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Design and Implementation of Non Real Time and Real Time Digital Filters for Audio Signal Processing

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ABSTRACT

An analog active filter can not provide a very sharp cut-off for both higher and lower frequency component, while a digital signal processor (DSP) using digital filter effectively reduces the unwanted higher or lower frequency components within an audio signal. In this paper, we present the design and implementation of non real time and real time digital filter for audio signal. Experimental results indicate that digital filters including low-pass, high-pass and band-pass effectively eliminate both low and high frequency components contained in human voice, providing high quality voice. The performances of these implemented digital filters are evaluated both by listening the filtered output and also by analyzing the frequency response curves.

Keywords: *Digital Signal Processor, FIR Digital Filter*.

1. INTRODUCTION

One of the most dynamic areas of digital systems today is in the field of digital signal processing or DSP. A DSP is a very specialized form of microprocessor that has been optimized to perform repetitive calculations on stream of digitized data [1]. The digitized data are usually being fed to the DSP from an ADC (Analog to Digital converter). A calculation is performed by the DSP to process these digitized data that come in. This calculation involves the most recent data point as well as several of the preceding data samples. The result of the calculation produces a new output data point, which is usually sent to a DAC (Digital to Analog) converter.

A major application of DSP is in filtering and conditioning of analog signals. Filtering of a signal can be done by taking samples of the signal with an ADC, performing mathematical operation on the samples with a microcomputer and output the result to a DAC [3]. This digital filter approach can easily produce filter response. The digital approach has the further advantage in that the filter response can be changed under program control.

Today processing of real time digital audio on personal computer is becoming more and more common. Real time audio processing allows modified audio to be “judged by hearing” while it is processed. A real time digital filter is needed to process this audio signal.

The rest of the paper is organized as follows. Section II presents characteristics of digital filters. Sections III and IV describes the design and implementation of our non real time and non real time digital filters respectively. Section V analyzes the frequency response curves of these filters, and Section VI discusses the limitation and future work for this study. Finally, section VII concludes this paper.

2. DIGITAL FILTER

A digital filter is a filter that works by performing digital mathematical operations on an intermediate form of a signal. It takes a digital input, gives a digital output and consists of digital components. A real time digital filter can perform filtering operation on real time signal. There are two classes of digital filters, depending on the duration of the impulse response [2]. For finite-duration impulse response (FIR) digital filter, the operation is governed by linear constant-coefficient difference equations of a non-recursive nature. The transfer function of a FIR digital filter is a polynomial in z^{-1}.

Infinite-duration impulse response (IIR) digital filters whose input-output characteristics are governed by linear constant-coefficient difference equations of a recursive nature. The transfer function of an IIR digital filter is a rational function in z^{-1} [2]. Consequently, for a prescribed frequency response the use of an IIR digital filter usually results in a shorter filter length than the corresponding FIR digital filter.

2.1 Structure of FIR Digital Filter

An inherent property of FIR digital filters is that they can realize a frequency response with linear phase. Recognizing that a linear phase response corresponds to a constant delay, the approximation problem in the design of FIR digital filters is therefore greatly simplified. Specifically, the design simplifies to that of approximating a desired magnitude response.

In general, an FIR system is described by the difference equation [4]

$$y(n) = \sum_{k=0}^{M-1} b_k \ x(n-k) \qquad (1)$$

or equivalently by the system function

Volume 2 No. 3 ISSN 2079-8407
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$$H(z) = \sum_{k=0}^{M-1} b_k \ Z^{-k} \cdot \quad (2)$$

where, x(n), y(n), and b_k are the input, output and co-efficient of the FIR system, respectively.

The unit impulse response of the FIR system is identical to the co-efficient $\{b_k\}$ that is,

$$h(n) = \begin{cases} b_n \ , 0 \le n \le M-1 \\ 0 \quad , otherwise \ . \end{cases} \quad (3)$$

where M is the length of the filter.
Thus, using (1) and (3) we can write,

$$y(n) = \sum_{k=0}^{M-1} h(k) \ x(n-k) \quad (4)$$

2.2 Filtering of Speech signal

The preprocessing of speech signal is fundamental to many applications such as the digital transmission and storage of speech, automatic speech recognition, and automatic speaker recognition systems. FIR digital filters are well suited for the preprocessing of speech signals.

