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Efficient design of non-uniform cosine modulated filter banks for digital hearing aids

Shaeen Kalathil*, Elizabeth Elias

Department of Electronics and Communication Engineering, National Institute of Technology Calicut, India

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ABSTRACT

In this paper, efficient and simple designs of non-uniform filter banks for digital hearing aid applications are proposed. Hearing aids should be individually tuned to satisfy the requirements of hearing impaired persons. Cosine modulated filter banks are one popular filter bank having simple design procedure with efficient implementation structure. In the proposed structure for hearing aid, non-uniform subbands are obtained using two methods. The first method is by merging the adjacent channels of a uniform filter bank and the second method, is by using transition filters between two filters with different bandwidths. The advantages of the proposed structure are simple design procedure, less implementation complexity, greater flexibility in tuning the subbands for various types of audiograms and improved performance in terms of matching error.

1. Introduction

Hearing aids are devices, designed to amplify the sounds for hearing impaired persons for making speech more comprehensible. Different hearing aids available in the market are analog, programmable analog and digital hearing aids. Analog hearing aid is the initial type of hearing aid, which is less flexible in discriminating sounds in noisy environment [1,2]. Analog filters have limited abilities in terms of frequency shaping and it is difficult to obtain linear phase characteristics. Programmable analog hearing aid has an embedded circuitry to adjust the gain required by frequency bands [1]. Digital hearing aid is the popular hearing aid which utilizes advanced digital signal processing algorithms for improved performance and efficient implementation [1]. Hearing aids require a bank of filters to suitably adjust the gain characteristics in the required band of frequency. Digital filter banks decompose the input signal into different frequency regions. This facilitates to give appropriate gain to specific frequencies. As a result efficient designs for digital hearing aid can be achieved.

An audiogram is the graph showing the frequency versus intensity (loudness of the sound required by the person) measured in decibels. Smaller variations in an audiogram are normal, but larger variations are the indication of hearing impairment. Hearing impairment in different patients varies with respect to frequency, or in other words, the audiograms of different patients will be different. Otosclerosis disease results in an audiogram with significant loss at all frequencies. Ménière's disease results in severe loss at low frequencies. Age related hearing loss or presbycusis has normal sensitivity at low frequencies, but progressively poorer sensitivity for higher frequencies. The hearing aid should adjust the volume of received signal at different selected frequencies within a given spectrum. Initially the input signal is passed through a filter bank that divides them into different frequency components. The hearing aid adjusts the gain for various frequency bands for improving the hearing.

Different uniform multi-rate filter banks have been proposed for hearing aid applications [3–7]. In [4,5] oversampled or non-maximally decimated filter banks are used, which will result in an increased implementation complexity. Non-uniform filter banks are the preferable choice for hearing aid applications, since the sharp transitions in the audiograms can be perfectly matched with narrow passband filters and slower transitions can be matched with wider passband filters. The number of subbands required can be reduced and as a result, the implementation complexity will be less. Improved matching errors can also be obtained by using non-uniform filter banks. Tree structured filter bank structure is one method to obtain non-uniform decomposition in a simple manner [2]. But it has some limitations in the design of the bandwidths of subband filters such as restricted sampling rates and high signal delay.

Another approach to obtain non-uniform subbands for hearing aid application is by using interpolated FIR filter [8]. The same

* Corresponding author. Tel.: +91 9447100244.
E-mail address: shaeen_k@yahoo.com (S. Kalathil).

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method is modified using frequency response masking approach in [9,10]. Here, the prototype filter is designed using a halfband filter and interpolated using different factors. The required non-uniform subbands are obtained by using masking filters and combining different branches using adders. This structure is designed for a particular set of audiograms. The design time is also more, as the prototype and masking filters with different interpolation factors are to be separately designed. This structure cannot be used as a general one for all audiograms. Hence there is no design flexibility and the design will not be able to efficiently fit the audiograms of all types of hearing defects.

