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DESIGN AND ANALYSIS OF DIGITAL FILTERS FOR SPEECH SIGNALS USING MULTIRATE SIGNAL PROCESSING

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Abstract
Digital filters provide an important role in the world of communication. This paper proposes the design of digital filters for audio application using multi rate signal processing. One of the important applications in multi rate signal processing is sub band coding. The main objective of this paper is to analyze various techniques for designing digital filters for speech signals. Additive White Gaussian Noise is added with the input speech signal. The input speech signal spectrum is divided into frequency sub-bands using down sampling by a factor 2. Various transforms like FFT, FWHT and DWT are applied to the signal and its sub bands. Then the low pass and high pass FIR filters are designed and implemented using windowing techniques and IIR filters are designed and implemented using Butterworth and Chebyshev filters. Finally quantization is performed on the filter coefficients of signal and its sub bands. The performances of digital filters are measured by calculating Signal to Quantization Noise Ratio. From the performance measures this paper concludes that, which filtering technique is most suitable for designing digital filters for speech signals.

Keywords:
Digital Filters, Sub Band Coding, FIR, IIR, DWT, FFT, FWHT, Quantization, SQNR

1. INTRODUCTION

Speech is the most basic and preferred means of communication among humans. In speech processing, a filter removes the unwanted signal and allows the desired signal. Filters may be analog or digital. Digital filtering is one of the important tools for digital signal processing applications. Digital filters are capable of performing that specifications which are extremely difficult, to achieve with an analog implementation. Multiple filtering is possible and it can be operated over wide range of frequencies, because the characteristics of digital filters can be easily changed under software control. Digital filters are classified either as Finite duration impulse response (FIR) filters or Infinite duration impulse response (IIR) filters, depending on the form of impulse response of the system.

In this paper, these FIR and IIR filters for speech signals are designed and implemented using multi rate signal processing. The systems that employ multiple sampling rates in the processing of signals are called multi rate signal processing systems. Multirate digital signal processing systems use down sampler and up sampler, the two basic sampling rate alteration devices in addition to conventional elements like adder, multiplier and delay to change the sampling rate of a digital signal.

2. LITERATURE SURVEY

Saini, et al. [4] displayed comparative analysis of speech signal utilizing different windowing methods such as Hamming, Hanning and Blackman window. For investigation, first of all a signal out of audio frequency range is chosen and after that a little partition of this signal is extracted utilizing framing technique. The resulting signal frame is gone through Hamming, Hanning and Blackman window and their particular power spectral densities are calculated. To investigate power content of signal FFT is utilized. It can be obtained from the simulated 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. The drawback of this paper is that the losses happen while travelling long distance.

Babu, et al. [14] utilized windowing techniques and the performance of Hamming, Hanning and Blackman windows are mainly compared depending on their magnitude response, phase response for designing the FIR low pass and high pass filters using matlab. The responses of FIR low pass, high pass filters can be obtained from its design. In this paper, on looking at the simulation results utilizing different windows, it is observed that the Blackman window creates better results among them and the response of Blackman window are more smooth and perfect.

Singh [5] built up speech signal analysis strategy taking into account Fast Fourier Transform (FFT) and Linear Predictive Coding (LPC). In this paper five samples of single word are taken by same individual. These samples are examined utilizing FFT and LPC as a part of matlab. After investigation various parameters of tests are gotten for FFT and LPC range exclusively. The primary restriction of this paper is that the spectrum analysis is a complex process of decomposing the speech signal into similar parts.

Naik, et al. [6] exhibited a very low bit rate speech coder taking into account sub-band coding technique. The audible frequency range 20Hz-20kHz is split into frequency sub-bands utilizing a bank of finite impulse response filter. Then the output of every filter is sampled and encoded. At the output, the signals are de-multiplexed, decoded and demodulated and afterwards summed to recreate the signal. This paper mainly focusing the comparison of correlation values for various clean speech signals and correlation values for after adding high amplitude noise to the same speech signals. Taking correlation tests demonstrate that its execution is fulfilling.

Podder, et al. [7] proposed windowing techniques for the comparison of performance of Hamming, Hanning and Blackman window based on their magnitude response, phase response and equivalent noise bandwidth in time and frequency domain utilizing matlab simulation. To observe the responses, FIR filter of low pass, high pass, band pass and band stop filters have been designed. Looking simulation consequences of various windows, Blackman window has best execution among them and the response of the Blackman window is more smooth and perfect when compared with Hamming and Hanning windows. The fundamental downside is that the Blackman window has higher equivalent noise bandwidth.

