Efficient Spectral Analysis Through Effective Reduction of Adjacent Channel Interference in Multirate Processing

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Abstract

Spectral analysis is considered to be important area of consideration in which the volume reconstruction and visualization takes place, besides amounting to increase in computational efficiency. The reception of selected frequency and amplitude level takes an important role for further processing at the succeeding stages of Digital Signal Processing. The number of developments are undertaken through research work for effective reception of signals. In this paper, the proposed method is based on Multirate technique, specified by authors [1][2], in Finite Impulse Response digital filter bank through **Modified Kaiser window**, subsequently followed by Fast Fourier transform.   A remarkable spectral output is achieved by way of increase in magnitude, linear phase response, constant width and sharp rise of response, less adjacent channel interference and better stopband attenuation as compared to authors of [3][4][5][6] besides significant improvement of specific parameters listed among the respective methods are elicited.   In addition, a better reduction in computational complexity is achieved. This method of spectral analysis is suitable in most of applications, especially in digital hearing aid applications as it is compatible with low frequency, low delay and low power requirements. The simulation results are added on account of satisfactory performance and comparison is drawn to enlighten the advantages in the proposed method.

Keywords: Mutirate Processing, Bandpass Filtering, Decimation, Modified Kaiser Window, Net Input Samples, Spectral output.

1. INTRODUCTION

Many applications need effective detection of signals mainly in terms of amplitude and frequency components, which become easier in processing of signals and implementation of hardware at the succeeding stages. In addition, the received signals become effective if they are received at maximum peak response, free from adjacent frequency interference and better stopband attenuation. More over, the reception becomes significant if the computational complexity is less. In order to meet the specified requirements, this paper proposes a method in Multirate Digital filter bank followed by Fast Fourier transform (FFT). Thus, this paper is emerged with a novel technique in spectral analysis to best suit among various applications. There is a possibility for the enhancement in selectivity and less adjacent channel interference as mentioned in the results, depicted by authors [3][4][7].   In addition to these characteristics, further there is a possibility of improvement in characteristics like peak response and stopband attenuation as per

the results shown in [5][6][8]. For these reasons, the proposed method of spectral analysis is suggested to overcome such disadvantages among these methods, depicted for various applications. The Finite Impulse Response (FIR) band pass filter is used as it has a great number of advantages including precise control over the phase response which greatly facilitates signal reconstruction. The filter is designed based on '**Modified Kaiser window**' approach which improves sharp rise and constant width response characteristic. Among others, particularly in digital hearing aid applications, the filter bank provides a natural decomposition of the input signal into frequency bands which may be processed independently to best compensate for the hearing loss and meet prescriptive targets, as hearing loss is a function of both frequency and input level. A reduction in computational complexity and hardware complexity is achieved through heterodyning operation and arrival of response using the techniques in Fast Fourier transform. The heterodyning operation shifts the original band of input frequencies to the lower frequencies starting from '0' Hz. In addition, the techniques in FFT algorithm presents further minimizing of computations rather than specified one in the reference [8]. These techniques in the proposed method have led to specific advantages in respect of linear phase response, constant width and sharp rise of response, minimum adjacent channel interference and better stopband attenuation as compared to the existing methods apart from great reduction in computational complexity. A novel technique is appended to the actual part of work resulting to further enhancement in stopband attenuation and peak output response.   This paper is a description of an extremely flexible frame work for separating the input into frequency bands as it forms the basis for a Multichannel compression in hearing aid development as demonstrated by authors [9][10], in addition to analysis of spectral components. This method is well described in this paper with the help of analytic equations. The simulation results and performance characteristics are drawn to verify with the facts. This method is particularly verified at lower frequencies because of quality assurance, implemented even at higher frequencies and so chosen with the input signal as '**Cosine function**' as this function represents the real time signals in most of applications.

2. METHODOLOGY AND RELATED EQUATIONS

2.1   Design of Bandpass Filter
This method of approach is arrived through partial modification of designated parameters in the existing Kaiser window method [8].
The frequency response of bandpass filter is given as

$$h(z) = \begin{cases} 1...............for \cdots f_{p1} \leq |f| \leq f_{p2} \\ 0.............for \cdots |f| \langle f_{p1} \cdots and \cdots |f| \rangle f_{p2} \end{cases} \quad \text{---------------------------(2.1.1)}$$

Where f_{p1} is lower pass band frequency and f_{p2} is upper passband frequency .
The desired impulse response is given by $h_d(n)$.

$$h_d(n) = \begin{cases} \dfrac{1}{n\pi}\left[\sin\left(\dfrac{2\pi n f_{c2}}{F} \right) - \sin\left(\dfrac{2\pi n f_{c1}}{F} \right) \right] \cdots\cdots for \cdots n \rangle 0 \\ \dfrac{2}{F}(f_{c2} - f_{c1}) \cdots\cdots for \cdots n = 0 \end{cases} \quad \text{--------------(2.1.2)}$$

Where $f_{c1} = f_{p1}-(df/2)$, $f_{c2} = f_{p2}+(df/2)$
$df_2 = f_{p1}-f_{s1}$, $df_3 = f_{s2}-f_{p2}$
where $df = \min\{df_2, df_3\}$,
F = Sampling frequency
f_{s1} = Lower stop band frequency
f_{s2} = Upper stop band frequency
n = integer
N' = Order of the filter
The frequency response of the $h_d(n)$ is given by h(z).

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$$h(z) = z^{-\frac{(N'-1)}{2}} \left[h_d(0) + \sum_{n=1}^{(N'-1)/2} 2h_d(n)\cos(2\pi f n T) \right] \text{------------------------(2.1.3)}$$

The equation, $h_d(0) + \sum_{n=1}^{(N'-1)/2} 2h_d(n)\cos(2\pi f n T)$ represents the magnitude response equation,

where T = 1/F.

2.2   Design of Window

The basic Kaiser Window function, $a_k(n) = \begin{cases} \dfrac{I_o(B)}{I_o(A)} ----- & for \cdots n \le \dfrac{(N'-1)}{2} \\ 0 -------otherwise \end{cases}$ -------- (2.2.1)

Where 'A' is an independent variable empirically determined by Kaiser

$$B = A\left(1 - \left(\frac{2n}{(N'-1)}\right)^2\right)^{0.5}$$

$$I_o(x) = 1 + \frac{(0.5x^2)}{(1!)^2} + \frac{(0.5x^2)^2}{(2!)^2} + \ldots\ldots\ldots$$

This series converges rapidly and can be computed up to 25 terms to attain the desired accuracy. Considering the bandpass specifications such as the passband ripple (Ap) and minimum stopband attenuation (As) in decibels are given by

$$Ap = 20\log_{10}\frac{(1+dp)}{(1-dp)}$$

$$As = -20\log_{10}(ds)$$

The 'ds' and 'dp' can be determined as

$$ds = 10^{(-0.05As)}$$

$$dp = \frac{(10^{Ap} - 1)}{(10^{0.05Ap} + 1)}$$

$$da = \min(dp, ds)$$

The actual stopband attenuation is arrived to be $As = -20\log_{10}(da)$

The value of 'A' is determined from the empirical design equation

$$A = \begin{cases} 0 ----- for \cdots\cdots As \le 21 \\ 0.5942(As - 21)0.4 + 0.7986(As - 20) ---- for \cdots 21 \le As \le 50 \\ 1102(As - 8.7) ------ for \cdots\cdots As\rangle 50 \end{cases}$$

A parameter 'D' is determined from the empirical design equation,

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$$D = \begin{cases} 0.9222 ----- for \cdots\cdots As \le 21 \\ \dfrac{(As - 7.95)}{14.36} --- for \cdots\cdots As \rangle 21 \end{cases}$$

