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Removing Noise from Voice Signals Using Height Adjustable Triagular (Hat) Window Basedfirdigital Filters

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Abstract: *Passing message by human voice is the most effective means of communication for mankind. With advancement in technology, voice signals are now carried over transmission lines to send messages to remote ends where ordinarily the voice cannot reach. Unfortunately these voice signals are interfered with by unwanted signals from either the source or along these transmission lines. Such unwanted signalsinclude Additive White Gaussian Noise (AWGN), Random Noise, power line noise, low frequency and high frequency noise components. If these noise components are not removed, the integrity of the voice signal can be compromised and its messagecontent becomes unreliable. In this paper a FIR filter is designed with height adjustable triangular (HAT) window using matlab and deployed to filtering out high frequency noise 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. The signal is transferred to a matlab workspace using "audioread" instruction. A noise signal of 4500Hz and above is also generated with matlab and added to the voice signal. When thecontaminated signal is applied to the filter, result shows that the window is effective in removing high frequency noise from voice signals.*

Keywords: *HAT window, voice signal, high frequency noise, power spectral density.*

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I. Introduction

Using voice signal to pass information or message for mankind is a very effective means.Technology has so advanced that voice signals are now carried over transmission lines transmission lines include mobile communication, telephone communication, multimedia communication and public address system transmission, lines. Incidentally this voice signals can suffer interference from either the source or along these transmission lines by noise components such as additive white Gaussian noise [1], random noise, power line noise, high frequency and low frequency noise components, and some other form of noise components.

For the integrity of the message in the voice signals to be preserved at the receiving end, any interfering signal must be removed. The aim of this paper is to remove noise above the upper cut of frequency of voice signals using finite impulse response (FIR) digital filter. Normally the frequency range of voice signal is from about 200Hz to 3400Hz. In order to improve performance, FIR filters are modified with one widow or the other depending on exact purpose. Without windowing the FIR digital filter will tend to distort the voice signal on application because of the complex nature of the signal, due to differential phase shift the voice signal will suffer. Some researchers have used various windows to remove these noise signals.

In [2] Gopika and Supriya used Han, hamming, Blackman and Bartlett windows individually to design high pass filters filtering out low frequency noise from human voice with a sampling frequency of 8000Hz, cutoff frequency of 600Hz and transition width of 200Hz. A voice statement "I am Gopika" with a duration of 2.4 sec is recorded in a noisy environment after which it is applied to the individual four filters and the result of each observed. The result showed that each of the filters provided satisfactory performance. In [3] Ritesh and Rajesh designed a low pass FIR filter individually with hamming and hanning windows to remove high frequency noise from audio signals. Three different noisy signals were analysed and hanning window provided better results compared to hamming window.Authors in [4] demonstrated how hamming, hanning and Blackman windows can be used to analyse speech signals. The authors designed low pass and high pass filters with windows to remove high frequency noise and low frequency noise respectively from speech signals with a sampling rate of 22050, number of bits per sample of 16 and order of 64. The Blackman window out-performs among the three windows.Rajput and Bhadauria [5] in filtering speech signal modified low pass filters with hanning, hamming Blackman and adjustable generalised window functions. The generalised window function is shown in (4) with adjustment parameter of **α**=0.07. The order of the filters is 31, sampling frequency, 8000Hz and cutoff frequency,1200Hz. The result shows that the low pass filter designed with each of the windows significantly reduced high frequency components above 1200Hz. This implies that low pass filters for removing high frequency components in speech signals can be designed with hanning, hamming, Blackman or adjustable window function of (4) with **α**=0.07.In another publication [6] Rajput and Bhadauria used another adjustable window function as shown in (5) to design low pass filter for filtering speech signals. Four different values of the adjustment α were used and the performance observed. The values for adjustment used are 0.71, 0.78, and 0.5 (hanning window) and 0.54 (hamming window) while the sampling frequency is 16000Hz, filter order, 33 and cutoff frequency, 3200Hz. The result showsthat each of the windows is able to significantly reduce high frequency components above 3200Hz, which makes them suitable for use in speech signal filtrations.It was showed in[7] that rectangular window can be used to design FIR low pass filter for noise reduction in speech signals. The filter has a cutoff frequency of 5000Hz, transition band of 300Hz and sampling frequency of 10 KHz. A high frequency noise is generated with matlab and used to corrupt a speech signal and the corrupt signal applied to the low pass filter. The filter significantly reduced the noise signal. Pranab and Mohammand [8] designed low pass, high pass and band pass filters with rectangular, triangular, Kaiser, hamming, hanning and Blackman windows for processing audio signals confined within voice signal frequency range. The cut off frequency for the high pass filter is 600Hz and that for low pass filter is3400Hz. The upper and lower cutoff frequencies for the band pass filter are 600Hz and 3400Hz respectively. Results indicate that these filters provide sharp cut off for removing both higher and lower frequency components of voice signal. In [9] Saseendra and Rajesh used Kaiser Window to design low pass filter for audio applications. They used three different values of the window adjustment parameter **β (that's β=0.5, 3.5 and 8.5)**. The order of the filter is 20, cutoff frequency, 10800Hz and sampling frequency, 40800Hz. An audio signal is first recorded in a wave format and loaded into matlabat a default frequency of 8192Hz for the input signal. The filter is used to remove higher frequencies above the cutoff frequency for each value of **β**. The simulation result shows that the FIR filter with **β=8.5** is better in performance than other values of **β**.Sangeetha and Kannan [10] in using multirate signal processing for speech signals designed low pass and high pass FIR filters by different windowing techniques such as hamming, hanning, Blackman, Rectangular and 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 provided a good performance.

