

Speech Enhancement by Spectral Subtraction through Kaiser Window Shape

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ABSTRACT: This paper presents the effect of Window shape on the improvement of Speech quality by reducing the noise using Kaiser Window with optimum shape. In Speech Enhancement process, signal degraded due to noise is segmented into frames and each segment is Windowed using Kaiser Window with variation in the shape parameter. The Windowed Speech segments are applied to the Spectral Subtraction Speech Enhancement algorithm and the Enhanced Speech signal is reconstructed in its time domain. The focus is to study the effect of Kaiser Window shape on the Speech Enhancement process. For different shapes of the Kaiser Window, it is observed that the Window shape plays an important role on the enhancement process. The quality of the Enhanced Speech was measured using six objective measures. The results are compared with the measures of Hanning window and an optimum shape constant for the Kaiser Window is proposed for better speech quality.

Keywords: DFT, Kaiser Window, objective measures, Speech Enhancement, Shape parameter ALPHA

1. Introduction

Speech is the most common way of human communication. The study of speech signals and its processing methods is known as speech processing. Speech enhancement aims to improve speech quality by using various algorithms. The objective of enhancement is improvement in intelligibility and/or overall perceptual quality of degraded speech signal using audio signal processing techniques. Speech enhancement is the most important field of speech processing. It is used for many applications such as mobile phones, teleconferencing systems, speech recognition and hearing aids. The processed speech signals should be more comfort for listening and also should give better performance in tasks like automatic speech and speaker recognition [1]. Several algorithms are proposed in the literature [2] for speech enhancement such as spectral subtraction methods, MMSE methods, Weiner algorithm etc. In STSA based speech enhancement method, the noisy speech signal is partitioned into frames. Each frame is multiplied by a window function prior to the spectral analysis and applied to the speech enhancement algorithm. The focus of the study is on the effect of windowing in the Speech Enhancement process in terms of six objective measures for Spectral Subtraction Speech Enhancement process.

The purpose of windowing is to reduce the effect of discontinuity introduced by the framing process. Commonly used windows include Hamming and Hanning [3]. Although these windows have a reduced side lobe levels, they have also reduced frequency resolution. Hence several factors enter into the choice of Window selection to frame the Speech for Enhancement. In this paper an attempt has been made to explore the possibility of improving the quality of speech signal by employing different shapes of Kaiser Window.

To study the performance of any algorithm, combinations of subjective and objective measures have to be carried on. Currently, the accurate method for evaluating speech quality is through

subjective listening tests. But it is costly and time consuming. Hence, six objective measures are chosen to evaluate the performance of the Kaiser Window in an enhancement system. P. Loizou has presented a correlation analysis of objective quality measures for evaluating speech enhancement algorithms [4]. The six measures namely SNR, Segmental SNR (Seg-SNR), Log Likelihood Ratio (LLR), Weighted spectral slope distance (WSS), frequency weighted segmental SNR (fwseg-SNR) and Cepstral distance (Cep) are selected for performance evaluation test, considering the fact that fwseg-SNR, LLR, Cep and WSS have high correlation with overall speech quality. The correlation coefficients for these measures with speech quality are 0.84, 0.85, 0.79 and 0.64 respectively [4]. These objective measures also have good correlation with subjective scores. Although the correlation coefficient of SegSNR is 0.36, it is chosen as a time domain measure where as all other measures evaluated are of frequency domain.

This paper explains the Kaiser Windowing effect on the noisy speech for Enhancement in terms of the six objective measures using Spectral Subtraction Speech Enhancement algorithm. The rest of the paper is organized as follows: Section-2 briefly reviews various windows for noisy Speech Enhancement. In Section-3 presents the Six Objective measure used in this study. In Section-4 Spectral Subtraction method for noisy speech enhancement is explained. Implementation of the scheme is explained in Section-5 and Section-6 explains the results and conclusion.

