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ORIGINAL PAPER

Design and synthesis of filter bank structures based on low order constrained least square and minimum phase methods for audiogram matching in digital hearing aids

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Abstract

Frequency decomposer or filter bank plays acritical role in the design of digital hearing aid systems.This paper proposes filter bank structures suitable for the design of optimal hearing aid systems.The performance indices such as complexity, power dissipation, area utilization, delay, matching error of the filter banks are very competitive when compared to the designs in literature.Constrained least square finite impulse response filter with smooth transition band characteristics is used in the first method.The second method is based on Minimum phase filters with reduced number of coefficients which exhibit the smallest passband group delay in the family of filters with the same magnitude response characteristics.Hardware complexity is reduced very much since there is significant reduction in multiplications per sample in both the methods.Full cycle responses of individual filters are considered for calculation of overall response of the filter bank.Coefficient decimation method is used in the reconfigurable structure to enable the filter banks to adapt variation in audiogram pattern during the course of time.Synthesis results by Cadence Encounter(R.) R.T.L. Compiler software also justify the theoretical claims

Keywords Constrained least square filters · Minimum phase filters · Finite impulse response filter bank · Coefficient decimation · Audiogram

1 Introduction

Hearing aids improve the speech intelligibility by rectifying hearing loss with auditory compensation methods [1]. Low cost hearing aids are still need of the day as millions of hearing loss cases are reported from the under developed countries and World Health Organization (WHO) studies reveals that, around 10% of the affected are children [2]. Hardware complexity of hearing aid device should be reduced to manufacture the item in a large volume at a lower cost, affordable by the poor. In digital hearing aid design, filter banks play a critical role. A filter bank decomposes the input signal in to frequency bands which may be processed independently to best compensate for the hearing loss. A person with impaired hearing tends to have a low sensitivity towards certain frequencies and the filter bank should be able to adjust sound levels at arbitrary frequencies within a given spectrum. This is achieved by passing input signal from microphone through a set of filters that divides audio signals in to different frequency bands. The gains of each sub band filters are adjustable to match the audiogram of the hearing impaired person. The design considerations of a compact hearing aid involve minimum hardware, low power consumption, low delay and higher flexibility. Long delays more than 10 ms may cause annoying effect in hearing aid users. So a filter bank structure with low delay and reduced hardware complexity are the requirements for the design of a compact hearing aid with longer battery life.

The filter bank design for digital hearing aids in literature can be classified as uniform and non uniform filter banks [3]-[6]. An 8 band filter bank with equal bandwidth finite impulse response (FIR) filter is implemented in [3]. The frequency bands are equally divided in uniform filter banks without any consideration to the non uniform scaling of human auditory system. So the computational complexity of uniform filter bank is high to meet the nonuniform frequency

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resolution characteristics of human ears. Consequently nonuniform filter banks are more suitable for digital hearing aids. The nonuniform filter bank can be classified into octave band, critical band and 1/3 octave band filter banks. The 8 band octave filter bank with frequency response masking technique (FRM) reported in [2] lowers the hardware complexity at the cost of delay. The critical bands used in the method reported in [3] match the human hearing perception, but increases the implementation difficulty due to irregularity of the sub bands. The IIR structure fails to provide linear phase characteristics in 1/3 octave filter bank realized in [3]. In [4], a three channel variable filter bank was proposed where the three digital filters were obtained from a normalized analog prototype Chebyshev type I low pass filter using analog frequency transformations along with a modified bilinear transformation. This type of filters is not always stable and could not provide linear phase property.

Computation complexity is low in filter bank design with IIR structure; however FIR filters are preferred not only for their linear phase but also for the stability and regular structure. A low-power design and implementation of ANSI S1.11 filter bank for digital hearing aids is presented in [6]. But, the drawback of the realization method is the long group delay, which is due to the multirate structure and the stringent filter specification. Studies have shown that delays more than 10 ms may cause disturbing perception for hearing aid users. The computational complexity is reduced in the FIR reconfigurable filter bank reported in [7] by using interpolation, decimation and frequency response masking methods. But the price paid is longer delay, greater than 20 ms, which is not suitable for a real time hearing aid. The long delays will cause various problems such as the disparity between aid generated and air- borne vented sound, etc.A recursive low pass filter and discrete cosine modulation method are used to generate a nine band uniform filter bank in [8]. The derived 1/3 octave filter bank had low group delay and low complexity but the recursive nature of the prototype filter introduces problems with stability and linear phase characteristics of the system. Hardware complexity is high in the design reported in [9] in which sample rate conversion technique based 16 band uniform and nonuniform filter bank is proposed. Farrow subfilter based variable bandwidth filter is presented in [10], but the fixed point implementation become difficult as there is exponential increase in coefficient values of the subfilters with their order. Canonic signed digit representation of filter coefficients is proposed in [11], in which multipliers are reduced to adders and shifters. In this system, Farrow based tunable filter characteristics are improved by hybrid evolutionary algorithms. A multirate filter bank structure with relaxed ANSI S1.11 specification is reported in [12], in which delay oriented cases, complexity oriented cases and patient oriented cases are presented. Huang et al. [13] proposed a reconfigurable filter bank based on nonlinear transformation. A reconfigurable filter bank based on fractional interpolation and symmetry property of linear phase filter is proposed in [14]. A nonuniform Modified Discrete Fourier Transform (MDFT) filter bank is proposed in [15] in which order of the filter is varied for different bands. Farrow structure based variable bandwidth digital filter for hearing aid application is proposed in [16] in which the hardware complexity is reduced by compromising the precision of Canonic Signed Digit (CSD) arithmetic. Additional circuitry involving accumulator is required to apply tunable factor value to the system which will increase the overall system complexity. A constrained least square (CLS) FIR filter bank based structure is reported in [17], in which the multipliers required is large as the filter order is high.

The review of the reported methods shows that, it is very difficult to trade off between hardware complexity and the effect of delay of filter banks for hearing aid applications. To this end, two methods are proposed in this paper. In the first method, a low order CLS FIR filter bank is proposed for audiogram matching. In the second method, a minimum phase filter bank is proposed. A reconfigurable CLS FIR Filter bank structure is also presented. CLS FIR Filter minimizes the square error over the entire frequency range from 0 to π subjected to the peak constraints. The minimum phase filter is derived from a prototype filter and the number of coefficients in a minimum phase filter is nearly half of that of the prototype filter. So hardware complexity of the filter is reduced without any effect on stability. The group delay of the minimum phase filter is very small in the pass band compared to that of linear phase FIR filter. This in turn preserves the envelope fidelity of the speech signal. Envelope fidelity is a primary factor in determining the intelligibility and quality of speech in a noisy environment [18]. The proposed systems thus have a low hardware complexity and delay which are the required characteristics of a good hearing aid. Cost of the hearing aid can be reduced as the area and power requirement are minimized in the proposed systems. The minimum phase filter shows tolerable variations in phase in the pass band. Indeed researches in psychoacoustics had shown that the human ear is not sensitive to such small phase distortions. The experiments conducted by Lipshitz et al. [19] so far lead to the conclusion that on normal music or speech signals phase distortion appears not to be generally audible. In the reported methods in literature, only the overlapping areas of individual filters are considered for calculating the overall response of the filter bank. The complete cycle of individual filter responses are considered in the proposed systems. Otherwise the audiogram matching become imperfect as the ripples in the stop band will also be boosted while the individual filter gains are selectively amplified to minimize the matching error. Most of the comparisons of such systems in literature are in terms of either number of multiplications or theoretically estimated gate count values only. In this work, synthesis result of filter bank structures for hearing aids are also presented.

The paper is organized as follows. An overview of constrained least square FIR filter is given in Section II. Minimum phase FIR filter is discussed in Section III. In Section IV, hearing loss and audiogram are presented. Filter bank implementation is given in Section V. Audiogram matching is given in Section VI. Performance evaluation is discussed in Section VII. Finally, Section VIII concludes the paper.

