# Comparison between Kalman Filter and Weiner Filter

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.

## Working:

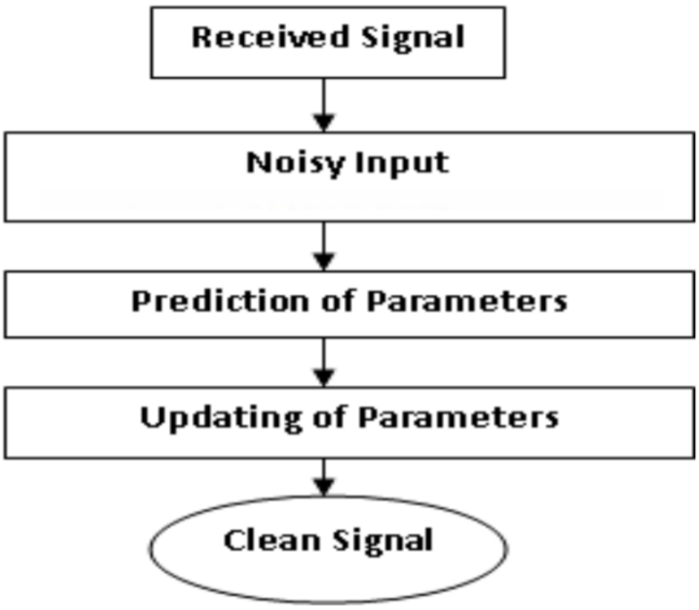
Initially, we have taken the audio input signal from MATLAB sample audios and added a random noise to it which produced appropriate output. The clean speech used in this work is a sentence pronounced by a group of people laughing together (name of the file being "Laughter-16-8-mono-4secs.wav" from MATLAB 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 and Weiner 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 and Weiner gain for each loop for upgradation 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 and Weiner filter is calculated and compared to each other.

**Steps Followed:**

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 & Weiner gain
    - Desired signal
    - Predicted state error
    - Estimated error sequence
    - Process noise covariance

