Review Paper on Noise Cancellation using Adaptive Filters

Deepanjali Jain*[1]
Research Scholar, Om Sterling Global University
Hisar, Haryana, India

Dr. Poonam Beniwal[2]
Associate Professor, Om Sterling Global University
Hisar, Haryana, India

***Abstract :-* This paper reviews the past and the recent research based on adaptive noise cancellation system using Adaptive filter algorithms. Adaptive noise cancellation is a wide area of research in the field of communication and is used for noise reduction in speech signals. In many applications, the change in the received signals could be very fast which requires the use of adaptive algorithms that converge rapidly. This paper deals with cancellation of noise in speech signal using Least Mean Square (LMS) adaptive algorithms that provides efficient performance with less computational complexity.**

Keywords- Noise; Adaptive noise cancellation (ANC); Least Mean square (LMS) Algorithm; Adaptive filtering.

I. INTRODUCTION

Any signal which travels in the environment gets distorted by the noise present in it. Removal of this noise, so as to obtain the original signal, is one of the biggest challenges for everyone [1]. Due to these reasons, noise control has gained considerable importance in recent years which has led to tremendous amount of research on this topic to find some technique for its reduction. Many researchers have proposed different definitions of noise as a result of its broad category of existence. In general, the most common definition states- "Noise is arbitrary, unwanted electrical energy that enters the communications system through the communicating medium and obstructs with the conveyed message".

As the noise from the surrounding environment severely reduces the quality of speech and audio signals, it is quite necessary to suppress noise and enhance speech and audio signal quality, hence the acoustic applications of noise cancellation have become the thrust area of research [2,3]. As the noise from the surrounding environment severely reduces the quality of speech and audio signals it is quite necessary to suppress noise and enhance speech and audio signal quality, hence the acoustics applications of noise cancellation has become the thrust area of research.

The traditional approach to acoustic noise cancellation uses passive techniques such as using earplugs, ear-protector, sound insulation walls, muffler ,enclosures, barriers and silencers to remove the unwanted noise signal [4]. Passive Noise Cancellation (PNC) uses physical object that is installed in the system to isolate the background noise from the surrounding. PNC techniques are effective in reducing noise over a wide frequency range. However, they require relatively large and costly materials, and are ineffective at low frequencies [5].

Therefore, the Active Noise Control (ANC) was proposed in the early 20th century, which has gained intensive development in the last two decades to reduce low-frequency noise [6]. ANC employs an electro-acoustic system to cancel the primary noise based on the principle of superposition, where an anti-noise (secondary signal) of equal amplitude but with antiphase is generated by secondary source(s) and combined with the unwanted primary noise. Thus, achieving the cancellation of noise [7].

ANC has greater advantages over PNC in terms of the capability to attenuate the low frequencies noise due to the presence of embedded control system whereas the passive noise control techniques are expensive, bulky, and ineffective. ANC had been used in the microcontroller and achieved the cancellation of 20dB to 25dB of the 60Hz periodic noise component in the electronics signal measurement [8]. However, the implementation of ANC is complicated although it has high reliability in cancelling noise.

Signals are carriers of information, both useful and unwanted. Extracting or enhancing the useful information from a mix of conflicting information is a simplest form of signal processing. Signal processing is an operation designed for extracting, enhancing, storing, and transmitting useful information. The known technique used to estimate the signals distorted by noise is to pass it through the system or filter that have a tendency to overturn the noise while leaving the signals unchanged. This type of signal estimation is known as direct filtering. The design of such filter was originated by wiener and has emerged as the most important techniques in this area. Filters which are accessible for direct extracting are called fixed filter, which needs previous information of both signal and noise, which implies that if we identify properties of the signal and noise beforehand, we implement the system that permits frequency that contains the wanted signal and block the frequency band taken by noise signal. This type of filters are working under stationary conditions.