2 Constrained least square filters

The method enables to define upper and lower thresholds that include the maximum allowable ripple in the magnitude response of a filter. With this constraint, the least square error minimization technique is applied over the whole frequency range of the filter's response, instead of over specific bands [20–22]

$$\mathrm{H}(\omega) = \sum_{\mathrm{n}=0}^{\mathrm{N}-1} \mathrm{h}(\mathrm{n})\mathrm{e}^{-\mathrm{j}\omega \mathrm{n}} \tag{1}$$

If $h(n) = h(N - 1 - n)$, then $\mathrm{H}(\omega)$ has linear phase and can be written as.

$$\mathrm{H}(\omega) = \mathrm{C}(\omega)\mathrm{e}^{-\mathrm{jM}\omega} \tag{2}$$

where $\mathrm{C}(\omega)$ is the real valued amplitude, and $\mathrm{M} = \frac{(\mathrm{N}-1)}{2}$ for filter of length N [23]. For symmetric odd length filters $\mathrm{C}(\omega)$ can be represented as.

$$\mathrm{C}(\omega) = \frac{1}{\sqrt{2}}\mathrm{c}(0) + \sum_{\mathrm{n}=1}^{\mathrm{M}} \mathrm{c}(\mathrm{n}) \cos n\omega \tag{3}$$

where the impulse response coefficients h(n), in terms of the cosine coefficients c(n) are.

$$h(n) = \begin{cases} \frac{1}{2}c(M-n) & for \quad 0 \le n \le M-1 \\ \frac{1}{\sqrt{2}}c(0) & for\ n = M \\ \frac{1}{2}c(n-M) & for\ M+1 \le n \le N-1 \\ 0 & otherwise \end{cases} \tag{4}$$

Let $\mathrm{D}(\omega)$ be the desired amplitude of an ideal lowpass filter. Approximating this discontinuous function by the cosine polynomial $\mathrm{C}(\omega)$ is the fundamental filter design problem. The approximation error can be written as.

$$\mathrm{E}(\omega) = \mathrm{D}(\omega) - \mathrm{C}(\omega) \tag{5}$$

The primary measures of approximation used in filter design are.

The weighted integral square error is given by

$$\|E(\omega)\|_2 = \left(\frac{1}{\pi}\int_0^{\pi} W(\omega)(\mathrm{C}(\omega) - \mathrm{D}(\omega))^2 \mathrm{d}\,\omega\right)^{\frac{1}{2}} \tag{6}$$

The weighted Chebyshev error is given by.

$$\|E(\omega)\|_\infty = \max_{\omega\epsilon|0,\pi|} |W(\omega)(\mathrm{C}(\omega) - \mathrm{D}(\omega))| \tag{7}$$

where $\mathrm{W}(\omega)$ is a nonnegative error weighting function. The simplest method to design optimal FIR filters is to minimize $\|\mathrm{C}(\omega) - \mathrm{D}(\omega)\|_2$.

The CLS FIR filter design minimizes the unweighted integral square error with the constraint that the local minima and maxima of $\mathrm{C}(\omega)$ lie within the specified lower and upper bound functions $\mathrm{BL}(\omega)$ and $\mathrm{BU}(\omega)$. The lower and upper bound functions are given by.

$$\mathrm{B_L}(\omega) = \begin{cases} c1 - \delta_p & for\ all\ \omega\ \epsilon\left[0, \omega_p\right] \\ -\delta_s & for\ all\ \omega\ \epsilon\left[\omega_p, \pi\right] \end{cases} \tag{8}$$

and by

$$\mathrm{B_U}(\omega) = \begin{cases} c1 + \delta_p & \text{for all } \omega\ \epsilon\left[0, \omega_p\right] \\ \delta_s & \text{for all } \omega\ \epsilon\left[\omega_p, \pi\right] \end{cases} \tag{9}$$

where δp and δs are the maximum allowed deviations in the passband and stopband. This method of filter design is more suitable for situation where the maximum peak error size is to be controlled and there is no reason to assume that the input signals to be filtered have no energy in the transition band.

3 Minimum phase filters

Linear Phase FIR filters with narrow transition bands are of very high order and have a very long group delay that is half the filter order [24]. In hearing aid applications such a long delay is not tolerable. By relaxing the linear phase characteristics, it is possible to design an FIR filter of lower order. This reduces the overall group delay and the computational cost by saving the multipliers required for the design. Group delay of the minimum phase filter has the smallest value among the family of filters with the same magnitude response characteristics. A causal stable filter can be considered as minimum phase if all of its zeros are inside the unit circle.

Consider an arbitrary FIR transfer function of degree M.

$$H(z) = \sum_{n=0}^{M} h(n)z^{-n} \tag{10}$$

The mirror image polynomial to H(z) is the transfer function given by.

$$\widehat{H}(z) = z^{-M}H(z^{-1}) = \sum_{n=0}^{M} h(M-n)z^{-n} \tag{11}$$

The zeros of the transfer function (11) are reciprocal to those of H(z). As a result.

$$K(z) = H(z)\widehat{H}(z) = z^{-M}H(z)H(z^{-1}) \tag{12}$$

has zeros with mirror image symmetry in the z plane and is thus a Type I linear phase transfer function of order 2 M. Double zeros are present on the unit circle for K(z) as it is factorizable in to H(z) and $\widehat{H}(z)$. From the Eq. (12), it follows that the magnitude squared function can be expressed as.

$$\left|H(e^{j\omega})^2\right| = K(\omega) \tag{13}$$

The amplitude response of K(z) is thus non negative. As K(z) has double zeros on the unit circle, K(ω) has double zeros on [0,π]. Based on this idea a minimum phase filter can be derived by the following procedure [14].

Step1: Design a Type I linear phase FIR transfer function E(z) of degree 2 M, satisfying the amplitude response $\widehat{E}(\omega)$ specifications.

$$1 - \delta_p(E) \le \widehat{E}(\omega) \le 1 + \delta_p(E) \qquad \text{for } \omega\ \varepsilon\ [0,\ \omega p]$$

$$-\delta_s(E) \le \widehat{E}(\omega) \le \delta_s(E) \qquad \text{for } \omega\ \varepsilon\ [\ \omega s,\ \pi]$$

This E(z) has single unit circle zeros.

Step2: Determine the linear phase transfer function.

$$K(z) = \delta_s(E)z^{-M} + E(z) \tag{14}$$

Its amplitude response $\widehat{K}(\omega)$ satisfies.

$1 + \delta_s(E) - \delta_p(E) \le \widehat{K}(\omega) \le 1 + \delta_s(E) + \delta_p(E)$ for ω ϵ [0, ωp].

$0 \le \widehat{K}(\omega) \le 2\delta_s(E)$ for ω ϵ [ωs, π].

This K(z) has double zeros in the stop band with all other zeros situated with a mirror image symmetry. Thus it can be expressed in the form.

$$K(z) = z^{-M}H(z)H(z^{-1}) \tag{15}$$

Such that H(z) is a minimum phase FIR transfer function containing the zeros of K(z) that are inside the unit circle and one each of the double zeros of K(z) on the unit circle.

Step3: Determine H(z) from K(z) by spectral factorization.

The direct approach is to simply pick up the zeros of K(z) and the filter obtained by selecting the zeros that lie inside the unit circle is called a minimum phase filter.

The pass band ripple δp (E) and the stop band ripple δs(E) of E(z) should ensure that the specified pass band ripple δp and stop band ripple δs of H(z) are satisfied. It can be shown that

$$\delta_p(E) = \sqrt{1 + \frac{\delta_p}{1+\delta_s}} - 1 \tag{16}$$

$$\delta_s(E) = \sqrt{\frac{2\delta_s}{1+\delta_s}} \tag{17}$$

Non linear phase IIR filters with smaller group delay can be easily implemented, but the issue is that, they are not stable. At the same time smaller group delay can be achieved in the passband region by using stable minimum phase FIR filters.