2. The Data Weighting Windows

2.1 Windowing

Windows are time-domain weighting functions that are used to reduce Gibbs' oscillations resulting from the truncation of a Fourier series [5-6]. Their roots date back over one-hundred years to Fejer's averaging technique for a truncated Fourier series and they are employed in a variety of traditional signal processing applications including power spectral estimation, beam forming, and digital filter design. The effect of a time window can be described in the frequency domain as a convolution of the frequency response of the window with the frequency response of the signal. The convolution smears frequency features, with the amount of smearing depending on the width of the main lobe of the window frequency response. For signal frequencies, observed through the rectangular window, which do not correspond exactly to one of the sampling frequencies, the pattern is shifted such that non-zero values are projected onto all sampling frequencies. This phenomenon of spreading signal power from the nominal frequency across the entire width of the observed spectrum is known as spectral leakage. In addition, spectral leakage from distant frequency components will occur if the sidelobe level of the window response is large [7]. Ideally, the window spectrum should have a narrow main-lobe and small side-lobes. However, there is an inherent trade-off between the width of the main-lobe and the side-lobe attenuation. A wide main-lobe will average adjacent frequency components and large sidelobes will introduce contamination (or spectral leakage) from other frequency regions. The main lobe for rectangular window is narrower than that of the Hanning window, while its side-lobes are higher. Some of the commonly used windows in speech processing are symmetric (e.g., Hamming and Hanning windows) or asymmetric (such as the hybrid Hamming-Cosine window). The goal of asymmetric windows is to reduce the algorithmic delay in speech coders.

2.2 Fixed and Adjustable Windows

In literature many windows have been proposed [7-10]. They are known as suboptimal solutions, and the best window is depending on the applications. Windows can be categorized as fixed or adjustable [11]. Fixed windows have only one independent parameter, namely, the window length that affects the main lobe width without substantially changing the sidelobe heights. For a fixed N , the time-bandwidth product trades off main lobe width against sidelobe suppression. Adjustable windows have two or more independent parameters, namely, the window length, as in fixed windows, and one or more additional parameters that can control other window characteristics [7, 11-15]. The Kaiser and Saramaki windows [8-9] have two parameters and achieve close approximations to discrete prolate functions that have maximum energy concentration in the main lobe. The Dolph-Chebyshev window [10, 12] has also two parameters and produces the minimum main-lobe width for a specified maximum side-lobe level. With adjusting their two independent parameters, namely the window length and the shape parameter, it can be controlled the spectral parameters of main lobe width and ripple ratio for various applications.

Kaiser window has a better sidelobe roll-off characteristic than the other well known adjustable windows such as Dolph-Chebyshev [10] and Saramaki [11], which are special cases of ultra spherical window [16]. The effect of the Window length in the speech Enhancement was studied and proposed a Window length of 15-35 ms (at a sampling rate of 8 kHz), will be a reasonable compromise in terms of the dynamics of speech production [17]. In the present work it is observed that, by properly selecting the shape parameter of the adjustable Window of fixed length 20ms, it is made possible to achieve the speech enhancement and using the objective measures, it can be established quantitatively. In this paper, study is restricted to Kaiser Window. The related work using other variable windows is under process.

2.3 Kaiser-Bessel Windows

The Kaiser-Bessel window was formulated to be a simple-to-compute approximation to the zero'th order discrete prolate spheroidal sequences in the main lobe [8]. As such, it tends to maximize the energy in the main lobe for a given sidelobe level. The discrete-time Kaiser-Bessel window is a sampled version of a continuous function with a parameter " α ". This parameter is analogous to the time-bandwidth product and trade off main lobe width against side lobe attenuation.

In discrete time domain, Kaiser Window is defined by [9]

$$w(n) = \begin{cases} I_0\left(\alpha\sqrt{1 - \left(\frac{2n}{N-1}\right)^2}\right) & |n| \leq \frac{(N-1)}{2} \\ 0 & \text{Otherwise} \end{cases} \quad (2.1)$$

Where " α " is the adjustable shape parameter, and $I_0(x)$ is the modified Bessel function of the first kind of order zero and it is described by the power series expansion as

$$I_0(x) = 1 + \sum_{k=0}^{\infty} \left[\frac{1}{k!} \left(\frac{x}{2}\right)^k \right]^2 \quad (2.2)$$

where $I_0(\cdot)$ is the zero'th-order modified Bessel function of the first kind. As “ α ” increases, the main lobe width widens and the side lobe attenuation increases. For “ α ” = 0, the Kaiser-Bessel window is a rectangular window. For “ α ” = 5.3, the Kaiser-Bessel window is close to a Hamming window. The two parameters useful to obtain the desired amplitude response pattern of the above defined windows are the length of the sequence N and a shape parameter “ α ”. In this study the number of FFT points considered in the Windowed speech frame is fixed to 160 or 240 hence the window length N is fixed. Therefore, the shape parameter “ α ” can only be varied to achieve the desired pattern of the magnitude response of the Window used. As the shape parameter “ α ” increases, the side lobe level of the magnitude response decreases at the cost of main lobe width.

