International Journal of Advanced Computer Research (ISSN (print): 2249-7277 ISSN (online): 2277-7970)
Volume-4 Number-2 Issue-15 June-2014

Application of High Order X-LMS Filter for Active Noise Control

N.V.K.Mahalakshmi[1], D.Sindhu[2], B. Ranjith Kumar[3], N.Santhisri[4], G. Sowjanya Rao[5]

Abstract

In this paper active noise is controlled by using higher order X-LMS (least mean square) filter. This technique is based on X-NLMS (normalized least mean square), also known as traditional acoustic noise cancellation (ANC) scheme. It cancels the wideband noise from the corrupted speech signal. The active noise reducing headphone is probably the most successful application of active control of sound – the technology of cancelling sound with sound i.e., by using anti-noise signal. This report presents an outlined technical review of noise cancellation in headphones. The principles of passive noise attenuation are presented after which active attenuation is introduced showing how the two complement the attenuation performance. In real-time environment, the number of different applications in which adaptive techniques are being successfully used that are echo cancellation, equalization of dispersive channels, system identification, signal enhancement, noise cancelling and control.

Keywords

X-LMS filter, ACTIVE NOISE, X-NLMS filter, corrupted speech signal.

1. Introduction

In many of the situations that today's consumers find themselves hard to achieve the needed peace and quiet required to focus on the task at hand. Being able to rest during a crowded commute is another important concern for many people. Advancements in technology and design have made overcoming these obstacles easier using noise-cancelling headphones. They allow wearers to go about their daily activities without the intrusion of the noisy, disruptive outside world. The technique used in the noise cancelling headphones is active noise control.

Active noise control (ANC) is a method of reducing the unwanted disturbances by the introduction of controllable secondary sources. In this process the outputs are arranged to interfere destructively with the disturbance from the original primary source. If the output generated by the active noise control system will not give the anti-noise signal completely to compensate the noise then the new anti-noise signal will be generated.

Noise cancellation in headphones relies on the acoustic isolation characteristic of headphones with active noise reduction. Active headphones are used mainly in highly noise environments to protect the user from the excessive noise. Attenuation of noise in headphones can be done in two ways. They are:-
1) Passive attenuation.
2) Active attenuation.

Passive attenuation takes place when the incoming signal is blocked or attenuated by the headphone shell covering the ear. This is most effective at high frequencies. In the active attenuation of sound a loudspeaker placed inside the headphone shell produces the anti-noise signal to cancel the external noise. Active attenuation works well at low frequencies. A good headphone will effectively combine low frequency active attenuation with high frequency passive attenuation to provide high attenuation of the external noise.

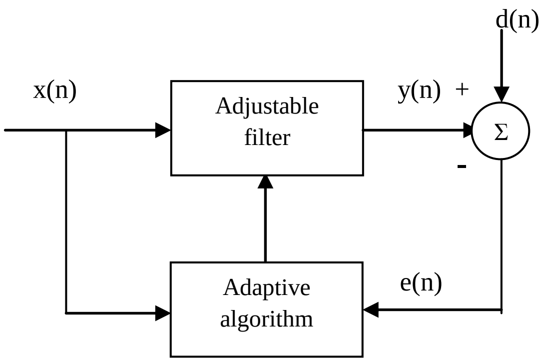
Applications of active noise control are mainly used in industries, automobiles, and in military [1], [2], [3].

2. Adaptive Filter

Noise cancelling headphones uses adaptive filters for generating anti-noise signal to reduce the noise by using destructive interference. An adaptive filter is a self adjusting filters in which the coefficients of the filter will be based on the input signal i.e., error

Manuscript received June 16, 2014.
N. V. K. Mahalakshmi, Assistant Professor, ECE Department, SRKIT, Vijayawada, India.
D. Sindhu, UG Scholar, ECE Department, SRKIT, Vijayawada, India.
B. Ranjith Kumar, UG Scholar, ECE Department, SRKIT, Vijayawada, India.
N. Santhisri, UG Scholar, ECE Department, SRKIT, Vijayawada, India.
G. Sowjanya Rao, UG Scholar, ECE Department, SRKIT, Vijayawada, India.

signal given to it. The block diagram of adaptive filters is as follows:



x(n)-input signal y(n)-output signal
e(n)-error signal d(n)-desired signal

Figure: 1 General adaptive filter

Initially, when the input signals x(n) is given to the filter and the estimate of d(n) i.e., y(n) is given by the filter which has same amplitude but opposite phase shift when compared to d(n). The error signal is given as

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

To generate the anti-noise signal an adaptive algorithm is used in the adaptive filter. There are mainly two types of algorithms that are used in the adaptive filter. They are:

1) RLS algorithm.
2) LMS algorithm.

