Implementation of FIR Filter using Adjustable Window Function and Its Application in Speech Signal Processing

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Abstract—Digital filter plays an important role in digital signal processing applications. Digital filter can also be applied in speech processing applications, such as speech filtering, speech enhancement, noise reduction and automatic speech recognition among others. In this paper Matlab program is used to implement a low-pass FIR filter using adjustable window function based on Hamming window. In some applications such as frequency spectral analysis of a signal, a small main lobe width of the window function in frequency domain is required for increasing the ability to distinguish two closely spaced frequency components without increasing the length of the filter. In this adjustable window function the width of main lobe and amplitude of the side lobe can be varied by changing the value of $\alpha$ for a fixed length. The comparisons of low pass FIR filters for different values of $\alpha$ are shown. Designed low-pass FIR filter is used in the speech filtering application. The filter is applied to a recorded speech signal to remove high-frequency components of the speech signal. Signal comparison in time and frequency domain between filtered speech and original speech shows that a high frequency component of the original speech signal successively eliminates by using this low pass filter.

Keywords—FIR filters, Window function, Hamming Window, Low-pass filter, FFT, IFFT, Speech Signal.

1. INTRODUCTION

A fundamental aspect of digital signal processing is filtering. Digital filters are widely used for filtering purposes in the different applications of digital signal processing. Digital filter is a system that performs mathematical operations on sampled discrete-time signal to reduce or enhance certain aspects of that signal. It can be used to pass or attenuate the signals according to the specified frequency. Digital filters are classified either Infinite Impulse Response (IIR) filters or Finite Impulse Response (FIR) filters. Finite Impulse Response filter has strictly linear phase, non-recursive structure, arbitrary amplitude-frequency characteristic, high stability and real-time stable signal processing requirements etc. [1,2]. Thus, it was widely used for different applications of digital signal processing such as speech processing. In speech processing Filters have many applications such as speech filtering, noise reduction and digital audio equalizing among others [3]. In the speech filtering, filter is used to pass or attenuate some frequency components of the speech signal.

FIR filter is described by differential equation.

\[
y(n) = \sum_{k=0}^{M-1} (h(k)x(n-k)) \tag{1}
\]

$x(n)$ is the input signal
$h(n)$ is the impulse response of fir filter
The transfer function of FIR filter is obtained by taking the $z$-transform of $h(n)$.

\[
H(z) = \sum_{k=0}^{M-1} (h(k)z^{-k}) \tag{2}
\]

The direct-form realization is the simplest structure in several realization structures for implementing an FIR system. The realization structure was shown in Fig.1 [1].
The most straightforward approach to designing FIR filters is based on truncating the Fourier series representation of the prescribed frequency response by using a Window function. The main advantage of this design technique is that the impulse response coefficient can be obtained in closed form without the need for solving complex optimization problems and can be determined very fast [5].

Some commonly used Fixed window functions are; Rectangular window, Bartlett window, Hanning window, Hamming window and Blackman window. In the literature, these different windows are used for the applications of Digital FIR filter design and spectral performance analysis [6-9]. Pranav Kumar, Mohammad Khan paper deals with the design of FIR filter and its application in audio signal processing [10]. In the study of Fourier transform of these different Fixed window functions, for the fixed length the Rectangular window provides smallest main lobe width but the highest peak of side lobe among them. Hanning, Hamming and Bartlett windows have significantly lower side-lobe amplitude compared with rectangular window but the main lobe width is wider. The main lobe width of these three windows approximately the same but Hamming window provides the smallest peak of side lobe. Blackman window has smallest side lobe peak compared to the other window, but its main lobe width is wider [4,12]. The Kaiser window is a kind of adjustable window function which provides independent control of the main lobe width and ripple ratio. But the Kaiser window has the disadvantage of higher computational complexity due to the use of Bessel functions in the calculation of the window coefficients [2].

In some application such as frequency spectral analysis of signal a small main lobe width of widow function is required because as the main lobe width of the window function increases the ability to distinguish two closely spaced frequency components decreases. Rectangular window provides smallest main lobe width but it has the largest relative side lobe amplitude, causes considerable leakage [3]. The main lobe width can be reduced by increasing the length of the window but which results in a larger filter with increasing computational complexity.

In this paper an adjustable window function based on hamming window function [11] is used for designing an FIR filter. In this window function for a fixed length the main lobe width and amplitude of side lobe can be varied in the frequency domain by changing the value of $\alpha$, which provides greater flexibility according to different applications. The suitable value of $\alpha$ is selected according to different digital signal processing applications. the design filter is compared for some different values of $\alpha$. the application of this filter in speech signal filtering is also presented.

