NOVATEUR PUBLICATIONS
International Journal Of Research Publications In Engineering And Technology [IJRPET]
ISSN: 2454-7875
VOLUME 2, ISSUE 6, June-2016

NOISE CANCELLATION USING LEAST MEAN SQUARE AND WAVELET TRANSFORM FOR SPEECH ENHANCEMENT

Tanmay A. Lonare
M.E.(VLSI and Embedded System)
G.H.R.I.E.T. Wagholi, Pune

Prof. Mrinal R. Bachute
M.E.(VLSI and Embedded System)
G.H.R.I.E.T. Wagholi, Pune

ABSTRACT:

This paper presents about Adaptive Filter Algorithms used in Embedded Signal Processing for Speech Enhancement. Filters are generally used to select or to remove or to separate out particular fixed frequency, but in Adaptive Filters the frequency selection is important as well as the coefficients of Adaptive filter are being updated by the Adaptive Algorithms. Adaptive Filters are the filters whose filter coefficients are updated automatically by the process of steepest descent algorithm. An Adaptive Filter is defined as a self-adjusting system that relies for its operation on a recursive algorithm, which makes it possible for the Filter to perform satisfactorily in an environment where knowledge of the relevant statistics is not available. Least Mean Square (LMS) is the algorithm used to update filter coefficients by subtracting the desired signal from input signal producing error signal which updates the algorithm variables at each iteration repeated iterating process trains itself to the input signal and cancels noise. Wavelet transform is taking the overlapped windowed frames of input signal transforming it from time domain to frequency to understand the spectrogram of signal apply thresholding depending upon the parameters to consider and denoise the signal. Databases of clean speech and Noise speech can be downloaded freely from TIMIT, NOIZEUS, and SpEAR database. Implement the both the filters LMS and Wavelet and compare them to conclude which algorithm works well.

KEYWORDS: Adaptive Filter, Least Mean Square, Wavelet Transform.

INTRODUCTION:

Signal need to be processed for proper scaling of information received by the Sensors. Signals in the form of Continuous in case of Analog and Discrete in case of Digital. Signal received contains noise along with information. That's why there is a need of Signal Processing.

Signal Processing is widely used in RADAR communication, Digital Still Cameras, Signal Processing in Telephone Network, Wireless Communication, Digital Satellite Television, etc. Signal processing is useful in removing unwanted information from useful information. It is used to increase the strength of signal. As the signals in the form of 0 and 1, signal processing used to remove unwanted sequence of bits.

Signal processing is important because the needed signal parameters such as the low level parameters and high level parameters are extracted from the available signal using different equations. This signal may be in Time or Frequency domain. Signal in the form of Light, Sound, Heat, Pressure, etc.

Challenge of Signal Processing is to made signal easily understood to the listener or receiver. If the generated signal in form of light then receiver will be able to decode or read that sent signal is the challenge. Features present in the signal are properly extracted by Feature Extraction Methods so that signals in the form of Feature or template are properly identified or stored for further identification.

In Signal Processing, Filter is a process or electronic device used to remove unwanted features or components from signal. Filtering is a part of Signal Processing where it is used to remove or suppress unwanted parameters partially or completely. Means removing some frequency components such as background Noise and suppress interfering signals in order to filter out information signal. [1]

Filters are classified as: Linear or Non-Linear , Time Invariant or Time Variant, Analog or Digital, Discrete Time (Sampled) or Continuous Time, Passive(Using R, L,C) or Active(Using Op-Amp) Type of Continuous Time

IJRPET

Filter, Infinite Impulse Response or Finite Impulse Response, Causal or Not Causal. But there are two types of digital Filter on the basis of Impulse Response of the Filter: Infinite Impulse Response, Finite impulse Response. [1]

FIR Filter Design carried out in three methods: Window Method, Frequency Sampling Method, and Optimal Filter Design Method. The Window method basically begins with a desired unit sample response which is then truncated by a Finite Duration Window. In the Frequency Sampling Method the frequency response of the FIR filter is specified in terms of the samples of desired Frequency Response. Optimal Filter Design Method is available with an Algorithmic Design Procedure which generates Optimum Equiripple FIR Filter design. IIR filter Design methods classified as: The Impulse Invariant Method, The Bilinear Transformation Method.[1]