The filter order for the lowest odd value of 'N' ', calculated to be

$$N' \ge \frac{(F)(D)}{df} + 1$$

The order of the filter increases with increase in the sampling frequency or decrease in the transition width. Finally the impulse response of the filter is determined as $h(n) = a_k(n) \, h_d(n)$. Where $a_k(n)$ and $h_d(n)$ are window coefficients and desired impulse response respectively. The frequency response and the output response, for an input signal band of frequencies from 480 Hz to 530 Hz, of the filter bank are plotted according to Modified Kaiser window approach, as shown in Figure-1 and Figure-2 respectively. It is observed that a better spectral response with improvement in passband ripple and quality of output are attained as compared to the methods of reference [5][6][8]. Accordingly, this novel approach of filter bank design can bring attractive advantages in the proposed method of filter bank design, which follows.
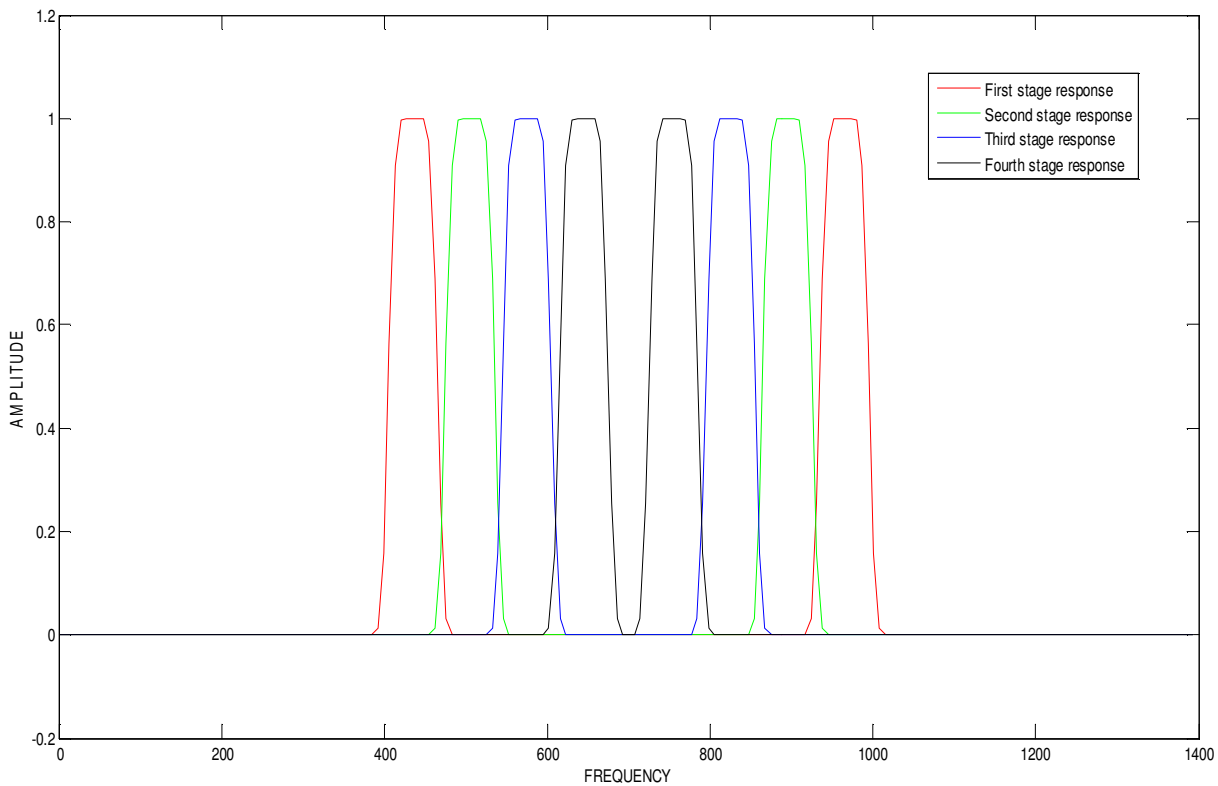
FIGURE 1: Modified Kaiser Window Based Frequency Response of Bandpass Filter Bank.
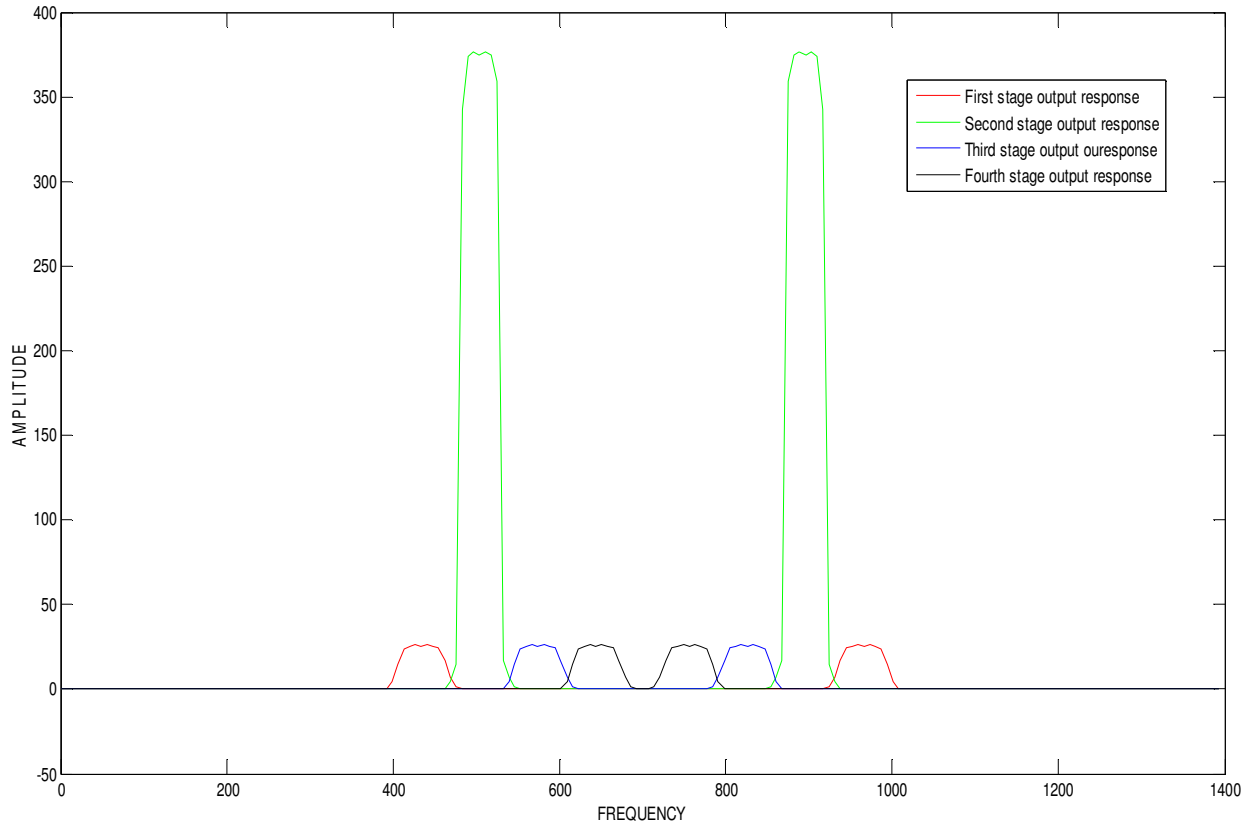
FIGURE 2: Output Response of Modified Bandpass Filter Bank For The Input Signal Band of Frequencies From 480 Hz to 530 Hz (With Existing Method).

2.3 PROPOSED METHOD WITH NEW TECHNIQUE IN MULTIRATE PROCESSING

The proposed method is a new technique in Multirate processing that is appended to the filter bank which causes an improvement in selectivity, constant width and sharp rise of response causing minimum adjacent channel interference as compared to methods [3][4][7]and additional improvement in characteristics like peak response and stopband attenuation as compared to methods [5][6][8]. More over, the specified advantages in the proposed method are undertaken at minimum number of computations that takes optimal time in processing.

In the proposed method, each sample of input is multiplied by a constant, derived by cosine function which is related to starting frequency of original input signal band and starting frequency of the shifted input signal band as shown in Figure-3. The novel method is introduced to further modify these values in such a way that the shifted input sample values are added with sample values of modified input function at the respective intervals of time so as to derive the **Net input samples (x5(n))** at reduced sampling frequency. The samples of modified input function are arrived at same intervals with negative sample values and having the magnitude $(1/14)^{th}$ of original input, taken at stopband frequencies. The Net input samples are significantly processed, so that the specified advantages in the passband frequencies are met in the proposed method. These Net input samples are taken as inputs to the individual stage of bandpass filter bank and each bandpass filter is designed to meet the desired specifications of passband and stopband frequencies. The response to the individual stage bandpass filter is arrived by performing convolution between the input and impulse response of bandpass filter as shown in Figure-3.