Variable bandwidth filters are proposed in [11,12], for hearing aid application with non-uniform subbands. Here, the variable bandwidth filters are implemented using two arbitrary sampling rate converters and a fixed bandwidth FIR low pass filter. By adjusting the sampling rate converters, the bandwidths of the filters are changed, without changing the coefficient values. Using frequency shifting, the filters are placed at appropriate frequency regions. The overall complexity of the filters is very high, but the design has flexibility in getting different bandwidths from a single low pass FIR filter. Variable bandwidth filters using three IIR filters are proposed in [12]. The individual bandedge frequencies and gains of the three filters are variable and hence can be tuned to match the different audiograms. However, the phase response does not have exact linear phase, and may result in different delays for different frequencies.

Uniform maximally decimated cosine modulated filter banks (CMFB) are proposed for hearing aid application in [7]. The main objective is to design a low delay uniform CMFB and the design is done by solving a constrained non-linear optimization problem. Compared to uniform decomposition of subbands, non-uniform decomposition of subbands is the preferred choice for hearing aid application. In perfect reconstruction (PR) filter banks, the output will be a weighted delayed replica of the input. In case of near perfect reconstruction (NPR) filter banks, a tolerable amount of aliasing and amplitude distortion errors are permitted. Design of NPR CMFB is easier and constitutes less number of constraints compared to the corresponding PR CMFB. Moreover, for NPR CMFBs by allowing small aliasing and amplitude errors, the filter performance can be significantly improved [13]. It is difficult to attain high stopband attenuation with PR CMFB. For hearing aid applications, it is reported that, most people are not sensitive to errors ≤3 dB [12]. Hence NPR-CMFB will be sufficient for hearing aids, provided the amplitude error is within tolerable limits.

In this paper, two different approaches are proposed for the efficient design of NPR non-uniform cosine modulated filter banks for digital hearing aid applications. One approach is to design a uniform CMFB and merge the adjacent channels properly by satisfying the feasibility conditions [14]. Only the prototype filter is designed. All the other subbands are obtained by cosine modulation and merging of the subbands. In the second approach, different prototype filters are designed for the channels with different bandwidths [15]. By allowing a wider transition width to wider passband filter, the number of non-zero coefficients can be reduced. Transition filters are used whenever a transition of channels with different bandwidths occurs. For different audiograms, the non-uniform filter banks are designed and matching error is determined. The results show that the proposed techniques are very well suited for hearing aid application with design ease, simple design procedure, reduced implementation complexity and the flexibility to adjust the bandwidths of the subbands.

The remaining part of the paper is organized as follows. Section 2 explains the different methods for designing non-uniform filter banks using cosine modulation. A brief introduction of uniform CMFB is given and then the non-uniform filter banks are derived using merging of adjacent channels as well as using different prototype filters. The proposed design methodology is explained in Section 3. The proposed block diagram and the design procedure adopted are also explained. The design examples for various types of audiograms and the performance comparison are given in Section 4. Finally Section 5 gives the conclusion.

2. Cosine modulated non-uniform filter banks

2.1. Design of cosine modulated uniform filter banks

The different errors encountered in multirate filter banks are phase distortion, amplitude distortion and aliasing distortion. Suppose $p_0(n)$ is the impulse response of the prototype filter and $P_0(z)$ is the corresponding transfer function. The M analysis and synthesis filters of an M channel uniform CMFB are given by Eqs. (1) and (2), respectively [16]

$$H_k(z) = a_k c_k P_0(zW^{(k+0.5)}) + a_k^* c_k^* P_0(zW^{-(k+0.5)}) \tag{1}$$

$$F_k(z) = a_k^* c_k P_0(zW^{(k+0.5)}) + a_k c_k^* P_0(zW^{-(k+0.5)}) \tag{2}$$

Here $W = e^{-j(\pi/M)}$ and $k = 0, 1, 2, \ldots, M-1$. a_k and c_k are unity magnitude complex constants. The proper choice of a_k and c_k ensures aliasing cancelation between adjacent channels and also eliminates phase distortion. The impulse response coefficients of the analysis and synthesis filters are given by $h_k(n)$ and $f_k(n)$ for $a_k = e^{j\theta_k}$, $c_k = W^{(k+0.5)(\frac{N}{2})}$ and $\theta_k = (-1)^k \frac{\pi}{4}$ [16].