The performance of these windows in processing voice signals are found to be satisfactory to a large extent but no researcher has used height adjustable triangular (HAT) window to process voice signals. In this work therefore HAT window will be used to design FIR low pass filter for removing high frequency components from voice signals.

II. Adjustable Windows

The desired frequency response of the ideal filter can be denoted by $H_d\left(e^{jw}\right)$. If it is a Fourier transform of desired impulse response $h_d(n)$ of the filter, then [11, 12]

$$H_d\left(e^{jw}\right) = \sum_{n=-\infty}^{\infty} h_d(n) e^{-jwn} \qquad (1)$$

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d\left(e^{jw}\right) e^{jwn}\, dw \qquad (2)$$

From (2) it can be seen that the desired impulse response $h_d(n)$ is infinite and as such any filter designed with it isunrealizable. Therefore the best approach to make the filter realizable is to truncate the impulse response at a desired length and smoothen it to prevent distortions that will arise from filters designed with such impulse response because of the sudden truncation. The function required to do the smoothening is called window function w(n). The required impulse response h(n) is obtained from the dot product of $h_d(n)$ and w(n) [13] as presented in (3)

$$h(n)=h_d(n).w(n) \qquad (3)$$

There are different types of windows which can be fixed or adjustable. The window is fixed when any part of it cannot be altered or varied whereas if any part of it such as shape or amplitudecan be varied by varying an adjustment parameter the window is said to be adjustable.

A. Generalised Adjustable Window

Two generalised adjustable windows are presented as [5] in (4) and as [6] in (5)

A1. $w(n)=\frac{1-\alpha}{2} - 0.5\cos\left(\frac{2\pi n}{M-1}\right) + \frac{\alpha}{2}\cos\left(\frac{4\pi n}{M-1}\right), 0 \le n \le M-1$

(4)

A2. $w(n)=\alpha-(1-\alpha)\cos\left(\frac{2\pi n}{M-1}\right), 0 \leq n \leq M-1$

(5)

where the adjustment parameter **α**varies between 0 and 1 in both of them. In (4) if α=0.16 it becomes Blackman window and hanning window if α=0 both of which are fixed windows.Also in (5) if α=0.5, it becomeshanningwindow and hamming window if α=0.54 both of which are equally fixed windows.We can easily observe the difference between **A1** and **A2** which is only in terms of cosine term.

B. Non generalized Adjustable Window

Three non generalised adjustable windows presented here are Kaiser Windowas in (6) and (7), height adjustable triangular (HAT) windowas in (8) and height adjustable sine(HAS) window as in (9).

B1. Kaiser Window

The Kaiser Window function is [9, 14, 15, 16]

$$W_k(\beta, n) = \frac{J_o\left[\beta\left[1-\left(\frac{2n}{M-1}\right)^2\right]^{1/2}\right]}{J_o\beta} \qquad (6)$$

$$\text{Where} -\frac{M-1}{2} \leq n \leq \frac{M-1}{2} \text{ Where } J_o(P) = 1+\sum_{k=0}^{\infty}\left[(P/2)^k/k!\right]^2 \quad (7)$$

J_o (x) is the modified Bessel function of the first kind of order zero [9, 15,17]. M is the length of the window. **β**is a parameter that determines the shape of the window and can be selected independently.

B2. HAT Window

$$w(n) = \begin{cases} \alpha+(2-2\alpha)n/(M-1), \ 0 \leq n \leq \frac{M-1}{2} \\ 2-[\alpha+(2-2\alpha)n/(M-1)], \ \frac{M-1}{2} \leq n \leq M-1 \end{cases}$$

(8)

wherethe adjustment parameter**α** varies from 0 to 1.

B3. HAS Window

$$w(n) = \begin{bmatrix} \alpha+\sin\left[\frac{2\sin^{-1}(1-\alpha)}{L}n\right], 0 \leq n \leq \frac{M-1}{2} \\ \alpha+\sin\left[\frac{(L-n)2\sin^{-1}(1-\alpha)}{L}\right], \frac{M-1}{2} \leq n \leq M-1 \end{bmatrix} \qquad (9)$$

wherethe adjustment parameter**α** varies from 0 to 1. M is the length of the window and L=M-1.

III. Design of Low Pass Digital FIR Filter

In this design HAT 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 and for such length the HAT window of (8) becomes as in (10) below.

$$w(n) = \begin{cases} \alpha+(2-2\alpha)n/34, \ 0 \leq n \leq 17 \\ 2-[\alpha+(2-2\alpha)n/34], \ 17 \leq n \leq 34 \end{cases}$$

(10)

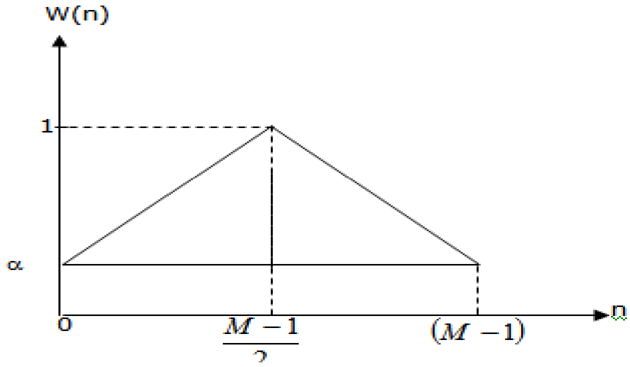

Fig. 1: Amplitude Response of HAT Window

With specification of filter order as 34, cutoff frequency as 3200Hz and sampling frequency as 44100Hz, five different values of height adjustment parameter **α=** (**0.00, 0.05, 0.1, 0.2** and **0.3**) are considered and in each value the impulse, magnitude and phase responses of the filter are obtained and are depicted below. Note that at **α= 0.0,** the function becomes a complete triangular windowfunction.The sampling frequency value chosen is because the voice being used here is recorded in windows media audio (.wma) format. The impulse, magnitude and phase responses of the filter at different values of **α** are shown below.