3. PROPOSED METHOD

The speech signal is taken as the input signal. AWGN is added with the input speech signal. The noisy speech signal spectrum is down sampled into multiple sampling rates using sampling rate conversion. Various transforms like Fast Fourier Transform (FFT), Fast Walsh Hadamard Transform (FWHT) and Discrete Wavelet Transform (DWT) are applied to the noisy speech signal and its sub bands. The FIR filters are designed and implemented using different window functions such as Rectangular, Hanning, Hamming, Blackman and Kaiser windows and the IIR filters are designed using Butterworth and Chebyshev filters. Then Quantization is applied to the filter coefficients. Finally the performance of the filter coefficients are measured based on the Signal to Quantization Ratio (SQNR). The Fig.1 shows the overall process of the proposed method.

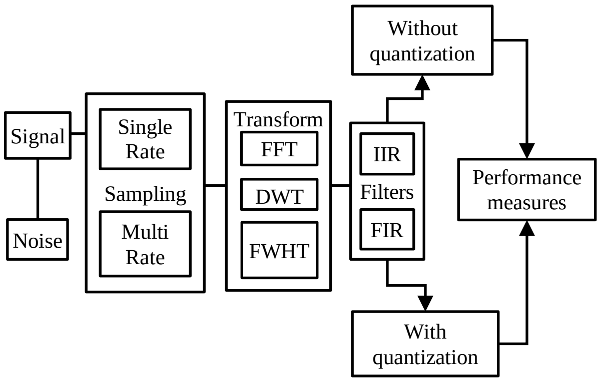

Fig.1. Block diagram of proposed method

3.1 SAMPLING RATE CONVERSION

The noisy speech signal spectrum is divided into four frequency sub bands by down sampling by a factor of 2. The below Fig.2 shows the down sampler.

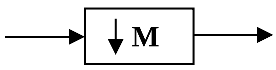

Fig.2. Downsampler

Input Sampling Frequency = $F = 1/T$

Output Sampling Frequency = $F' = F/M = 1/MT$

3.2 TRANSFORM

In speech processing, the speech signal should be first transformed and compressed for further processing. They are not necessary, but they do make some calculations much simpler and more convenient. It is possible to do all computation and analysis of a signal in either time domain or the frequency domain. However some operations are much simpler and more intuitive in one than the other. Transforms are the tools to convert time domain into frequency domain. There are many techniques available which are used to extract the important features and compress the signal without losing any important information. Some of the techniques are FFT, DWT and FWHT.

3.2.1 Fast Fourier Transform (FFT):

FFT algorithm computes the DFT of a sequence or its inverse. It is an efficient algorithm to compute DFT and its inverse which compute the same result quickly. Because DFT requires N^2 operation for N-point sequence while FFT requires only $N\log N$ operation.

Formula to compute FFT is,

$$X_k = \sum_{n=0}^{N-1} x_n e^{-i2\pi nk/N}; k = 0,1,....N-1 \tag{1}$$

where,

k - index of frequencies, n - index of signal and N – bandwidth.

3.2.2 Fast Walsh Hadamard Transform (FWHT):

FWHT is an efficient algorithm to compute the WHT (Walsh Hadamard Transform). FWHT is a divide and conquer algorithm that recursively breaks down a WHT of size n into smaller WHT of size $n/2$. WHT is a non-sinusoidal orthogonal transform that uses Walsh function which is determined from the number of zero crossing per unit time interval. Walsh function is rectangular or square value and real value. It requires only $N\log N$ additions or subtractions.

Formula to compute FWHT

$$X[k] = \sum_{m=0}^{N-1} x[m] \prod_{i=0}^{n-1} (-1)^{(k_i + k_{i+1}) m_{n-1-i}} \tag{2}$$

3.2.3 Discrete Wavelet Transform (DWT):

In numerical analysis and functional analysis, a Discrete Wavelet Transform is any wavelet transform for which the wavelets are discretely sampled. A key advantage of wavelet transform over Fourier transform is temporal resolution. Temporal resolution-it captures both frequency and location information (location in time).

3.3 FILTERING TECHNIQUES

Filtering techniques are useful for separating, extracting and restoring the signal. It helps to remove the sources of noise. There are two types of digital filtering techniques.