In speech processing applications, it is essential to maintain precise time alignment. The exact linear phase property inherent to a FIR digital filter caters to this requirement in a natural way [5].

To design a FIR digital filter with a sharp cutoff characteristic, the length of the filter has to be large. This in turn will produce an impulse response with a long duration.

3. DIGITAL FILTER DESIGN

The basic principle of digital filter is to take continuous samples of the output waveform from the ADC, process the samples with the microcomputer and resultant output to the DAC to get analog signal. The processing done by the microcomputer, determines the filter response. FIR filters can be designed using the following method:

- Design of FIR filter using window method
- Design of FIR filter using frequency sampling method
- Design of Optimum Equiripple FIR filters

Window method is chosen for designing FIR digital filter since the design procedure is very straightforward and easy to use and the window characteristics completely define the filter characteristics.

Truncating an infinite-length impulse response is equivalent to multiplying it by a finite-length window function. The window determines how much of the impulse response can be 'seen' and is sometimes refers to as observation window [6]. There are different types of windows such as rectangular, hamming, hanning, Kaiser etc.

In order to get a satisfactory response of the intended digital filter, the following procedural steps are used for the FIR filter using "WINDOW METHOD".

- Appropriate number of coefficient for the digital filter is decided to determine the point at which the unit impulse response is to be truncated.
- A window function is selected and applied to the unit impulse response.
- The resulting amplitude response is thoroughly checked to ensure that the resulting response is satisfactory. If the response is not satisfactory, then the value of M is increased or one of the many other different window functions is used.

The most straightforward approach to FIR filter design is to obtain a finite length impulse response by truncating an infinite duration impulse response sequence. If $H_d(e^{jw})$ be the ideal desired frequency response, then

$$H_d(e^{jw}) = \sum_{n=-\infty}^{\infty} h_d(n) e^{-jwn} \ , \quad (5)$$

where $h_d(n)$ is the corresponding impulse response sequence, which is defined as

$$h_d(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(e^{jw}) e^{jwn} \, dw \, . \quad (6)$$

The sequence $h_d(n)$ is of infinite duration, and it must be truncated to obtain a finite duration impulse response. If $h_d(n)$ has infinite duration, one way to obtain finite duration impulse response is to simply truncate $h(n)$, which is defined as

$$h(n) = \begin{cases} h_d(n) \ , 0 \le n \le M-1 \\ 0 \qquad otherwise \ . \end{cases} \quad (7)$$

In general, $h(n)$ can be represented as the product of desired impulse response and a finite duration of rectangular window, $w(n)$, is defined as

$$h(n) = h_d(n) \ w(n) \quad (8)$$

where

$$w(n) = \begin{cases} 1, & 0 \le n \le M-1 \\ 0, & otherwise. \end{cases} \quad (9)$$

4. DIGITAL FILTER IMPLEMENTATION

Figure 1 shows a block diagram of our digital filter design for the transmission and processing of voice signal.

Volume 2 No. 3 ISSN 2079-8407
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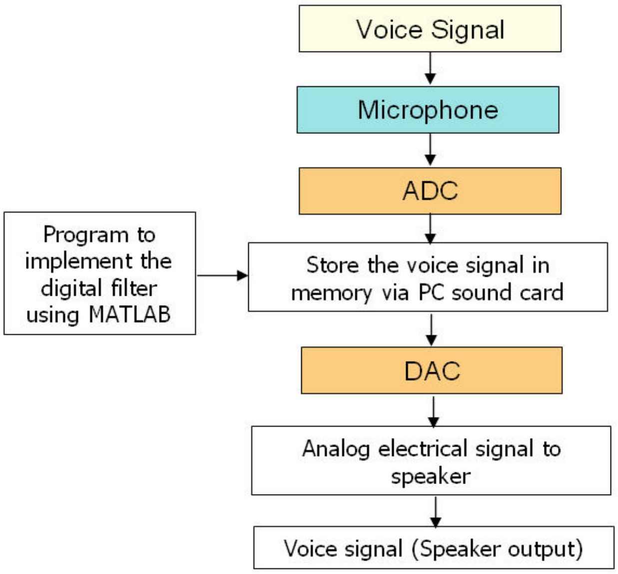

Fig.1 Filter design procedure.

