ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

An Adaptive Noise Cancellation Using Wavelet Based Grazing Estimation of Signal Method: Special Reference to Speech Signal



Nahid Jabeen
Research Scholar,
Dept of ECE,
Sun Rise University,
Alwar, Rajastan, India.

Dr.Sachin Saxena
Professor,
Dept of ECE,
Sun Rise University,
Alwar, Rajastan, India.

Dr.R.Murali Prasad
Professor,
Dept of ECE,
Vardhaman College of
Engineering, Hyderabad.

Abstract:
This paper introduce the reducing the content of noise present in the received Speech signals for wireless communication medium by using different methods such as adaptive filtering and wavelet thresholding for speech noise cancellation are compared and evaluated on their performance for noise cancellation in speech. Evaluation of various methods is done in terms of PSNR(Peak Signal to Noise Ratio), MSE(Mean Square Error), Correlation.This output is cascaded with wavelet transforms techniques with compare the available control algorithms output error signals. Compared to other available control algorithms the proposed method is Simple to implement, yields good performance and converges quickly. This proposed technique is implemented using Matlab software and DSP processor. This computer output simulation results confirm the effectiveness of our proposed algorithm.

Keywords:
Speech noise cancellation, Adaptive and FIR and filtering, Wavelet Transformation, MSE, PSNR, Correlation.

I .INTRODUCTION:
Active noises are nothing but real time noise whereas they cannot be predictable Noise is non-informative and plays the role of sucking the originality of signal. Any kind of processing of the signal contributes to the noise addition. A signal traveling through the channel Speech noise cancellation is the most important field of speech enhancement. It is used for various applications such as mobile phones, year phone etc. Reduction of noise from speech signals plays a important role in modern communication systems.

Adaptive filtering is a powerful technique for signal detection because of the random pattern of the noise and the non-deterministic sources of the interference.and time domain. Various algorithms such XLMS, thresholding are derived for noise cancellation and comparison is made.

II.METHODOLOGY:
A.Adaptive Filtering:
Adaptive filtering means that filter parameters such as bandwidth and resonant frequency change with time. Noise cancellation technology is a growing field that capitalizes on the combination of disparate technological advancements. This aims to cancel or at least minimize unwanted signal and so to remedy the excess noise that one may experience. Adaptive noise cancellation is widely used to improve the Signal to Noise Ratio (SNR) of a signal by removing noise from the received signal. In this configuration the input x(n), a noise source, N1(n), is compared with a desired signal, d(n), which consists of a signal, s(n) corrupted by another noise, N0(n). The adaptive filter coefficients adapt to cause the error signal to be a noiseless version of the signal s(n) as shown in Fig. 1.

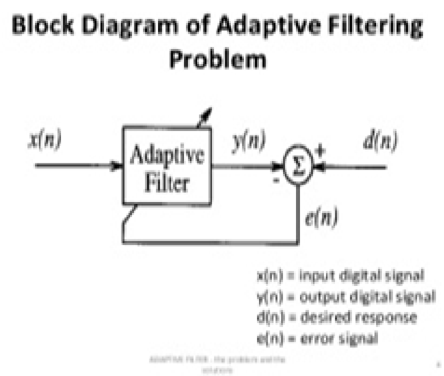

Fig1:Block diagram of an Adaptive Filte

ISSN No: 2348-4845
International Journal & Magazine of Engineering,
Technology, Management and Research
A Peer Reviewed Open Access International Journal

The blunder sign ought to merge to the sign s(n), however it won't unite to the accurate sign. At the end of the day, the contrast between the sign s(n) and the mistake signal e(n) will dependably be more prominent than zero.

B.Noise cancellation in single channel headset:

The fundamental components of the framework are rehashed for the left and right channels like the DSP,simple to computerized (A/D) and advanced to simple (D/A) converter. A speaker is mounted inside of the ear cup for accepting and acoustically transducing a sign, mix of clamor cancelation sign and sound sign. An amplifier is likewise mounted inside of the ear cup for transducing acoustic weight inside of the ear cup to a relating simple mistake signal. A simple channel gets the simple mistake flag and rearranges it to create a simple broadband commotion cancelation signal. The simple blunder sign is likewise given to an A/D converter, gets the simple receiver mistake flag and changes over it to an advanced mistake signal. A high pass (HP) and low pass (LP) channels are given in the sound sign to every channel. The low recurrence part of the sound sign is given to DSP through A/D where it is subtracted from the amplifier yield generally acoustic sign scratchs off the sought sound sign. The high recurrence sign is straightforwardly given to summing enhancer.

