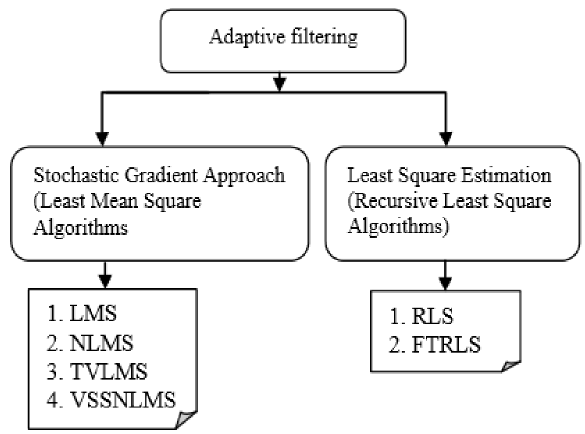
Comparative study of adaptive noise cancellation algorithms in the application of hearing aids.

Himanshu Jaiswal, Naman Arora, Krishnaa Gopakumar
Department of electronics and communication,
VIT University,Vellore

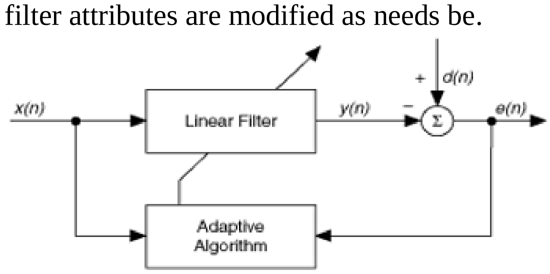
Abstract: Adaptive noise cancellation is an approach used for noise reduction in signal. As received signal is continuously corrupted by noise where both the received signal and noise signal changes continuously, thus there arises a need of adaptive filtering. Adaptive filters adjust their coefficients to minimize an error signal. As most of the portable electronic devices such as cellular phones, personal digital assistants, and hearing aids require digital signal processing for high performance. In the era of communication systems, the processing of signal is of main concern as the transmission signals suffered from various interferences and noise because of their sinusoidal nature. Thus, to improve the quality of communication, signal processing is required for the noise cancellation from the signal by using filters In this Paper, we will use a few Algorithms to design the adaptive filter and we aim to eliminate the noise in the input signal by designing an adaptive filter using MATLAB and simulate the working of the adaptive filter in MATLAB using noise corrupted signals.

Introduction: There are many digital signal processing applications in which second order statistics cannot be specified. Such application includes channel equalization echo cancellation and noise cancellation. In these applications, filters with adjustable coefficients called Adaptive filters are employed. An adaptive filter is a filter that self adjusts its transfer function according to an optimizing algorithm.

It adapts the performance based on the input signal. Such filters incorporate algorithms that allow the filter coefficients to adapt to the signal statics. There are different approaches used in adaptive filtering, which are as follows:



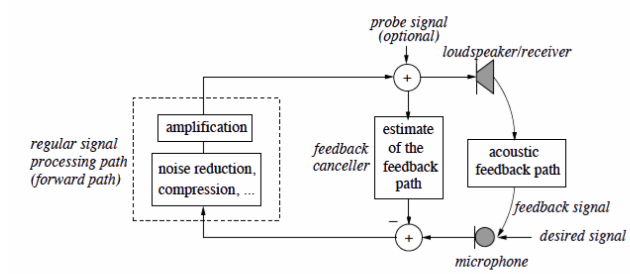
Adaptive techniques use algorithms, which enable the adaptive filter to adjust its parameters to produce an output that matches the output of an unknown system. An adaptive filter algorithmically modifies its parameters keeping in mind the end goal to minimize a component of the contrast between the desired yield d (n) and its real yield y (n). This capacity is known as the expense capacity of the adaptive algorithm. Figure 1 demonstrates a block diagram of the adaptive noise cancellation. The adaptive filter plans to compare its yield y(n) to the fancied yield d(n). At every cycle the error signal, e (n) =d(n)-y(n), is sustained over into the filter, where the filter attributes are modified as needs be.



The point of an adaptive filter is to ascertain the contrast between the desired signal and the adaptive filter output i.e. e(n). This error signal is feedback once more into the adaptive filter and its coefficients are changed algorithmically with a specific end goal to minimize an element of this distinction, known as the expense capacity.
On account of acoustic noise cancellation, the ideal output of the adaptive filter is measure up to in worth to the undesirable noise. At the point when the adaptive filter output is equivalent to desired signal the error signal goes to zero. In this circumstance the noise signal would be totally wiped out and the far client would not hear any noise.