2. Objective Measures:

i) **Signal-To-Noise Ratio (SNR):** The Signal-to-Noise Ratio (SNR) is the ratio of signal energy to noise energy and is expressed in decibels dB given by [2-4] as

$$SNR_{dB} = 10 \log_{10} \left[\frac{\sum_n s^2(n)}{\sum_n [s(n) - \hat{s}(n)]^2} \right] \quad (3.1)$$

Where $s(n)$ is the undistorted or clean signal and $\hat{s}(n)$ is the degraded or processed / enhanced speech signal, N is the frame length.

ii) **Segmental Signal-to-noise Ratio:** The Seg-SNR is the frame-based SNR and is estimated as It is defined [2-4] as

$$SegSNR_{dB} = \frac{1}{M} \sum_{m=0}^{M-1} 10 \log_{10} \left[\frac{\sum_{n=0}^{N-1} |s(n+mN)|^2}{\sum_{n=0}^{N-1} |s(n+mN) - \hat{s}(n+mN)|^2} \right] \quad (3.2)$$

Where $s(n)$ is the undistorted or clean signal and $\hat{s}(n)$ is the degraded or processed / enhanced speech signal, N is the frame length. M represents the number of frames. The Seg-SNR poses a problem if there are intervals of silence in the speech utterance. In segments in which the original speech is nearly zero, any amount of noise can give rise to a large negative SNR for that segment, which could appreciably bias the overall measure of Seg-SNR. This problem is resolved by including the SNR of the frames only if the frame energy is above a specified threshold. Generally, the frames with segmental SNR between -10 dB to 35 dB are only considered in the average.

iii) **Weighted Spectral Slope Distance:** WSS distance measure computes the weighted difference between the spectral slopes in each frequency band. The spectral slope is obtained as the difference between adjacent spectral magnitudes in decibels. The WSS measure is defined and evaluated [18] as

$$WSS = \frac{1}{M} \sum_{m=0}^{M-1} \left[\frac{\sum_{j=1}^k W(j,m) [s_c(j,m) - s_p(j,m)]^2}{\sum_{j=1}^k W(j,m)} \right] \quad (3.3)$$

Where $W(j, m)$ are the weights computed. $s_c(j, m)$ and $s_p(j, m)$ are the spectral slopes for j^{th} frequency Band at m^{th} frame of clean and processed speech signals respectively.

iv) **Log Likelihood Ratio:** The LLR measure is defined [19] as

$$LLR = \log_{10} \left[\frac{a_p R_s a_p^T}{a_s R_s a_s^T} \right] \quad (3.4)$$

Where a_p and a_s are the LP coefficient vectors for the clean and degraded or enhanced speech segments, respectively. R_s denote the autocorrelation matrix of the clean speech segment.

v) **Cepstrum distance:** The Cepstrum distance [19] provides an estimate of the log spectral distance between two spectra. It is defined as

$$CD(m) = \sqrt{\sum_{k=1}^N [c_s(k, m) - c_p(k, m)]^2} \quad (3.5)$$

Where $C_s(n)$ and $C_p(n)$ represent the cepstrum of clean and the degraded or enhanced speech respectively.

$$C_s(k, m) = \text{Re}[\text{IDFT}\{\log|DFT(s(k, m))|\}] \quad (3.6)$$

The cepstrum coefficients can also be obtained recursively from the LPC coefficients using the following expression [2, 4]

$$c(m) = a_m + \sum_{k=1}^{m-1} \frac{k}{m} c(k) a_{m-k} \quad \text{for } 1 \leq m \leq p \quad (3.7)$$

vi) **Frequency weighted segmental SNR:** It is computed [4, 20] using the following equation

$$fwSNRseg = \frac{10}{M} \sum_{m=0}^{M-1} \left\{ \frac{\sum_{j=1}^K W(j, m) \log_{10} \frac{s(j, m)^2}{[s(j, m) - \hat{s}(j, m)]^2}}{\sum_{j=1}^K W(j, m)} \right\} \quad (3.8)$$

where $W(j, m)$ is the noise-dependent weight applied on the j th frequency band, K is the number of bands, M is the total number of frames in the signal, $s(j, m)$ is the weighted (by a Gaussian-shaped window) clean signal spectrum in the j th frequency band at the m th frame, and $\hat{s}(j, m)$ in the weighted enhanced signal spectrum in the same band.