RLS algorithms are a class of adaptive filter which recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to input signals [1].

LMS algorithms are a class of adaptive filters that are used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal [1].

Applications of adaptive filters are system identification, channel equalization, signal prediction, noise cancellation.

3. LMS Algorithm

LMS algorithm has become one of the widely used algorithms in adaptive filtering. LMS algorithm generally consists of two basic processes:

1. A filtering process, which involves
   a) Computing the output of a linear filter in response to an input signal.
   b) Generating an estimation error by comparing this output with a desired response.
2. An adaptive process, which involves the automatic adjustment of the parameters of the filter in accordance with the estimation error.

The block diagram of noise cancellation using LMS algorithm is as follows:

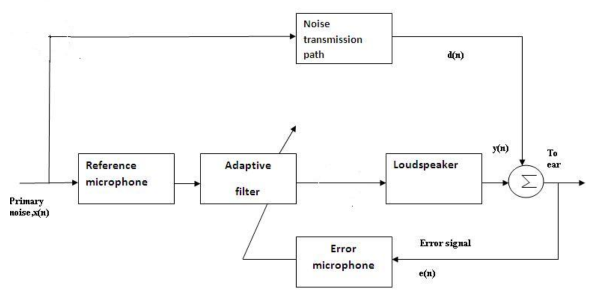


Figure: 2 Block diagram of noise cancellation using LMS algorithm

In LMS algorithm an input signal x(n) is given to the reference microphone and it acts as a transducer and converts sound signal to the electrical signal. The output of the transducer will be given to the adaptive filter and it will generate the estimate of d(n) i.e., y(n) and if the anti noise signal does not reduce the noise completely then the error signal will be given to the adaptive filter until the noise will be reduced.

We don’t know the exact parameters of the reference signal received at the reference microphone from the primary source and the signal at the error sensor is unknown. Also there are more than one disturbance sources and so there should be multiple-channel feed forward systems that control stochastic or random disturbances. Time domain formulation is needed in such cases. Active control at a number of error sensors is achieved by detecting the waveform of the primary sources with a number of noise sensors, and feeding these signals through a matrix of control filters to a set of secondary sources. Figure 2 shows the typical block diagram of such a case. The main aim of the sensors is to get the noise signal and the reference signal strength and if the noise signal strength is high then we will generate an anti-noise signal to reduce the noise signal. The reference signals are fed to a matrix of adaptive filters whose outputs are used to drive secondary sources, with output signals y(n). The output of the adaptive filter will be

$$y(n) = \sum_{i=0}^{L-1} wi(n)x(n-i)$$
$$= W^T(n)X(n)$$

where ,X(n)=[x(n) x(n-1)………… x(n-L+1))] is a vector of input signal samples.
$W(n)=[w_0(n)\ w_1(n)\ldots\ldots\ldots W_{L-1}(n)]^T$ is a vector containing the coefficients of the FIR filter at time n.

The LMS algorithm is not applicable in all the conditions. In active noise control applications the output of the adaptive filter drives the secondary path, and the error signal is derived only at the microphone. In such cases the simple LMS algorithm is unstable due to the phase shift caused by the secondary path. The problem is analysed and the solution is so-called filtered reference or filtered X-LMS (XLMS) algorithm. This algorithm requires a model of the secondary transfer function, which the reference signal is filtered by [2].

4. Filtered X-LMS Algorithm

In the ideal situation the error signal is found as the difference between the primary noise signal previously referred to as d(n) and the estimated signal y(n) in the absence of the last sub-section, using adaptive filtering for ANC in headsets involves a number of physical transducers. Another disadvantage is the loudspeaker which acts as a secondary source, can cause problems as the transfer function of the system must be reversible. To reduce the loudspeaker transfer function problem the filter must be placed in before the loudspeaker. However, the phase systems of the loudspeakers are not minimum and not invertible. The problem is reduced by filtering the input signal using adaptive filter algorithm by using the estimate of transfer function of the loudspeaker. There are mainly two types of adaptive algorithm. They are LMS algorithm or the normalized LMS algorithm, which is similar to LMS algorithm. This modification results in the filtered-x LMS algorithm.