Adaptive filter is a nonlinear filter since its attributes are subject to the input signal and thus the homogeneity and additive conditions are not fulfilled. The way to effective adaptive signal processing comprehends the major properties of adaptive calculations, for example, Least Mean Square LMS, RLS and so forth. Utilization of adaptive filter is the dropping of the noise segment, an undesired signal in the same frequency range.

Most dissonance diminution systems are pitch contour based systems, modulation-based Randomness Management analyses the overall floor and the presence of sound that change in storey over fourth dimension (such as lecture). Speech is a very different signal than noise, this deviation is what allows the system to analyse the sounds and make decisions. If the noise is loud enough, and language International Relations and Security Network 't detected, then the noise decrease will kick in and reduce the overall level of output to prevent the situation becoming uncomfortable. If talking to is detected, then the level will be turned back up to ensure audibility of these important and desirable sounds. These adjustments are done in different frequency circle to turn down frequency where noise is dominant allele , while going away the frequence that are useful and important for speech understanding alone. In an nonpareil noise reduction system, the auditory modality assist will only reduce the undesired noise while leaving desired speech sign completely intact. In this ideal scenario, the listening aid will understand what sounds it is expected to pass on to the listener and what sounds it should reject. In monastic order to reach this goal, the nidus has been on the acoustic difference of opinion between speech and noise. Noise can vary in an almost infinite number of ways, however the nature of speech is constant in its overall social organization.



1. LMS ALGORITHM: Least mean squares (LMS) algorithms are class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time.

The basic idea behind LMS filter is to approach the optimum filter weights (R-1 P), by updating the filter weights in a manner to converge to the optimum filter weight. The algorithm starts by assuming a small weights (zero in most cases), and at each step, by finding the gradient of the mean square error, the weights are updated. That is, if the MSE-gradient is positive, it implies, the error would keep increasing positively, if the same weight is used for further iterations, which means we need to reduce the weights. In the same way, if the gradient is negative, we need to increase the weights. So, the basic weight update equation is:

$$wn+1 = wn - \mu\Delta\varepsilon[n]$$

Where, ε represents the mean-square error. The negative sign indicates that, we need to change the weights in a direction opposite to that of the gradient slope.

LMS algorithm summary:

The LMS algorithm for a pth order algorithm can be summarized as

Parameters:

$$P = \text{filter order}$$
$$\mu = \text{step size}$$

Initialization:

$$\hat{h}(0) = 0$$

Computation: For n = 0, 1, 2...

$$X(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$$
$$e(n) = d(n) - \hat{h}H(n)\, X(n)$$
$$\hat{h}(n+1) = \hat{h}(n) + \mu\, e^*(n)\, X(n)$$

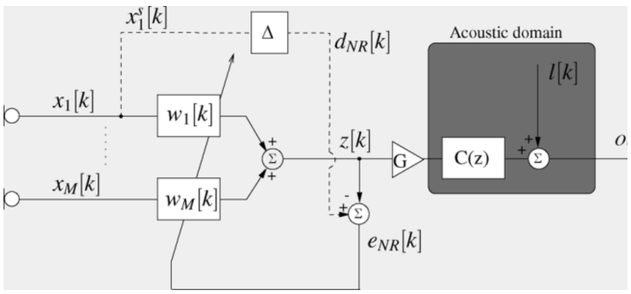


Fig. 1. Multichannel noise reduction system in the hearing aids context

2. NORMALISED LEAST MEAN SQUARE (NLMS) ALGORITHM : The main drawback of the "pure" LMS algorithm is that it is sensitive to the scaling of its input. This makes it very hard to choose a learning rate μ that guarantees stability of the algorithm. The Normalised least mean squares (NLMS) filter is a variant of the LMS algorithm that solves this problem by normalising with the power of the input.

NLMS algorithm summary:
Parameters:

$$P = \text{filter order}$$
$$\mu = \text{step size}$$

Initialization:

$$\hat{h}(0) = 0$$

Computation: For n = 0, 1, 2...

$$X(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$$
$$e(n) = d(n) - \hat{h}H(n)\, X(n)$$
$$\hat{h}(n+1) = \hat{h}(n) + (\mu\, e^*(n)\, X(n) / X^H(n)\, X(n))$$

SIGN-ERROR LMS ALGORITHM & SIGN-DATA LMS ALGORITHM :

When the sign function of error signal is taken for the filtering process, it's called the sign-error LMS algorithm, and if the sign function of the data signal is taken, it's called the sign-data LMS algorithm.

sgn(a)= 1; a > 0
0; a = 0
−1; a < 0

Formula used - w(n + 1) = w(n) + μu(n) sgn(e(n))

The Sign algorithm (other names: pilot LMS, or Sign Error) w(n + 1) = w(n) + μu(n) sgn(e(n))

• The Clipped LMS (or Signed Regressor) w(n + 1) = w(n) + μ sgn(u(n))e(n)

• The Zero forcing LMS (or Sign Sign) w(n + 1) = w(n) + μ sgn(u(n)) sgn(e(n))

The Sign algorithm can be derived as a LMS algorithm for minimizing the Mean absolute error (MAE) criterion

J(w) = E[|e(n)|] = E[|d(n) − wT u(n)|]

OBSERVED CALCULATIONS

We used a couple of parameters to find out which algorithm can be used for the most efficient noise cancellation mechanism in hearing aids.