In case of Analog Circuits the Filter consists of Resistors, Capacitors, Inductors, which are bulky in comparison to the Integrated Circuits that is why Analog is transformed to Digital. Analog work as it is transformed to Digital work is a kind of Challenge. Analog Radio is transformed to Software Defined Radio. Different signals have different parameters to suppress interfering signals or Noise so the challenge is removing this unwanted parameters make the Filter more Precious in Filtering. [1]

An Adaptive Filter is defined as a self- adjusting system that relies for its operation on a recursive algorithm, which makes it possible for the Filter to perform satisfactorily in an environment where knowledge of the relevant statistics is not available. Adaptive filters are classified into two main groups: linear and non linear. Linear Adaptive Filters compute an estimate of a desired response by using a linear combination of the available set of observables applied to the input of the Filter. Otherwise, the adaptive filter is said to be nonlinear. Adaptive Filters may also be classified into: Supervised Adaptive Filters, Unsupervised Adaptive Filters. [2]

Adaptive Filtering involves the changing of filter parameters (coefficients) over time, to adapt to changing signal characteristics. Over the past three decades, digital signal processors have made great advances in increasing speed and complexity, and reducing power consumption. As a result, real-time adaptive filtering algorithms are quickly becoming practical and essential for the future of communications, both wired and wireless.[3]

LITERATURE REVIEW:

Widrow et al. [6] described the concept of adaptive noise canceling, an alternative method of estimating signals corrupted by additive noise or interference. The method uses a primary input containing the corrupted signal and the reference input containing noise correlated in some way with the primary noise. The reference input is adaptively filtered and subtracted from the primary input to obtain the signal estimate.

Sambur [7], presented a Least Mean Square (LMS) adaptive filtering approach has been formulated for removing the deleterious effects of additive noise on the speech signal.

Widrow et al. [8] presented a new family of algorithms to adjust the weights of an adaptive filter so that the expected value of the error to the degree 2K would be minimized. This algorithm uses the concept of steepest descent so can be viewed as extension of the Widrow-Hoff LMS algorithm.

Williamson et al. [9] presented a paper explaining the performance characteristics of the median LMS Adaptive Filter. The performance of gradient search adaptive filters, such as the least mean square (LMS) algorithm, may degrade badly when the filter is subjected to input signals which are corrupted by impulsive interference. The median LMS (MLMS) adaptive filter is designed to alleviate this problem by protecting the filter coefficients from the impact of the impulses. They have addressed two important factors of MLMS algorithm behavior: 1) convergence in the mean, 2) a cost comparison with LMS.

Gazor et al. [10] presented a paper explaining a general analysis of the LMS based algorithms. There analysis covers all the fixed step-size versions of the LMS algorithms. These include the conventional LMS, transform-domain normalized LMS, and LMS/Newton algorithm but exclude those algorithms whose step-sizes are changing with time.

Xi et al. [11] presented a paper on computing running discrete cosine/sine transforms based on the adaptive LMS algorithm. The possibility of implementing orthogonal analyzers using adaptive filtering is an area of interest. In 1987, Widrow et al. investigated the relationship between the least mean square adaptive algorithm and the discrete Fourier transform which resulted in a new method for calculating the DFT using the adaptive LMS algorithm.

Gelfand et al. [12] presented a variable step-size LMS (VSLMS) algorithms are a popular approach to adaptive filtering which can provide improved performance while

IJRPET

maintaining the simplicity and robustness of conventional fixed step-size LMS.

Chi- Chou Kao [13] presented the new methods to cancel echo for short path (e.g. car) and eliminate noise for speech enhancement. An adaptive filter based on the delayed error least mean square algorithm is used to cancel echo.

Xiao et al. [14] presented a paper explaining the statistical performance of the conventional adaptive Fourier analyzers, such as the least mean square(LMS), the recursive least square (RLS) algorithm, and so on, may degenerate significantly, if the signal frequencies given to the analyzers are different from the true signal frequencies. This difference is referred to as frequency mismatch (FM).

PROPOSED WORK AND PROBLEM DEFINITION :

Study of the existing algorithms and suggesting new algorithm for Noise Cancellation and Speech Enhancement.

PROBLEM DEFINITION:

As the Algorithms are developing day by day and their parameters such as the Convergence Factor, Stability Factor, Step-size, Speed, performance, are certainly being improving at each Algorithm for achieving Accuracy and Precision.

This paper consists of Speech enhancement that is to remove Train Noise, Car Noise, different types of Noise from the Input Signal.