In contrast to the conventional filter design techniques, a basic concept was introduced by Widrow, in which noise was removed or suppressed from a signal using Adaptive Filters [9] which is a digital filter that has self-adjusting characteristics. It is capable of adjusting its filter coefficient automatically to adapt the input signal via an adaptive algorithm. Adaptive filters play an important role in modern digital signal processing (DSP) in areas such as telephone echo cancellation, noise cancellation, equalization of

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 11 Issue 01, January-2022

communications channels, biomedical signal enhancement, active noise control (ANC), and adaptive control systems. The usage of adaptive filters is one of the most popular proposed solutions to reduce the signal corruption caused by predictable and unpredictable noise. Adaptive filters have been used in a broad range of applications for nearly five decades. It includes adaptive noise cancellation, linear prediction, adaptive system identification, adaptive equalization, inverse modelling etc. Noise is assumed to be a random process and adaptive filters have the capability to adjust their impulse response to filter out the correlated signal in the input. They require modest or no a priori knowledge of the signal and noise characteristics [10]. In addition, adaptive filters have the potential of adaptively tracking the signal under non-stationary conditions. It has the unique characteristic of self-modifying [15] its frequency response to change the behaviour in time and allowing the filter to adapt the response to the input signal characteristics change.

II. ADAPTIVE NOISE CANCELLATION

Fig. 1 shows the basic problem and the adaptive noise cancelling solution to it. It contains 4 signals – Input signal $d(n)$, Reference signal $u(n)$, Filter output signal $y(n)$ and Error signal $e(n)$.

Adaptive noise canceller (ANC) receives two inputs namely primary input signal $d(n)$ and reference noise signal $u(n)$. Primary input is combination of source signal $s(n)$ and noise signal (n), uncorrelated with each other. While the reference noise signal $u(n)$, is correlated in some extent with noise n only. Reference input $u(n)$goes through adaptive filter producing $y(n)$ which is near estimate of primary input, which will be subtracted from the corrupted signal $d(n) = s(n) + n(n)$.

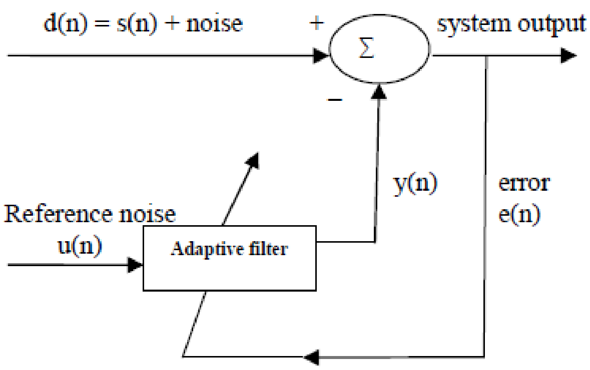

Fig 1: Adaptive Noise Cancellation

i.e., this signal and the primary signal are superimposed, so that the noise signal is cancelled. In Fig. 1 the reference input is processed by an adaptive filter, which is accomplished through an algorithm. When the noise estimate $y(n)$ equals or approximates the noise $n(n)$ in the corrupted signal, that is, y(n)= n(n), the error signal $d(n) = s(n) + n(n) - y(n) = s(n)$ will approximately give the clean speech signal $s(n)$. Hence, the noise is cancelled. The filter can operate under changing conditions and can readjust itself continuously to minimize the error signal. Main purpose of using adaptive noise canceller here is to have output which is best fit in the least squares sense to the signal $s(n)$. To achieve output, the error is fed back to adaptive filter that adjusts the filter using LMS adaptive algorithm to minimize total system output power [11]. Minimizing the total power at the output of the canceller, maximizes the output signal-to-noise ratio.