4 Hearing loss and audiogam

Audiogram is produced by audiometric testing and it charts hearing ability, more specifically, the softest sounds that can be heard at various low and high frequencies. Hearing threshold is defined as the softest sounds a person hears at each frequency. Fig. 1 represents the audiogram of a normal hearing person. The response of the ear measured in decibels (dB) is plotted in the Y axis and the corresponding frequency in Hertz (Hz) is represented in the X axis. Audiogram curves generally describe the individual

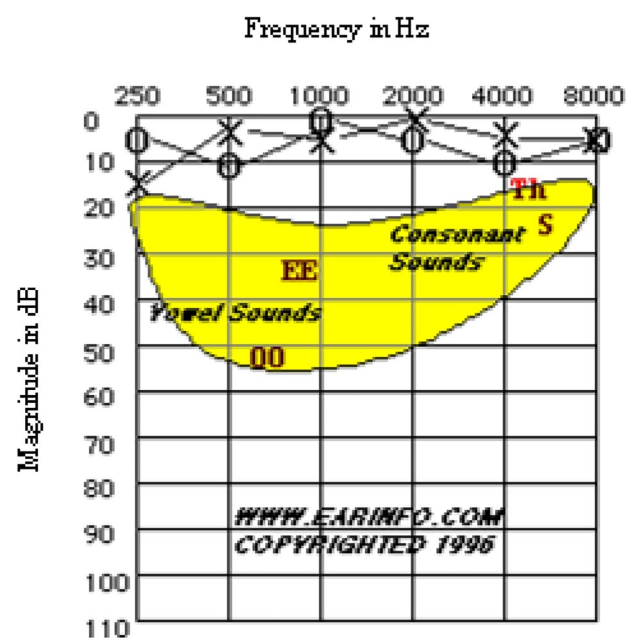


Fig. 1 Audiogram for normal hearing (www.earinfo.com)

hearing threshold compared to the normal hearing average.All thresholds upto 20 dB are treated as normal. An ‘O’ is used to represent the response of the right ear and an ‘X’ is used to represent the response of the left ear. A specific type of hearing loss is rectified by selectively amplifying sounds signals at different band of frequencies. The reference audiograms used in this paper are from the independent hearing aid information source https://www.earinfo.com. Hearing level measurements are done at each octave 250/500/1 k/2 k/4 k/8 k frequencies in a standard audiogram. Hence uniform filter bank may face difficulties to match the audiogram at low frequencies, unless a large number of bands are used. Thus nonuniform spaced digital FIR filter bank becomes very attractive for hearing aid applications. Nonuniform filter bank structures are used in the proposed methods. The response of the filter bank should match the audiogram of the patient so that the audio signals can be selectively amplified. In the audiogram where abrupt changes in slope occur, matching can be obtained by using large number of sub bands with narrow bandwidths. Three different audiograms of hearing loss cases are used in this paper to assess the performance parameters such as hardware complexity, delay and matching error etc. Matching error is the overall error between filter output and audiogram. Psycho acoustical studies accepted 3 dB as the tolerable limit [10], of matching error since most people are not sensitive to errors less than 3 dB.

5 Filter bank implementation

The main drawback of the reported FIR filter methods is the high computational cost due to the involvement of large number of multipliers. Group delay of the filter is another issue, as it is directly related to the order of the filter. This section proposes a filter bank design for hearing aid application with simple structure and low computational complexity. The midpoint value of a specific band of the audiogram of a hearing loss case is taken as the initial value of the sub band gain of the filter bank. Then it is varied to minimize the overall matching error. Data broadcast structure is used to realize the filter bank structure, so that the critical path can be reduced without using pipelining latches. The minimum time required to process a new sample is called as critical path. Reversing the direction of edges of a signal flow graph and interchanging the input and output ports keeps the functionality of the system. This transposition theorem leads to a structure called the data broadcast structure in which data are transmitted to all the multipliers at the same time. Thus the transformed form leads to a reduction in the critical path delay which will improve the clock speed or sample speed [25].

5.1 CLS FIR filter bank

In the first method, CLS FIR filter of order n = 32,$\delta_P = 10^{-5}$,$\delta_s = 10^{-5}$ is used as the prototype filter for the filter bank structure as shown in the Fig. 2. In which h represent filter coefficient and D represent D flipflop register. The smooth transition band characteristic of CLS FIR filter enables to reduce the order of the filter and number of bands required for the audio spectrum. The width of the individual bands used for the filter bank design for three different given in Table 1. Frequency response of low pass CLS FIR filter with cut off frequency 4 kHz for different values of n is plotted in Fig. 3. The sampling frequency used for the design is 16 kHz and filter delay is obtained by taking the product of half of the filter order (n) and sampling time. Audiogram matching for the filter order n = 32, n = 30 and n = 28 are carried out and the least matching error is obtained for the case with filter order n = 32 as shown in Table 2. So CLS FIR filter with order n = 32 is selected for the proposed filter bank design.

5.2 Minimum phase CLS FIR filter bank

In the second method CLS minimum phase FIR filter of order n = 32,$\delta_P = 10^{-5}$,$\delta_s = 10^{-5}$ is used as the prototype filter for the filter bank design. The number of coefficients required to meet the magnitude response characteristics in minimum phase FIR filter is only half of that of the FIR counterpart. Thus the multipliers required for filter implementation is reduced very much which will in turn reduce the hardware complexity. The bandwidth of individual filters used for the filter bank design for three different hearing loss cases is given in Table 3. Audiogram matching are done for filter order n = 32, n = 30 and n = 28 and the least matching error is obtained for the filter with order n = 32 as shown in Table 4. So Minimum phase CLS FIR filter with n = 32 is selected for the proposed filter bank design. Group delay characteristics of minimum phase low pass FIR filter derived from CLS FIR Filter of order 32 with $\mathbf{f_C} = 2\mathbf{kHz}$, and $\boldsymbol{\delta_P} = 10^{-5}$ and $\boldsymbol{\delta_s} = 10^{-5}$ is plotted in the Fig. 4. Frequency response of low pass minimum phase CLS FIR filter with cut off frequency 4 kHz for different values of n is plotted in Fig. 5. Group delay of the CLS minimum phase FIR filter varies from 3 to 6 samples (187.5 μS to 375 μS) in the pass band. This very small variation in group delay is not at all noticeable to human ear. At same time, group delay of the linear phase FIR filter of order 32 is 16 samples (1 ms). The system delay can be reduced drastically as the group delay of minimum phase filter is very small in the pass band. CLS FIR and Minimum phase CLS FIR filter can be applied in variable sample rate method [9], frequency

Fig. 2 CLS FIR Filter Bank structure

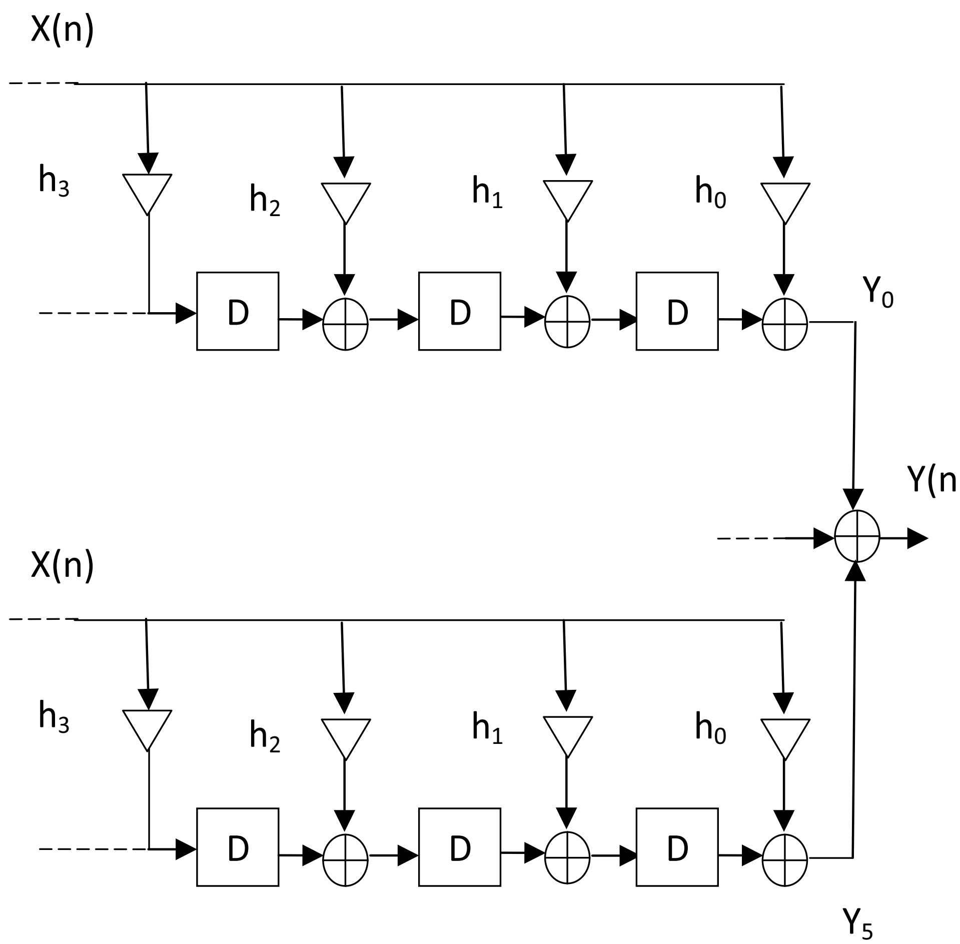


Table 1 Frequency bands used for CLS FIR Filter bank

Hearing loss type	No. of bands	Bandwidth (Hz)
Mild hearing loss at high frequencies	6	560,2240,1600,1760,2640, 3360
Mild hearing loss at all frequencies	6	960,720,2400,4000,2560, 1040
Mild to moderate hearing loss at low frequencies	6	600,700,3400,1680,1600, 1400

