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Performance Comparison of Various Noisy Audio Signals Analysis Using Different Sampling Rates

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ABSTRACT

The discrete time systems that process data at more than one sampling rate are known as multirate systems. The two basic operations in multirate signal processing are decimation and interpolation.One of the important applications of multirate signal processing is sub-band coding of speech signal. In the proposed system, speech signal is taken as input signal. Additive White Gaussian Noise is added with the input speech signal. The input speech signal spectrum is divided into frequency sub-bands using a bank of finite response filters. Hamming, Hanning, Blackman, Rectangular and Kaiser windowing methods are used to implement the low pass and high pass filters. Finally performance of the proposed system is evaluated on the TIMIT data base using the parameters like leakage factor, main lobe width, side lobe attenuation, peak amplitude of side lobe and signal to noise ratio. The performance evaluation shows which window is suitable for designing the finite impulse response filters and sub-band coding system.

Keywords - Windowing, Signal to Noise Ratio, FIR Filter, Multirate.

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I. INTRODUCTION

Multirate means multiple sampling rates. A multirate digital signal processing system uses multiple sampling rates among the system. Whenever a signal at one sampling rate has to be used by a system that expects a different sampling rate, the rate has to be increased or decreased, and some processing is required to do so. Therefore multirate digital signal processing refers to art or science of changing the sampling rates. Systems that use multiple sampling rates in the processing of digital signals are called multirate digital signal processing systems. In telecommunication systems that transmits and receives different kinds of signals as an example teletype, facsimile, speech, video, etc., there is a requirement to process the various signals at different rates. The processing of converting a signal from a given rate to a different rate is called sampling rate conversion.

Multirate digital signal processing systems use a down sampler and an up sampler. The two primary operations that enable the data rate to be altered in an efficient manner are decimation and interpolation. Decimation reduces the sampling rate, effectively compressing the information, and retaining only the required data. Interpolation on the other hand increases the sampling rate of the information subsequently. Multirate signal processing finds its applications in many fields like speech and audio processing, communication systems, antenna systems, radar systems, A/D, D/A converters, sub band coding, voice privacy using analog phone lines etc. During decimation process the sampling rate is reduced from Fs to Fs/M by deleting M-1 samples for every M samples within the original sequence. Interpolation works by inserting L-1 zero-valued samples for every input sample. The sampling rate thus increases from Fs to LFs.

II.LITERATURE SURVEY

Ashraf M. Aziz (2009) proposed a structure of two channel Quadrature Mirror Filter (QMF) with low pass, high pass filters, decimators and interpolators to perform sub-band coding of speech signals in advanced area. The execution of the proposed structure is contrasted with the execution of the delta-modulation encoding systems. The outcomes demonstrate that the proposed structure essentially diminishes the mistake and accomplishes significant execution change contrasted with delta-modulation encoding systems. Maurya A.K. and Deepak Nagaria (2011) exhibited decimation and interpolation procedures of multirate signal processing which are rate change methods. The favourable position is interpolation can change the sampling rate of the signal without changing its unique substance.

Saurabh Singh Rajput and S.S. Bhadauria (2012) designed low pass FIR filter utilizing an effective adjustable window function taking into account Blackman window. The burden of this paper is that Blackman window provides higher side lobe attenuation and the width of the main lobe is marginally more prominent than Hamming window function. Vijayakumar Majjagi (2013) used rate conversion procedures of multirate signal processing to design a sub-band coding arrangement of an ideal four channel Quadrature Mirror Filter (QMF) bank. QMF bank

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permits complete disposal of amplitude and phase distortion of the recreated signal. The remade signal is contrasted with the original input speech signal.

Dolly Agrawal and Divya Kumud (2014) presented multirate signal processing methods, for example, decimation and interpolation by integer factors and after that exhibited how the two procedures can be joined to acquire sampling rate change by any rational component. The impediment of this paper is that the impacts of aliasing for decimation and pseudo images for interpolation are made while designing the multirate systems. Prajoy Podder, Tanvir Zaman Khan, M.Muktadia Rahman and Mamdudul Haque Khan (2014) proposed windowing systems for the comparison of execution of Hamming, Hanning and Blackman window based on their magnitude response, phase response and equivalent noise bandwidth. Looking simulation consequences of various window, Blackman window has best execution among them and the response of the Blackman window are more smooth and perfect. The fundamental downside is that the Blackman window has higher equivalent noise bandwidth.

Jagriti Saini and Rajesh Mehra (2015) displayed comparative examination of speech signal utilizing different windowing methods such as Hamming, Hanning and Blackman window. It can be gotten from the reproduced results that the Blackman window contains almost double power when contrasted with Hamming and Hanning window. So for long distance communication Blackman window is utilized. Suresh Babu, D.Srinivasulu Reddy and P.V.N.Reddy (2015) utilized windowing techniques and the execution of Hamming, Hanning and Blackman windows are mainly compared depending on their magnitude response, phase response. In this paper, on looking at the simulation results utilizing different windows, we watched that Blackman window creates better results among them.

Lalima Singh (2015) built up speech signal analysis strategy taking into account Fast Fourier Transform (FFT) and Linear Predictive Coding (LPC). These techniques are utilized to extract and compress a few elements of the speech signal. The primary restriction of this paper is that the spectrum investigation is complex procedure of breaking down the speech signal into comparative parts. Lalitha R. Naik and Devaraja Naik R L (2015) exhibited a low rate speech coder taking into account sub-band coding technique. This paper mainly focusing the comparison of correlation values for various clean speech signals and correlation values for in the wakenals at more than one of adding high amplitude noise to the same speech signals. Taking correlation tests demonstrate that its execution is fulfilling.