Assume that $s(n)$, $n(n)$, $u(n)$ and $y(n)$ are statistically stationary. Assume that $s(n)$ is uncorrelated with $n(n)$and $u(n)$ and suppose that $u(n)$ is correlated with (n). The output $d(n)$ is

$$d(n) = s(n) + n(n) - y(n) \qquad (1)$$

Squaring, we obtain

$$d(n)^2 = s(n)^2 + (n - y(n))^2 + 2s(n)(n - y(n)) \qquad (2)$$

Taking expectations both side of equation (2),

$$E[d^2] = E[s^2] + E[(n - y)^2] + 2E[s(n - y)] \qquad (3)$$

Realizing that $s(n)$ is uncorrelated with (n),

$$E[d^2] = E[s^2] + E[(n - y)^2] \qquad (4)$$

The signal power $E[d^2]$will be unaffected as the filter is adjusted to minimize $E[d^2]$. Accordingly, the minimum output power is-

$$Min\,E[d^2] = E[s^2] + Min\,E[(n - y)^2] \qquad (5)$$

When the filter is adjusted so that $E[d^2]$is minimized, therefore $E[(n - y)^2]$ is, also minimized. The filter output $y(n)$is then best least squares estimate of the primary noise (n). Moreover, when $E[(n - y)^2]$ is minimized, $E[(d - s)^2]$ is also minimized, since, from (1),

$$d(n) - s(n) = n(n) - y(n) \qquad (6)$$

Adapting the filter to minimize the total output power is thus causing the output $d(n)$ to be best least squares estimate of the signal $s(n)$. From (l), the output noise is given by $(n - y)$. Since minimizing $E[d^2]$ minimizes $E[(n - y)^2]$ Therefore, minimizing the total output power minimizes the output noise power. Since the signal in the output remains constant, minimizing the total output power maximizes the output signal to noise ratio. From (4) the smallest possible output power is

$$E[d(n)^2] = E[s(n)^2] \qquad (7)$$

When, $E[(n - y)^2] = 0$

At $y(n) = n(n)$and $d(n) = s(n)$

Minimizing the output power causes the output signal to be perfectly noise free [11].

III. LEAST MEAN SQUARE (LMS) ALGORITHM

There are several algorithms that can be utilized in noise cancellation and implemented using MATLAB. An adaptive algorithm is an algorithm that alters its features at the execution time depending on availability of information and based on previous based methods. The famous algorithm for the adaptive systems which works as self- adjusting algorithm is LMS algorithm i.e., Least Mean Square algorithm. The LMS (Least Mean Square) algorithm [12], is a very straight-forward approach in noise cancelling and was introduced by Widrow and Hoff in 1959. The LMS adaptive filter algorithm that has been developed is shown in Figure 2 below [13].

International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 11 Issue 01, January-2022

The parameters $d(n)$and $x(n)$ are the inputs of the algorithm in the form of column vector. In this $d(n)$ is the noise corrupted signal and $x(n)$ is the reference noise signal. This algorithm uses a gradient descent to estimate a time-varying signal. The gradient descent method finds a minimum, if exists, by taking steps in the direction negative of the gradient and it does so by adjusting the filter coefficients in order to minimize the error. The gradient is the Del-operator and is applied to find the divergence of a function, which is the error with respect to the nth coefficient in this case [15]. The LMS algorithm has been accepted by many researchers for hardware implementation because of its simplicity, low computational complexity and a fast convergence rate.

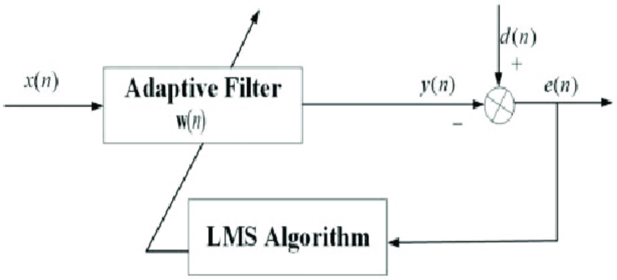

Fig 2: LMS adaptive filter algorithm [13]