- FIR filtering technique
- IIR filtering technique

3.3.1 FIR Filter Design:

In FIR Filter the present output depends on present input and past input. The impulse response of continuous time signal is sampled into impulse response of discrete time signal. There is no feedback required for implementing FIR filter. Therefore FIR filters are called Non Recursive Filters. FIR filters are always stable and finite. Here, FIR Filters are designed using Windowing Methods. A Window function is a mathematical function that is zero valued outside of some chosen interval. The basic principle of window function is to calculate $h_d(n)$ by the Fourier transform based on the filter frequency response.

$$h_d(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(e^{jw}) \tag{3}$$

Then $h(n)$ can be obtained by multiplying the window sequence, w(n) with the $h_d(n)$.

$$h(n) = w(n) \cdot h_d(n) \tag{4}$$

For low pass filter, $h_d(n)$ is given by,

$$h_d(n) = \frac{\sin w_c(n)}{n\pi} \tag{5}$$

For high pass filter, $h_d(n)$ is given by,

$$h_d(n) = \frac{-\sin w_c(n)}{n\pi} \tag{6}$$

The common types of windowing methods are,

- **Hamming Window:** Hamming Window is the raised cosine window which minimizes the maximum side lobe and provide a height of about one fifth of Hanning Window.

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

- **Hanning Window:** Hanning window is the raised cosine window which has minimum stop band attenuation. It can be used to reduce the side lobe while preserving the good frequency resolution. Hanning window touches zero at both ends and removes any discontinuities.

$$w(n) = \begin{cases} 0.5 - 0.5\cos\left(\dfrac{2n\pi}{N-1}\right) & 0 \le n \le N-1 \\ 0 & otherwise \end{cases} \tag{8}$$

- **Blackman Window:** Blackman Window is similar to Hamming and Hanning Window and the width of the main lobe is low when compared to Hamming and Hanning Window.

$$w(n) = \begin{cases} 0.42 - 0.5\cos\left(\dfrac{2n\pi}{N-1}\right) + 0.08\cos\left(\dfrac{4n\pi}{N-1}\right) & 0 \le n \le N-1 \\ 0 & otherwise \end{cases} \tag{9}$$

- **Rectangular Window:** Rectangular Window is the simplest window, taking a chunk of the signal without any modification.

$$w(n) = \begin{cases} 1 & 0 \le n \le N-1 \\ 0 & otherwise \end{cases} \tag{10}$$

- **Kaiser Window:** Kaiser Window maximizes the ratio of main lobe energy to side lobe energy. In Kaiser Window for a particular length, the particular parameter controls the side lobe height.

$$w(n) = \begin{cases} \alpha\sqrt{1 - \left(\dfrac{2n}{N-1} - 1\right)^2} & 0 \le n \le N-1 \\ 0 & otherwise \end{cases} \tag{11}$$

- **IIR Filter Design:** A filter whose impulse response is infinite can be considered as an IIR filter and it has a feedback loop. The disadvantages of IIR filter are that they usually have nonlinear phase. The transfer function of the IIR Filter is given by,

$$H(z) = \sum_{n=0}^{\alpha} h(n) z^{-n} \tag{12}$$

The commonly used filters are,

- Butterworth filters - No ripples at all,
- Chebyshev filters - Ripples in pass band or stop band, and
- Elliptic filter - Ripples in both pass band and stop band.

The magnitude function of the Butterworth filter is given by,

$$|H(j\Omega)| = \frac{1}{\left[1 + \left(\dfrac{\Omega}{\Omega_c}\right)^{2N}\right]^{0.5}} \tag{13}$$

The magnitude square function of the Chebyshev filter is given by,

$$|H(j\Omega)|^2 = \frac{1}{1 + \varsigma^2 C_N{}^2\left(\dfrac{\Omega}{\Omega_P}\right)} \tag{14}$$

where,

N - Order of the filter

Ω_c - Cut off frequency.

Ω_p - Pass band frequency

C_N - $\cos(N\cos^{-1}x)$, $|x| \le 1$ (Pass band)

C_N - $\cos(N\cos^{-1}x)$, $|x| > 1$ (Stop band)

3.4 QUANTIZATION

Quantization is the process of converting Discrete Time Continuous Amplitude into Discrete Time Discrete Amplitude signal. There are two types of quantization namely, truncation and rounding

The quantization error is given by,

$$e(n) = x_q(n) - x(n) \tag{15}$$

where,

$x_q(n)$ - Quantized sample of the signal.

$x(n)$ - Unquantized sample of the signal.

