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Understanding the Behavior of Band-Pass Filter with Windows for Speech Signal

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ABSTRACT: *The use of Digital Signal Processing has always been a field of research in the Electronics & Communication Engineering, and specially in the field of Speech & Audio Processing we can always see much of the research work and showing promising result with the use of Fast Fourier Algorithm in the relevant work. FFT is used in the process to analyze the response of speech in different filters using windows. Windowing is used to minimize the effect leakage and reduce the noise level. Many different windows have been proposed over time each with its own advantages & disadvantages relative to the others. Some are more effective for specific type of signals, types such as random or sinusoidal. In the following work we have represented the use of different types of windows to filter speech signal consisting of speech data and noise. Windows restrict the leakage and obtain the amplitude and frequency spectrum plot using Rectangular, Hamming & Hanning windows for Band pass filters.The result obtained shows significant reduction in noise level in speech signal using Band pass filter. Furthermore, it can be seen that the band pass filter in itself gives best result with Hamming window.*

Key words: *Hamming, Hanning, IIR, FIR, FFT, leakage.*

I. INTRODUCTION

Digital signal processing (DSP) technology and its advancements have dramatically impacted our modern society everywhere. Without DSP, we would not have digital/Internet audio or video; digital recording; CD, DVD, and MP3 players; digital cameras; digital and cellular telephones; digital satellite and TV; or wire and wireless networks. Medical instruments would be less efficient or unable to provide useful information for precise diagnoses if there were no digital electrocardiography (ECG) analyzers or digital x- rays and medical image systems. We would also live in many less efficient ways, since we would not be equipped with voice recognition systems, speech synthesis systems, and image and video editing

systems. Without DSP, scientists, engineers, and technologists would have no powerful tools to analyze and visualize data and perform their design, and so on. The analog signal process relies wholly on electrical and electronic devices such as resistors, capacitors, transistors, operational amplifiers, and integrated circuits (ICs). DSP systems, on the other hand, use software, digital processing, and algorithms; thus they have a great deal of flexibility, less noise interference, and no signal distortion in various applications. A digital signal processing technique has three basic blocks; an input block or signal (analog or digital), a system block (where processing is done) and an output block (where the desired or processed signal can be viewed).[1]

II. FILTER EQUATIONS

IIR filters are sometimes unstable; non-zero pole exists in its transfer function, lower filter order than a corresponding FIR filter, and usually has nonlinear phase property.[1]

Infinite Impulse Response (IIR) filters are the first choice when the speed is paramount and phase non-linearity is acceptable. IIR filters are computationally more efficient than FIR filters as they require fewer coefficients due to the fact that they use feedback or poles. However feedback can result in the filter becoming unstable if the coefficients deviate from their true values.[1] The general equation of an IIR filter can be expressed as follows:

$$H(z) = \frac{b_0 + b_1 z^{-1} + \ldots + b_N z^{-N}}{1 + a_1 z^{-1} + \ldots + a_M z^{-M}}$$

$$= \frac{\sum_{k=0}^{N} b_k z^{-k}}{1 + \sum_{k=1}^{M} a_k z^{-k}}$$

a_k and b_k are the filter coefficients.
The transfer function can be factorized to give:

$$H(z) = k \frac{(z - z_1)(z - z_2)\cdots(z - z_N)}{(z - p_1)(z - p_2)\cdots(z - p_N)} = \frac{Y(z)}{X(z)}$$

Where: z_1, z_2… z_N are the zeros,
p_1, p_2… p_N are the poles.

FIR filters are inherently BIBO (bounded-input, bounded-output) stable, nonzero pole does not exist in its transfer function, easy to implement and can be designed to have linear phase property.
An FIR filter is completely specified by the following input-output relationship:

$$y(n) = \sum_{i=0}^{K} b_i x(n-i)$$
$$= b_0 x(n) + b_1 x(n-1) + b_2 x(n-2) + \cdots + b_K x(n-K) \qquad \ldots 3$$

where b_i represents FIR filter coefficients and K + 1 denotes the FIR filter length.
Applying the z-transform on both sides of above equation (3) leads to

$$Y(z) = b_0 X(z) + b_1 X(z) z^{-1} + \cdots + b_K X(z) z^{-K} \qquad \ldots .4$$