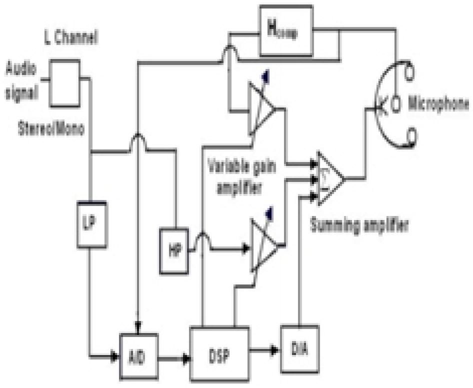

Fig. 2 Schematic diagram of noise cancellation circuit for a single channel headset

The DSP takes the digital error signal and generates a digital tonal noise cancellation signal using an adaptive digital feedback filter. A digital to analog converter then converts the digital tonal noise cancellation signal toan analog tonal noise cancellation signal so that it can be combined with the analog broadband noise cancellation Signal.

The active noise cancellation headset system, when an audio signal is given to headphone (for single channel). This method can also be used for noise cancellation without audio signal. The resultant composite cancellation signal is provided to the speakers in the ear cups to cancel noise within the ear cups. The broadband analog cancellation is effective to reduce overall noise within the ear cup. The DSP not only provides active control of the analog cancellation loop gain to maximize the effectiveness of the broadband analog cancellation but also uses the adaptive feedback filter/algorithm to substantially reduce at least the loudest tonal noises penetrating the ear cup.

C. FIR Filter Function:

this important to make up for the optional way exchange capacity S(z) from y(n) to e(n), which incorporates the D/A converter, recreation channel, power enhancer, amplifier, acoustic way from amplifier to blunder receiver, preamplifier, hostile to associating channel and A/D converter.

E9X)=[P(Z)- S(Z) W(Z)]X(Z) 1.0

The lingering mistake is in a perfect world zero i.e. E(Z)=0, The Optimal exchange capacity is

W0(Z) =0 1.2

It W(Z) needs to all the while model P(Z) contrarily display S(Z). It is a legitimate model of the plant is a key point of preference change is i/p signal brought about by changing in the clamor sources. It is FIR channel capacity 1/s(z) show in fig(3) P(z) does not contain a deferral of at any rate parallel length.z.

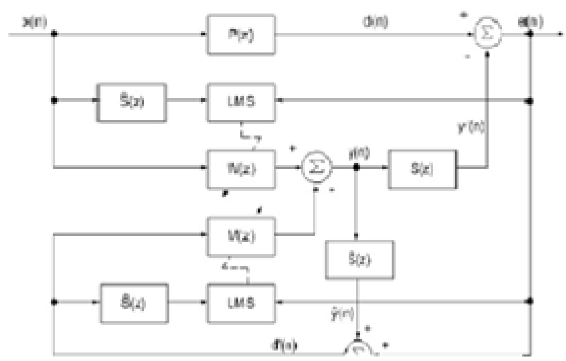

Fig. 3 FIR Filter Function

ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

D. FILTERED – XLMS ALGORITHM:

This algoriyhm explain in two ways to deal with tackling this issue, the primary arrangement is to put a reverse channel 1/s(z), is arrangement with s(z) to uproot its impact the second arrangement is to put an indistinguishable channel in the reference signal.

E. DERIVATION OF THE FXLMS ALGORITHM

$E(n)=d(n)-s(n)*[Wt(n)x(n)]$ 1.3

Where n is time index s(n) is impulse response of secondary path S(Z), * denoted convolution.

$W(n)=[wo(n),w1(n)\ldots\ldots WL-1(n)]^{T}$ 1.4

$X(n)=[x(n)\ x(n-1)\ldots\ldots x(n-L+1)]^{T}$ L is filter order mean sequence cost function $\sum(n=E(e^2(n)]$, $\sum\hat{}(n)=e^2(n)$
1.5
$W(n+1)=w(n)-\mu/2\ \ \sum\hat{}(n)$
1.6

$\sum\hat{}(n)=\nabla e^2(n)=2[\nabla e(n)]e(n)$ have $\nabla e(n)=-s(n)*x(n)=-x^1(n)$, where $x^1(n)=[x^1(n)\ x^1(n-1)$

$x^1(n-L+1)]^T$ and $x^1(n)=s(n)*x(n)$

$\sum\hat{}_{\nabla}(n)=-2x^1(n)\ e(n)$ 1.7.

F.(WT)Wavelet Thresholding:

Methods discussed above analyse the speech signal in frequency domain only, whereas Wavelet transformation analyse the signal both in frequency and time domain. It also helps in mult resolution analysis.De-noising using linear filters is not efficient for functions with discontinuities. This is due to the linear nature of this process, which prevents to efficiently estimate discontinuities. Wavelet approximation using thresholding allows an adaptive representation of signal discontinuities. We will thus use wavelet thresholding to perform a non-linear denoising.