The number of output samples of individual bandpass filter is rounded to 'N' samples by placing zeroes after relative magnitude values at successive intervals of time. The value 'N' is equal to a number, 2^{sts}, where 'sts' is an integer and also represents number of stages of Fast Fourier transform such that, the value 'N' is next immediate greater value to the number of output samples of bandpass filter. Finally, the frequency response of individual stage of bandpass filter is arrived by performing radix-2 Fast Fourier transform on 'N' samples of the filter response as shown in Figure-3. The related procedure of Fast Fourier transform in connection with Figure-3 is shown in Figure-4. From Figure-3, the signal conversions at various points upon input signal can be considered in the form as mentioned below.

x1(n) = x(n) (cos (nw$_p$'T))
x2(n) = x(n) (cos (nw$_p$'T))

where 'w$_p$' is the starting frequency of input signal band in rad/sec. and x(n) is the input signal. The input signal, taken at passband frequencies and stopband frequencies respectively as x1(n) and x2(n), obtained at n=0,1,2,-------(M-1), Where 'M' is the number of samples of decimated input signal as corresponding to sampling frequency 'F' '. The samples of x1(n) and x2(n) are taken at the decimated sampling frequency such that the decimation factor, D = F/F', where 'F' ' is the decimated sampling frequency.
Let x(nT) be the input signal with number of input samples be 'N1'. The decimated output of original input signal is taken by using the Interpolation formula as given below.

$$x_m(mT_y) = \sum_{n=0}^{(N1-1)} x(nT)g(mT_y - nT) \quad \text{------------------------------------- (2.3.1)}$$

Where ' T ' is the original sampling period and ' T$_y$ ' is the sampling period of decimated output. 'n' and 'm' are the sample numbers of original and decimated input signals respectively such that n=0,1,2,-------(N1-1) and m=0,1,2,3--------(M-1), where 'N1' and 'M' are the numbers of samples of original and decimated input signals respectively.
Where $g(t)$ is given as

$$g(t) = .\frac{\sin(\pi t/T)}{\pi t/T} \leftrightarrow G(F1) = \begin{cases} T, \cdots\cdots |F1| \leq \dfrac{F}{2} \\ 0 \cdots\cdots otherwise \end{cases}$$

x3(n) = x1(n)+x1(n)
x4(n) = (-x2(n))(1/14)
x5(n) = x3(n)+x4(n)

The x5(n) is the finally modified input signal and for the above equations the value 'n' is taken at 0,1,2,3,-----------(M-1).
The impulse responses of each bandpass filter is taken as h$_1$(n),h$_2$(n),----h$_i$(n) and output response of each bandpass filter is arrived by convolution, such that
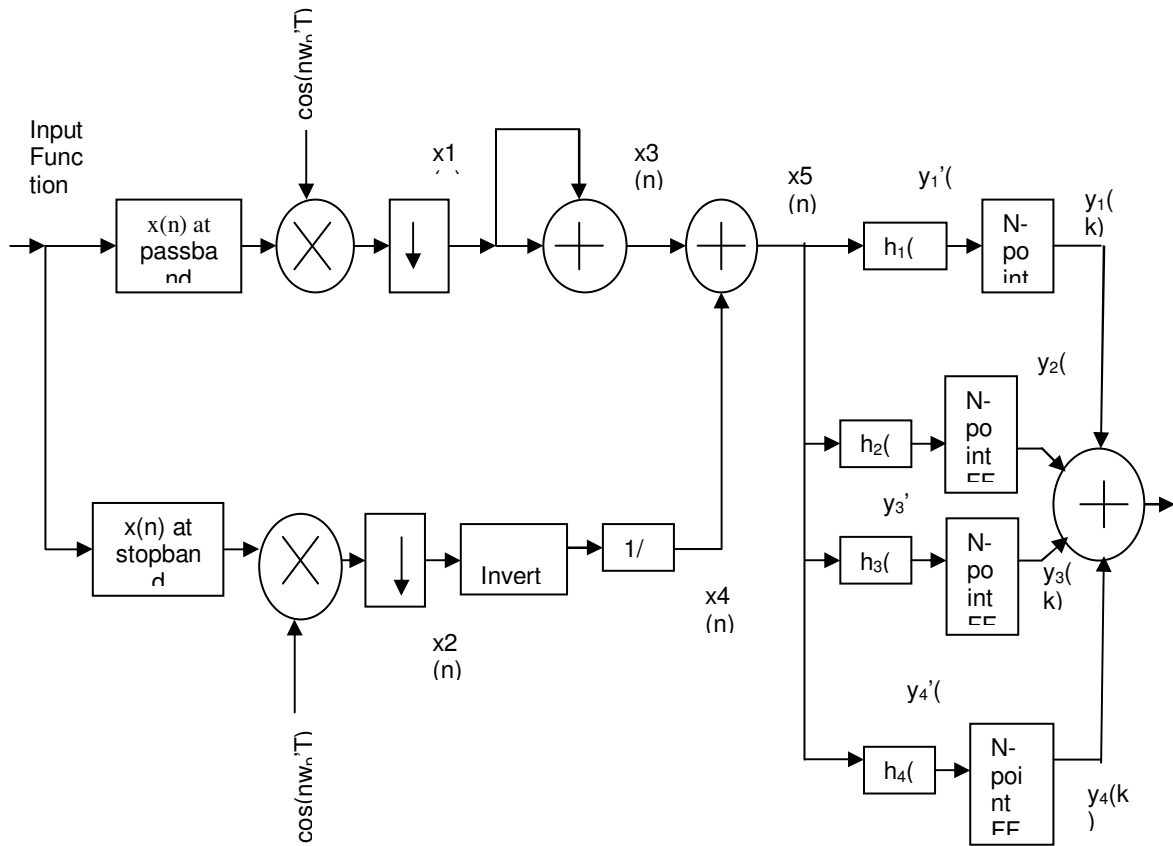
y$_1$'(n) = x5(n)*h$_1$(n)
y$_2$'(n) = x5(n)*h$_2$(n)
…………………
…………………
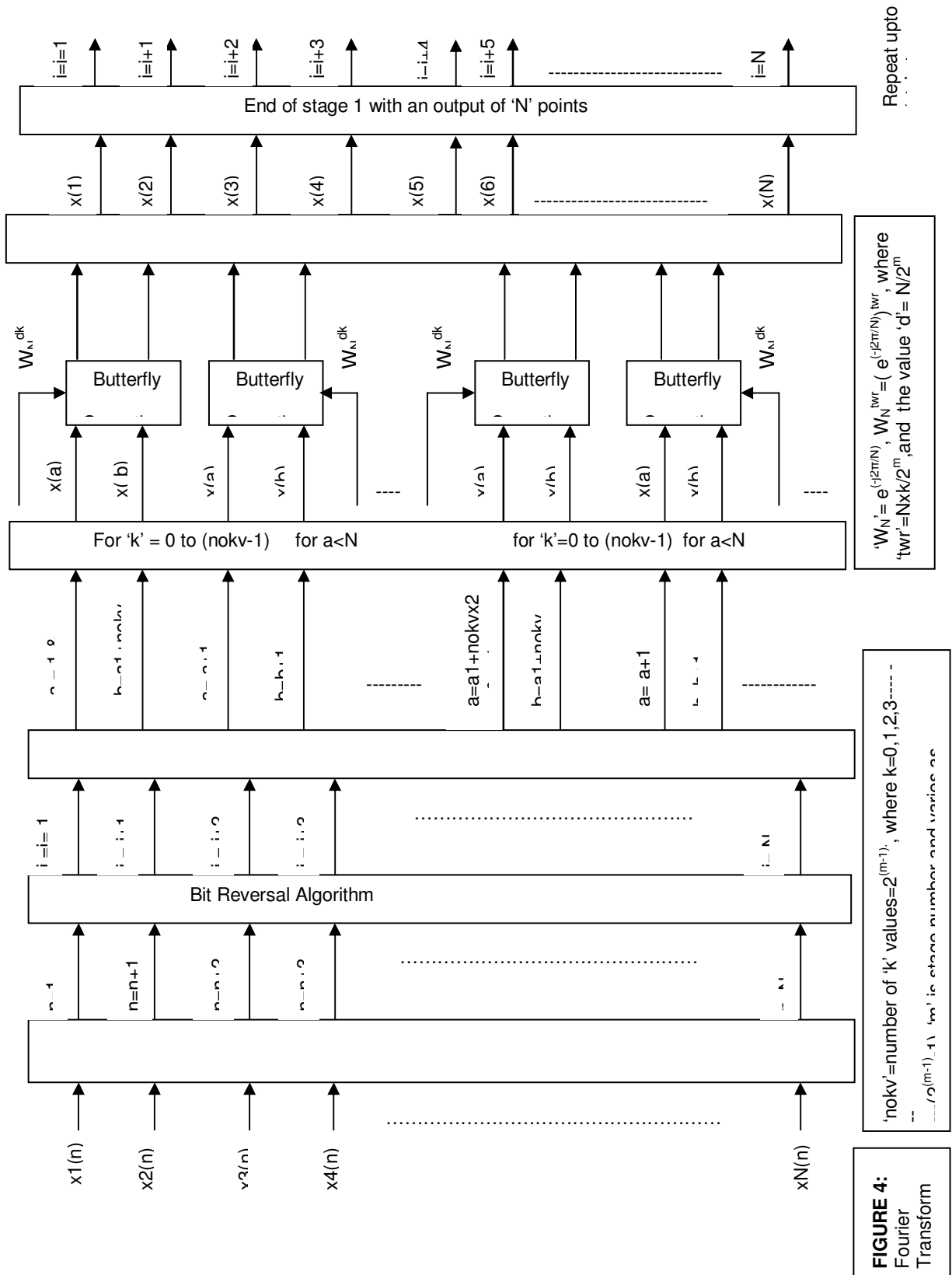…………………
y$_i$'(n) = x5(n)*h$_i$(n)

The h$_1$(n),h$_2$(n),h$_3$(n)----h$_i$(n) are designed at shifted frequencies of input signal and with respect to decimated sampling frequency. The 'i' refers to stage number and 'n' refers to a number of sampling interval at which impulse response is taken as it varies from n = 0 to (N'-1).

