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Enhanancing Voice Signal Processing Using Blackman-Haris Windowed Filtering Technique

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ABSTRACT- *The integrity of human voice can be compromised by high frequency noise and other transmission impairments. In this research, a static finite impulse response digital low pass filter for removing high frequency noise from voice signal was designed and implemented. It incorporates a new algorithm known as Blackman-Harris windowed filter algorithm. Based on the algorithm, filter order of 40, sampling frequency of 16000Hz, pass band frequency of 3200Hz and rejection frequency of 4800Hz, that is a transition band of 1600Hz were used in removing noise signals with frequencies above the pass band frequency of 3200Hz. The filter order and sampling frequency were largely varied to investigate their effects on the performance of the system. Their optimum values are of the order of 50 and 14500Hz for the sampling frequency. These values provided a pass band ripple of -1.93dB, stop band ripple of -69.06dB and rejection frequency of 4165.125Hz, that is, a transition band of 965.125Hz. A comparative analysis of the new window with three other algorithms individually obtained from Han, Hamming and Blackman windows by removing high frequency noise from voice signals shows that the stop band ripple for the Han window, Hamming window and Blackman window are -68.53dB, , -68.41dB and -69.01dB respectively, indicating that the performance of the new algorithm is very satisfactory.*

KEYWORDS: *Digital Filter, Blackman-Haris Windowed Filter and Blackman window Function*

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I. INTRODUCTION

The human voice is the most important and effective means for passing message to mankind. With advancement in technology, voice signals, which have frequency range of about 200Hz to 3200Hz, are now carried over transmission lines to send messages to remote ends where ordinarily the voice cannot reach. These transmission lines include mobile communication, telephone communication, multimedia communication and public address system transmission lines. Incidentally, this voice signals can be contaminated from either the source or along this transmission lines by unwanted signals such as additive white Gaussian noise, random noise, power line noise, high frequency noise and other noise signals. For the integrity of the message in the voice signals to be preserved at the receiving end, any noise contaminating the signal must be removed. The object of interest in this research is to remove high frequency noise from voice signals.

However, Static Finite Impulse Response (FIR) digital low pass filter can be used to remove high frequency noise effectively during voice generation and transmission. FIR filters without window weighting will tend to distort the voice signal on application because of the complex nature of the signal. The voice will suffer degradations due to differential phase shift as a result of high frequency noise. Several researchers have used rectangular window (Dhar and Khan, 2011), triangular window (Dhar and Khan, 2010 – 2011), Kaiser Window (Saseendran and Mehra, 2014, Dhar and Khan, 2010 – 2011), hamming (Window (Babu et al, 2015, pawar and Mehra, 2014, Dhar and Khan, 2010 – 2011), hanning window (Babu et al, 2015, Pawar and Mehra, 2014, Dhar and Khan, 2010-2011) and Blackman window (Babu et al, 2015, Dhar and Khan, 2010-2011). In this paper, our focus is on developing Blackman-Harris window function which is a reliable technique to further enhance the quality of output in voice signal processing.

II. DIGITAL FILTERS

A digital filter is small dedicated computer that can implement an algorithm capable of converting an input sequence into a desired output sequence (Sarkar, 2003). Digital filters can be used for signal separation

and restoration. Signal separation is necessary when a signal is corrupt with unwanted signals at that point, whereas signal restoration is used to restore a distorted signal. It can be of infinite impulse response (IIR) or finite impulse response (FIR) type. The impulse response of IIR filters is composed of sinusoids that exponentially decay in amplitude. In principle, this makes their impulse response infinitely long and hence the name infinite impulse response filter. However, the amplitude eventually drops below the round-off noise of the system, and the remaining samples can be ignored (Smith, 2001). On the other hand, a finite impulse response filter does not have such infinitely long impulse, rather they have finite duration.
A digital filter is characterized by its transfer function, or equivalently.
The general transfer function for a linear, time invariant digital filter can be expressed as transfer function in z-domain, if it is causal, then it has the form of equation 1

$$H(z) = \frac{B(z)}{A(z)} = \frac{b^0 + b_1 z^{-1} + b_2 z^{-2} + - - + b_n z^{-N}}{1 + a_1 z^{-1} + a_2 z^{-2} + --- + a_m z^{-m}} \quad \text{.....(1)}$$

where the order of the filter is the greater of N or M (Mbachu, 2014, Cavicchi, 2000). This form of filter represented by equation 1 is infinite impulse response filter, which typically leads to infinite impulse response behavior, but if the denominator is unity, or all the poles are at the origin as shown in 2, then this is the form for a finite impulse response filter (Mitra, 2001).

