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Enhancement of Degraded Speech Using Spectral Subtraction,Wiener Filter and Kalman Filter

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***Abstract*—In Speech communication, Degradation of speech due to background noise is very big problem. This paper presents three noise reduction techniques like Spectral Subtraction, Wiener Filter and Kalman filter and presents our simulation results of these noise reduction techniques. It is shown that system is removed the background noise almost completely from noisy speech and gives the best SNR values.**

***Keywords*—Noise Reduction; Speech Signal; Speech Enhancement; Spectral Subtraction; Wiener Filter.**

I. INTRODUCTION

In speech communication, the speech signal is always accompanied by some noise. In most of the cases, background noise of any environment where the source of speech lies, it is the main component of noise signal which adds to the speech communication. |The obvious effect of this noise addition is to make the listening task difficult for a direct listener, there are many negative effects when we process the degraded speech for some other applications. A related problem is processing degraded speech signal in the preparation for coding by the bandwidth compression system of speech. So, speech enhancement not only consists processing speech signals for human listening system but also for further processing prior to listening task. The Main objective of speech enhancement technique is to improve the perceptual aspects of speech signal like overall quality, intelligibility, or degree of listener fatigue. In our work, we study two speech enhancement techniques to enhance the quality of speech in the presence of additive and broadband acoustic noise which is spectral subtraction and wiener filtering method.

Spectral subtraction is the earliest popular method for enhancing the speech degraded by additive background noise. This technique estimates the spectrum of the clean means noise-free signal by the subtraction of the estimated noise magnitude spectrum from the noisy speech signal magnitude spectrum while keeping the phase spectrum of the noisy speech signal. The drawback of this technique is the residual noise.

Wiener filtering method designs the optimal filter which is minimizing the mean squared error (MSE) in the frequency domain. The musical noise is more reduced by the Wiener filter method than the Spectral subtraction method. If accurate power spectrum of the clean speech and accurate power spectrum of additive noise can be estimated, then the Wiener filtering method can be designed accurately, but here the power spectrum of the clean speech cannot be observed directly. So, the iterative Wiener filtering (IWF) method is processed to estimate the power spectrum more accurately. Here, First, the power spectrum of noise is estimated in silent segment of speech signal and the speech power spectrum is estimated by using LPC analysis for noisy speech signal. second, the Wiener filter is designed by using the estimated spectra and speech enhancement is carried out by filtering the noisy speech with the Wiener filter to obtain the enhanced speech signal. Next, LPC analysis is operated for the enhanced speech and the Wiener filter is designed again and the filter is processed for the noisy speech to obtain the enhanced speech signal. These procedures are repeated again to obtain the more accurate speech power spectrum and also to design more optimal Wiener filter. So, it is known that the spectrum of the enhanced speech is distorted after several iterations and then the optimal number of iteration cannot be determined.

Kalman filter is used to reduce the background noise using the prediction and correction mechanism.

In section II we have discuss Spectral subtraction method related to speech enhancement, in section III Wiener filter algorithm is discussed, in section IV Kalman filter algorithm is discussed Section V shows performance analysis of spectral subtraction, wiener filter and Kalman filter method.

II. SPECTRAL SUBTRACTION ALGORITHM

When we get the additive background noise from the signal then we can use spectral subtraction technique to estimate that noise from noisy speech signal. Sometimes when the speech signal has more random noise then sometimes it is not possible to reduce the fully noise but by the spectral subtraction technique it is possible to reduce the average effect of noise from that additive noise speech signal.

The noisy speech signal model in the time domain space is given by[15],

$$y(m)= x(m) +n(m) \tag{1}$$

where y(m) is the input signal , x(m) is the additive noise speech signal and n(m) is the noisy speech signal and m is the discrete time index or time variable state.

In the frequency domain method, the noisy speech signal model is given as[15],

$$Y(f)= X(f)+ N(f) \tag{2}$$

where $Y(f)$, $X(f)$ and $N(f)$ are the Fourier transforms of the noisy speech signal $y(m)$, the original speech signal $x(m)$ and the noise $n(m)$ respectively and f is the frequency variable.