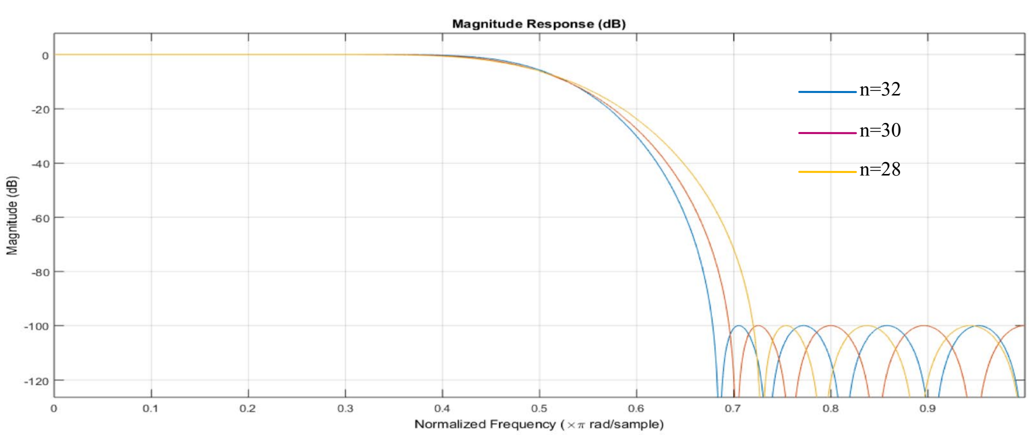


Fig. 3 Frequency response of CLS FIR low pass filter

Table 2 Transition width and matching error for different values of n for Mild hearing loss at high frequencies

Filter order(n)	Transition width (Radians)	Matching Error (dB)
32	0.1893π	2.09
30	0.1988π	3.12
28	0.2199π	3.23

response masking [26], all pass transformation [27], coefficient decimation and interpolation method [28, 29]

5.3 Reconfigurable CLS FIR filter bank

Coefficient decimation method (CDM) is used to vary bandwidth and edge frequencies of each filter in the bank in the proposed system. Thus the hearing aid can be dynamically reconfigured, based on the variations in hearing loss pattern of the user. The user can adjust the gain of each band with the help of an audiologist, based on the audiogram for a specific category of hearing loss. Frequency response of decimated filter obtained from the prototype filter with cut off frequency fc = 2 kHz, by using CDM is plotted in the Fig. 6. Frequency range of operation of a hearing aid is limited and the bandwidth requirements are also narrow, so Coefficient Decimation Method is used in the proposed structure to alter the bandwidth of

Table 3 Frequency bands used for Minimum phase Filter bank

Hearing loss type	No. of bands	Bandwidth (Hz)
Mild hearing loss at high frequencies	6	530,1120,1120,1600,1520,2880
Mild hearing loss at all frequencies	6	560,1120,880,2240,1600, 2800
Mild to moderate hearing loss at low frequencies	6	600,700,3400,2450,2000, 1400

Table 4 Transition width and matching error for different values of n of Minimum phase CLS FIR filter for Mild hearing loss at high frequencies

Filter order (n)	Transition width (Radians)	Matching Error(dB)
32	0.195π	2.51
30	0.1989π	3.22
28	0.220π	3.43

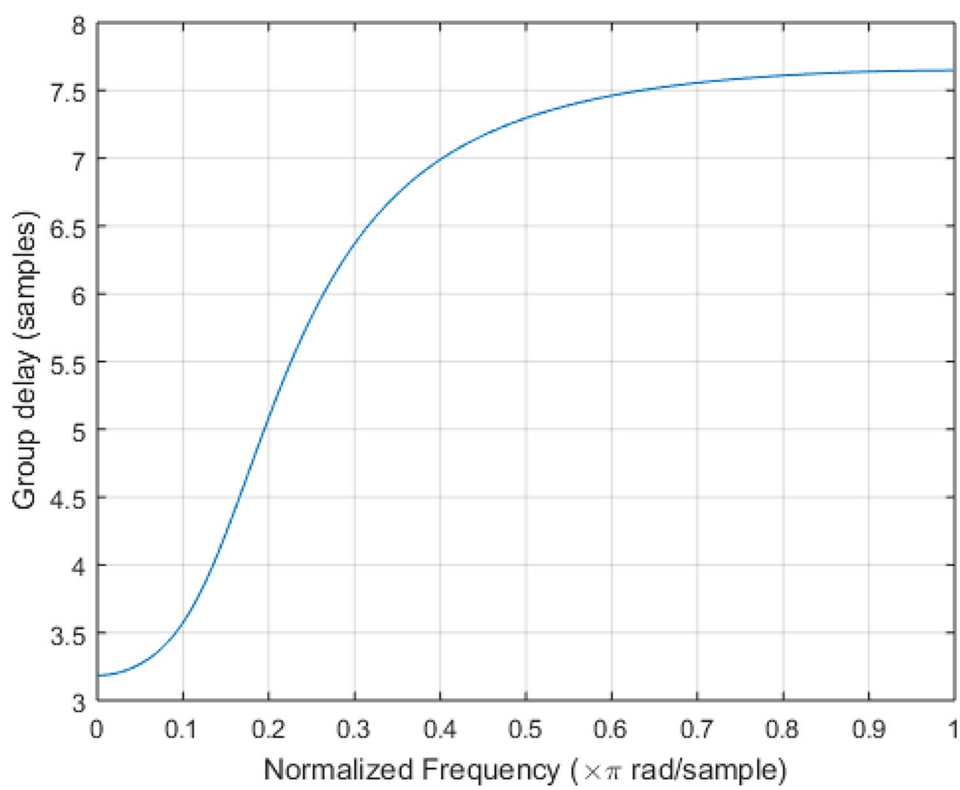


Fig. 4 Group delay characteristics of Minimum Phase CLS low pass FIR filter

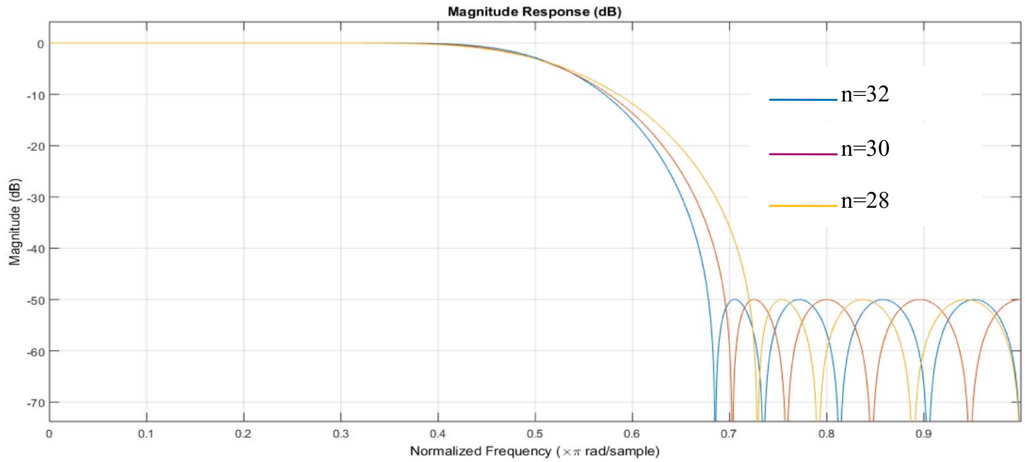
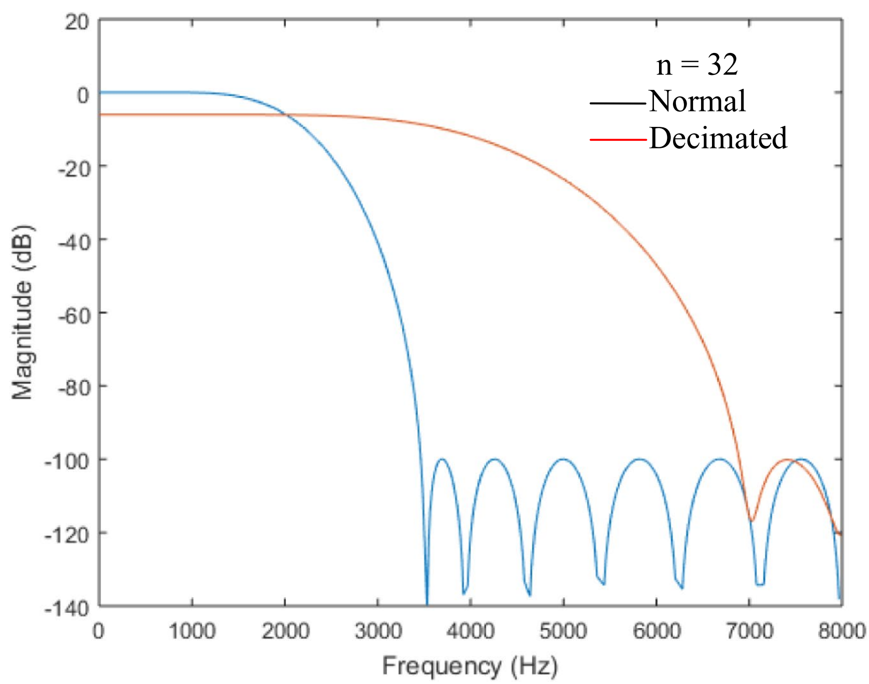