III.METHODOLOGY

Speech signal is taken as the input signal. AWGN is added with the input speech signal. The finite impulse response filters are designed and implemented using different window function. Window function is a mathematical function that is zero-valued outside of some picked interval. When another function is multiplied by a window function, the product is also zero-valued outside the interval. LPF and HPF are designed using Rectangular, Hanning, Hamming, Blackman and Kaiser windows. Then multirate signal processing is performed Finally performance of the proposed system is evaluated based on main lobe width, leakage factor, side lobe attenuation, peak amplitude of side lobe and Signal to Noise Ratio.

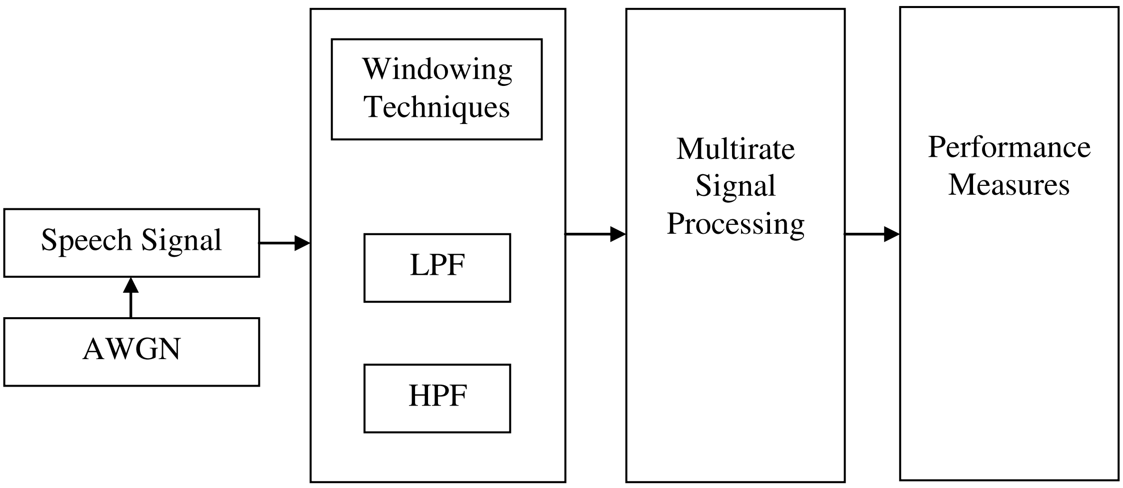

Fig 3.1 Block diagram of proposed method

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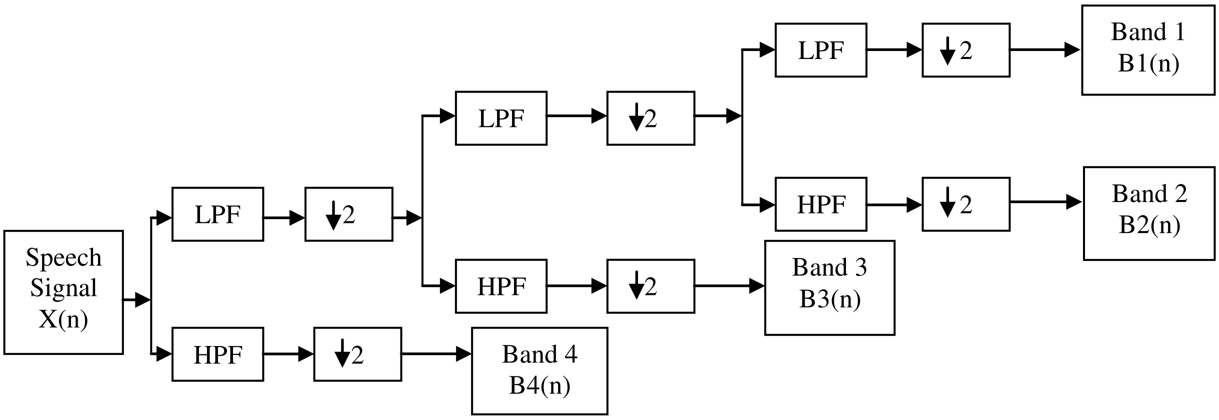

Fig 3.2 Speech signal sub band coding

The above Fig 3.2 shows speech signal sub band coding which has three frequency subdivision. Sub-band coding is a method where the speech signal is subdivided into several frequency bands and each band is digitally encoded separately. The primary frequency subdivision splits the input speech signal into two equal parts, a low pass signal and a high pass signal. Here the low pass signal from the primary stage is divided into two equal bands, a low pass signal and high pass signal. Here the signal is totally divided into four frequency bands. After the frequency subdivision, decimation by a factor of '2' is performed.

3.1 FILTER DESIGN USING FIR

FIR filter means Finite Impulse Response digital filter. This filter has linear phase. It is relatively easy to design and highly stable. This filter is mostly used in different digital signal processing applications. The output can be obtained by [27], [28],

$$y(n) = \sum_{k=0}^{N-1} h(k)\, x(n-k) \qquad (1)$$

Where,

$x(n)$ is the input signal.

$h(n)$ is the impulse response of FIR filter.

