COMPARISON OF SPEECH ENHANCEMENT USING KALMAN FILTER vs WEINER FILTER

In this project, we investigated the enhancement of speech by applying kalman filter. Noise removal is very important in many applications like telephone conversation, speech recognition, etc. The corruption of speech due to presence of additive background noise causes severe difficulties in various communication environments. The presence of this background noise in speech significantly reduces the intelligibility of speech. Degradation of speech severely affects the ability of person, whether impaired or normal hearing, to understand what the speaker is saying.
Here we used the Kalman filter which is an efficient recursive filter that estimates the internal state of a linear dynamic system from a series of noisy measurements. We have also Compared the results of kalman filter with spectral subtraction, wiener filter and found kalman filter has shown good improvement in SNR values.

INTRODUCTION:

The Kalman filter is a mathematical power tool that is playing an increasingly important role in computer graphics as we include sensing of the real world in our systems. The good news is you don't have to be a mathematical genius to understand and effective use Kalman filters.

Speech enhancement has been a hot research area in recent years with the fast development of multimedia communications and other application. The presence of background noise in speech significantly reduces the intelligibility of speech. Noise reduction or speech enhancement algorithms are used to suppress such background noise and improve the perceptual quality and intelligibility of speech. Removing various types of noise is difficult due to the random nature of the noise and the inherent complexities of the speech. Noise reduction techniques usually have a trade-off between the amount of noise removal and speech distortions introduced due to processing of the speech signal. Several techniques have been proposed for this purpose in the area of speech Enhancement, like spectral subtraction approach, wiener filter, Kalman filter, weighted filter. The performance of these techniques depends on the quality and intelligibility of the processed speech signal. The improvement in the speech signal to noise ratio is the target of most techniques.

COMPONENTS REQUIRED:

- Matlab.
- Audio Signal.

BLOCK DIAGRAM:

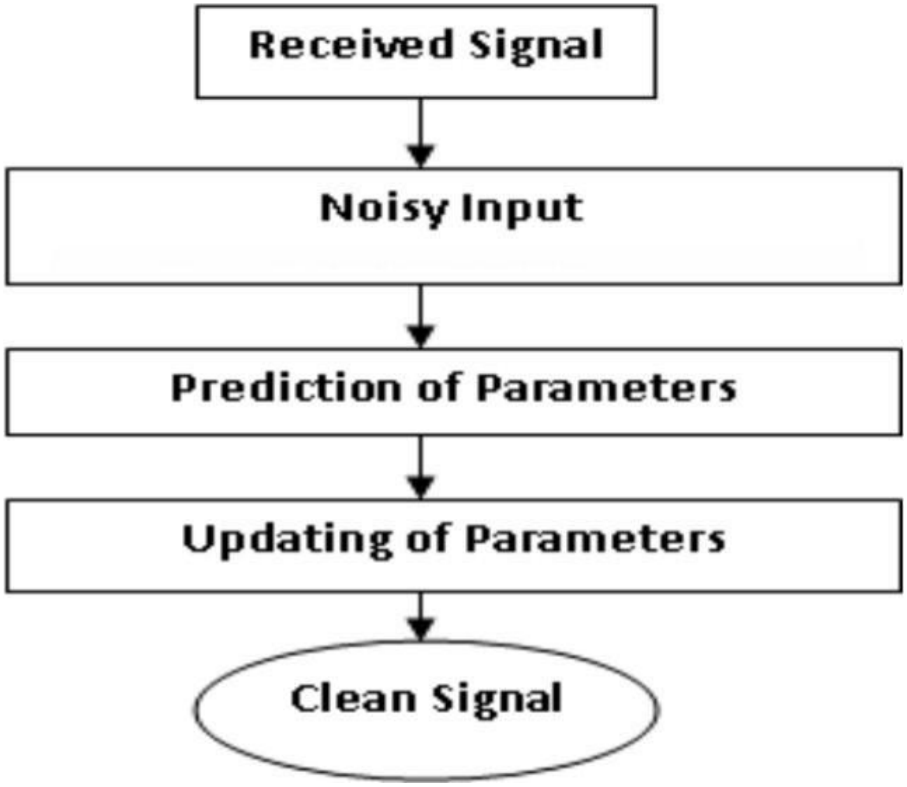


Fig: Flow chart of kalman filter

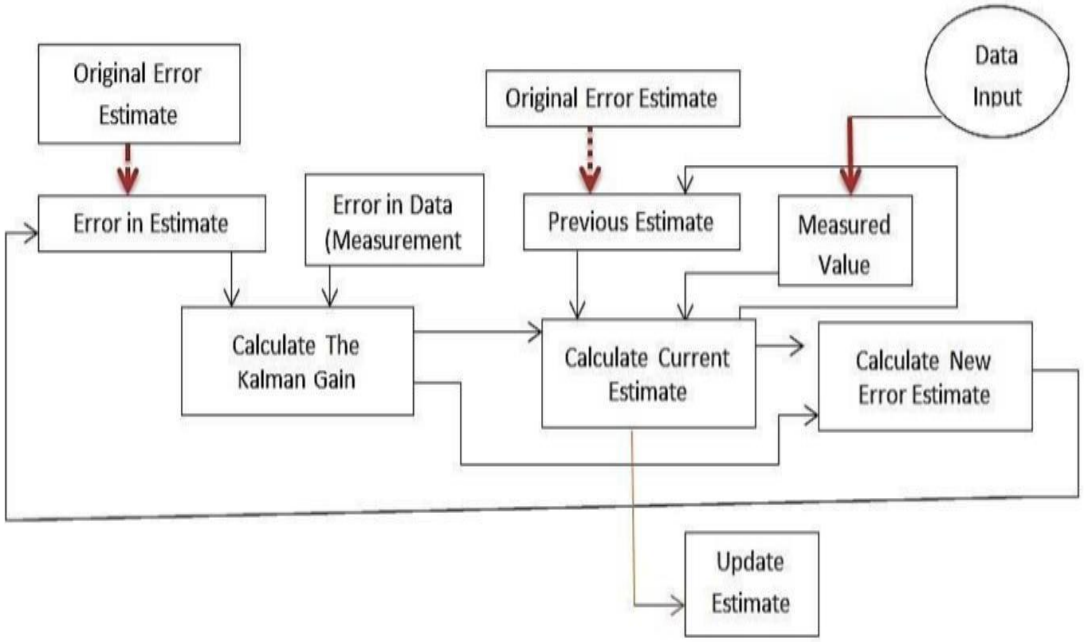


Fig: Mechanism of Kalman filters in speech enhancement

WORKING:

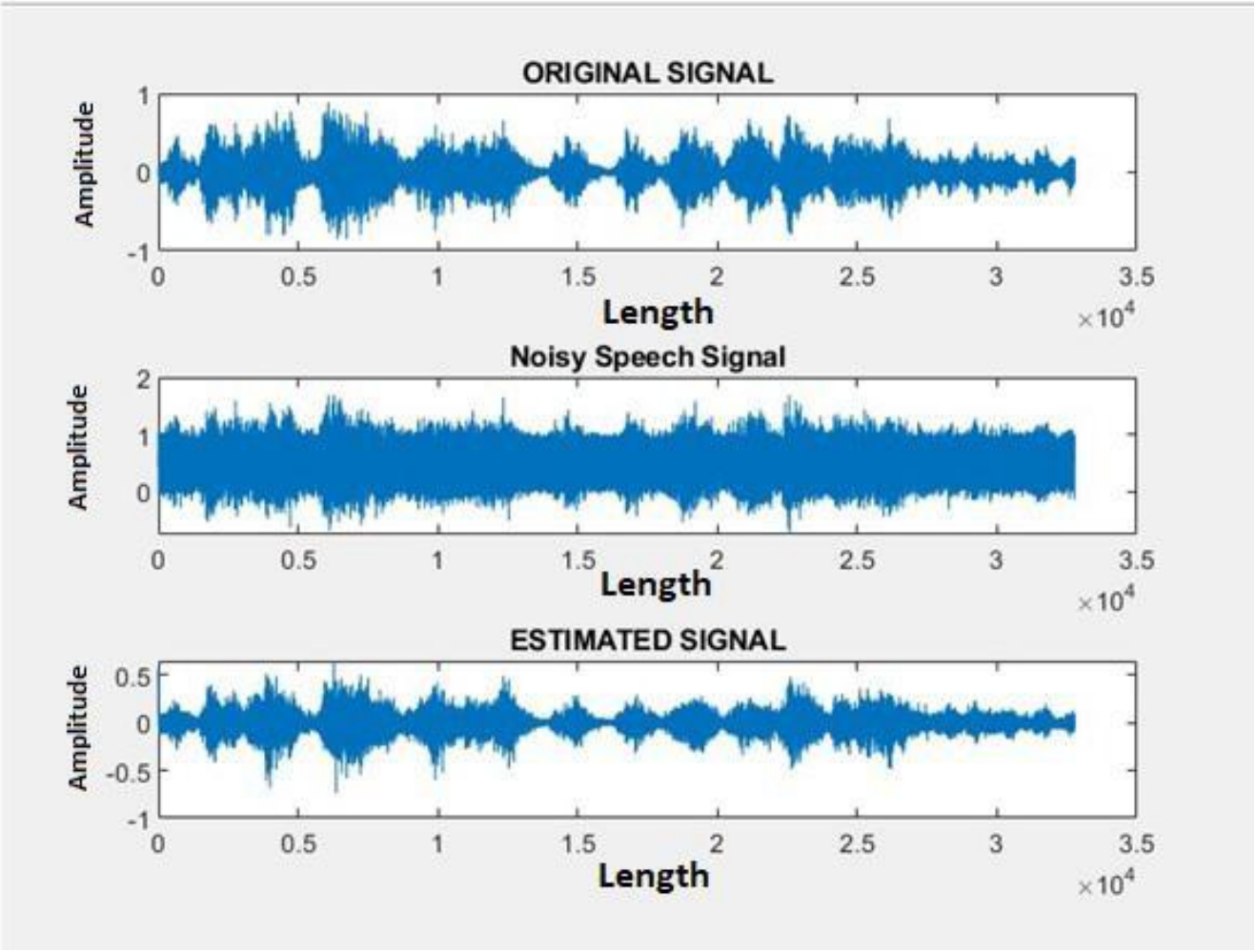
The main aim of the work is speech enhancement using Kalman filter. Initially, we have taken the audio input signal from matlab sample audios and added a random noise to it which produced appropriate output. The clean speech used in this work is a sentence pronounced by a group of people laughing together (name of the file being "Laughter-16-8-mono-4secs.wav" from matlab database). We have also taken a random noise with SNR 10dB. This is used as the noisy speech which is given as the input to the Kalman as the data observed. As speech is not stationary for a long time we took small frames of speech by windowing. Here in this work, we observed the algorithm by taking different windowing techniques, Rectangular and Hamming. We took each frame length to be 240 samples. Now the segmented noisy speech is saved as a matrix where each row consists of the value of each window, where each window is of 240 samples looping and taking one window at a time. We calculated the LPC coefficients of the original noisy speech signal and calculate the Kalman gain for each loop for updation of the next state. Looping is done as the past samples have an influence over the future samples. Finally, after iterative process, the SNR of the output of the Kalman filter is calculated and compared with different techniques.