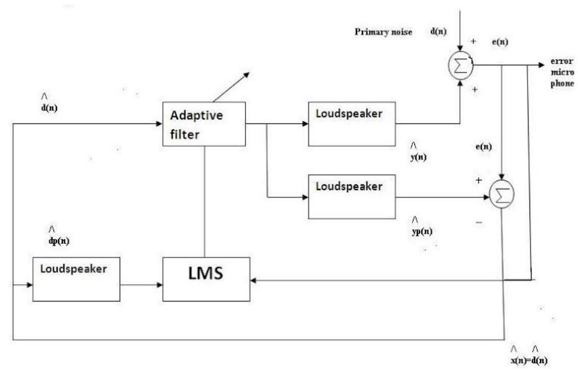


Figure: 3 Block diagram of the active noise control feedback configuration of filtered X-LMS algorithm

To account the effects of the secondary path transfer function the conventional LMS algorithm is modified. To ensure convergence of the algorithm input to the error correlated is filtered by secondary path estimate. This results in filtered X-LMS (FXLMS) algorithm. Filtered X-LMS algorithm is illustrated in figure 3.

In filtered X-LMS algorithm initially an estimated loudspeaker is used and the comparison of the primary noise with the estimated loudspeaker takes place. The main function of the estimated loudspeaker is to generate the anti-noise signal. At the initial stage the error obtained is very large because we don’t know the exact parameters of the primary noise signal. The error obtained after comparison will be given to the LMS algorithm and it gives the required coefficients to the adaptive filter and the output from the adaptive filter will be given to the new loudspeaker system for generating the anti-noise signal. At this stage the comparison of the signals from the output of the new loudspeaker and initially obtained error signal takes place and this process will continue until the error signal reduces to zero. The filtered X-LMS algorithm is explained by [4], [5], [6], [7].

The output y (n) is computed as

$$y(n) = \sum_{i=0}^{L-1} w_i(n)\,x(n-i)$$
$$= W^T(n)X(n)$$

Where
X(n)=[x(n) x(n-1)………… x(n-L+1))]is a vector of input signal samples,
$W(n)=[w_0(n)\ w_1(n)\ldots\ldots\ldots W_{L-1}(n)]^T$ is a vector containing the coefficients of the FIR filter at time *n*.

The filtered X-LMS algorithm can be expressed as
W(n+1)=W(n)+μe(n)X(n)
where μ is the step size of the algorithm that determines the convergence and stability of the algorithm.

The error signal is expressed as
e(n)=d(n)-y(n)
$e(n)=d(n)-W^T(n)X(n)$

Mean Square Error solution of the coefficient vector is obtained by minimizing the quadratic function

$$J(n) = E\left\{\left(e(n)\right)^2\right\}$$

$$= E\{\left(d(n) - W^T(n)X(n)\right)^2$$

$$J(n) = E[d^2(n)] - 2W^T E[d(n)X(n)] + W^T E[X(n)X^T(n)]$$

$$= E[d^2(n) - 2W^T P + W^T RW]$$

$R = E[X(n)X^T(n)]$ is the auto correlation matrix of the input vector.
$P = E[d(n)X(n)]$ is the cross correlation of the desired response signal and the input signal.

Since R and P are not time varying and stationary and this can be carried out by using the gradient vector for the mean square error J (n). From above equations

$$\nabla_w J(n) = -2P + 2RW$$

J(n) is a quadratic, non-negative function of the coefficient vector. To minimize the error function equates it to zero.

$$\nabla_w J(n) = 0$$

If the autocorrelation matrix R_{XX} (n) is invertible, the cost function has a unique minimum given by

$$W_{opt}(n) = R_{XX}^{-1}(n)P(n)$$

Our objective is to iteratively descend to the bottom of the cost function surface, so that W(n) approaches $W_{opt}(n)$, using a strategy analogous to that of the ball rolling in a bowl.

The weight update algorithm plays a major role in the filtered X-LMS algorithm. These facts suggest an iterative approach for finding the parameter value associated with the minimum of the cost function: simply moves the current parameter value in the direction opposite to that of the slope of the cost function at the current parameter value.