The z-transform of impulse response of FIR filter $h(n)$ is obtained by taking the transfer function of a causal FIR filter [27], [28],

$$H(z) = \sum_{k=0}^{N-1} h(k)\; z^{-k} \qquad (2)$$

3.1.1 Low Pass Filter

For the low pass filter the impulse response is given by [27], [28],

$$h_d(n) = \begin{cases} \frac{\sin w_c n}{n\pi} & ; \quad n \neq 0 \\ \frac{w_c}{\pi} & ; \quad n = 0 \end{cases} \qquad (3)$$

3.1.2 High Pass Filter

For the high pass filter the impulse response is given by the equation [27], [28],

$$h_d(n) = \begin{cases} \frac{-\sin w_c n}{n\pi} & ; \quad n \neq 0 \\ 1 - \frac{w_c}{\pi} & ; \quad n = 0 \end{cases} \qquad (4)$$

3.2 WINDOWING METHODS

A simple and efficient way to design an FIR filter is window method. A window is a finite array consisting of coefficients selected to satisfy the desirable requirements. While designing the finite impulse response filter using windowing method it is necessary to specify a window function to be used and the filter order according to the required specifications. These two requirements are interrelated. Each function is a kind of compromise between the two following requirements i.e. the higher the selectivity the narrower the transition region and the higher suppression of undesirable spectrum the higher the stop band attenuation. The main aim of a window function is to provide accurate type of responses with reduced side lobes and comparatively less pass-band and stop-band ripples. The Window method is the most popular and effective method because this method is simple, convenient, fast and easy to understand.

3.2.1 Rectangular Window

The rectangular window is expressed by using the below forumula which is given by,

$$w(n) = \begin{cases} 1 & ; \quad 0 \le n \le N-1 \\ 0 & ; \quad otherwise \end{cases} \qquad (5)$$

Where,

N is the order of the filter.

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3.2.2 Hanning Window

The hanning window is expressed by using the below formula which is given by [27], [28],

$$w(n) = \begin{cases} 0.5 - 0.5\, cos\left(\frac{2\pi n}{N-1}\right) & ; \quad 0 \le n \le N-1 \\ 0 & ; \; otherwise \end{cases} \tag{6}$$

3.3.3 Hamming Window

The hamming window is expressed by using the below formula which is given by [27], [28],

$$w(n) = \begin{cases} 0.54 - 0.46 \cos\left(\frac{2\pi n}{N-1}\right) & ; \quad 0 \le n \le N-1 \\ 0 & ; \quad othertwise \end{cases} \tag{7}$$

3.2.4 Blackman Window

The blackman window is expressed by using the below formula which is given by [27], [28],

$$w(n) = \begin{cases} 0.42 - 0.5\, cos\left(\frac{2\pi n}{N-1}\right) + 0.08\, cos\left(\frac{4\pi n}{N-1}\right); & 0 \le n \le N-1 \\ 0 & ; \quad otherwise \end{cases} \tag{8}$$

3.2.5 Kaiser Window

The Kaiser window with parameter α is expressed by the below formula which is given by [27], [28],

$$w(n) = \begin{cases} \frac{I_0\left[\alpha\sqrt{1-\left(\frac{2n}{N-1}-1\right)^2}\right]}{I_0(\alpha)} & ; \quad 0 \le n \le N-1 \\ 0 & ; \quad otherwise \end{cases} \tag{9}$$

Where,

α is the adjustable parameter which is used to determine the shape of the window and thus controls the trade-off between main-lobe width and side-lobe amplitude.

$I_0(\alpha)$ is the modified zeroth-order Bessel function of first kind.

3.3 PERFORMANCE MEASURES

The performance of the windowing techniques are evaluated by using leakage factor, main lobe width, side lobe attenuation, peak amplitude of side lobe, signal to noise ratio.

3.3.1 Leakage Factor

Leakage factor is the ratio of power in the side lobes to the total power in the window spectrum..

$$Leakage\ Factor\ (\%) = \frac{Power\ in\ side\ lobes}{Total\ Power} \tag{10}$$

3.3.2 Main Lobe Width

The point at which the power falls -3dB below the peak power is known as main lobe width. One of the important characteristics of the frequency response of window function is that the width of the main lobe should be small and it should contain as much of the total energy as possible.

3.3.3 Side Lobe Attenuation

Side lobe attenuation is the difference between the power of the main lobe peak and peak power in the side lobes. It is usually expressed in unit called decibels.

$$Attenuation = peak\ power\ in\ the\ main\ lobe - peak\ power\ in\ the\ side\ lobes \tag{11}$$

3.3.4 Peak Amplitude of Side Lobe

Peak amplitude of side lobe represents maximum side lobe magnitude in the window spectrum. One of the important characteristics of the frequency response of window function is that the side lobes should have very low magnitude for large attenuation in the frequency spectrum.

3.3.5 Signal to Noise Ratio

It is defined as the ratio of signal power to the noise power which can be expressed in decibels. The SNR is given by,

$$SNR = 10\ \log_{10} \frac{\sum_{n=0}^{M-1} S_i^{\,2}(n)}{\sum_{n=0}^{M-1} N_i^{\,2}(n)} \tag{12}$$

Where,

$S_i(n)$ is the input signal power.

$N_i(n)$ is the noise signal power.

IV.RESULTS AND DISCUSSION

The voice sample of the man is taken for analyzing the speech signal from the TIMIT (Texas Instruments and Massachusetts Institute of Technology) data base. TIMIT data base was compiled using 630 speakers and each speaker spoke ten utterances giving a total of 6300 sentences.