$$H(z) = b^0 + b_1 z^{-1} + b_2 z^{-2} + - - + b_n z^{-N} \quad \text{..................................(2)}$$

II. RELATED WORKS

Many researchers have studied and used several methods in removing noise from human voice signals. Babu et al (2015) employed hamming, hanning and Blackman windows to design low pass and high pass filters for filtering speech signals. The considered characteristics are impulse response, magnitude response, phase response and pole-zero response. The order of each filter is 64 while the cut-off frequency for the high pass filter is 200Hz. The sample is 22050 and number of bits per sample is 16.Kaiser window is an efficient window in designing low pass filter for audio applications. It can be used to design an FIR low pass filter to remove the high frequency noise in audio signals (Saseendran &Mehra, 2014). The authors used a cut-off frequency of 10,800Hz and sampling frequency of 40,800Hz. The number of taps of the filter is 21 while three different values of 0.5, 3.5 and 8.5 for β parameter are used to design the filter one after the other. The parameter β determines the width of the Kaiser function. The aim is to remove high frequency audio signals above 8192Hz. A contaminated audio signal is applied to the 21-taps designed filter for each value of β and the results recorded. The simulation result shows that in the FIR filter β = 8.5, is better in performance than other values of β. Podder et al (2014) also demonstrated how Butterworth, Chebyshev-1 and elliptic filters can be used to analyze speech signals. They designed low pass, high pass, band pass and band stop filters and applied them in the processing of the speech signal. The order of each filter is 6, sampling rate, 8000 and number of bit per sample, 16.

III. PROPOSED PROCESSING SYSTEM

The block diagram of the proposed system is presented in fig. 1. It consists of a human and high frequency noise sources. The signals from the two sources are combined to produce a noisy voice signal which is applied to a microphone that converts it to electrical form. Since the
filtering is to be done in frequency domain, the noisy signal is transformed to frequency domain using fast Fourier transform (FFT) technique. The high pass filter removes the low frequency noise such as the 50Hz powerline noise while the static FIR digital low pass filter which is the filter of interest removes the high frequency random noise from the voice signal. The filtered voice signal is transformed back to time domain because it is a time domain function, using inverse fast Fourier transform (IFFT) before finally converting to analogue form.