Voice signal is the actual input of the system. Micro-phone acts as an interface between the human audio voice signal and PC sound card. The input of the micro-phone is the human voice signal which is a mechanical wave. Microphone acts a transducer to convert this mechanical signal to an analog electrical one.

The ADC is used to convert the analog signal to digital signal while DAC performs the reverse operation.

The output of the speaker is the filtered voice signal which is the improved version of the original voice signal. It is a mechanical wave and human ear receives this mechanical wave in the form of hearing sound.

Using MATLAB programming, three types of non real time digital filters (high pass, low pass, and band pass) are implemented. Separate programs are written and executed for different filters.

The following functions are used to implement non real time FIR digital filter:

[Y, FS, NBITS]=WAVREAD('FILE') reads a windows wave file where Y is the sample data, FS is the sample rate in Hz, and NBITS is the number of bits per sample.
WAVWRITE(Y_1, FS, NBITS, 'WAVFILE') writes data Y_1 to windows wave file.
Y = fft(X, n) returns the n-point Discrete Fourier Transform (DFT).
[h,w] = freqz(b,a, l) returns the frequency response vector h and the corresponding frequency vector w for the digital filter whose transfer function is determined by the (real or complex) numerator and denominator polynomials represented in the vectors b and a, respectively.

The real time FIR digital filters (high pass, low pass and band pass) are implemented by using MATLAB DSP block set. This set contains DSP sources, DSP sinks and filter design tools. FDA (Filter Design and Analysis) tool is used for implementing digital filter. A model has been devised for real time as shown in figure 2.

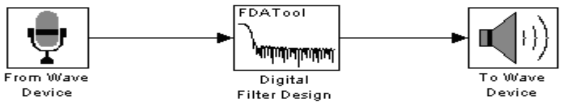

Fig.2. Implementation of real time digital filter using MATLAB DSP block sets.

5. EXPERIMENTATAL RESULTS

Different programs have been developed for different non real time FIR digital filters using windows. The outputs of these filters are shown in the Figures 3 to 8. Figures 3 and 4 show the input and output response of a low-pass filter for an audio voice signal in both time domain and frequency domain when cutoff frequency is 3400 Hz.

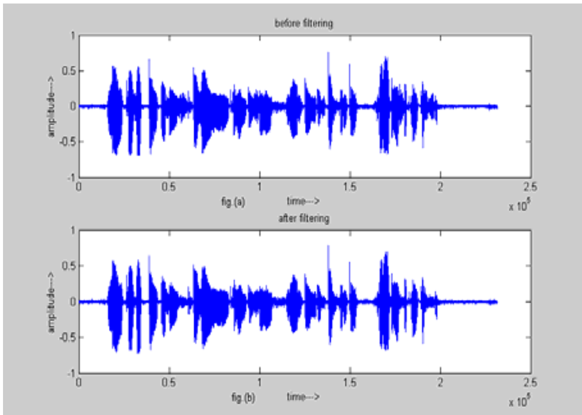

Fig.3: Input and output response of a low- pass filter in time domain.

