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Comparison of Different Adaptive Speech Algorithm to Correct Wow and Flutter in Audio Signals

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***Abstract*— This proposed research work describes a system which is used to remove Wow and Flutter from audio signal using Advanced Adaptive Speech Algorithm. It is impossible to completely remove Wow and Flutter from audio signal but its effect can be reduced significantly. It occurs during the process of sound reproduction. It is the group of tones created by the irregularities in turntables or tape drive speed during reproduction, duplication or recording. Wow occurs due to irregularities at low frequency whereas at high frequency irregularities, Flutter occurs.**

Wow and Flutter could be found in old gramophone recordings, wax cylinders, on the magnetic and optical sound tapes. Least Mean Square Algorithm uses Adaptive Filter which adjusts their coefficient in order to minimize the required wobble effects in audio signal. Results show that Advanced Adaptive Speech Algorithm can significantly diminish the effects of wow and flutter.

***Keywords*— Least Mean Square Algorithm, Normalized Least Mean Square Algorithm, Recursive Least Square Algorithm, Mean Square Error, Short Time Energy and Elapsed Time.**

I. INTRODUCTION

This research work describes a system which is used to remove Wow and Flutter from audio signal using Advanced Adaptive Speech Algorithm. It is impossible to completely remove Wow and Flutter from audio signals but the effect of same can be reduced. It occurs during the process of sound reproduction. It is the group of tones created by the irregularities in turntables or tape drive speed during reproduction, duplication or recording. Wow occurs at low frequency irregularities whereas at high frequency irregularities Flutter occurs. In this proposed work advanced adaptive algorithms will be used in order to minimize these effects in audio signal. Least mean square algorithm, Normalized Least mean square algorithm, Recursive Least Mean Square Algorithm, etc are the types of advanced speech adaptive algorithm. Hence by comparing the results obtained by the different algorithms we will find the better algorithm chosen for this purpose that provides efficient results. Wow and Flutter is the distortion of audio signal. Wow occurs in the frequency range of 0.5 Hz to 6 Hz hence called as low frequency distortion of audio signal whereas Flutter occurs at the frequency range of 6 Hz to 100 Hz so Flutter it called as high frequency distortion of audio signal [1].Wobble is caused by the speed fluctuation in tape. Both Wow and Flutter are separate terms. It is found that Flutter is at its peak value when actual frequency of wobble is 4 Hz and its value is least at below and above of the frequency 4 Hz. . This is clearly demonstrated by the following diagram. [2]

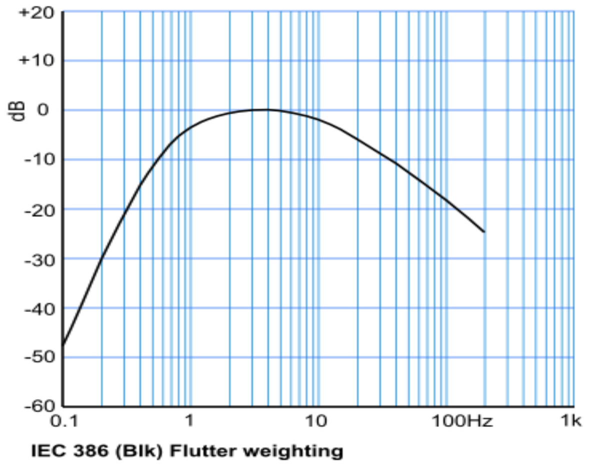


Fig. 1 Variation of wow and flutter

A. Wow and Flutter on the basis of their causes

Following are some of causes of wow and flutter [3]**:**

Tape Speed: Speed of tape is controlled by motor and the pinch roller. Pressure must be maintained within specification since a high take up tension will pull the tape between the motor and pinch roller. Due to this motor is not able to control the speed of tape. Similarly, if a high back tension is present then motor can not also control the speed of tape. Since the capstan motor is synchronous in nature then it leads to variation in tape speed if main frequency is not correct. These variations results in fluctuation and this become the cause of wow and flutter in audio signal.

Wrong Tape Tension: Wrong tape tension introduces error when tape is dragged at different standard value of tension. These error results in wow and flutter so, tape tension is important to control in all modes of tape machines.

a) During playback: A certain amount of tape tension is required during playback mode in order to provide a good head contact with tape.

b) During spooling: A certain amount of tape tension is required during spooling mode in order to provide a safe running of tape.

c) During spooling and breaking: A certain amount of tape tension is required during spooling and breaking in order to prevent from over stressing and stretching of tape.

Other Reasons: If moist and sticky tape, wrong break adjustment, wrong pinch roller pressure, dirty capstan, dirty pinch roller, bearing problems and worn out idler, etc.

II ADVANCED ADAPTIVE SPEECH ALGORITHM

A. Introduction to adaptive filter

Filtering is the process of obtaining the information from the signal. As the name suggests it filters the signal so that unwanted signal contaminated from the signal and desired signal obtain. An adaptive filter adjusts their parameters in order to minimize the error function between the desired output $d(n)$ and actual output $y(n)$ and hence the error free signal is obtained at the output. This error function is also known as the cost function. An adaptive filter is a digital filter with an adaptive algorithm which is used to modify the filter's coefficients [4]. Filters are used in biomedical instruments, as the frequency of biomedical instruments are very low so digital filters are much popular for low frequency applications.

For any application of the adaptive filters, the input signal and the reference input are used for example the least mean square is used to adjust the weight of the adaptive filter in order to minimize the error. The best solution to remove the unwanted signal or noise from the original input signal, the reference signal contains the noise or unwanted signal must be filter out by the use of adaptive filter due to its good performance and reliability.

B. Least mean square algorithm :

LMS stands for Least Mean Square. This algorithm was developed by Widrow and Hoff in 1959 through their studies in pattern recognition [5]. Since its invention, it has been in use for variety of signal processing applications including denoising of image. Since images are suffered from white Gaussian noise so, least mean square algorithm is used to denoise the images. It is also implemented in image compression which is very demanding application in telecommunication image compression. This technique is used to control the distortion in image after transmitting the image through channel. In the presence of numerical errors caused by finite precision arithmetic LMS performs robustly. There is efficient use of memory. It does not require prior knowledge of signal statics. In least mean square algorithm weights are obtained on the basis of estimation but their value improves constantly with time as weights are adjusted [6].Its convergence rate is not too express because of Eigen value spread. It is stochastic gradient descent method in which adaptive filter is used to minimize the error [7]. At each iteration, to make the exact approximations, weight vector is computed by steep descent algorithm and it converges to the optimum wiener solution. The exact measurement of gradient vector is impossible because it would require prior knowledge of both the matrix that are tap input autocorrelation matrix R and the cross correlation vector p. The gradient vector must be estimated from the given data when somebody is doing work in unknown environment. After the estimation of gradient vector a relation is obtained through which it can update the tap weight. The input signal $x(n)$ is an audio signal containing wow and flutter. $d(n)$ is the reference signal and act as a secondary input to the adaptive filter. Least mean square algorithm is used to subtracts the input signal $x(n)$ from the signal received by the filter that is $y(n)$. $w(n)$ is the current tap weight vector. Hence we get the following equation:

$w(n+1) - w(n) + \mu x(n)[d*(n) - x^H(n)w(n)$ ------- (1)

where, μ = step size parameter

x^H(n) = Hermit of a matrix u

$d*(n)$ = Complex conjugate of $d(n)$

The above mentioned equation may write the result in the form of three basic relations as follows:

Filter output

$y(n) = w(n)' * d(n)$-------- (2)

Estimation Error or Error Signal

$e(n) = x(n) - y(n)$------ (3)

Tap weight adaptation

$W(n) = w(n) + 2 * \mu * d(n) e * (n)$ ----- (4)

From equation (2) and (3) we can calculate the error $e(n)$ and its calculation is based on the current tap weight vector $w(n)$. In equation (4) the term $x(n)e*(n)$ represents an adjustment to calculate the current tap weight vector $w(n)$.

C. Least mean square algorithm using adaptive filtration

Adaptive filter is a self adjusted filter. The term self adjusted is used because its coefficients are not fixed, it reduce the error in the original signal by adjusting its coefficients. Transversal filter is the most common form of adaptive filter so, we use least mean square algorithm [5]. By using the concept of adaptive filtrations LMS has been implemented in equalization of data communication channel [6]. Data can be transmitted reliably through a communication channel and demodulate the signal perfectly at the output without causing any error if the channel is ideal. It is used in time varying system identification [8]. System identification is the central issue that occurs in various applications of least mean square such as in echo cancellation, channel equalization and teleconferencing etc. To find the parameter of unknown model the input- output desired response can be calculated and error that is optimized. There is a trade-off effect with the stepsize choice. If the value of μ is large then it gives better tracking ability in a non-stationary environment but it suffers from large effect of noise and if the value of μ is small then it has poorer tracking ability but it suffers from smaller effect of noise.

