IOSR Journal of Electrical and Electronics Engineering (IOSR-JEEE)
e-ISSN: 2278-1676,p-ISSN: 2320-3331, Volume 14, Issue 4 Ser. II (Jul. – Aug. 2019), PP 15-29
www.iosrjournals.org

Improving Speech Signal Quality in a Noisy Environment Using Fir Filter Designed With Kaiser Window

Mbachu, C. B[1], Akaneme S. A.[2] ,Okpagu P. E.[3]
[1, 2, 3] *Department of Electrical and Electronic Engineering, ChukwuemekaOdumegwuOjukwu University,Uli,Anambra State, Nigeria.*
Corresponding Author: Mbachu, C B

Abstract: *Processing voice signal has become very important in the present generation of communication because it is used in sending information across various locations of any distance on daily basis. But along its path of propagation or source of generation the signal can be contaminated by different noise components including Additive White Gaussian Noise (AWGN), Random Noise, power line noise, high and low frequency noise components. If these noise components are not eliminated, they degrade the integrity of the voice signal and render its information content unreliable. In this paper a FIR filter is designed with Kaiser Window using matlab to remove high frequency noise component in voice signal. The optimal parameters of the filter are sampling frequency of 44100Hz and filter order of 34. A real voice statement, "Education is the Key to the Development of any Nation" is converted into electrical voice signal using the system in-built microphone and recorded in windows media audio (.wma) format and stored in one of the files of the system. With "audioread" instruction the stored voice is loaded into a matlab edit window. The signal is contaminated with noise components of 4500Hz and above generated with matlab. Applying the contaminated signals to the filter gives a resulting output that confirms the effectiveness of the window in drastically reducing high frequency noise components from voice signals.*

Keywords: *Kaiser Window, voice signal, high frequency noise, power spectral density.*

Date of Submission: 03-08-2019 Date of Acceptance: 19-08-2019

I. Introduction

Widows are widely used in FIR filters for processing voice signals because they improve the quality of the signal by drastically reducing the noise components that degrade the quality of the voice signal. Some of the noise components include additive white Gaussian noise (AWGN), Random Noise, power line noise, high frequency and low frequency noise components. Some researchers have used Kaiser Window to reduce these noise components. Saseendra and Rajesh [1] implemented a low pass filter with Kaiser Window. The window is an adjustable window because the bandwidth of the main lobe and side lobe amplitude can be varied by changing the value of its adjustment parameter $\boldsymbol{\beta}$forfixed length of the window. The sampling frequency is 40800Hz, cutoff frequency of 10800Hz and filter order is 20 (21 filter taps) and three different values of $\boldsymbol{\beta}$ equalto 0.5, 3.5 and 8.5are applied in sequence to filter high frequency noise from audio signals. Simulation results show that FIR filter with $\boldsymbol{\beta}$=8.5 gives a better performance than the other three values of $\boldsymbol{\beta}$. In [2] Sangeetha and Kannan in using multirate signal processing for speech signals designed low pass and high pass FIR filters by different windowing techniques which included Kaiser Windowing. A voice signal of 8000Hz is recorded and stored as a wave file for use in a matlab. An additive white Gaussian noise (AWGN) is added to the speech to form a noisy speech signal. The noisy speech is filtered with the designed filters. Results show that each of the filters including the Kaiser Window-based filter provided a good performance. In [3] Pranab and Mohammand designed low pass, high pass and band pass filters to eliminate both low and high frequency noise components contained in human voice so as to provide high quality voice, using Kaiser Window and other windows. First the cutoff frequency for the low pass filter is made to be 3400Hz and the voice signal is applied to ascertain the effectiveness of the filter. Secondly the cutoff frequency of the high pass filter is made to be 600Hz and the voice signal applied to it to ascertain its effectiveness and thirdly the lower and upper cut of frequencies of the band pass filter are made to be 600Hz and 3400Hz and the voice signal applied to it to observe its filtration. Results show that each of these filters including the Kaiser-based filter eliminated the corresponding noise from the voice signal. Furthermore, the cutoff frequency of low pass, high pass and band pass were made variable in a DSP block to determine real time optimum cutoff frequencies. For the low pass the cutoff frequency was varied from 2.6KHz to 4KHz and the intensity of voice varies a little around 3400Hz. Similarly for a high pass filter the cutoff frequency was varied from 300Hz to 1000Hz and no significant change is observed around 600Hz. Variation of cutoff frequencies for the band pass gives a bandwidth that is between

600Hz and 3400Hz. The filters show sharp cutoff for removing high and low frequency components of noise from voice signal with non real time digital filters and real time digital filters.

No researcher has clearly shown any indication of applying the Kaiser window on voice signal of windows media audio (.wma) format which is a double column vector. In this paper therefore a Kaiser window will be used to remove high frequency noise component from a real time voice signal of '.wma' format.

II. Kaiser Window

The Kaiser Window function is [1, 4, 6, 7, 8]

$$W_k(\beta, n) = \frac{J_o\left[\beta\left[1-\left(\frac{2n}{M-1}\right)^2\right]^{1/2}\right]}{J_o\beta} \qquad (1)$$

$$\text{Where } -\frac{M-1}{2} \le n \le \frac{M-1}{2} \quad \text{Where } J_o(P) = 1 + \sum_{k=0}^{\infty}\left[(P/2)^k / k!\right]^2 \qquad (2)$$

J_o (x) is the modified Bessel function of thefirst kind of order zero [1, 6] and M, the length of the window. $\boldsymbol{\beta}$is a parameter that determines the shape of the window as shown in fig.1 and can be selected independently. The bandwidth of the main lobe and side lobe amplitude of the window can be varied by changing the value of $\boldsymbol{\beta}$for

fixed length of the window, thus making Kaiser Window an adjustable window function.