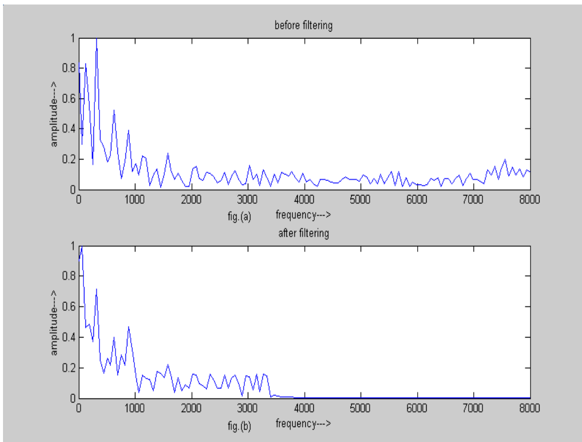

Fig.4: Input and output response of a low-pass filter in frequency domain.

Figures 5 and 6 show the input and output response of a high-pass filter for an audio voice signal in both time domain and frequency domain when cutoff frequency is 600 Hz.

Volume 2 No. 3 ISSN 2079-8407
Journal of Emerging Trends in Computing and Information Sciences

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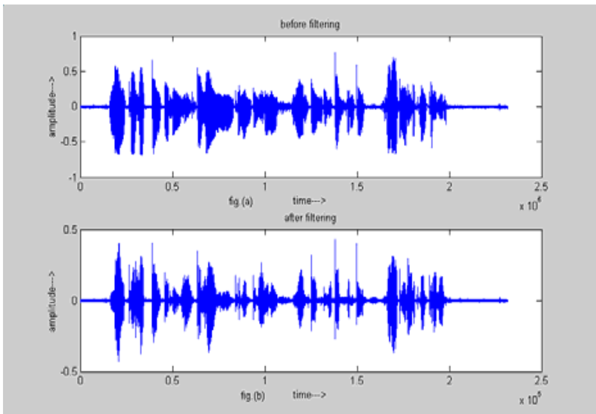


Fig.5: Input and output response of a high-pass filter in time domain.

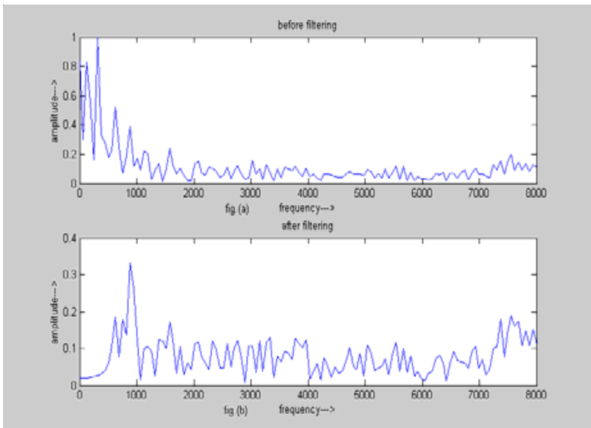


Fig.6: Input and output response of a high-pass filter in frequency domain.

Figures 7 and 8 show the input and output response of a band pass filter for an audio voice signal in both time domain and frequency domain when pass band frequencies are 600 Hz and 3400Hz.

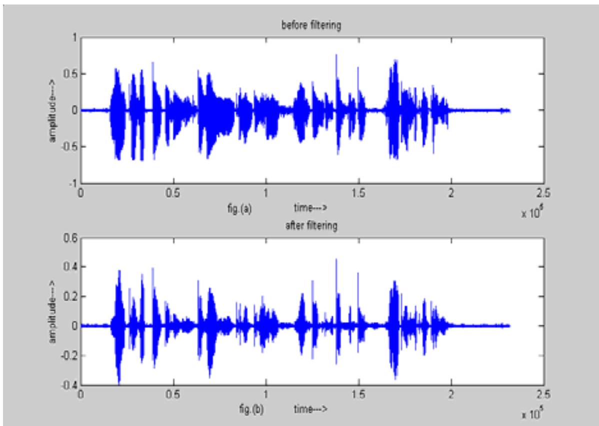


Fig.7: Input and output response of a band-pass filter in time domain.

