Design and Analysis of Adaptive FIR Filter for Different Step Size

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Abstract - An adaptive filter has capability to operate under an environment which is not known. This feature of adaptive filter makes a useful filter for various applications of signal processing and control. The adjustable filter coefficients of adaptive filter can be controlled by the estimated error which is further computed with the help of input signal and desired response. In this paper effect of different sizes has been analyzed a design of adaptive filter. The error reduction has been compared only by variable step size. The developed adaptive filter has been design and simulated in MATLAB. It can observe from the simulated results, the chosen step size has minimum error as compare to others.

Keywords- ADAPTIVE FILTER, DIGITAL FILTER, FIR, LMS, STEP SIZE

I. INTRODUCTION

An adaptive filter is a filter which self-adjusts is transfer function according to an algorithm driven by an error. Known adaptive has a static transfer which is known as wiener filter. Adaptive filter are used for some applications where some parameter are not known in advance. It uses feedback in form of error signal generated by the filter output and noise corrupted signal to adjust its transfer function to match the changing parameter. This error is minimized by updating the coefficients of a digital filter according to an adaptive LMS algorithm. LMS algorithm is used due to its robustness and simplicity leading to its implementation in many applications.

Error Signal is obtained by subtracting the output signal from primary signal. It is given by

$$e_n = x_n - y_n \qquad (1)$$

Where n = Iteration number
e_n = Error signal

x_n = Input to adaptive filter

y_n = Output of adaptive filter

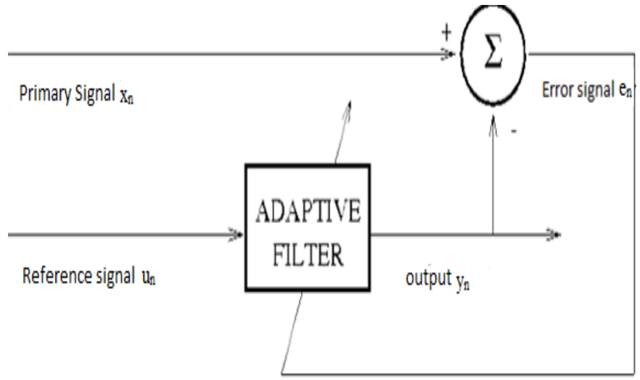

Figure 1. Adaptive Filter Structure

In LMS filter, the algorithm starts by assuming a weight and each step by finding the gradient of the mean square error, the weights are updated. According to the Widrow- Hopf LMS algorithm by updating the weights from sample to sample is given by

$$W_{n+i}=W_n+2\mu e_n u_{n-i} \qquad (2)$$

Where i=0, 1…….. , M-1

M=filter length

μ=step size

The weight updates by the LMS algorithm are only estimates which improve gradually with time [1]

In practice for fast convergence speed large step size is chosen which causes large steady state misadjustment error due to this adaptive algorithm sensitive to interference present in the input. Small step size increases number of iterations results slow convergence speed and less steady state misadjustment error [2].

The condition for convergence is

$$0 < \mu > 1/\lambda_{max}$$

Where λ_{max} is the maximum Eigen value of the input signal auto correlation matrix.

In this paper we propose the value of step size for which misadjustment error is less that desired signal approximately same as the wiener filter signal. If step size deviates either side from the proposed step size value the desired signal deviates wiener filter signal.

II. Implementation of basic LMS algorithm

Least Mean Square (LMS) algorithms are used to find the filter coefficients to produce the least mean squares of the error signal which is the difference the desired signal and the actual signal. The method having filter adaption is based on the error at the current time is known as stochastic gradient descent method.

In adaptive system digital filter is realized using a finite impulse response FIR structure.

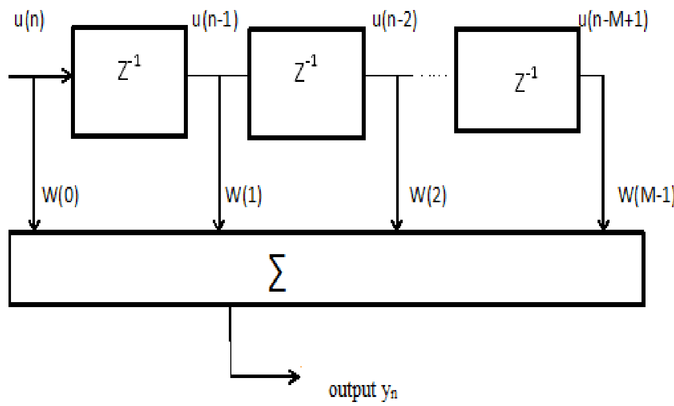

Figure 2.Finite Impulse Response Filter Structure

FIR structure is widely used due to its simplicity. The computational procedure for the LMS algorithm is given sequentially as:

1. Initially set each weight $W_n(i)$, i= 0, 1……, M-1.

For each subsequent sampling n=1, 2 …………….

