ADAPTIVE NOISE CANCELLATION

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***Abstract*—This paper represents the importance and applications of adaptive filters in noise cancellation. The LMS adaptive filter is a self-designing system that can change their coefficients by a negative feedback from the output and therefore it is also called as negative feedback algorithm. LMS and its normalized form NLMS are the one of the best adaptive filter algorithm for noise reduction in communication. In this paper the analysis and implementation is achieved by varying the stability factor such as step size, SNR values and changing the noise level of the input signal. Finally a system with step size 0.0005 and order 2 is developed, which showing good results at different noise power level with limited influence of distortion.**

Keywords—LMS (Least Mean Square), SNR (Signal to Noise Ratio), Variance, Step Size, ANC (Adaptive noise cancellation), NLMS (Normalized Least Mean Square), RLS (Recursive Least Square)

1.1 INTRODUCTION

Noise cancellation is one of the most important process in the signal processing because noise can degrade the performance of the desired signal. There are numerous techniques to minimize the effects of noise, but among these techniques adaptive filtering is one of the best option for noise cancellation [1]. In this project I have implemented the adaptive filter in MATLAB in order to combat the effects of noise in the desired signal. By using adaptive filters it increases the SNR values of the primary signal and at the same time it reduces the mean square error between the primary and reference signal because the reference signal is correlated with primary signal in some way, thereby adaptive filter enhances the input signal [1].

1.2 BACKGROUND OVERVIEW

Unlike conventional filters like FIR or IIR filter, adaptive filters have variable filter coefficients which means they are self-designing system using recursive algorithm. The main concept of this system is to use the LMS algorithm for designing adaptive filter for better noise cancellation. About the principles of the LMS adaptive filter it is based on stochastic gradient descent which primarily for optimization and its cost function is defined as mean square error apart from this the basic principle is derived from steepest descent method. This is a minimization technique and works according to the gradient. The algorithm describes that if the initial points moves in the highest slope direction it leads to either local maxima or minima (Figure 1) [1]. So the next step is to find the direction of local minima for getting minimum error signal for that take the negative of gradient function at the present point and updating the current weights of the filter coefficients in the direction opposite to the gradient direction in each iteration therefore it doesn't require to know about the gradient (Figure 2) [1].

(Update weight vector) = (current weight vector) + (learning parameter) (tap input vector) (error signal)

i.e, $W^{n+1} = W^n + \mu * e * U'$

Another important parameter is the step size which determines whether the LMS algorithm convergent in the mean square and the condition for that is 0< step size (μ) < ($2/lamda_{max}$) [2].

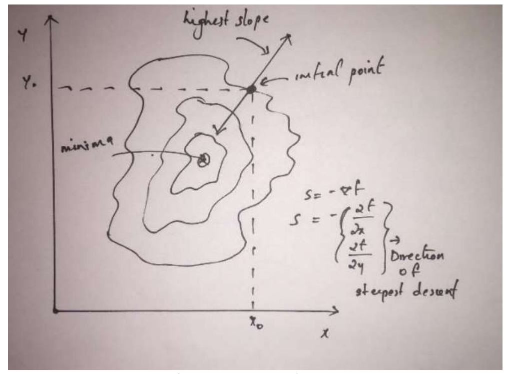

Fig.1. Steepest descent

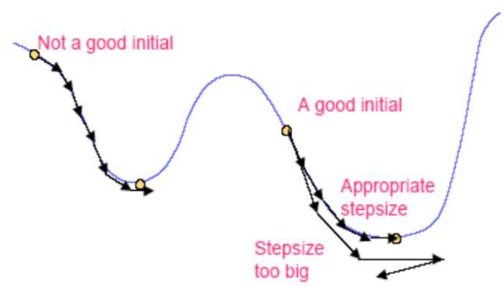

Fig. 2. Gradient descent

In generally LMS algorithm consists of two stages that is filtering process and adaptation process. In filtering process, estimating the FIR filter output by convolving input signals and filter weights then compare it with the desired signal to determine the error signal. In the adaptation process, updating the filter coefficients or weights by estimated error signal recursively [1].

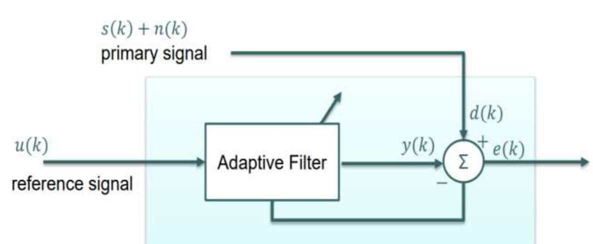

Fig. 3. Adaptive filter

2. THEORY

The ANC technique has two inputs one is the primary signal and reference signal. The reference signal that used here is a additive white gaussian having the same length of the primary signal, which is just the summation of original (recorded sound for 25 seconds) and additive white gaussian noise. The default sampling rate of MATLAB is 8000 therefore the length of the original signal is 2 lakhs.