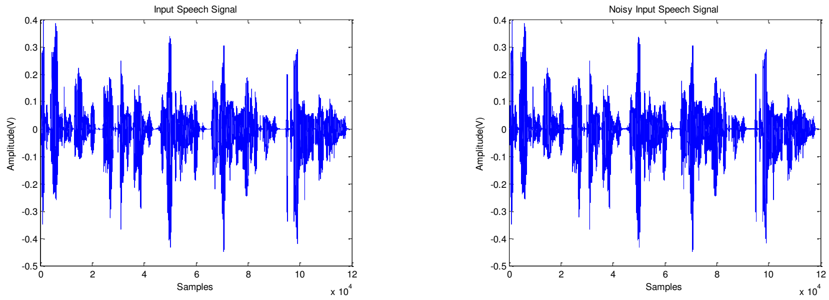

Fig 4.1 Original and noisy input of speech signal 1

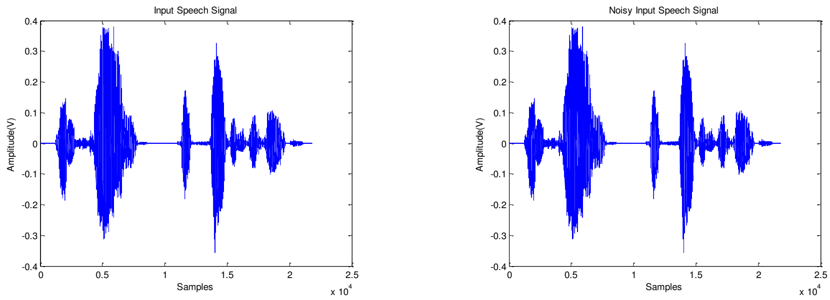

Fig 4.2 Original and noisy input of speech signal 2

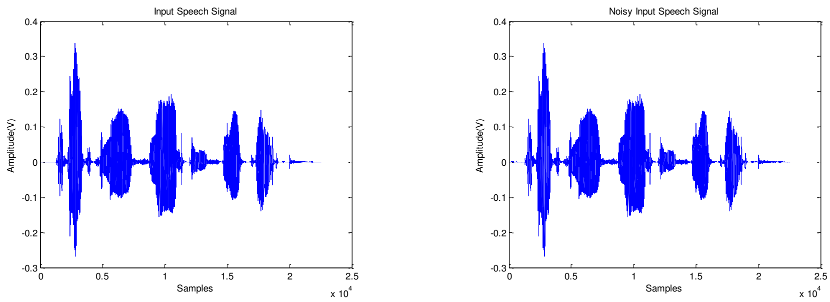

Fig 4.3 Original and noisy input of speech signal 3

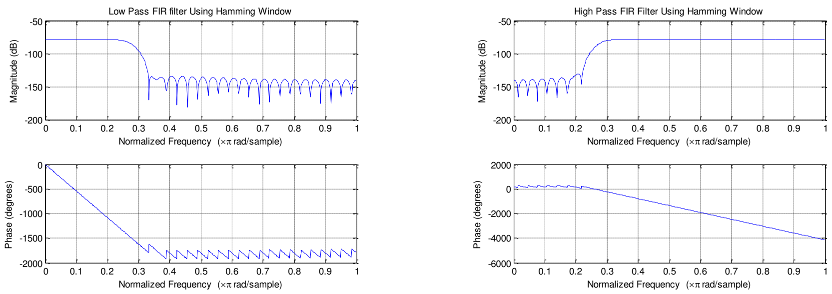

Fig 4.4 Response of LPF and HPF using Hamming Window

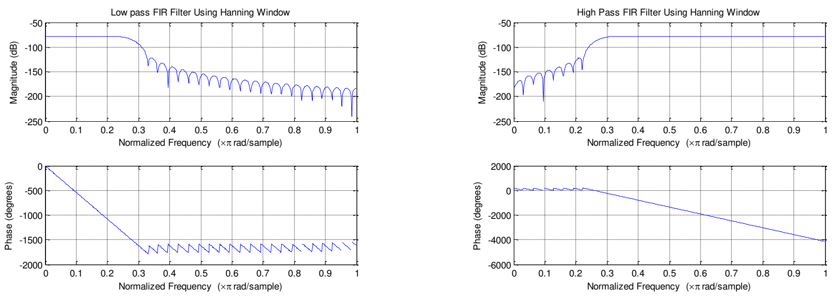

Fig 4.5 Response of LPF and HPF using Hanning Window

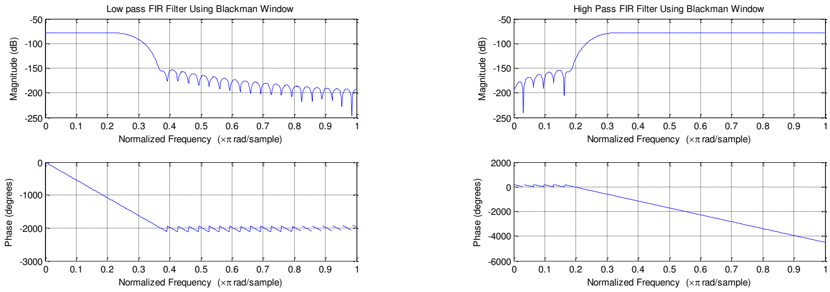

Fig 4.6 Response of LPF and HPF using Blackman Window

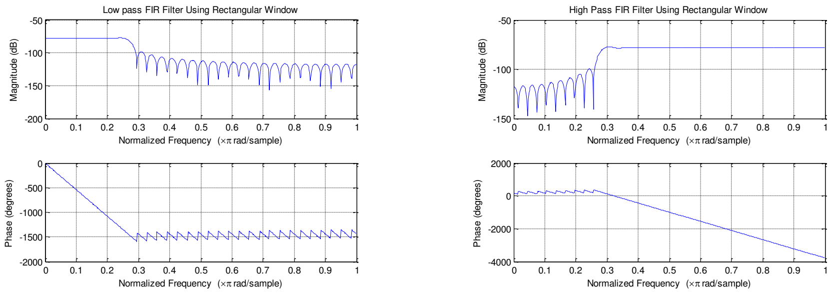

Fig 4.7 Response of LPF and HPF using Rectangular Window

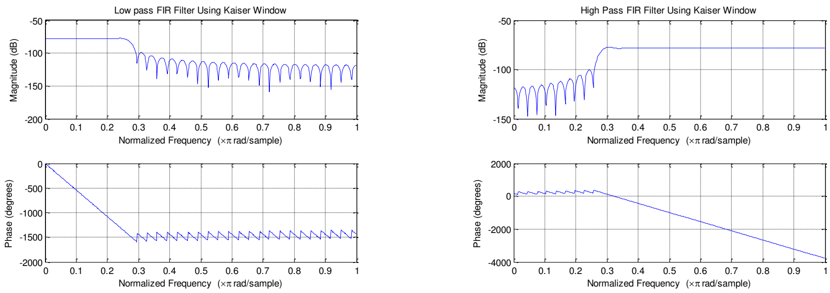

Fig 4.8 Response of LPF and HPF using Kaiser Window

