Design and Performance Analysis of FIR Filter for Audio Application

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ABSTRACT
One of the important applications of digital signal processing is digital filter used in communication system. Digital filters are commonly used in audio processing applications like audio filtering, audio enhancement, noise reduction and automatic audio recognition etc. When noise is mixed with the audio signal then it distracts the original signal. Digital FIR filters are used to reduce the unwanted noise from audio applications. This paper presents design of FIR filter for audio application using different window techniques. Hamming and Hanning window techniques are used for design analysis and comparison using MATLAB. The developed filters performance is compared in terms of signal to noise ratio (SNR). The result shows that Hanning window based filter shows 8% to 11% better SNR as compared to Hamming based FIR filter design.

Keywords: *DSP, FIR, IIR, ISI.*

1. INTRODUCTION
Digital Signal processing (DSP) related with the digital representation of the signals and these signals are modified, extract or analyze the information from it [1]. Digital signal Processing has a property to manipulate the information signal and modifying or improve in some manner that the signal characteristics has to be changed. Digital Filters gives the option of removing the noise, shape of the spectrum and also change the inter-symbol interference (ISI) in communication structure.
Digital signal processing has various applications in which Digital Filters are mostly used as one of them. The basic Digital Filters are FIR and IIR (Finite impulse Response, Infinite Impulse Response). These Filters are depends upon their Impulse Responses [2]. Digital filters are selected upon the nature of the problems and their specification of the required frequency response and their magnitude in pass band. When a linear phase with integer slope has a simple delay in the time domain and reduces the phase warping to minimum frequency domain [3].

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

Where h(l) is the impulse response of the filter. The transfer function of FIR filter for coefficient calculation is

$$H(p)= \sum_{p=0}^{M-1} h(p)z^{-p} \quad (2)$$

H(p) is the transfer function of the filter.

FIR filter are finite and IIR filters are infinite. FIR filters are non- recursive, that is there is no feedback involved. The Impulse Response of an FIR filter will eventually reach Zero. The impulse response of an IIR filter may very well keep “ringing” ad- infinitum.IIR filters can designed to accurately simulate “classical” analog filter responses where as FIR filters in general cannot do this. FIR filters s having a transfer function of a polynomial in z- and having all zero filters in the sense that the zeroes in the z- plane by obtaining the frequency response magnitude characteristic.

2. FIR DESIGN METHODS
FIR filters are mostly used for application where exact linear phase response is required. The FIR used in a non- recursive way which is mostly a stable filter. FIR filter design essentially consist Approximation problem and Realization. Approximation method chooses the ideal response from frequency domain also allowed the class of filters is chosen (e.g. the length N for filters).Approximation method or algorithm is used to find best filter transfer function. The realization part helps to choose the structure to implement the transfer function which may be in the form of the form of a program [4]. FIR filter designed by three well known methods namely as Window method, Frequency sampling technique, Optimal filter design method. The window technique describes the various method of designing the FIR filter. In window techniques the frequency responses of digital filters are periodic in frequency ranges and could be expand in Fourier series.

$$W_d(e^{j\omega})=\sum_{n=-\infty}^{\infty} w_d(n)e^{-j\omega n} \quad (3)$$

From the above equation the Fourier coefficients of the series w (n) whose impulse response of digital filter are identical is-

$$W(n)=\frac{1}{2\pi}\int_0^{2\pi} W(e^{j\omega})e^{j\omega n}d\omega \qquad (4)$$

When designing of a digital filter two types of the problems that is impulse response of a digital filter are finite duration and second and second problem is that the filters are non- causal and unrealizable. The reliability of impulse response cannot make by any finite amount of delay. So from the Fourier series representation of $W(e^{j\omega})$ the filter formed is an unrealizable IIR filter. To overcome from these two types of the problem the impulse response of the infinite time period would be converted to a finite duration impulse response by converted the infinite series at $n=\pm N$. By this conversion it does not gives the result in required oscillations in the pass band and stop band of the digital filter. This effect occurs by slow convergence of the Fourier series near the point where maximum discontinuities occurs. These types of the discontinuities for oscillations is removed by group of time-limited weighting function w(n), this weighting function is defined as window function and also used to modified the Fourier coefficients.

