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Analysis of Adaptive Filtering for Noise Reduction in Communications Systems

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Abstract- In this paper, a comprehensive and comparative anlaysis of three adaptive filtering algorithms is done to help improve modern noise reduction mechanisms in communication systems. Noise cancellation as we know is done by various algorithms and to ascertain the best of them we have tried to implement noise cancellation on different noises through different algorithms. The implementation of Adaptive Filtering is the need of the modern communication systems and thus we desire to implement various algorithms for adaptive filters using MATLAB by Mathworks. The MATLAB programs for the following three adaptive filtering algorithms -LMS, QLMS and HR-QLMS was implemented on MATLAB and simulation results for the same are compared for different input noises to a hello voice sample. The basis of analysis is SNR measured before filtering, SNR measured after filtering and the absolute errors in implementation of different adaptive filtering algorithms.
Keywords-Adaptive filter, LMS, QLMS, SNR

I-Introduction

Today, Digital filtering has become the need of the hour for fast and reliable modern communication systems. It has become of immense importance for speech and audio signal processing as most of the world’s communication takes place through voice calls and video conferences. Modern communication systems rely heavily on the Digital Signal Processing applications. Digital Signal Processors consist of microprocessors and embedded systems designed for specific applications. As far as application of noise reduction from speech is concern, adaptive filters have seen considerable increase in their applications as they are found to give excellent performance. Adaptive filters are being used for many a applications in DSP and their performance is getting enhanced by the day. They deal with the noise affected digital signals by continuously adapting to the incoming signal points and tracking the desired output sequence[1-4]. The parameters of the adaptive algorithm are adjusted continuously with time-varying signal environment. Adaptive signal processing uses a filter, usually an FIR filter whose coefficients are adjusted continuously with time to track the desired output signal. The adaptive systems are judged by their performance factors which include mean square error, rate of convergence, mathematical robustness, algorithm complexity and system execution time. Many of the applications, noise cancellation is one of the most sought application today in the communication industry. It is categorized as passive noise control and active noise control. The process of noise cancellation is done to mitigate unwanted background noise from the corrupted signal. The need for faster, better and more efficient algorithms has led to the continuous surge in development of new technologies to tackle the issue of active noise control [5-8]. Though the LMS algorithm is still widely used, its main disadvantage is its slow rate of convergence. Hence recent research in adaptive filtering has found out algorithms such as Quaternion LMS (QLMS), Filter edx LMS (FxLMS), HR-QLMS (HR-QLMS) which have shown significant improvements. The filtering and processing of the noise-corrupted speech signal captured by devices like headphones is facilitated by use of multiple sensors that help generate a counter-noise signal called the reference signal which is reduced from the received audio signal to give an almost similar audio signal as the one sent so that it is free from background interferences.

Active noise cancellation is a modern and progressing technological advancement. It has been enhancing ever-since the start of 20th century. Active noise cancellation is a significant requirement as noise is a health issue today. Studies have found that prolong exposure to unpleasant noises and even to loud music can lead to serious prolonged health problems. The noise cancellation algorithm traditionally uses an input signal which is correlated to the background noise that has corrupted the original audio and is called the reference signal. This reference input is then passed through the adaptive filter system which traces the original audio/voice input by successive subtraction of the reference noise from the primary input to the filter. This process helps getting rid of noise from the corrupted signal and translates it into the noise free signal. The fundamental motto behind this noise reduction process is to genearte an input signal for the filter which is equivalent to the noise corrupting the initial input audio .It is generated such that it has similar properties like that of input's noise but has an opposite phase to help in destructive interference with the primary input. The adaptive noise reduction system makes use of sensors to read incoming audio and the background noise to help produce reference signal for proper cancellation [9-10].

II. Brief overview of Digital Filters

The various superior characteristics of digital filters are lower power consumption, small area requirements and fast speed of operation as compared to the analog filters. The digital filters are widely used in many applications of DSP(digital signal processing) and their demand is growing rapidly due to the new inventions and fast-growing technological advancements in the field. Digital filters employ use of Analog to Digital conversion and Digital to Analog conversion done using typical ADCs and DACs. Digital filters are categorized in two categories as follows-

1.FIR (Finite impulse response) Filter- This filter has a finite impulse response or transfer function. FIR filters have a linear phase response .The FIR filters do not have any kind of feedback from output and are non-recursive filters. They have a better stability performance as compared to the IIR filters.