2. FIR FILTER DESIGN USING WINDOW METHOD

In the FIR filter design process, we determine the coefficient of a causal FIR filter that closely approximates the desired frequency response specifications. There are different methods to find the coefficients of digital filter from frequency specifications.

1) Fourier series method
2) The window method
3) Frequency sampling method
4) Optimal filter design method

Window method is one of the main FIR filter design techniques, because of its simple operation and ability to reduce of Gibbs’ oscillations present in the Fourier series method by using a class of time-domain functions known as window functions. In the Window Method, first the infinite-duration impulse response is determined by expanding the frequency response of an ideal filter in a Fourier series and then we develop an FIR filter by multiplying the impulse response of the ideal IIR filter with a finite duration window function in the time domain or by convolving the frequency response of the ideal IIR filter with the frequency response of the window function.

**DESIGNING PROCEDURE FOR LOWPASS FIR FILTER USIND WINDOW METHOD:**
Implementation of FIR Filter using Adjustable Window Function and Its Application in Speech Signal Processing

1. Suppose the transfer function of the ideal desired filter is given by the equation.

\[
H_d(e^{jw}) = \begin{cases} 
1, & \text{when } |w| \leq w_c \\
0, & \text{when } w_c < |w| \leq \pi 
\end{cases}
\]  

(3)

2. Impulse response \( h_d(n) \) of ideal filter was obtained by applying inverse Fourier transform to the ideal characteristics \( H_d(e^{jw}) \) of digital filter [12].

\[
h_d(n) = \frac{1}{2\pi} \int_{-\omega_c}^{\omega_c} e^{jnw} dw
\]

(4)

\[
h_d(n) = \begin{cases} 
\sin \left( \frac{\omega_c n}{\pi} \right), & n \neq 0 \\
\frac{\omega_c n}{\pi}, & n = 0
\end{cases}
\]

(5)

3. The window function \( w(n) \) and window length \( M \) are selected according to performance indicators. These both parameters are selected according to requirements of transition bandwidth and stop-band attenuation. In this paper adjustable window function based on hamming window function given in equation (9) is used for designing FIR filter.

4. In the window method design procedure we design a causal linear-phase FIR filter by multiplying an ideal filter that has an infinite-duration impulse response (IIR) by a finite-duration window function.

\[
h(n) = h_d(n)w(n)
\]

(6)

5. The transfer function of designed FIR filter is obtained by taking the Fourier transform of \( h(n) \).

\[
H(e^{jw}) = \sum_{n=0}^{M-1} (h(n)e^{-jnw})
\]

(7)

3. ADJUSTABLE WINDOW FUNCTION BASED ON HAMMING WINDOW.

Two desirable characteristics of a window function are [2]:

(1). Fourier transform of the window function should have a small width of the main lobe.
(2). Fourier transform of the window function should have side lobes that decrease in energy rapidly as \( \omega \) tends to \( \pi \).

Windows can be categorized as fixed or adjustable window function. Fixed windows such has Rectangular, Hanning, Hamming and Blackman window have only one independent parameter window length which controls the main-lobe width. Adjustable windows have two or more independent parameters such as window length and one or more additional parameters that can control the other window’s characteristics [5,12]. The Kaiser window is a kind of two parameter window function. In a Kaiser window width of main lobe can be controlled by adjusting the length of the filter and side lobe level can be controlled by varying the other parameter \( \alpha \). But the Kaiser window has the disadvantage of higher computational complexity due to the use of Bessel functions in the calculation of the window coefficients.

An adjustable window function equation based on hamming window function is given by FREDRIC J. HARRIS [11]. In this window function the width of main lobe can be varied by changing the value of \( \alpha \) for a fixed length of the filter. \( \alpha \) is selected according to different applications. This generalized window is referred to as the Hamming window for \( \alpha = 0.54 \) and Hann or Hanning window for \( \alpha = 0.5 \). They are both commonly used in speech processing and other digital signal processing applications. But in some application such as spectral analysis of a specified frequency spectrum, if the frequency of interest contains two or more signals very near to each other, then frequency resolution is very important. In such cases a small main lobe width of the window function in frequency domain is required. For an efficient value of \( \alpha \), this window function provides a lesser main lobe width compares to Hanning (\( \alpha = 0.5 \)) and Hamming window (\( \alpha = 0.54 \)), however the amplitude of side lobe and ripples in pass band is also increased.
\[ w(n) = \begin{cases} 
\alpha - (1 - \alpha) \cos \left( \frac{\pi n}{M-1} \right), & 0 \leq n \leq M - 1 \\
0, & \text{otherwise}
\end{cases} \quad (8) \]