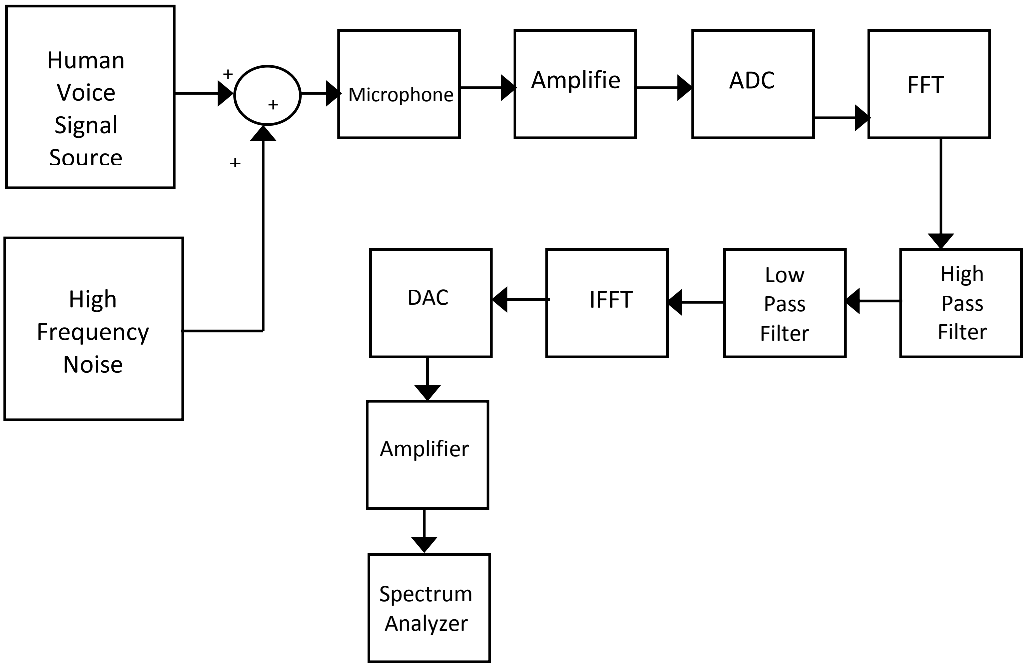

Figure 1: Block Diagram of Voice Signal Processing

IV. DESIGN AND REALISATION OF THE PROPOSED FIR BLACKMAN-HARRIS WINDOWED FILTER

Eight steps are involved in this design and implementation. They are as follows: mathematical modeling of the Blackman Harris window function, calculation of the order of the filter based on selected sampling frequency and attenuation values, obtaining responses of the filter based on the calculated order and cut-off frequency, determining the optimum order of the filter, determining the optimum sampling frequency of the filter, structural realization of the filter, generation of results, calculation of signal to noise ratio of the filter, and validating the proposed system

V. MATHEMATICAL MODELING OF A BLACKMAN HARRIS WINDOW FUNCTION

A typical unit Blackman-Harris window function is represented in fig. 1 as obtained from its mathematical model of equation 3 (Chinchkledeetal, 2011).

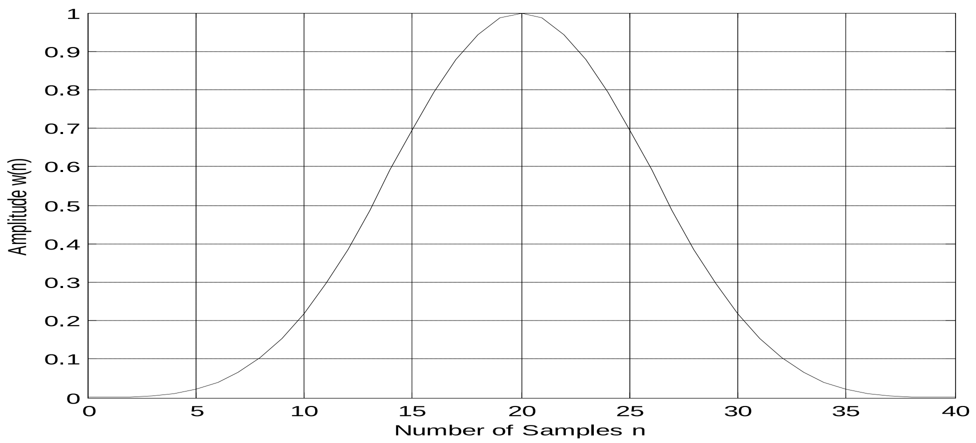

Fig.2: Blackman-Harris Window Function (equ 3)

$$w(k) = 0.35875 - 0.48829\cos\left(\frac{2\pi k}{M-1}\right) + 0.14128\cos\left(\frac{4\pi k}{M-1}\right) - 0.01168 cos\left(\frac{6\pi k}{M-1}\right) \qquad \text{............ 3}$$

$0 \leq k \leq M\text{-}1$

where M is the number of samples or taps of the filter.

M=L+1 ………………………………………..4

Where L is the order of the filter and M begins from 0 to L.

VI. RESULTS AND ANALYSIS

The results from our algorithm show that it is possible to remove noise from human voice signal. Firstly, we considered the proposed Blackman-Harris windowed filter. The uncontaminated and filtered voice signals are shown below in Fig. 3 & Fig. 6 respectively.