Furthermore, if we make the magnitude of the change in the parameter value proportional to the magnitude of the slope of the cost function, the algorithm will make large adjustments of the parameter value when its value is far from the optimum value and will make smaller adjustments to the parameter value when the value is close to the optimum value. This approach is the essence of the steepest descent algorithm.

$$W(n+1) = W(n) - \frac{\mu}{2}\frac{\partial^2 E(e^2(n))}{\partial W^2}$$

$$\frac{\partial^2 E[e^2(n)]}{\partial w^2} = \nabla_w J(n)$$

$$\nabla_w J(n) = 2e(n)\frac{\partial e(n)}{\partial w}$$

$$\nabla_w J(n) = 2e(n)\nabla_w e(n)$$

$$\nabla_w J(n) = 2e(n)\nabla_w[d(n) - W^T(n)X(n)]$$

$$= -2e(n)X(n)$$

$$W(n+1) = W(n) - \frac{\mu}{2}(-2e(n)X(n))]$$

$$W(n+1) = W(n) + \mu e(n)X(n)$$

$$W(n+1) = W(n) + \mu X(n)[d(n) - y(n)]$$

$$W(n+1) = W(n) + \mu X(n)[d(n) - W^T(n)X(n)]$$

$$W(n+1) = W(n) + \mu[R - WP]$$

When this algorithm is implemented, the convergence of the filter can be achieved much more quickly than theory suggests, and the algorithm appears to be very tolerant of errors made in the estimation of the secondary path by the filter.

5. Results

The results obtained for active noise control is done by using MATLAB software. In this MATLAB software we will estimate the delays and the functioning of the primary and secondary propagation path which plays a major role in reducing the noise signal. The steps that are used in reducing the noise signal in MATLAB software is as follows:

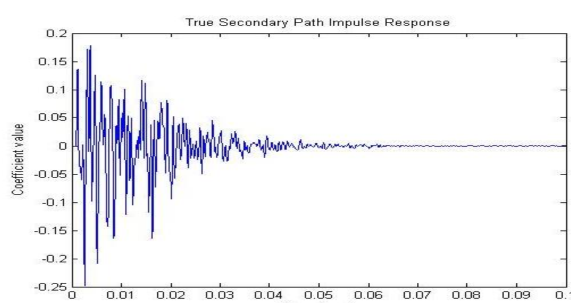


Figure: 4 Secondary propagation path impulse response

Figure 4 shows the output for the anti-noise signal generated from the output loudspeaker to error microphone within the quiet zone.

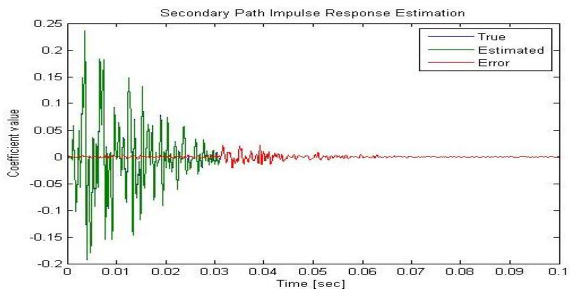


Figure: 5 Impulse response estimation of the secondary path

Figure 5 shows the coefficients of the true, error, and the estimated path. The impulse response estimation can be generated for the secondary path.

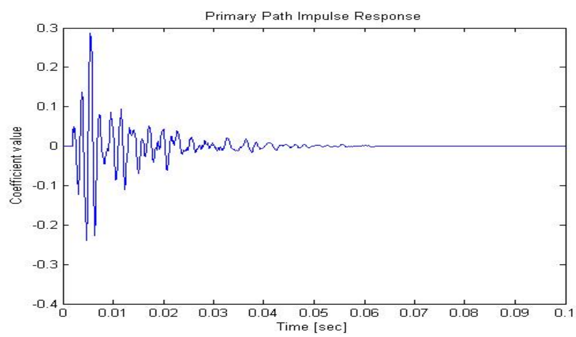


Figure: 6 Primary propagation path

Figure 6 shows the impulse response of the primary propagation path of the noise to be cancelled can also be characterized by using a linear filter. The primary propagation path shows the output from input to the error microphone.