The parameter $W(n)$ is the column weight vector of the filter at n[th] time, which is used in the algorithm to update the subsequent column weight vector and can be represented as,

$$W(n) = \begin{bmatrix} W_0(n) \\ W_1(n) \\ \vdots \\ W_{L-1}(n) \end{bmatrix} \tag{8}$$

The error signal $e(n)$ at n^{th} time is defined as the difference between the noise corrupted signal and the weighted-noise signal as shown the equation below,

$$e(n) = d(n) - y(n) \tag{9}$$

With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula:

$$w(n+1) = w(n) + 2\mu * e(n) * x(n) \tag{10}$$

where, $x(n)$ is the input vector of time delayed input values, $w(n)$ represents the coefficients of the adaptive FIR filter tap weight vector at time n, $w(n+1)$ is the filter coefficient for the next iteration, $e(n)$ is the error value and μ is known as the step size which is introduced here to control the step width of the iteration and thus the stability and convergence or divergence rate of the algorithm.

Step size, μ is one of the important parameters for LMS algorithm. So, selection of a suitable value for μ is important for the performance of the LMS algorithm, if the value μ is too small, the time adaptive filter takes time to converge on the optimal solution i.e., the rate of convergence is too slow and if μ is too large the adaptive filter becomes unstable and its output diverges [12].

Here, the LMS (Least Mean Square) adaptive filter algorithm will be terminated when column weight vector iterates up to the length of filter and the final output of the algorithm is $e(n)$.

The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise while leaving the signal unchanged.

IV. EXPERIMENTAL RESULTS

In this section we study the performance of the LMS algorithm as noise canceller. The algorithm is implemented according to the steps. Figure 3 shows, the Input sinusoidal signal and random noise signal. Figure 4 shows, the noise present in the sinusoidal signal and is eliminated using LMS algorithm [14].

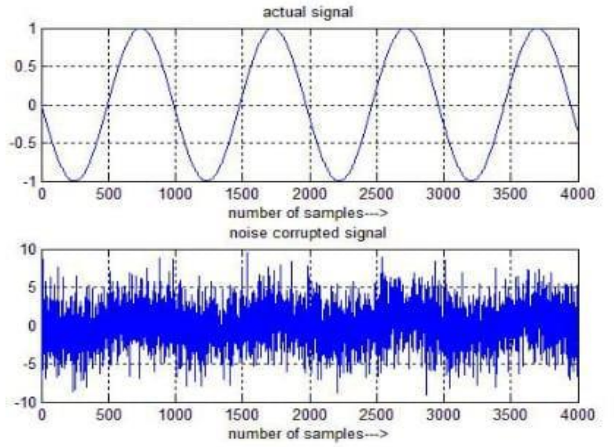

Fig 3: Input and noise signal [14]

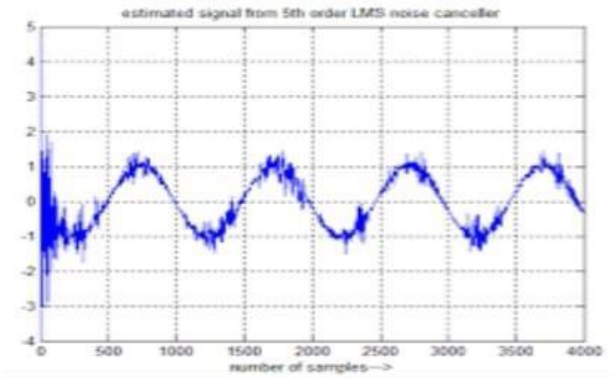

Fig 4: LMS filter output [14]

V. CONCLUSION

When the information signal travels in the free environment, it gets corrupted by the noise present in it. Removing this noise emerges out to be one of the most important concern for everyone. There are many conventional techniques to suppress the noise present in the information signal of which Adaptive Noise Cancellation (ANC) is one of the most important technique. ANC uses adaptive filters so as to analyze real time signals which continuously vary with respect to time. ANC involves many algorithms which are implemented so as to cancel the noise. This research paper is been focused on the Least Mean Square (LMS) algorithm. A comprehensive review has been carried out to identify the existing literature related to adaptive filtering in noise reduction using LMS adaptive algorithm. LMS is preferred by many researchers due to its robust and reliable nature. Also, LMS is simple to implement, has low computational complexity and shows a faster convergence rate. The LMS algorithm has been shown to produce good results in a noise cancellation problem.