Fig 4.1, 4.2, 4.3 shows the original and noisy input signal representation of three speech signals which was spoken by a man. The voice is recorded and it is stored as a wave file for further usage in MATLAB.

Fig 4.4, 4.5, 4.6, 4.7 and 4.8 shows the magnitude and phase response of low pass and high pass filters using Hamming, Hanning, Blackman, Rectangular and Kaiser windows. The peak amplitude of side lobe for Hamming, Hanning, Blackman, Rectangular and Kaiser windows are -135dB, -125dB, -155dB, -100dB and -100dB respectively.

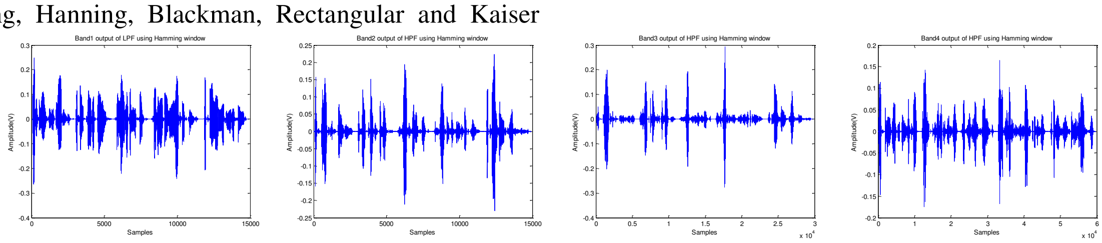

Fig 4.9 Band 1, 2, 3 and 4 outputs of Hamming window for speech signal1

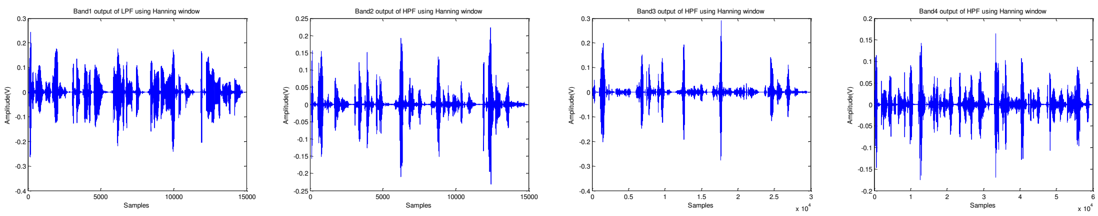

Fig 4.10 Band 1, 2, 3 and 4 outputs of Hanning window for speech signal1

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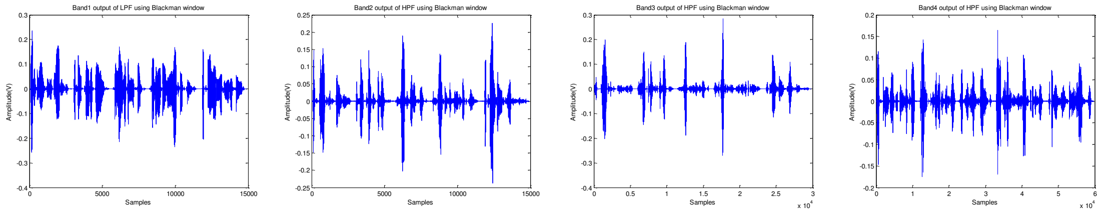