2. IIR (Infinite Impulse Response) Filter-These filters unlike FIR filters have an infinite impulse response. These filter have a feedback path from output towards input ,so that the response of the filter in this case relies upon past values of the output response y(n) and also on the present and past values of input signal x(n). The advantage of thses filters is derived from the feedback property present in the system. However this also produces a disadvantage in the IIR filter which is stability. The IIR filters are tough to maintain stable as compared to FIR filters and hence their performance is affected if the filter is not well designed.

III. Adaptive Filters

Filters whose coefficients of the transfer function's numerator and denominator are adjustable according to the optimization alogorithm coefficients are called adaptive filters. Thus, many computationally efficient algorithms for adaptive filtering have been developed. Adaptive filters are being implemented for the recovery of data corrupted by noise and other disturbances .The adaptive algorithm depends on four factors given as :-

1. Error- The difference between optimal and actual value of error.

2. Rate of Convergence- The maximum number of loops executed to finish adaptation and updation process.
3. Stability of the system – The update equation used for the algorithm must provide stable updation throughout the algorithm execution.
4. Complexity – The complexity of the the system and the hardware where the filter is to be used. The filtering process is a mechanism to eliminate noise from a received corrupted signal.The filtering problem is characterized by some terms like smoothing, filtering, prediction and deconvolution.

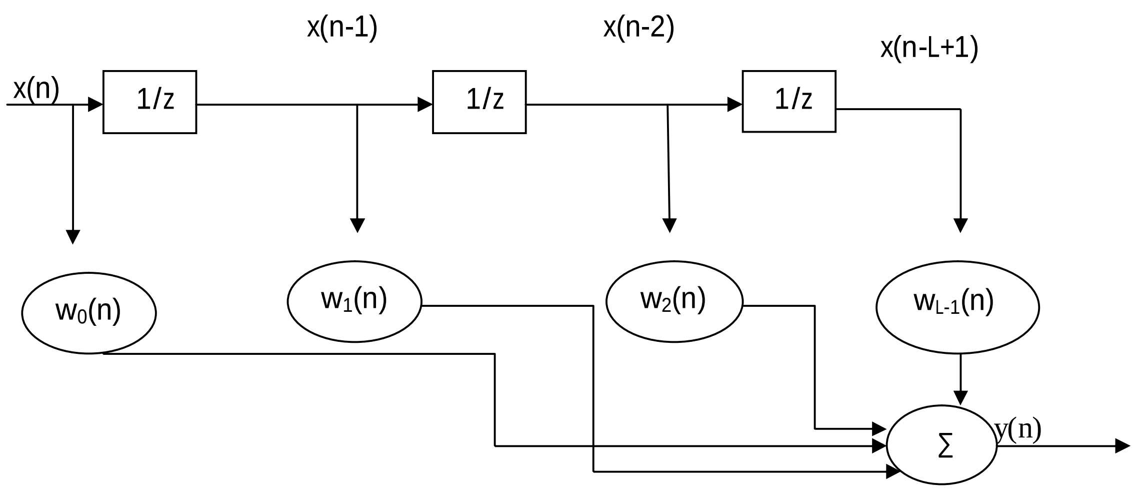

Fig.1 FIR based Adaptive Filter

This work is focused on developing and analyzing noise canceller algorithm and determining their efficiency in terms of error and numerical complexity. The approach and methodology used to achieve the intended results was to start with analysis of the most common ,widely used and reliable adaptive algorithm –LMS(or Least Mean Square) algorithm .Then variants of LMS like QLMS , HR-QLMS have been implemented to check upon the enhancements possible in the future.