Hamming window function: The causal hamming window function is defined by

$$W_{hamm}(h)=\begin{cases}0.54-0.46\cos\frac{2\pi h}{N-1}, 0\le h< N-1\\ 0, \qquad \text{otherwise}\end{cases} \qquad (5)$$

The non causal hamming window function is

$$W_{hamm}(h)=\begin{cases}0.54\ +\ 0.46\cos\frac{2\pi h}{N-1}, |n|\ \le\ \frac{N-1}{2}\\ 0, \qquad \text{otherwise}\end{cases} \qquad (6)$$

Hanning window function: Causal hanning window function is described by

$$W_{hn}(y)=\begin{cases}0.5 - 0.5\cos\frac{2\pi y}{N-1}, 0\ \le\ y\ \le\ N-1\\ 0, \qquad \text{otherwise}\end{cases} \qquad (7)$$

For non causal hanning window function

$$W_{hn}(y)=\begin{cases}0.5\ +\ 0.5\cos\frac{2\pi y}{N-1}, 0<|y|<\frac{N-1}{2}\\ 0, \qquad \text{otherwise}\end{cases} \qquad (8)$$

Generally window function is defined in the two forms that are fixed and Adjustable window functions. But mostly fixed window function is used. These windows are Rectangular window, Hanning Window, Hamming Window, and Blackman window. The Kaiser window is a type of adjustable window function. The described windows are used in the designing and performance analysis of the FIR filter. These different windows give the frequency response in the form of the main lobe and side lobe structure in frequency domain. In the frequency response of the different windows we describe the width of the main lobe and side lobe of the frequency response. Hanning, Hamming they gives us better result in the form of the responses. They give actual side lobe peaks of the filter as compare to the rectangular window. Due to the above reasons these two windows are commonly used in the different application of the DSP. In Hamming, Hanning window the computational complexity is very less. [5, 6].

3. MATLAB BASED DESIGN SIMULATION

The window method is used to design a filter for signal to noise ratio in audio applications. In this paper designing of filter used two types of window Hamming and Hanning. In the MATLAB simulation an audio signal is loaded in the filter having fixed order of N=66 with noise of 50 Hz and samples are taking up to 2000. The input audio signals are of the different length. According to the length of the signal the amplitude vary from low to high or high to low. The magnitude and phase response of the hamming and hanning window are shown in Figure 1 & Figure 2.

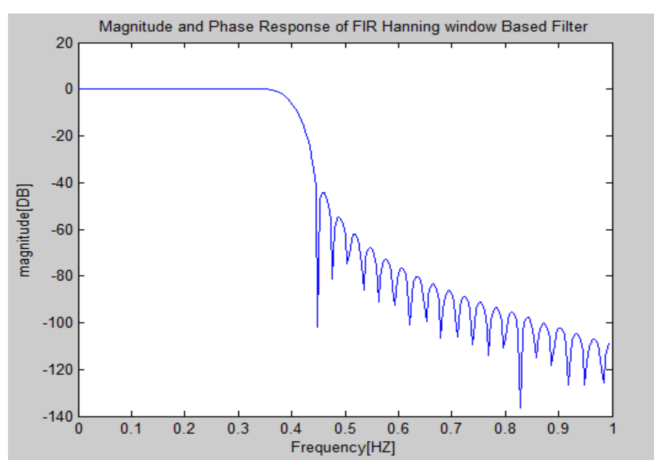

Figure1: Response of Hanning window.