Fig 4.11 Band 1, 2, 3 and 4 outputs of Blackman window for speech signal1

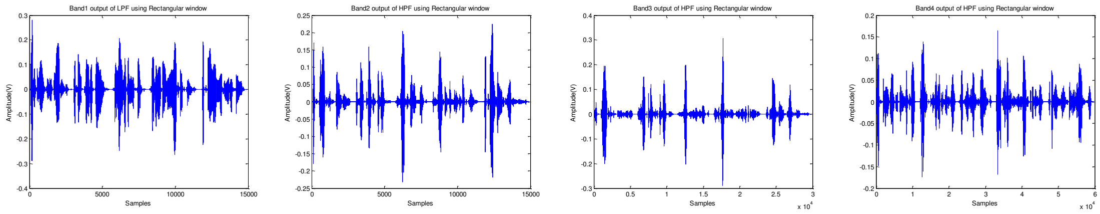

Fig 4.12 Band 1, 2, 3 and 4 outputs of Rectangular window for speech signal1

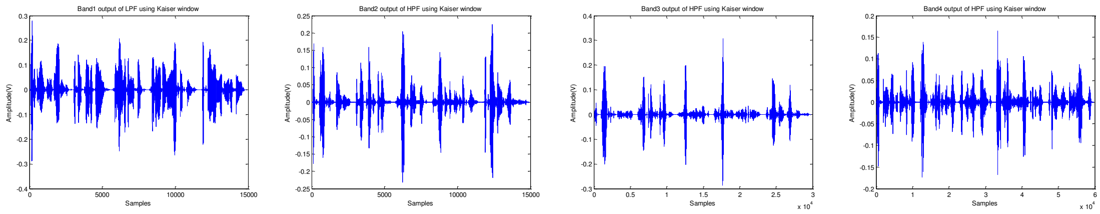

Fig 4.13 Band 1, 2, 3 and 4 outputs of Kaiser window for speech signal1

From the Fig 4.9, 4.10, 4.11, 4.12 and 4.13 it is observed that most of the information is present in the band 1. The band 2 contains little less information and also band 2 signals slightly deviates from the original signal. In band 3 the amount of information that is present is very less and also the information is scattered. In band 4 most of the signal that is present is noise and amplitude levels of this band are also less. Since most of the information is present in the lower frequency band 1, this band almost resembles the original signal.

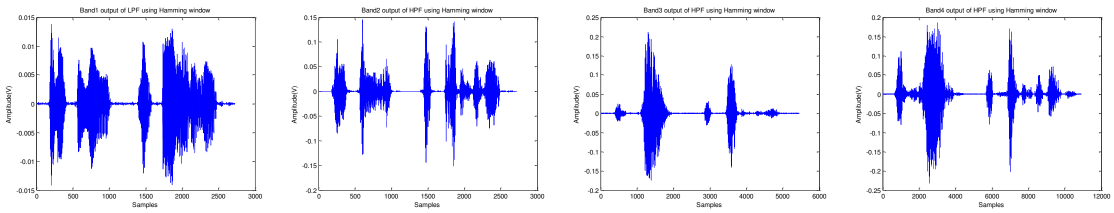

Fig 4.14 Band 1, 2, 3 and 4 outputs of Hamming window for speech signal2

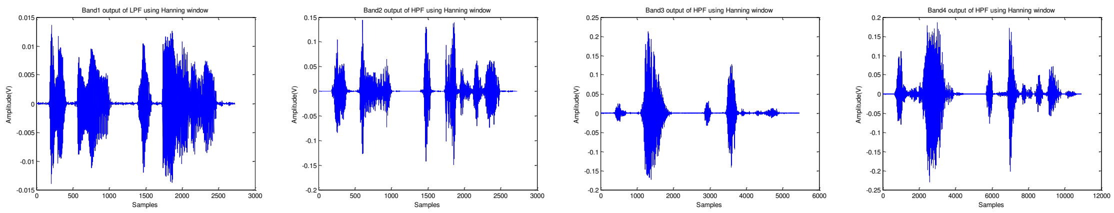

Fig 4.15 Band 1, 2, 3 and 4 outputs of Hanning window for speech signal2

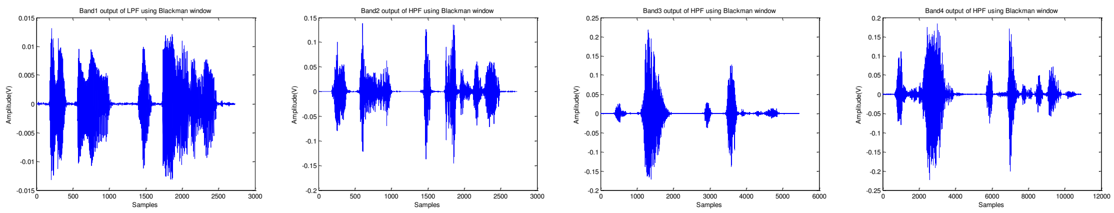

Fig 4.16 Band 1, 2, 3 and 4 outputs of Blackman window for speech signal2

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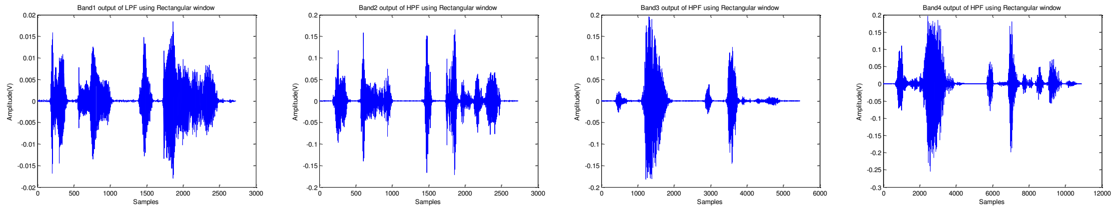