In spectral subtraction, the incoming speech signal $x(m)$ is divided into segments of N samples length. Each segment is windowing, using a Hanning or a Hamming window, then transformed via discrete Fourier transform (DFT) to the N spectral samples of speech signal. The windows remove the effect of discontinuities at the end points of each segment in speech signal.

The windowing signal is given as[15]:

$$\begin{aligned} y_w(m) &= w(m)\, y(m) \\ &= w(m)\, [\, x(m) + n(m)\,] \\ &= x_w(m) + n_w(m) \end{aligned} \tag{3}$$

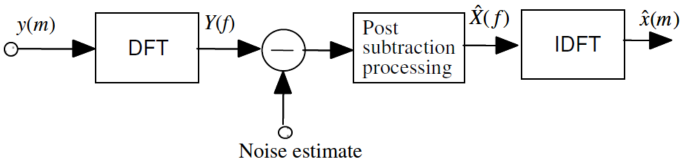

Fig.1 A block diagram illustration of spectral subtraction [15].

The equation describing spectral subtraction is given as[15],

$$|\hat{X}(f)|^b = |Y(f)|^b - \alpha |N(f)|^b \tag{4}$$

The parameter α is control the amount of noise subtracted from the noisy speech signal. For fully noise subtraction, α=1 and for over-subtraction α>1. For magnitude spectral subtraction, the exponent b=1, and for power spectral subtraction, b=2. But in this method, the background noise is still remains, So, to overcome of this problem another method wiener filtering is used to remove that background noise from the noisy speech signal.

III. WIENER FILTER ALGORITHM

The Wiener filter is a popular technique and it is used in many signal enhancement methods. Wiener filter technique designs the filter minimizing the mean squared error (MSE) in the frequency domain. The musical noise is reduced by the Wiener filter method than the spectral nsubtraction method. If accurate power spectrum of clean speech signal and accurate power spectrum of additive noise speech signal can be estimated, then the Wiener filter can be designed accurately. However, the power spectrum of clean speech signal cannot be observed directly. So, the iterative Wiener filter (IWF) method is adopted to estimate the power spectrum more accurately.

Assume that the clean speech signal $s(t)$ degraded by an additive noise $w(t)$, the noisy speech $x(t)$ is defined as [5],

$$x(t) = s(t)+w(t) \tag{5}$$

Wiener filter is a filter that minimizes the Mean Squared Error (MSE) criterion, the filter can be defined as [5],

$$S(w) = H(w)X(w) \tag{6}$$

Wiener filtering is a method to minimize noise in speech signals. It is based on minimizing the mean square error between the estimated speech signal magnitude spectrum and the original signal magnitude spectrum S(ω). The formulation of the optimal wiener filter is by follows H(w) is [5],

$$\frac{Ps(w)}{Ps(w) + Pv(w)} \tag{7}$$

where Ps(ω) and Pv(ω) are the power spectral densities of the clean speech and the noise signals, respectively. This formula can be derived considering the speech signal s and the noise signal v as uncorrelated and stationary signals. The SNR is defined as[5],

$$SNR = \frac{Ps(w)}{Pv(w)} \tag{8}$$

The enhanced speech signal is estimated in the frequency domain as[5],

$$S(w) = \frac{Ps(w)}{Ps(w) + Pv(w)} X(w) \qquad (9)$$

The enhanced speech signal can be obtained by inverse FFT for *S*(w) and Over-Lap Add procedure is used in the time domain method between adjacent frames to avoid the click sound.
The main drawback of the Wiener filter is the fixed frequency response at all frequencies and the requirement to calculate the power spectral density of the clean speech signal and noise signal prior to filtering.