4. Spectral Subtraction

The noise corrupted speech is processed by the Spectral Subtraction method to get processed or enhanced speech. Spectral Subtraction [21-24] is a popular frequency domain method to reduce the effect of additive uncorrelated noise in a signal. The noise spectrum is estimated, and updated, from the periods when the signal is absent and only the noise is present. For restoration of the time-domain signal, an estimate of the instantaneous magnitude spectrum is combined with the phase of the noisy signal, and then transformed via an Inverse Discrete Fourier Transform (IDFT) to the time domain.

If $y(n)$ is the discrete noise corrupted input signal which is composed of the clean speech signal $s(n)$ and $v(n)$ the uncorrelated additive noise signal, then the noisy signal can be represented as:

$$y(n) = s(n) + v(n) \quad (4.1)$$

Since the speech is not a stationary signal, the processing is carried on short-time basis (frame-by-frame).

$$y(n, k) = s(n, k) + v(n, k) \quad (4.2)$$

Where n is the time index and k is the frame index, $y(n, k)$ is the k th frame. In the frequency domain, with their respective Fourier transforms, the power spectrum of the noisy signal can be represented as:

$$P_{yy}(w, k) = P_{ss}(w, k) + P_{vv}(w, k) \quad (4.3)$$

$$|Y(w, k)|^2 = |S(w, k)|^2 + |V(w, k)|^2 \quad (4.4)$$

Here $Y(w, k)$ is the DFT of $y(n, k)$ given by

$$Y(w, k) = \sum_{n=0}^{N-1} y(n, k) e^{-j \frac{2\pi wn}{N}} = |Y(w, k)| e^{j\varphi(w)} \quad (4.5)$$

Where $\varphi(w)$ is the phase of the corrupted noisy signal, and N is the number of samples in the framed speech signal.

Thus from Eq.12 the estimation of clean speech signal can be given as

$$|\hat{S}(w, k)|^2 = |Y(w, k)|^2 - |\hat{V}(w, k)|^2 \quad (4.6)$$

Once the estimate of the clean speech is obtained in the spectral domain, the enhanced speech signal is obtained according to:

$$\hat{s}(n, k) = IDFT\{|\hat{S}(w, k)| e^{j\varphi(w)}\} \quad (4.7)$$

Here, the phase information from the corrupted signal is used to reconstruct the time domain signal by taking the IDFT.

One may generalize the technique of spectral subtraction by replacing the magnitude squared of the DFT by some power of the magnitude. The exponent, '2', in Equation.14, can be replaced by 'a' as given below:

$$|\hat{S}(w, k)|^a = |Y(w, k)|^a - |\hat{V}(w, k)|^a \quad (4.8)$$

In this paper, all the measures were evaluated with $a = 1.7$.

A method of exponential averaging proposed in [22] is used to estimate the noise. The frame-by-frame update scheme using the exponential averaging method is given below:

$$|\hat{V}(w, k)|^a = \begin{cases} \mu |\hat{V}(w, k-1)|^a + (1-\mu) |\hat{Y}(w, k)|^a & \text{for Noise only} \\ |\hat{V}(w, k)|^a & \text{Speech and Noise} \end{cases} \quad (4.9)$$

Where $0 \leq \mu \leq 1$ is exponential averaging constant. In this work " μ " is selected as 0.85.

The block diagram for the overall spectral subtraction algorithm is shown in Figure.1 below.

5. Implementation

Phonetically balanced clean speech signals and noise corrupted signals at different SNR levels have been taken from a speech corpus called "NOIZEUS". Noise corrupted speech signal is segmented into frames containing 160 samples of 20ms length (at 8 KHz Sampling rate). 160- point Discrete Fourier Transform (DFT) of each segment is obtained after applying the Kaiser window with variable shape parameter " α ".