2. Computation of filter output

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3. Computation of error estimate is given by eq. (1)
4. Filter weights are updated as eq. (2)

Above computation process can be shown by as the flow chart for LMS adaptive filter [1].

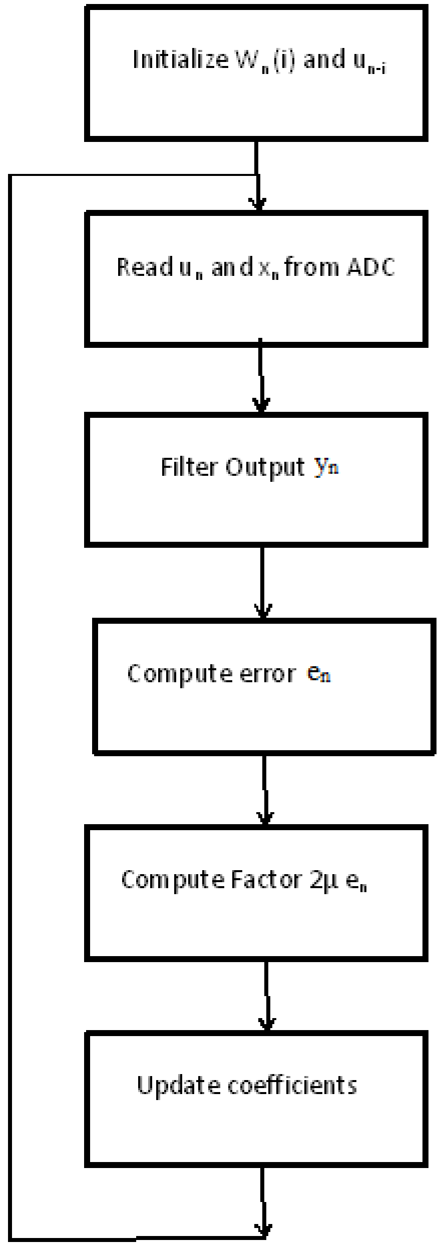


Figure 3. Flowchart of LMS Adaptive filter

III. Result And Discussion

In the simulation we provide an audio signal which is represented by the equation

$$S_n = \sin(2*pi*(f_a/f_s)*n)$$

Where f_a = frequency of an audio signal
f_s = sampling frequency

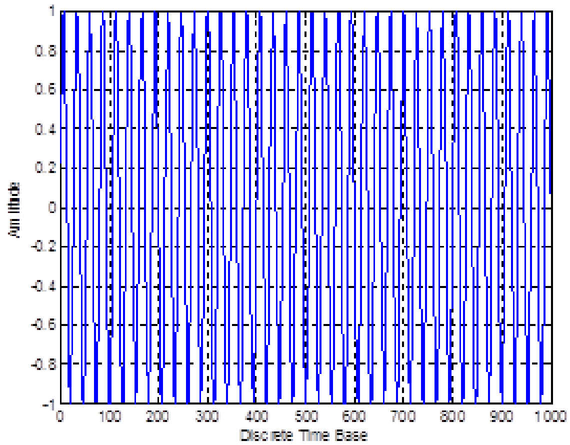


Figure 4. Audio Signal

To make an input signal to adaptive filter added a noise signal then it becomes $x_n = \sin(2*pi*f_a/f_s*n)$ + noise v_n [4]

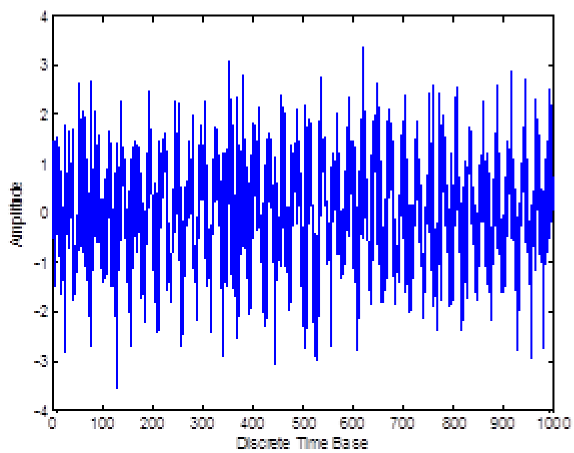


Figure 5. Audio Signal with Noise

A reference signal u_n which is a noise signal uncorelated With input signal s_n and have corelation with noise v_n