$$h_k(n) = 2p_0(n)\cos\left(\frac{\pi}{M}(k+0.5)\left(n-\frac{N}{2}\right) + (-1)^k\frac{\pi}{4}\right) \tag{3}$$

$$f_k(n) = 2p_0(n)\cos\left(\frac{\pi}{M}(k+0.5)\left(n-\frac{N}{2}\right) - (-1)^k\frac{\pi}{4}\right) \tag{4}$$

$$k = 0, 1, 2, \ldots, M-1$$
$$n = 0, 1, 2, \ldots, N-1$$

N is the order of the prototype filter. The prototype filter is chosen as a linear phase filter with real coefficients. To reduce the error in amplitude distortion, the condition to be satisfied is as given below [16]

$$|P_0(e^{j\omega})|^2 + |P_0(e^{j(\omega-\frac{\pi}{M})})|^2 = 1 \quad \text{for } 0 \le \omega \le \frac{\pi}{M} \tag{5}$$

From the above relation it can be shown that when $\omega = \frac{j\pi}{2M}$ [17]

$$|P_0(e^{\frac{j\pi}{2M}})| \approx 0.707 \tag{6}$$

The passband edge and stopband edge frequencies can be iteratively varied in small step size to satisfy the condition given by (6) within a given tolerance value.

The structure of an $\tilde{M}$ channel cosine modulated non-uniform filter bank is shown in Fig. 1. A set of $\tilde{M}$ analysis filters $\tilde{H}_k(z)$,

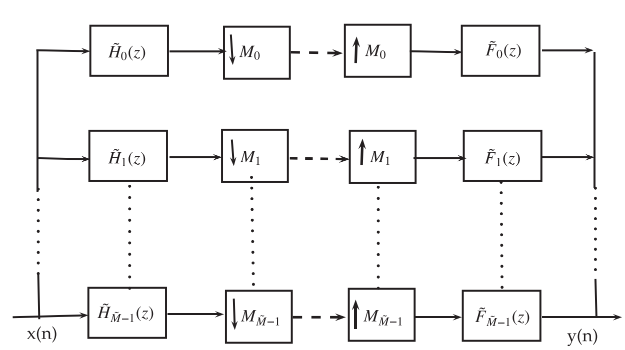

Fig. 1. Cosine modulated non-uniform filter bank.

Table 1
Performance comparison of non-uniform CMFB.

Work	Stopband attenuation (dB)	Filter order	Amplitude distortion error
Lee et al. [14]	46.3	40	2.7×10^{-2}
Ogale et al. [19]	110	163	6.5×10^{-3}
Kumar et al. [20]	110	143	2.6×10^{-3}
Shaeen et al. [18]	110	133	1.2×10^{-3}

$0\le k\le \tilde{M}-1$ decomposes the input signal into $\tilde{M}$ subbands. A set of synthesis filters $\tilde{F}_k(z)$, $0\le k\le\tilde{M}-1$ combines the $\tilde{M}$ subband signals. For critically sampled filter banks, the decimation factors should satisfy the condition $\sum_{k=0}^{\tilde{M}-1}(1/M_k)=1$.

2.2. Merging method

A simple and efficient design of NPR non-uniform cosine modulated filter banks with integer sampling factors is by merging the adjacent channels of a uniform CMFB [14]. Initially an NPR uniform CMFB is designed using linear search technique [18]. Only the prototype filter is designed and optimized. All the analysis and synthesis filters are obtained from this filter by cosine modulation. A linear phase prototype filter, results in the total linear phase response of the filter bank. Adjacent channel aliasing elimination is inherent in the filter design itself. 2Mth band constraint on prototype filter eliminates the amplitude distortion [16].

The non-uniform bands are obtained by merging the adjacent analysis and synthesis filters. Consider the analysis filter $\tilde{H}_i(z)$, which are obtained by merging l_i adjacent analysis filters.

$$\tilde{H}_i(z)=\sum_{k=n_i}^{n_i+l_i-1}H_k(z),\quad i=0,1,\ldots,\tilde{M}-1 \tag{7}$$

Here, n_i is the upper band edge frequency ($n_0=0<n_1<n_2<\cdots<n_{\tilde{M}}=M$) and l_i is the number of adjacent channels to be combined. The synthesis filter $\tilde{F}_i(z)$, is obtained in a similar way.

$$\tilde{F}_i(z)=\frac{1}{l_i}\sum_{k=n_i}^{n_i+l_i-1}F_k(z),\quad i=0,1,\ldots,\tilde{M}-1 \tag{8}$$

The corresponding decimation factor M_i, is given by $M_i=M/l_i$. To satisfy the adjacent channel alias cancelation constraint, the l_i and n_i are chosen such that n_i is an integral multiple of l_i, for all $i=0,1,\ldots,\tilde{M}-1$ [14]. This is termed as the feasibility condition. Table 1 shows the performance comparison of different non-uniform cosine modulated filter banks using merging method. It is observed that the method proposed in [18] gives minimum amplitude distortion error with minimum filter order.