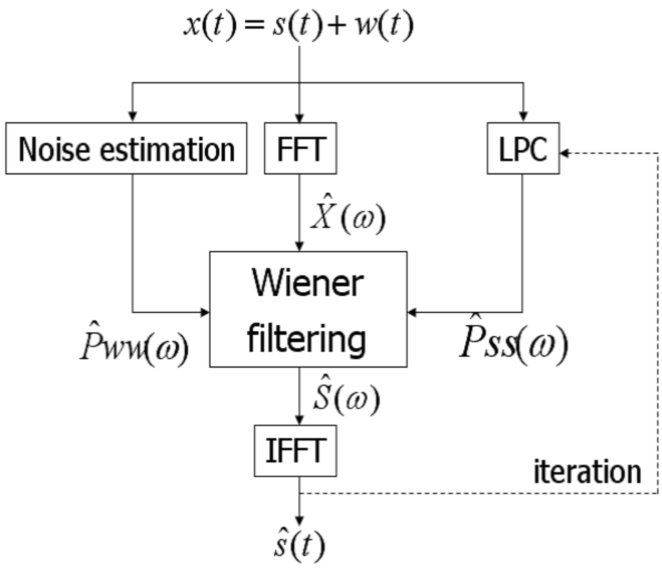

Fig.2 Block diagram of the WF algorithm[5]

The performance of the Wiener filter depends on the accuracy of speech power spectral estimation which is *Pss*(w). It is used to make the estimated spectrum close to the true one by repeating the Wiener filtering technique. Figure 2 shows the block diagram of the iterative Wiener filter algorithm.
There are two kinds of power spectra can be estimated by Linear Predictive Coding (LPC) analysis. Noise power spectrum, P_{ww}(w) are estimated in the first non-speech section from speech signal.
Speech power spectrum, P_{ss}(w) is estimated by LPC analysis for input noisy speech signal, *x*(*n*). Using the Wiener filtering and Inverse FFT operation, enhanced speech signal is estimated and then it is analyzed to estimate the more accurate speech power spectrum, P_{ss}(w) by means of LPC analysis and the Wiener filter is operated again. The iterative procedure is again repeated to obtain more clean speech.

A. Two Step Noise Reduction (TSNR)

To enhance the performance of the noise reduction technique, we propose the method to estimate the *a priori* SNR[20] in a two-step procedure. The DD algorithm[20] introduces a frame delay when the parameter β is very close to one. The spectral gain computed at current frame p, that matches the previous frame of p−1. Based on this procedure, we propose to calculate the spectral gain for the next frame p + 1 using the DD approach[20] and to apply it on to the current frame because of the frame delay in signal. This leads to an algorithm in two steps.
In the first step, using the DD algorithm[20], we compute the spectral gain GDD(p, k)[20]. In the second step, this gain[20] is used to estimate the *a priori* SNR[20] at frame p + 1[20]:

$$\widehat{SNR}_{Prio}^{TSNR}(p,k) = \beta \frac{[GDD(p,k)X(p,k)^2]}{Yn(p,k)} + (1-\beta)P[\widehat{SNR}post(p+1,k) - 1] \qquad (10)$$

And Gain of Spectrum by TSNR is find as[20],

$$G_{TSNR}(p,k) = ((SNR)\hat{}\_Prio\hat{}TSNR\,(p,k))/(1 + (SNR)\hat{}\_Prio\hat{}TSNR\,(p,k)\,)$$

(11)

B. Harmonic Regeneration of Noise reduction (HRNR)

Some harmonics are considered as noise-only components and are subtracted. Here, We propose to take advantage of the harmonic structure of voiced speech signal to prevent the harmonic distortion. For that purpose, we propose to process the distorted signal to create a fully harmonic signal where all the missing or degraded harmonics are regenerated. This signal will then be used to compute a spectral gain which is able to preserve the speech harmonics. This will be called the speech harmonic regeneration step and can be used to improve the results of any noise reduction technique.
The Priori SNR by HRNR is find as[20],

$$\widehat{SNR}_{Prio}^{HRNR}(p,k) = \frac{p(p,k)[S(p,k)]^2 + [1-p(p,k)][Sharmo(p,k)]^2}{\Upsilon n(p,k)} \quad (12)$$

And Gain of Spectrum by HRNR is find by[20],

$G_{HRNR}(p,k)=$ ((SNR)^_Prio^HRNR (p,k))/(1+ (SNR)^_Prio^HRNR (p,k))

(13)

Here, The two-step noise reduction technique is remove the annoying reverberation effect from noisy speech signal while maintaining the benefits of the decision-directed approach[20]. So, classic short-time noise reduction techniques, including TSNR, introduce harmonic distortion in the enhanced speech. To overcome of this problem, a method called harmonic regeneration noise reduction technique (HRNR) is implemented to refine the a priori SNR used to calculate a spectral gain which is able to preserve the speech harmonics from speech signal.