BLOCK DIAGRAM:

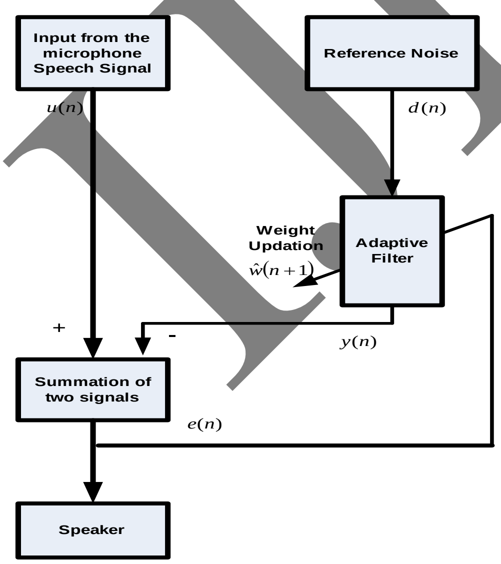

Figure 1: Flow diagram of Noise Canceller using Adaptive Filter Algorithm

Approaches to the development of Linear Adaptive Filters:

1) Stochastic Gradient Approach
2) Least Square Estimation:

a) Least Mean Square (LMS): Block by Block (Block of equal length),
b) Recursive Least Square (RLS): Sample by Sample (requires less storage).

Three basic kinds of Estimation:

a) Filtering ,
b) Smoothing and
c) Prediction.

LMS (LEAST MEAN SQUARE) ALGORITHM:

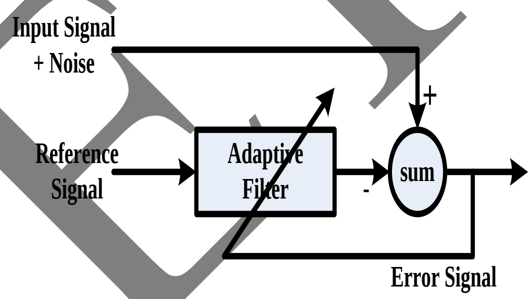

Figure 2 : Block Diagram of LMS Algorithm

This Algorithm was invented by Widrow and Hoff in 1960. The algorithm uses equations as, for n = 0, 1, 2… n,

EQUATIONS:

- Eq. for updating the tap-weight vector :

$$\hat{w}(n+1) = \hat{w}(n) + \mu u(n)\left[d^*(n) - u^H(n)\hat{w}(n)\right] \quad \text{---- (1)}$$

- Filter Output :

$$y(n) = \hat{w}^H(n)u(n) \quad \text{--- (2)}$$

- Estimation Error or Error Signal :

$$e(n) = d(n) - y(n) \quad \text{--- (3)}$$

- Tap-weight Adaptation :

$$\hat{w}(n+1) = \hat{w}(n) + \mu u(n)e^*(n) \quad \text{--- (4)}$$

Where,
μ = step-size parameter
$w(0)$ = initial condition
n = total number of iterations

Algorithm is defined as, M = number of taps (i.e., filter length), μ = step-size parameter $0 < \mu < \frac{2}{MS_{max}}$,

where S_{max} is the maximum value of the power spectral density of the tap inputs u(n) and the filter length M is moderate to large. Initialization of LMS Algorithm is

IJRPET

NOVATEUR PUBLICATIONS
International Journal Of Research Publications In Engineering And Technology [IJRPET]
ISSN: 2454-7875
VOLUME 2, ISSUE 6, June-2016

done, if the prior knowledge of the tap-weight vector $\hat{w}(n)$ is available; use it to select an appropriate value for $\hat{w}(0)$. Otherwise, $\hat{w}(0)$ =0. Input provided to LMS Algorithm is given as,

u(n) = M-by-1 tap-input vector at time n.

$$u(n) = [u((n), u(n-1)), \ldots, u(n-M+1)]^T \text{ , - (5)}$$

d(n) = desired response at time n.

Output is calculated as, $\hat{w}(n+1)$ = estimate of tap-weight vector at time n+1.

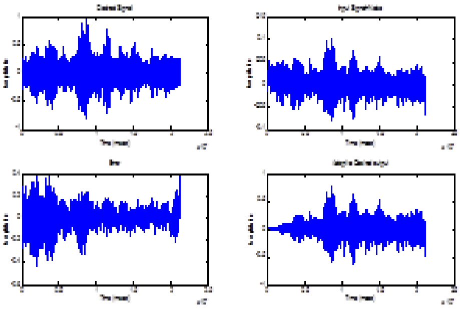

Figure 3: Showing Desired Signal, Input Signal + Noise, Error Signal and Adaptive Desired Output using LMS Algorithm.