2.3. Transition filter approach

The merging method for generating cosine modulated non-uniform filter bank will result in the same number of non-zero coefficients for all the channels (narrow passband as well as wider passband filters). The transition width of all the subbands are equal. For hearing aid applications this need not be an essential requirement. The design of wider channels can be relaxed by having a wider transition band. For example, the number of non-zero coefficients required for 4 channel filter bank will be half of that of 8 channel filter bank. The filters with the same transition width for different passband width will have equal number of non-zero coefficients.

Transition filters allow the design of non-uniform modulated filter banks with different transition bands for filters with different passbands [15]. Transition filters are complex coefficient low pass filters with asymmetrical transition widths on the positive and negative frequencies. A typical frequency response characteristics is shown in Fig. 2. ω_{p1} and ω_{s1} are the passband and stopband edge frequencies on the negative frequency axis and ω_{p2} and ω_{s2} are the passband and stopband edge frequencies on the positive frequency axis.

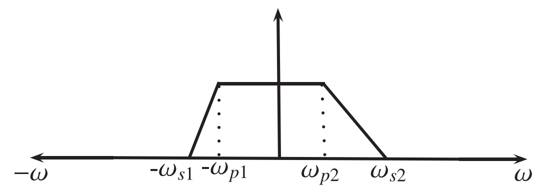

Fig. 2. Typical frequency response of transition filter.

Transition filters are used in between an M_0 channel filter bank and M_1 channel filter bank or vice versa, to get a non-uniform decomposition as shown in Fig. 3. In order to make the total filter bank response as that of uniform NPR CMFB, the magnitude response of the prototype transition filter should be as given in Fig. 2. The unsymmetrical, complex coefficient transition filter matches the transition band of $H_0(z)$ on the left side and $H_2(z)$ on the right side. For channel i the impulse responses of the analysis and synthesis filters are as given below [15].

$$h_i(n)=a_ip_i(n)e^{-j\left(\frac{\pi}{M_i}(k+0.5)\left(n-\frac{N_i}{2}\right)\right)}+a_i^*p_i^*(n)e^{-j\left(\frac{\pi}{M_i}(k+0.5)\left(n-\frac{N_i}{2}\right)\right)} \tag{9}$$

$$f_i(n)=a_i^*p_i(n)e^{-j\left(\frac{\pi}{M_i}(k+0.5)\left(n-\frac{N_i}{2}\right)\right)}+a_ip_i^*(n)e^{-j\left(\frac{\pi}{M_i}(k+0.5)\left(n-\frac{N_i}{2}\right)\right)} \tag{10}$$

$n=0,1,2,\ldots,N_i-1$

Here, $p_i(n)$ is the impulse response of the prototype filter for the ith channel of order N_i. To cancel the adjacent channel aliasing, a_i can be chosen as $a_i=e^{j(-1)^i\frac{\pi}{4}}$. To avoid distortions at 0 and π, the transition filters are not allowed at these frequencies. Transition filters can be used in the middle subbands, other than those including 0 and π. The analysis filter, $H_t(z)$ is obtained by the complex modulation as given by Eq. (9). The decimation set should be a compatible set, to avoid aliasing distortion [14,21]. For filters with real coefficients, Eqs. (9) and (10) will reduce to that of cosine modulation as given in Eqs. (3) and (4), respectively. For prototype filters with complex coefficients, complex modulation as given by Eqs. (9) and (10) is done and the modulated filters will have real coefficients.