IV About the algorithms implemented

1. THE BASIC LEAST MEAN SQUARE (LMS) ALGORITHM

LMS(Least mean square) is a type of adaptive program based on stochastic adaptations. It is helpful in two ways. This algorihm and its variants are popularly used for acquiring knowledge about a system or for the purpose of linear prediction and it allows us to do so without the need for exact knowledge a signal's charcateristics such as the covariance which are not easily comprehended by a system. Moreover, these methods uniquely track down any signal from its variations quite remarkably.
The LMS algorithm is a type of adaptive algorithm used to emulate a digital filter by continuously adapting its coefficients to the changes in input primary signal by the help of an optimization algorithms.Though the LMS algorithm is stiil widely used ,its main disadvantage is its slow rate of convergence.

2. QLMS (QUATERNION LEAST MEAN SQUARE) ALGORITHM

The widely used LMS algorithm has a slight disadvantage of executing a slow convergence rate than many other adaptive algorithms. The QLMS Algorithm is used for adaptive filtering of 3-D and 4-D processes. These processes show complex non-linear dynamics and coupling between the dimensions. In this algorithm the complex vector space is extended to make use of three imaginary parts of the quaternion. Quaternions are extension of complex numbers and as the name suggests ,they can only contain maximum four variables. The QLMS algorithm is superior to the LMS algorithm since it makes use of the signal properties like covariance and pseudo-covariance of input signal unlike that of LMS and hence allows the incorporation of unsymmetrical data in its update analysis. QLMS is used image processing and wind vector processing.

3. HR –QLMS ALGORITHM

The HR-QLMS is an extension of the QLMS algorithm . The HR-QLMS is obtained by modifying the derivative rules of the quaternion algebra. HR-QLMS is used in the industry for purposes like image processing and wind vector processing.

V. Block diagram for noise reduction system using adaptive filter

The block diagram for adaptive noise cancellation system is as given below :-

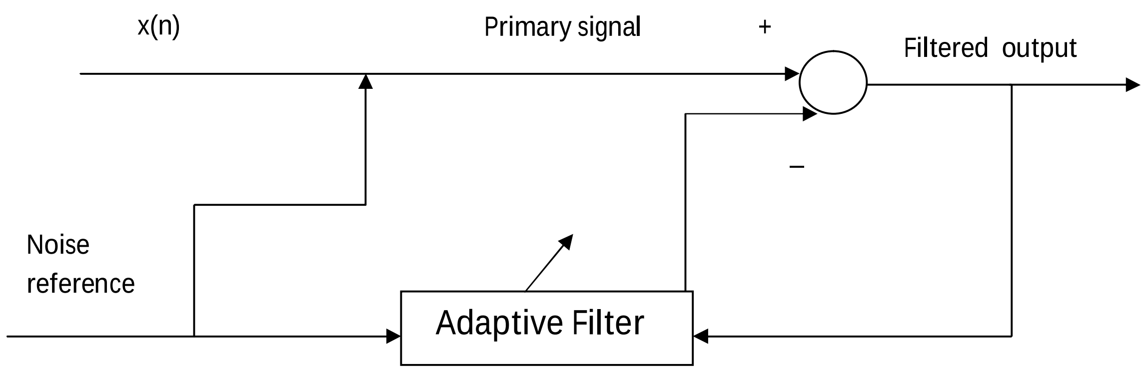

Fig.2 Adaptive Noise Cancellation System

VI. Simulation and results

The simulations of the mentioned algorithms have performed using MATLAB (Matrix Laboratory) by Mathworks. MATLAB is a software used for computational analysis in a variety of scientific and technical calculations.It is used in the fields of control and automation, image processing and speech signal processing and other digital signal processing schemes. MATLAB allows for simple matrix manipulations , plotting various mathematical functions and creation of user interfaces.

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1. SIMULATION RESULTS FOR LMS ALGORITHM