Factoring out X(z) on the right-hand side of equation (4) and then dividing X(z) on both sides, we have the transfer function, which depicts the FIR filter, as

$$H(z) = \frac{Y(z)}{X(z)} = (b_0 + b_1 z^{-1} + \cdots + b_K z^{-K}) \qquad \ldots .5$$

We can conclude that
1. The transfer function in Equation (5) has a constant term, all the other terms each have a negative power of z, and all the poles are at the origin on the z-plane. Hence, the stability of filter is guaranteed. Its impulse response has only a finite number of terms.
2. The FIR filter operations involve only multiplying the filter inputs by their corresponding coefficients and accumulating them; the implementation of this filter type in real time is straightforward.
From the FIR filter format, the design objective can be to obtain the FIR filter b_i coefficients such that the magnitude frequency response of the FIR filter H(z) will approximate the desired magnitude frequency response, such as that of a lowpass, highpass, bandpass, or bandstop filter.[1]

FFT based measurements are subject to errors from an effect known as leakage. This effect occurs when the FFT is computed from a block of data which is not periodic. To correct this problem appropriate windowing functions must be applied. we must choose the appropriate window function for the specific application. When windowing is not applied correctly, then errors may be introduced in the FFT amplitude, frequency or overall shape of the spectrum.[4]

III. FILTERING TYPES

The basic filter types can be classified into four categories: lowpass, highpass, bandpass, and bandstop. Each of them finds a specific application in digital signal processing. One of the objectives in applications may involve the design of digital filters. In general, the filter is designed based on specifications primarily for the passband, stopband, and transition band of the filter frequency response. The filter passband is the frequency range with the amplitude gain of the filter response being approximately unity. The filter stopband is defined as the frequency range over which the filter magnitude response is attenuated to eliminate the input signal whose frequency components are within that range. The transition band denotes the frequency range between the

passband and the stopband. The design specifications of the lowpass filter are illustrated in Figure (), where the low-frequency components are passed through the filter while the high-frequency components are attenuated.[1]
A low-pass filter is a filter that passes low-frequency signals and attenuates (reduces the amplitude of) signals with frequencies higher than the cutoff frequency. The actual amount of attenuation for each frequency varies depending on specific filter design. It is sometimes called a high-cut filter, or treble cut filter in audio applications. A low-pass filter is the opposite of a high-pass filter. A band-pass filter is a combination of a low-pass and a high-pass.[1]

The high pass filter, remains high-frequency components and rejects low frequency components.

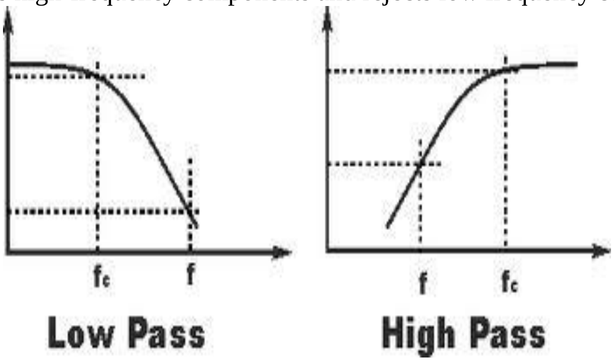

Figure 2. Low pass and high pass filter responses.

A high-pass filter (HPF) is an electronic filter that passes high-frequency signals but attenuates (reduces the amplitude of) signals with frequencies lower than the cutoff frequency. The actual amount of attenuation for each frequency varies from filter to filter. A high-pass filter is usually modeled as a linear time-invariant system. It is sometimes called a low-cut filter or bass-cut filter. High-pass filters have many uses, such as blocking DC from circuitry sensitive to non-zero average voltages or RF devices. They can also be used in conjunction with a low-pass filter to make a band pass filter. The bandpass filter attenuates both low- and high-frequency components while remaining the middle-frequency component. Finally, the bandstop (band reject or notch) filter, rejects the middle-frequency components and accepts both the low- and the high-frequency component.[1]