Spectral Subtraction Algorithm is applied to the spectral components of each segment using Eq.4.1-Eq.4.9. The signal is reconstructed in its time domain with the help of IDFT and overlap adds method (with 40% overlap between frames). The signal thus obtained is the enhanced signal. The performance of the enhanced signal is analyzed by using six objective measures for speech enhancement. The measures are WSS, LLR, fwseg-SNR, Cep, Seg-SNR, and SNR defined in Eq.3.1-Eq.3.8. All the measures are computed by segmenting the sentences using 30-ms duration Hamming windows with 75% overlap between adjacent frames. A tenth order LPC

analysis was used in the computation of LPC- based objective measure LLR. The performance of Kaiser windowed signal is studied under two real world noise conditions namely “Car noise” and “Train noise” at 0dB, 5dB, 10dB and 15dB SNR levels and presented in Table.6.1(a)-6.4(d) and Table.6.5(a)-6.8(d),

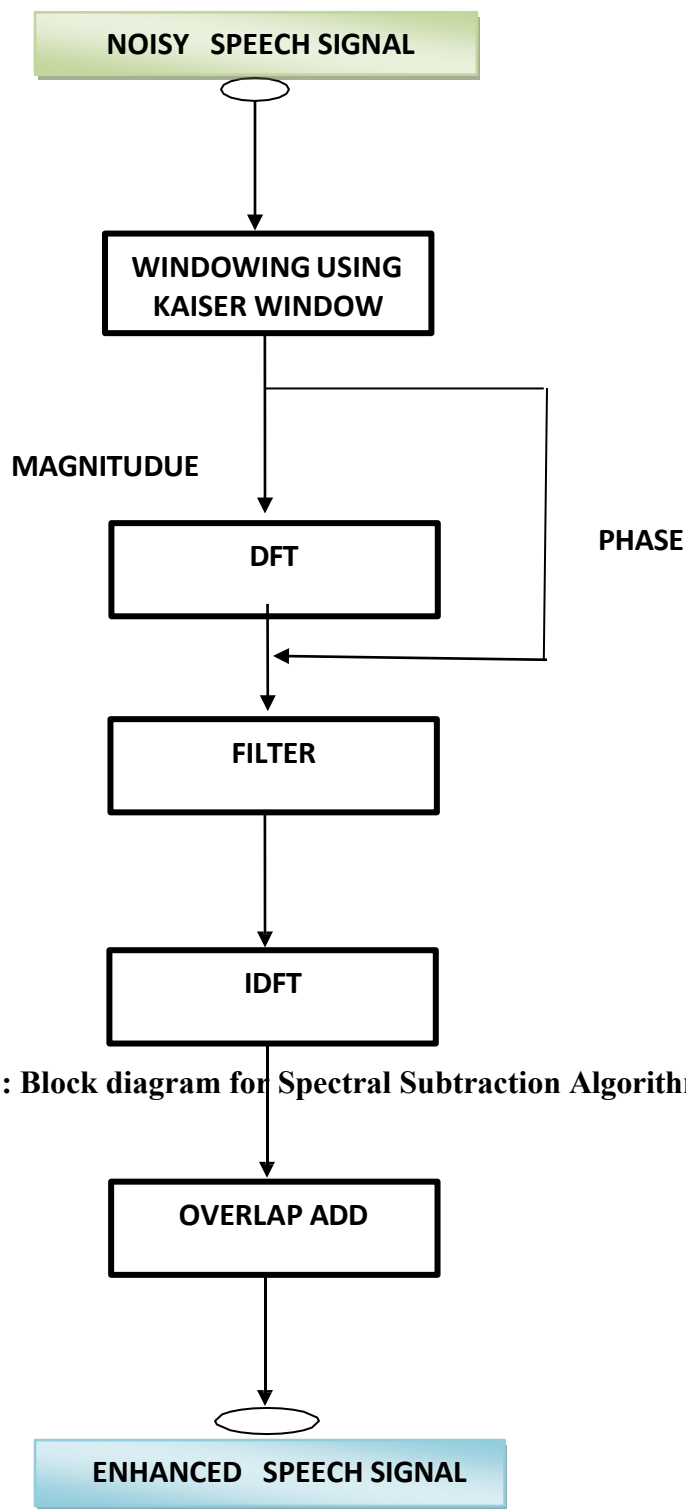


Fig.5.1: Block diagram for Spectral Subtraction Algorithm

6. Results and Conclusion

The Objective measure scores are shown in Fig. 6(a) - 6(b) as a function of the shape parameter " α " of the Kaiser window used as analysis window. The following observations can be made based on these results. For the narrow main lobe width of the analysis Window's frequency response, the Objective measure scores are good and as expected. The measures WSS, LLR and CEPSTRAL DISTANCE increase with the increase of " α ". This result indicates that, the main lobe width of the analysis Window increases and contributes the unwanted noise for the noise corrupt signal which results degradation in the Speech Enhancement process.