(a) BIKE NOISE

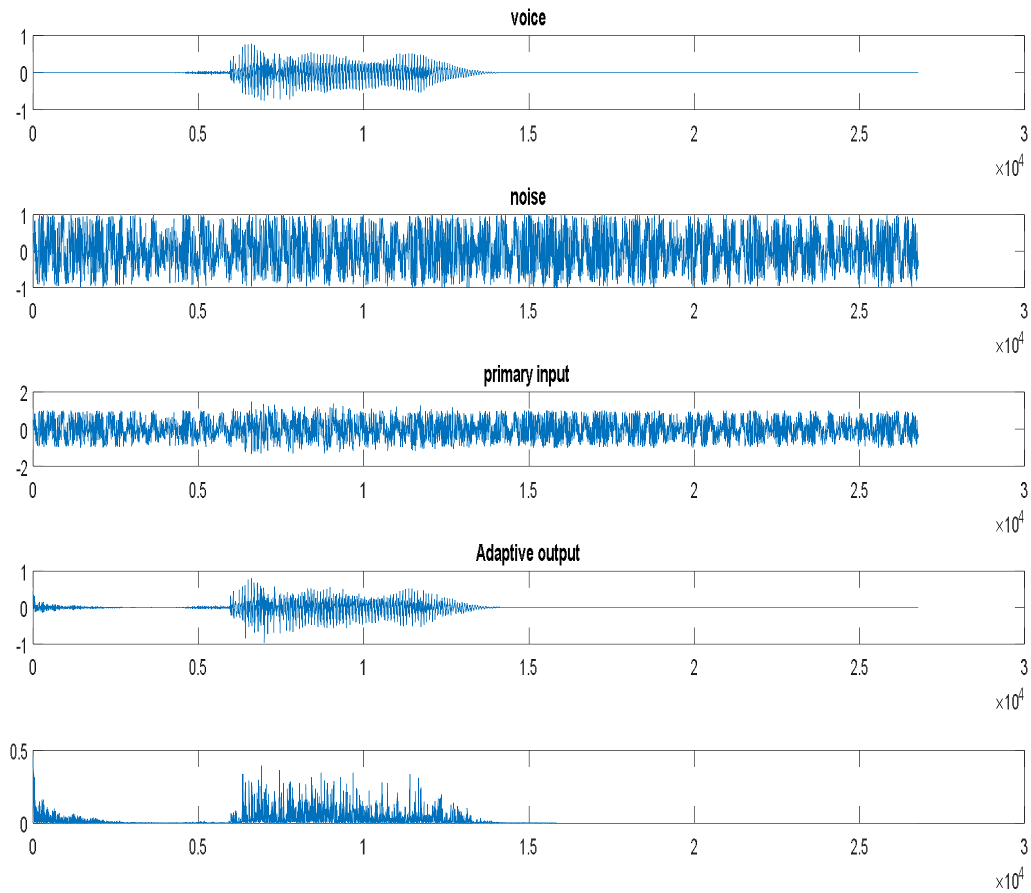

Fig.3

1.SNR measured before filtering process through the algorithm is 0.2018 dB.

2.SNR measured after filtering process through the algorithm is 7.7800 dB.

3.The maximum absolute error is found to be 0.38.