- In the first step the samples of reference signal is convolved with the filter weights of FIR having order 2.

  Output of filter, $Y = U^T W$

- In the second stage the error signal is estimated by comparing the primary signal with the output of filter.

  Error signal, $e = d – Y$

- In the final stage updating the filter coefficients using gradient descent method for optimising the output and repeat the whole process until receive the last m samples of reference signal where m is the length of W.

  i.e, $W^{n+1} = W^n + \mu * e * U'$

The things I have learned by this project is what is adaptive filter and how its more efficient than other filters because it's a self-designing system. This project helped me to know more about LMS algorithm, stochastic gradient descent and steepest descent method not only for designing filter but how I can use this for designing machine learning systems because this algorithm has vast applications in many field.

The difficulty I faced in this project on two areas one in taking rms value and another in finding the right value for step size. When I try to normalise the input signal the output showing more distortions which means without doing normalising it giving good result, but once I increased the period of recording to 25 seconds it giving good results with the normalized input. Second difficulty is in choosing the step size for getting good output and finally I found a better value by using this equation i.e, $0 < \mu < (2/\text{input power signal})$.

3. RESULTS AND DISCUSSION

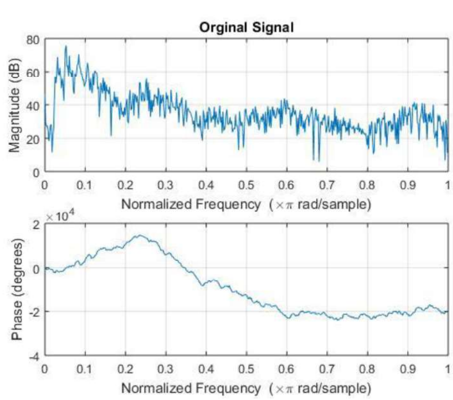

Fig. 4. Frequency response of original signal

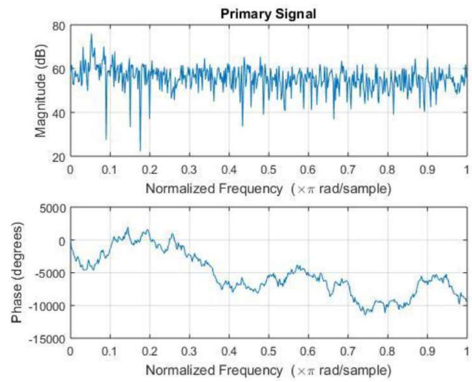

Fig. 5. Frequency response of primary signal

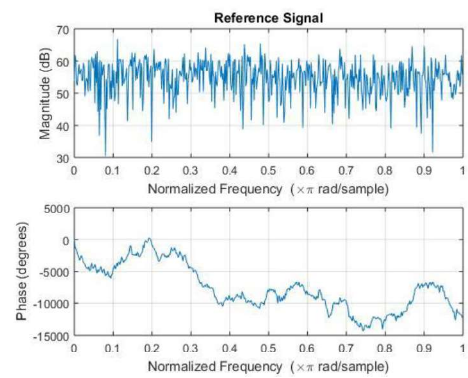

Fig. 6. Frequency response of reference signal

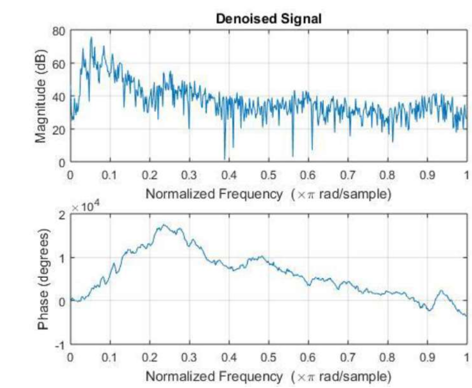

Fig. 7. Frequency response of denoised signal

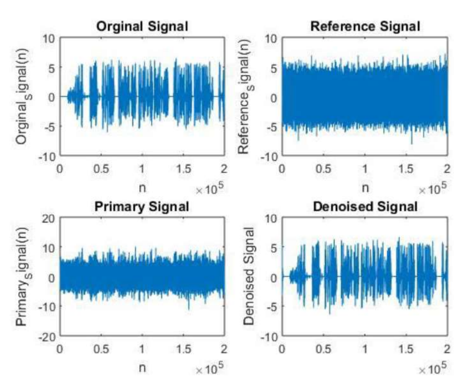

Fig. 8. All signals in time domain

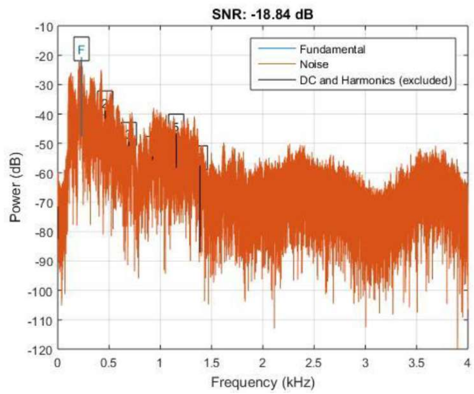

Fig. 9. SNR of original signal

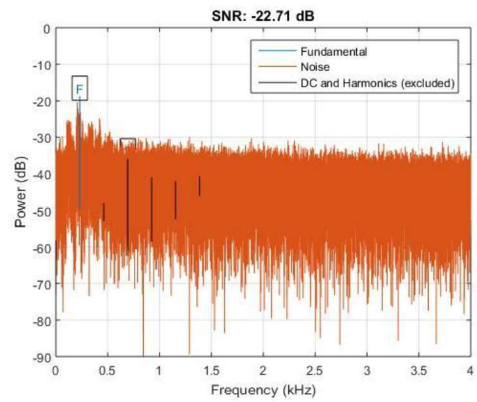

Fig. 10. SNR of primary signal

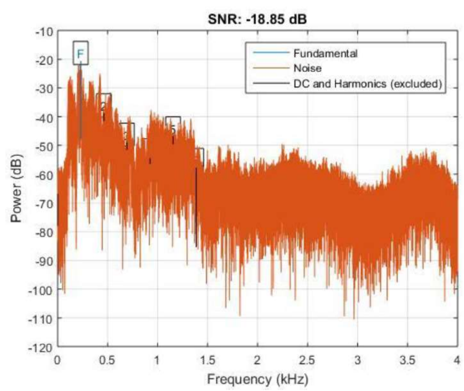

Fig. 11. SNR of denoised signal

	Test Signal 1	Test Signal 2	Test Signal 3	Test Signal 4
Noise Power (db)	+ 5 db	+5 db	+ 10 db	-7 db
Step Size µ	0.0005	0.035	0.0005	0.0005
Elapsed Time	2.1435	4.0927	2.1532	2.0815
Variance of added noise	2.0343	2.4327	8.108	0.1628
Snr value of original signal	-18.84 db	-14.51 db	-15.82 db	-17.54 db
SNR value of denoised signal	-18.85 db	-15.37 db	-15.89 db	-17.53 db

Table 1