3. Design of Low Pass Digital FIR Filter With Kaiser Window

In this design Kaiser Window function is used on FIR filter. The function is depicted in fig. 1. Using the window on a FIR filter of order 34 implies that the corresponding window length is M=35.

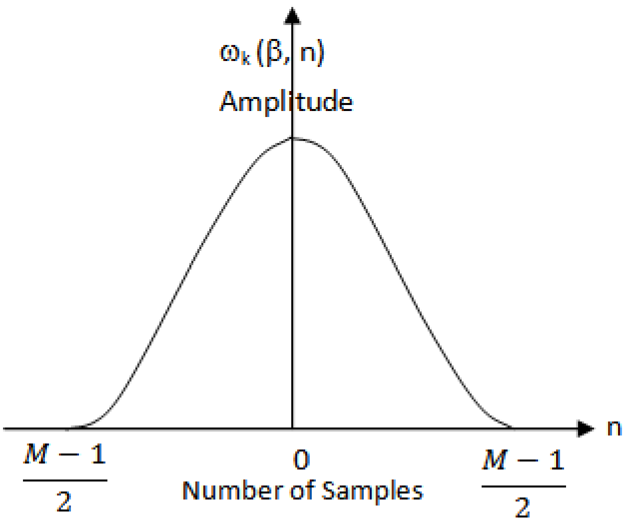

Fig.1: Kaiser Window Function

With specifications of filter order as 34, cutoff frequency as 3200Hz and sampling frequency as 44100Hz, six different values of height adjustment parameter $\boldsymbol{\beta}$**=** (**0.5, 3.5, 5.5, 8.5, 12** and **16**) are considered and in each value the impulse, magnitude and phase responses of the filter are obtained and are depicted below. The sampling frequency value chosen is because the voice signal in this circumstance is recorded in windows media audio (.wma) format. The impulse, magnitude and phase responses of the filter at different values of $\boldsymbol{\beta}$ are shown below.