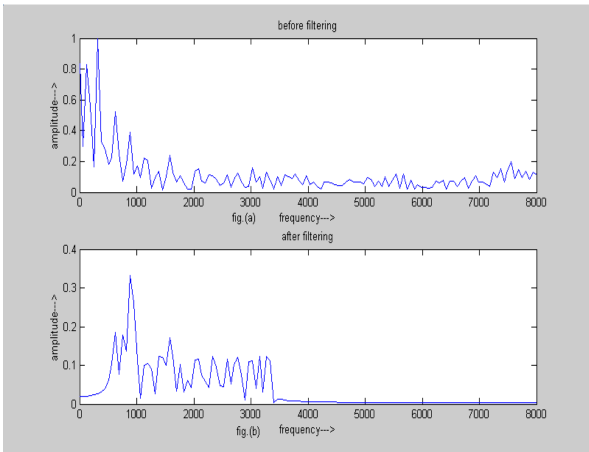


Fig.8: Input and output response of a band-pass filter in frequency domain.

Using DSP block sets different types of FIR digital filters have been implemented and both input and output listened simultaneously on real time. The output of these filters is shown in figures 9 to 11 in terms of frequency response. Figures 9(a) to 9(f) represent the frequency response of a voice signal for a low-pass filter when cutoff frequency is 3400 Hz for different windows.

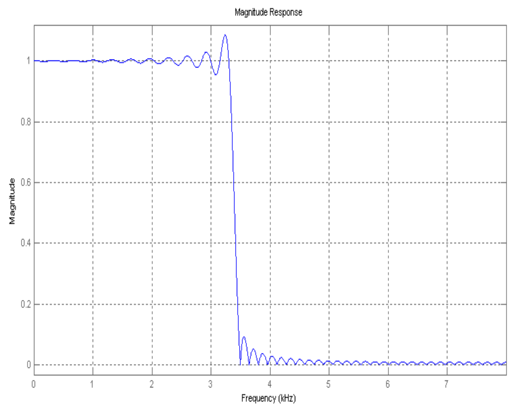


Fig. 9 (a). Rectangular window

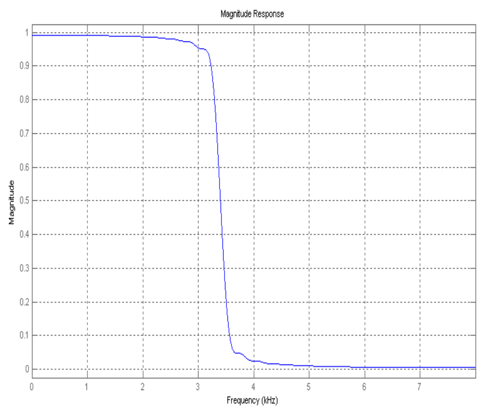


Fig. 9 (b). Triangular window

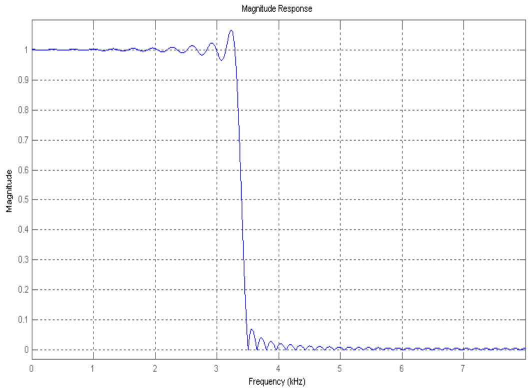
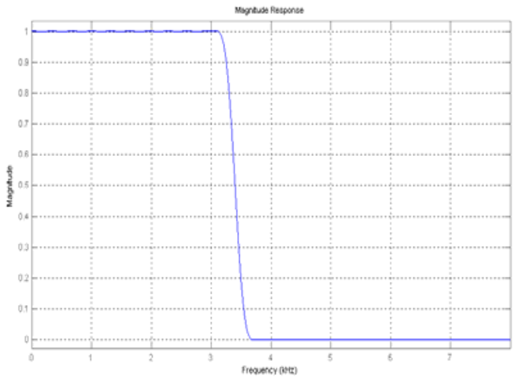