Thresholding is done by the following definition:

$$d_{jk}^{*}=\begin{cases} d_{jk}, & |d_{jk}|\geq\lambda \\ 0\ , & |d_{jk}|<\lambda \end{cases}$$

where $\lambda \geq 0$ is threshold value/parameter It is of two types:

Soft Thresholding and Hard Thresholding:

The hard thresholding method keeps some coefficients fixed and sets others to 0; in contrast the soft thresholding method either ‘shrinks’ or sets them to 0.The noisy signal is basically of the following form:

$s(n) = f(n) + e(n)$ 1.8

where time n is equally spaced. In the simplest model, suppose that e(n) is a Gaussian white noise N(0,1) and the noise level is supposed to be equal to1.

The de-noising objective is to suppress the noise part of the signal s and to recover f. The de-noising procedure proceeds in three steps:

Decomposition. Choose a wavelet, and choose a level N. Compute the wavelet decomposition of the signal s at level N. Detail coefficients thresholding. For each level from 1 to N, select a threshold and apply soft thresholding to the detail coefficients. Reconstruction. Compute wavelet reconstruction based on the original approximation coefficients of level N and the modified detail coefficients of levels from 1 to N.

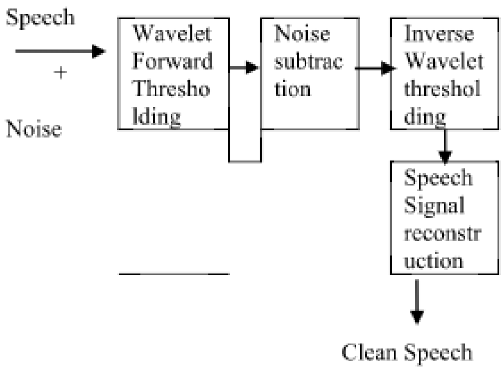

Fig 4: Block diagram of wavelet thresholding

III. RESULTS:

ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