IV. KALMAN FILTER ALGORITHM

The Equations to estimate the error or noise for Kalman filter is divided into two groups, first is time update equations and second is measurement update equations.
Time update equations are used for projecting a speech in a forward in time the current state and error covariance estimates to obtain the priori estimates of the next time step as a predictor equations.
Measurement update equations are used for the feedback to solve the error or noise like integrating into the a priori estimate state a new measurement to get an enhanced speech signal a posteriori estimate as a corrector equations.

Symbol	Quantity	Time update equation
Xk	Priori state estimate at step k	$X_k^- = AX_{k-1}^- + BV_k$
P_k^-	Priori estimate error covariance	$P_k^- = AP_{k-1}A^T + Q$

Table [1] Discrete Kalman Filter Time Update equations[21]

Symbol	Quantity	Measurement update equation
Kk	Kalman Gain	$K_k = P_k^{-T}/(HP_k^- H^T + R)$
Xk	Posteriori state estimate	$\hat{X}_k = \hat{X}_k + Kk(Z_k - H_k\hat{X}_k)$
P_k^-	Posteriori error covariance	$P_k = (I - K_k H_k)P_k^-$

Table [2] Discrete Kalman Filter Measurement Update equations[21]

V. PERFORMANCE ANALYSIS

A. Simulation Results of Spectral Subtraction

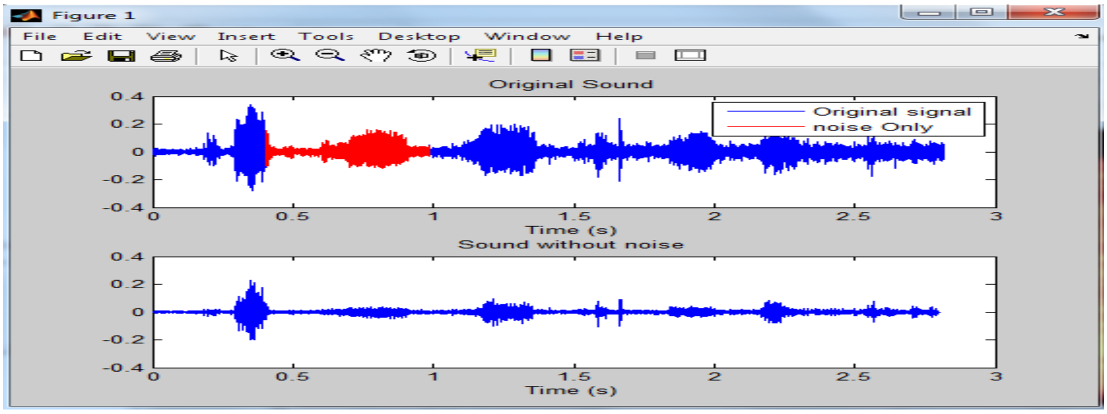