Fig. 9 (c). Kaiser window



F
ig. 9 (d). Hamming window

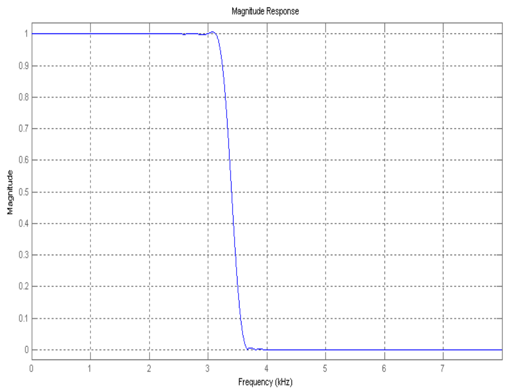


Fig. 9 (e). Hanning window

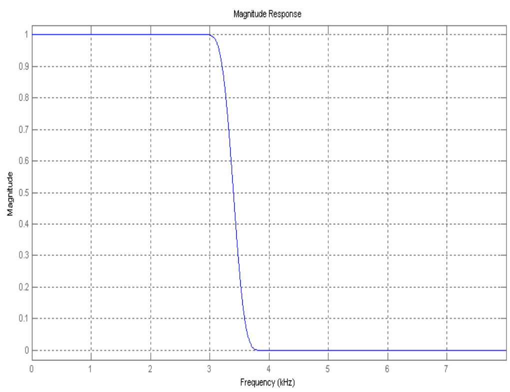


Fig. 9 (f). Blackman window

Figures 10(a) to 10(f) represent the frequency response of a voice signal for a high-pass filter when cutoff frequency is 600 Hz for different windows.

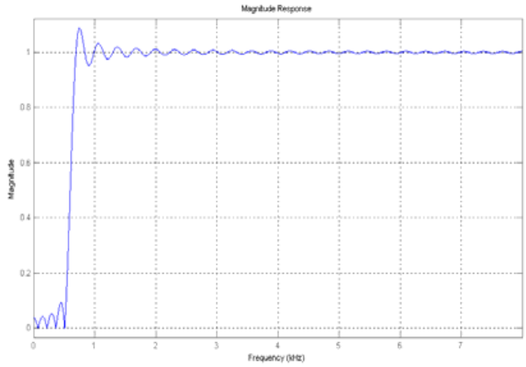


Fig. 10 (a). Rectangular window

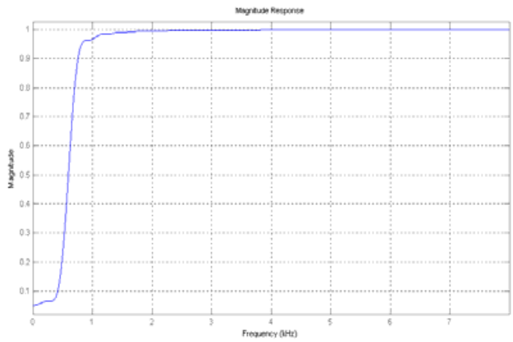


Fig. 10 (b). Triangular window

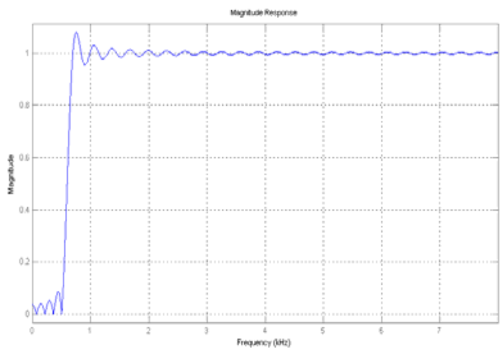


Fig. 10 (c). Kaiser window

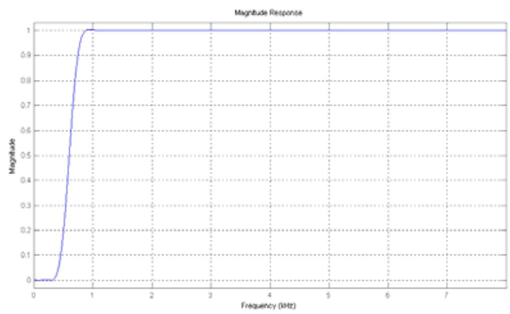


Fig. 10 (d). Hamming window

Volume 2 No. 3 ISSN 2079-8407
Journal of Emerging Trends in Computing and Information Sciences