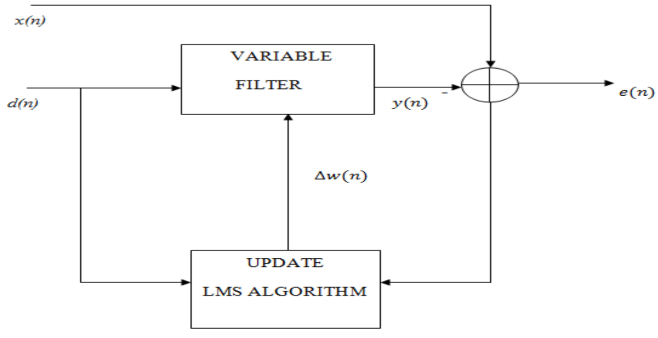


Fig. 2 Block diagram of Least mean square algorithm

D. Normalized Least Mean Square Algorithm

Normalized Least Mean Square Algorithm is an extension of the standard LMS Algorithm. Its practical implementation is similar to that of the LMS algorithm [6]. Least Mean Square Algorithm suffers from the major drawback that it is sensitive to the scaling of its input signal $d(n)$. So, it is very difficult to choose a learning rate μ. The learning rate μ ensure the stability of algorithm. Normalized Least Mean Square Algorithm solves this problem by normalizing with the power of input signal.

Following are the equations of Normalized least Mean Square Algorithm [6]:

$w(n+1) = w(n) + \mu(n)x(n)e(n)$ .... (5)

where $\mu(n)$ is adaptive step size which is computed as

$\mu(n) = \frac{\alpha}{L * P(n)}$, 0 < α < 2

$P(n)$ is the estimated power of $x(n)$ at time n , L is order of filter , and α is the normalized step size. An exponential window is used to estimate the power of $x(n)$.

$P(n) = (1-\beta) * p(n-1) + \beta * x^2(n)$........ (6)

Filter output

$y(n) = e(n) * d(n) * w(n)$ .... (7)

Estimation Error or Error Signal

$e(n) = x(n) - y(n)$ .... (8)

Tap weight adaptation

$W(n) = w(n) + d(n)$.......... (9)

where β is a smoothing parameter , which is in terms of its equivalent (exponential) window length $(M \equiv \frac{1}{\beta})$

E. Recursive least square algorithm

In RLS algorithm the weight vector $W(n)$ updates continuously with each set of new data without solving matrix inversion (thesis). The least square algorithms required all the past samples of the input signal as well as the desired output at every iteration.RLS algorithm is a simple adaptive filter and time update version of Weiner filter. For non-stationary signals, this filter detects the time variations but in case of stationary signals, the convergence behavior of this filter is same as that the convergence behavior in Weiner filters. RLS filter has fast convergence rate and it is widely used in the application such as echo cancellation, channel equalization, speech enhancement and radar where the filter should do fast changes in signal.

F. Implementation of RLS algorithm

There are two main factors of the RLS implementation and the first is that although the matrix inversion is essential to derive the derivation of RLS algorithm but there is no matrix inversion calculations are required for its implementation, hence it reduces the computational complexity of the algorithm. Secondly, its current variables are not updated within the iteration using values from the previous iteration as it occurs in LMS algorithm [9]

Filter output

$y(n) = x(n) * w^T(n)$ .... (10)

Calculates the error signal *e* (*n*)

$e(n) = d(n) - y(n)$.......... (11)

3. Updates the filter coefficients as:

$W(n) = w(n) + e(n).k(n)$............ (12)

Where $W(n)$ is the filter coefficients vector and $k(n)$ is the gain vector

$k(n) = \frac{P(n).u(n)}{\lambda + u^T P(n).u(n)}$...................... (13)

Where λ is forgetting factor and $P(n)$ is inverse correlation matrix of input signal

$P(n+1) = \frac{1}{\lambda}$

$[P(n-1) - k(n)u^T(n)P(n)]$.............. (14)

k=1 is the time at which the RLS algorithm commences and l is a small positive constant which is very close to1 but smaller than 1.

III METHODOLOGY

In the current work, we will take a sample of audio signal which is suffered by wow and flutter. This signal act as input signal which is represented by $d(n)$. We will also take another signal $x(n)$ that is in the frequency range of 0.5 to 100 Hz and it act as a reference input. Advance Speech Adaptive Algorithm like Least mean Square Algorithm, Normalized least mean square algorithm etc will used to minimize the effect of Wow and Flutter from audio signal and hence as a result signal is recovered.

Input Signal Input signal $d(n)$ contains the audio signal which is affected by wow and flutter. Variation occurs in the speed of playback recorded signal which causes variation and distortion in pitch. These variations and distortions result in fluctuation of signal. Due to fluctuations low frequency irregularities and high frequency irregularities occurs which produces wow and flutter. These signals are taken from analog tape recorders and gramophones

.

Analog to digital converter Analog to digital converter is a device which converts analog signals into digital signals. Analog to Digital converter is used so that computer can understand and process the coming wow ant flutter signal. The accuracy of converter to convert the audio signal into digital one depends on the sampling rate. Higher the sampling rates mean higher accuracy of conversion. Thus computer digital signal can now be processed using any computer algorithm.

Advanced adaptive speech algorithm The digital signal thus obtained from above step is now fed as an input to Advanced Adaptive Speech algorithm. It minimizes the effect of wow and flutter in digital signal.

Digital to analog converter Digital to analog converter device converts the digital signal back to analog form. This signal gets relaxed from the effects of audio wobbles and hence can be heard accurately.

IV RESULTS AND DISCUSSIONS

The result section is divided into three subsections.

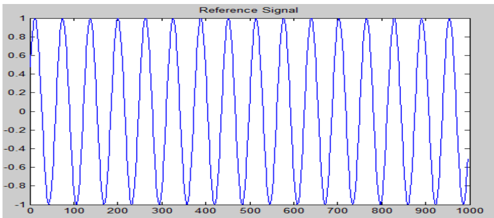
1. Generation of reference signal for wow and flutter.
2. Detection of wow and flutter in original input signal.
3. Correction of wow and flutter
   3.1 By using simple subtraction
   3.2 By using advanced adaptive speech algorithm
4. Comparison of results obtained from different algorithm on the basis of some parameters.

A. Generation of reference signal

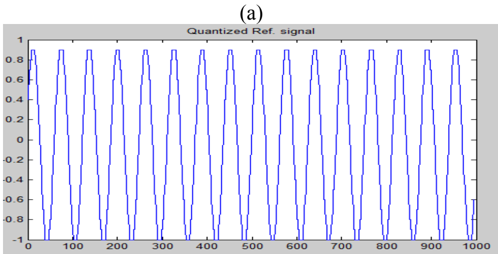
In noise cancellation usually a reference signal is generated to cancel out the the corrupted signal so noise free signal is obtained at the output. Similarly to remove of wow and flutter from audio signal we have taken the reference signal whose range lies between 0.5 to 100 Hz and this is also equal to the range of wow and flutter. After obtaining reference signal we quantized the above mentioned signal. So, we get the Quantized reference signal.

B. Generation of reference signal

To correct wow and flutter from audio signal advanced adaptive speech algorithms used two signal one is input signal containing wow and flutter and another signal is reference signal. So, first of all there is a need to generate reference signal. The reference signal should be in the range of wow and flutter. So, that algorithm can subtract the reference signal from the audio signal and this process will be repeated again and again until minimum error is not obtained at the output and hence the signal obtained at the output is corrected from wow and flutter.



(a)



(b)

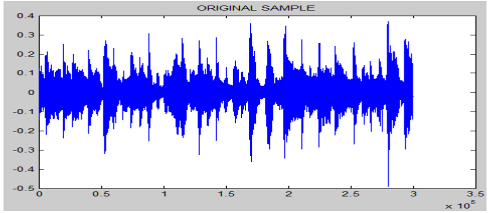
Fig 5.1: Waveform for the Generation of reference signal (a) Reference signal (b) Quantized reference signal.

C. Detection of wobble in input signal

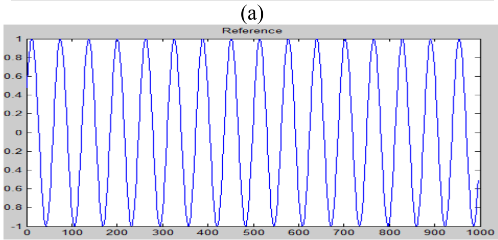
Detection of wobble in input signal is done on the basis of frequency of signal. The input signal is quantized and then compared with the quantized value of the reference signal generated in previous section.

D Detection of wobble in input signal (SIGNAL 1)

SIGNAL 1: For signal1 [5] following waveforms are obtained for length of signal is 751104 samples. Signal 1 contains a large amount of wobble in audio signal. Figure (d) shows the wobble irregularities in audio signal 1.