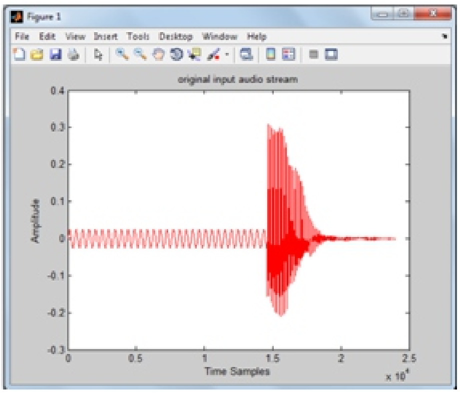

Fig.5:Original input audio signal

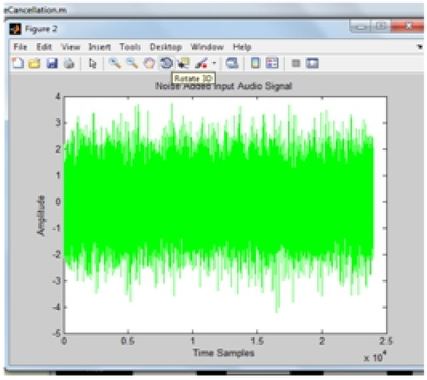

Fig. 6: Noise added speech

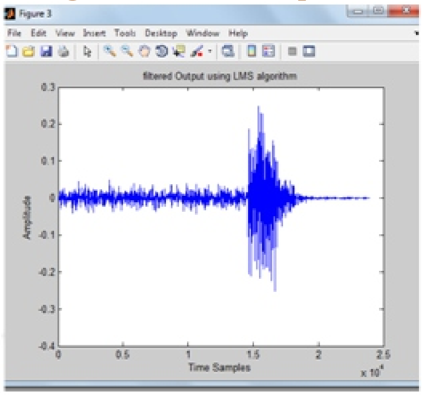

Fig. 7:Filtered output using LMS algorithm

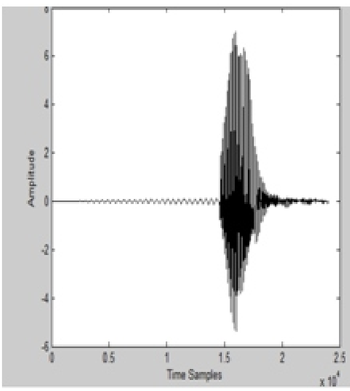

Fig.8: Filtered output using XLMS algorithm

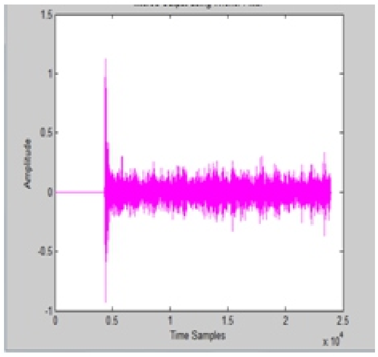

Fig.9: FIR Filter output

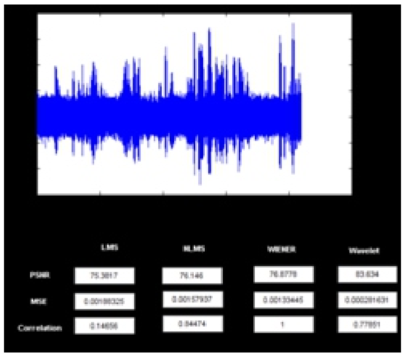

Fig. 10: Wavelet output and Adaptive Noise Cancellation Methods

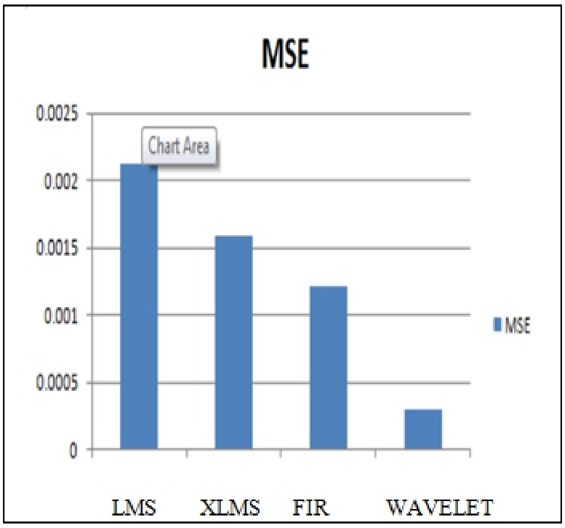

Fig 11: Graphical Representation of MSE

ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

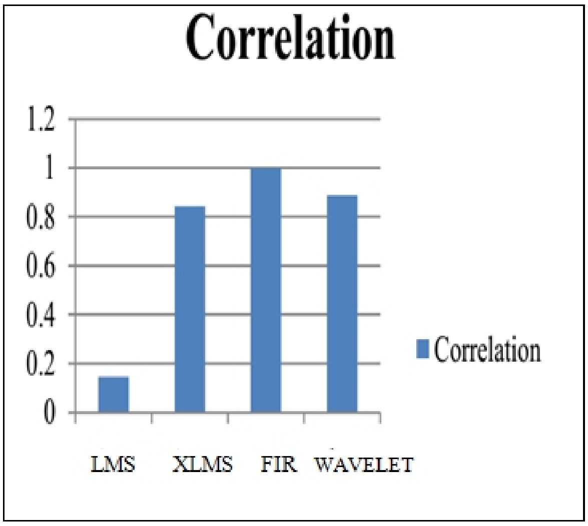

Fig. 12: Graphical Representation of correlation for For different method Different method

Method	LMS	XLMS	FIR filter	WAVELET
PSNR	75.381	76.146	76.8778	83.634
MSE	0.0018	0.0015	0.0013	0.0002
CORRELATION	0.146	0.146	1	0.7789

TABLE. 1: Performance Evaluation of different

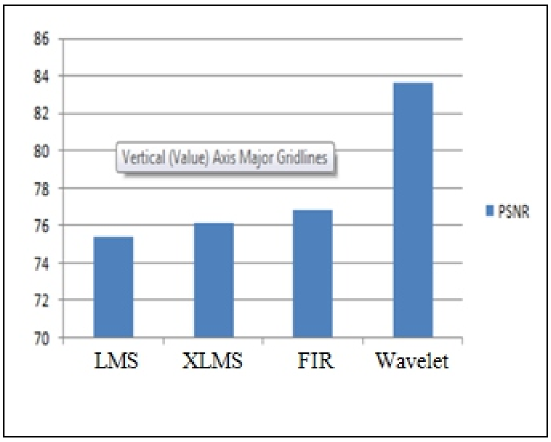

Fig. 8: Graphical Representation of PSNR for different method

V. CONCLUSION:

As shown in above screen shots and graphs among the outputs of three algorithm: LMS, XLMS and Threshold, most stable algorithm is Wavelet thresholding. Among the adaptive filters: LMS and XLMS, as shown in output that performance of XLMS algorithm is more reliable than LMS algorithm.