Fig. 5 Frequency response of Minium phase CLS FIR low pass filter

Fig. 6 Decimated CLS FIR low pass FIR filter



filters [28]. In CDM, every Mth coefficient of the prototype is grouped together by removing in between coefficients, so that the passband width of the prototype filter is increased by M times, where M is the integer decimation factor. The stopband attenuation characteristics deteriotes as the value of M increases, hence M = 2 is selected for the proposed system. A simple 2:1 multiplexer is used in the filter architecture to select normal or increased bandwidth filter. The hardware complexity of the CDM structure is very low compared to the reported reconfigurable filter bank structures. The reconfigurable filter bank structure is shown in Fig. 7.

6 Audiogram matching and performance evaluation

Based on the requirement of the hearing loss pattern, a set of bandwidths that could be used to fit audiograms is chosen as the initial step. A bank of digital filters is designed to obtain

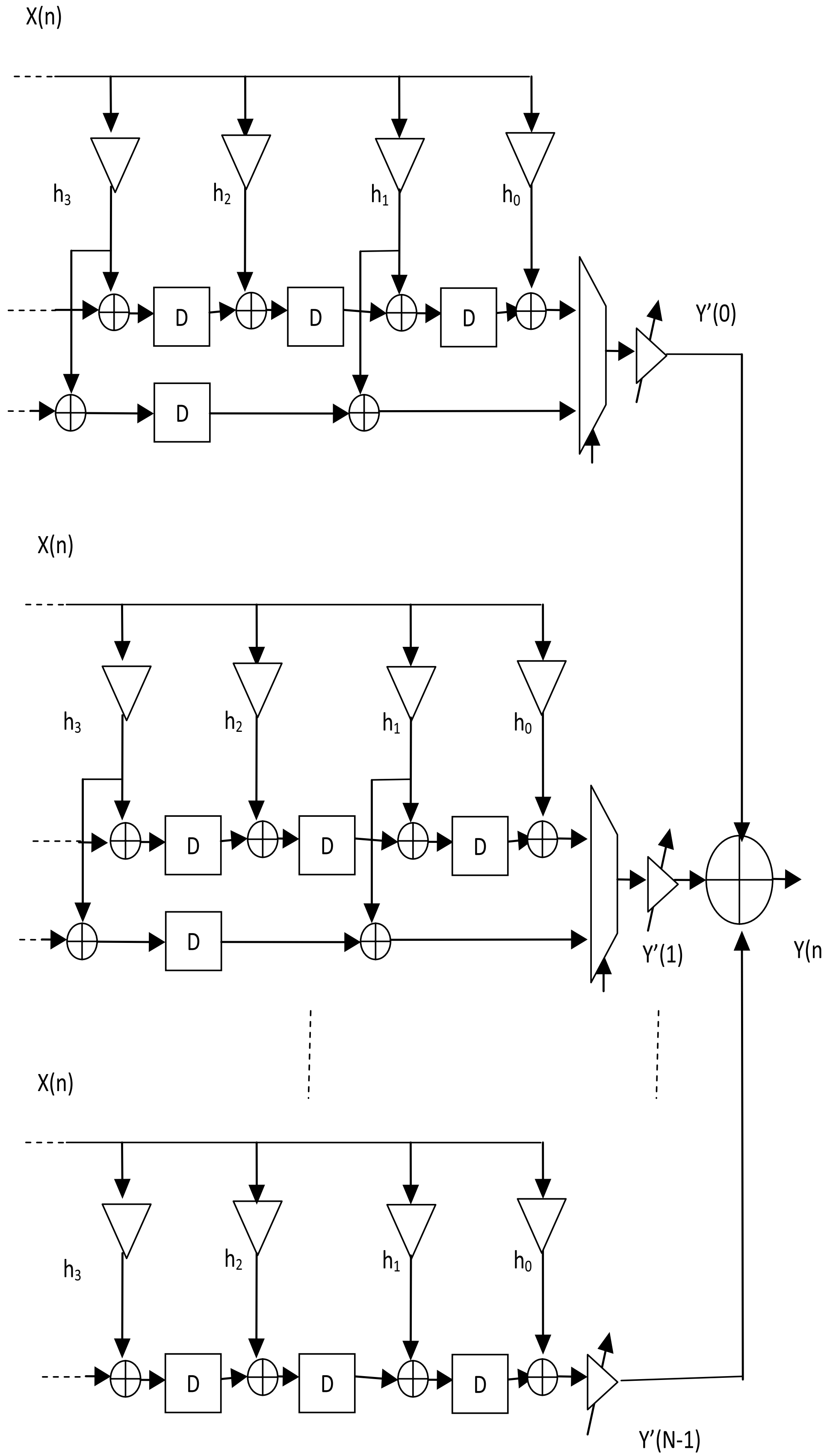


Fig. 7 Reconfigurable Filter bank structure

audiogram matching on various types of hearing losses. CLS FIR filter of order 32 with $\delta_p = 10^{-5}$ and $\delta_s = 10^{-5}$ is used as the prototype filter. The stop band attenuation of the CLS FIR Filter is 100 dB and the transition bandwidth is 0.1893 π radians. Minimum Phase FIR Filter with 17 coefficients is derived from the prototype CLS FIR filter. The stop band attenuation of the Minimum Phase FIR Filter is 50 dB and the transition bandwidth is 0.195 π radians. Optimal bandwidth for audiogram matching is selected by simulating them individually for a minimum matching error. Performance of filter bank structures is evaluated by MATLAB software. The overall response is obtained by adding the individual responses of each filter in the filter bank over the complete cycle. Only overlapping areas of individual filters are considered for calculating the total response of filter bank in the reported methods. This will lead to imperfect matching as the ripples in the stop band is also boosted while adding gain to individual filter response for minimizing matching error. Filter banks for matching audiogram pattern of mild hearing loss at all frequencies, audiogram pattern of mild to moderate hearing loss at low frequencies and mild hearing loss at high frequencies are designed. The frequency bands and gains required for the filter bank for each audiogram matching is obtained by a number of simulation trials.

6.1 Audiogram for mild hearing loss at high frequencies

People suffering from this kind of hearing loss cannot hear s's, z's,th's, v's and other soft, high frequency consonants. In this case, six bands are used in the bank structure for getting best results. The audiogram is shown in Fig. 8(a). The frequency response of the CLS FIR filter bank is shown in Fig. 8(b). Frequency response of the filter bank after separate gain adjustment to each band in order to match the audiogram is shown in Fig. 8(c). Matching error is plotted in Fig. 8(d) and maximum matching error obtained is 2.09 dB. The frequency response of the Minimum phase CLS FIR filter bank is shown in Fig. 9(a).The overall Frequency response of the filter bank is shown in Fig. 9(b). Matching error is plotted in Fig. 9(c) and 2.51 dB is the maximum matching error obtained in this case.