3. Design methodology

3.1. Audiogram matching

Audiograms are the graphs which show the audible thresholds for different frequencies. An audible threshold is the softest sound

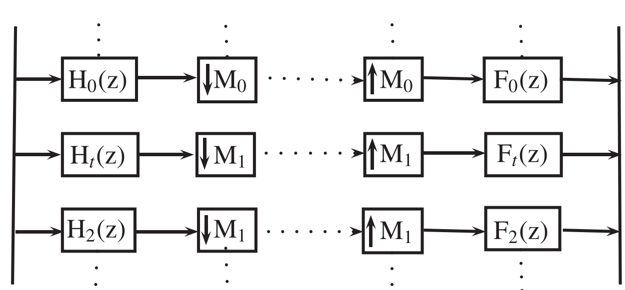

Fig. 3. Transition filter $H_t(z)$ in between two uniform channels of different widths.

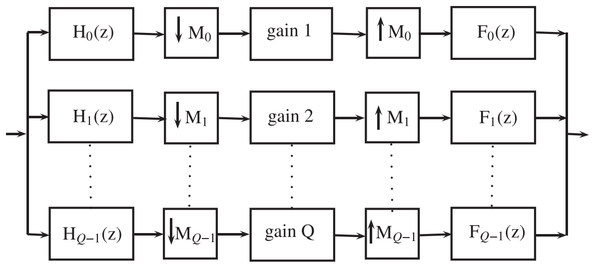

Fig. 4. Proposed structure for digital hearing aid using cosine modulated non-uniform filter banks.

a person is able to hear. The symbol 'O's indicate the estimations for right ear and 'X's indicate the estimations for left ear. The hearing loss based on intensity is classified as mild hearing loss that extends from 25 to 40 dB, moderate loss which is around 41–55 dB and moderately severe that extends from 56 to 70 dB. Further classification of hearing loss is severe loss which extends from 71 to 90 dB and profound loss (greater than 90 dB).

Audiograms for different patients will be different and hence the hearing aids should be suitably adjusted to satisfy the individual requirements. The audiogram of a patient with presbycusis may have smaller variations from normal at low frequencies, but with progressively larger variations at high frequencies. Some of the common hearing impairments seen in patients are mild hearing loss at high frequencies, mild hearing loss at all frequencies, mild to moderate hearing loss at low frequencies, severe hearing loss in the middle to high frequency range and profound hearing loss. Depending on the hearing deficiency, the slopes of an audiogram may have very steep slopes for some frequency ranges and low or gradual slopes for other frequency ranges. In order to match the audiograms for steep slopes, filters with narrow passband widths are required. Otherwise the matching error will cross the required tolerance of ±3 dB [12]. The small slopes in an audiogram can be efficiently matched with filters having wider passband widths. If uniform filter bank is used, the passband width of each filter should be very narrow and hence the total number of filters required will be high in order to keep the matching error a minimum. Therefore, the total number of filters required to match a given audiogram can be reduced significantly using non-uniform filter banks.

3.2. Design procedure

The block diagram of the proposed structure for digital hearing aid using cosine modulated non-uniform filter banks is shown in Fig. 4. A Q channel analysis filter bank decomposes the input into Q non-uniform subbands. After decimation, appropriate gain is given to each subband. The different subbands after amplification are interpolated and combined by the synthesis filter banks. The proposed design procedure is given below.

1. Initially a uniform filter bank is designed for audiogram matching with sufficient gains in each subband. The width of the passbands should be narrow in the regions with steep slopes in the audiograms. The total number of channels required is determined by the steepest slope. An error of ±3 dB will be tolerable by the human brain [12], as most people are not sensitive to lower errors.
2. The subbands are suitably merged in the frequency regions with very small slope. Care should be taken while merging the subbands such that the feasibility condition for the decimation factors is satisfied. Tolerable matching errors can be obtained with less number of subbands, when compared to the number of subbands using uniform filter banks. This can reduce the hardware implementation complexity.
3. The transition width of narrow passband filter is limited. While merging a number of channels, the passband width will be increased, but retains the same transition width and same number of coefficients. A wide passband filter can be designed with a wide transition width with lesser number of coefficients. The non-uniform filter bank can also be designed using different prototype filters for different channels. Transition filters are used whenever a transition of two different bandwidths occurs. The aliasing cancelation is achieved by the magnitude response of the transition filter and the compatible set of decimation factors. The zero phase transition filter is complex modulated to obtain the required subband and the corresponding coefficients will be real.