3.1. Responses When β=0.5

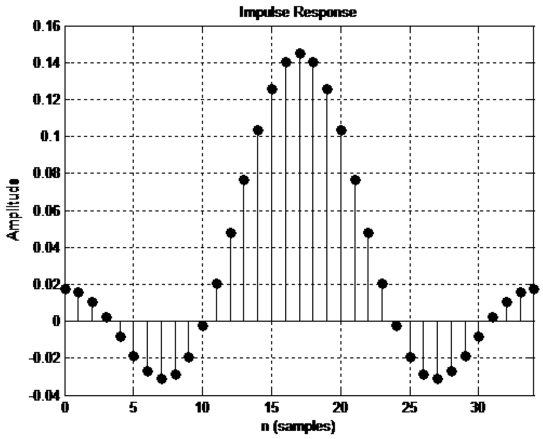

Fig 2a: Impulse Response When β=0.5

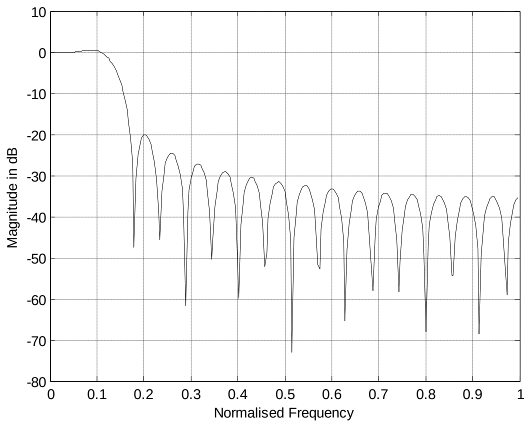

Fig 2b: Magnitude Response When β=0.5

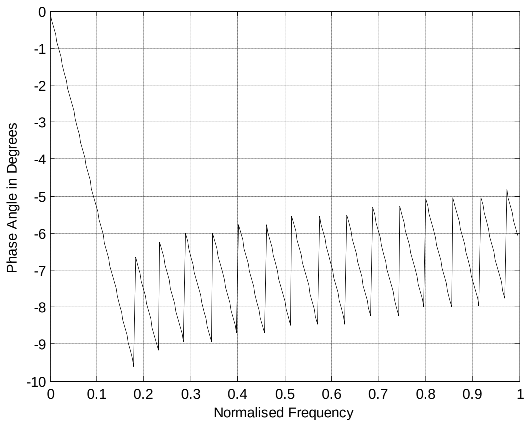

Fig 2c: Phase Response When β=0.5

3.2. Responses When β=3.5

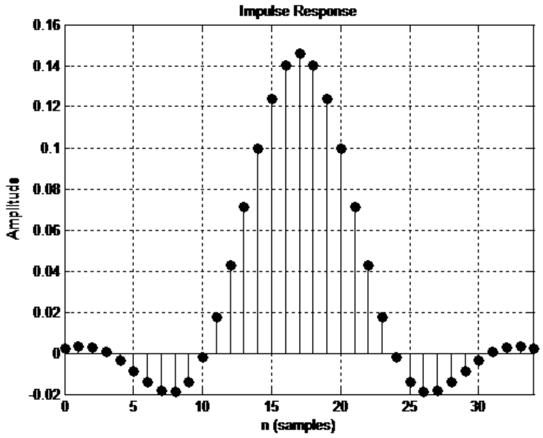

Fig 2d: Impulse Response When β=3.5

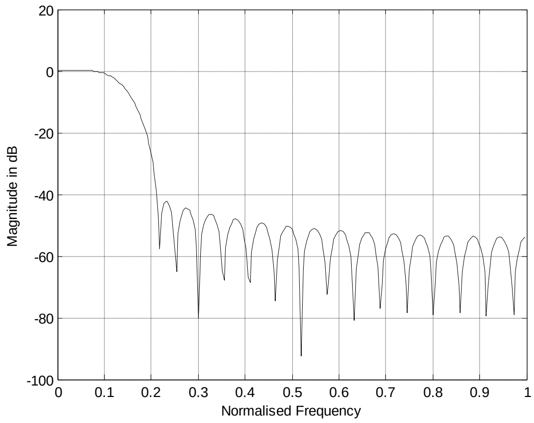

Fig 2e: Magnitude Response When β=3.5

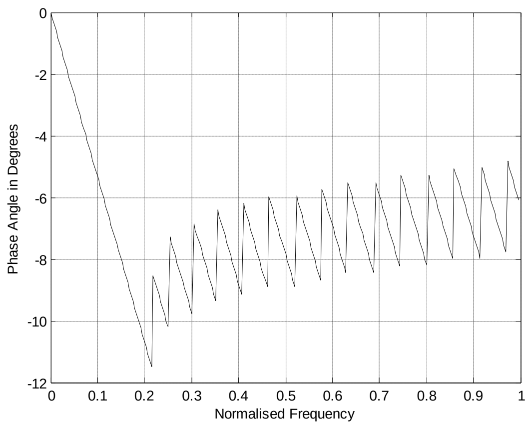

Fig 2f: Phase Response When β=3.5

3.3. Responses When β=5.5

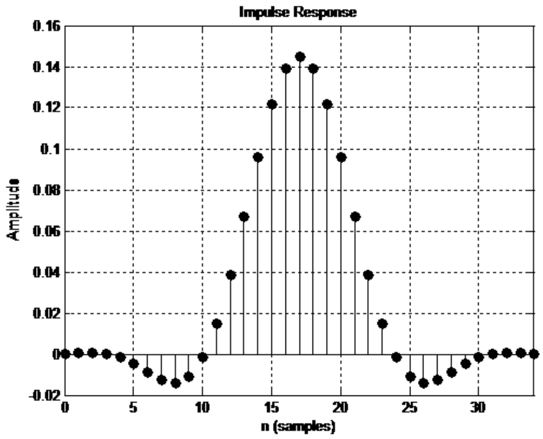

Fig 2g: Impulse Response When β=5.5

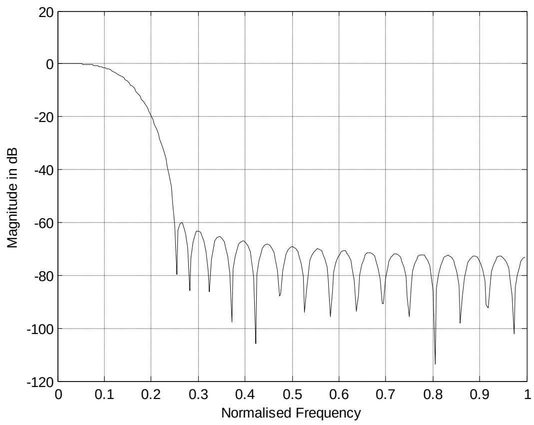

Fig 2h: Magnitude Response When β=5.5

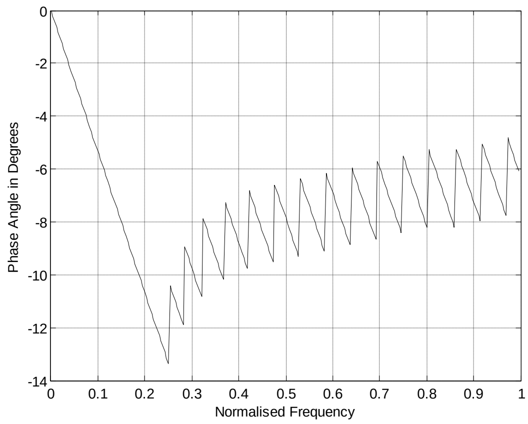

Fig 2i: Phase Response When β=5.5

3.4. Responses When β=8.5

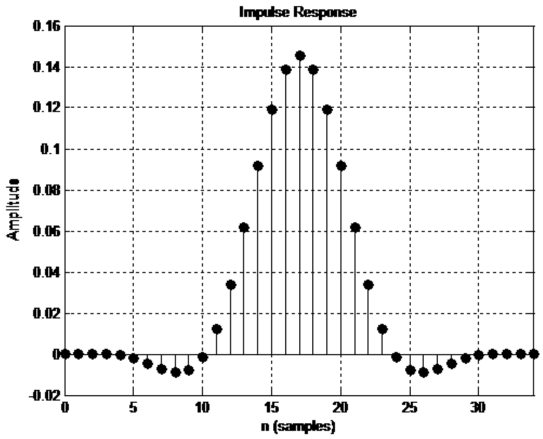

Fig 2j: Impulse Response When β=8.5

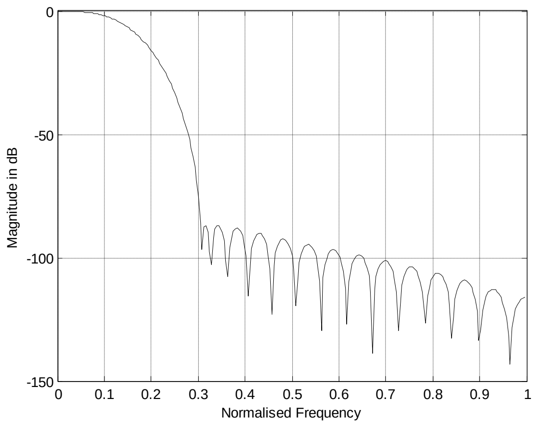

Fig 2k: Magnitude Response When β=8.5