3.5 PERFORMANCE MEASURES

3.5.1 Signal to Quantization Noise Ratio (SQNR):

SQNR can be applied to the Quantized signal. SQNR can be calculated between the Normalized Signal power and the Quantization Noise power.

Signal to Quantization Noise Ratio (SQNR) is given by,

$$SQNR = 10\log_{10}\left(\frac{\text{Normalized Signal Power}}{\text{Quantization Noise Power}}\right) \tag{16}$$

4. RESULTS AND DISCUSSIONS

The experimental results of the proposed methods are shown in the below figures. The voice sample of the male is taken for analysis of 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.

4.1 INPUT SIGNAL

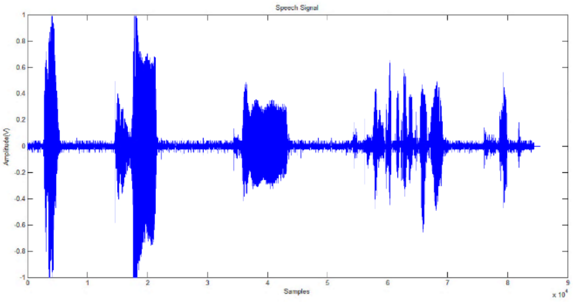

Fig.3. Speech signal

The Fig.3 shows the original signal representation of speech signal which was spoken by a man. The voice is recorded and it is stored as a wave file for further usage in matlab. The frequency of the input signal is 8000Hz.

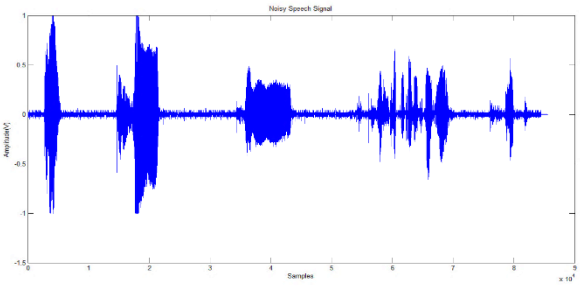

Fig.4. Noisy Speech signal

The Fig.4 shows the noisy input signal representation which is treated as the original signal. Noisy speech signal is the AWGN added speech signal.

4.2 SAMPLING RATE CONVERSION

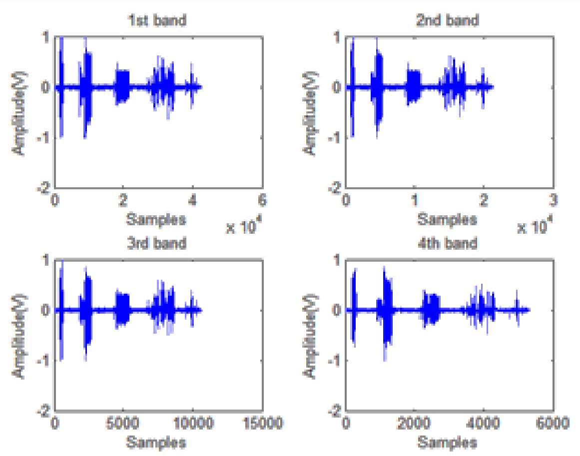

Fig.5. Sub bands of original signal

The Fig.5 shows the sub bands of original signal which is a down sampled signal of original speech signal by a factor 2. Band 1 has the frequency of 4000Hz, Band 2 has the frequency of 2000Hz and so on.

4.3 TRANSFORM

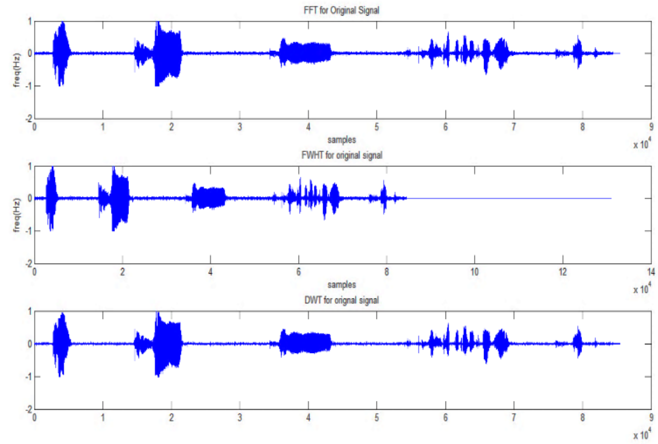

Fig.6. Various Transforms applied Original signal

The Fig.6 shows the responses of original signal transformed by various transforms techniques such as FFT, FWHT and DWT.