(a)



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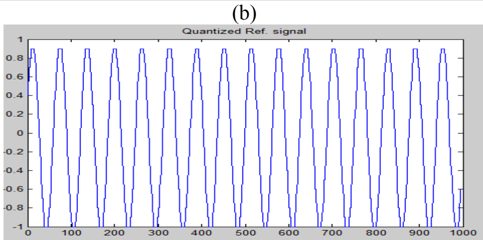

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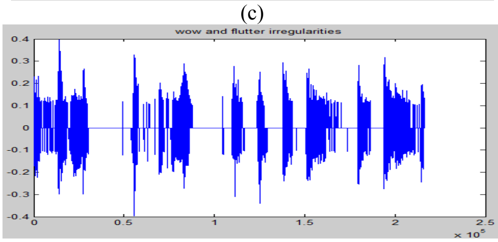


(d)

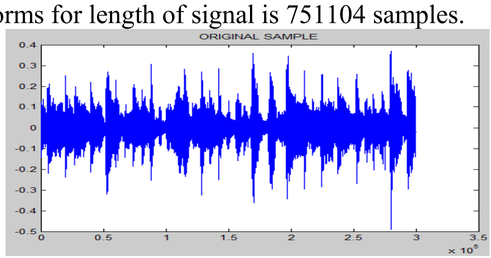
Fig. 5.2: Detection of wobble in input signal (a) (b) reference signal (c) quantized reference signal (d) Detected wobble irregularities

E. Correction of wobble in input signal by using two methods

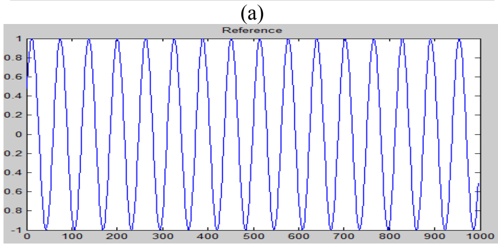
We have taken a signal which contains wobble. So, First of all we detected the presence of wobble in input signal and now the correction is done by the use of advanced adaptive speech algorithm.

F. Correction of wobble in input signal (SIGNAL 1)using direct subtraction method

SIGNAL 1: For signal 1 [5] a wave/mp3 file, the following waveforms for length of signal is 751104 samples.



(a)



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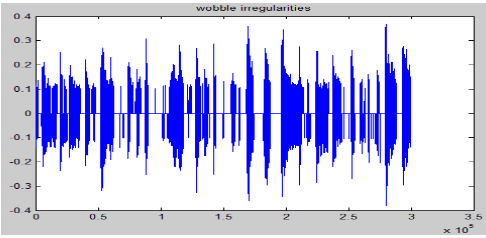

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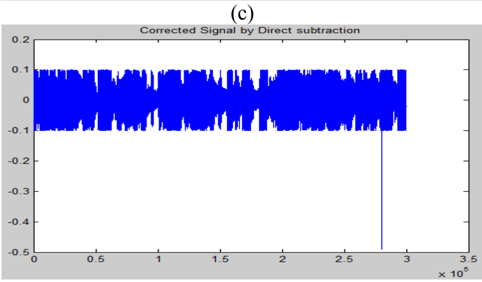

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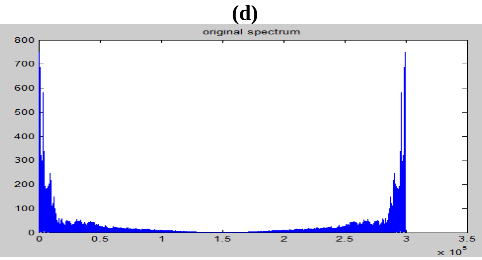

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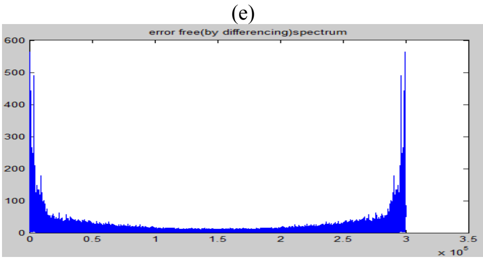

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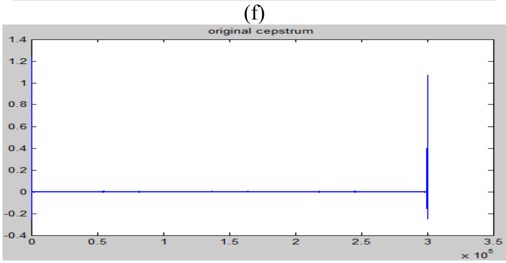
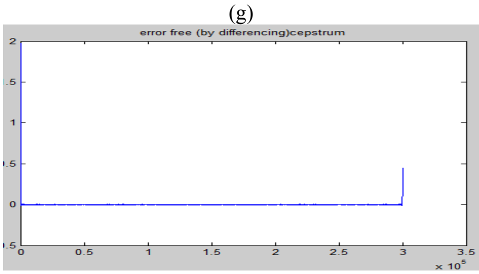


(g)

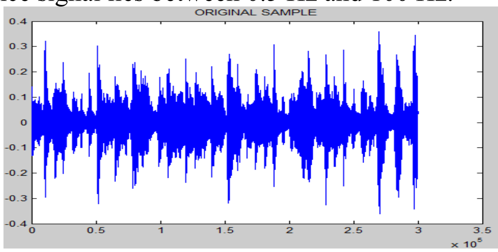


(h)

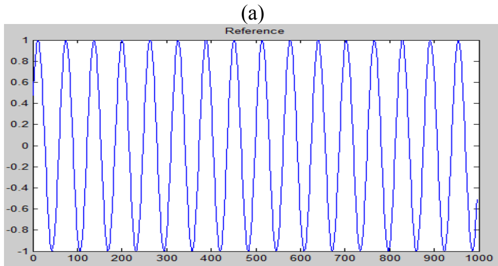
Fig. 5.5 Correction of wobble in input signal (SIGNAL 1)using Direct Subtraction Method (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by direct subtraction (e) Spectrum of Original Input signal (f) Spectrum of error free(by differencing) (g) Cepstrum of original Input signal (h) Cepstrum of error free(by differencing)

F. Correction of wobble in input signal (SIGNAL 2) using direct subtraction method:

SIGNAL 2: For signal 2 [5] following waveforms are obtained for length of signal is 751104 samples and a reference signal lies between 0.5 Hz and 100 Hz.



(a)



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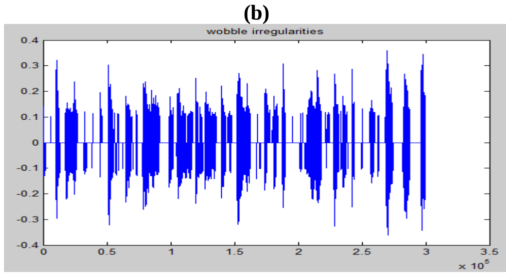

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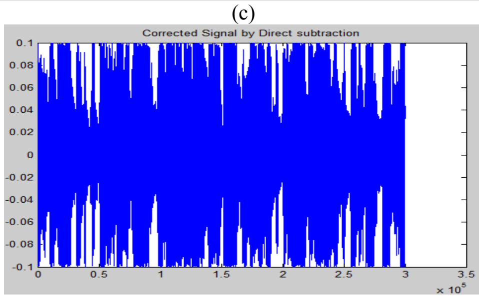

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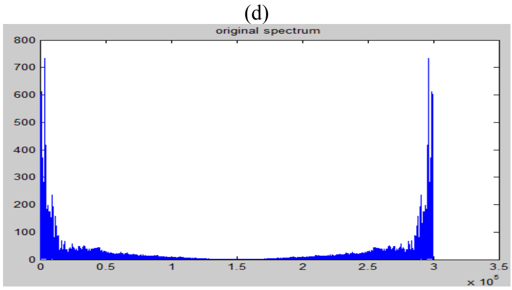

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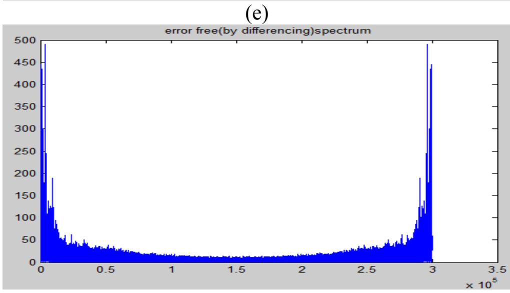
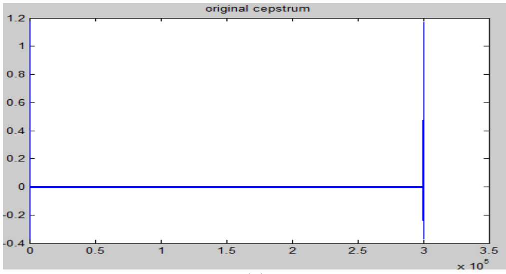
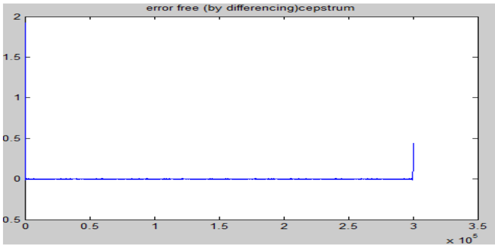


(f)



(g)

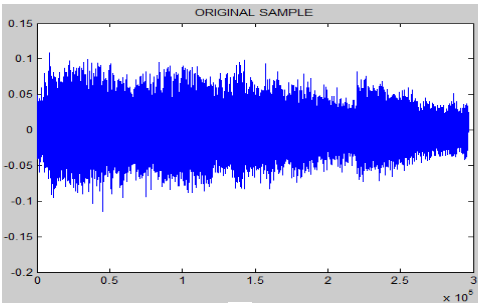


(h)

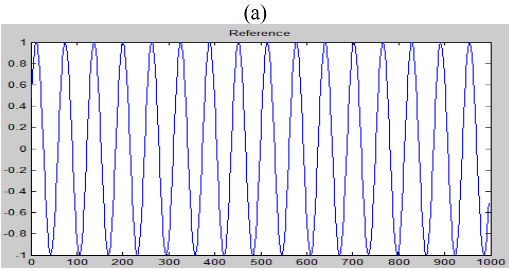
Fig. 5.6 Correction of wobble in input signal (SIGNAL 2)using Direct Subtraction Method (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by direct subtraction (e) Spectrum of Original Input signal (f) Spectrum of wobble (g) Spectrum of error free(by differencing) (h) Cepstrum of original Input signal (i) Cepstrum of wobble (j) Cepstrum of error free(by differencing)

G. Correction of wobble in input signal (SIGNAL 3) using direct subtraction method:

SIGNAL 3: For signal 3 [5] a wave/mp3 file, the following waveforms for length of signal is 751104 samples.