4. Design examples and performance comparison

In order to verify the efficacy of the filter bank proposed for hearing aid application, it is tested for various types of audiograms available in the internet and also on the audiograms of some hearing impaired persons collected from National Institute of Speech and Hearing, Trivandrum, India. All the simulations are done using a Dell precision T3610 work station using MATLAB R2013b. The prototype filter of the uniform CMFB is designed using least square approximation. For merging method, the subbands of the uniform filter bank are carefully combined, so that the feasibility condition is satisfied. For the non-uniform CMFB design using transition filters, different prototype filters are required for channels with different bandwidths. Real coefficient FIR filter is designed using least square approximation for the prototype filter with symmetrical transition width on both sides. The complex coefficient linear phase prototype filter for asymmetrical transition width, is designed using MATLAB Signal Processing Toolbox. Suppose $p_0(n)$ is the impulse response of the prototype filter with complex coefficients, then $p_0(n)$ should satisfy the complex conjugate symmetry as $p_0(n) = p_0^*(N_1 - 1)$, where N_1 is the filter order. All the prototype filters are designed iteratively by adjusting the edge frequencies in small step sizes to satisfy the condition given in Eq. (6) within a given tolerance value.

Audiogram for mild to moderate hearing loss at low frequency as given in Fig. 5 is considered first. Initially a uniform CMFB is designed and then sufficient gain is given to individual subbands. The audiogram matching with uniform filter banks is shown in Fig. 6. An 8 channel filter bank is required for the uniform filter bank so that the matching error is within ±3 dB. For narrower slopes the subbands can be merged. By merging the two subbands, which satisfy the feasibility conditions, the matching error remains the same or in some cases even lesser. Hence with less number of

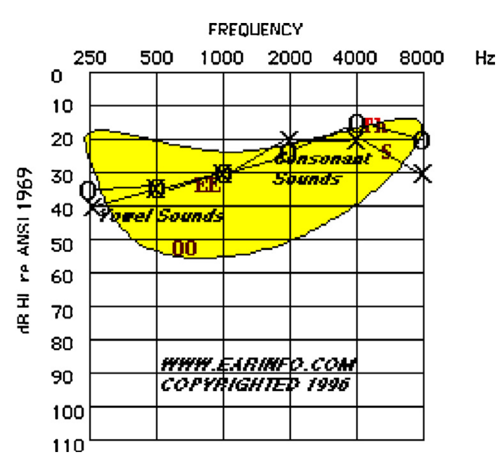

Fig. 5. Audiogram for mild to moderate hearing loss at low frequency [22].

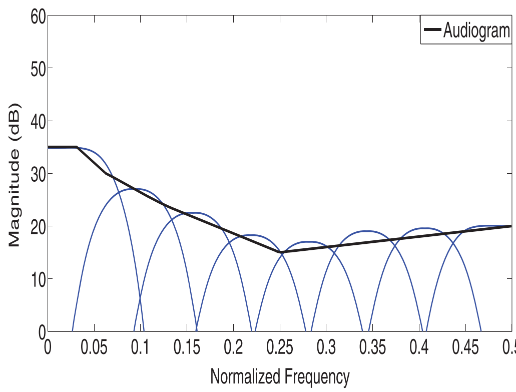

Fig. 6. Mild to moderate hearing loss at low frequencies audiogram matching using uniform CMFB.

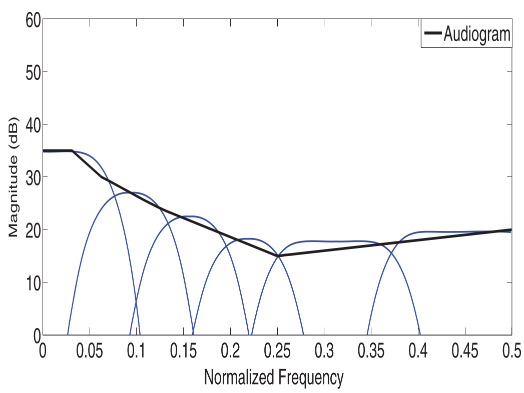

Fig. 7. Mild to moderate hearing loss at low frequencies audiogram matching using non-uniform CMFB – merging.

coefficients, the filter banks can be implemented. Instead of eight channels required with uniform filter bank, the non-uniform filter bank requires only six channels for the case of mild to moderate hearing loss at low frequency, as shown in Fig. 7. The number of filter coefficients required for each channel is the same for both uniform and non-uniform filter banks. The audiogram matching is shown in Fig. 7.

