Mathematical Model for De-Noising of Audio Signals using Adaptive Filtering

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Abstract

Nets and filters are important conclusions of functional spaces which have got variety of applications in data transmission. Mathematical modeling of the input and output in wiener filter can result in control of noise during data transmission. We have unscrupulously applied least mean square theorem for optimal control in noise.

Keywords: Wiener filter, Additive White Gaussian Noise, Least Mean Square theorem, Minimum Mean Square Error

I. INTRODUCTION

Noise and distortion are main factors which limit the ability of data transmission in broadcastings and these factors also affect the precision of outcomes in signal measurement methods, while, modeling and reducing noise and distortions are the primary considerations in theory and practical in communications and signal processing. One additional concern is that, reducing noise and eliminating distortion are major complications in applications such as; cellular mobile communication, speech recognition, image processing, medical signal processing, radar, sonar, and any other application where the it is not possible to isolate the desired signals from noise and distortion [1]. The coefficients of Wiener filter are calculated to minimize the average squared distance between the filter output and desired signal. In its basic form, it is assumed in the Wiener theory that the signals are stationary processes. Nevertheless, if the coefficients of filter are calculated periodically for every block of N signal samples then the filter adapts itself to the average characteristics of the signals within the blocks and becomes block-adaptive. A block-adaptive also called as segment adaptive filter can be used for signals for instance speech and image which may be considered practically stationary over a relatively small block of samples [2].

The foundation of data-dependent linear least square error filters is formed using Wiener theory which was expressed by Norbert Wiener in 1940 [3].The problem is to propose a linear filter with one input as noisy data and the constraint of minimizing the consequence of noise at the filter output according to some statistical criterion. A useful approach to this filter optimization problem is to reduce mean square value of error signal that is defined as the difference between some desired response and the actual filter output. For stationary inputs, the resulting solution is commonly known as the Weiner filter. Its main purpose is to reduce the amount of noise existing in a signal by comparison with an estimation of the desired noiseless signal [4].

II. MATHEMATICAL MODEL

- Read an audio file and add white Gaussian noise to it.
- Find out noise by subtracting signal from the noisy signal.
- Frequency modulates the noisy signal using sampling rate of 8000Hz and carrier frequency of 3000 Hz.

Let the carrier be $x_c(t) = X_c \cdot \cos(f_c t)$, and
the modulating signal be $x_m(t) = B \cdot \sin(f_m t)$
Then $x(t) = X_c \cdot \cos[f_c t + B \cdot \sin(f_m t)]$
Where, $B = \dfrac{\text{maximum carrier frequency deviation}}{\text{Modulation frequency}}$

- And then demodulate the signal using frequency demodulation.
- After demodulation pass the signal through the filter.
- Assuming that all the signals involved are real-valued signals the elements for LMS algorithm are :

Weight vector $h = [h_0, h_1, h_2, h_3 \ldots\ldots h_{n-1}]^T$ (1)
Signal input $X = [x(n),x(n-1),x(n-2),\ldots\ldots,x(n-N+1)]^T$ (2)
Filter output $y(n)= h^T X(n)$ (3)
Error signal $e(n) = d(n)-y(n)$ (4)
Where d(n) is desired output and y(n) is actual filter output .
Equation (4) is the error or cost function and LMS is used in order to minimize it.

$$\text{Mean square error} = \frac{e_1^2+e_2^2+e_3^2+\cdots+e_n^2}{n}$$

Where n is number of bits of reference signal.
A simple algorithm can be written as
for c=1:nk,
e(c)=d(c)-y(c);
h(c+1)=h(c) + (u * e(c) * X^T)
end
Where y is filter output and u is the step size.
In figure 1 a simple process of LMS algorithm is shown:

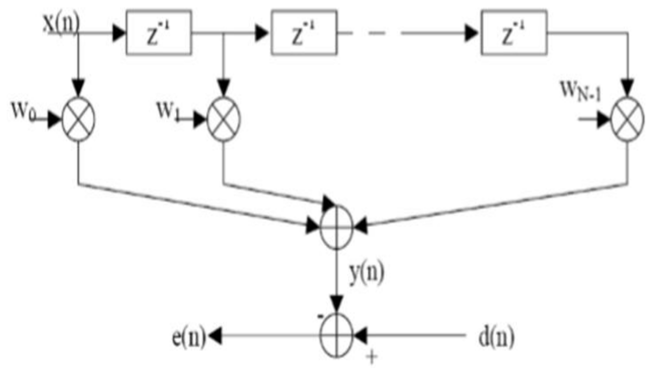
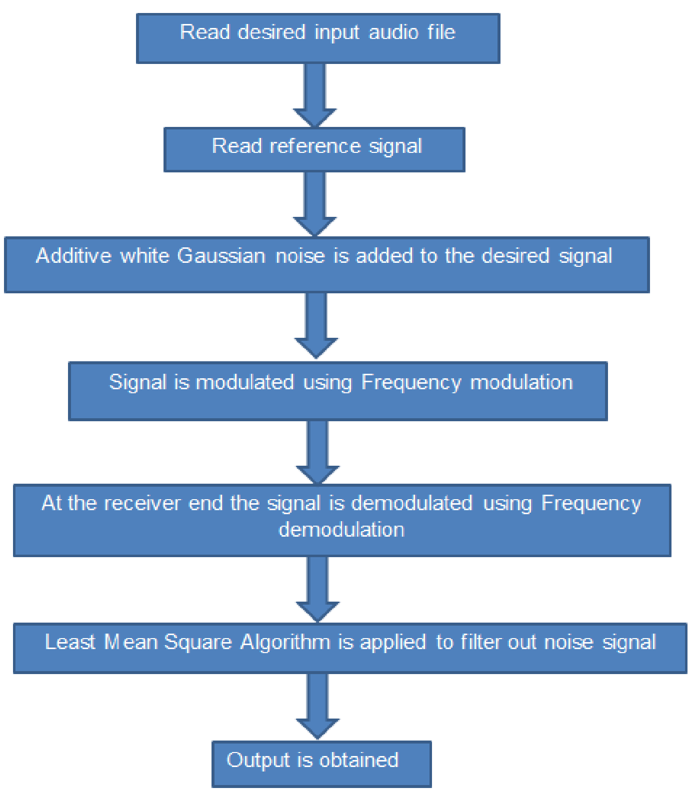
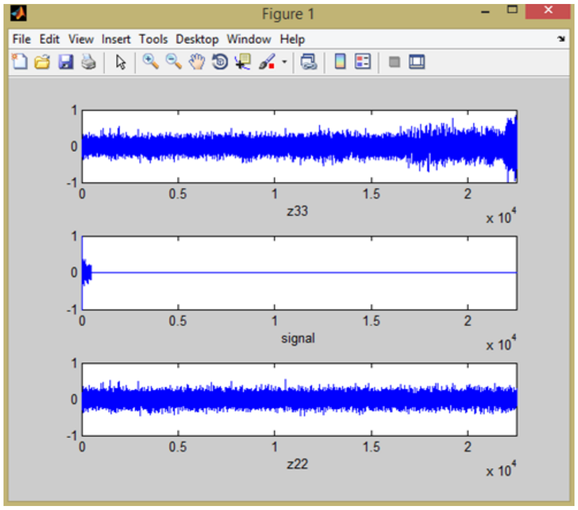


Fig. 1: LMS algorithm and traversal filter

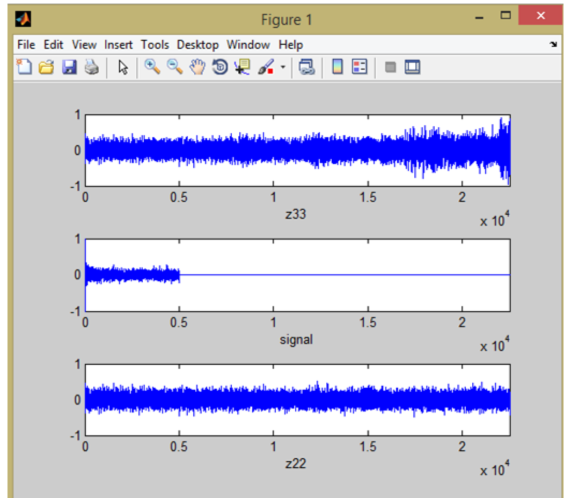
III. ALGORITHM



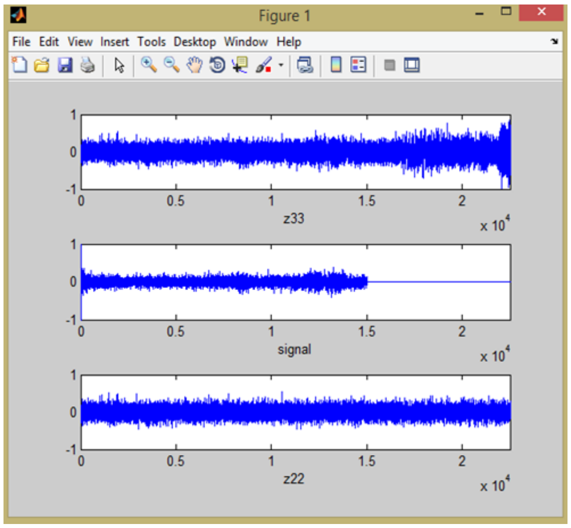
IV. RESULTS AND DISCUSSION



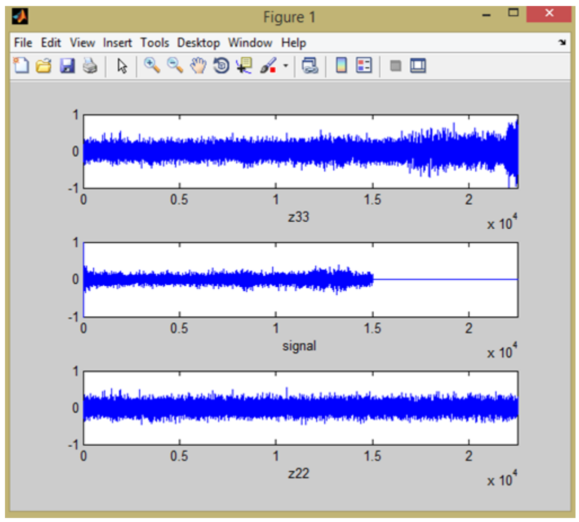
(a)