Fig 4.17 Band 1, 2, 3 and 4 outputs of Rectangular window for speech signal2

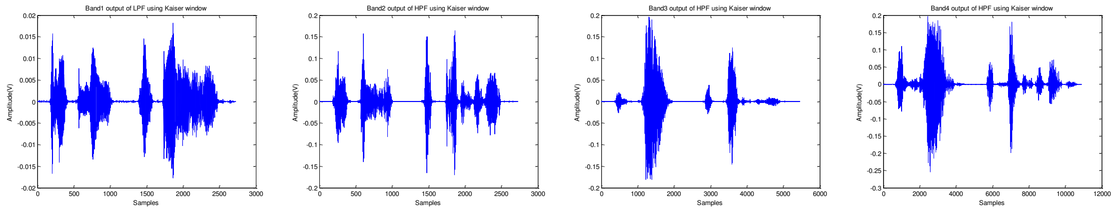

Fig 4.18 Band 1, 2, 3 and 4 outputs of Kaiser window for speech signal2

From the Fig 4.14, 4.15, 4.16, 4.17 and 4.18 it is observed that most of the information is present in the band 1. The band 2 contains little less information and also band 2 signals slightly deviates from the original signal. In band 3 the amount of information that is present is very less and also the information is scattered. In band 4 most of the signal that is present is noise and amplitude levels of this band are also less. Since most of the information is present in the lower frequency band 1, this band almost resembles the original signal.

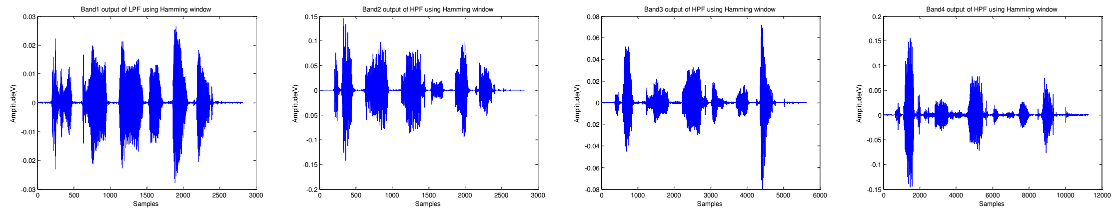

Fig 4.19 Band 1, 2, 3 and 4 outputs of Hamming window for speech signal3

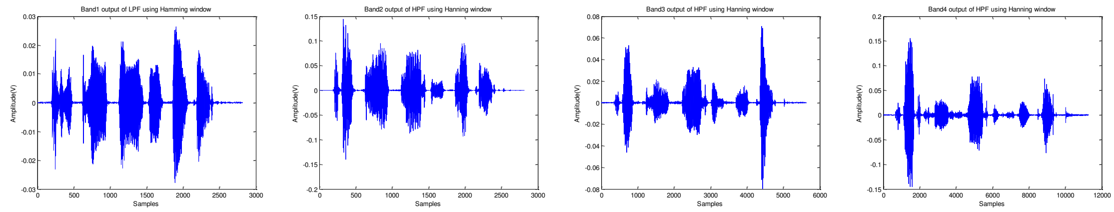

Fig 4.20 Band 1, 2, 3 and 4 outputs of Hanning window for speech signal3

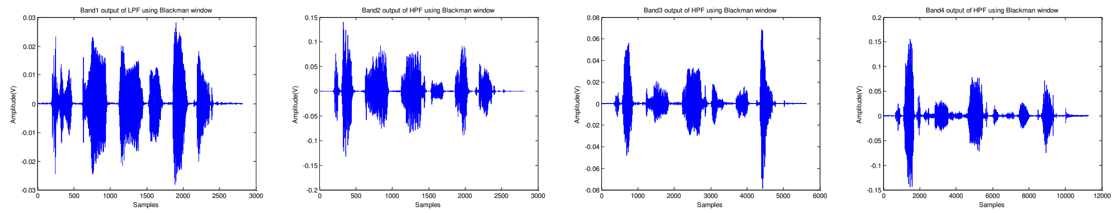

Fig 4.21 Band 1, 2, 3 and 4 outputs of Blackman window for speech signal3

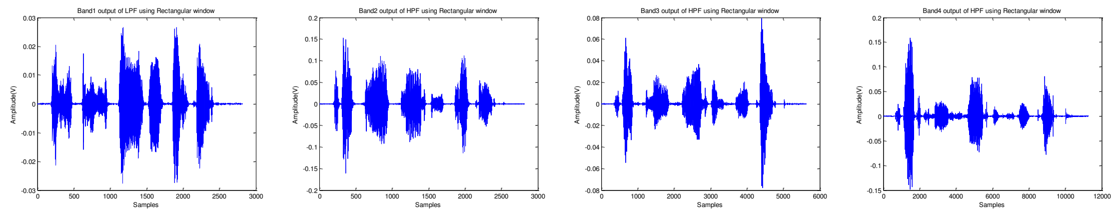

Fig 4.22 Band 1, 2, 3 and 4 outputs of Rectangular window for speech signal3

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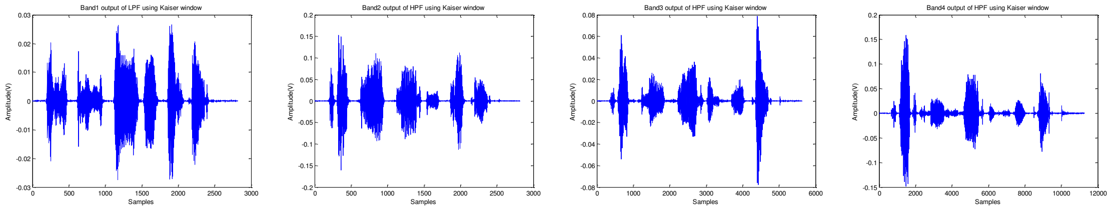