3.1. Responses When α=0.00

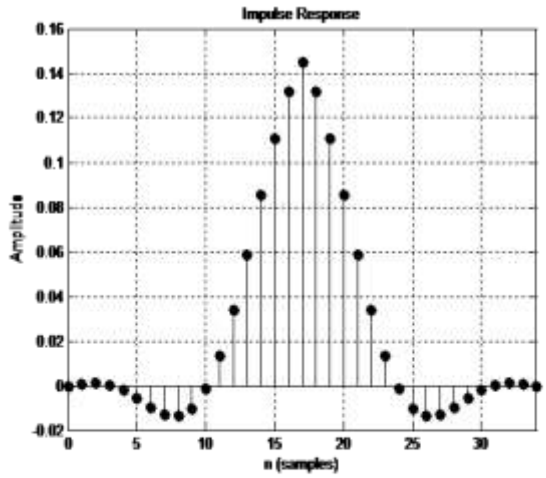

Fig 2a: Impulse Response When α=0.00

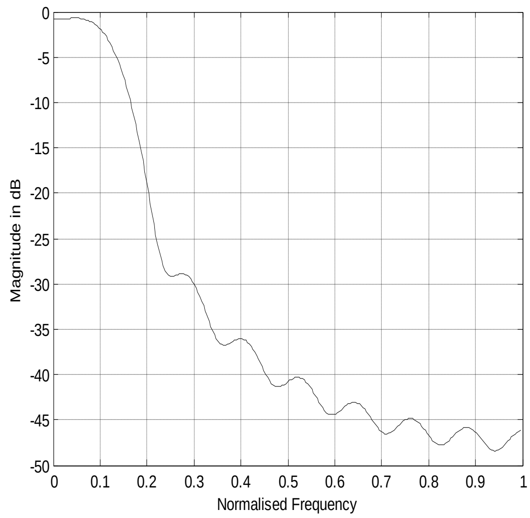

Fig 2b: Magnitude Response When α=0.00

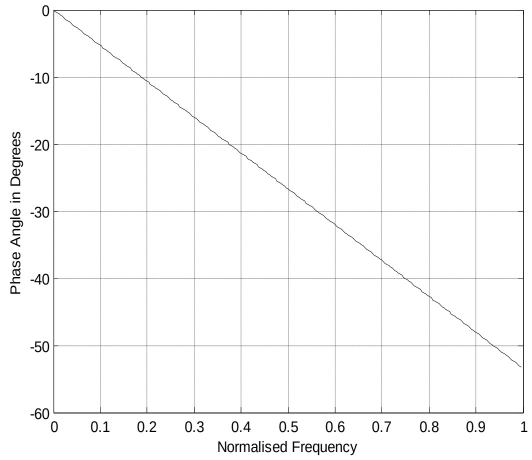

Fig 2c: Phase Response When α=0.00

3.2. Responses When α=0.05

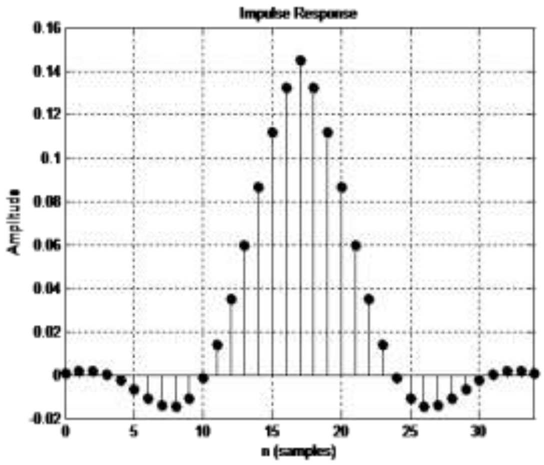

Fig 2d: Impulse Response When α=0.05

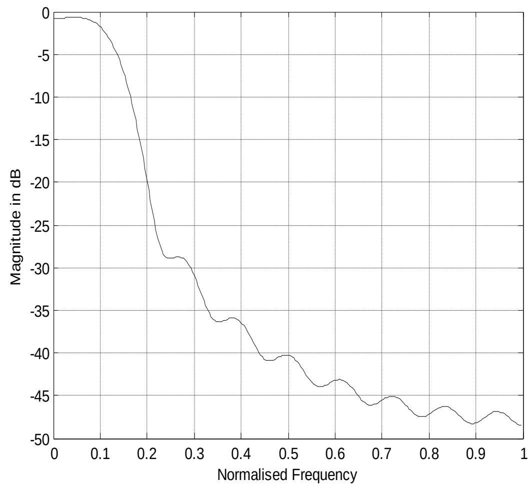

Fig 2e: Magnitude Response When α=0.05

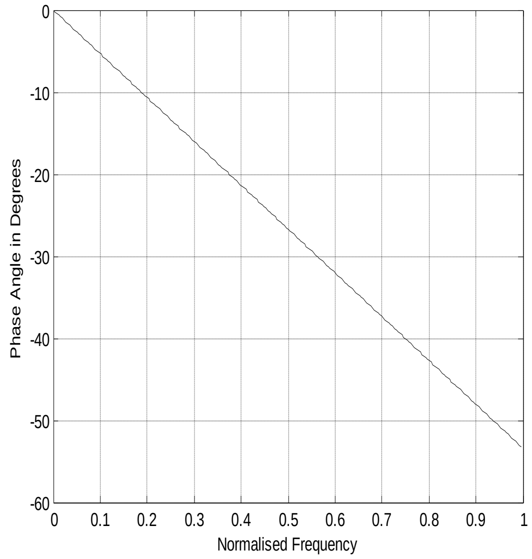

Fig 2f: Phase Response When α=0.05

3.3. Responses When α=0.1

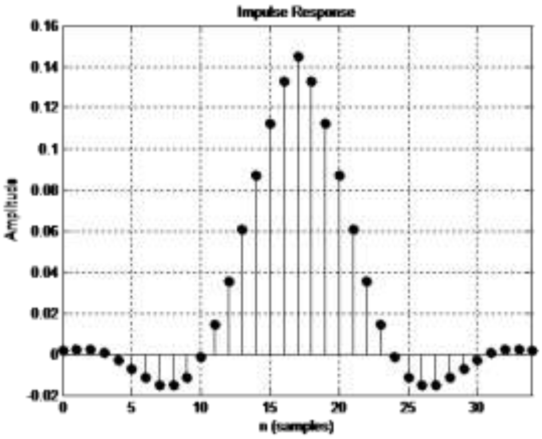

Fig 2g: Impulse Response When α=0.1

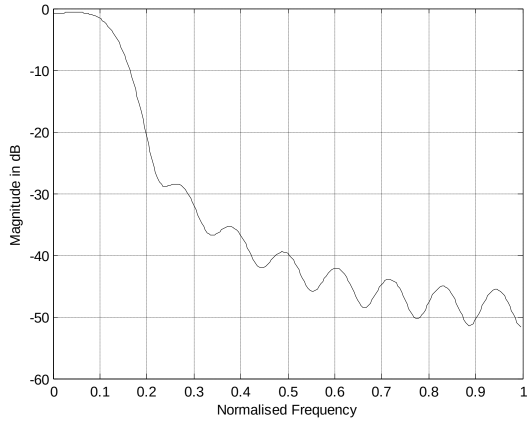

Fig 2h: Magnitude Response When α=0.1

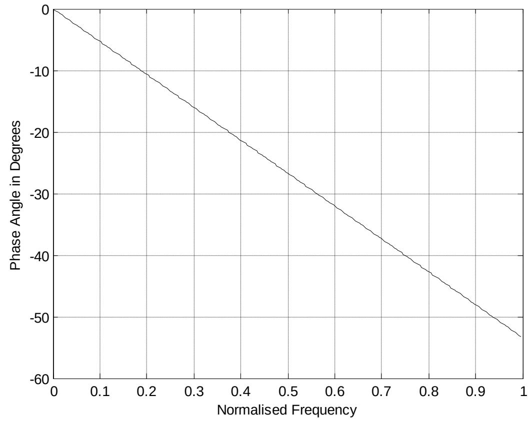

Fig 2i: Phase Response When α=0.1

