic microphones. The periodicity of the noise removes the causality constraint, as each harmonic can be controlled independently and a much shorter FIR filter can be used to model the secondary path.

2.3 Feedback Active Noise Control

Vijayan D (1994) in his research thesis discussed, the Feed forward ANC systems (broadband and narrowband) use a reference sensor to measure the primary noise signal, a feed forward adaptive filter and an error sensor to measure the residual error signal. However, in some applications, it is not feasible to have a sensor to measure or internally generate the error signal. This section describes a class of algorithms known as feedback ANC in which the reference signal is generated from the output of the error sensor. This is used in applications that combat spatially incoherent noise generated from turbulence, noise generated from many sources and propagation path induced resonance where no coherent reference signal is available.

2.3.1 Non-Adaptive Feedback Active Noise Control

Jacobson C A, Johnson Jr. C R, McCormick D C and W A Sethares (2001) is explained the non-adaptive feedback ANC system that used an amplifier carefully matched to the response of the error sensor and the secondary source. Hong
and Eghtesadi studied the application of feedback ANC for noise reduction in a duct. Veit, Carme and Wheeler studied the application of feedback ANC for noise compensation in personal hearing protectors. The basic block diagram of a classical feedback ANC system is as shown in Figure 2.6

![Fig. 2.6. Classical feedback ANC system.](image)

In Figure 2.6 above, \( d(n) \) is the primary noise at the error sensor location, \( e(n) \) is the residual noise, \( y(n) \) is the secondary antinoise signal, \( W(z) \) is the transfer function of the controller and \( S(z) \) is the transfer function of the secondary path. Under steady state conditions, the z-transform of the error signal can be expressed as

\[
E(z) = D(z) - S(z)W(z)E(z)
\]

\[
E(z) = D(z) / 1 + S(z)W(z)
\]

Therefore the closed loop transfer function \( H(z) \) from the primary noise to the error signal can be expressed as

\[
H(z) = E(z) / D(z) = 1 / 1 + S(z)W(z)
\]

From equation 2.9 the power spectrum of the error signal is given by

\[
S_{ee}(w) = \frac{1}{|1 + S(w)W(w)|^2} S_{dd}(w)
\]

Where \( S_{ee}(w) \) and \( S_{dd}(w) \) are the power spectra of the error signal \( e(n) \) and the reference noise \( d(n) \) respectively. Therefore, in order to minimize \( S_{ee}(w) \), we need to minimize \(|1+S(z)W(z)|^2\), or the gain of \( S(z)W(z) \) should approach infinity. If the
frequency response of $S(w)$ is flat, then the gain of $W(w)$ can be increased without limit so that the overall transfer function of the feedback loop becomes marginal. However, this is rarely the case as the response of the secondary source introduces a significant phase shift and there is some propagation delay from the output of the control filter to the error sensor.

These effects introduce a phase shift in $S(w)$ that increases with frequency. As the phase shift approaches $180^\circ$, the desired negative feedback becomes positive feedback leading to instability. Therefore as the frequency and phase shift increase, the gain of $W(w)$ should decrease. Hence it is possible to design an inverting amplifier $W(w)$ provided the gain is not large enough to make the net loop gain greater than unity when the phase shift is $180^\circ$. Therefore, if

$$S(w)w(w) = G(w)e^{j\phi(w)}$$  \hspace{1cm} (2.12)

$$|1+S(z)W(z)|^2 = 1 + G^2(w) + 2G(w)\cos\phi(w)$$ \hspace{1cm} (2.13)

Given a secondary path $S(w)$, $W(w)$ needs to be chosen such that the net gain $G(w)$ is maximized when $-180^\circ < \phi(w) < 180^\circ$. A more detailed explanation of the design of feedback ANC system is available in the literature.

2.3.2 Single channel Adaptive Feedback Active Noise Cancellation

Tohma S (1991) in his journal, the adaptive single channel Active Noise Cancellation system was first proposed by Eriksson and then extended to the multi channel scenario by Popovich. This technique is generally viewed as an adaptive feedforward ANC system that in effect synthesizes its own reference signal. Under certain conditions, the system can also be interpreted as an adaptive predictor.

In the feedback ANC system shown in Figure 2.6, the primary noise signals $d(n)$ is not available. Therefore, the main idea of an adaptive feedback ANC system is to regenerate the reference signal $d(n)$ from the error signal. From Fig 2.6, we can see that the primary noise can be expressed in the z-domain as

$$D(z) = E(z) + S(z)Y(z)$$  \hspace{1cm} (2.14)

Where $E(z)$ is the residual error signal obtained from the error signal and $Y(z)$ is the output of the adaptive filter. The secondary path transfer function $S(z)$ can also be estimated as $S(z)$. Thus we can estimate the primary noise $d(n)$ and use this as a synthesized reference signal $x(n)$ as follows
\[ X(z) \equiv \hat{D}(z) = E(z) + \hat{S}(z)Y(z) \]  

This is illustrated in Figure 2.7.

\[ Y(z) = W(z) X(z) \]  