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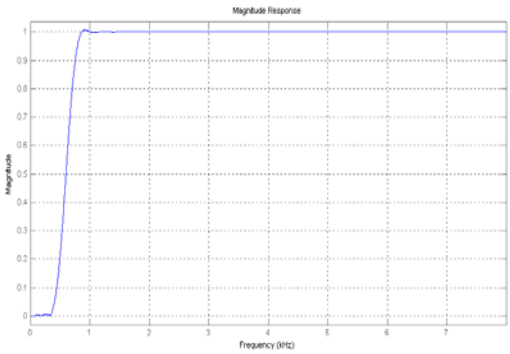

Fig. 10 (e). Hanning window

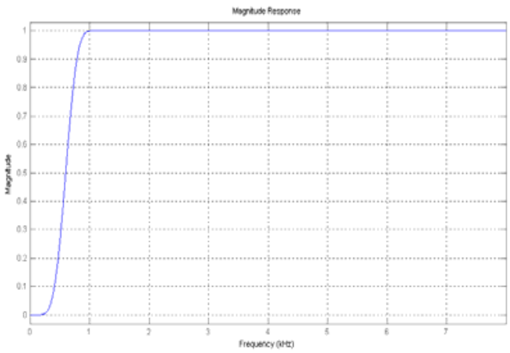

Fig. 10 (f). Blackman window

Figures 11(a) to 11(f) represent the frequency response of a voice signal for band pass filter when pass band frequencies are 600Hz and 3400Hz for different windows.