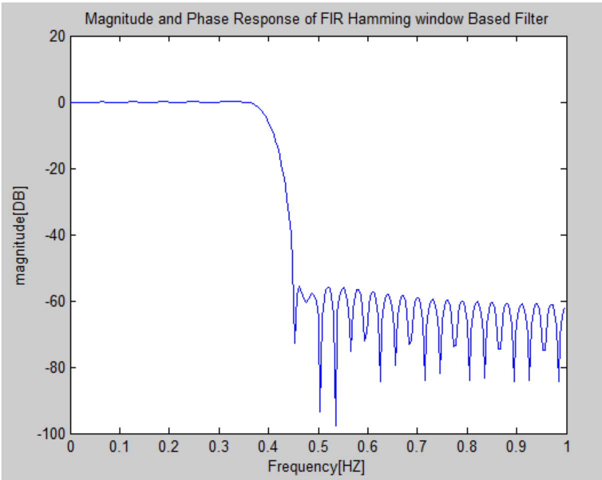

Figure 2: Response of Hamming window.

Hanning filter reduce the noise from first input Audio signal as shown in figure 3. Hamming filter reduce the noise from first input Audio signal as shown in figure 4.

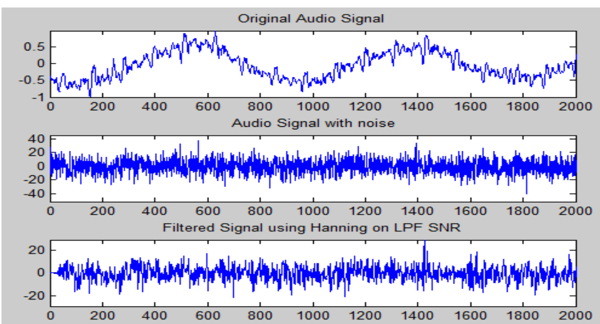

Figure 3: Filtered audio signal 1 for Hanning

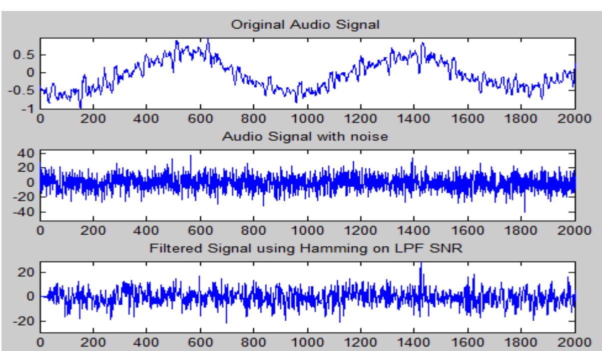

Figure 4: Filtered audio signal 1 for Hamming

Hanning filter reduce the noise from second input Audio signal of different length as shown in figure 5. Hamming filter reduce the noise from second input Audio signal of different length as shown in figure 6.

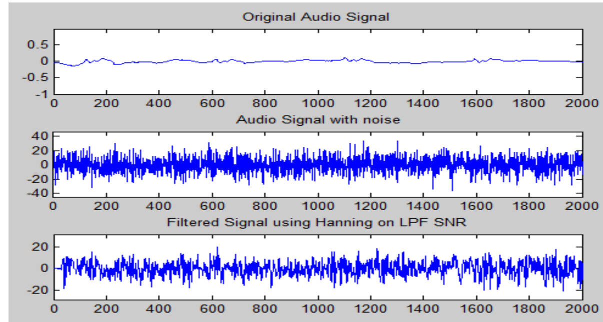

Figure 5: Filtered audio signal 2 for Hanning

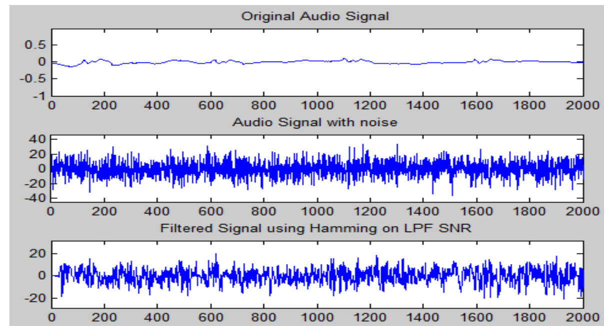

Figure 6: Filtered audio signal 2 for Hamming

Hanning filter reduce the noise from third input Audio signal as shown in figure 7. Hamming filter reduce the noise from third input Audio signal as shown in figure 8.