Fig.1(A) Original and clean signal

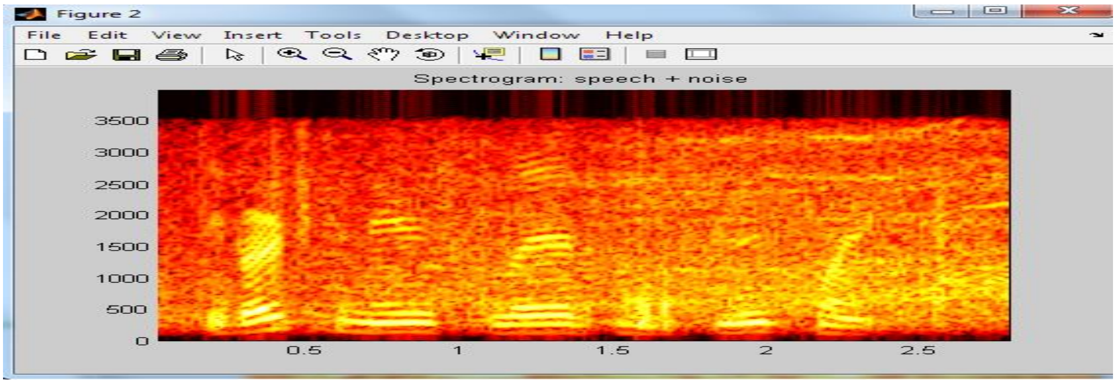

Fig.2(A) Noisy Speech Signal

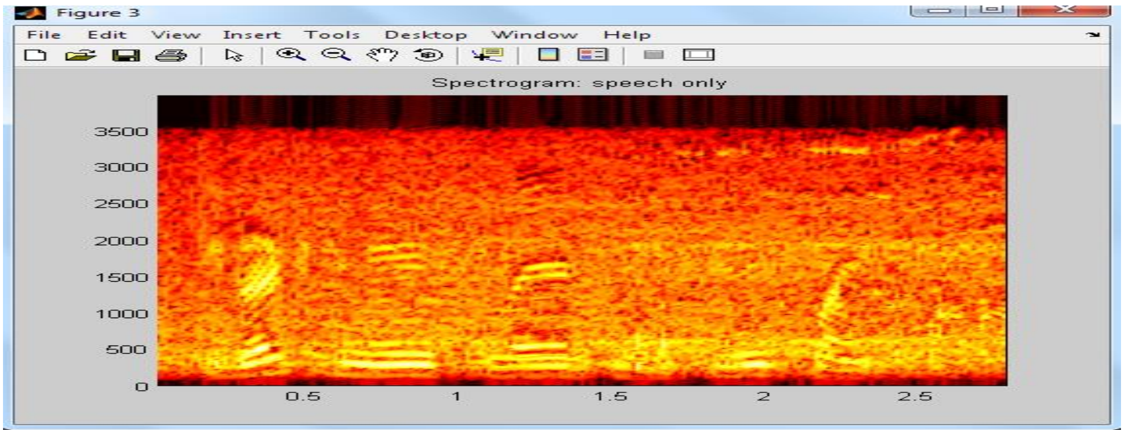

Fig.3(A) Clean speech signal

Using the standard noisy data base, below table 1(A) shows the SNR before filtering and SNR after Spectral subtraction method is applied on it.

Noise	SNR Before Filtering(db)	SNR After Filtering (db)
Street	0	13.7409
	5	9.1697
Airport	0	16.0357
	5	9.979
Babble	0	15.4177
	5	10.0933
Car	0	15.6772
	5	11.3941
Train	0	15.2425
	5	10.9907
Train Station	0	13.3801
	5	11.4095
Restaurant	0	13.1219
	5	10.6951
Exhibition Hall	0	15.1337
	5	8.2183

Table 1(A) SNR of noisy and enhanced speech by Spectral subtraction

B. Simulation of Wiener filter using TSNR and HRNR

Here, The two-step noise reduction (TSNR) technique is used to remove the annoying reverberation effect from speech while maintaining the benefits of the decision-directed approach[20]. However, classic short-time noise reduction techniques which including TSNR and introduce harmonic distortion in the enhanced speech signal. So, to overcome of this problem, a method called harmonic regeneration noise reduction Technique (HRNR) is implemented in order to refine the a priori SNR[20] used to compute a spectral gain which is able to preserve the speech harmonics.