VII. OPERATION OF THE PROPOSED FILTER

An uncontaminated voice signal is generated using matlab function and it is shown in fig.3. High frequency noise of 6000Hz, 8000Hz and 10000Hz are generated and added together to form the noise signal of fig. 4. The uncontaminated voice signal is corrupt with the high frequency noise and the corrupt signal is presented as fig. 5. The contaminated voice signal is applied to the proposed filter and the output is presented in fig. 6. The magnitude responses of the contaminated and filtered signals are shown as figures 7 and 8 respectively.

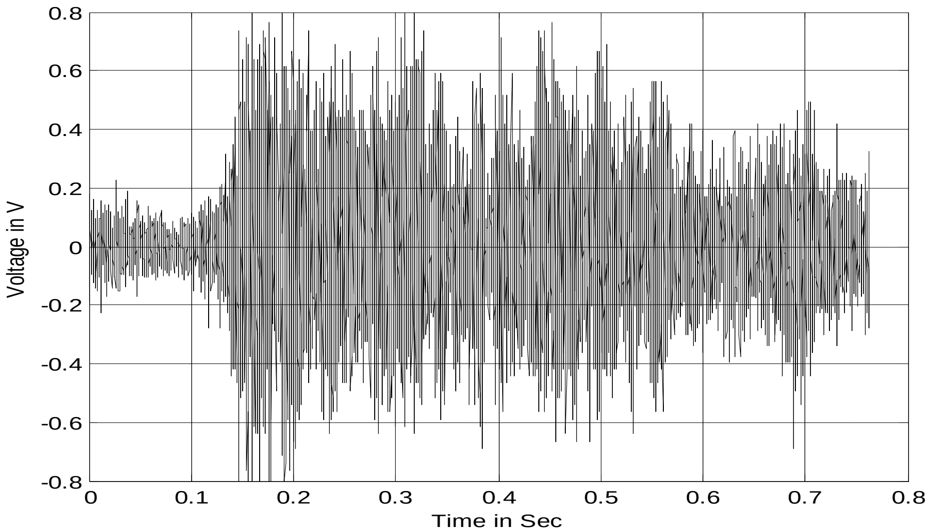

Fig. 3: Uncontaminated Voice Signal

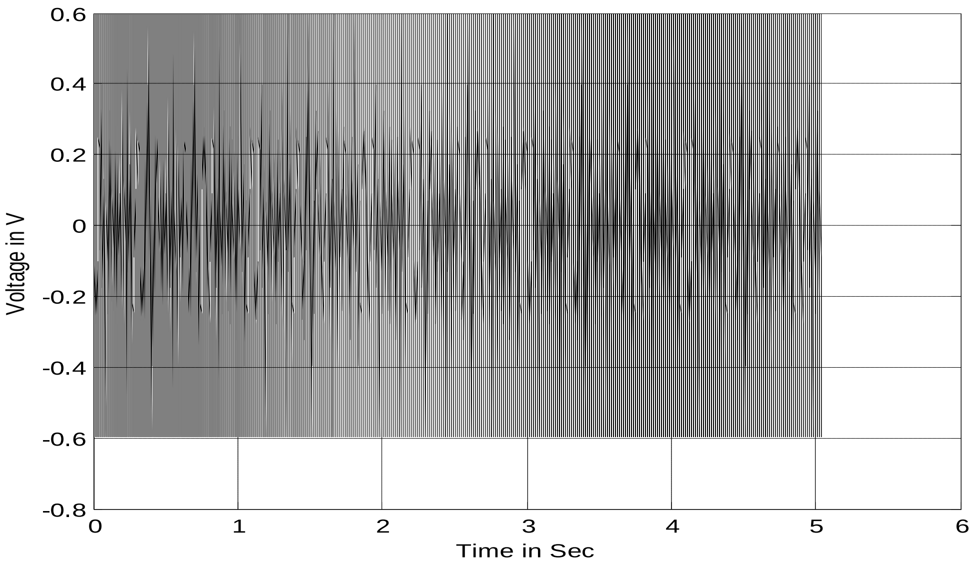

Fig. 4: High Frequency Noise

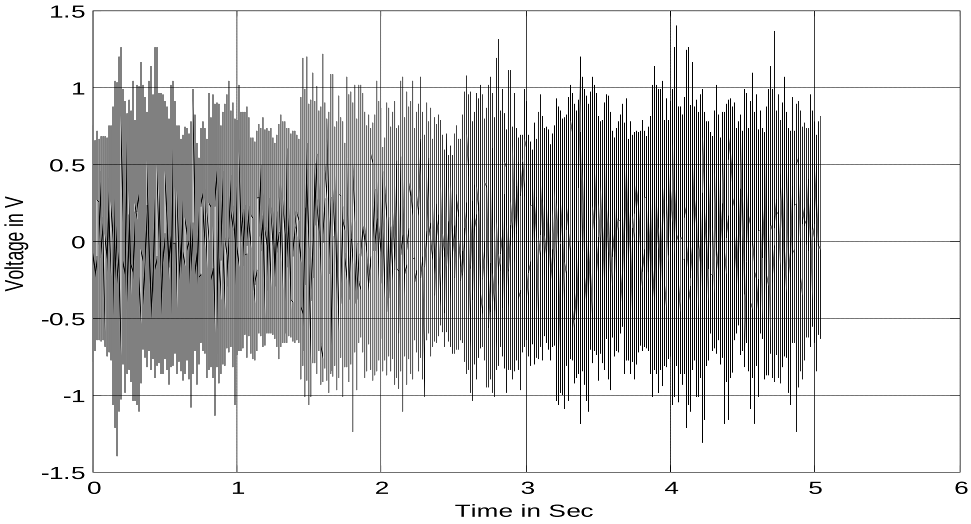

Fig. 5: Contaminated Voice Signal

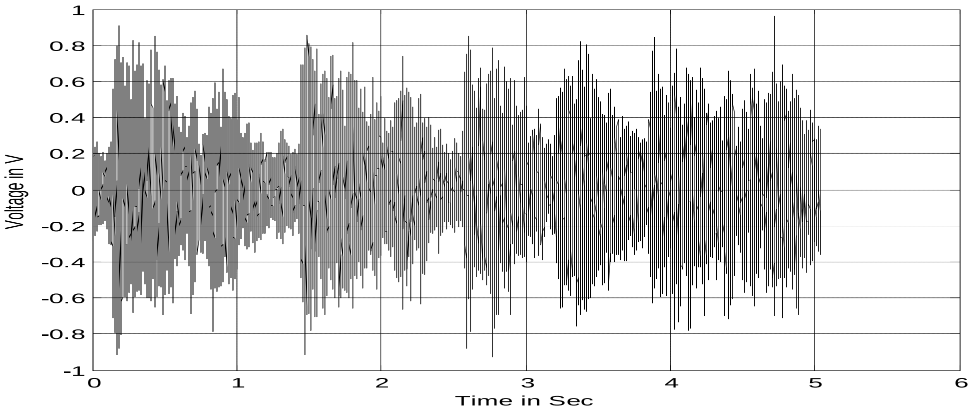

Fig.6: Filtered Voice Signal

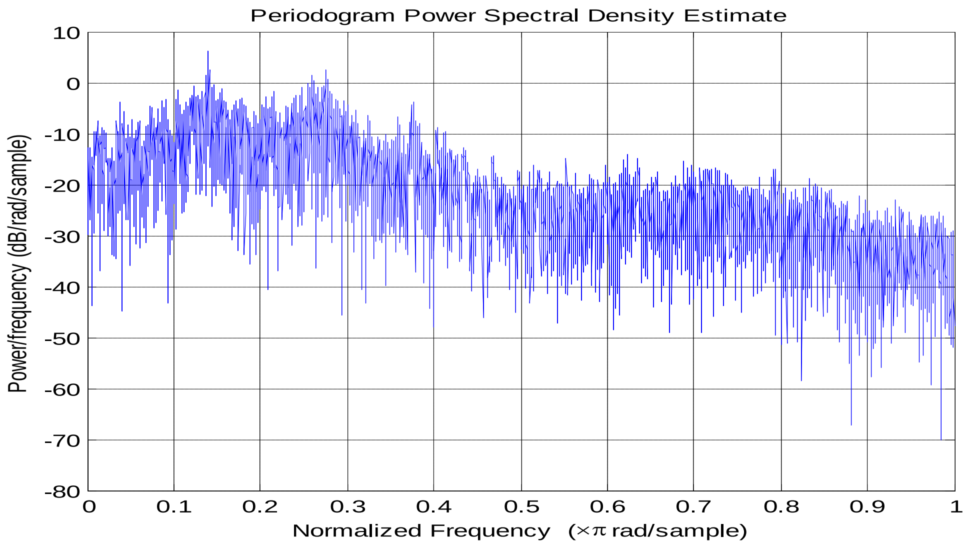

Fig. 7: Power Spectral Density of the Contaminated Signal

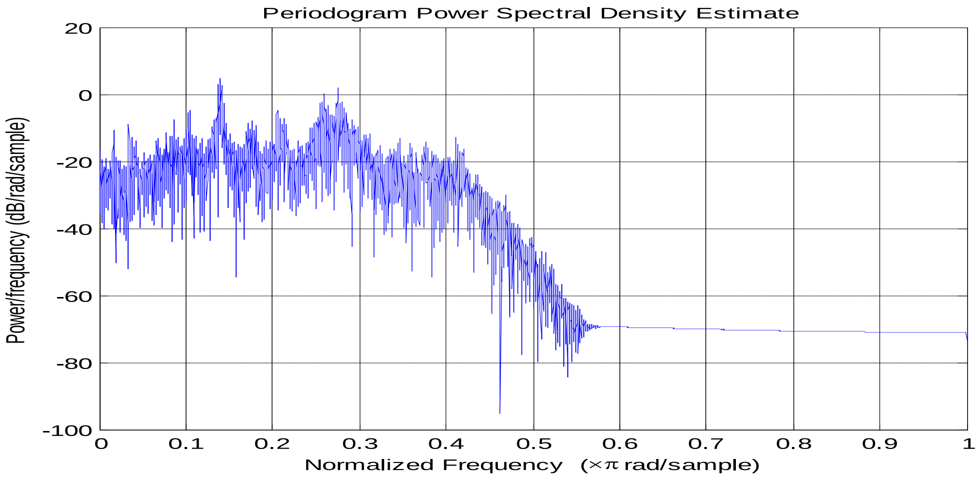