(b) BEAR GROWLING NOISE-

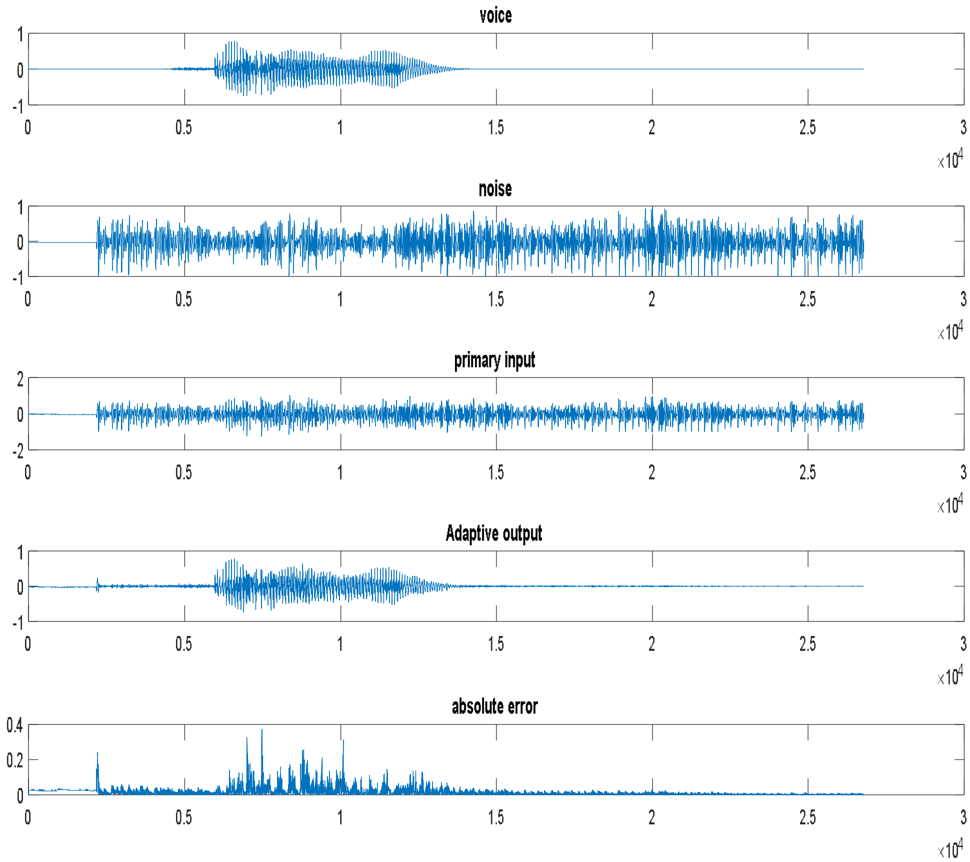

Fig.4

1.SNR measured before filtering process through the algorithm is 0.5795 dB.
2.SNR measured after filtering process through the algorithm is 10.93 dB.
3.The maximum absolute error is found to be 0.42.

In the above figures , the voice used is a 1.3 sec long "**hello**" taken as "**.wav"** file .The noise input for the first case is bike-noise with a duration of around 2.75 seconds and for the second case is bear's growl noise with a duration of around 2.75 seconds.
The error in the adaptive outputs in both the cases at the starting of the plots is high as the adaptive weight vector value initially is set as zero and it gets reduced as the weight vector updates itself. Hence ,the error after the voice ends is near to zero.

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2. SIMULATION RESULTS FOR QLMS ALGORITHM