4. APPLICATION OF FIR FILTER IN SPEECH SIGNAL PROCESSING

FIR filter can also be used in speech processing applications such as speech filtering, noise reduction, frequency boosting and digital audio equalizing etc. In speech filtering filter are used to modify the frequency response of a speech signal according to requirements of speech processing. Hanning and Hamming window function are commonly used in speech and voice processing applications. Designed low-pass FIR filter for Hamming window (\(\alpha = 0.54\)) is used to eliminate the high-frequency spectrum of the speech signal. Fast Fourier transforms (FFT) is used for analysis of speech signals in the frequency domain. Experimental results show that high-frequency part of speech signal was significantly reduced after applying design low-pass filter.

PARAMETER SPECIFICATION FOR SPEECH SIGNAL FILTERING APPLICATION:

Take a recorded speech signal in wave format and load into Matlab by using the following command

\[
[x \ fs]=\text{wavread('speech.wav');}
\]

\(x\) is the sample speech signal and \(fs\) is the sampling frequency of speech signal in Hz. the Sampling frequency of recorded speech signal 16000 Hz.

A low-pass filter was designed to filter out the high-frequency components above 3200 Hz of the speech signal. Cutoff frequency of the filter \(f_c = 3200\) Hz.

The normalized cutoff frequency of the filter is calculated as \(w_c = f_c / (fs / 2) = 0.4\).

A cutoff frequency of low-pass filter can be varied according to requirements.

Consider order of the filter is 33.

According to requirements the order of the filter can be varied according to transitional bandwidth and stop band attenuation.

5. SIMULATION RESULTS

The comparison of design low-pass filter using adjustable window function for different values of alpha is shown in figure 2. Consider the length of the filter \(M=33\) and normalized cutoff frequency is \(w_c=0.4\). For length \(M=33\) and normalized cutoff frequency \(w_c=0.4\), two new efficient values of \(\alpha\), \(\alpha=0.71\) and \(\alpha=0.78\) are selected for designing a low-pass FIR filter, which provides lesser main lobe width compare to Hanning \((\alpha = 0.5)\) and Hamming window \((\alpha = 0.54)\), however the amplitude of side lobe is also increased. But these two new values are providing higher side lobe attenuation compare to rectangular window function.
Implementation of FIR Filter using Adjustable Window Function and Its Application in Speech Signal Processing

Figure 2. Amplitude frequency characteristic curve of FIR filter for different values of $\alpha$ for length $M=33$.

Designed low-pass FIR filter for $\alpha = 0.54$ is used to eliminate the high-frequency spectrum of the speech signal. The Amplitude frequency and phase-frequency characteristic curve of low-pass FIR filter is shown in fig3.

Figure 3. Amplitude frequency and phase-frequency characteristic curve of FIR filter using Hamming window ($\alpha = 0.54$) for $M=33$.

The impulse response of low-pass FIR filter is shown in fig4.

Figure 4 Impulse response of FIR filter.

Representations of recorded speech signal in time domain and frequency domain before and after filtering are shown in fig 5 and 6.
As shown in Figure 6, the designed low-pass filter significantly reduces high-frequency components, above 3,200 Hz, while letting the signal pass through the filter. A cutoff frequency of low-pass filter can be varied according to different speech processing applications.

6 CONCLUSION

In this paper, low-pass FIR filter designed using adjustable window function and compared for different values of $\alpha$. This type of window function is simple in operation and provides greater flexibility in digital signal processing applications. In frequency resolution problems a small main lobe width of window function in frequency domain is required. In the range $0 \leq \alpha \leq 1$, for large values of $\alpha$, as the value of $\alpha$ increases the main lobe width is continuously decreasing compared to Hanning and Hamming windows; however the amplitude of side lobe is also increased. $\alpha$ is selected according to requirements based on different applications. Digital filter plays a major role in digital signals processing with broad applications in speech processing. Application of designed low-pass filter in speech filtering is also presented in this paper. This filter can also be used for removing the noise contained in speech signal. On the bases of the requirements filter characteristic parameters such as filter order and the cutoff frequency can also be changed to...
Implementation of FIR Filter using Adjustable Window Function and Its Application in Speech Signal Processing

meet the desired Engineering requirements.

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