FIGURE 3: Block Diagram of The Proposed Method.

FIGURE 4: Fourier Transform

The succeeding intervals of output samples are zero padded so that the total samples 'N' represents the following equation.

$y_i(n) = y_i'(n)\big|$ for n=0,1,2---------(N'+M-2) $+y_i'(n)=0\big|$ for n=(N'+M-1),(N'+M)-------(N-1)

Finally the frequency response of individual stage is carried out by performing FFT on $y_1(n),y_2(n),\ldots\ldots,y_i(n)$, whose output is given in the form as given by the relation.

$$y_1(k) = \sum_{n=0}^{(N-1)} y_1(n)(e^{-j2\pi f T_y})^n$$

$$\ldots\ldots\ldots\ldots\ldots\ldots$$
$$\ldots\ldots\ldots\ldots\ldots\ldots$$

$$y_i(k) = \sum_{n=0}^{(N-1)} y_i(n)(e^{-j2\pi f T_y})^n$$

2.4 Fast Fourier Transform Method
The Fast Fourier transform method is adopted to obtain the frequency response of the Digital filter bank. The number of points upon which the FFT is to be performed is dependent on the number of output points of each bandpass filter. The type of FFT which is adopted is In-place radix-2 decimation in time algorithm. The techniques incorporated in FFT algorithm have been arrived with minimum number of computations as well as reduced hardware complexity. The number of points in FFT are 'N', such that it is equal to 2^{sts}, where 'sts' is an integer. The methodology used in the FFT algorithm is shown in Figure-4.

3. COMPUTATIONAL COMPLEXITY
The computational complexity is also utmost form of concern for designing a particular method of spectral analysis because less computational complexity and techniques involved in the algorithm greatly minimizes the hardware complexity besides increasing the system accuracy. Calculation of computational efficiency is carried out by performing number of computations required. The number of multiplications and additions required for individual stage and subsequent computations for number of stages are calculated.

Assuming the number of original input samples, decimated input samples and the filter order to be ' N1', ' M ' and ' N' ', the number of computations required for the proposed method are given by equations as follows.
The number of multiplications required for Heterodyning operation = N1+N1.
The number of additions required for arriving Net input samples = M+M.
The multiplications required for arriving Net input samples = M+M.
The number of multiplications required for time domain output response of single bandpass filter stage = (M)(N').
The number of multiplications required for time domain output response of 'i' bandpass filter stages = (M)(N')(i).
The number of additions required for time domain output response of single bandpass filter stage = (M-1)(N'-1).
The number of additions required for time domain output response of 'i' bandpass filter stages = (M-1)(N'-1)(i).
The number of multiplications required to perform FFT on time domain response of single bandpass filter stage = $\dfrac{(M+N'-1)}{2}\log_2(\dfrac{(M+N'-1)}{2})$

The number multiplications required to perform FFT on time domain response of 'i' bandpass filter

$$\text{stages} = (i)\frac{(M + N'-1)}{2}\log_2(\frac{(M + N'-1)}{2})$$

The number of additions required to perform FFT on time domain response of single bandpass filter stage $= (M + N'-1)\log_2(M + N'-1)$

The number of additions required to perform FFT on time domain response of 'i' bandpass filter stages $= (i)(M + N'-1)\log_2(M + N'-1)$

As per above equations, the computations required are calculated for the proposed method and comparison is drawn with existing methods. The number of computations required is dependent on number of input samples, filter order and number of output frequency points (' f_n '). In addition, the quality of response is dependent upon the selection of number of input samples. For this reason, the number of input samples is taken at half of the sampling frequency so that the maximum quality in the response is achieved. More over, the number of computations required is reduced in obtaining the response at reduced sampling frequency. Thus, the number of input samples are arrived as N1 = 700, M = 400 for existing and proposed methods respectively. Considering ' f_n ' = 200 for existing method of 'Filter bank', the number of multiplications required for single stage of existing methods of 'Filter bank' and 'Fast Fourier transform' are 394500 and 2958, subsequently the additions required are 392536 and 6616. But, for the proposed method, the number of multiplications and additions for single stage are arrived to be 69686 and 70572 only. In this example, for four stages, the number of multiplications required for existing methods is 1578000 and 11832; subsequently the additions required are 1570144 and 26464. But for the proposed method the number of multiplications and additions are arrived to be 272144 and 279888 only.

4. RESULTS AND DISCUSSION
The methodology and advantages have been discussed by the following example. Assuming the input signal to be in the frequency range between 410 Hz and 670 Hz, the sampling frequency becomes (2)(700) =1400 Hz. The sampling frequency is reduced to 800 Hz by a decimation factor 'D' such that the value 'D' can be arrived as (1400/800). The decimated sampling frequency is so selected that the frequency modified input signal result in without aliasing. The filter bank is designed between 0 Hz and 280 Hz, corresponding to the input signal band between 410 Hz to 670 Hz as the original frequency band is shifted correspondingly between 10 Hz and 270 Hz. The filter bank consists of 4 stages and each stage has got a bandwidth of 50 Hz. The shifted passband and stopband frequencies of these stages are as shown in Table-1.

Stage	Lower stopband frequency (f_{s1}) (Hz)	Lower passband frequency (f_{p1}) (Hz)	Higher passband frequency (f_{p2}) (Hz)	Higher stopband frequency (f_{s2}) (Hz)
Stage 1	0	10	60	70
Stage 2	70	80	130	140
Stage 3	140	150	200	210
Stage 4	210	220	270	280

**TABLE 1:** Passband and Stopband Frequencies of Proposed Method.

Initially the specifications of passband ripple and stopband attenuation are assumed as 0.5 and 35dB respectively. Using these specifications, the frequency response and the output response