From figure 4 to figure 7 shows the frequency responses of different signals. By comparing figure 7 and figure 1 say that the output signal or denoised signal has almost same frequency response of original signal, neglecting the small phase difference in denoised signal. Figure 8 showing all the signals in time domain and from that its clearly understood that the waveform of denoised signal is 95% similar to the original signal. Here the power level of noise (white gaussian) is 5 db and the system showing good result. One more reason for system showing good performance is because of the snr value of denoised signal. By comparing figure 9 and figure 11, we can say that snr is almost same which means output same as original signal. In the figure 10 its shows that the snr value of primary signal decreased to -22.71 db which means the influence of noise increased and by doing adaptive filtering the snr value increased to -18.85 db which is equal to the snr value of original signal. I had recorded this speech for 25 seconds and system took 2.1435 seconds, but when I recorded the signal for 10 seconds the system gives output within 1.273 seconds. Variance of added noise is 2.5350 and one thing about the variance is that the power level of noise is proportional to the variance. I had tested the system with 5 different signals with different power levels of noise and step size. When the step size increased from 0.005 to 0.035 and power level set to +5 db for noise then it shows more distortion and the difference between snr value of denoised and original signal is 1.32 db even the frequency response of both denoised and original has significant difference due to the influence of noise and poor selection of step size. Even if the power level of noise signal increases to +10 db the output is still clear if the step size is 0.005 and for other value of step size the influence of noise is high. Table 1 shows the comparison of different test signals for different values of noise power level and step size.

By testing different signals with different parameters helped me to understand the influence or importance of each parameter and how to choose parameters according to different situations. It also helped me to evaluate the system whether it is efficient or not based on analysing the frequency response, snr values, variance and signal waveforms.

4. CONCULSION

ANC is a effective technique for noise cancellation and it is based on adaptive algorithm such as LMS, NLMS and RLS, which monitor the environment and change the filter weights accordingly to give the best result. Adaptive filters based on LMS algorithm can shows excellent results by carefully selecting the filter parameters such as filter order and step size. They are widely used because of reduced time complexity and easiness in implementation by using gradient descent method.

5. REFERENCES

[1] "Adaptive Filter Theory", by Simon Haykin, 4ed,2002 Prentice Hall.

[2] B. Widrow and S.D. Stearns, Adaptive Signal Processing, Prentice Hall, Englewood Cliffs, NJ, 1985.