Computational complexity	lms	nlms	sdlms	senlms
Additions	2N	3N	2N+1	2N+1
Multiplications	2N+1	3N+1	2N+3	2N+2
Divisions	-	N	N	N
Comparisons	-	-	-	3

We also computed the SNR values to find out which algorithm would have the best amplification ability when it comes to hearing aids.

	SNR
LMS	16.333
NLMS	18.65
SDLMS	17.873
SELMS	17.8926

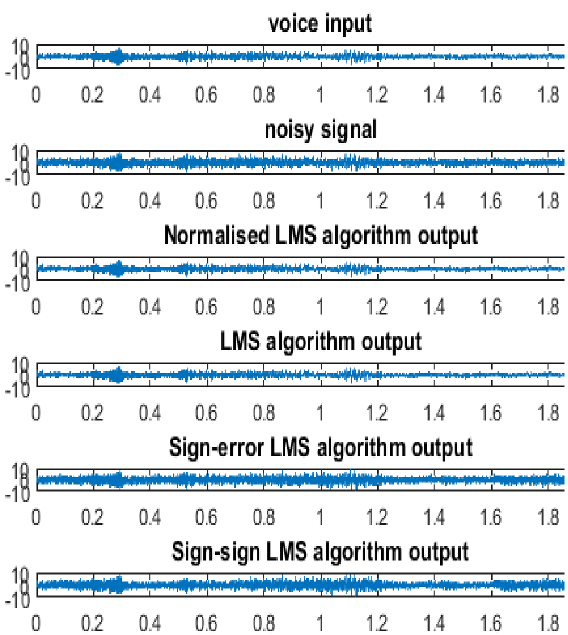
Performance Comparison

Parameter	*LMS algorithm*			
	Conventional LMS	*Sign-Data*	*Sign-Error*	*Sign-Sign*
MSE	2.29 $*10^{-16}$	9.155 $*10^{-16}$	3.6226	0.6369
Complexity	High	Low	Low	Very low
Stability	Less stable	Less stable	Less stable	Less stable

OUTPUT OBTAINED USING MATLAB:

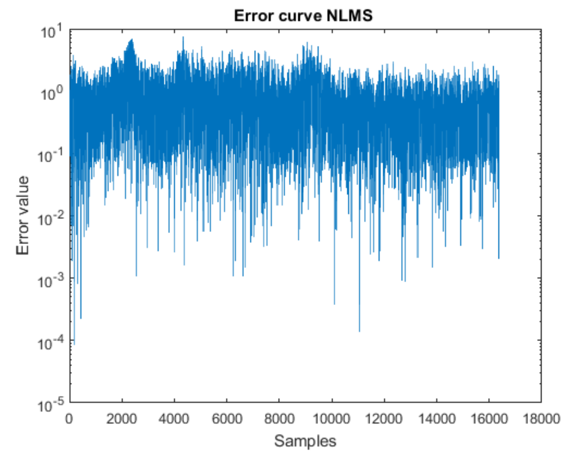
The proposed algorithms were implemented and simulated on MATLAB and the following figure shows the output of the code executed when a random audio signal was given as the input.

The first graph shows the voice input. To this, noise is added which is then given to the filter is shown in the graph. Then the outputs of NLMS, LMS, sign-error LMS and sign-sign LMS algorithm are given respectively.

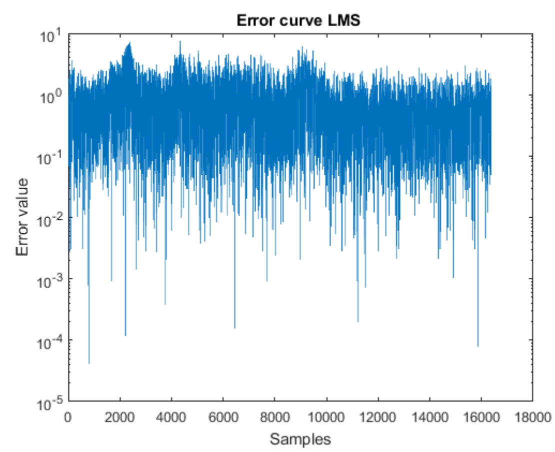


CONCLUSION:

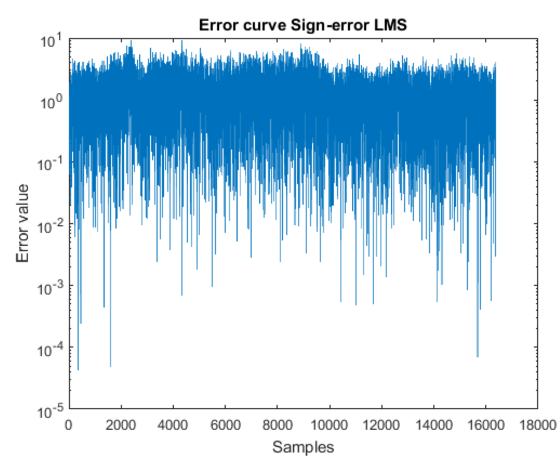
The error graphs of the four algorithms are as given below.



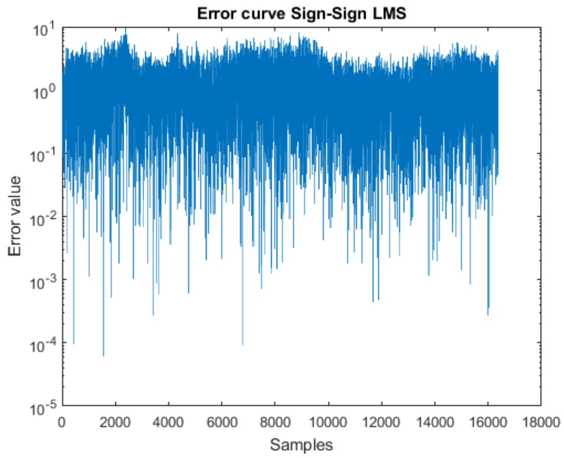
(a) Error curve for NLMS algorithm



(b) Error curve for LMS algorithm



(c) Error curve for sign-error LMS algorithm.



(d) Error curve for sign-sign LMS algorithm.

As seen from different graphs it is clear that the choice of step size between the specified range is very important .If it is too low or near to upper range convergence is poor and desired signal is not obtained as shown in different figure. Thus different variants of LMS indicate different performance properties according to the choice of step size.

Here through rigorous simulation we observed that with increase in step size the MSE and Misadjustment increases and hence SNR decreases for adaptive algorithms. As the filter length increases the selection for optimum step size decreases in LMS, ENLMS, SENLMS whereas it increases for NLMS algorithm. MSE and Misadjustment increases with filter length whereas SNR decreases. Hence for better results we choose a smaller filter order. MATLAB results show that the LMS algorithm has low SNR but it is simple to implement. ENLMS reduces the number of computation due to use of norm of e(n) but does not show good results as NLMS. NLMS is simple to implement and also provides good results but proposed SENLMS algorithm shows same with less number of computations. Proposed algorithm proves to overcome all these problems by providing better results with low computational cost hence can be implemented on TMS320C6713 DSP board for real time signals.

REFERENCES

[1] Raj Kumar Thenua and S.K. AGARWAL" Simulation and Performance Analyasis of Adaptive Filter in Noise Cancellation" International Journal of Engineering Science and Technology Vol. 2(9), 2010, 4373-4378.

[2] Sayed.A.Hadei and M.lotfizad,"A Family of Adaptive filter Algorithms in Noise Cancellation For Speech Enhancement", International Journal of Computer and Engineering, vol.2, No.2, April2010, 1793-8163.

[3] Raj Kumar Thenua, "Hardware Implementation of Adaptive Algorithms for Noise Cancellation", 2011 International Conference on Network Communication and Computer (ICNCC 2011)

[4] John G. Proakis, "Digital Signal Processing Principles, Algorithms and Applications", Pearson Prentice Hall, fourth Edition, page No. 909-911.

[5] Monson H. Hayes: Statistical Digital Signal Processing and Modeling, Wiley, 1996, ISBN 0471-59431-8.

[6] Simon Haykin: Adaptive Filter Theory, Prentice Hall, 2002, ISBN 0-13-048434-2.

[7] Simon S. Haykin, Bernard Widrow (Editor): Least-Mean-Square Adaptive Filters, Wiley, 2003, ISBN 0-471-21570-8.

[8] Analysis and comparison of RLS adaptive filter in signal De-noising, GuoQuan Xing; YuXia Zhang, Dept. of Biomed. Eng., Xianning Coll.

[9] Paulo S.R. Diniz: Adaptive Filtering: Algorithms and Practical Implementation, Kluwer Academic Publishers, 1997, ISBN 0-7923-9912-9.