of existing method of filter bank approach with Modified Kaiser window are simulated as shown in Figure-1 and Figure-2, as part of comparative advantages to be met in the proposed method. Further, using these specifications, the frequency response of the proposed method of filter bank is simulated as shown in Figure-5, to insight the advantages.
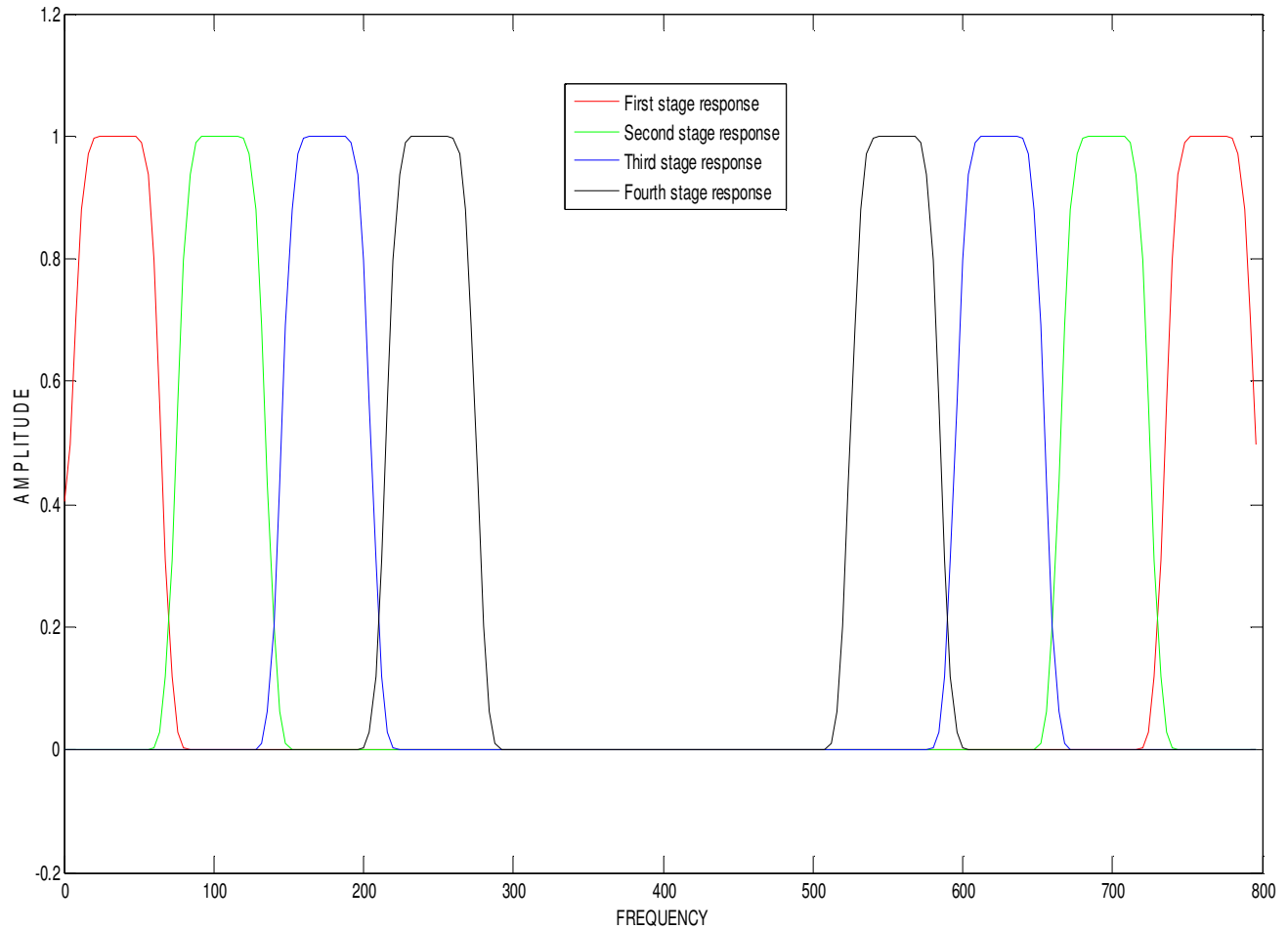
FIGURE 5: Frequency Response of Bandpass Filter Bank of Proposed Method.

Unlike the method of Figure-2, the other existing method [8] is simulated by taking the FFT output for the same input signal, as shown in Figure-6. In addition, the variation of design parameters of 1st and 2nd existing methods is shown from Table-2 to Table-5. Even in both the responses of Figure-2 and Figure-6, it is observed that the output response need to be enhanced in respect of constant width cum sharp rise of response and stopband attenuation as it amounts to minimum interference to adjacent frequencies. In this connection the proposed method is drawn with improved characteristics in the response. As per design constraints in the proposed method, the specifications are arrived with better values in respect of peak response and stopband attenuation as compared to the existing methods and rather designed with Modified Kaiser window as shown in Table-6 and Table-7. The time domain output and its frequency response of the proposed method are shown in Figure-7 and Figure-8 respectively, when the input signal band of frequencies taken from 410 Hz to 460 Hz.
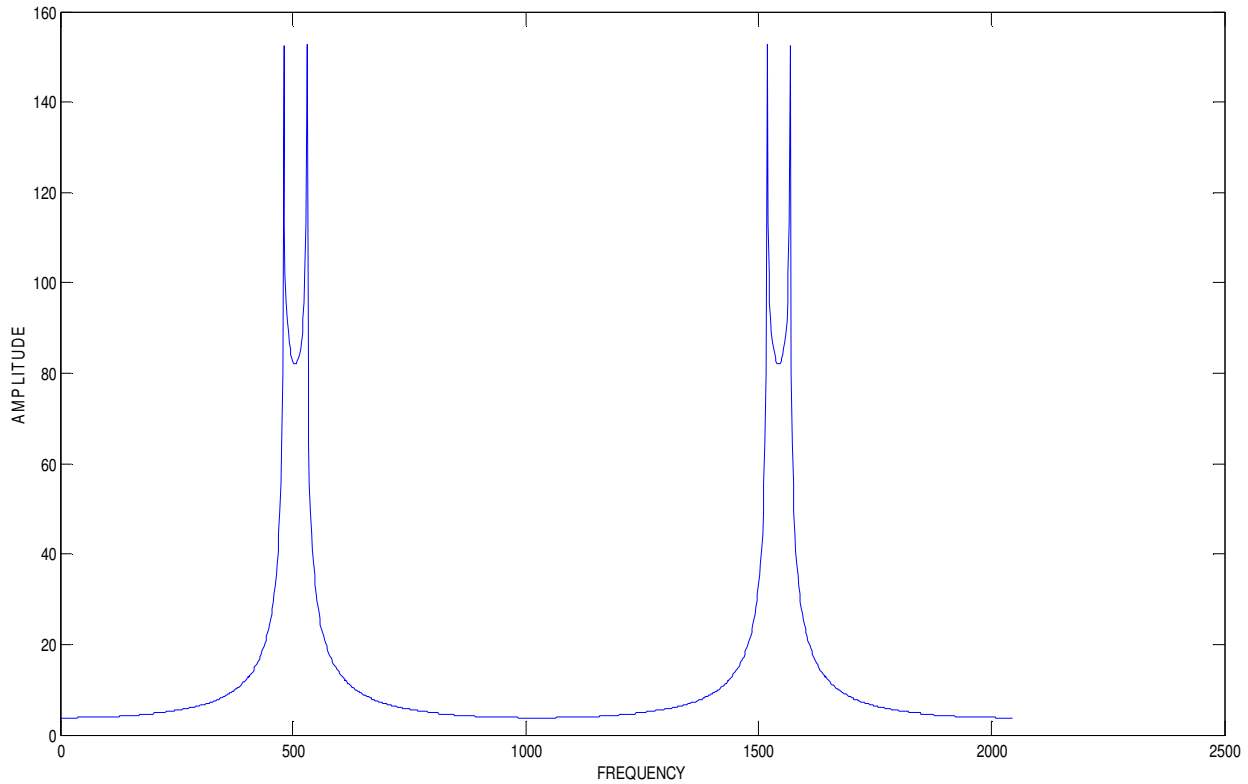
FIGURE 6: Fast Fourier Transform Output For The Input Signal Band of Frequencies From 480 Hz to 530 Hz (Existing Method-2).

As a result, the first stage of filter bank only exhibited response with an improvement in specifications like better selectivity, minimum adjacent channel interference as compared to methods [3][4][7] and more peak response, better stopband attenuation as compared to methods [5][6][8] respectively. An additional distinguished advantage to be mentioned hereby that a characteristic of linear amplitude variation in stopband is arrived, because of which the spectral output became more selective due to result of effective extraction of selected passband low frequency components, attached to adjacent sidebands of assumed passband. Thus, the combined effect of this special feature other than enhanced characteristics of spectral parameters, the proposed method could able to levy the efficiency of spectral output.

S.No.	Input frequency band (Hz)	Normalized amplitude
1	410 – 460	0.104
2	480 – 530	1
3	550 - 600	0.104
4	620 -670	0.104

TABLE 2: Amplitude Response of Existing Method of Filter Bank For The Input Signal Band of Frequencies From 480 Hz to 530 Hz (Using Modified Kaiser Window)(As Per Figure 2).

S.No.	Input frequency band (Hz)	Normalized amplitude
1	410 – 460	0.193
2	480 – 530	1
3	550 - 600	0.2
4	620 -670	0.129

TABLE 3: Amplitude Response of Existing Method of Fast Fourier Transform For The Input Signal Band of Frequencies From 480 Hz to 530 Hz (As Per Figure 6).