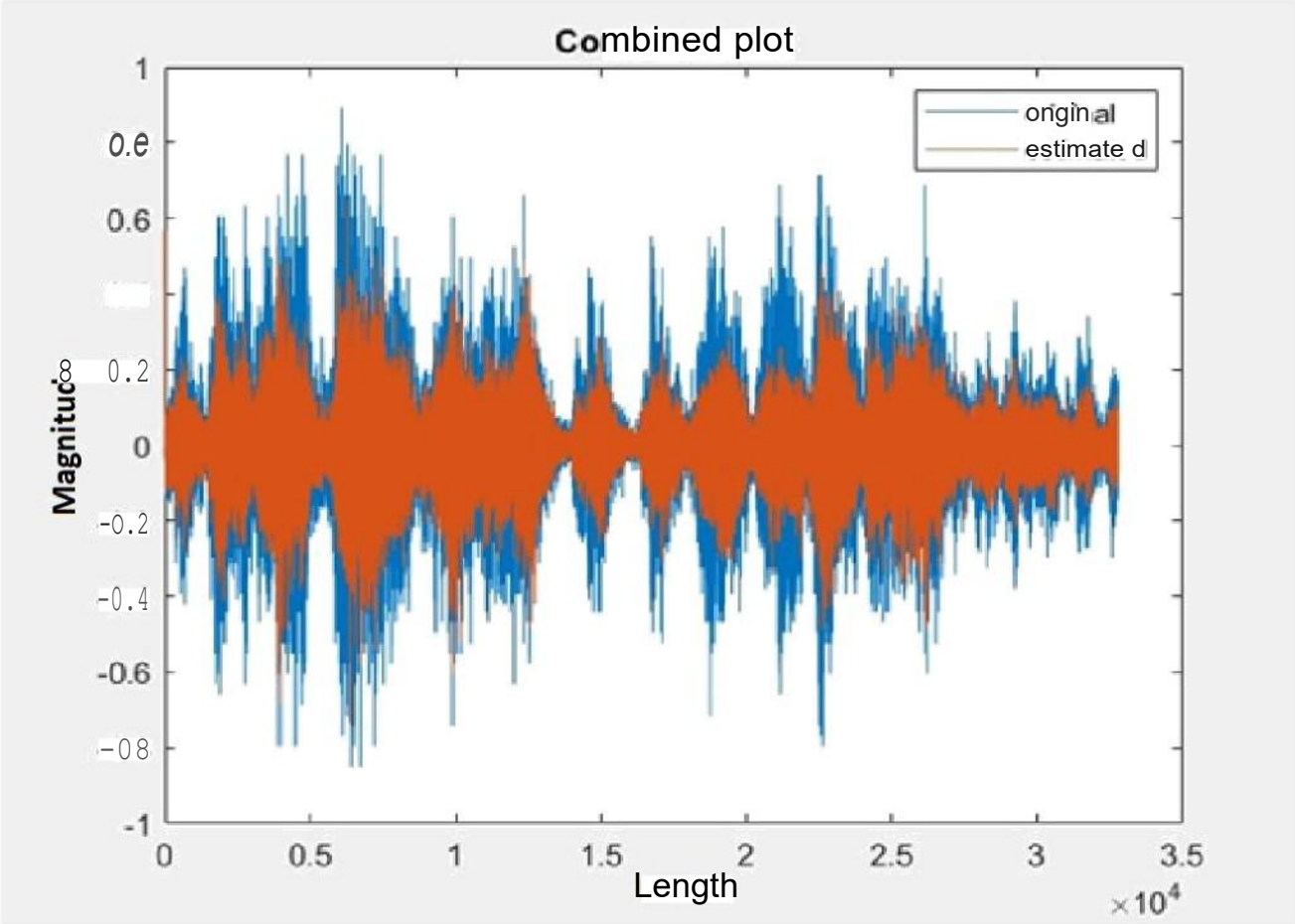
Steps:

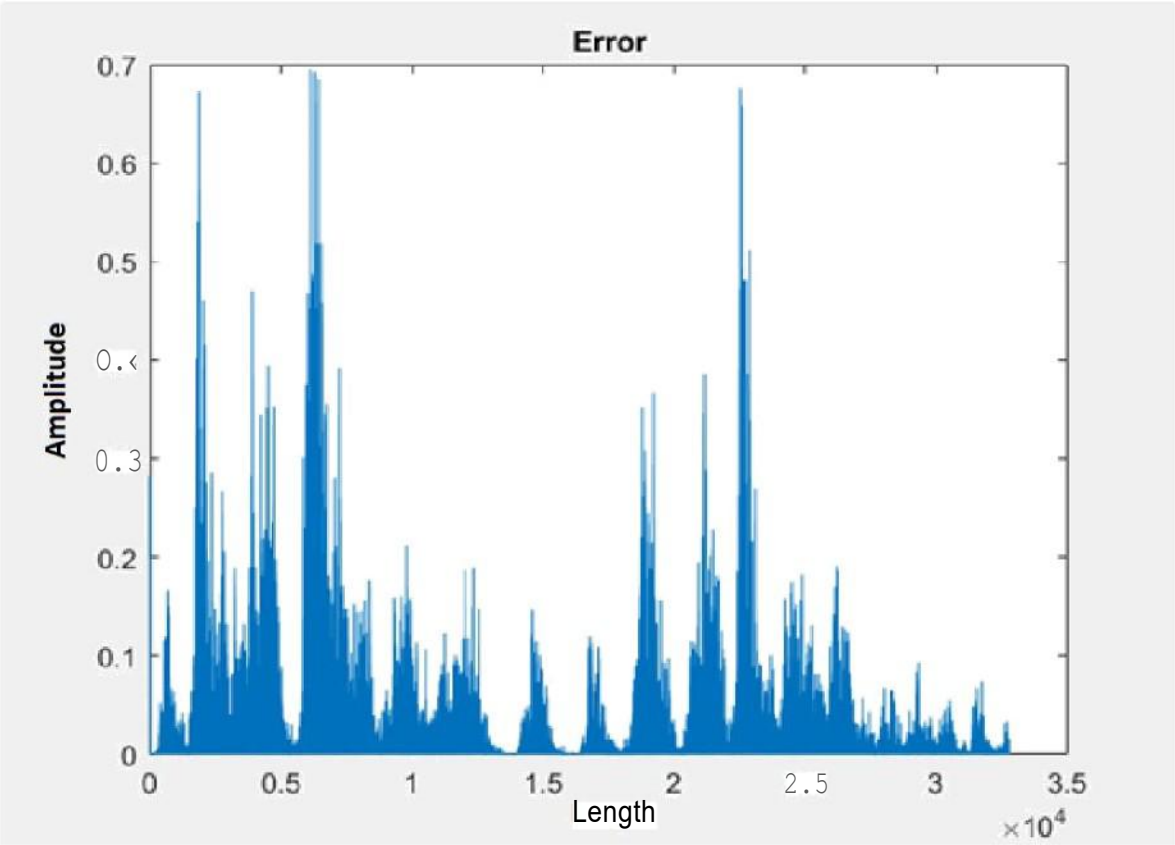
1. First we have taken the audio input signal from MATLAB database.
2. Then, we have given an input of signal containing random Gaussian noise.
3. An instruction to play the noisy speech with 0 SNR.
4. An instruction for noisy speech eradicating random noise.
5. Then we can calculate the following data:

- Length of the input signal
- Initialization of standard transition matrix
- Transition matrix
- Priori or posterior covariance matrix
- Kalman gain
- Kalman coefficient for yy.
- Desired signal
- Predicted state error
- Estimated error sequence
- Process noise covariance
- Measurement noise covariance
- Output of the signal