3.4. Responses When α=0.2

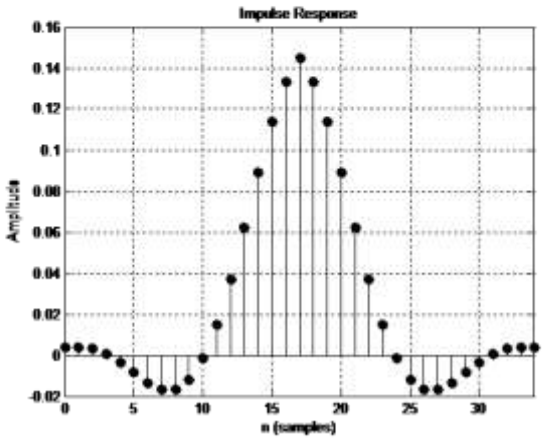

Fig 2j: Impulse Response When α=0.2

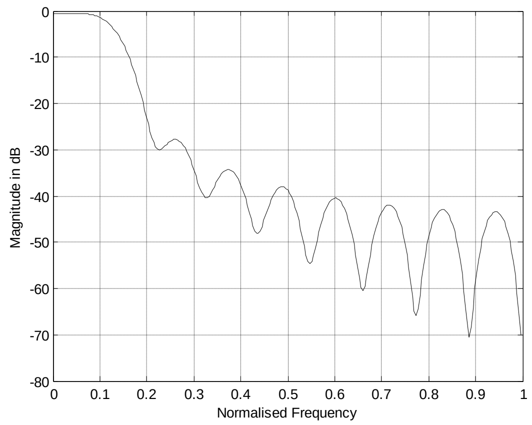

Fig 2k: Magnitude Response When α=0.2

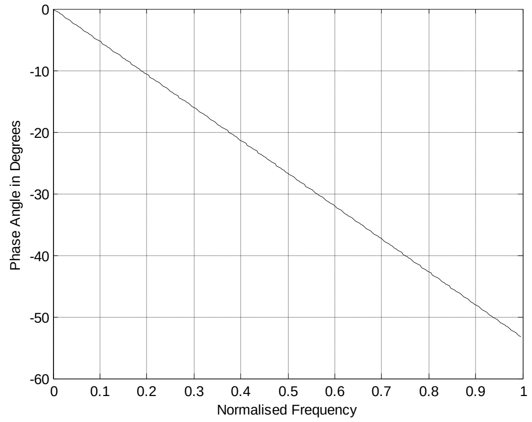

Fig 2l: Phase Response When α=0.2

3.5. Responses When α=0.3

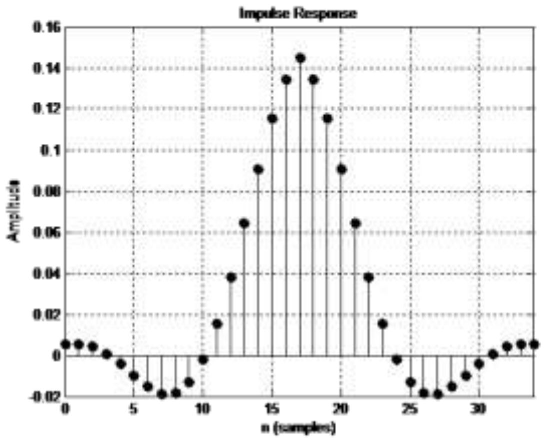

Fig **2m**: Impulse Response When α=0.3

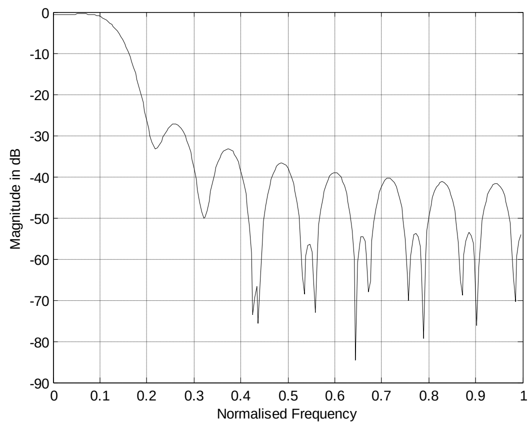

Fig 2n: Magnitude Response When α=0.3

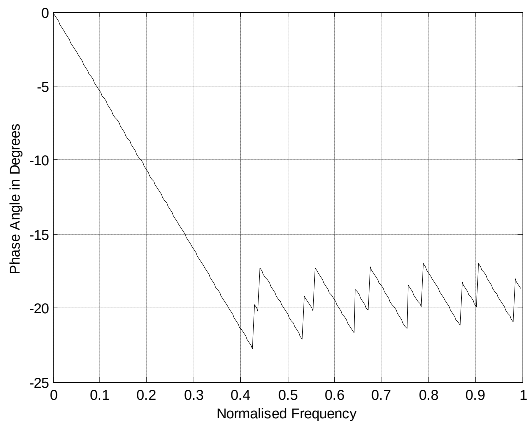

Fig 2p: Phase Response When α=0.3

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 linearity. However the phase response when **α=0.3** has started indicating some degree of non-linearity which will widen should **α** be increased further. Based on that the optimum value of **α**can be said tolie between 0.0 and 0.2.

IV. 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 environment using "audioread"instruction after which it is made noisy by adding sine wave of 4500Hz and aboveto it as noise. The voice signal is shown in fig.3 while fig.4 depicts the contaminating noise, and the noisy voice signal depicted in fig.5. The noisy voice signal is filtered with each of the designed low pass filters and the output recorded. Figures 6, 7, 8, 9 and 10 show the voice signal after filtering. Comparing the clean voice signal of fig.3, the noisy voice signal of fig.5 and the filtered voice signals of fig6 to fig.10 it can be seen that the filter largely removed the noise contained in the noisy voice signal. Also listening to the signals confirms the effectiveness of the filters.