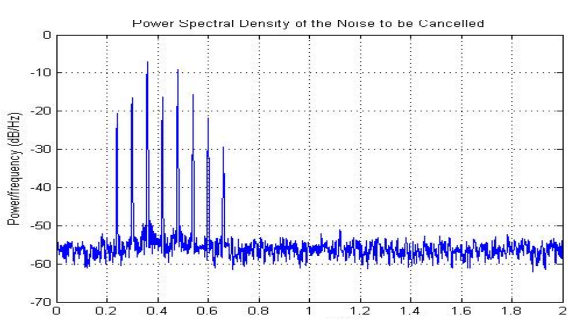


Figure: 7 Noise to be cancelled

Figure 7 shows the spectrum of the at the error microphone before cancellation is plotted. Here we will generate the noise which is to be cancelled.

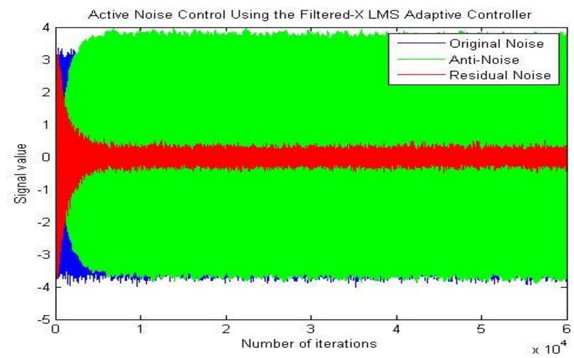


Figure: 8 Active noise controls using the Filtered X-LMS algorithm

The most popular adaptive filter used is Filtered X-LMS algorithm and it uses the secondary path estimate method for reducing the noise using destructive interference. Figure 8 shows the reduction of annoying sound by using filtered X-LMS algorithm.

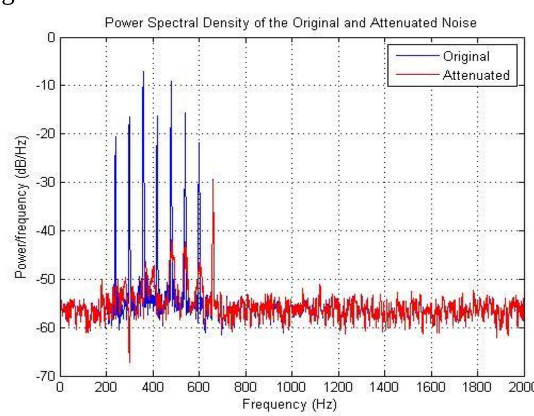


Figure: 9 Residual error signal spectrum

Figure 9 shows the comparison of spectrum of the residual error signal with that of original noise signal. The noise signal that is present at the initial stage will not be attenuated because some delay is present in the system at the initial stage for generating the anti-noise signal using the secondary path estimate based on the primary noise signal given to the system.

6. Conclusion

The passive and active attenuation methods used in headphones for noise cancelling have been discussed. It is found from the study that both the passive and active attenuation complements each other, good passive attenuation at high frequencies and good active control at low frequencies. Also the digital model is best suited at narrowband frequencies or tonal frequencies whereas the analog model gives good performance for broadband noise cancellation. The best and the optimal solution is to combine all the three characteristic models in a single headphone to use it at a wide range of frequencies. The development of active noise control systems in the field of acoustics, particularly headphones, has reached a stage where commercial systems are available for protection from noise in a wide variety of applications.

7. Future Prospects

Some of the algorithms like filtered X-LMS algorithm have been tested for headphones on DSP processors. Experiments have shown drastic improvement in the cancellation of noise in headphones. However for most of these algorithms, the noise cancellation parameters are taken in the absence of audio signal and hence they have to be updated in case of change of noise source or addition of another noise source. There are other algorithms like recursive least squares (RLS) algorithm, which is deterministic in the nature of the noise and hence can be used for real time cancellation of noise without the need for offline updating of its parameters, and Kalman filtering approach, which has a faster rate of convergence. However due to the computational complexity of these algorithms and the cost involved, there have not been much research on the implementation of these algorithms in noise cancelling headphones. So it will take a few more years to implement these algorithms on the DSP processors to achieve a better improvement in the cancellation of noise in headphones.

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N. V. K. Mahalakshmi was born on 22nd August 1984. She received her B.Tech degree in Nimra College of Engineering in the year 2005 and completed her M.Tech in Godavari institute of technology 2012. She is presently working as Assistant Professor in SRK Institute of Technology and she has 5 years of teaching experience.