4.4 FIR FILTERING

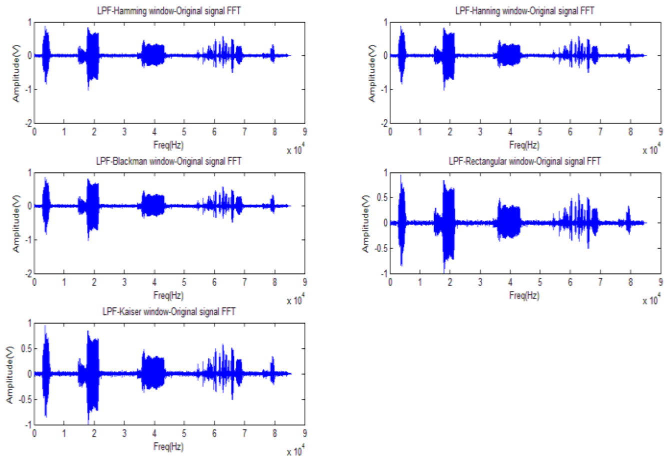

Fig.7. FFT applied Original signal filtered by FIR-LPF

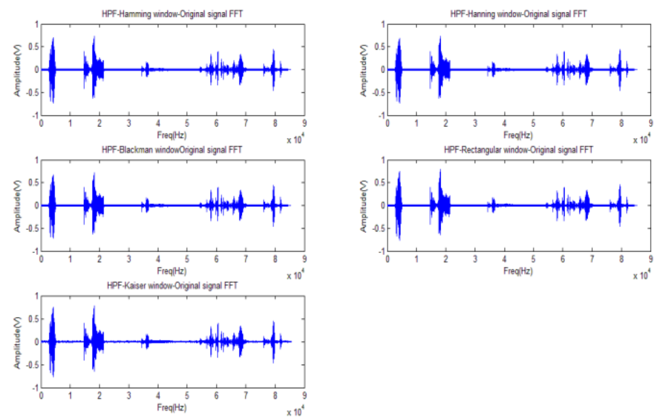

Fig.8. FFT applied Original signal filtered by FIR-HPF

The Fig.7 and Fig.8 shows the responses of FIR LPF and HPF designed by different windowing such as Hamming, Hanning, Blackman, Rectangular and Kaiser Window techniques of FFT applied original signal. From these windows Blackman HPF provides high SQNR than other windowing techniques.