(a)



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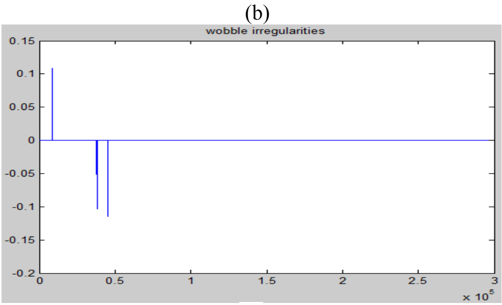

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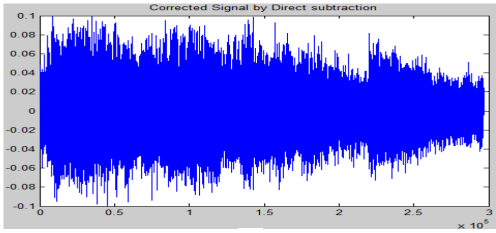

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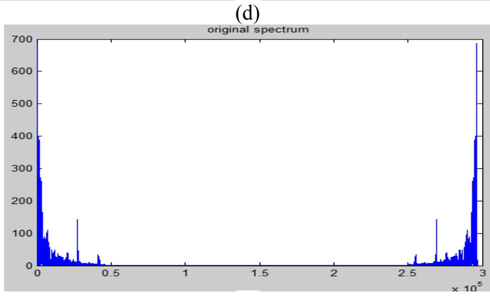

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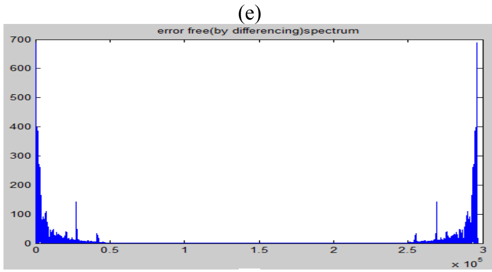


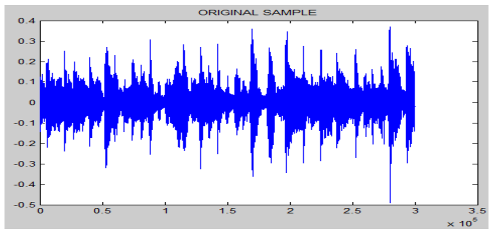
(f)

Fig. 5.7 Correction of wobble in input signal (SIGNAL 3)using Direct Subtraction Method (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by direct subtraction (e) Spectrum of Original Input signal (f) Spectrum of error free(by differencing)

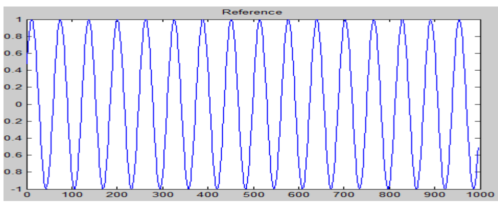
A. Correction of wobble in input signal (SIGNAL 1)using Least Mean Square Algorithm

Since using LMS, we have one parameter in hand to train the filter adaptively. It is learning parameter µ

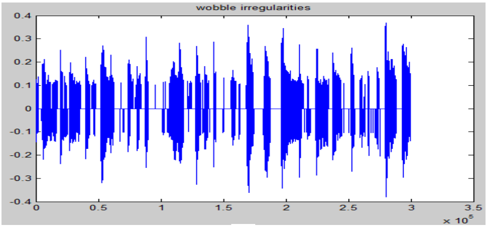
SIGNAL 1: For signal 1[5] following waveforms are obtained for **µ = 0.8** and a reference signal lies between 0.5 Hz to 100 Hz and length of the signal is 751104 samples.



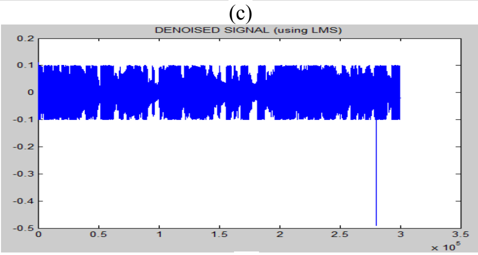
(a)



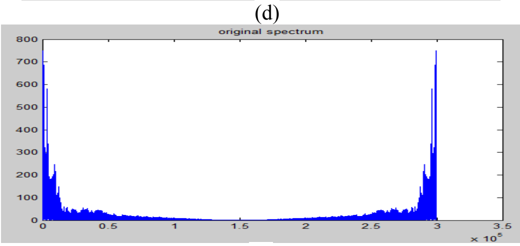
(b)



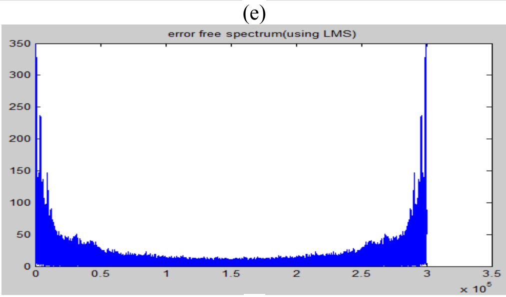
(c)



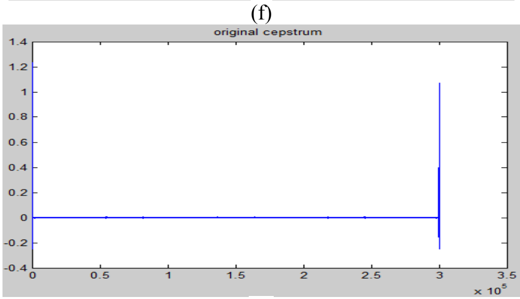
(d)



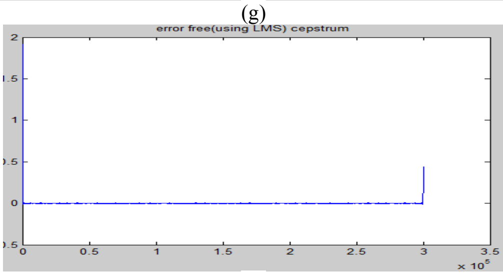
(e)



(f)



(g)



(h)

Fig 5.9 Correction of wobble in input signal (SIGNAL 1)using Least Mean Square Algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using LMS algorithm (e) Spectrum of Original Input signal (f) Spectrum of wobble (g) Spectrum of error free(by using LMS) (h) Cepstrum of Original Input signal (i) Cepstrum of wobble (j) Cepstrum of error free(by using LMS)

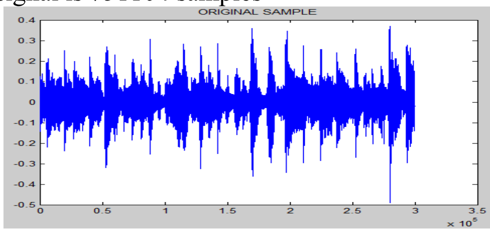
B. Correction of wobble in input signal (SIGNAL 1) using Normalized Least Mean Square Algorithm

Since using NLMS, we have following parameters in hand to train the filter adaptively:

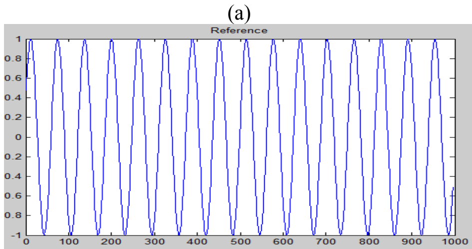
1. Smoothing parameter
2. Normalized step size
3. Filter order

So, the above three parameters can be varied in order to minimize the effects of wobble.