6.2 Audiogram for mild hearing loss at all frequencies

The audiogram shown in Fig. 10(a) represents mild hearing loss at all frequencies and in this case, the communication distance is reduced to 2 m where as the distance is up to 12 m for a normal person. Figure 10(b) shows the frequency response of the filter bank. Best matching results are obtained by properly adjusting the magnitude gains of 6 bands as shown in Fig. 10(c). Matching error is plotted in

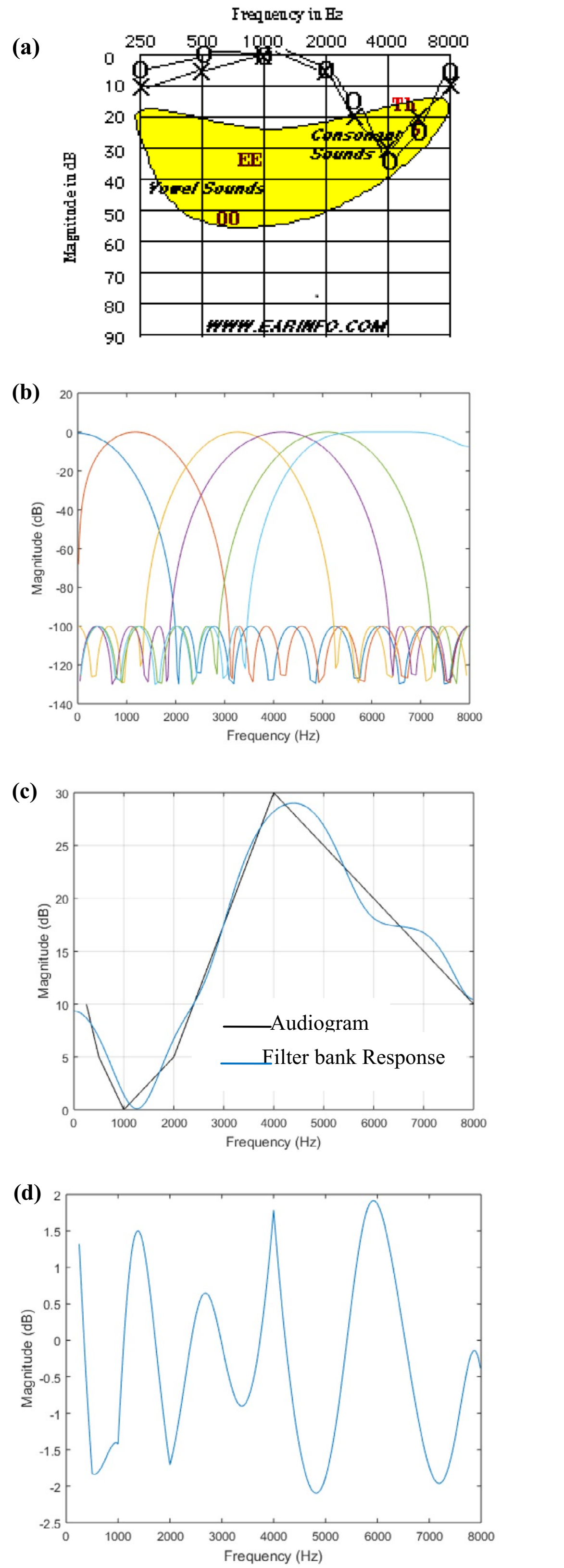


Fig. 8 Audiogram and experiment results. (a) Audiogram for mild hearing loss at high frequencies, (b) Frequency response of the filter bank, (c) Frequency response of the filter bank after separate gain to each band. (d) Matching Error

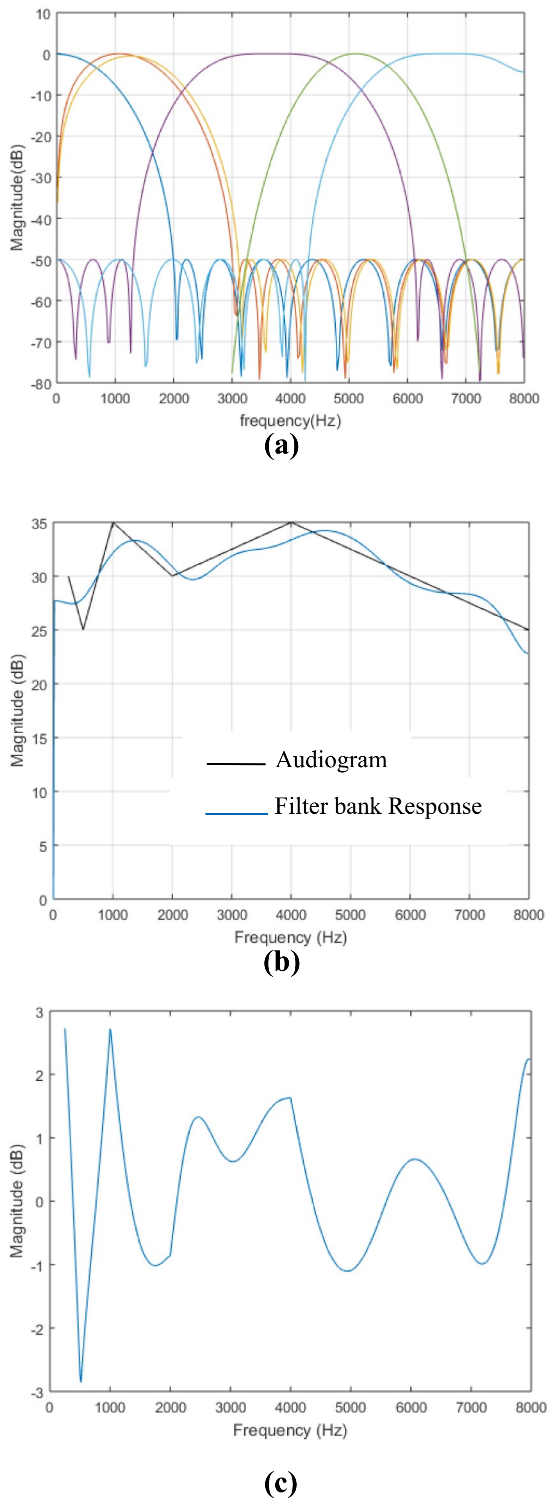


Fig. 9 Experiment results for mild hearing loss at high frequencies, using CLS Minimum phase filter bank (a) Frequency response of the filter bank, (b) Frequency response of the filter bank after separate gain to each band, (c) Matching Error

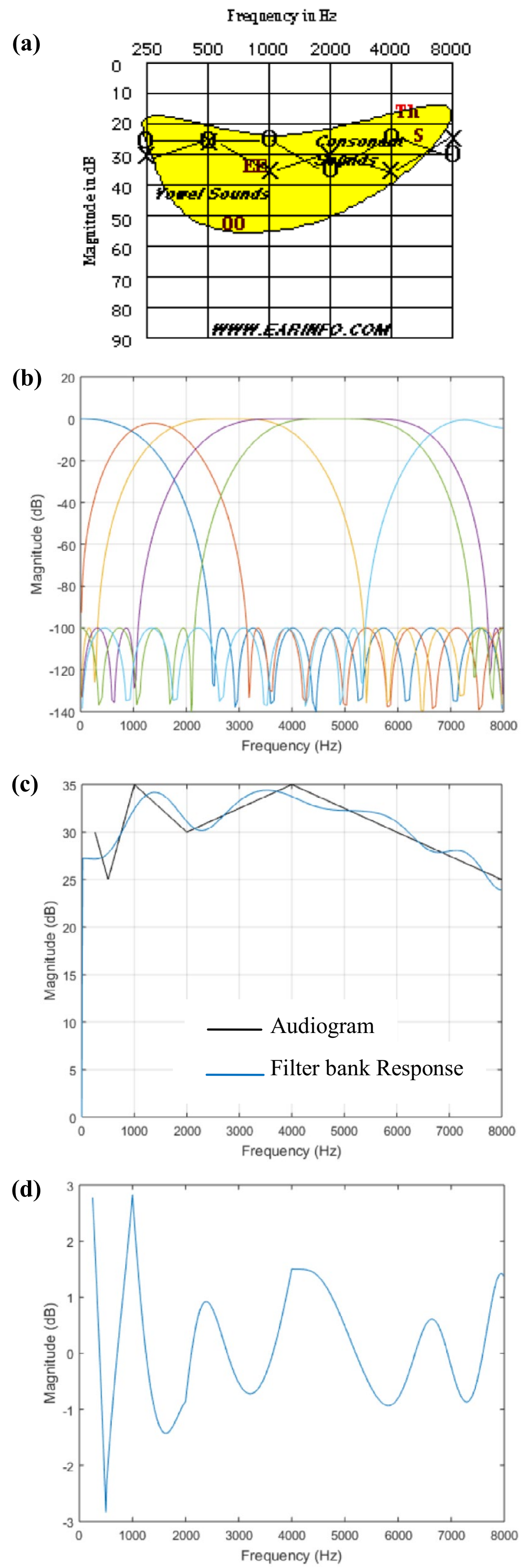


Fig. 10 Audiogram and experiment results. (a) Audiogram for mild hearing loss at all frequencies, (b) Frequency response of the filter bank, (c) Frequency response of the filter bank after separate gain to each band, (d) Matching Error

Fig. 10(d) and maximum matching error obtained is 2.83 dB. Minimum phase CLS FIR filter bank frequency response is shown in Fig. 11(a).The overall Frequency filter response of the filter bank is shown in Fig. 11(b). Matching error is plotted in Fig. 11(c) and 2.86 dB is the maximum matching error obtained in this case.