S.No.	Input frequency band (Hz)	Assumed passband and stopband frequencies (Hz)	Simulated results of passband and stopband frequencies (Hz)	Filter order	Passband and stopband attenuation of output response (dB) (As per Fig.2)
1	410 -460	$f_{s1} = 400$ $f_{p1} = 410$ $f_{p2} = 460$ $f_{s2} = 470$	$f_{s1} = 400$ $f_{p1} = 410$ $f_{p2} = 460$ $f_{s2} = 470$	283	19.66
2	480 – 530	$f_{s1} = 470$ $f_{p1} = 480$ $f_{p2} = 530$ $f_{s2} = 540$	$f_{s1} = 470$ $f_{p1} = 480$ $f_{p2} = 530$ $f_{s2} = 540$	283	0
3	550 – 600	$f_{s1} = 540$ $f_{p1} = 550$ $f_{p2} = 600$ $f_{s2} = 610$	$f_{s1} = 540$ $f_{p1} = 550$ $f_{p2} = 600$ $f_{s2} = 610$	283	19.66
4	620 - 670	$f_{s1} = 610$ $f_{p1} = 620$ $f_{p2} = 670$ $f_{s2} = 680$	$f_{s1} = 610$ $f_{p1} = 620$ $f_{p2} = 670$ $f_{s2} = 680$	283	19.66

TABLE 4: Values of Filter Parameters As Per Simulation Results of Existing Method of Filter Bank (Using Modified Kaiser Window).

| S.No.	Input frequency band (Hz)	Assumed passband and stopband frequencies (Hz)	Simulated results of passband and stopband frequencies (Hz)	Passband and stopband attenuation of output response (dB) (As per Fig.6)
1	410 -460	$f_{s1} = 400$ $f_{p1} = 410$ $f_{p2} = 460$ $f_{s2} = 470$	$f_{s1} = 400$ $f_{p1} = 410$ $f_{p2} = 460$ $f_{s2} = 470$	14.33
2	480 – 530	$f_{s1} = 470$ $f_{p1} = 480$ $f_{p2} = 530$ $f_{s2} = 540$	$f_{s1} = 470$ $f_{p1} = 480$ $f_{p2} = 530$ $f_{s2} = 540$	0
3	550 – 600	$f_{s1} = 540$ $f_{p1} = 550$ $f_{p2} = 600$ $f_{s2} = 610$	$f_{s1} = 540$ $f_{p1} = 550$ $f_{p2} = 600$ $f_{s2} = 610$	13.98
4	620 - 670	$f_{s1} = 610$ $f_{p1} = 620$ $f_{p2} = 670$ $f_{s2} = 680$	$f_{s1} = 610$ $f_{p1} = 620$ $f_{p2} = 670$ $f_{s2} = 680$	17.79

TABLE 5: Values of Spectral Parameters As Per Simulation Results of Existing Method of Fast Fourier Transform.

S.No.	Input frequency band (Hz)	Translated band of input frequencies (Hz)	Normalized amplitude at passband and stopband frequencies
1	410 - 460	10 - 60	1
2	480 - 530	80 - 130	0.273
3	550 - 600	150 - 200	0.103
4	620 - 670	220 - 270	0.043

TABLE 6: Amplitude Response of The Proposed Method For The Input Signal Band of Frequencies From 410 Hz to 460 Hz (As Per Figure 8).

S.No.	Input frequency band (Hz)	Assumed passband & stopband frequencies (Hz)	Translated band of input frequencies (Hz)	Simulated results of passband and stopband frequencies (Hz)	Filter order & passband ripple (dB)	Passband and stopband attenuation of output response (dB) (As per Fig.8)
1	410 – 460	$f_{s1} = 400$ $f_{p1} = 410$ $f_{p2} = 460$ $f_{s2} = 470$	$f_{s1} = 0$ $f_{p1} = 10$ $f_{p2} = 60$ $f_{s2} = 70$	$f_{s1} = 0$ $f_{p1} = 10$ $f_{p2} = 60$ $f_{s2} = 70$	163 & 0	0
2	480 -530	$f_{s1} = 470$ $f_{p1} = 480$ $f_{p2} = 530$ $f_{s2} = 540$	$f_{s1} = 70$ $f_{p1} = 80$ $f_{p2} = 130$ $f_{s2} = 140$	$f_{s1} = 70$ $f_{p1} = 80$ $f_{p2} = 130$ $f_{s2} = 140$	163 & 0	11.28
3	550 - 600	$f_{s1} = 540$ $f_{p1} = 550$ $f_{p2} = 600$ $f_{s2} = 610$	$f_{s1} = 140$ $f_{p1} = 150$ $f_{p2} = 200$ $f_{s2} = 210$	$f_{s1} = 140$ $f_{p1} = 150$ $f_{p2} = 200$ $f_{s2} = 210$	163 & 0	19.71
4	620 - 670	$f_{s1} = 610$ $f_{p1} = 620$ $f_{p2} = 670$ $f_{s2} = 680$	$f_{s1} = 210$ $f_{p1} = 220$ $f_{p2} = 270$ $f_{s2} = 280$	$f_{s1} = 210$ $f_{p1} = 220$ $f_{p2} = 270$ $f_{s2} = 280$	163 & 0	27.31