- Measurement noise covariance
- Output of the signal

**<u>Block Diagram:</u>**



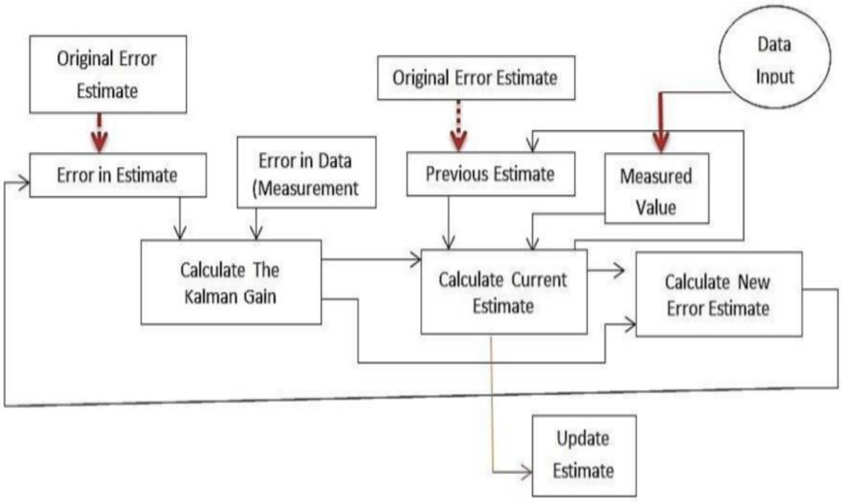




**Fig.1:** Mechanism of Kalman filters in speech enhancement

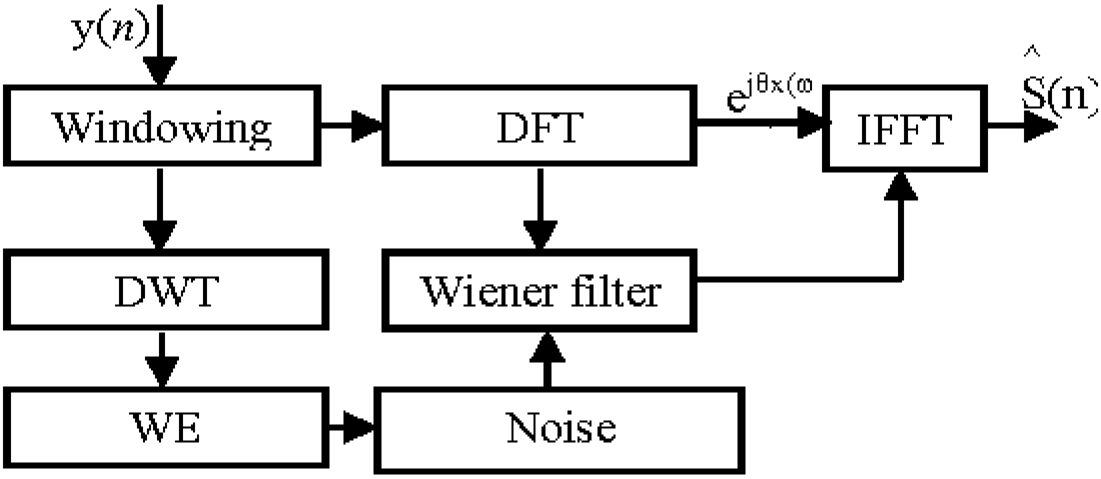
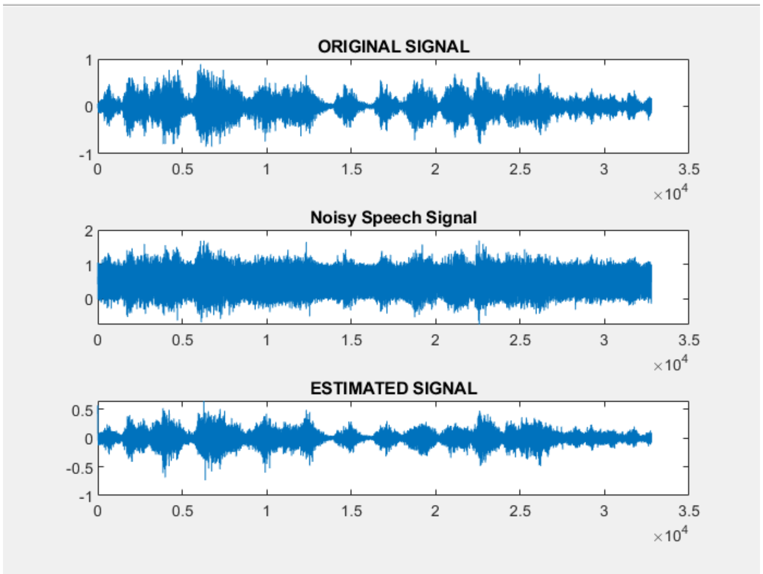
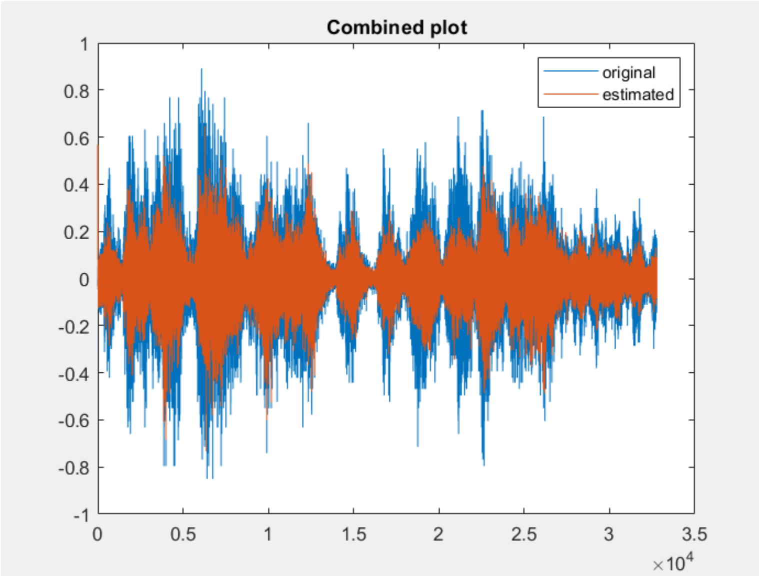
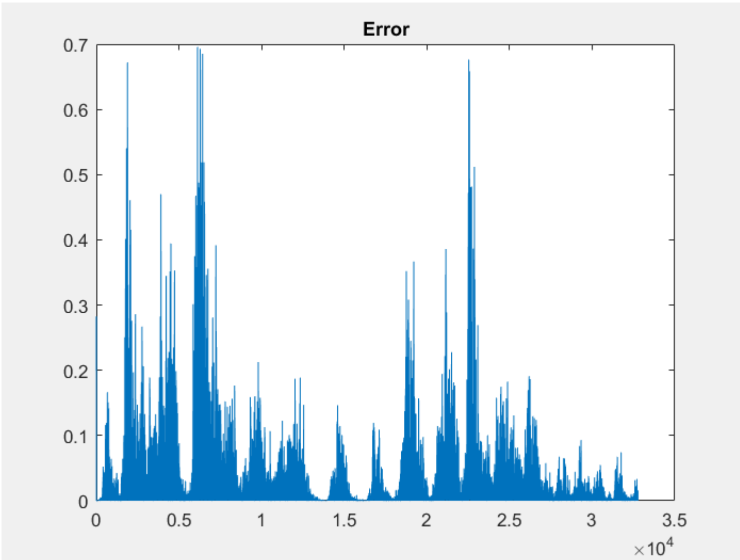