WAVELET TRANSFORM:

Wavelet is a section of speech wave signal. Speech signal in time domain represent time dependent information while frequency domain consists of frequency dependent information. Frequency is inversely proportional with Time. If time is minimum then maximum information is obtained in frequency. Fourier Transform is used to convert time domain signal to frequency domain signal. Wavelet Transform is used to represent to represent speech signal in time domain as well as in frequency domain.

Wavelet Transform performs scaling and shifting in order to adjust the shape of the wavelet for filtering. Wavelets are derived from basic wavelet called mother wavelet by scaling and shifting.

$$\psi_{a,b}(t) = \frac{1}{\sqrt{a}}\psi\left(\frac{t-b}{a}\right) \quad \text{----(6)}$$

Where, a is scaling parameter and b is shifting parameter.

Wavelet family consists of following types of wavelets having different shapes as follows:

- Haar function
- Daubechies function
- Coeflit coefficient
- Symlet function

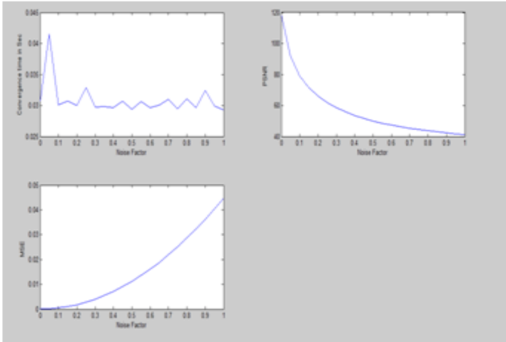

Figure 4: Graph showing the effect of variable noise percents on Speech signal in terms of MSE(Mean Square Error) and SNR(Signal to Noise Ratio).

Table 1: Showing the Convergence Time, PSNR and MSE for different Noise percent

Sr.No.	Time	SNR	MSE
1	0.033707	118.1768	8.57E-06
2	0.046166	91.98467	0.00012
3	0.035008	79.04207	0.000456
4	0.034142	71.40266	0.001015
5	0.032415	66.04438	0.001797
6	0.032089	61.95073	0.002803
7	0.034443	58.6596	0.004033
8	0.033688	55.9223	0.005487
9	0.032551	53.58972	0.007164
10	0.031076	51.56546	0.009064
11	0.030706	49.82209	0.011188
12	0.030559	48.49881	0.013536
13	0.031169	47.32474	0.016107
14	0.030657	46.2743	0.018902
15	0.031483	45.32768	0.021921
16	0.030778	44.46931	0.025163
17	0.029548	43.68671	0.028629
18	0.033004	42.96975	0.032318
19	0.034065	42.31009	0.036231
20	0.030224	41.70081	0.040367
21	0.030313	41.13609	0.044727

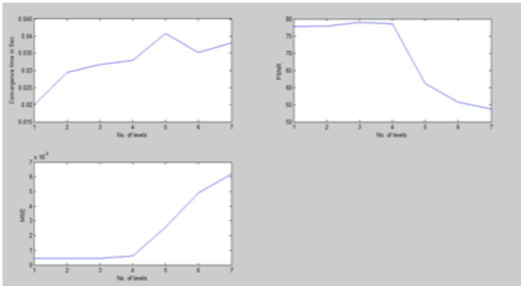

Figure 5: Graph showing the effect of different levels of Wavelet on Speech signal in terms of MSE and SNR.

Table 2: Showing the Convergence Time, SNR and MSE for different wavelet levels.

Sr.No.	Time	SNR	MSE
1	0.019343	77.7385	0.000458
2	0.031901	77.87669	0.000456
3	0.032142	79.04207	0.000456
4	0.032845	78.63024	0.000608
5	0.034492	61.20259	0.002556
6	0.035214	55.78066	0.004911
7	0.043603	53.7045	0.006182

CONCLUSION:

A detailed review on transform domain adaptive filters has been studied and presented. It provides better computational speed, fast convolution, enhances convergence performance as compared to time domain algorithms.

REFERENCES:

[1] Md. Saiful Islam, et al., "*Design of FIR Filter Using Hamming Window,"* International Journal of Emerging Research in Management &Technology, ISSN: 2278-9359, (Volume-3, Issue-2), Research Article February 2014.