(a) BIKE NOISE-

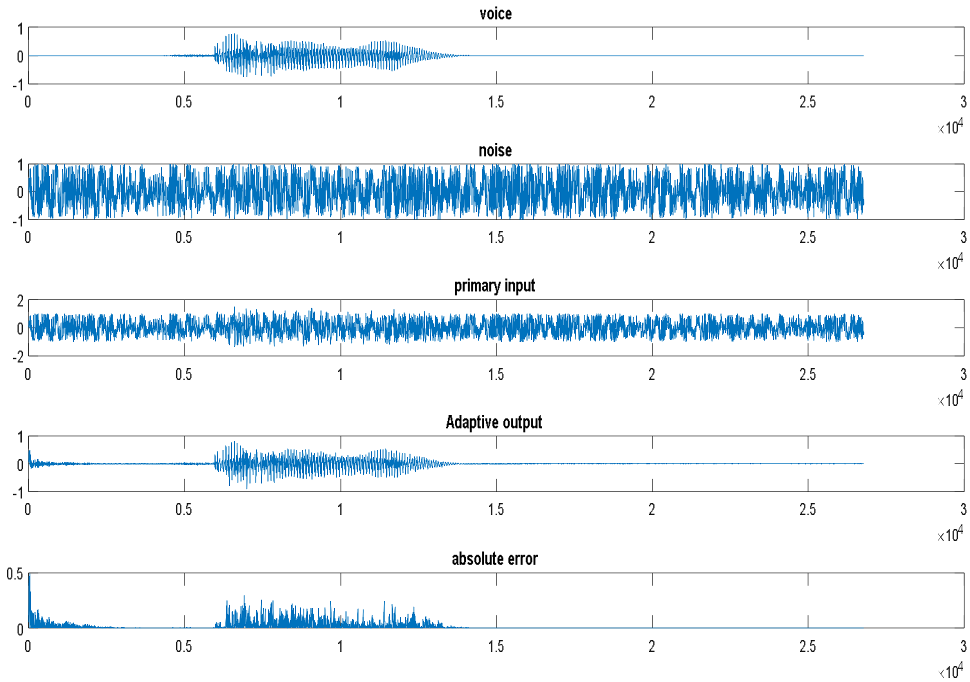

Fig.5

(b) BEAR GROWLING NOISE-

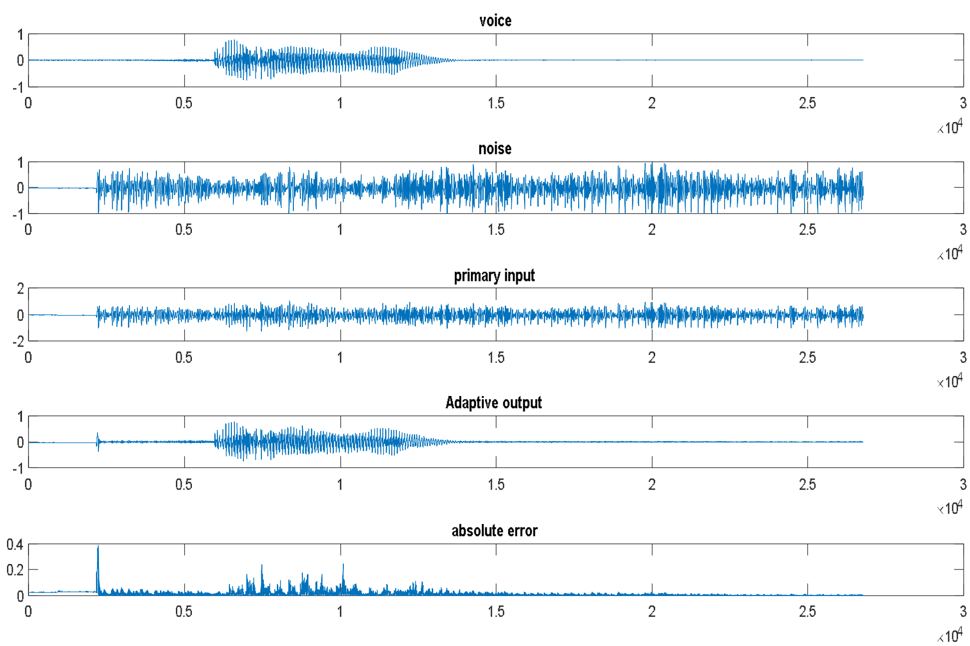

Fig.6

In the first case (fig. 5) where bike noise is taken the following data is recorded :-

1.SNR measured before filtering process through the algorithm is 0.2018 dB.
2.SNR measured after filtering process through the algorithm is 10.08 dB.
3.The maximum absolute error is found to be 0.29.

In the second case (fig.6) where bear growling noise is taken the following data is recorded :-

1.SNR measured before filtering process through the algorithm is 0.5795 dB.
2.SNR measured after filtering process through the algorithm is 12.58 dB.
3.The maximum absolute error is found to be 0.40.

In the above figures , the voice used is a 1.3 sec long “**hello**” taken as “**.wav”** file .The noise input for the first case is bike-noise with a duration of around 2.75 seconds and for the second case is bear’s growl noise with a duration of around 2.75 seconds.
The error in the adaptive outputs in both the cases at the starting of the plots is comparatively lower as compared to the LMS algorithm as the QLMS algorithm makes use of both the covariance and pseudo-covariance of the signal unlike that in the LMS. The error gets reduced as the weight vector updates itself.