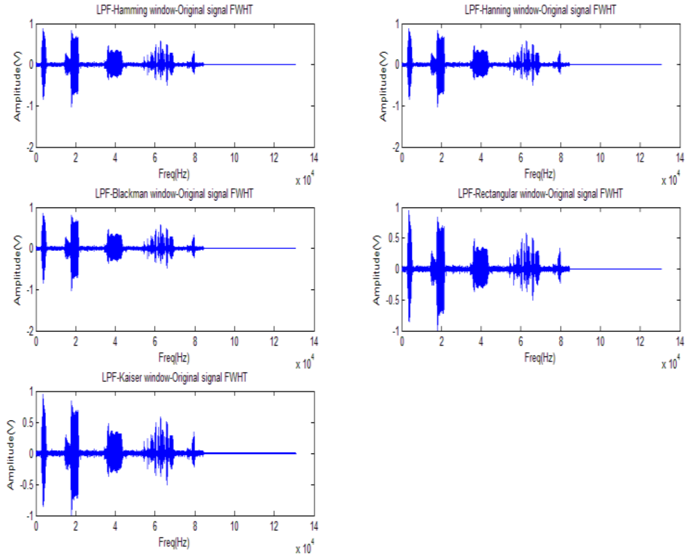

Fig.9. FWHT applied Original signal filtered by FIR-LPF

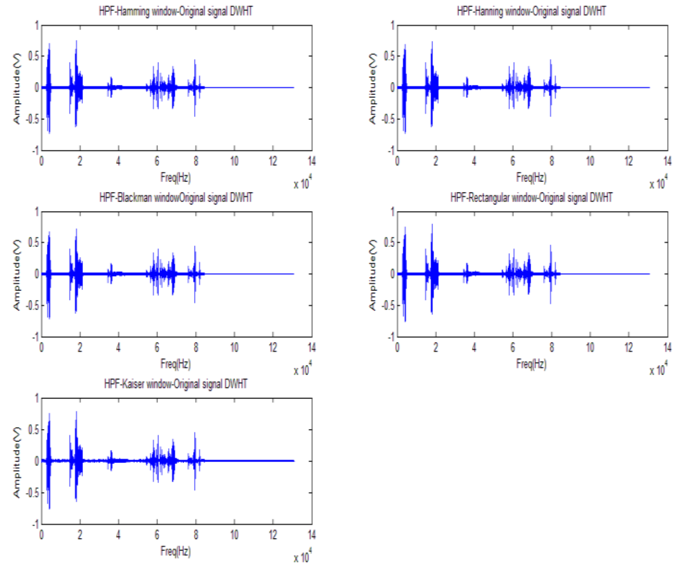

Fig.10. FWHT applied Original signal filtered by FIR-HPF

The Fig.9 and Fig.10 show the responses of FIR LPF and HPF designed by different windowing such as Hamming, Hanning, Blackman, Rectangular and Kaiser Window techniques of FWHT applied original signal. From these windows Blackman HPF provides high SQNR than other windowing techniques.

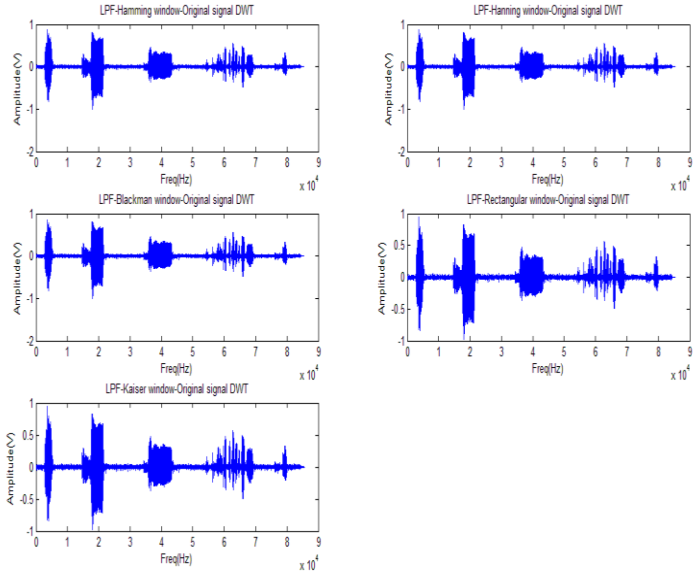

Fig.11. DWT applied Original signal filtered by FIR-LPF

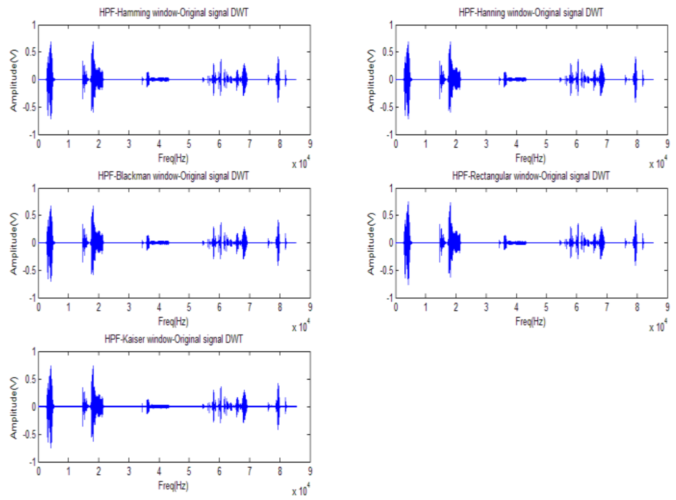

Fig.12. DWT applied Original signal filtered by FIR-HPF

The Fig.11 and Fig.12 show the responses of FIR LPF and HPF designed by different windowing such as Hamming, Hanning, Blackman, Rectangular and Kaiser Window techniques of DWHT applied original signal. From these windows Blackman HPF provides high SQNR than other windowing techniques.

4.5 IIR FILTERING

The Fig.13 shows the response of IIR Filter designed by Butterworth LPF, Butterworth HPF, Chebyshev LPF and Chebyshev HPF of FFT applied original signal. From these results Chebyshev HPF provides high SQNR than other techniques.