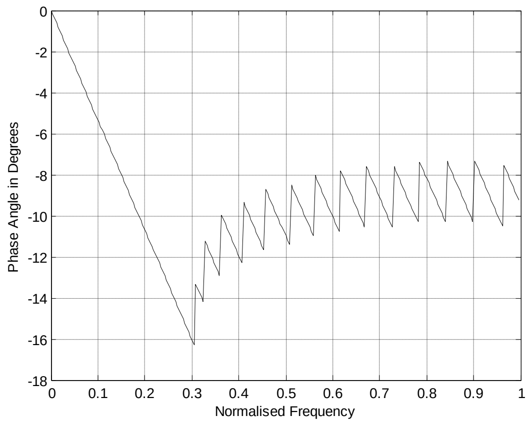

Fig 2l: Phase Response When β=8.5

3.5. Responses When β=12

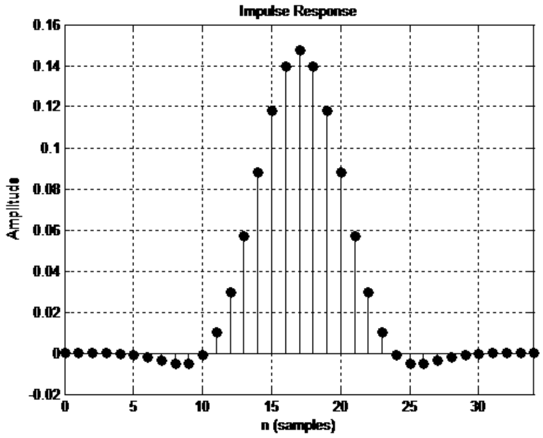

Fig 2m: Impulse Response When β=12

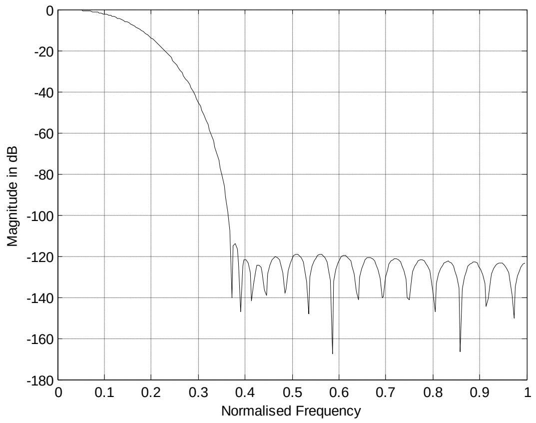

Fig 2n: Magnitude Response When β=12

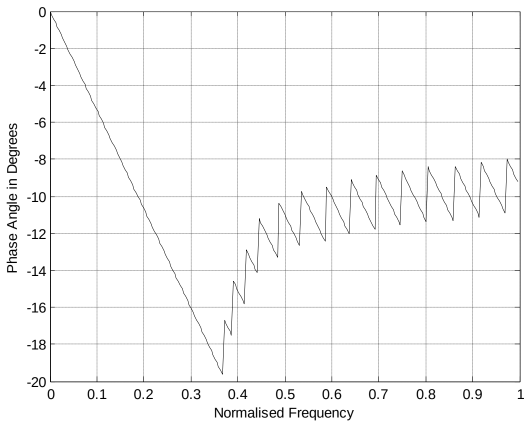

Fig 2p: Phase Response When β=12

3.6. Responses When β=16

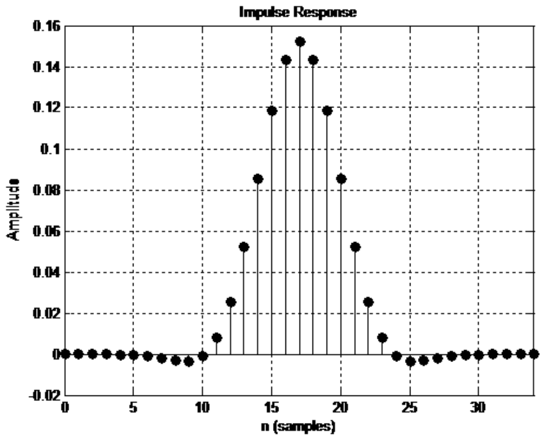

Fig 2q: Impulse Response When β=16

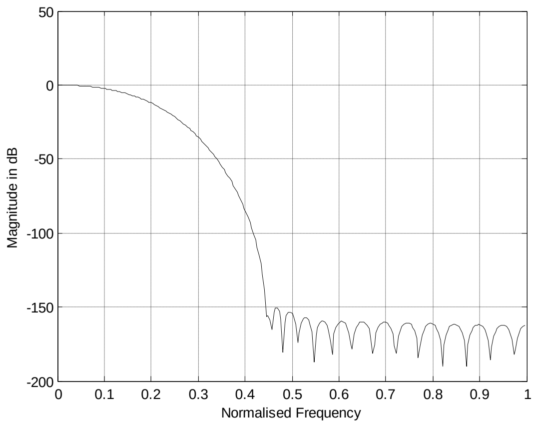

Fig 2r: Magnitude Response When β=16

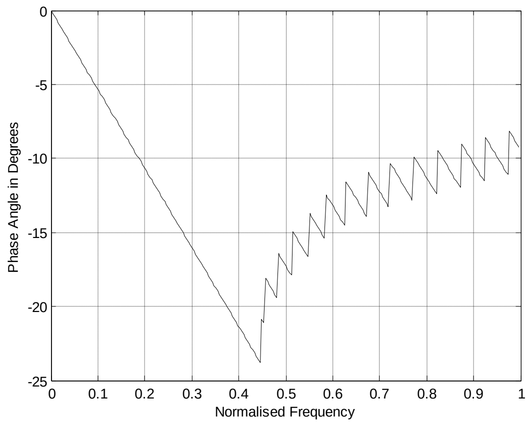

Fig 2s: Phase Response When β=16

Analysing the responses above it can be seen that all the impulse and magnitude responses exhibit stability in that there are no sustained oscillations while the phase responses exhibit clear linearity within the required frequency range. It can also be seen that as β increases the width of the main lobes of the magnitude responses increase and the attenuation levels increase.