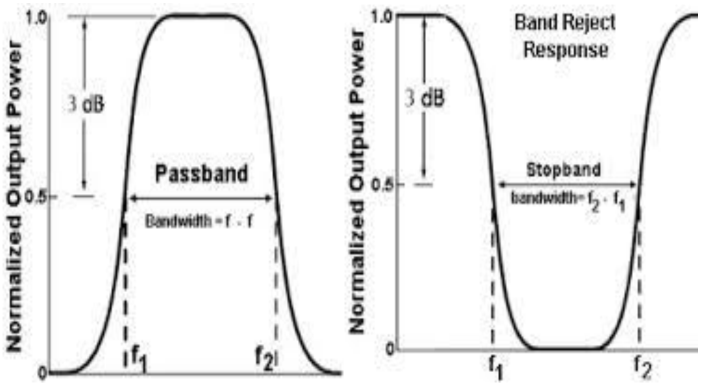

Figure 3. Band pass and Band stop filter responses.

IV. WINDOWING

Window technique implicates a function called window function. It is also known as tapering function. It states that if some interval is chosen, it returns with finite non-zero value inside that interval and zero value outside that interval. A major effect of windowing is that the discontinuities of the frequency response are converted into transition bands between values on either side of the discontinuity.[4]
There are many window techniques available for designing the FIR filter and they are:
1. Hanning window 2. Hamming window
3. Blackman window 4.Rectangular
5. Bartlett window 6. Kaiser window

All the programmings are done on MATLAB and the results are obtained according to the sample speech signal fed to the software.

V. RESULT

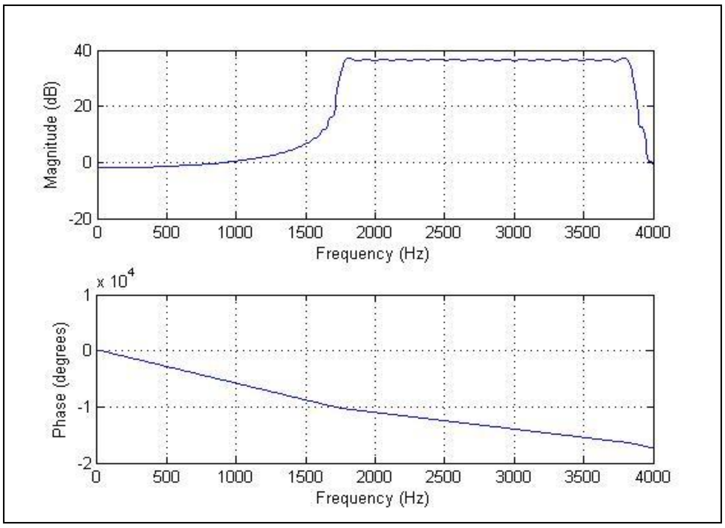

Figure 4. Frequency responses of the designed band pass filter using rectangular window.

Above figure shows that the magnitude response of the band pass filter with hamming window has many side lobes and does not completely cut-off the lower frequency components, thus allowing them to appear in the side lobes, unlike the rectangular window which totally cut-off the side lobes.

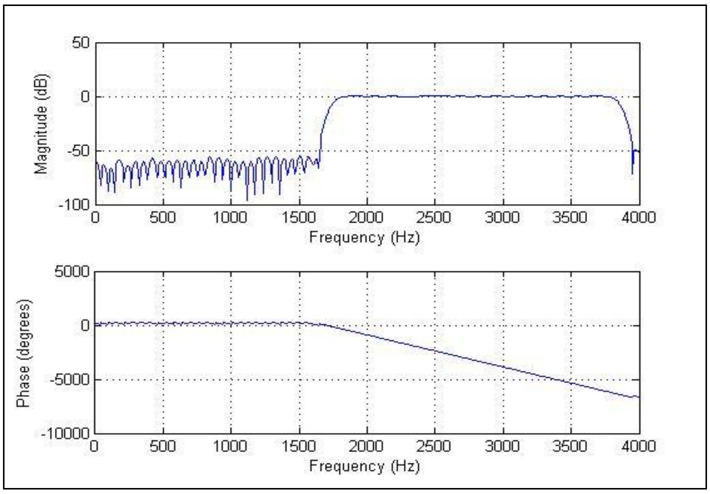

Figure 5. Frequency responses of the designed band pass filter using hamming window.