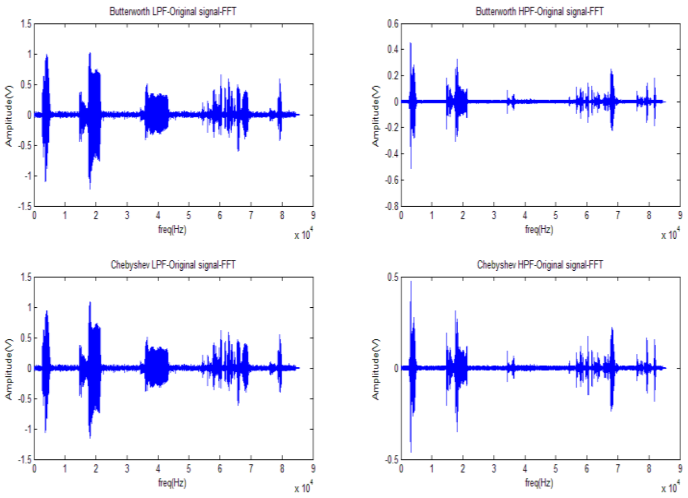

Fig.13. FFT applied Original signal filtered by IIR filter

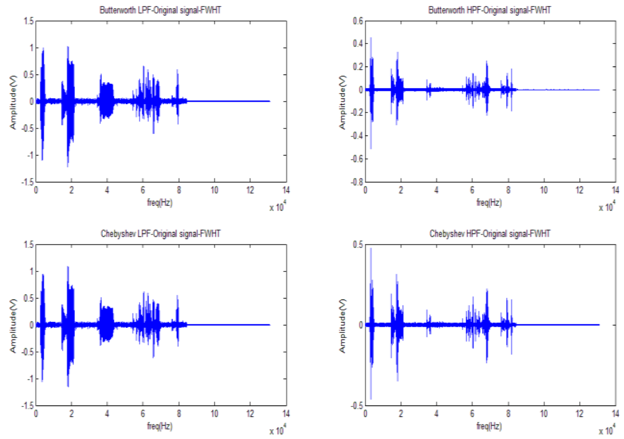

Fig.14. FWHT applied Original signal filtered by IIR filter

The Fig.14 shows the response of IIR Filter designed by Butterworth LPF, Butterworth HPF, Chebyshev LPF and Chebyshev HPF of FWHT applied original signal. From these results Chebyshev HPF provides high SQNR than other techniques.

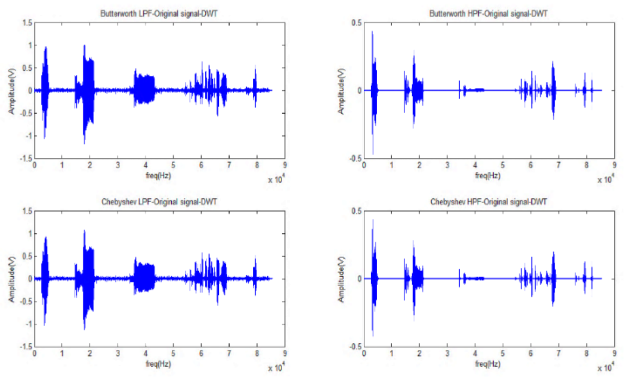

Fig.15. DWT applied Original signal filtered by IIR filter

The Fig.15 shows the response of IIR Filter designed by Butterworth LPF, Butterworth HPF, Chebyshev LPF and Chebyshev HPF of DWT applied original signal. From these results Chebyshev HPF provides high SQNR than other techniques.

5. PERFORMANCE ANALYSIS

Similarly, FIR and IIR filters are designed for each sub bands separately and their filter coefficients are quantized. After quantization the Signal to Quantization Noise Ratio for each filter coefficients of original signal and their sub bands are calculated and the comparison tables are made for the performance analysis. The signal to noise ratio value of signal after quantization is more compared to signal to noise ratio value of signal before quantization.

Table.1. Table of SQNR for FFT applied signals filtered by FIR (LPF and HPF) Windowing methods

Window methods	Filter	Signals				
		Original	Band1	Band2	Band3	Band4
Hamming	LPF	168.18	155.634	146.465	128.333	114.784
	HPF	170.435	152.882	136.871	123.589	108.253
Hanning	LPF	168.193	155.648	146.455	128.371	114.75
	HPF	170.478	152.888	136.865	123.591	108.314
Blackman	LPF	168.269	155.778	146.543	128.498	114.582
	HPF	170.644	153.002	136.878	123.673	108.55
Rectangular	LPF	168.027	155.453	145.933	127.908	114.542
	HPF	169.971	152.691	136.935	123.572	107.58
Kaiser	LPF	168.031	155.459	145.966	127.924	114.562
	HPF	169.986	152.706	136.929	123.567	107.603

From the Table.1 it inferred that, Blackman HPF window provides high SQNR (170.644) for original signal and Blackman LPF window provides high SQNR for all four band signals (155.778 for Band1, 146.543 for Band2, 128.498 for Band3, 114.582 for Band4 signals).