TABLE 7: Values of Filter Parameters As Per Simulation Results of Proposed Method.
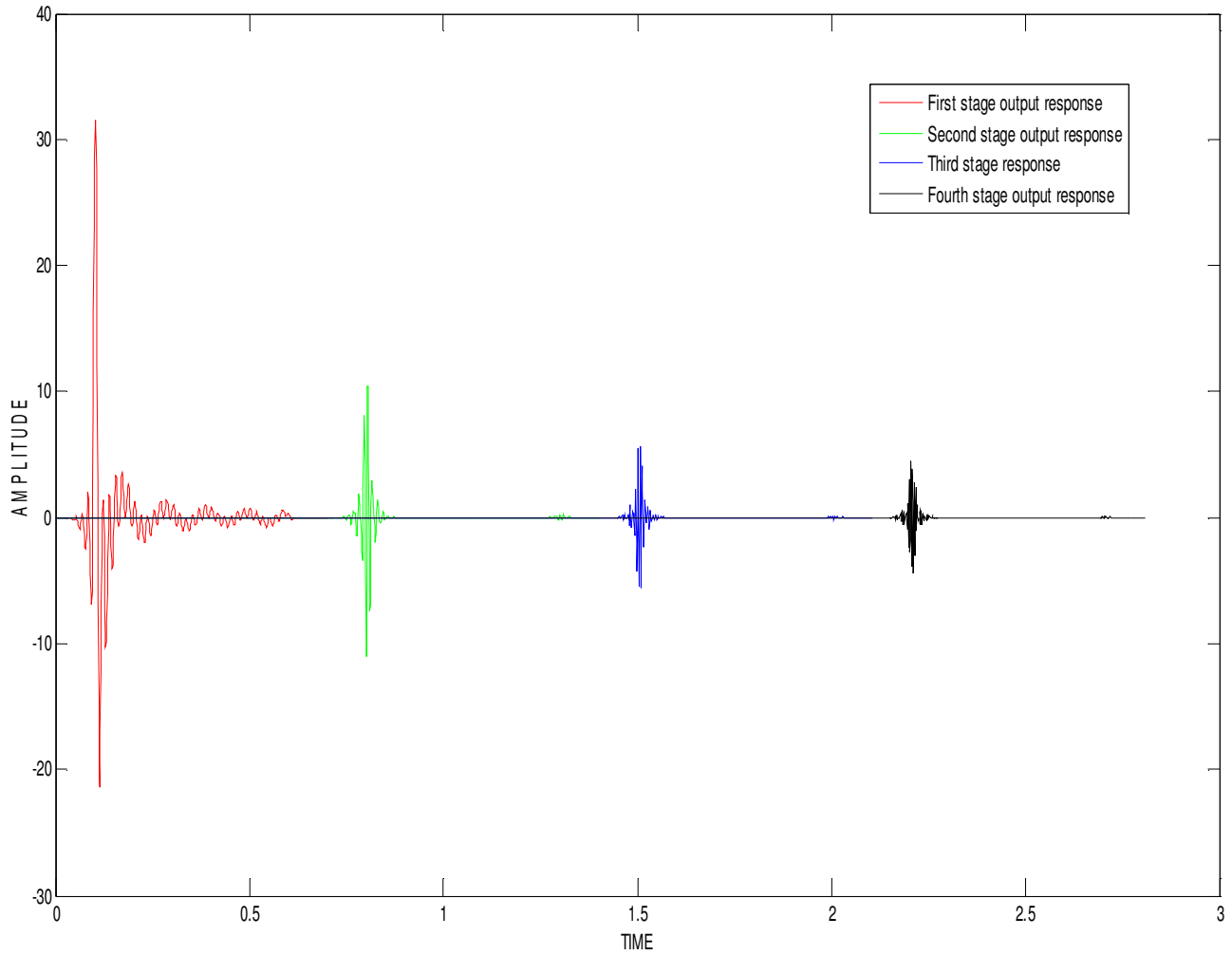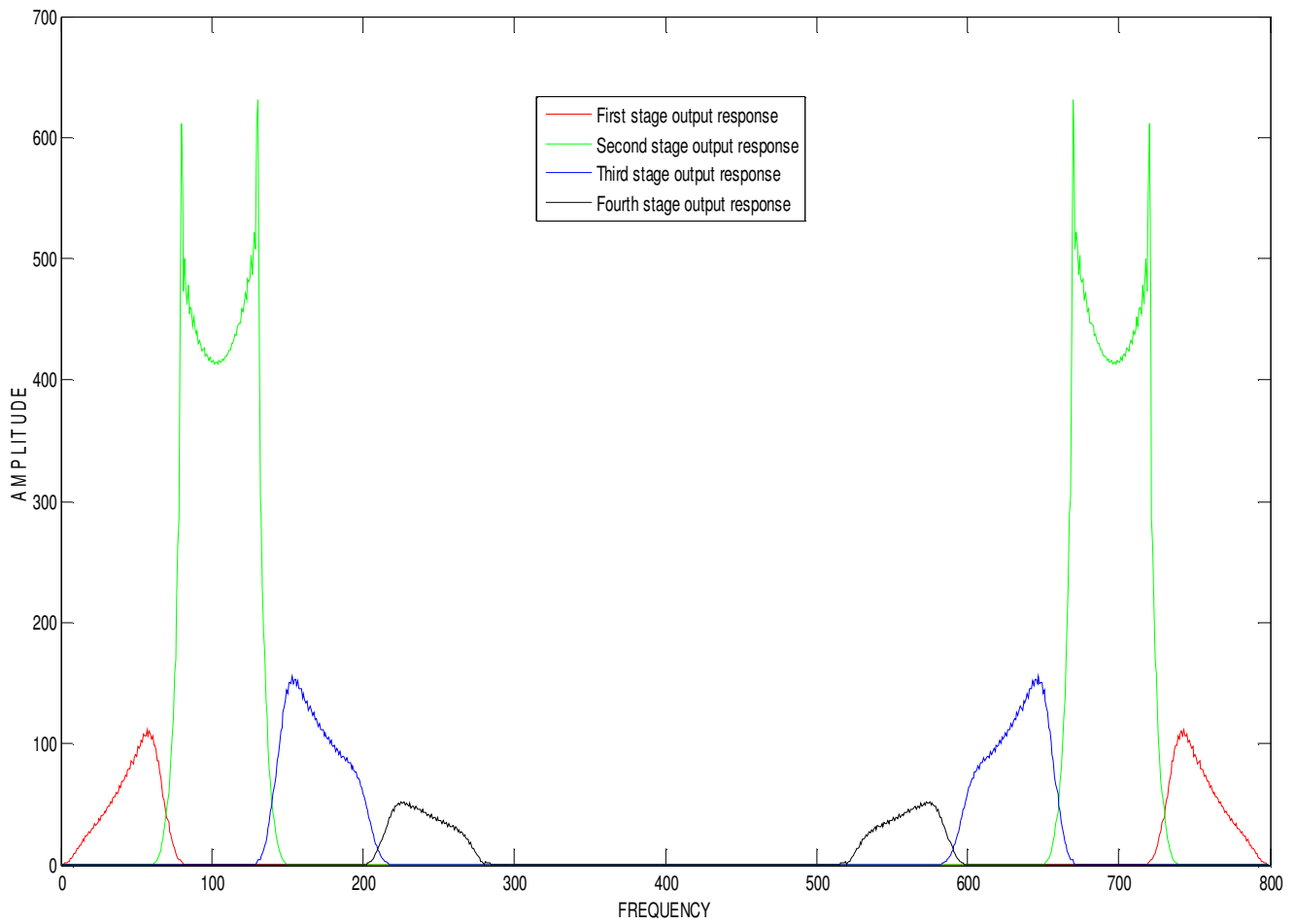
FIGURE 7: Time Domain Response of Frequency Shifted Filter Bank For The Input Band of Frequencies From 410 Hz to 460 Hz of Proposed Method.
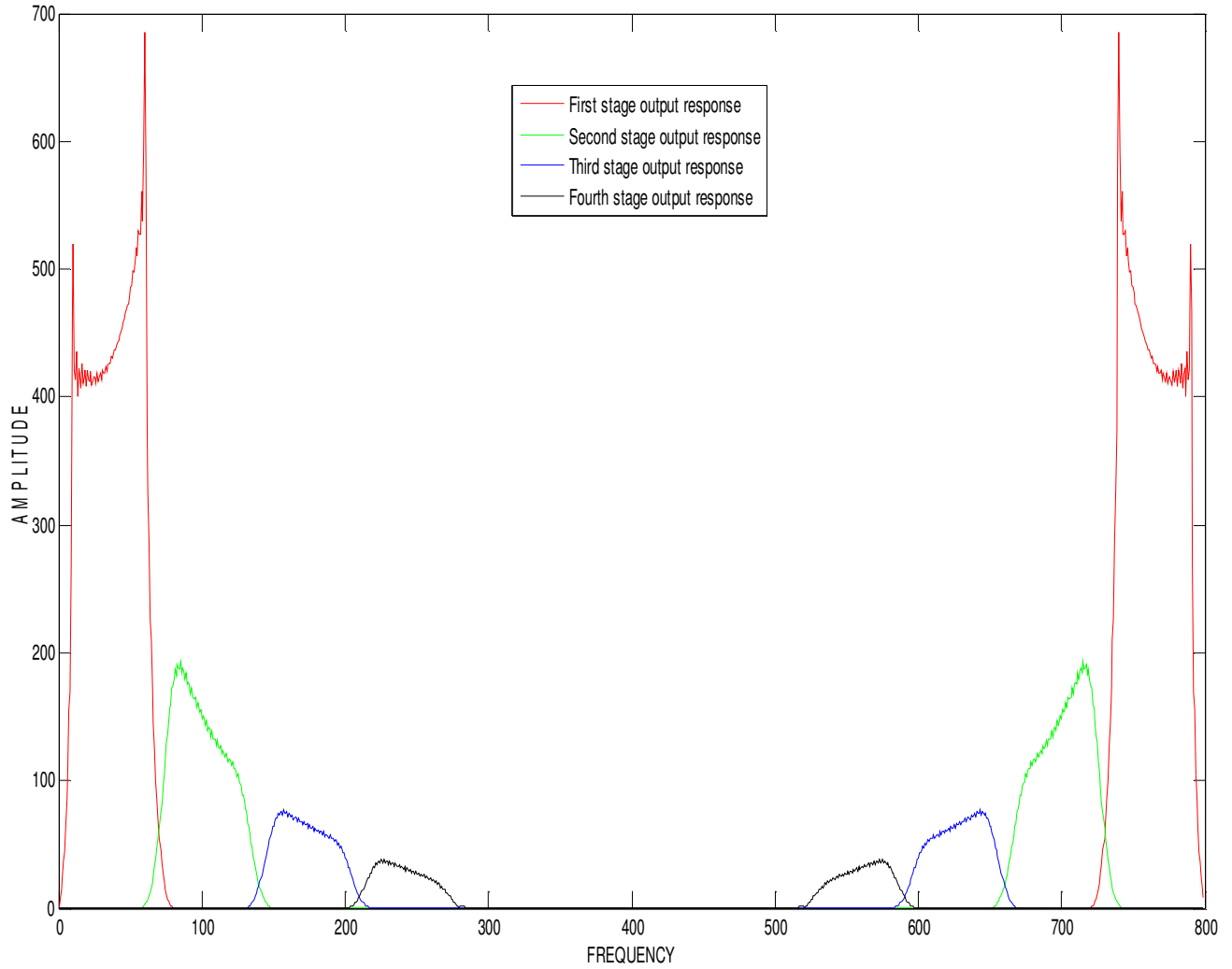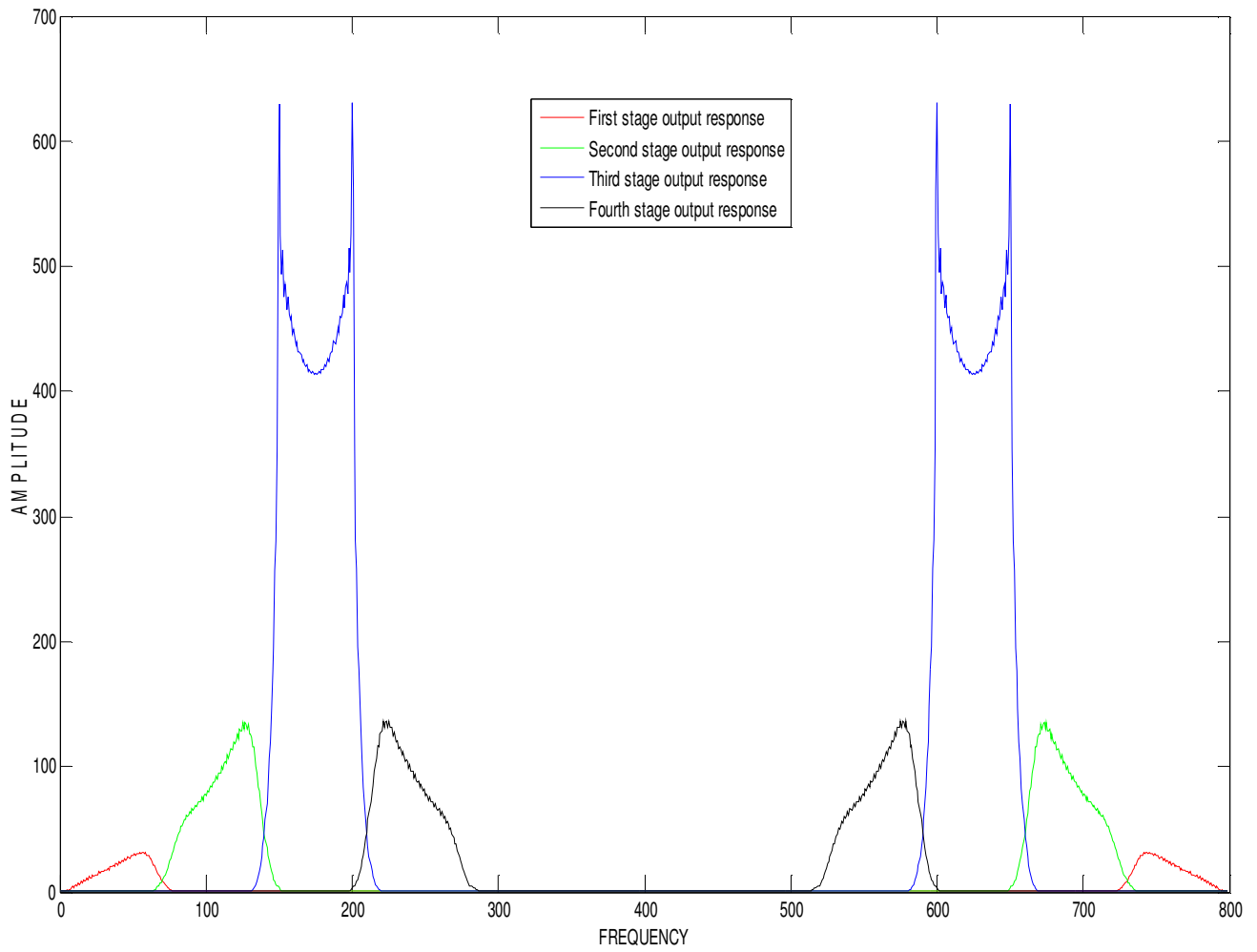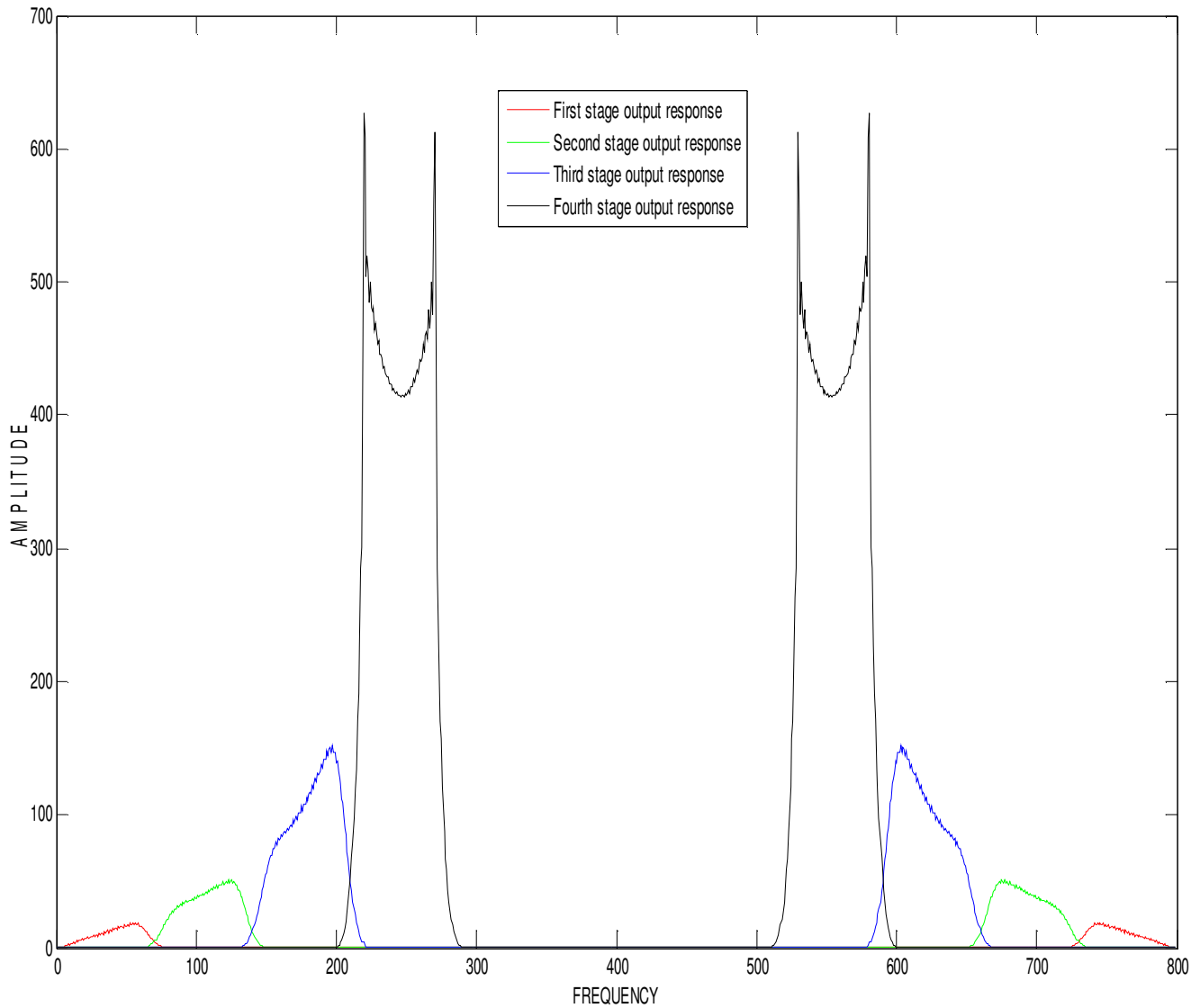
FIGURE 8: Output Response of Proposed Method For The Input Band of Frequencies From 410 Hz to 460 Hz.