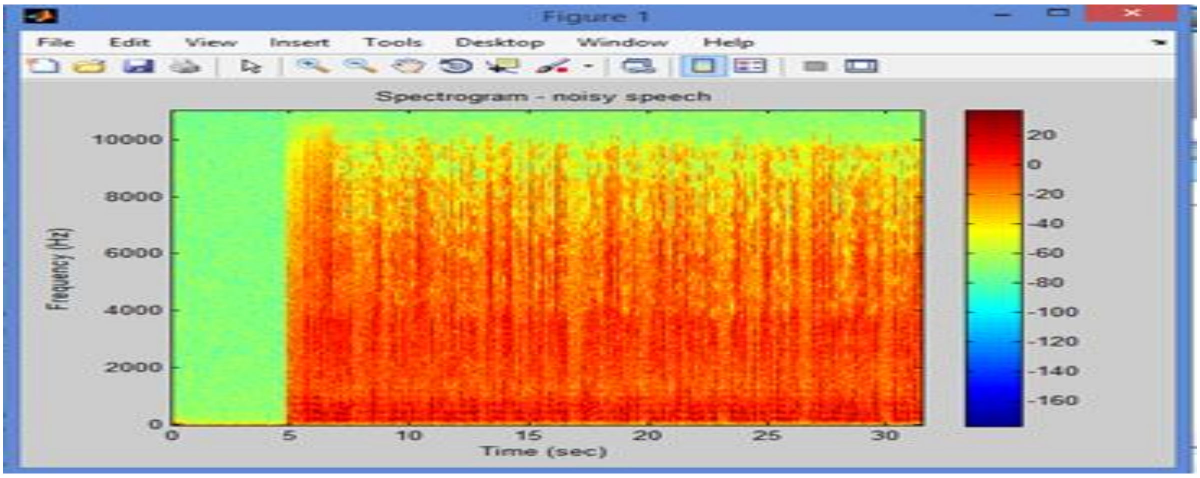

Fig.1(B) Noisy speech signal

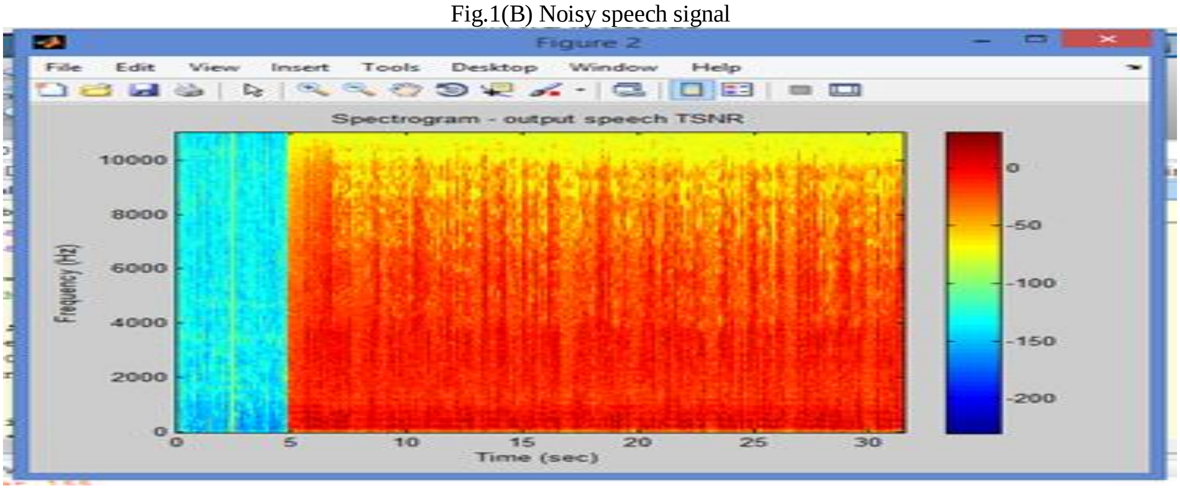

Fig.2(B) Enhanced speech using TSNR

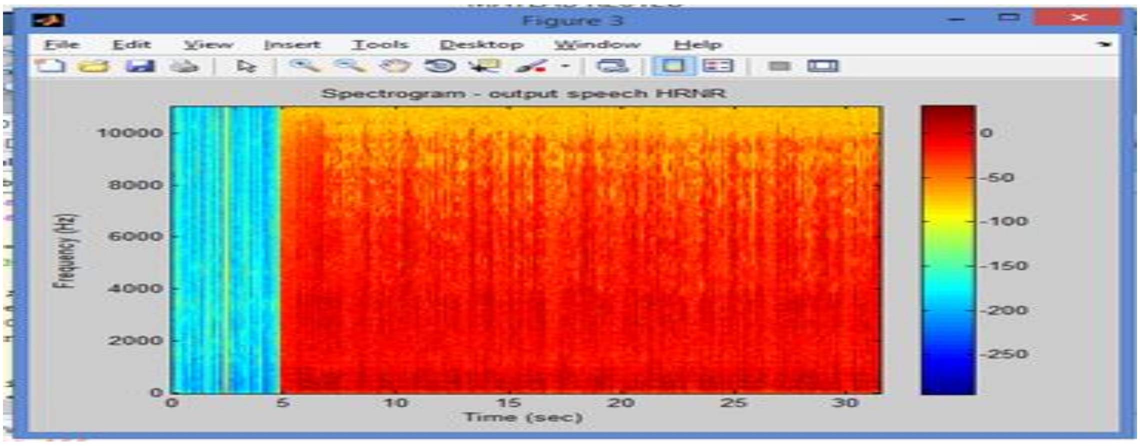

Fig.3(B) Enhanced speech using HRNR

Using the standard noisy data base, below table 1(B) shows the SNR before filtering and SNR after Wiener Filtering Using TSNR and HRNR is applied on it.

Noise	SNR Before Filtering(db)	SNR After Filtering (db)	
		TSNR	HRNR
Street	0	9.6194	10.342
	5	5.3112	5.5723
Airport	0	9.0117	9.7666
	5	6.6202	7.062
Babble	0	10.8116	11.8298
	5	6.1383	6.5486
Car	0	10.1083	10.7984
	5	6.88	7.3188
Train	0	8.7584	9.3516
	5	6.5658	6.9388
Train	0	9.2734	10.1614

Station	5	7.061	7.5836
Restaur ant	0	10.7194	11.7519
	5	5.5651	5.8376
Exhibit ion Hall	0	9.7586	10.4369
	5	5.4802	5.8125