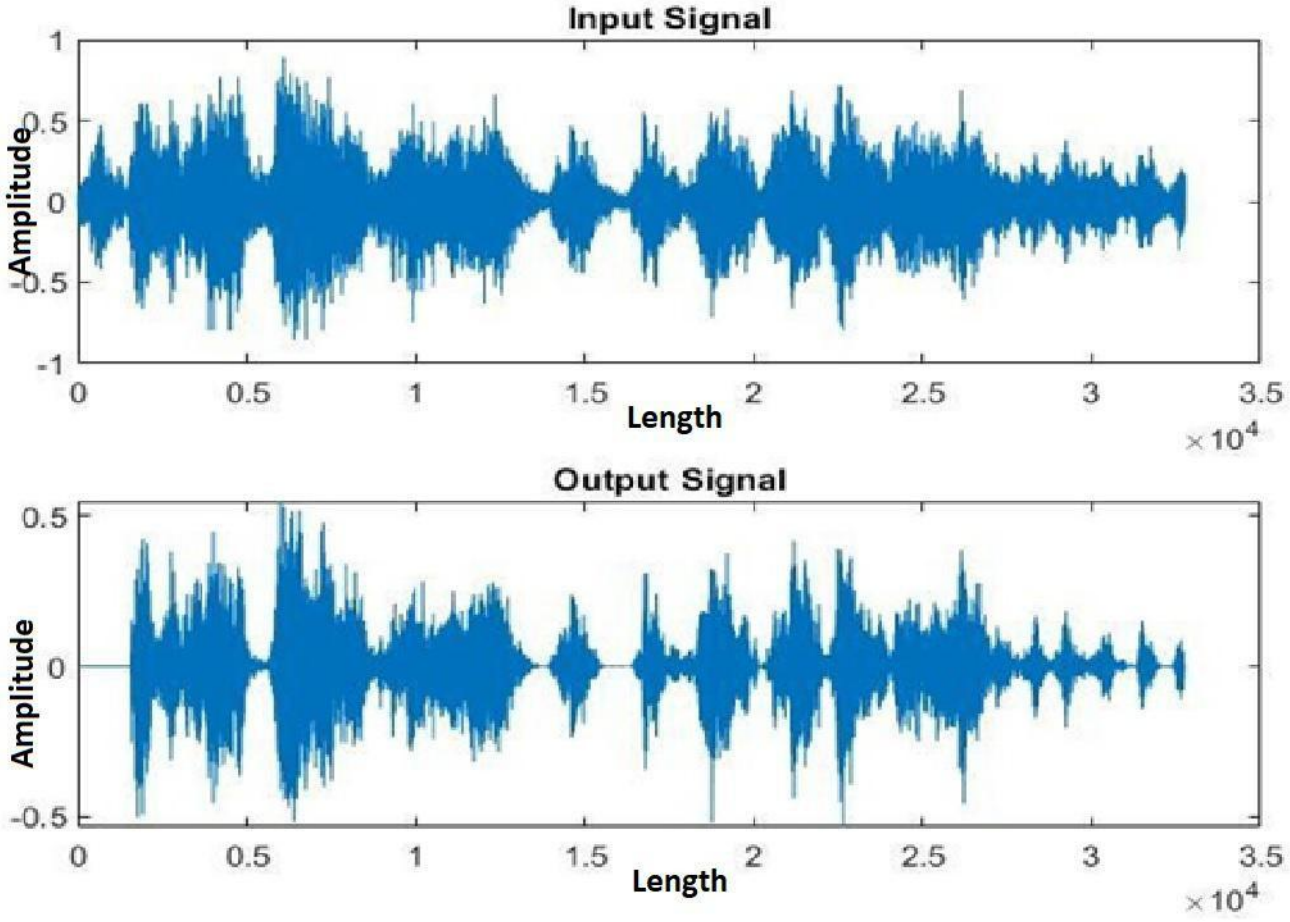
OUTPUT OF KALMAN FILTER:

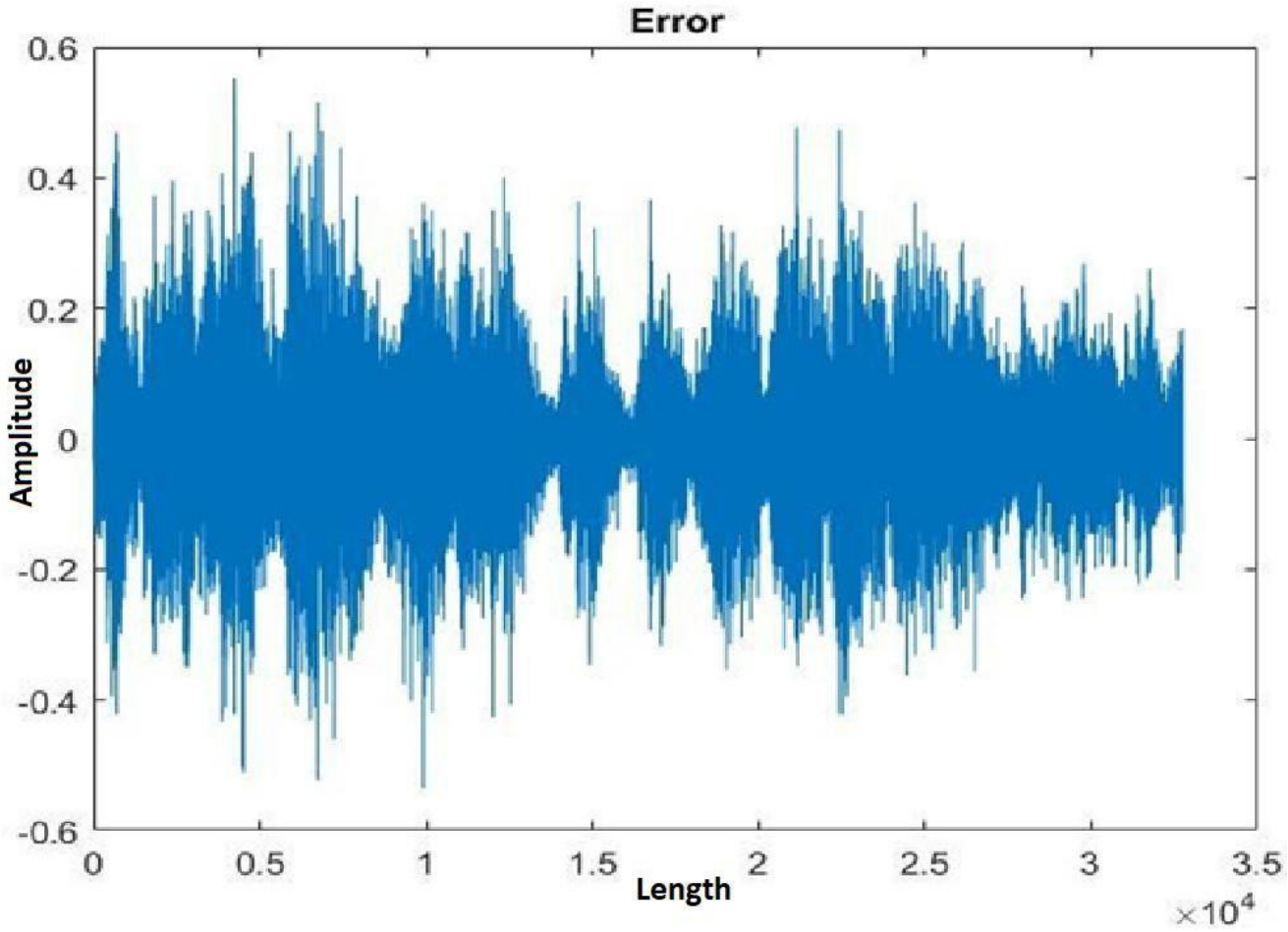



Combined plot
original
estimate d
Magnitude
Length
×10^4
1
0.8
0.6
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0
0.5
1
1.5
2
2.5
3
3.5


Error
Amplitude
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0
0.5
1
1.5
2
2.5
3
3.5
Length
×10^4

OUTPUT FOR WEINER FILTER:





CONCLUSION:

Speech enhancement is implemented using two methods namely Wiener filter and Kalman filter. In our daily life, the signals are not stationary and will vary randomly. Wiener filter is suitable for stationary signals only but it has problem like musical noise. To overcome these problems, Kalman filter is applied, which is time domain in nature. It has overcome the disadvantages mentioned in early two methods. Each method is implemented in MATLAB and SNR values of respective methods are compared. It is observed that, among two methods, performance of the Kalman filter is comparatively good for both stationary and non-stationary signals and from the results.