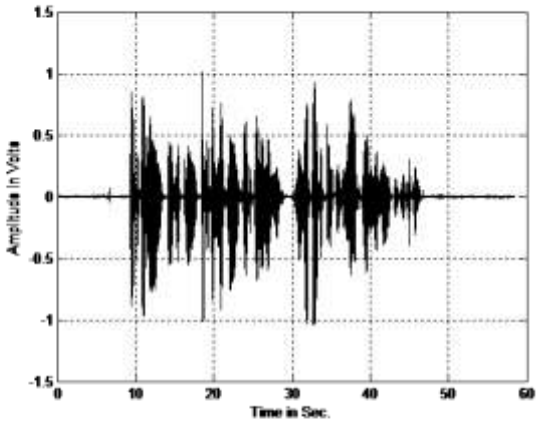

Fig. 3: Noise Free Voice Signal

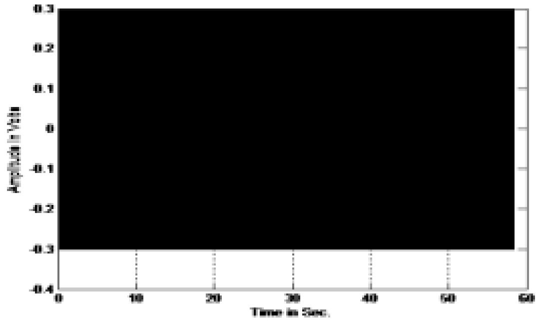

Fig. 4: Noise Signal

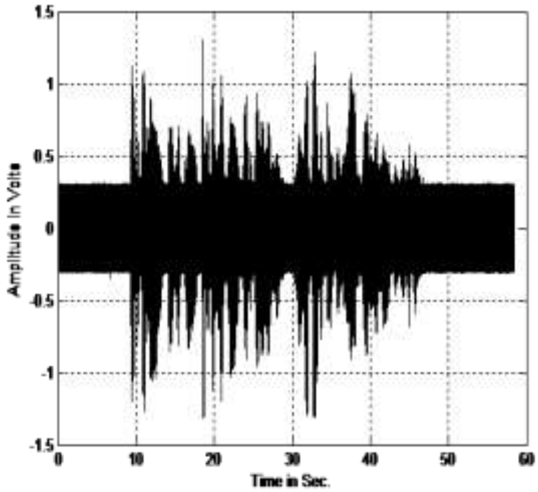

Fig. 5: Noisy Voice signal

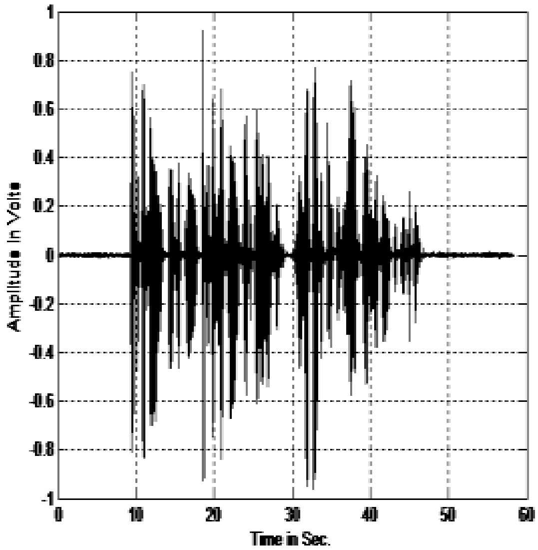

Fig.6: Voice Signal Filtered Low Pass Filter

When α=0.00

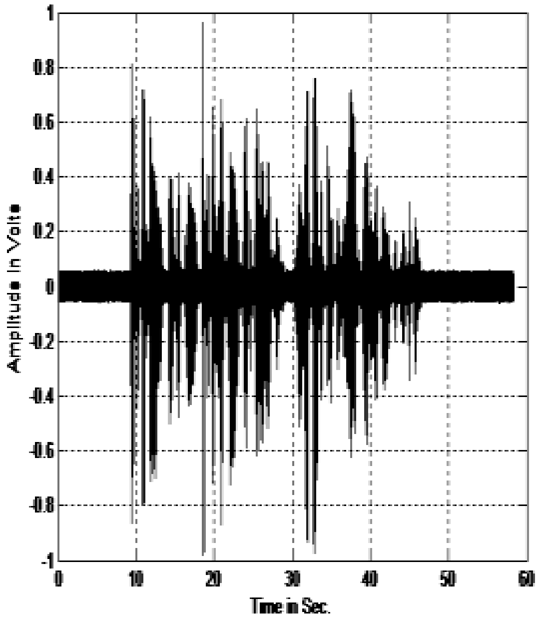

Fig.7: Voice Signal Filtered With Low Pass Filter When α=0.05

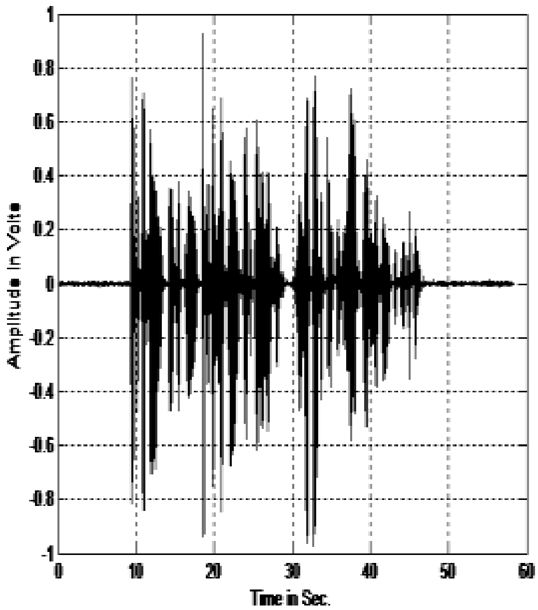

Fig.8: Voice Signal Filtered With Low Pass Filter When α=0.1

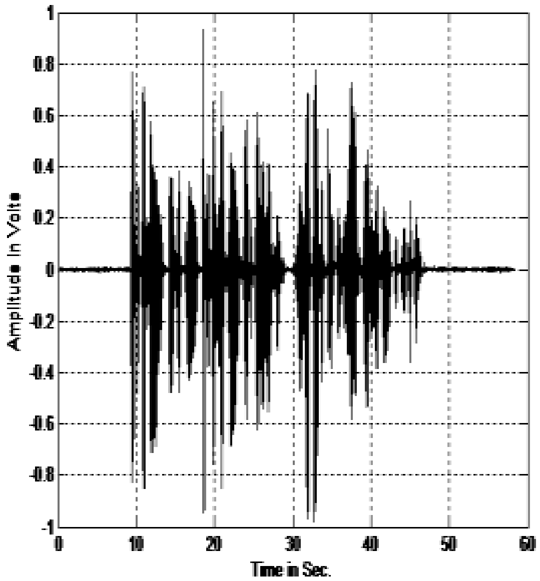

Fig.9: Voice Signal Filtered With Low Pass Filter When α=0.2

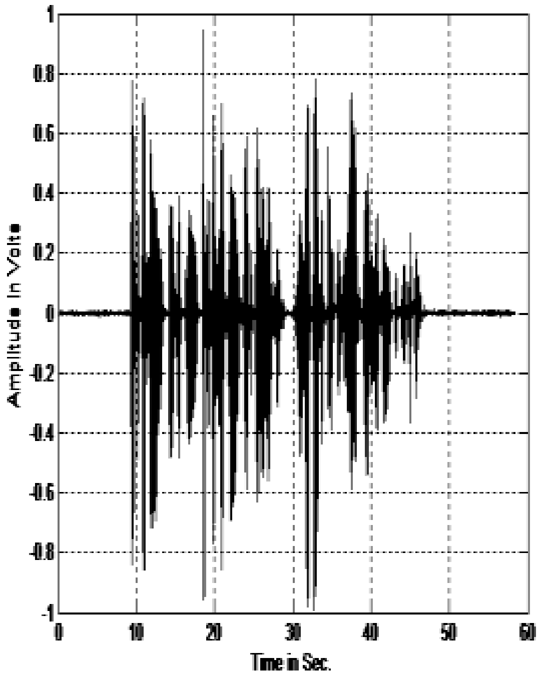

Fig.10: Voice Signal Filtered With Low Pass Filter When α=0.3

4.1. Signal Power Level