3. SIMULATION RESULTS FOR HR-QLMS ALGORITHM

(a) BIKE NOISE

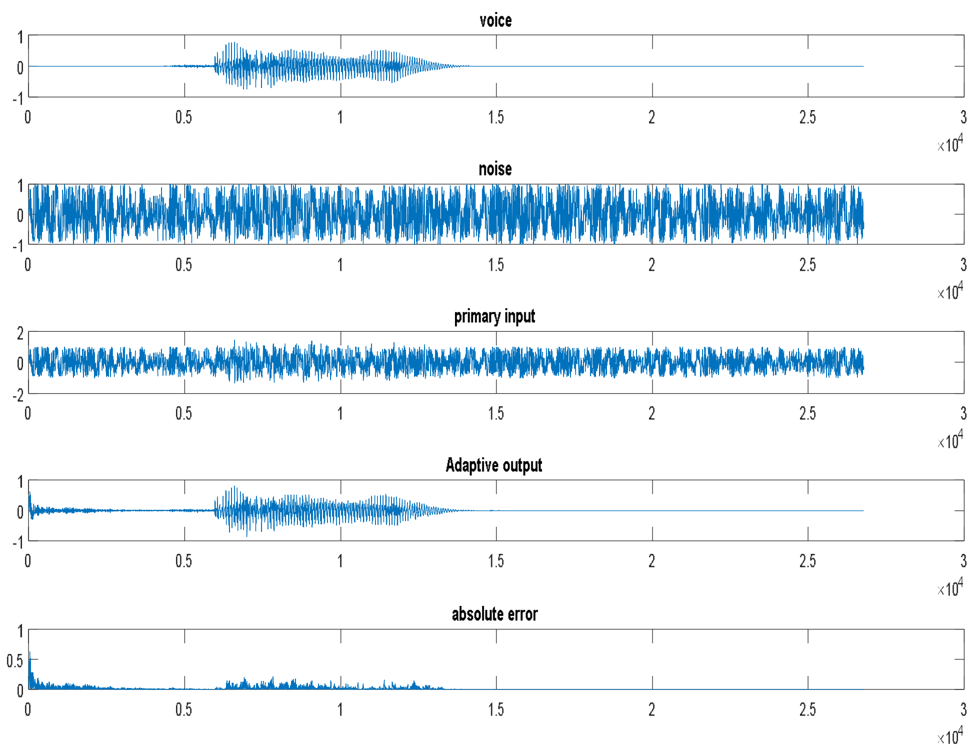

Fig.7

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(b) BEAR GROWLING NOISE

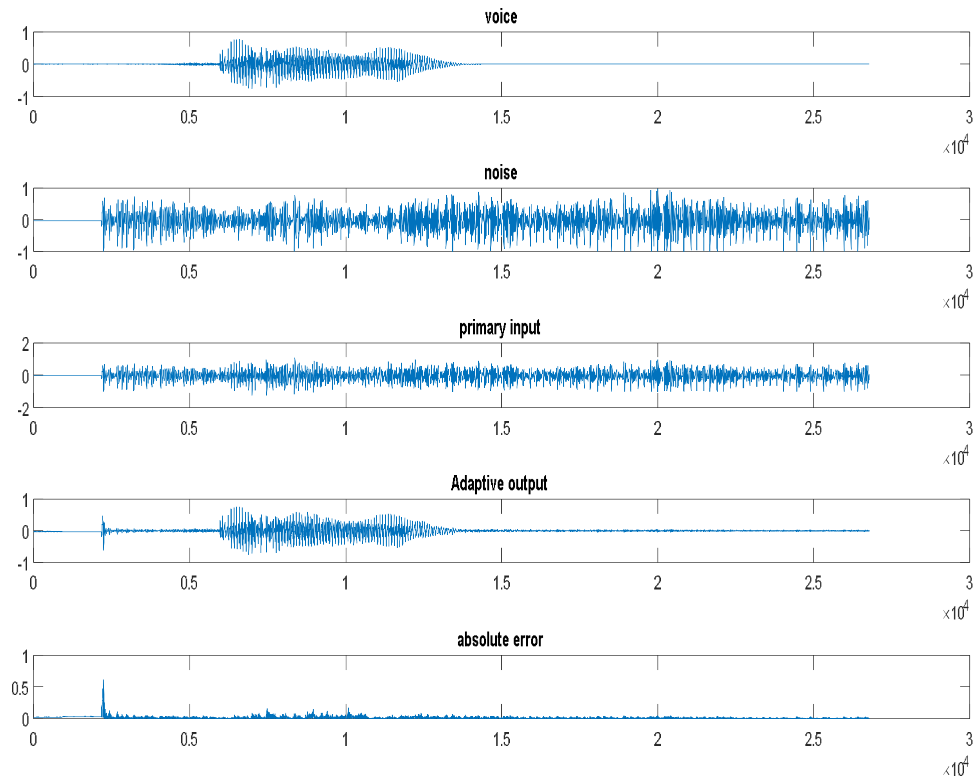

Fig.8

In the first case (fig. 7) where bike noise is taken the following data is recorded :-

1.SNR measured before filtering process through the algorithm is 0.2018 dB.
2.SNR measured after filtering process through the algorithm is 10.91 dB.
3.The maximum absolute error is found to be 0.22.