6.3 Audiogram for mild to moderate hearing loss at low frequencies

The audiogram shown in Fig. 12(a) represents the hearing loss of a person having mild to moderate hearing loss at low frequencies, with close to normal hearing in the high frequencies and the primary effect of such a type is the loss of overall loudness. The frequency response of 6 bands generated by the CLS FIR filter bank is shown in Fig. 12(b), the matching is shown in Fig. 12(c). Matching error is plotted in Fig. 12(d) and maximum matching error obtained is 2.34 dB. The frequency response of the Minimum phase CLS FIR filter bank is shown in Fig. 13(a).The overall frequency response matching of the filter bank is shown in Fig. 13(b). 2.38 dB is the maximum matching error obtained in this case and matching error is plotted in Fig. 13(c).

7 Performance comparison

Low power, area utilization and delay are the important requirements of digital hearing aid. The battery life and size of hearing aid device depends on the power and area utilization of the filter bank structure. Low pass filters and filter bank structures for matching of audiogram for mild hearing loss at high frequencies are synthesized using Cadence Encounter(R) RTL compiler v11.1. Synthesis results are shown in Table 5 and 6. From Table 6, it is clear that, around 45% saving in area utilization is obtained in the proposed method 2 compared to method 1. Similarly the percentage saving in power is 43% in method 2. The percentage saving in area utilization and power in the proposed method 1 compared to earlier work [17] are 48% and 50% respectively. So the proposed filter bank structures consumes less area and low power and thus reduces size of hearing aid and increases its battery life. Minimum phase filters are free from the stability issues associated with low complex IIR filters reported in the literature. The delay is also minimum in the proposed method 2. But price paid is the minute phase variation in the pass band and reduction in stop band attenuation in the minimum phase filter characteristics. It is reported that human ear is insensitive to such minute variations in phase of audio signals. If the selective gain boosting required for audiogram matching is less than 50 dB, stopband attenuation characteristics of minimum phase filter does not affect the performance of hearing aid. These two issues are resolved

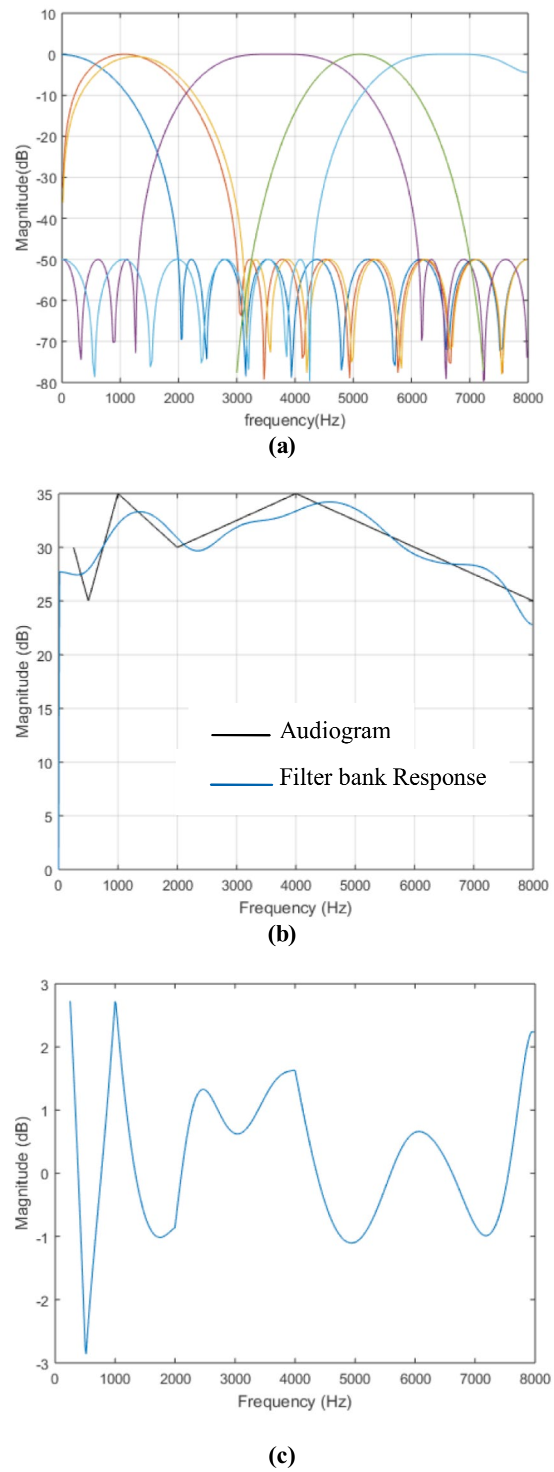


Fig. 11 Experiment results. for mild hearing loss at all frequencies using CLS Minimum phase filter bank, (a) Frequency response of the filter bank, (b) Frequency response of the filter bank after separate gain to each band, (c) Matching Error

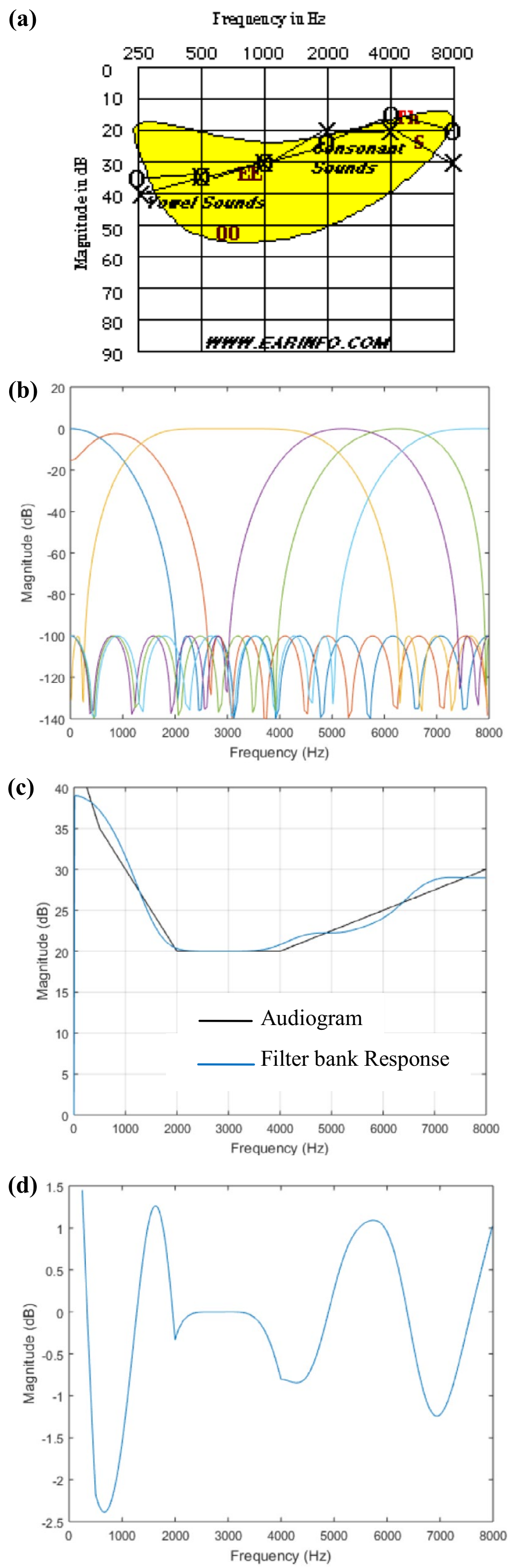


Fig. 12 Audiogram and experiment results. (a) Audiogram for mild to moderate hearing loss at low frequencies, (b) Frequency response of the filter bank, (c) Frequency response of the filter bank after separate gain to each band, (d) Matching Error