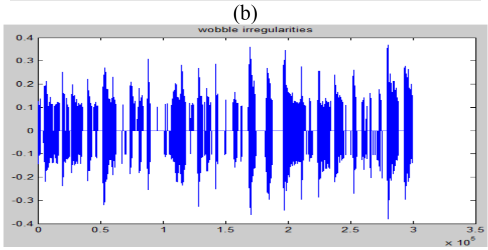
SIGNAL 1: For signal 1[5], a wave/mp3 file, the following waveforms are obtained for smoothing parameter = 4, Normalized step size = 1 and Filter order = 20 and a reference signal lies between 0.5 Hz to 100 Hz and length of the signal is 751104 samples



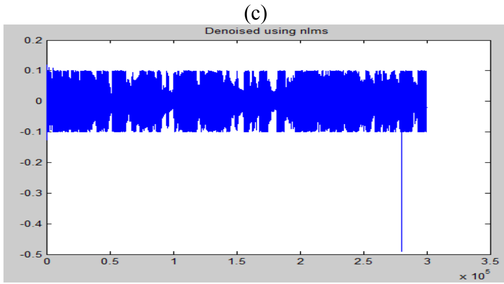
(a)



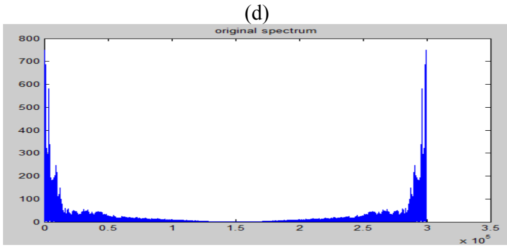
(b)



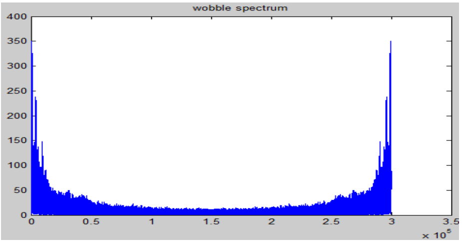
(c)



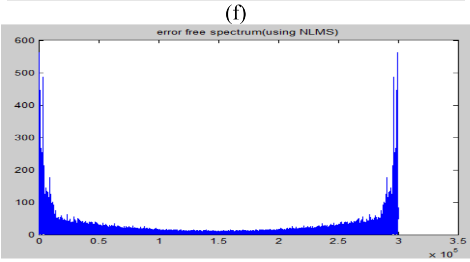
(d)



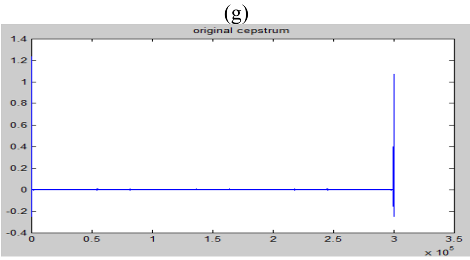
(e)



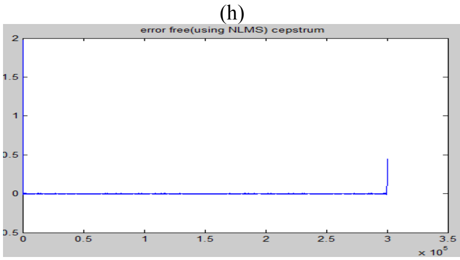
(f)



(g)



(h)

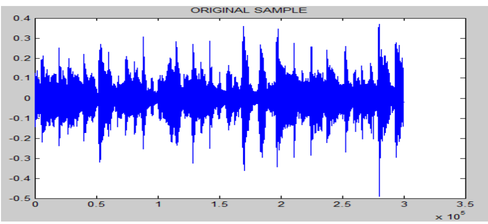


(j)

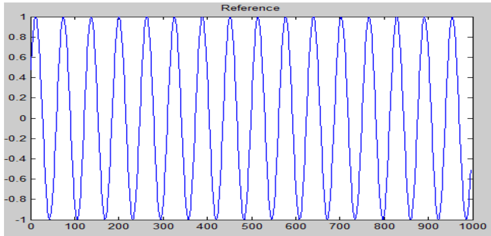
Fig. 5.10 Correction of wobble in input signal (SIGNAL 1)using Normalized Least Mean Square Algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using NLMS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using NLMS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using NLMS)

C. Correction of wobble in input signal (SIGNAL 1) using Recursive Least Square Algorithm

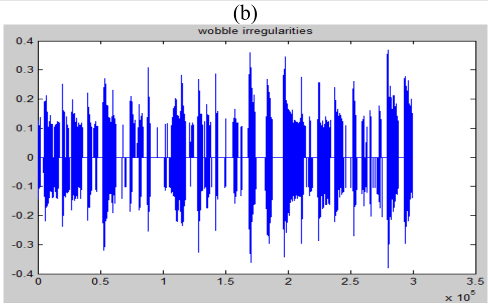
SIGNAL 1: For signal 1[5], a wave/mp3 file, the following waveforms are obtained for frequency of signal = 44100, forgetting factor = 0.99 and a reference signal lies between 0.5 Hz to 100 Hz and length of the signal is 751104 samples.



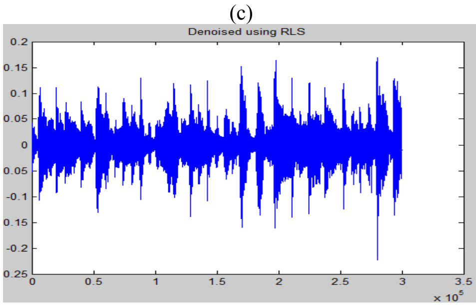
(a)



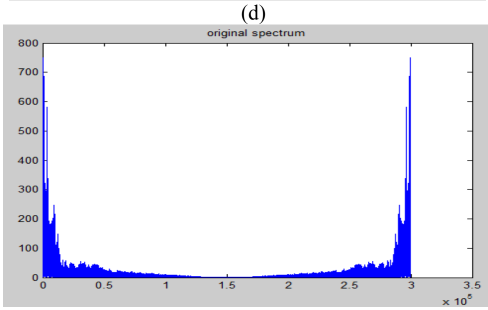
(b)



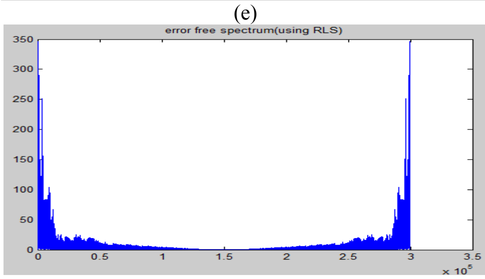
(c)



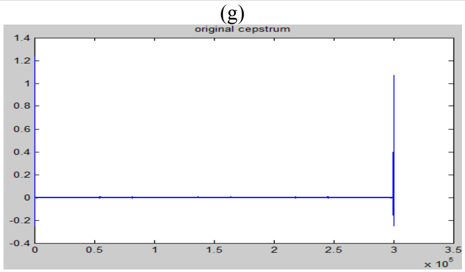
(d)



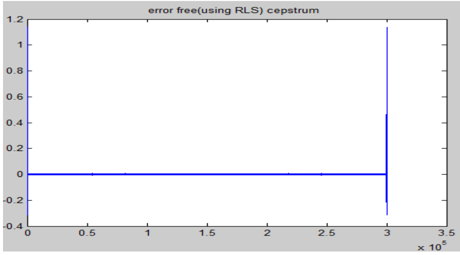
(e)



(g)



(h)

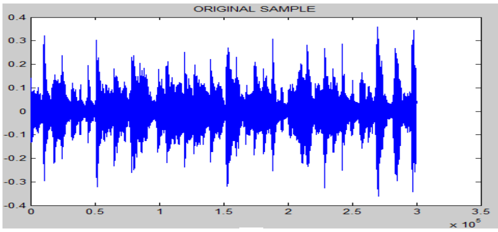


(l)

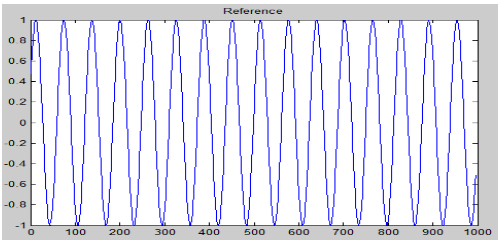
Fig. 5.11 Correction of wobble in input signal (SIGNAL 1) using Recursive Least Square Algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using RLS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using RLS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using RLS)

D. Correction of wobble in input signal (SIGNAL 2) using LMS algorithm

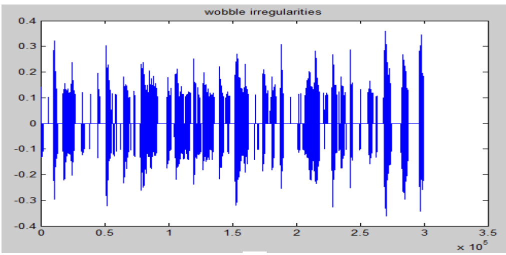
SIGNAL 2: For signal 2 [5] the following waveforms are obtained for μ = 0.8 and a reference signal lies between 0.5 Hz and 100 Hz and length of the signal is 751104 samples.