The total number of non-zero coefficients, as well as matching error can be reduced using transition filters for getting a non-uniform decomposition. The audiogram matching using transition filters is shown in Fig. 8. The audiogram matching error comparison for the three types of filter banks is shown in Fig. 9. Matching error is also found to be less for non-uniform CMFB designed using transition filters. The maximum matching error for mild to moderate

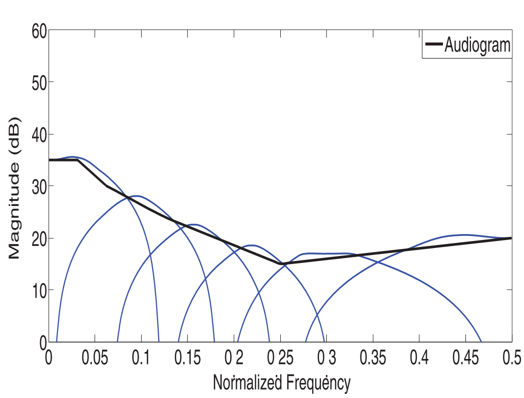

Fig. 8. Mild to moderate hearing loss at low frequencies audiogram matching using non-uniform CMFB – transition filters.

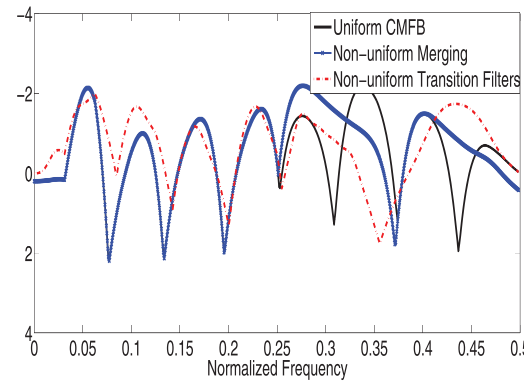

Fig. 9. Comparison of matching errors (mild to moderate hearing loss at low frequencies).

Table 2
Comparison of maximum matching error for mild to moderate hearing loss at low frequencies.

	Max. matching error (dB)	Prototype filter order	No. of channels
Method in [10]	4.1		16
Method in [11]	2.08	250 + 639	8
Uniform CMFB	2.19	63	8
Non-uniform merging method (proposed)	2.19	63	6
Non-uniform transition filter method (proposed)	1.9	49[a] + 69[b] + 79[c]	6

[a] Prototype filter order of 4 channel CMFB.
[b] Prototype filter order of 6 channel CMFB.
[c] Prototype transition filter order.

hearing loss at low frequency for the proposed method and the existing method is given in Table 2. Table 2 also compares the total number of channels required and order of the prototype filters. For uniform 8 channel CMFB, the order of the prototype filter is 63. For the method proposed in [11], 250 is the order of the fixed FIR filter and 639 is the order of the filters for sampling rate conversion. The design of non-uniform CMFB using transition filters requires a minimum of two prototype filters. The narrow band prototype filter can be obtained from the wider passband filter by interpolation [14]. In this paper, two real coefficient FIR prototype filters are designed for 4 channel and 8 channel filters and one complex coefficient FIR prototype filter is designed to be used in between the 8 and 4 channel filter banks. The transition width matches with that of 8 channel in the negative frequency region and transition width matches with that of 4 channel in the positive frequency region. The order of the transition filter is 79. The width of passband region is equivalent to the width of the 8 channel prototype. The prototype filter order of 8 channel filter bank is 69 and that of 4 channel filter bank is 49. The total number of non-zero coefficients for uniform CMFB is 504 and for non-uniform CMFB with merging is 378, whereas non-uniform CMFB using transition filter requires only 344 non-zero coefficients.

The filter banks are also designed for mild hearing loss at all frequencies, moderate hearing loss at high frequencies and mild hearing loss at low frequencies. Fig. 10 shows the audiogram matching for uniform filter bank. Figs. 11 and 12 show the audiogram matching for non-uniform filter bank design using merging method and transition filter approach, respectively. Table 3 gives the performance comparison of the uniform and non-uniform filter banks for different audiograms. It can be observed that non-uniform decomposition minimizes the total number of subbands for satisfying the required matching error. Also the transition filter

Table 3
Comparison of different audiogram matching using proposed approaches.