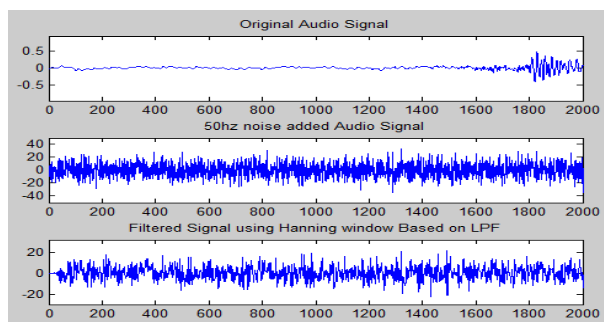

Figure 7: Filtered audio signal 3 for Hanning

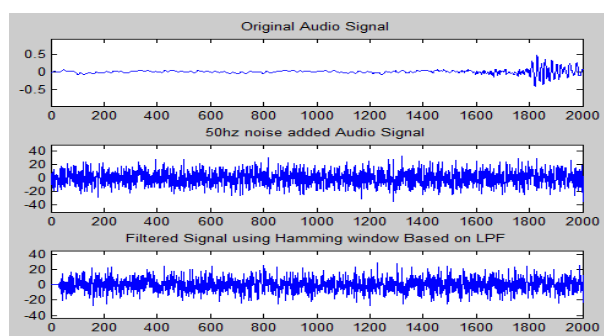

Figure 8: Filtered audio signal 3 for Hamming

4. RESULT AND DISCUSSION

In this section the signal to noise ratio (SNR) of different audio samples has been compared using hamming and hanning window. Table 1 shows that first audio sample (audio 1) provides 10.76% and second sample provides 9.47%, third sample provides 8.87% better SNR performance of hanning in comparison to hamming.

Table 1: Comparison of SNR.

Input Signal	Signal to Noise Ratio (SNR)		% Reduction in SNR
	Hanning	Hamming	
Audio 1	-24.8832	-27.8852	10.76
Audio 2	-28.0611	-31.0729	9.47
Audio 3	-30.7922	-33.7921	8.87

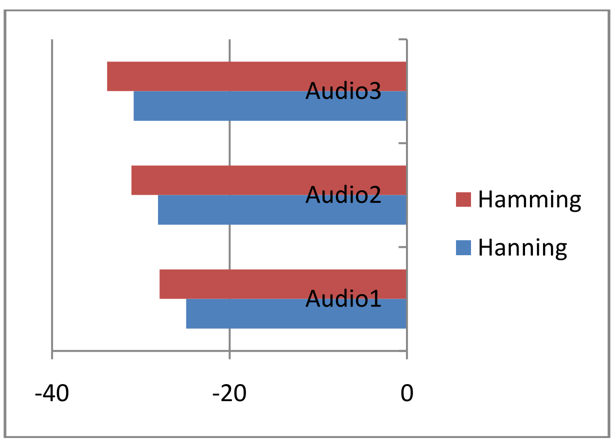

Figure 9: Bar Chart Comparison of SNR

5. CONCLUSION

In this paper three different noisy audio signals have been analyzed and compared after passing through two different filters. These filters are designed through Hamming and Hanning windowing techniques. Comparisons are in terms of the signal to noise ratio (SNR). From results it can be observed that the hanning window technique provides better results as compared to hamming windowing techniques. The simulation result shows that signal to noise ratio (SNR) varies according to the length of the audio signal. It is clear from the results that hanning windowing techniques provides better filtering as compared to hamming windowing technique. This means that Hanning reduces the maximum noise from the signal.

ACKNOWLEDGMENT

The author would also like to thank Director, National institute of Technical Teachers' Training & Research Chandigarh, India for their inspirations and support on this research work.

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