III. Results

In order to ascertain the quality of the designed filter a real voice statement "Education is the Key to the Development of any Nation" is converted into electrical voice signal using the system in-built microphone and recorded in windows media audio (.wma) format and stored in one of the files of the system. The signal is transferred to a matlab edit window using "audioread" instruction. In the edit window a noise component of 4500Hz and above is added to the voice to constitute noise-corrupt voice signal. The corrupt-free voice signal is shown in fig.3 while fig.4 depicts the noise component, and the noise-corrupt voice signal depicted in fig.5. The noise-corrupt voice signal is filtered with each of the designed low pass filters and the outputs recorded. Figures 6, 7, 8, 9, 10 and 11 show the voice signal after filtering. Comparing the clean voice signal of fig.3, the noise-corrupt voice signal of fig.5 and the filtered voice signals of fig6 to fig.11 it can be seen that the filter reducedthe noise contained in the noise-corrupt voice for each value of $\boldsymbol{\beta}$. The best value of $\boldsymbol{\beta}$ in this circumstance can be determined from power spectral analysis of the filtered signals from fig.12 to fig.19 below

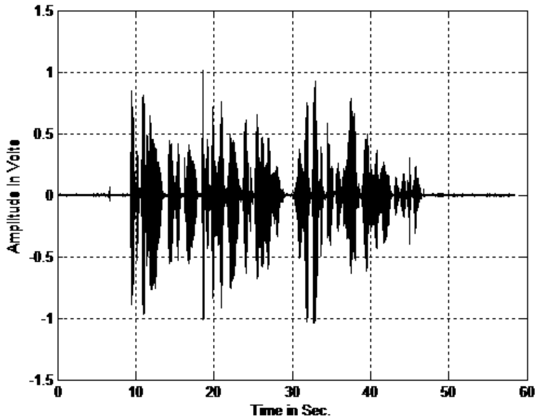

Fig. 3: Noise Free Voice Signal

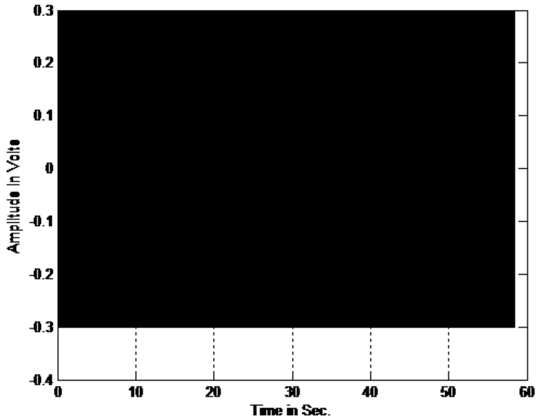