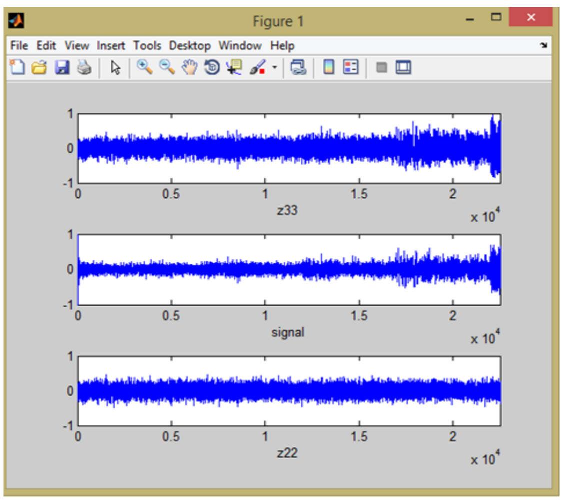
(b)



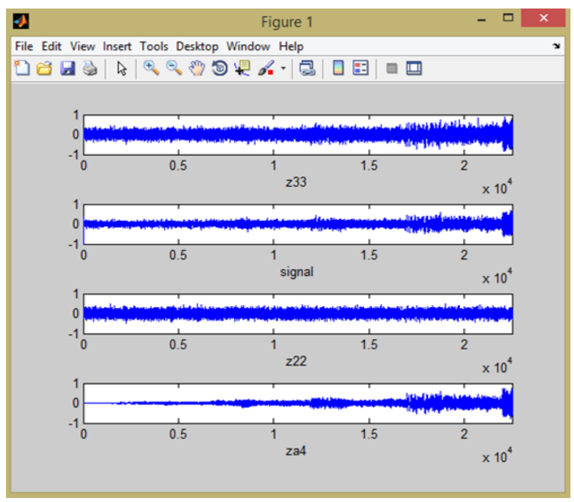
(c)



(d)



(e)



(f)

Fig. 2: Simulation Result for iteration number (a) 500,(b) 1000,(c) 5000,(d) 15000,(e) 200000,(f) 1323227, here z22 is desired signal, z33 is Signal corrupted with awgn and za4 is error signal.

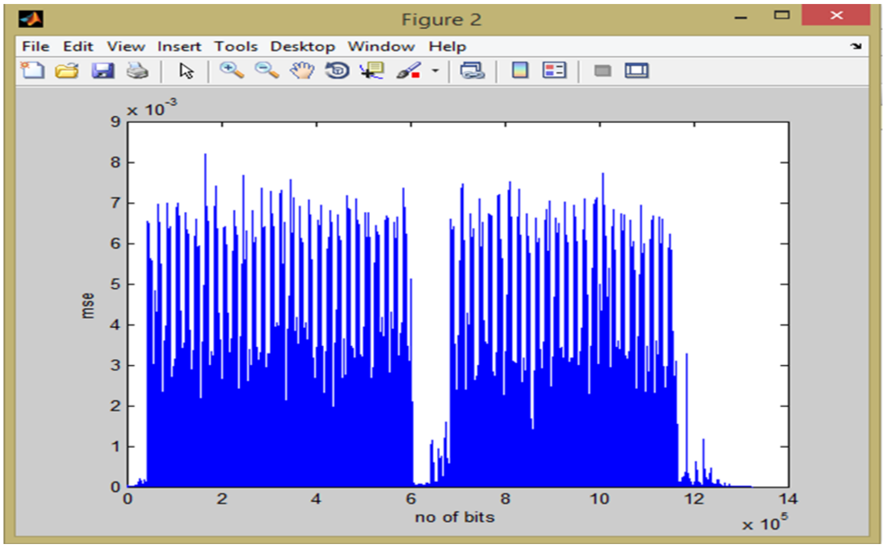


Fig. 3: Simulation Result for mean square error

V. CONCLUDING REMARKS

By the above discussion it is evident that Weiner filter is of utmost importance in controlling the noise during data transmission the error function as shown in figure 3 is minimum when least mean square model is applied to it. The value of mean sis found to be less than 9×10^{-3}. If the data transmission is allowed at a rate of 6 to 6.35×10^{5} bits the mean square error is seen to be minimum. It is thus apparent that filters are input sensitive with respect to noise parameter at output node.

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