VI.FUTURE WORK:

In future we have considered doing change in our calculations for testing them with video flags and pictures moreover. We have additionally considered actualizing some de-noising or clamor cancelation systems in equipment hardware, contrast them and our embraced procedures, and make a relative study between them in future. The work could be amplified towards different ecological clamor conditions with longer discourse sections. The SNC method could be suitable for ongoing applications, for example, portable hearing assistants, versatile information transfers, sans hand sound gadgets, biomedical, and numerous others.

REFERENCES:

1.Ahmad N.A. and Okwonu F.Z. (2011), Least Squares Problem for Adaptive Filtering, Australian Journal of Basic and Applied Sciences, 5 No. 3, 69-74.

2.Akhtar M.T. (2009), Improving Performance of Active Noise Control Systems in the Presence of Uncorrelated Periodic Disturbance at Error Microphone, IEEE Transaction on Audio, Speech and Language Processing, No. 3, 2041-2044.

3.Jacobson C A, Johnson Jr. C R, McCormick D C and W A Sethares (2001), Stability of Active Noise Control Algorithms, IEEE Signal Processing Letters, Volume 8, Number 3.

4.Kuo S.M. and Wu H.T (2005), Nonlinear adaptive bilinear filters for active noise control systems, IEEE Trans. Circuits Syst. I, Reg. Papers, vol. 52, no. 3, pp. 1–8.

5.Kuo S.M., Wu H.T., F.–K. Chen, and M.R. Gunnala, (2004), Saturation effects in active noise control systems, IEEE Trans. Circuits Systems I, Reg. Papers, vol. 51, no. 6, pp. 1163–1171.

6.Kuo SM (1995), Adaptive Active Noise Control Systems- Algorithm and DSP Implementation Proc. Digital

ISSN No: 2348-4845
International Journal & Magazine of Engineering, Technology, Management and Research
A Peer Reviewed Open Access International Journal

Signal Processing Technology Conference 17th – 18th, Florida.

7.Kuo. S.M, Morgan D.R. (1996), Active Noise Control Systems-Algorithms and DSP Implementations. pp: 37.

8.Manikandan S, Dr. S. Subharani, (2005), Design of Active Noise Control for Speech Signal Using Adaptive Techniques, Academic Open Internet Journal, Volume 16.

9.Manikandan S (2007), Effective Noise Reduction for Speech Signal Using Wavelet, International Journal of Systems Simulation, Vol.1 No: 1, by the Serials Publications.

10.Manikandan.S (2006), Literature Survey Of Active Noise Control Systems, Academic Open Internet Journal, ISSN 1311-4360, Issue no: 17.

11.Manikandan S, Madheswaran M (2008), Design of Adaptive noise cancellation for speech signals using GES method, International journal of soft computing 3 (1), p 63-68.

12.Manikandan S, Mythili S, (2006), A new design of active noise fed forward control systems using Delta rule algorithms, Academic Open Internet Journal, Volume 19.

13.Napoli R., Piroddi L.(2010), Nonlinear active noise control with NARX models, IEEE Transactions on Audio, Speech and Language Processing, Vol. 18, n. 2, pp. 286–295.

14.Oppenheim A V, Weinstein E, Zangi K C, Feder M, Gauger D (1994), Single-Sensor Active Noise Cancellation, IEEE Transactions on Speech and Audio Processing, Vol. 2 No. 2.

15.Park S J, Yun J H, Park Y C Youn D H (2011), A Delayless Subband Active Noise Control System for Wideband Noise Control, IEEE Transactions on Speech and Audio Processing, Vol. 9, No. 8.

16.Petraglia M R and Alves R G (1997), New Results on Adaptive Filtering Using Filter Banks, IEEE International Symposium on Circuits and Systems.

17.Kirei B.S., Topa M.D., Muresan I., Homana I. and Toma N., (2014), Blind Source Separation for Convolutive Mixtures with Neural Networks, Advances in Electrical and Computer Engineering (AECE), No. 1, 63-68.

Author's Details:

Nahid Jabeen received M.Tech. in DSCE from JNTUH, and having more than 10 years of experience in both teaching and industry currently pursuing Ph.D and working as an Asst. Professor at NSAKCET, Hyderabad.

Dr. Sachin Saxena Received his PhD and M.Tech from I.I.T Roorki, and having more than 18 years of teaching experience. Currently he is working as Professor at Sun Rise University, Alwar, Rajastan, India.

Dr. R. Murali Prasad Received his PhD from JNTUA, and having more than 22 years of teaching experience. Currently he is working as Professor at Vardhaman College of Engineering, Hyderabad, India.