Fig. 4: Noise Signal

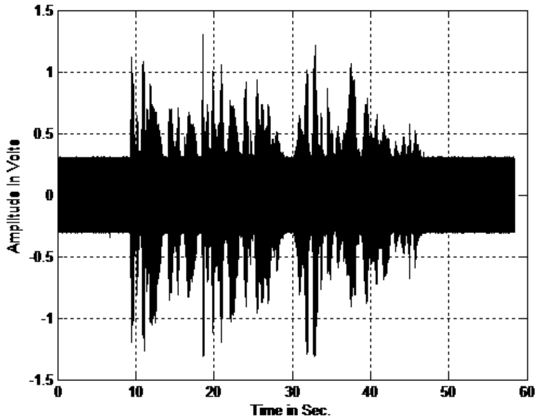

Fig. 5: Noisy Voice signal

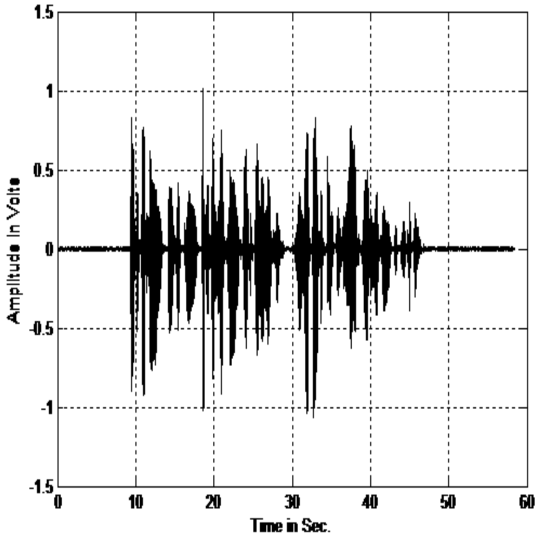

Fig.6: Voice Signal Filtered Low Pass Filter When $\boldsymbol{\beta}$=0.5

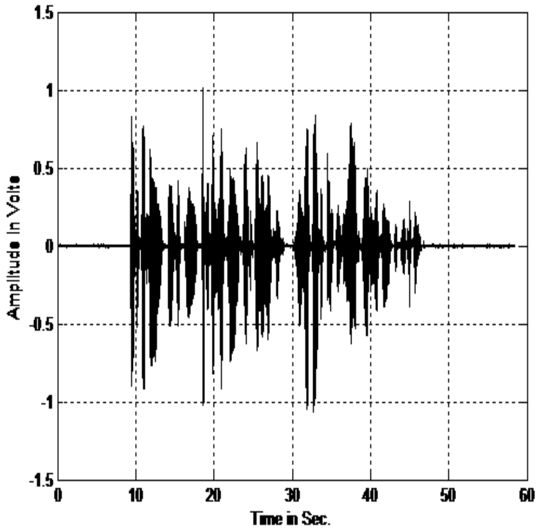

Fig.7: Voice Signal Filtered With Low Pass Filter When $\boldsymbol{\beta}$=3.5

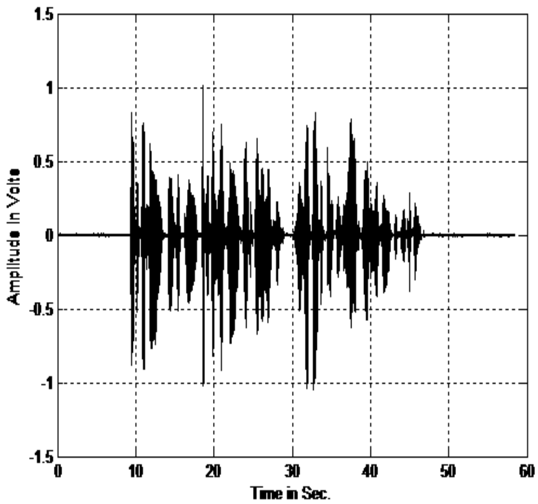

Fig.8: Voice Signal Filtered With Low Pass Filter When $\boldsymbol{\beta}$=5.5

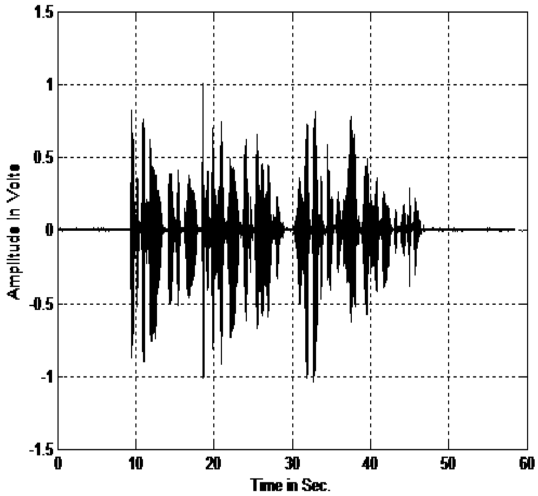

Fig.9: Voice Signal Filtered With Low Pass Filter When$\boldsymbol{\beta}$**=8.5**

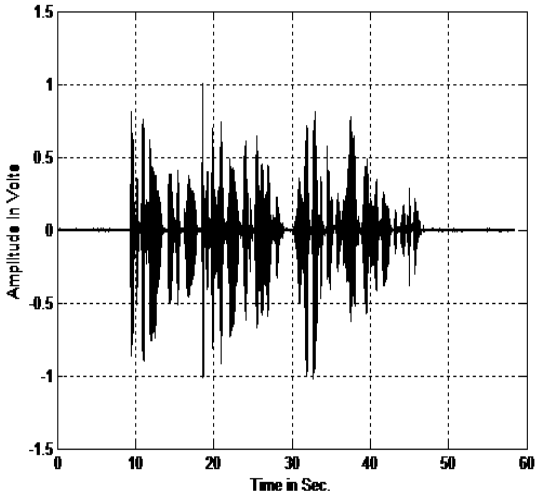

Fig.10: Voice Signal Filtered With Low Pass Filter When $\boldsymbol{\beta}$=12

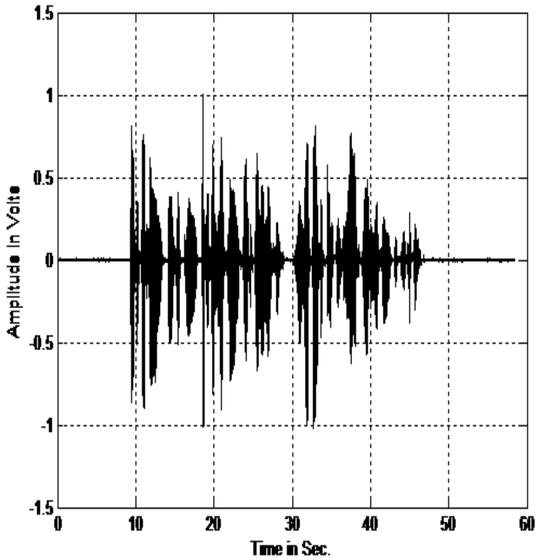

Fig.11: Voice Signal Filtered With Low Pass Filter When $\boldsymbol{\beta}$=16.

4.1. Signal Power Level

The performance of the filters can be analysed by considering the power levels of the filtered signals [9, 10]. Fig. 12 is the spectral density of a noise-free voice signal while the spectral density of the noise-corrupt voice signal is depicted in fig. 13. Fig.14 to fig.19are the power spectral densities of the filtered voice signal as β changes values.Choosing a normalised frequency of 0.875 for the analysis, table 1 below shows the summary of the power levels in dB of the filtered signals as β varies.