It is important to note that Fig. 6(a)-6(d) shows steep variation in Objective measures is observed when the shape parameter is between 4 and 15 which indicates the shape of the analysis window 's frequency response plays significant role on the speech quality measures in terms of the above objective measures. It is also observed that, further increase beyond 15 in the shape parameter " α " there is no significant improvement in the objective intelligibility scores as a function of analysis window shape. This can be attributed by the fact that the window shape gives rise to the frequency response with main lobe width and side lobe levels are compliment to each other. Narrower level main lobe width will tend to increase the side lobe levels and vice versa. Hence the contribution of undesired spectral components due to the main lobe width will be compensated due to the reduced side lobe levels. Hence the overall improvement in the Objective measures is not observed as in the case of variation of the shape parameter values between 4 and 15. Based on objective intelligibility scores, it can be seen that the optimum window shape parameter for speech analysis is between 2 and 4. For speech applications based solely on the short-time magnitude spectrum, the Kaiser window with the above shape parameter is expected to be the right choice.

From Table.6.1(a)-6.4(d) and Table.6.5(a)-6.8(d), the comparison of six objective measures of speech quality is made for the case of signal contaminated with Car noise and Train noise at various SNRs of 0dB, 5dB, 10dB and 15dB and clean speech signals using the Hanning Window for six " α " values. In this analysis six Objective measures are evaluated for $\alpha=1, 2, 3, 4, 5, 6$ and the results are presented in Table.6.1(a)-6.4(d) and Table.6.5(a)-6.8(d), Considering the fact that, higher SNR, Seg-SNR and fwseg-SNR values give better quality where as WSS, Cep and LLR measures, lower values indicate a better quality. From the Table.6.1(a)-6.4(d) and Table.6.5(a)-6.8(d), shown, it can be noticed that all the measures are improved for Kaiser Window with shape parameter " α " less than 3 in contrast with Hanning windowed measure. From WSS measure it is observed that, a significant improvement is achieved with the proposed window scheme. Observing the results presented in the Tables, the proposed window method shown is able to remove the residual noise in better manner compared to the Hanning window method and observing all the results, one can have a judicious choice for " $\alpha=2.5$ " as optimum value and can be used for Speech Enhancement process for better results using Spectral Subtraction method.