The performance of the filters can be analysed by considering the power levels of the filtered signals [18, 19]. Fig. 11 is the spectral density of a clean voice signal while the spectral density of the noisy voice signal is depicted in fig. 12. Fig.13 to fig.17 are the power spectral densities of the filtered voice signal at different values of **α.**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 at different values **α.**

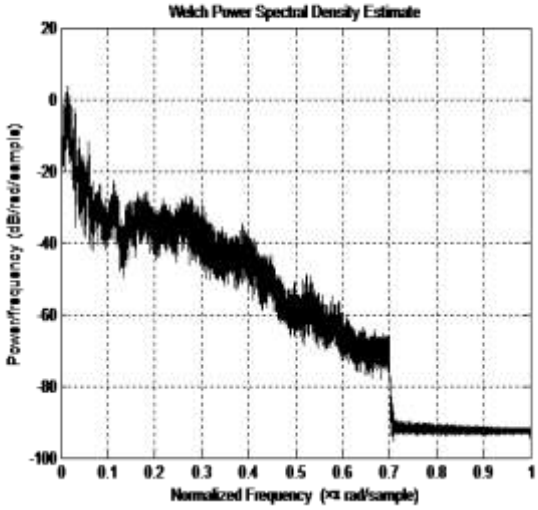

Fig.11: Power Spectral Density of Noise Free Voice Signal

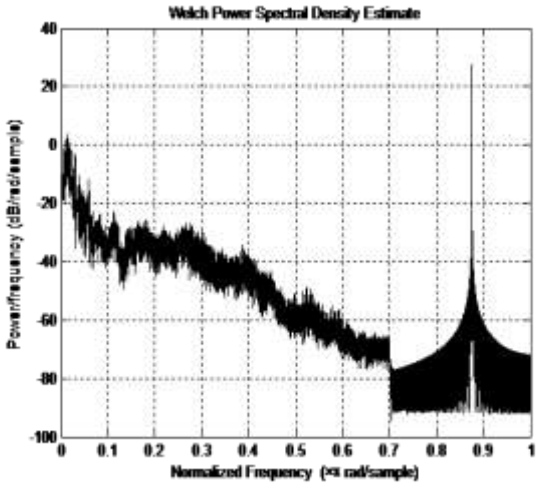

Fig.12: Power Spectral Density of Unfiltered Voice Signal

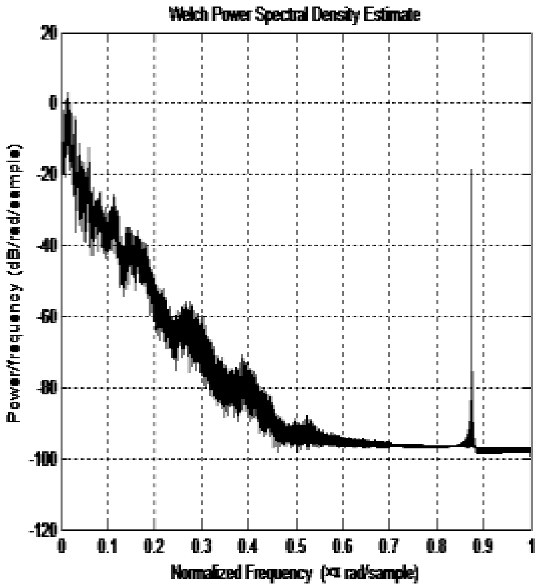

Fig.13: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When α=0.00

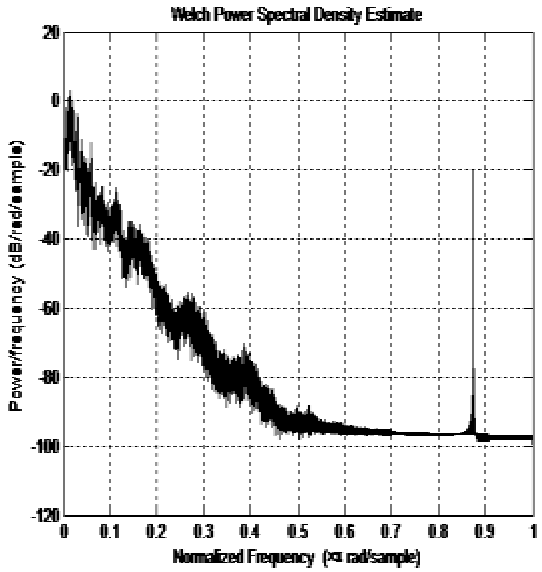

Fig.14: Power Spectral Density of Voice Signal Filtered With Low Pass Filter When α=0.05