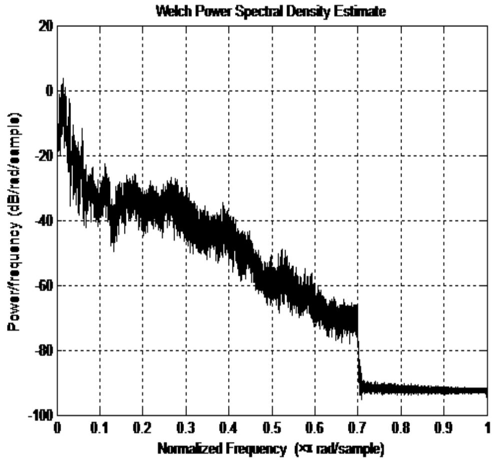

Fig.12: Power Spectral Density of Noise Free Voice Signal

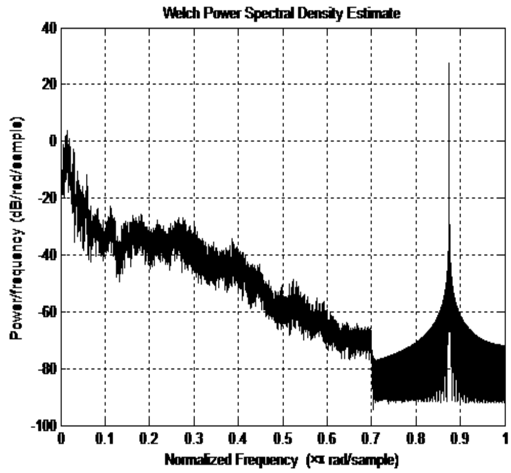

Fig.13: Power Spectral Density of Unfiltered Voice Signal

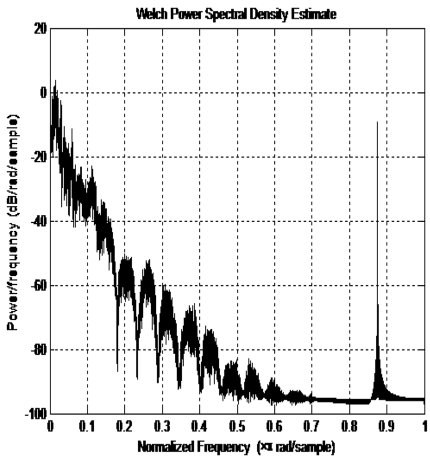

Fig.14: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When β=0.5

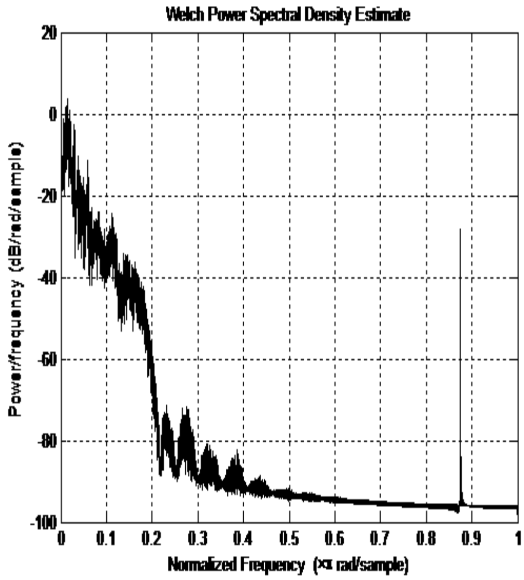

Fig.15: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When β=3.5

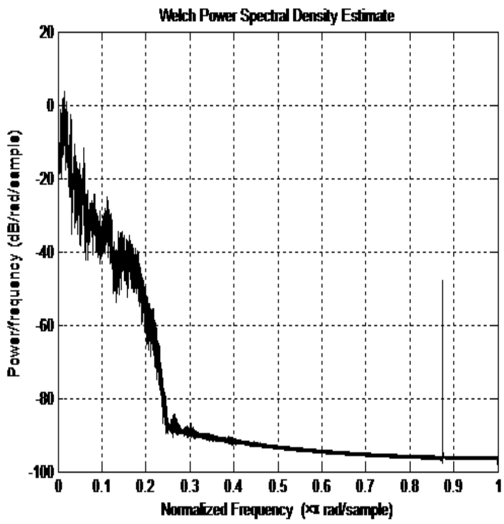

Fig.16: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When β=5.5

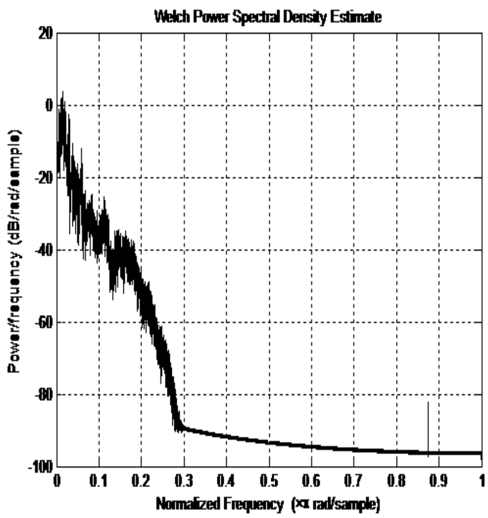

Fig.17: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When β=8.5

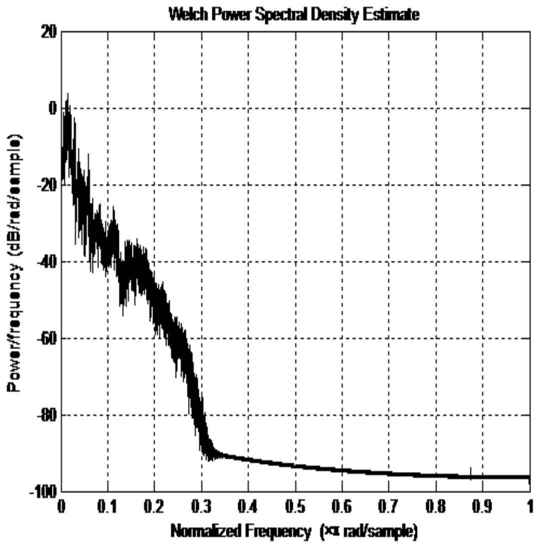

Fig.18: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When β=12

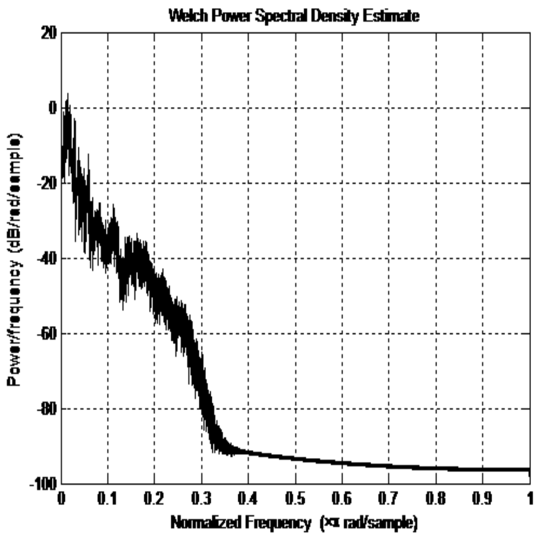

Fig.19: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When β=16

Table 1: Power Levels of Voice Filtered Signalsat 0.875 Normalised Frequency

Power level of unfiltered voice	+27.37dB
Power level of filtered voice when α=0.5	-9.35dB
Power level of filtered voice when α=3.5	-28.21dB
Power level of filtered voice when α=5.5	-47.86dB
Power level of filtered voice when β=8.5	-82.23dB
Power level of filtered voice when β=12	-93.76dB
Power level of filtered voice when β=16	-95.87dB

From table 1, power level of the noise-corrupt voice signal is +27.37dB. Comparing it with the corrupt-free voice signal power level of -91.19dB which can be determined from fig. 12 shows that noise signal added a lot of noise power of 27.37-(-91.19)=118.56dB. It can also be seen from table1 that the signal power increases asβincreases and some fraction of noise showed up in a decreasing quantity until at$\boldsymbol{\beta}$**=16** where the noise completely disappeared. It can therefore be concluded that the best value of the adjustment parameter β =16 and the corresponding power level is -95.87dB.

IV. Conclusion

It can be concluded that Kaiser window is a very effective window in designing FIR filters for voice signal processing. For the voice signal, the noise type and level in this circumstance, the best value of $\boldsymbol{\beta}$is 0.1. This value may vary if a different type of signal, noise type or window length is under consideration.

References