Table 1(B) SNR of noisy and enhanced speech by Wiener Filter using TSNR and HRNR

C. Simulation Results of Kalman filter

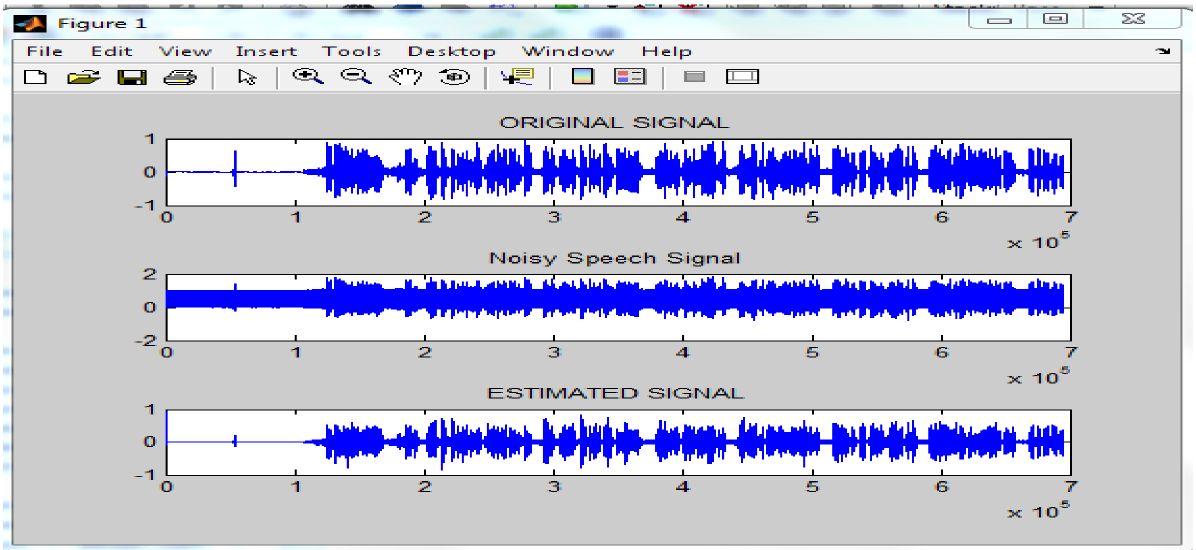

Figure 1(C). Original signal, Noisy signal and Estimated Signal

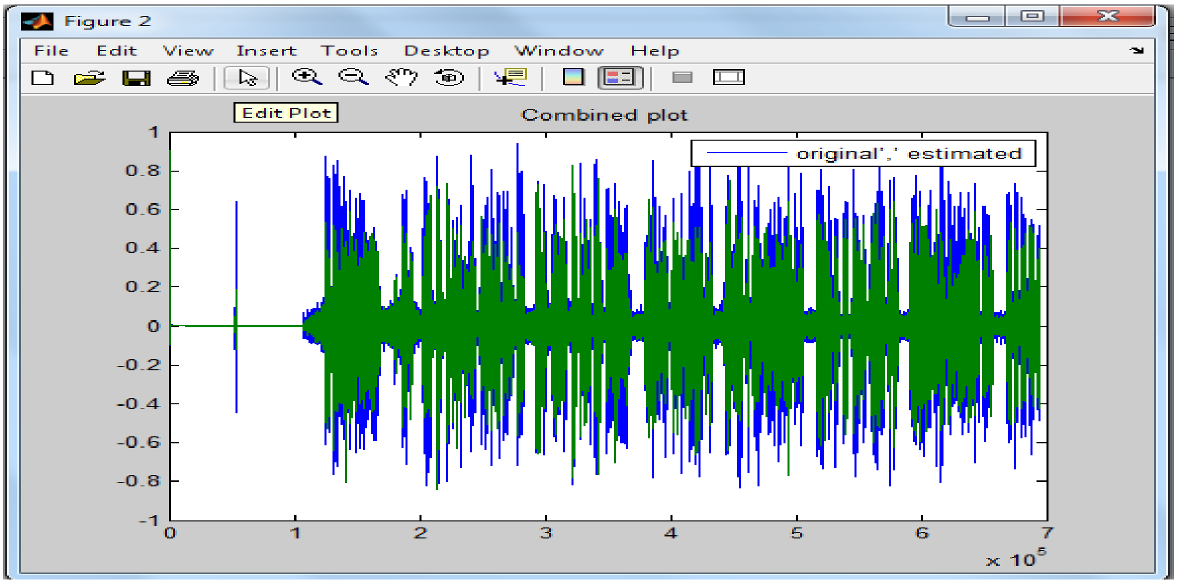

Figure 2 (C). Combined Plot Of Original and Estimated Signal

Noise	SNR Before Filtering(db)	SNR After Filtering (db)
Street	0	2.268
	5	4.477
Airport	0	5.0235
	5	4.8659
Babble	0	3.0304
	5	3.39541
Car	0	0.6424
	5	4.0106
Train	0	2.4872
	5	0.6859
Train Station	0	1.7962
	5	2.9297
Restaurant	0	5.3479
	5	4.5321
Exhibition Hall	0	2.0076
	5	0.73654

Table 1(C) SNR of noisy and enhanced speech by Kalman Filter

V. CONCLUSION

This paper investigated the problem of noisy conditions assuming absence of information about the noise. The works reported are carried out using a standard database like street, airport, babble, car, train, station, exhibition hall and restaurant noise. Three different methods, namely spectral subtraction, Wiener filter and Kalman filter has been used for noise elimination from the speech signal. From the performance, it has been observed that spectral subtraction is a very efficient method for elimination of all noise in high SNR condition. However, its performance rapidly degrades with increasing of SNR. Further, in the present study, it has been observed that at all SNR condition Wiener filter is an efficient noise reduction technique. Kalman filter is efficient only for the low SNR values.

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