Fig 4.23 Band 1, 2, 3 and 4 outputs of Kaiser window for speech signal3

From the Fig 4.19, 4.20, 4.21, 4.22 and 4.23 it is observed that most of the information is present in the band 1. The band 2 contains little less information and also band 2 signals slightly deviates from the original signal. In band 3 the amount of information that is present is very less and also the information is scattered. In band 4 most of the signal that is present is noise and amplitude levels of this band are also less. Since most of the information is present in the lower frequency band 1, this band almost resembles the original signal. The performances of the different windows are evaluated by measuring the leakage factor, main lobe width, side lobe attenuation, peak amplitude of side lobe and Signal to Noise Ratio. Here the performance of Hamming, Hanning, Blackman, Rectangular and Kaiser windows are compared for both the low pass and high pass FIR filters.

Table 4.1 Comparison of leakage factor for different windowing methods

Windowing Methods	**Leakage Factor (%)**	
	LPF	**HPF**
Hamming	99.26	100
Hanning	85.11	100
Blackman	84.91	100
Rectangular	99.3	100
Kaiser	99.3	100

Table 4.2 Comparison of main lobe width for different windowing methods

Windowing Methods	**Main Lobe Width (-3dB)**	
	LPF	**HPF**
Hamming	0.51953	1.9961
Hanning	0.51953	1.9961
Blackman	0.51172	1.9961
Rectangular	0.53516	1.9961
Kaiser	0.53516	1.9961

Table 4.1 and 4.2 shows the comparison of leakage factor and main lobe width for different windowing methods such as Hamming, Hanning, Blackman, Rectangular and Kaiser. From the comparison it is observed that Blackman window has minimum leakage factor and smaller main lobe width when compared to other windowing methods.

.Table 4.3 Comparison of side lobe attenuation for different windowing methods

Windowing Methods	**Side Lobe Attenuation (dB)**	
	LPF	**HPF**
Hamming	0	61
Hanning	0.1	105.3
Blackman	0	116.7
Rectangular	0.8	39.9
Kaiser	0.7	40.5

Table 4.4 Comparison of peak amplitude of side lobe for different windowing methods

Windowing Methods	**Peak Amplitude of Side Lobe (dB)**	
	LPF	**HPF**
Hamming	-135	-135
Hanning	-125	-125
Blackman	-155	-155
Rectangular	-100	-100
Kaiser	-100	-100

Table 4.3 and 4.4 shows the comparison of side lobe attenuation and peak amplitude of side lobe for different windowing methods such as Hamming, Hanning, Blackman, Rectangular and Kaiser. From the comparison it is observed that Hamming and Blackman window have minimum side lobe attenuation for the LPF and Blackman window has minimum peak amplitude of side lobe for both the LPF and HPF.

Table 4.5 Comparison of signal to noise ratio of different windowing methods for speech signal 1

Windowing Methods	**Signal to Noise Ratio (dB)**			
	Band 1	**Band 2**	**Band 3**	**Band 4**
Hamming	12.0196	15.4032	14.2775	14.6716
Hanning	12.0638	15.4452	14.3096	14.6662
Blackman	12.1626	15.6137	14.4682	14.6829
Rectangular	11.4783	14.8236	13.8619	14.7123
Kaiser	11.5023	14.8469	13.8782	14.7092

Table 4.6 Comparison of Signal to Noise Ratio of different windowing methods for speech signal 2

Windowing Methods	Signal to Noise Ratio (dB)			
	Band 1	Band 2	Band 3	Band 4
Hamming	30.8282	14.7157	11.7933	7.7887
Hanning	30.8056	14.7780	11.8018	7.8552
Blackman	30.8222	15.0322	11.8498	8.1252
Rectangular	30.4593	13.9377	11.4457	6.9922
Kaiser	30.4993	13.9677	11.4665	7.0208

Table 4.7 Comparison of Signal to Noise Ratio of different windowing methods for speech signal 3

Windowing Methods	Signal to Noise Ratio (dB)			
	Band 1	Band 2	Band 3	Band 4
Hamming	23.3328	11.8081	18.3550	10.7771
Hanning	23.2459	11.8749	18.3457	10.7775
Blackman	22.8583	12.1366	18.3814	10.8609
Rectangular	23.6469	11.0254	18.0072	10.7427
Kaiser	23.6925	11.0533	18.0301	10.7412

Table 4.5, 4.6 and 4.7 shows the comparison of Signal to Noise Ratio for different windowing methods such as Hamming, Hanning, Blackman, Rectangular and Kaiser. From the comparison it is observed that Blackman window has maximum Signal to Noise Ratio values when compared to other windowing methods.

V. CONCLUSION

In this paper, the bank of finite impulse response filters are used to design the sub-band coding system. Here different windowing methods are used to implement the low pass and high pass finite impulse response filters. Different windowing methods are Hamming, Hanning, Blackman, Rectangular and Kaiser windows. The performance of the different window methods are evaluated based on leakage factor, main lobe width, side lobe attenuation, peak amplitude of side lobe and signal to noise ratio. Finally, the results of different windows are compared and it is observed that Blackman window has minimum leakage factor, side lobe attenuation and peak amplitude of side lobe, smaller main lobe width and provides maximum signal to noise ratio values when compared to other windowing methods.

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