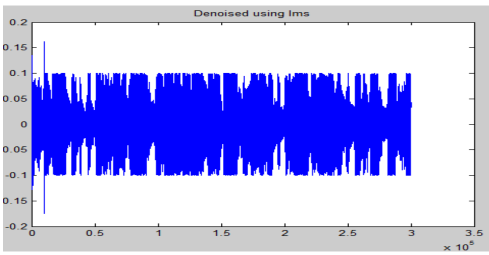
(a)



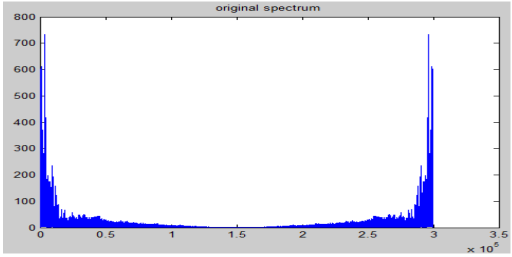
(b)



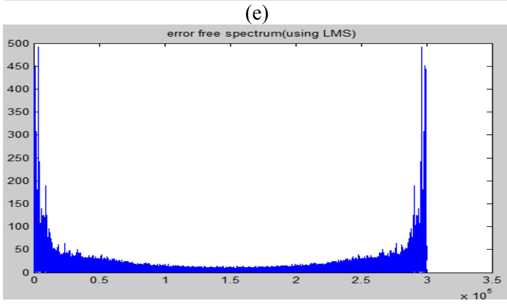
(c)



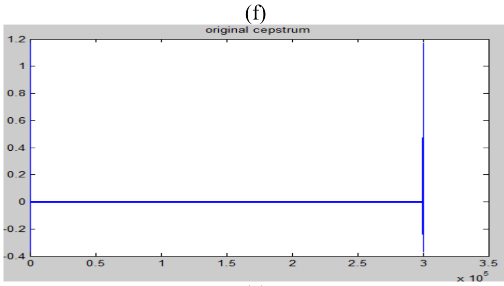
(d)



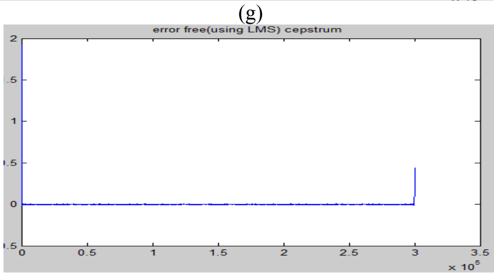
(e)



(f)



(g)

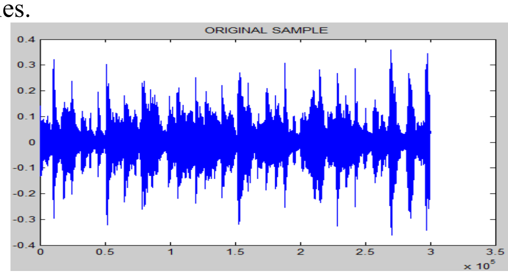


(h)

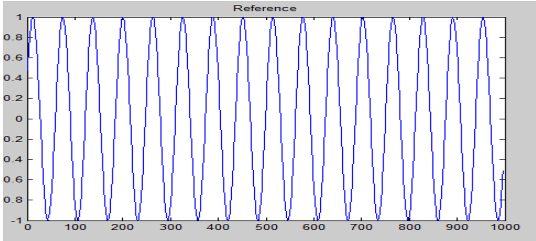
Fig 5.12 Correction of wobble in input signal (SIGNAL 2) using LMS algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using LMS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using LMS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using LMS)

E. Correction of wobble in input signal (SIGNAL 2) using NLMS algorithm

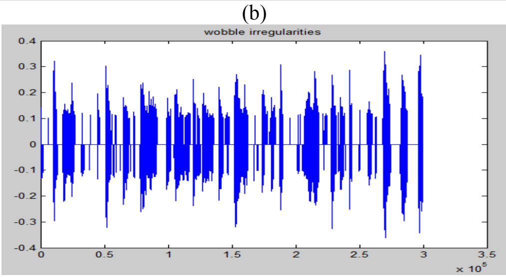
SIGNAL 3:For signal 3[5] following waveforms are obtained for smoothing parameter =4 Normalized step size = 1 and Filter order = 20 and a reference signal lies between 0.5 Hz to 100 Hz and length of the signal is 751104 samples.



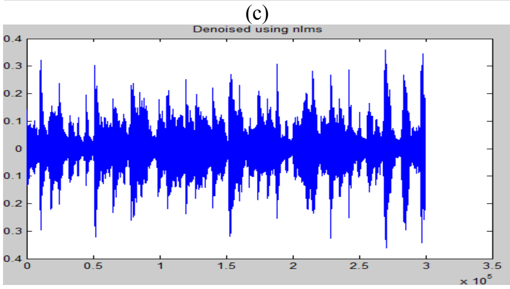
(a)



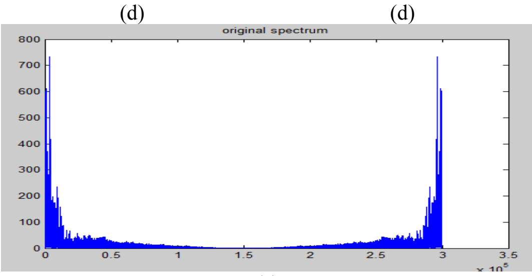
(b)



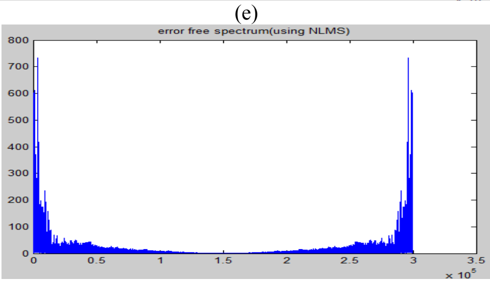
(c)



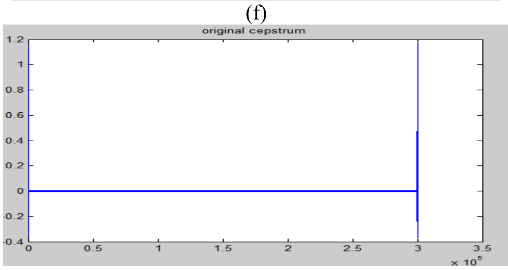
(d) (d)



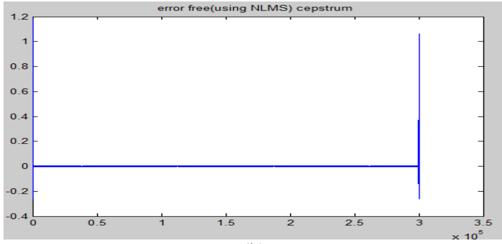
(e)



(f)



(g)

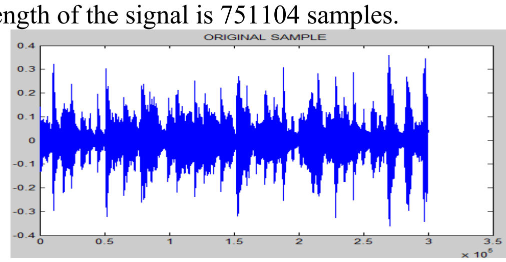


(h)

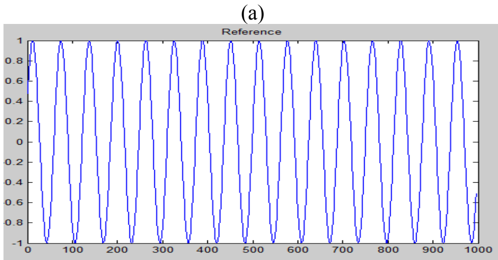
Fig. 5.13 Correction of wobble in input signal (SIGNAL 2) using NLMS algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using NLMS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using NLMS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using NLMS)

F. Correction of wobble in input signal 2 using RLS algorithm

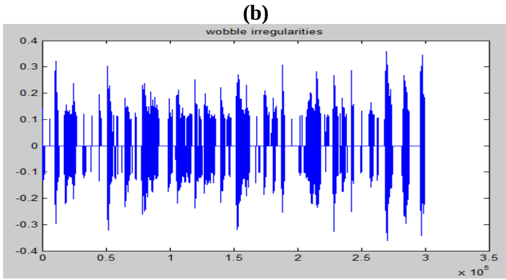
SIGNAL 2: For signal 3[5] following waveforms are obtained for frequency of signal = 44100, forgetting factor = 0.99 and a reference signal lies between 0.5 Hz to 100 Hz and length of the signal is 751104 samples.