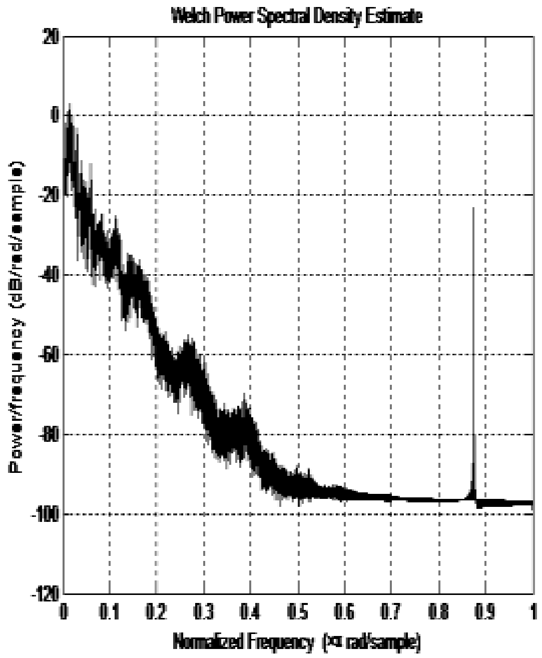

Fig.15: Power Spectral Density of Filtered

Voice Signal When α=0.1

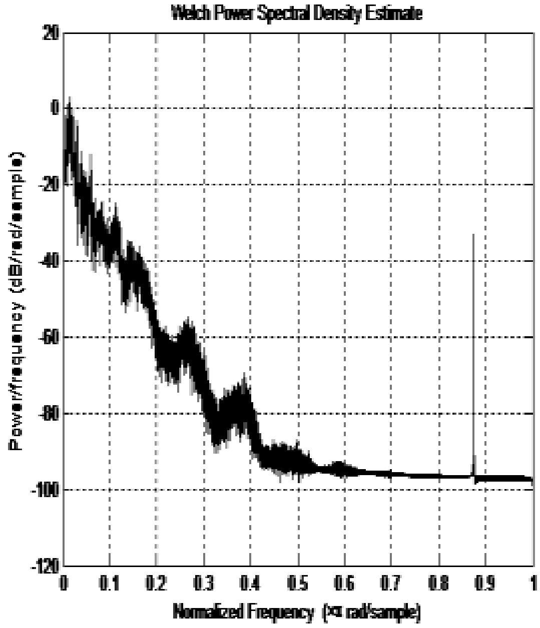

Fig.16: Power Spectral Density of Filtered

Voice Signal When α=0.2

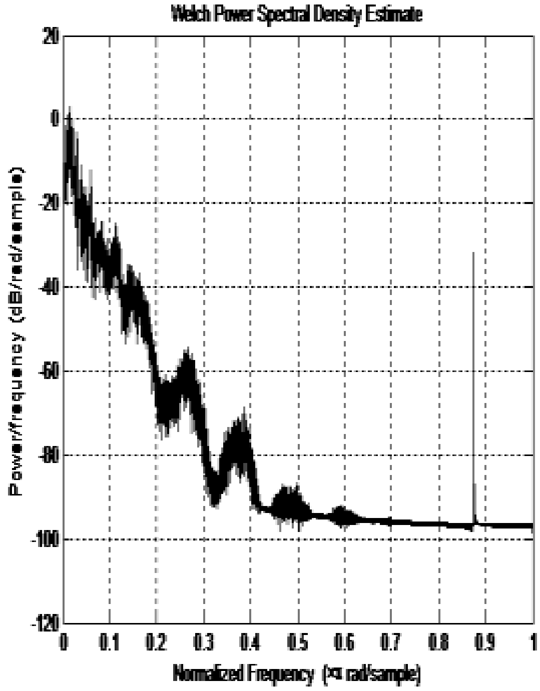

Fig.17: Power Spectral Density of Filtered

Voice Signal When α=0.3

From table 1, power level of the contaminated voice signal is +29dB. Comparing it with the clean voice signal power level of -91.55dB which can be deduced from fig. 11 shows that noise signal added a lot of noise power of 29-(-91.55)=121.07dB. It can also be seen from table1 that the signal power increases as**α** increases. Recall that in the magnitude and phase responses non linearity of the phase started manifesting whenα=0.2. It can therefore be concluded that the optimum value of the height adjustment parameter **α**is 0.1 in this circumstance.

Table 1: Power Levels of Filtered Signals at Normalised Frequency of 0.875

Power level of unfiltered voice	+29.52dB
Power level of filtered voice when α=0.0	-16.49dB
Power level of filtered voice when α=0.05	-20dB
Power level of filtered voice when α=0.1	-23dB
Power level of filtered voice when α=0.2	-33.10dB
Power level of filtered voice when α=0.3	-34.67dB

V. Conclusion

It can be concluded that HAT window is a reliable window in designing FIR filters for voice signal processing. For the voice signal, the noise type and level in this circumstance, the optimum value of **α**is 0.1. This value may vary if a different type of signal, noise type or filter length is under consideration

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