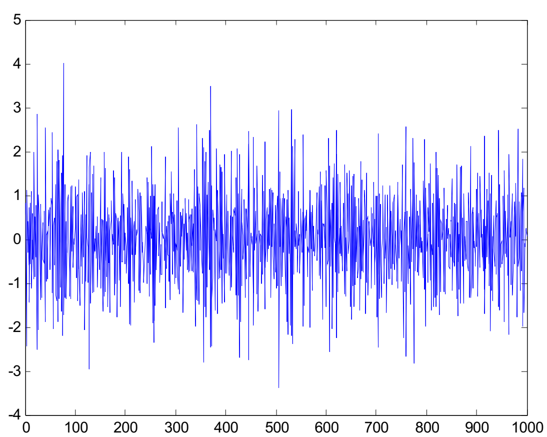
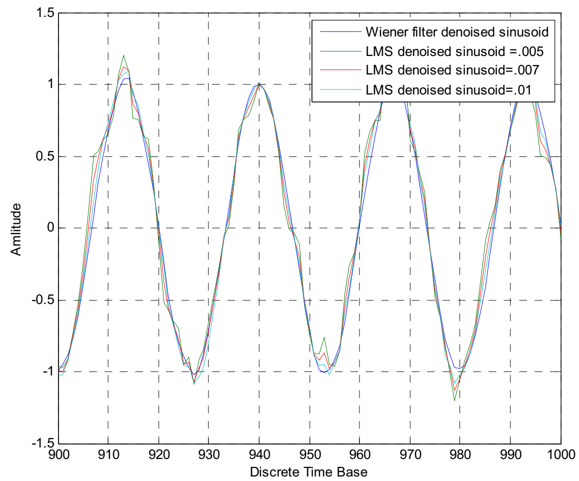


Figure 6. Reference Signal

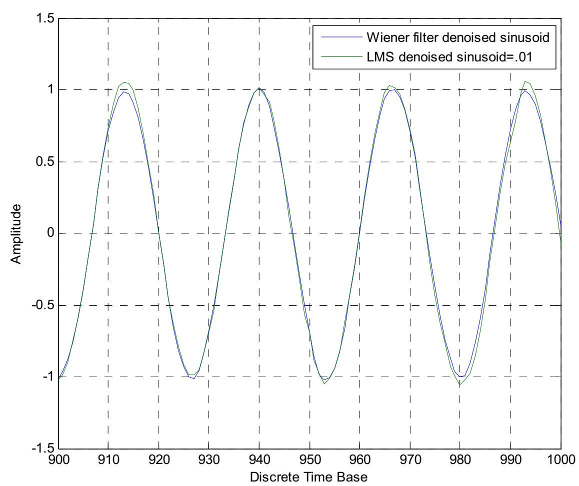
For analysis purpose, set the different values of step size to get the output waveform

These output waveforms of the system are plotted for step size less than 0.01



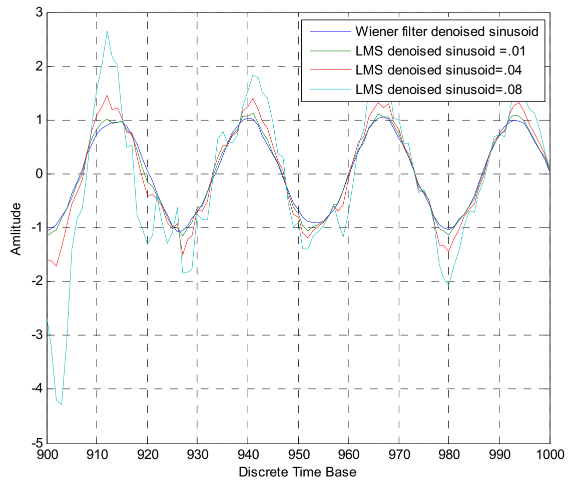
(a)

Output waveform of the system is plotted for step size 0.01.



(b)

Output waveform of the system are plotted for step size greater than 0.01.



(c)
Figure 7. Output Signal at different step size

IV. CONCLUSION

This paper analysis the output waveform of the system for different step size. For the selected value of step size 0.01, the result is very close to the wiener filter response. Output waveforms of the system for step sizes less and greater than
the selected value deviates from the wiener filter response. More deviation in step size results more deviated output waveform, so the optimum value of step size is 0.01. For this size adaptive noise canceller is very efficient and useful system in many applications with sound.

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