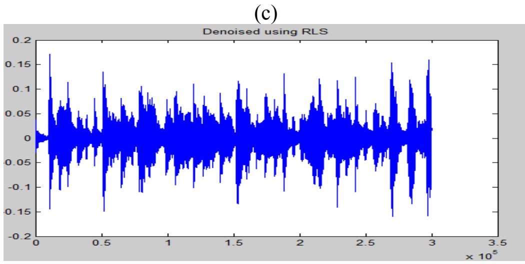
(a)



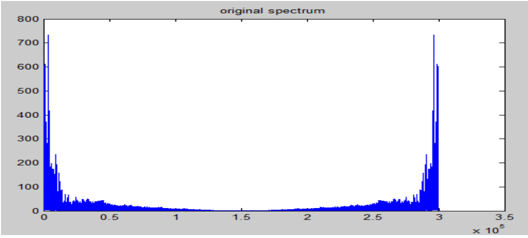
(b)



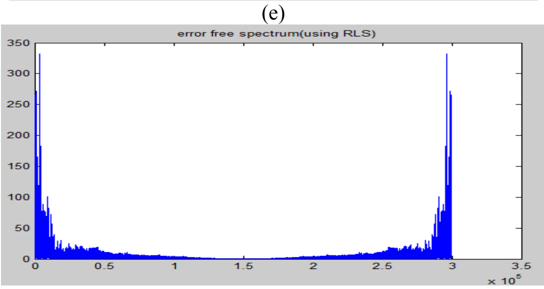
(c)



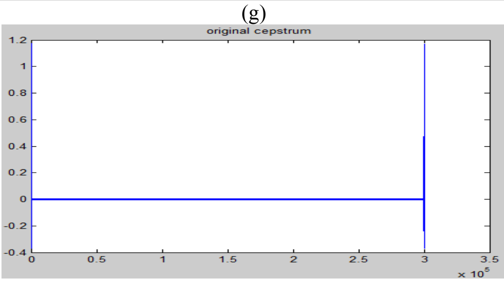
(d)



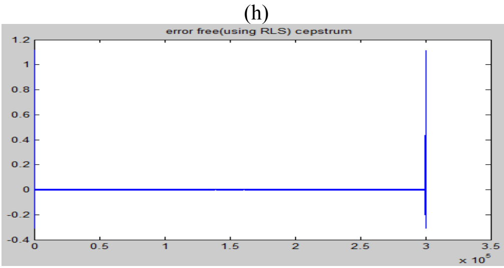
(e)



(g)



(h)

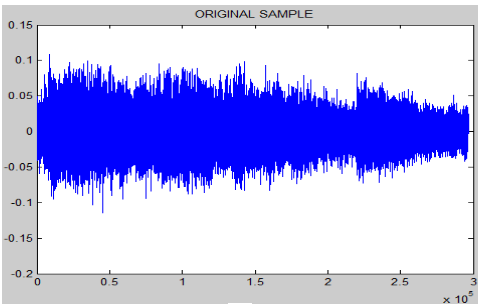


(j)

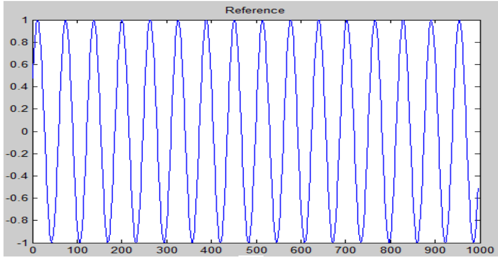
Fig. 5.14 Correction of wobble in input signal 2 using RLS algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using RLS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using RLS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using RLS)

G. Correction of wobble in input signal (SIGNAL 3) using LMS algorithm

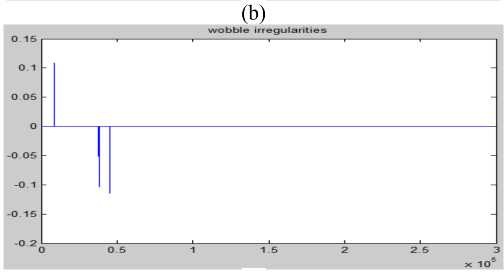
SIGNAL 3: For signal 3[5] following waveforms are obtained for $\mu = 0.8$ and a reference signal lies between 0.5 Hz and 100 Hz and length of the signal is 751104 samples.



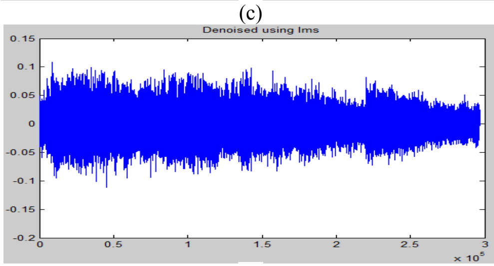
(a)



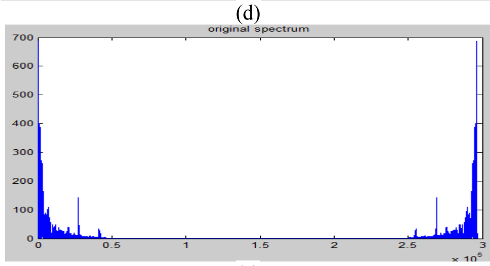
(b)



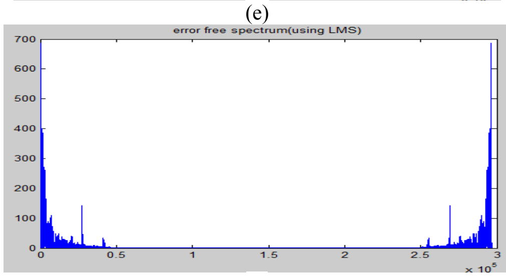
(c)



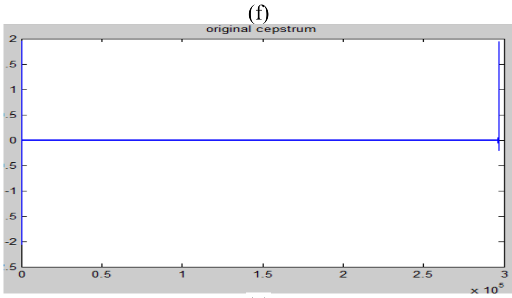
(d)



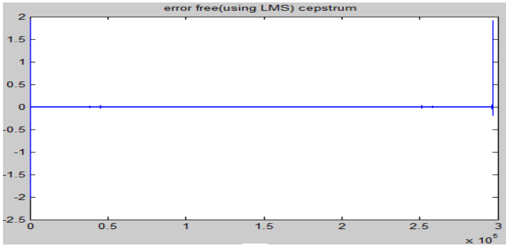
(e)



(f)



(g)

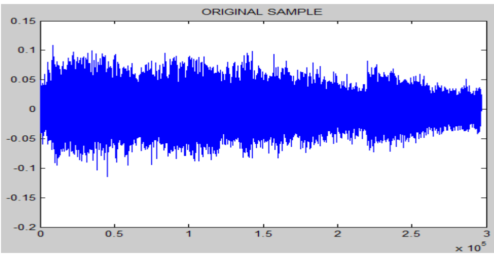


(h)

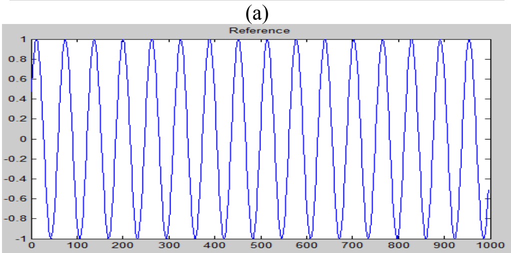
Fig 5.15 Correction of wobble in input signal (SIGNAL 3) using LMS algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using LMS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using LMS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using LMS)

H. Correction of wobble in input signal 3 using NLMS algorithm

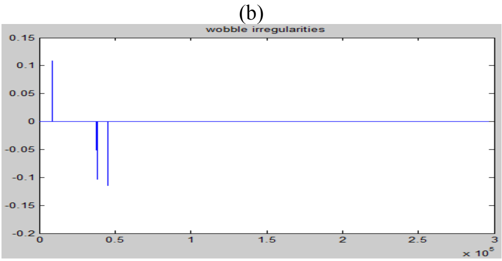
SIGNAL 3: For signal 3[5] following waveforms are obtained for smoothing parameter = 4, Normalized step size = 1 and Filter order = 20 and a reference signal lies between 0.5 Hz to 100 Hz and length of the signal is 751104 samples



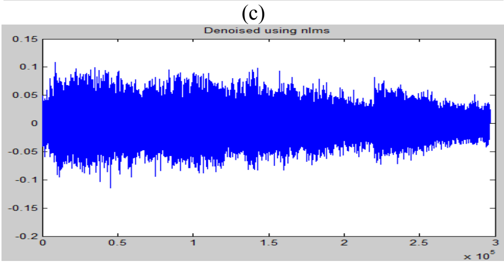
(a)



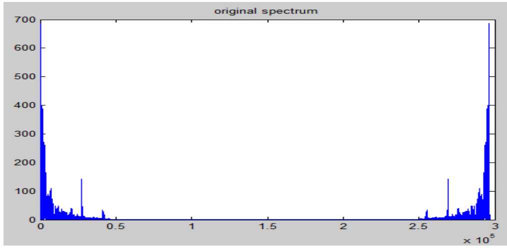
(b)



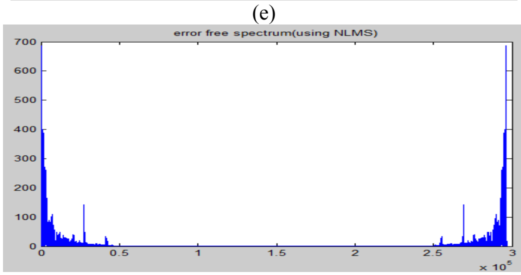
(c)