6.1 Simulation Results

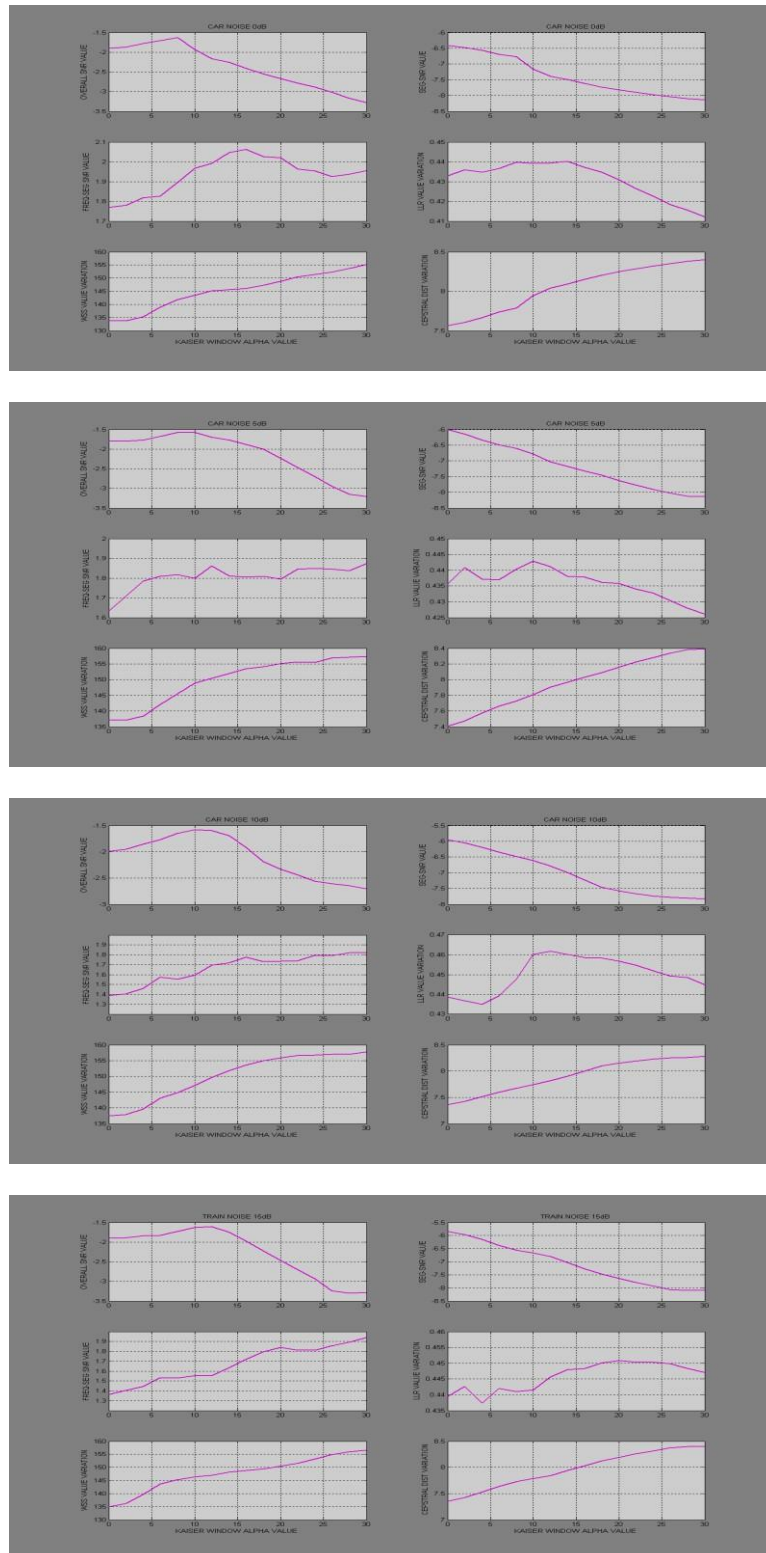


**Fig.6 (a): Variation of Objective Measures with Kaiser Window Shape Parameter “ α ”.
CAR-NOISE of 0dB, 5dB, 10dB and 15dB SNR Values**

**Table.6.: Variation of Objective Measures with Kaiser Window Shape Parameter “ α ”.
TRAIN-NOISE of 10dB SNR Value**

S.No	Objective Measure	Hanning Window	Kaiser window with different Alpha values					
			$\alpha=1$	$\alpha=2$	$\alpha=3$	$\alpha=4$	$\alpha=5$	$\alpha=6$
1	LLR	0.4552	0.4364	0.4386	0.4353	0.4384	0.4430	0.4478
2	WSS	140.5990	135.6660	135.8000	135.9970	137.4359	139.7408	141.2053
3	Cep	8.1183	7.7006	7.7599	7.8227	7.8865	7.9534	8.0190
4	Seg-SNR	-7.1574	-6.4890	-6.5805	-6.6543	-6.7413	-6.8404	-6.9481
5	Fwseg-SNR	2.1807	2.2305	2.1297	2.1990	2.2455	2.3011	2.1813
6	Overall-SNR	-1.8641	-1.8113	-1.8277	-1.7800	-1.7422	-1.7267	-1.7201

Conclusion

Speech signal is enhanced with the help of Spectral Subtraction method using Kaiser Window with suitable shape parameter “ α ”. It is found that the speech quality in terms of the Objective measures is improved by applying the Kaiser Window with optimum shape parameter “ $\alpha=2.5$ ”. From the figure.6, it is observed that the objective measure WSS has significant improvement for Kaiser Windowed signal when compared to Hanning Windowed signal. All the measures evaluated indicate that Kaiser Window with suitable shape parameter is superior to the Hanning Window in speech enhancement application using Spectral Subtraction method. It is also observed that the Window presented in this paper works out good for different types of noise, like babble, airport, street and noise, at different SNR levels. Hence it may be concluded that Kaiser Window with suitable shape parameter is an attractive option compared with Hanning Window for the Speech Enhancement using Spectral Subtraction method for better Speech quality and intelligibility.

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