In the second case (fig.8) where bear growling noise is taken the following data is recorded :-

1.SNR measured before filtering process through the algorithm is 0.5795 dB.
2.SNR measured after filtering process through the algorithm is 12.84 dB.
3.The maximum absolute error is found to be 0.60.

VII. Conclusion

The work presented in this report describes the performance analysis of the LMS, QLMS and HR-QLMS algorithms. The simulation results obtained during the analysis show that the HR-QLMS algorithm is superior to the QLMS and LMS algorithms. And the QLMS algorithm is better as compared to the LMS algorithm. The observations taken during analysis show that the SNR measured for the adaptive output is best for HR-QLMS algorithm followed by the QLMS algorithm and then by the LMS algorithm. The results have been obtained for two different noises augmented upon a “hello” voice of duration 1.3 seconds and they suggest that the HR-QLMS algorithm is the most efficient .However the complexity of the LMS algorithm is the least and that of the HR-QLMS is the most among the three adaptive filtering algorithms.

For the bike-noise case ,the maximum absolute error is minimum for the HR-QLMS case .However, for the bear growl noise case ,the maximum absolute error is minimum for the QLMS algorithm. The HR-QLMS algorithm uses complex partial derivatives for the updation purpose .It can be safely concluded that HR-QLMS algorithm is superior and will give good results for complex systems but for simple and cheap system purposes LMS or QLMS can also be used. The future of adaptive filtering is quite open to discovery as this field has much to offer to the new technological advancements in sectors like image processing, ANC (active noise cancellation) ,wind vector analysis and other areas of DSP .

VIII. References

1. N. Sireesha, K. Chithra and T. Sudhakar, "Adaptive filtering based on least mean square algorithm," 2013 Ocean Electronics (SYMPOL), Kochi, 2013, pp. 42-48.
2. S. A. Hosseini and S. A. Hadei, "A novel noise cancellation method for speech enhancement using variable step-size adaptive algorithm," 2014 9th International Symposium on Communication Systems, Networks & Digital Sign (CSNDSP), Manchester, 2014, pp. 249-254.
3. Mirza, Alina & Kabir, Sumrin Mehak & Ayub, Sara & Sheikh, Shahzad. (2015). Impulsive Noise Cancellation of ECG signal based on SSRLS. Procedia Computer Science. 62. 196-202. 10.1016/j.procs.2015.08.440.
4. Patel, Shruti R. et al. “Comparative Study of LMS & RLS Algorithms for Adaptive Filter Design with FPGA.” (2014).
5. Rad Sanz, Pablo Corral and Antonio CBzar de Castro Lima ‘Adaptive Beam forming techniques for OFDM based WLAN systems: A comparison between RLS and LMS’, IEEE 2004.
6. Quanzhen, Huang et al. “Comparison of LMS and RLS Algorithm for Active Vibration Control of Smart Structures.” 2011 Third International Conference on Measuring Technology and Mechatronics Automation 1 (2011): 745-748.
7. E. Eweda, "Comparison of RLS, LMS, and sign algorithms for tracking randomly time-varying channels," in IEEE Transactions on Signal Processing, vol. 42, no. 11, pp. 2937-2944, Nov. 1994.
8. E. C. Mengüç and N. Acir, "An adaptive noise canceller based on QLMS algorithm for removing EOG artifacts in EEG recordings," 2017 International Artificial Intelligence and Data Processing Symposium (IDAP), Malatya, 2017, pp. 1-4.
9. C. C. Took and D. P. Mandic, "The Quaternion LMS Algorithm for Adaptive Filtering of Hypercomplex Processes," in IEEE Transactions on Signal Processing, vol. 57, no. 4, pp. 1316-1327, April 2009
10. Neto, Fernando G. Almeida and Vitor H. Nascimento. “Second-order analysis of the RC-WL-QLMS algorithm.” 2012 International Symposium on Wireless Communication Systems (ISWCS) (2012): 466-470.