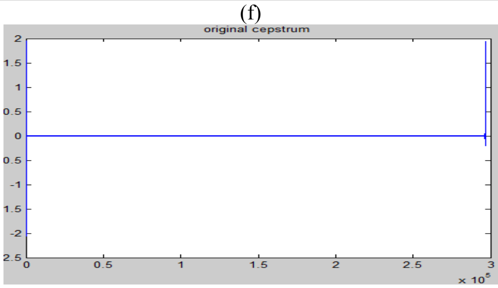
(d)



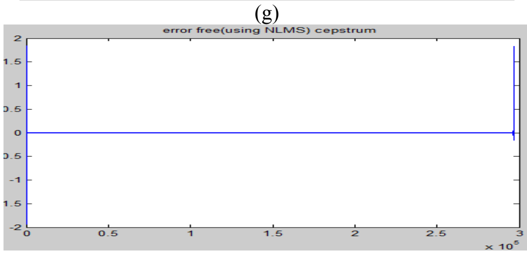
(e)



(f)



(g)

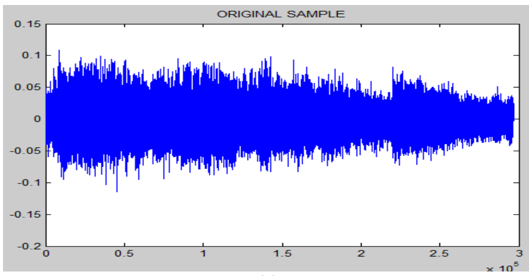


(h)

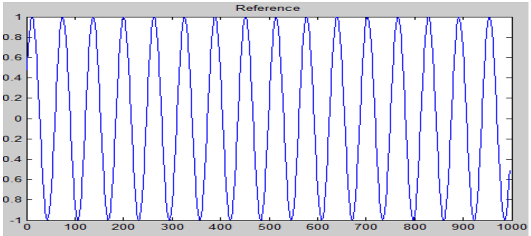
Fig 5.16 Correction of wobble in input signal 3 using NLMS algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using NLMS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using NLMS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using NLMS)

I. Correction of wobble in input signal 3 using RLS algorithm

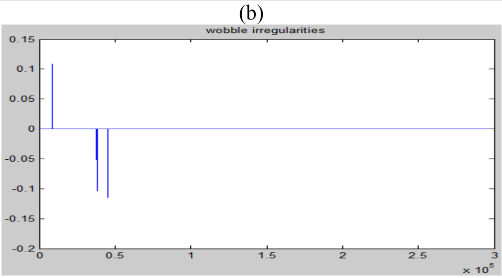
SIGNAL 3: For signal 3[5], a wave/mp3 file, the following waveforms are obtained for frequency of signal = 44100, forgetting factor = 0.99 and a reference signal lies between 0.5 Hz to 100 Hz and length of the signal is 751104 samples.



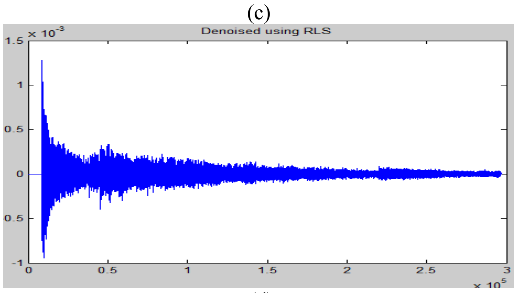
(a)



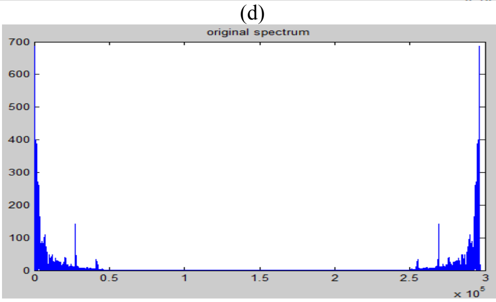
(b)



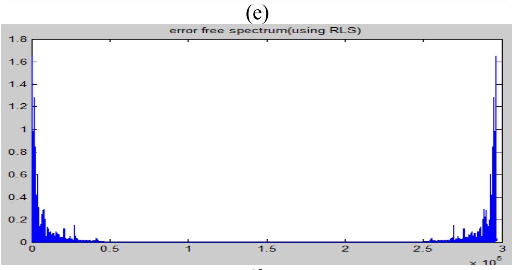
(c)



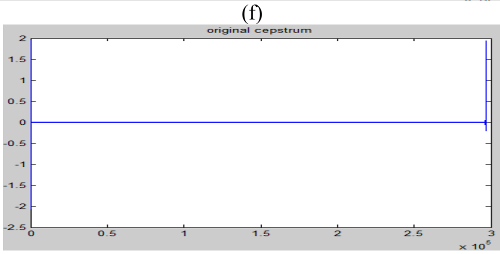
(d)



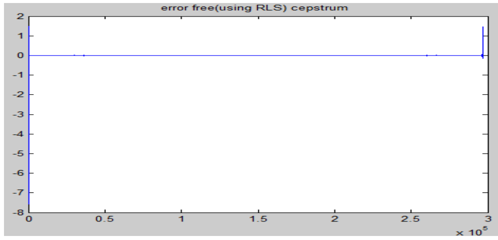
(e)



(f)



(g)



(h)

Fig 5.17 Correction of wobble in input signal 3 using RLS algorithm (a) Original Input signal containing wow and flutter (b) Reference signal (c) Wobble irregularities present in Input signal (d) Recovered corrected output signal by using RLS algorithm (e) Spectrum of Original Input signal (f) Spectrum of error free(by using RLS) (g) Cepstrum of Original Input signal (h) Cepstrum of error free(by using RLS)

J. Comparison on the basis of results obtained by different algorithm

Comparison of different algorithm has done on the basis of Mean Square error (MSE), Elapsed time (ET) and short time energy.

Mean square error

Mean square error is important factor to calculate the performance of algorithm [10]. It is also useful to know the concepts of bias, precision and accuracy in statistical estimation. In above tables we have calculated the mean square error between original signal and recoded signal and also calculate the mean square error between original signal and signal obtained at the output of advanced adaptive speech algorithms. The ideal value of mean square error is zero but practically it is not possible. MSE zero means algorithm predicts accurate observations.

Formula to calculate mean square error

If $\hat{Y}_i$ is a vector of n predictions, and Y_i is the vector of the true values, then the MSE of the predictor can be calculated as :

$$MSE = \frac{1}{n}\sum_{i=1}^{n}(\hat{Y}_i - Y_i)^2$$

Elapsed Time

The time spent to complete a program is known as Elapsed time. Elapsed time can be calculated in matlab by using following command.

Short time energy

Whenever the energy level of input signal suddenly increased then there is a need to measure this increment in energy level [11]. This is known as short time energy. To calculate the short time energy feature the input signal is divided into number of windows and after then calculates the windowing function for each window w. It is calculated by the following formula:

$$S_e = \sum_{i=-\infty}^{\infty} x(i)^2 . h(w - i)$$

Where, $x(i)$ is the input signal and $h(w)$ is the impulse response.

COMPARISON TABLE FOR SIGNAL 1				
S.N0.	**ALGORITHM**	**MSE**	**ET (in sec)**	**STE**
1.	Differencing	0.0043	8.095290	0.1487
2.	LMS	0.0040	12.883316	0.1071
3.	NLMS	0.0283	146.329657	0.0205
4.	RLS	0.0033	1083.703431	0.0141

COMPARISON TABLE FOR SIGNAL 2				
S.N0.	**ALGORITHM**	**MSE**	**ET (in sec)**	**STE**
1.	Differencing	0.0040	7.980041	0.1381
2.	LMS	0.0038	12.904133 s	0.0532
3.	NLMS	0.0140	150.422091	0.0200
4.	RLS	0.0031	815.981749	0.0150

COMPARISON TABLE FOR SIGNAL 3				
S.N0.	**ALGORITHM**	**MSE**	**ET (in sec)**	**STE**
1.	Differencing	0.0013	7.938536	0.0652
2.	LMS	6.422	13.583130	0.0550
3.	NLMS	6.422×10^{-4}	148.407144	0.0543
4.	RLS	6.48×10^{-4}	769.242572	4.92×10^{-5}

COMPARISON TABLE FOR SIGNAL 4				
S.N0.	**ALGORITHM**	**MSE**	**ET (in sec)**	**STE**
1.	Differencing	0.017	5.869024	0.0823
2.	LMS	8.29×10^{-4}	10.323959	0.0719
3.	NLMS	8.29×10^{-4}	67.556918	0.0695
4.	RLS	8.24×10^{-4}	349.637503	9.23×10^{-5}

V. CONCLUSIONS AND FUTURE SCOPE

The From this research work it is clear that wow and flutter can be removed by the use of advanced adaptive speech algorithm. Among LMS, NLMS and RLS algorithm, Recursive Least Square Algorithm gives better results on the basis of MSE and STE. But from the result of elapsed time it is clear that RLS algorithm takes more time to give results as comparison to LMS and NLMS algorithm. For future scope other advanced adaptive speech algorithm or Hybrid of these algorithm or a new algorithm can be used which will give better results than RLS and take less time to give results as comparison to RLS algorithm.

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