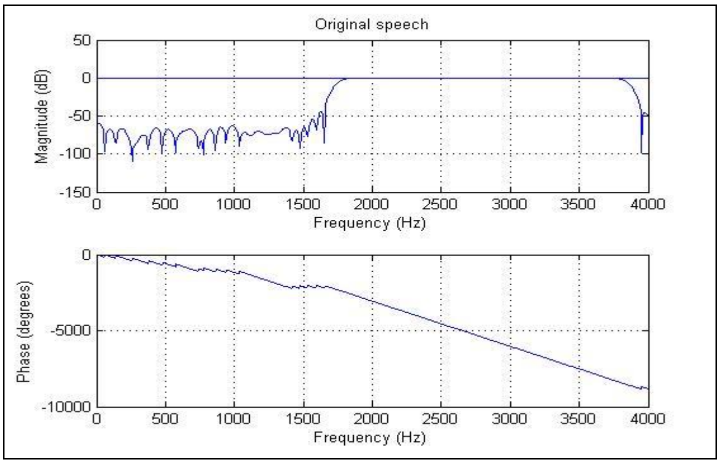

Figure 6. Frequency responses of the designed bandpass filter using Hanning window.

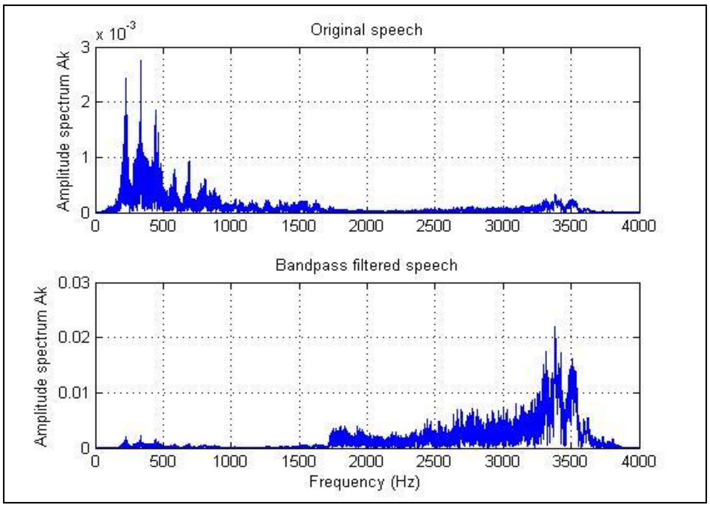

Figure 7. Amplitude spectra of the original speech and band pass filtered speech using rectangular window.

Above figure shows the amplitude spectrum of the band pass filter using the rectangular window. It can be seen that the spectrum is more relevant in the audio frequency range of human hearing (1kHz-4kHz) and the low frequency range signals are suppressed, but not completely which results in some noise and leakage.
We can see that the spectrum is completely flat in the non hearing range of humans and the noise and leakage is absent unlike the band pass filter with rectangular window.

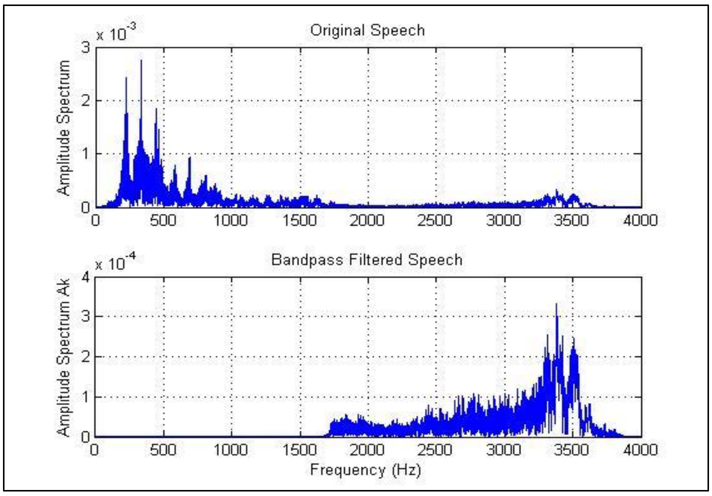

Figure 8. Amplitude spectra of the original speech and band pass filtered speech using Hamming window.