[2] Adaptive Filters Simon Haykin McMaster University Hamilton, Ontario, Canada L8S 4K1.

[3] [Online].Available:http://in.mathworks.com/help/dsp/ug/overview-of-adaptive-filters-and-applications. html[Accessed: Aug. 02, 2015].

[4] Woon-Seng Gan and Sen M. Kuo, Embedded signal processing with the Micro Signal Architecture. Hoboken, New Jersey: John Wiley & Sons.

[5] Simon Haykin and Thomas Kailath, Adaptive Filter Theory. South Asia: Pearson Education, 2009.

[6] Bernard Widrow, et al., *"Adaptive Noise Cancelling: Principles and Applications,"* IEEE, vol 63, no. 12, pp. 1692-1716, Dec 1975.

[7] Marvin R. Sambur, "*Adaptive Noise Canceling for Speech Signals,"* IEEE Transactions on Acoustics, Speech and Signal Processing, VOL. ASSP-26, No.5, October 1978.

[8] Eugene Walach and Bernard Widrow, "*The Least Mean Fourth (LMF) Adaptive Algorithm and its Family,"* IEEE Transactions on Information Theory, vol IT- 30, no.2, March 1984.

[9] Geoffrey A. Williamson, et al., *"Performance Characteristics of the Median LMS Adaptive Filter,"* IEEE Transactions on Signal Processing, vol. 41, no.2, February 1993.

[10] B. Farhang-Boroujeny and S. Gazor, "*Performance of LMS based Adaptive Filters in Tracking a Time Varying plant,"* IEEE Transactions on Signal Processing, vol. 44, no. 11, November 1996.

[11] Jiangtao Xi and Joe F. Chicharo, "*Computing Running Discrete Cosine/Sine Transforms based on the Adaptive LMS Algorithm,"* IEEE Transactions on Circuits and Systems for Video Technology, Vol. 8, No. 1, February 1998.

[12] Saul B. Gelfand, et al., "*The Stability of variable step-size LMS algorithm,"* IEEE Transactions on Signal Processing, Vol. 47, No. 12, December 1999.

[13] Chi- Chou Kao, "*Design of Echo Cancellation and Noise Elimination for Speech Enhancement*," IEEE Transactions on Consumer Electronics, Vol. 49, No. 4, November 2003.

[14] Yegui Xiao, et al., *"Statistical properties of the LMS Fourier Analyzer in the presence of Frequency Mismatch,"* IEEE Transactions on Circuits and Systems—I: Regular Papers, Vol. 51, No. 12, December 2004.

[15] Yoichi Hinamoto and Hideaki Sakai, "*Analysis of the Filtered-X LMS Algorithm and a related new algorithm for Active Control of Multitonal Noise,"* IEEE Transactions On Audio, Speech, And Language Processing, Vol. 14, No. 1, January 2006.

[16] Esfandiar Zavarehei, et al., "*Noisy Speech Enhancement using Harmonic–Noise model and Codebook-based Post-Processing,"* IEEE Transactions on Audio, Speech, and Language Processing, vol.15, no.4, May 2007.

[17] Andrea L. Kraay and Arthur B. Baggeroer, "*A Physically Constrained Maximum-Likelihood Method for Snapshot-Deficient Adaptive Array Processing,"* IEEE Transactions on Signal Processing, VOL. 55, NO. 8, August 2007.

[18] Cassio G. Lopes and Ali H. Sayed, "*Incremental Adaptive Strategies Over Distributed Networks,"* IEEE Transactions on Signal Processing, vol.55, no.8, August 2007.

[19] Weifeng Liu, et al., "*The Kernel Least-Mean-Square Algorithm,"* IEEE Transaction on Signal Processing, vol. 56, no.2, February 2008.

[20] Leonardo Rey Vega, et al., "*A New Robust Variable Step-Size NLMS Algorithm,"* IEEE Transactions On Signal Processing, Vol. 56, No. 5, May 2008.

[21] Philipos C. Loizou and Gibak Kim, *"Reasons why Current Speech-Enhancement Algorithms do not Improve Speech Intelligibility and Suggested solutions,"* IEEE Transactions on Audio, Speech and Language Processing, Vol. 19, No.1, January 2011.

[22] Jacob Benesty, et al., "*On Regularization in Adaptive Filtering,"* IEEE Transactions On Audio, Speech, And Language Processing, Vol. 19, No. 6, August 2011.

IJRPET