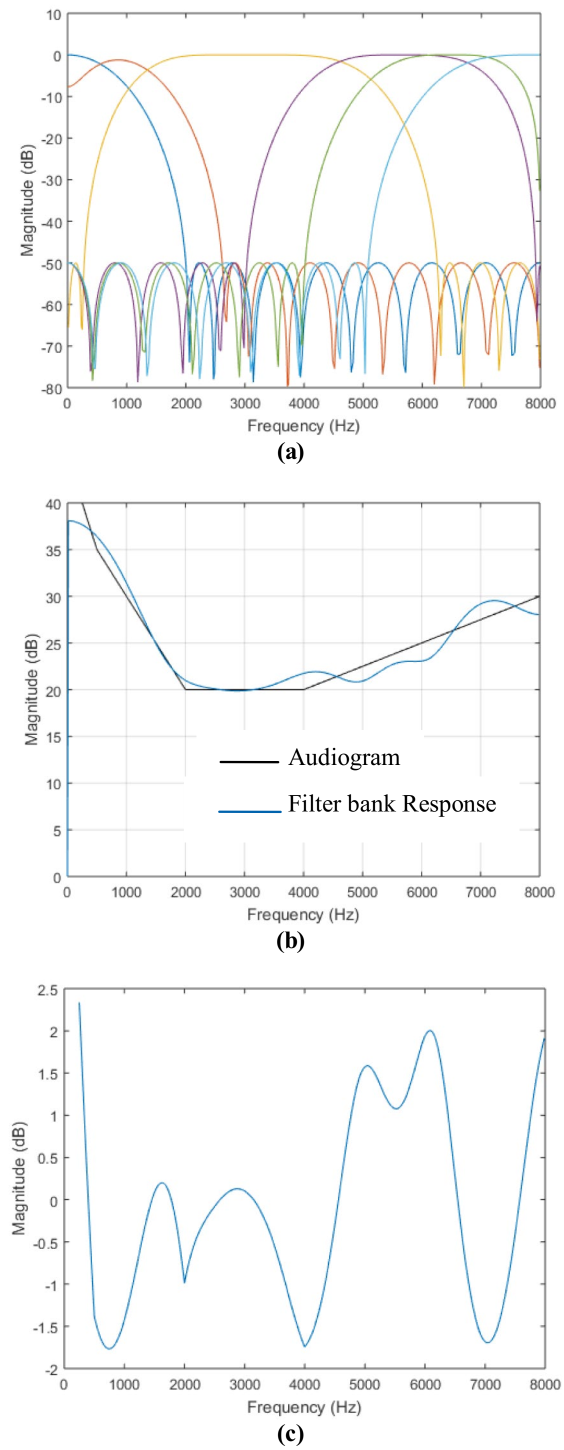


Fig. 13 Experiment results. for mild to moderate hearing loss at low frequencies using CLS Minimum phase filter bank, (a) Frequency response of the filter bank, (b) Frequency response of the filter bank after separate gain to each band, (c) Matching Error

Table 5 Low pass filter synthesis results

	Mahesh [17]	Proposed 1	Proposed 2
Area(μm2)	473,965.43	417,170.48	142,632.71
Power(nW)	100,059,902	56,522,573.6	32,658,881.1
Delay(pS)	4452	4452	4452

Table 6 Synthesis results of Filter bank for Mild hearing loss at high frequencies

	Mahesh [17]	Proposed 1	Proposed 2
Area(μm2)	2,499,210.81	1,295,260.24	708,130.68
Power(nW)	513,682,049	257,771,987	147,658,201
Delay(pS)	4919.40	4801.50	4771.20

in the proposed method 1, but at the cost of increase in hardware. Still the proposed method 1 consumes less area, low power and takes minimum delay compared to the reported methods in literature so far. The area utilization and power consumption increases with the increase in number of multipliers of filter bank. A performance comparison of proposed structures per band with the reported filter bank structures used for digital hearing aid design is given in the Table 7. Huang et al. [13] proposed a filter bank structure based on non linear transformation in which number of multipliers/band required is 187 and the delay is 7.77 ms. In Filter bank based on fractional interpolation reported in [14], number of the multipliers/band used is 84 and the delay is 13.5 ms. Non uniform MDFT filter bank is used in the method by Sakthivel et al. [15] in which complexity is expressed in terms of the order of filter and the minimum number of multipliers required for a filter with order n is generally n + 1. Farrow structure based variable bandwidth filter is detailed in [16] in which the number of multipliers required for filter per band is 33 and the delay is 2 ms. But the accumulator and additional circuitry required to transfer the variable parameter makes the system complicated. In this paper, a CLS FIR filter bank structure is proposed in method 1 and a Minimum phase CLS FIR filter bank structure is proposed in method 2. The number of multipliers per filter is only 17 and the maximum pass band group delay is only 0.375 ms for the second method. There is minute group delay variation is of the order of 0.1875 ms is observed in method 2, but it is not at all noticeable by human ear. The number of multipliers required in CLS FIR filter in method 1 is 33 and the delay is 1 ms. The stop band attenuation of CLS FIR filter is 100 dB which is thus more preferable for profound hearing loss cases in which the selective amplification of more than 80 dB is required for audiogram matching. In most of the reported methods, stop band attenuation of individual filter

Table 7 Performance comparison

Hearing Loss	Method	Multipliers	No. of bands	Delay (ms)	Matching Error (dB)
Mild hearing loss at high frequencies	Huang et al. [13]	187	7	7.77	1.65
	Sakthivel Elias [15]	102	10	3.18	2.2
	Amir,Bindiya [14]	84	13	18.61	2.54
	Mahesh,Shahana [17]	74	10	2.25	1.82
	Raghu,Nisha [16]	36	6	2.06	1.08
	Proposed 1	33	6	1	2.09
	Proposed 2	17	6	0.375	2.51
Mild hearing loss at all frequencies	Huang et al. [13]	187	7	7.77	1.35
	Sakthivel Elias [15]	56	9	1.75	1.69
	Amir,Bindiya [14]	84	-	13.55	1.49
	Mahesh,Shahana [17]	74	8	2.25	1.61
	Raghu,Nisha [16]	36	6	2.06	0.98
	Proposed 1	33	6	1	2.83
	Proposed 2	17	6	0.375	2.86
Mild to moderate hearing loss at low frequencies	Huang et al. [13]	187	7	7.77	3.75t
	Sakthivel Elias [15]	29	6	0.87	1.87
	Amir,Bindiya [14]	84	-	13.55	1.49
	Mahesh,Shahana [17]	74	8	2.25	2.6
	Raghu,Nisha [16]	36	6	2.06	2.42
	Proposed 1	33	6	1	2.34
	Proposed 2	17	6	0.375	2.38

is always less than 80 dB. Matching error of the proposed structures are greater than 2 dB but within the tolerable limit of 3 dB. In most of the reported methods, only overlapping areas of individual filter responses are considered for calculating the overall response of the filter bank. This may lead to unrealistic audiogram matching because ripples in the pass band are also boosted while the pass band gains are selectively increased to minimize the matching error. But in the proposed structures, the overall response of the filter bank is obtained by considering the complete cycle responses of individual filters. From the comparison table, it is evident that the proposed filter bank structures have better performance than the reported methods. So hearing aid using the proposed filter bank structures consumes less area and low power thus improves battery life and minimizes size of the device. The proposed method can be easily extended to other common hearing loss patterns also. The proposed structures can be made reconfigurable by using decimation technique. Hardware complexity of this multiplexer based structure is less compared to the reported methods. As the number of bands used in the filter bank is only 6 and bandwidths used are also narrow, decimation method used in this work is more cost effective than complex systems.

8 Conclusion

Small size and low power requirements are the important features of an optimal digital hearing aid. Filter bank structures proposed in this work consume less area and low power than the reported methods and thus suitable for the design of optimal hearing aids. The hardware requirement is reduced drastically as the proposed designs have significant reduction in multiplications per sample. The proposed filter bank structures based on CLS FIR filter and Minimum phase filter show very low delay and the matching errors of filter banks are within the tolerable limit. The complete cycle responses of individual filters are considered for the calculation of overall filter bank response. The reconfigurable structure proposed is based on CDM and enables the system to adapt variations in audiogram pattern that can happen in future, without much increase hardware complexity. The delay and hardware efficient CLS FIR and Minimum phase filter bank structures are synthesized using Cadence Encounter(R) RTL Compiler v11.10 software.

Compliance with ethical standards

Conflict of interest The authors declare that they have no conflict of interest.

Informed consent Declare that the research work did not include any data that requires informed consent.

Research involving human and animal participants Declare that the research work did not involve any human participants or animals.

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