[1]. Saseendran, T.K. and Rajesh Mehra. Kaiser Window Based 21 Tap FIR Filter for Audio Applications. International Journal of Advanced Engineering Research and Technology (IJAERT),2014, pp.99-103.
[2]. S. Sangeetha and P. Kannan. Design and Analysis of Digital Filters for Speech
[3]. Signals Using Multirate Signal Processing. ICTACT Journal on Microelectronics, Vol. 03, Issue 04, January, 2018, pp. 480-486.
[4]. Pranab Kumar Dharand Mohammand Ibrahim Khan. Design and Implementation of Non Real Time Digital Filters for Audio Signal Processing. Journal of emerging Trends in Computing and Information Science, Vol.2, No.3, 2011, pp. 149-155.
[5]. Poulami Das, Subhas Chandra, Sudip Kumar and Sankar Narayan. An Approach for Obtaining Least Noisy Signal Using Kaiser Window and and Genetic Algorithm. International Journal of Computer Applications, Vol. 150, No. 6, September, 2016, pp. 16-21.
[6]. Chinckhede K. D., GovindSharanYadav, Hirekhan S. R. and Solanke D. R. (2011). On the Implementation of FIR Filter with Various Windows for Enhancement of ECG Signal. International Journal of Engineering Science and Technology, Vol. 3, No. 3, pp. 2031 – 2040.
[7]. Sharma S. Book on Digital Signal Processing with Matlab Programs, 2009.
[8]. Akanksha Raj, ArshiyanazKhateeb and FakrunnisaBalaganur. Design of FIR Filter for Efficient Utilisation of Speech Signal.International Journal for Scientific Research and Development, Vol. 3, Issue 3, 2015.
[9]. Mbachu C. B. Performance Analysis of Various Windows in the Reduction of Powerline Interference in ECG Signal. International Journal of Engineering Technology-UK, Vol., 5, No. 1, 2015, pp. 18-25.
[10]. Mbachu C. B. and Nwosu A. W. Performance Analysis of Various Infinite Impulse Response (IIR) Digital Filters in the Reduction of Powerline Interference in ECG Signals. International Journal of Scientific and Engineering Research, Vol. 5, Issue 11, November, 2014, pp. 498-506.
[11]. Marc Karam, Hasan F. Khazaal, HeshmatAglan and Cliston Cole. Noise Removal in Speech Processing Using Spectral Subtraction.Joiurnal of Signal and Information Processing, Vol. 5, 2014, pp. 32-41.

IOSR Journal of Electrical and Electronics Engineering (IOSR-JEEE) is UGC approved Journal with Sl. No. 4198, Journal no. 45125.

Mbachu, C B. "Improving Speech Signal Quality in a Noisy Environment Using Fir Filter Designed With Kaiser Window." IOSR Journal of Electrical and Electronics Engineering (IOSR-JEEE) 14.4 (2019): 15-29.