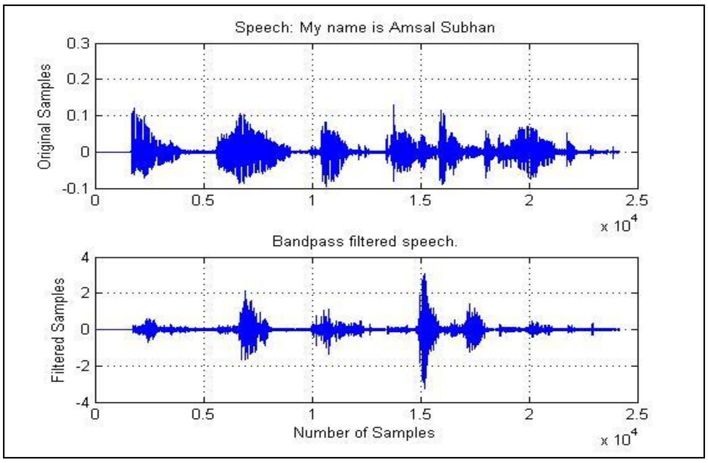

Figure 9. Plots of the original speech and band pass filtered speech using Rectangular window.

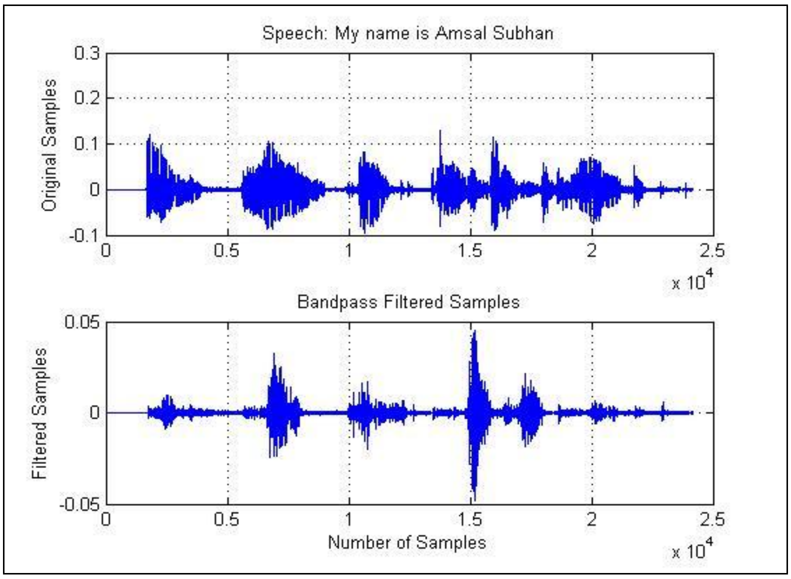

Figure 10. Plots of the original speech and band pass filtered speech using Hamming window.

From the above figure we can see that the noise signal after passing through band-pass filter with hamming window gives a sharp output free from any noise and leakage, unlike the band-pass filter with rectangular window (figure 9) which passes noise and the output plot can be seen much like the input speech signal with noise.

VI. CONCLUSION

From the above analysis we concluded that the band pass filter is more useful in speech filtration and with the use of Hamming or Hanning window we can even eliminate the noise and leakage and can have pure speech signal free from any noise and leakage as a Windowing function minimizes the effect of leakage to better represent the frequency spectrum of the data. A window is shaped so that it is exactly zero at the beginning and end of the data block and has some special shape in between. This function is then multiplied with the time data block forcing the signal to be periodic. A special weighting factor must also be applied so that the correct FFT signal amplitude level is recovered after the windowing.

For band-pass filters rectangular window allows some noise to pass through it but Hamming and Hanning windows are quite useful in suppressing the noise signal and reducing leakage. Practical implementation of ban-pass filter with Hamming window for hearing aid application can be done with some more researches and modifications in the filter specification regarding speech signal.

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