ACKNOWLEDGMENT

I Take this opportunity to express my regards and sincere thanks to my advisor and guide Dr Poonam Beniwal. Her constant encouragement and moral support gave me the motivation to carry out the project successfully.

REFERENCES

[1] Rajshree Tiwari, Abhinav Singh et. al. "A Review Paper on Adaptive Noise Cancellation Implementation using TMS320C6713 DSP Board", International Research Journal of Engineering and Technology (IRJET) Volume: 05 Issue: 02 | Feb-2018, Pages 2246-2250
[2] Lakshmikanth.S, Natraj.K.R, Rekha.K.R, "Noise Cancellation in Speech Signal Processing-A Review" , International Journal of Advanced Research in Computer and Communication Engineering Vol. 3, Issue 1, January 2014.
[3] Navneet Upadhyay, Abhijit Karmakar, "An Improved Multi-Band Spectral Subtraction Algorithm for Enhancing Speech in Various Noise Environments", International Conference On Design and Manufacturing, IConDM 2013, Elsevier, Vol. 64, Pages 312-321, 13 November 2013.
[4] Kajikawa Y, Gan WS and Sen MK 2012 Recent advances on active noise control: open issues and innovative applications APSIPA Transactions on Signal and Information Processing 1(3)
[5] Chen X and Yijun M 2016 Passive control of centrifugal fan noise by employing open-cell metal foam Applied Acoustics 103 Part A pp 10-9
[6] Hu S, Rajamani R and Yu X 2013 Directional cancellation of acoustic noise for home window applications Applied Acoustics 74(3) 467-77
[7] Wu L, Qiu X and Guo Y 2014 A simplified adaptive feedback active noise control system Applied Acoustics 81 pp 40-6
[8] Wesson KD, Ochshorn RM and Land BR 2009 Low-cost, high-fidelity, adaptive cancellation of periodic 60 Hz noise J Neurosci Methods 185(1) pp 50-5
[9] LizheTan, JeanJiang, "Adaptive Filters and Applications", Digital Signal Processing (Third Edition), Fundamentals and Applications, 2019, Pages 421-474.
[10] Sekata Olika , Dr.A.Rajani, "Adaptive Noise Cancellation For Speech Signal", International Journal of Scientific and Research Publications, Volume 10, Issue 9, September 2020 ,ISSN 2250-3153
[11] Gyanendra Singh, Kiran Savita, Shivkumar Yadav and Vaibhav Purwar, "Design of Adaptive Noise Canceller using LMS algorithm", IJATER, vol 3, Issue 3, pp. 85-89, May 2013.
[12] Yang Liu, ''A Noise Reduction Method Based on LMS Adaptive Filter of Audio Signals'', 3rd International Conference on Multimedia Technology, ICMT, 2013.
[13] Jia-Haw Lee, Lu-Ean Ooi1, Ying-Hao Ko and Choe-Yung Teoh, "Simulation for noise cancellation using LMS adaptive filter", IOP Conf. Series: Materials Science and Engineering 211(2017) 012003.
[14] Archit Prakash Raut, Aayesha Raj et. Al. "Adaptive Noise Cancellation using Least Mean Sqaure Filter Algorithm (Matlab)", International Journal of Engineering Research & Technology (IJERT) Vol. 9 Issue 08, August-2020
[15] Shubhra dixit, Deepak nagaria "LMS Adaptive Filters for Noise Cancellation: A Review", October 2017 International Journal of Electrical and Computer Engineering 7(5):2520