**Fig.2:** Mechanism of Weiner filters in speech enhancement

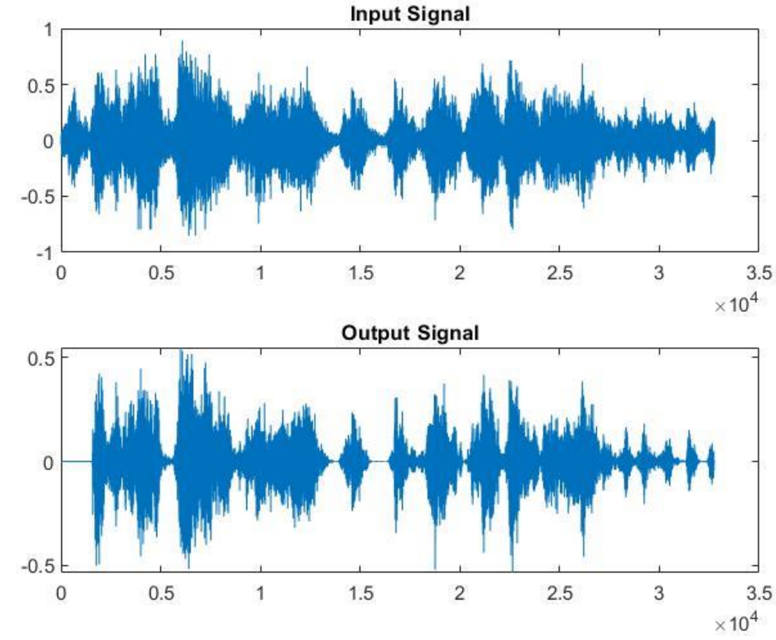
## Output Signal (Kalman Filter):

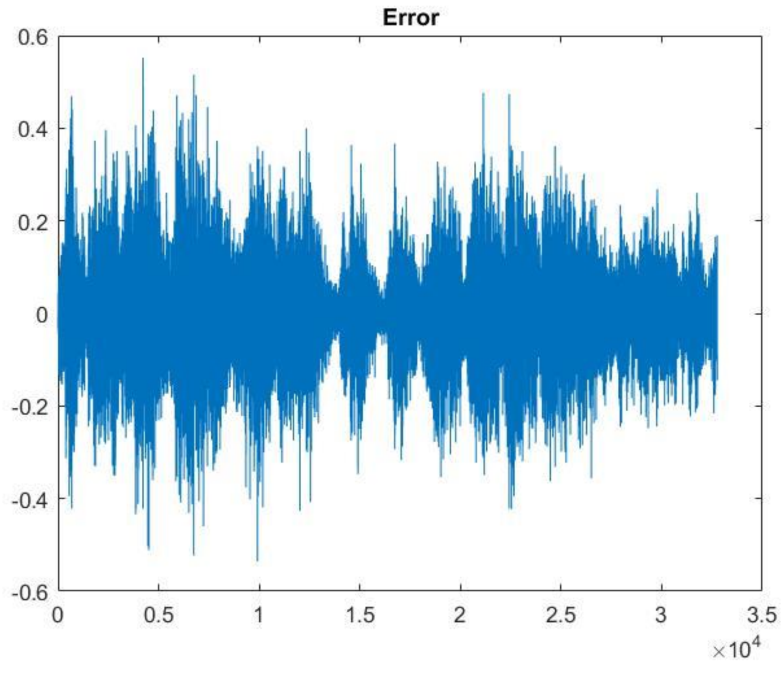



Combined plot
original
estimated
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0
0.5
1
1.5
2
2.5
3
3.5
×10^4


Error
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
×10^4

## Output Signal (Weiner Filter):



## **Error Comparison:**

Command Window

```
MSE_kalman =

   3.9056e-05

>> Weiner_Filter

MSE_weiner =

    0.0014

>>
```