Fig.8: Power Spectral Density of the Filtered Signal

Comparing fig. 3, fig. 5 and fig. 6, that is the uncontaminated, contaminated and filtered voice signals respectively it can be seen clearly that the filter substantially removed the noise in the corrupt voice signal. Also comparing fig. 7 and fig. 8 it can be seen that from normalised frequency of 0.5745, which corresponds to rejection frequency of about 4165.125Hz, the filter removed noise components of such frequencies with a stopband attenuation of -69.06dB and above.

VIII. CONCLUSION

In this research, a new algorithm known as Blackmann-Harris windowed filter was developed and used in a static finite impulse response digital low pass filter to effectively remove noise from voice signal. The linearity property of the filter was exploited because it helped to complex process signals such as voice signal. From the information gathered about magnitude response, it shows that the filter was stable as there was no oscillation. The response shows a cutoff frequency of 3200Hz and rejection frequency of 4165.125Hz which agrees substantially with voice signal characteristics. Based on the algorithm, filter order of 40, sampling frequency of 16000Hz, pass band frequency of 3200Hz and rejection frequency of 4800Hz, that is a transition band of 1600Hz, removing noise signals with frequencies above the pass band frequency of 3200Hz was extensively performed. The result shows that their ultimate optimum values are 50 for the order and 14500Hz for the sampling frequency. These values provided a pass band ripple of -1.93dB, stop band ripple of -69.06dB and rejection frequency of 4165.125Hz, that is, a transition band of 965.125Hz.

RECOMMENDATION FOR FURTHER STUDIES

The Blackman-Harris window filter should be applied in filtering and processing other signals such as electrocardiographic and electroencephalographic signals, baseline wander, electromyographic and electrooculographic signals.

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