Table.2. Table of SQNR for FWHT applied signals filtered by FIR (LPF and HPF) Windowing methods

Window methods	Filter	Signals				
		Original	Band1	Band2	Band3	Band4
Hamming	LPF	168.18	155.635	146.468	128.333	114.785
	HPF	170.426	152.878	136.877	123.585	108.254
Hanning	LPF	168.194	155.649	146.458	128.37	114.751
	HPF	170.467	152.885	136.871	123.587	108.315
Blackman	LPF	168.27	155.779	146.546	128.498	114.584
	HPF	170.634	152.999	136.884	123.669	108.551
Rectangular	LPF	168.028	155.454	145.934	127.908	114.543
	HPF	169.961	152.6929	136.941	123.567	107.582
Kaiser	LPF	168.032	155.46	145.967	127.924	114.564
	HPF	169.977	152.7076	136.935	123.562	107.604

From the Table.2, it inferred that, Blackman HPF window provides high SQNR for original signal (170.634), Blackman LPF window provides high SQNR for Band1 signal (155.779), Kaiser LPF window provides high SQNR for Band2 signal (145.967), Blackman LPF window provides high SQNR for Band3 signal

(128.498) and Hamming LPF window provides high SQNR for Band4 signal (114.785).

Table.3. Table of SQNR for DWT applied signals filtered by FIR (LPF and HPF) Windowing methods

Window methods	Filter	Signals				
		Original	Band1	Band2	Band3	Band4
Hamming	LPF	168.279	155.862	146.548	128.549	114.981
	HPF	170.802	153.681	137.368	124.654	108.626
Hanning	LPF	168.293	155.88	146.556	128.587	114.939
	HPF	170.84	153.711	137.358	124.621	108.684
Blackman	LPF	168.369	156.017	146.617	128.724	114.814
	HPF	171.023	153.772	137.356	124.61	108.911
Rectangular	LPF	168.115	155.663	145.852	128.109	114.675
	HPF	170.277	161.246	137.481	124.79	107.996
Kaiser	LPF	168.119	155.669	145.885	128.126	114.696
	HPF	170.295	162.391	137.474	124.785	108.017

From the Table.3 it inferred that, Blackman HPF window provides high SQNR for original signal (171.023), Kaiser HPF window provides high SQNR for Band1 signal (162.391), Hanning LPF window provides high SQNR for Band2 signal (146.556), Blackman LPF window provides high SQNR for Band3 signal (128.724) and Hamming LPF window provides high SQNR for Band4 signal (114.981).

Table.4. Table of SQNR for different transforms applied signals filtered by IIR (LPF and HPF)

Window methods	Filter	Signals				
		Original	Band1	Band2	Band3	Band4
Hamming	LPF	166.795	152.43	138.431	124.217	109.092
	HPF	166.794	152.431	138.433	124.218	109.091
Hanning	LPF	166.994	152.97	139.061	124.958	109.928
	HPF	174.106	156.623	139.207	124.123	113.47
Blackman	LPF	174.104	156.62	139.208	124.119	113.472
	HPF	174.649	157.169	139.979	125.129	115.333
Rectangular	LPF	166.776	152.819	138.324	124.836	107.89
	HPF	166.778	152.8196	138.324	124.838	107.89
Kaiser	LPF	166.938	153.085	139.128	125.541	109.134
	HPF	174.353	156.484	138.76	124.073	114.038

From the Table.4, it inferred that Chebyshev HPF provides high SQNR for original signal and Band4 signal, Butterworth HPF provides high SQNR for Band1 and Band2 signals and Chebyshev LPF provide high SQNR for Band3 signal.

6. CONCLUSION

The proposed digital filter for speech signals using multi rate signal processing has been designed. After filtering, the quantization of the input signal and filter coefficients were analyzed. The performance of the proposed system was evaluated based on Signal to Quantization Noise Ratio. From the performance measures, it was observed that DWT applied signals filtered by Chebyshev HPF provides high SQNR in IIR Filtering and DWT applied signals filtered by Blackman Window provide high SQNR in FIR filtering. In future, the proposed digital filters will be further implement for various signals such as ECG signal, OFDM signal, etc.

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