FIGURE 9: Output Response of Proposed Method For The Input Band of Frequencies From 480 Hz to 530 Hz.

FIGURE 10: Output Response of Proposed Method For The Input Band of Frequencies From 550 Hz to 600 Hz.

FIGURE 11: Output Response of Proposed Method For The Input Band of Frequencies From
620 Hz to 670 Hz.

5. CONCLUSIONS

The simulated results of proposed method are well furnished and relative advantages are tabulated with respect to some important parameters of spectral analysis.   These results conclude that there is a large achievement with respect to parameters like constant width and sharp rise of response, peak output, minimal interference with adjacent channel frequencies and stopband attenuation as compared to existing methods, discussed earlier.   Further, it is noticed that the computational complexity is arrived to be optimally minimum in addition to these advantages.   Thus, the proposed method of spectral analysis is improved by virtue of techniques presented in this paper and these techniques are implemented in MATLAB-7.01 and results are simulated.   Therefore, this method can particularly draw numerous advantages especially in digital hearing aid applications, because the merits in this method are able to bring attractive advantages with regard to compression techniques described in [9][10] and hearing loss compensation. In addition, the feature of linear amplitude response in stopband of proposed

method is well supportive for arriving vestigial sideband modulated sequences so that the proposed method of spectral analysis is more suitable in multicarrier applications [11]. More over, this method is well suited to extract variety of advantages in various real time applications. The over all performance of the proposed method of spectral analysis is improved   in respect of specifications, Spectral response and Computational complexity as per the results shown in figures and tables.

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