We know that the error signal in Figure 2.8 can be expressed in the z-domain as

\[ E(z) = D(z) - S(z) Y(z) \]  

Where

\[ Y(z) = W(z) [E(z) + \hat{S}(z)Y(z)] \]  

Rearranging equation 2.22 we get

\[ [1 - \hat{S}(z)W(z)]Y(z) = W(z)E(z) \]  

\[ Y(z) = \frac{W(z)E(z)}{1 - \hat{S}(z)W(z)} \]  

Substituting equation 2.24 in 2.20, we get

\[ E(z) = D(z) - S(z) \left[ \frac{W(z)E(z)}{1 - \hat{S}(z)W(z)} \right] \]  

\[ \left[ 1 + \frac{S(z)W(z)}{1 - \hat{S}(z)W(z)} \right]E(z) = D(z) \]  

Fig. 2.7. Adaptive feedback ANC system using synthesized reference signal.
Assuming $S(z) = S(z)$, equation 2.28 simplifies to
\[ H(z) = \frac{E(z)}{D(z)} = 1 - S(z) W(z) \]
Therefore under ideal conditions, the feedback ANC system is transformed to a feedforward ANC system.

A number of alternate schemes have been proposed for feedback ANC. Oppenheim and Zangi proposed a feedback ANC scheme based on the block Expectation Maximize algorithm. Openheim et al proposed a scheme based on the RLS algorithm. However, these algorithms have been generated with ideal conditions and initial conditions need to be very carefully generated to ensure that the algorithm converges. Eriksson et al proposed a generalized recursive ANC scheme that uses three adaptive filters that accurately model the primary path, the feedforward path and the feedback path for the filtered X or filtered U algorithms. Performance analysis of all these algorithms have shown that the noise attenuation is very noticeable in the lower frequency regions below 1 kHz and deteriorates very rapidly as the center frequency of the noise increases.

### 2.4. Hybrid Active Noise Control systems

Kuo S M and Morgan D R (1996) on their book a combination of the feed-forward and feedback ANC schemes is known as the Hybrid ANC scheme. Here, the canceling signal is generated based on the inputs of both the reference sensor and the error sensor. This method was first proposed by Swanson. The motivation behind this method is to increase the correlation between the primary noise and the signal picked up by the reference sensor. Since the error sensor is generally placed downstream from the source of the primary noise and the reference sensor is places as close as possible to the primary source. However, it is often the case that the reference sensor may not pick up all the acoustic cues of the primary noise source and this can be rectified by using the signal at the error sensor also to generate the canceling output.
The block diagram of the Hybrid ANC scheme is as shown in Figure 2.10.

Fig. 2.8. Active Noise Cancellation system with both feedback and feedforward control loops

The system in Figure 2.10 can be configured with the feedforward path as any one of the following:

1. Filtered-X LMS algorithm
2. Filtered-X LMS algorithm with feedback cancellation
3. Filtered-U recursive LMS algorithm

The feedback path can be configured to be

1. Classical feedback ANC
2. Feedback ANC using Filtered-X LMS algorithm
3. Output whitening method

The canceling signal fed to the speaker is generally an unweighted sum of the outputs of both algorithms. This hybrid scheme was better at canceling broadband noise, when compared to the feedforward or feedback schemes by themselves. Furthermore, she showed that it is possible to reduce the length of the filters in the hybrid scheme.
2.5 Multi channel Active Noise Cancellation

Morinushi K (1991) in his journal, it is desirable to cancel noise at several locations in a three dimensional space. Single channel systems are effective when the area of interest is restricted and there is only a single primary source that can be accurately located and a single “quiet zone” where the error sensor needs to be located. However, many practical applications involve relatively large multidimensional spaces where the noise source cannot be accurately pointed to be at one single location. The complexity of multi channel ANC in a multi dimensional space is however significantly higher and the system needs to be carefully ported to a practical real world application.

Elliott S J and Nelson P A (1993) in his journal explained, the multichannel feed forward ANC system can thus be viewed to be a combination of single channel feedforward systems, with the exception that there are multiple secondary paths from each of the adaptive filters to each of the error sensors. Multichannel feedforward algorithms are also generally prone to feedback due to the larger number of error and reference sensors. It has also generally been found that IIR adaptive filters are more effective than FIR filters in multi channel systems. Multi channel systems have also been implemented using the feedback ANC algorithm using either a K X 1 system with K reference sources and a single error source or a K X M system with K reference sensors and M error sensors.

2.6 Design of Adaptive noise cancellation for speech signals using GES method

Manikandan S, Madheswaran M (2008) were dealt in their paper about reducing the content of noise present in the received Speech signals for wireless communication medium are using Wavelet Grazing Estimation of signal Method. The received signal is corrupted due to mixing of white Gaussian noise. This new method is designed based on superposition principle and eight cases of signal varied positions. This new method output is fed to the wavelet technology compare the existing LMS, RLS algorithm using Matlab6.5 simulation results. Finally, this new technology is more effective for noise reduction application as compare to the existing methods.