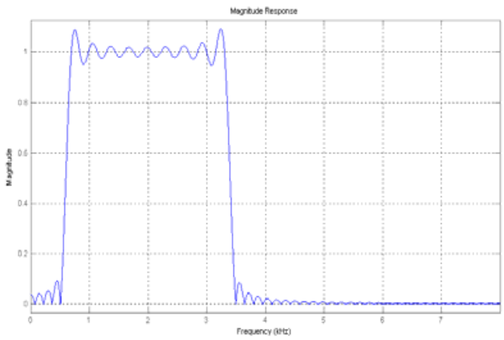

Fig. 11 (a). Rectangular window

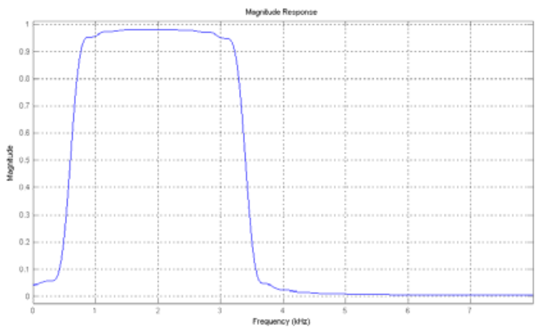

Fig. 11 (b). Triangular window

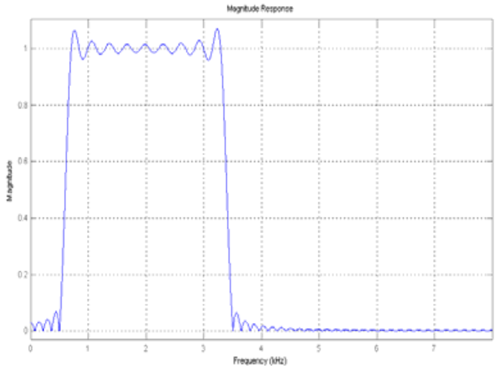

Fig. 11 (c). Kaiser window

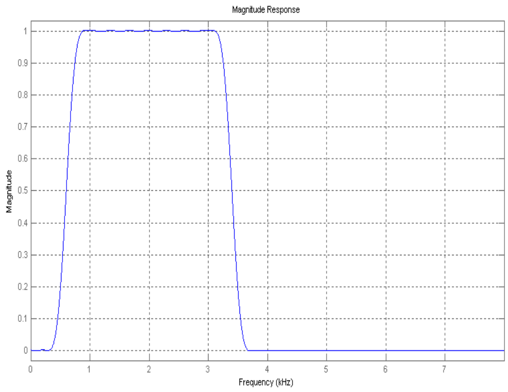

Fig. 11 (d). Hamming window

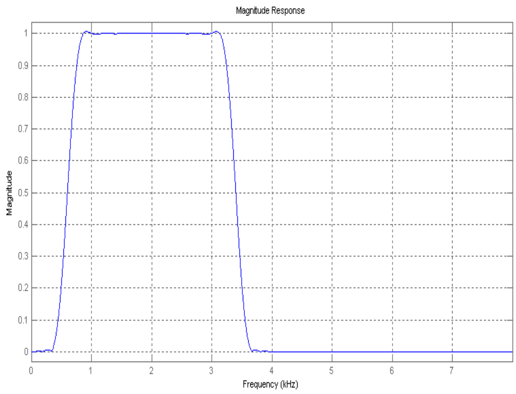

Fig. 11 (e). Hanning window

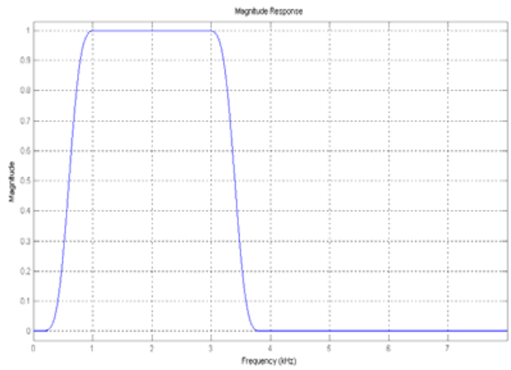

Fig. 11(f). Blackman window

The cut off frequency of low-pass, high-pass and band-pass filter were considered as a variable. For a low pass filter the cutoff frequency was varied from 2.6 kHz to 4 kHz. The quality and intensity of voice varies a little around the cutoff of 3.4 kHz. Similarly for a high pass filter cutoff frequency was selected ranging from 300 Hz to 1000 Hz. A significant change has not been noticed around 600 Hz. Thus, for the high pass filter 600 Hz is selected as cutoff frequency. Also, for band pass filter the bandwidth is 2800 Hz (3400 Hz – 600 Hz).

The frequency response curves of non real time digital filters show almost no ripples in stop band. In case of real time digital filters some ripples exist in pass band and stop band for Rectangular and Kaiser Window but almost no ripples in other windows. These filters provide sharp cut off for removing both higher and lower frequency components of voice signal.

6. LIMITATIONS AND RECOMMENDATIONS

A digital filter is used as a constituent part of digital signal processor the input of which is an analog signal. Digital filter takes discrete data from ADC. But the speed of operation of ADC and subsequently digital signal processor puts the practical limitations on its use. The video signals or signals having other extremely wide band width requires fast sampling rate of ADC and high performance digital signal processor. The real time and non real time FIR digital filters are implemented here to process the human voice signal. Any other low frequency audio signal can also be analyzed using these filters. These filters can also be used to remove the noise signals contained in human voice signal. In the future, we will modify this filtering program to support high frequency wide band signal such as video signal.

7. CONCLUSION

In this paper, we have introduced non real time and real time digital filters for audio signal processing, especially for voice signal. Moreover, we have analyzed the effects of different types of FIR digital filtering for human voice. We have observed insignificant changes in voice quality and intensity since human ear is a low pass filter having a massive diaphragm. It is not able to change the movement of diaphragm in accordance with the change in frequency of the voice. These filters can also be used for removing the noise contained in an audio signal. The order of these filters and the cut off frequency can be varied. These filters can be used for processing and transmitting of audio signal for specific purpose.

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