Type of hearing loss	Uniform CMFB		Non-uniform[a]		Non-uniform[b]	
	Max. matching error (dB)	No. of channels	Max. matching error (dB)	No. of channels	Max. matching error (dB)	No. of channels
Mild to moderate hearing loss at low frequencies	2.19	8	2.19	6	1.9	6
Mild hearing loss at all frequencies	2.2	16	2.15	9	2.12	9
Moderate hearing loss at high frequencies	2.49	28	2.49	16	2.2	16
Mild hearing loss (left ear)	2.55	28	2.38	10	2.23	11
Mild hearing loss at all frequencies (right ear)[c]	2.29	24	2.4	11	2.3	11
Mild hearing loss at all frequencies (left ear)[c]	2.3	24	2.3	8	2.4	8

[a] Merging method.
[b] Transition filter method.
[c] Audiogram collected from National Institute of Speech and Hearing, Trivandrum, India.

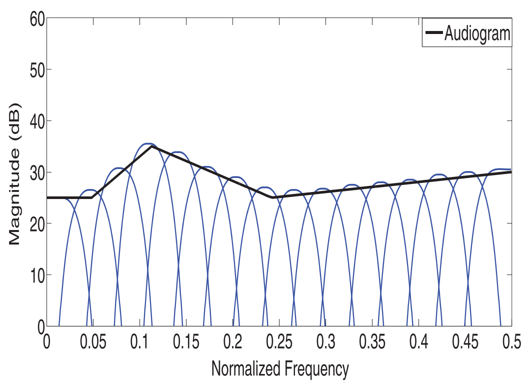

Fig. 10. Mild hearing loss at all frequencies audiogram matching using uniform CMFB.

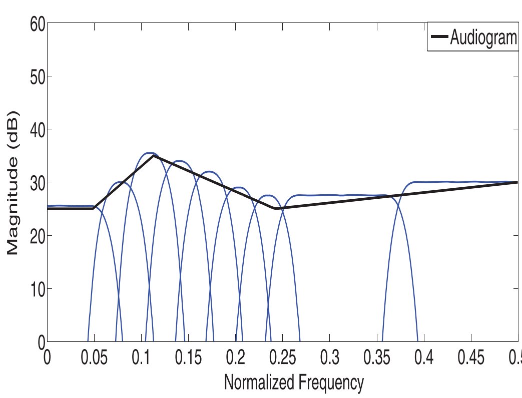

Fig. 11. Mild hearing loss at all frequencies audiogram matching using non-uniform CMFB – merging.

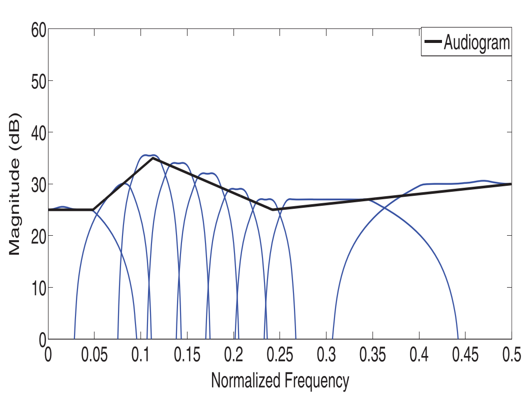

Fig. 12. Mild hearing loss at all frequencies audiogram matching using non-uniform CMFB – transition filters.

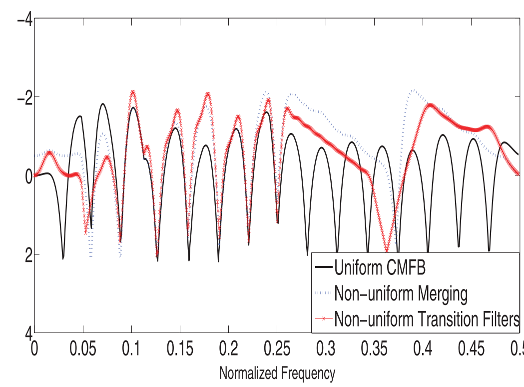

Fig. 13. Comparison of matching errors (mild hearing loss at all frequencies).

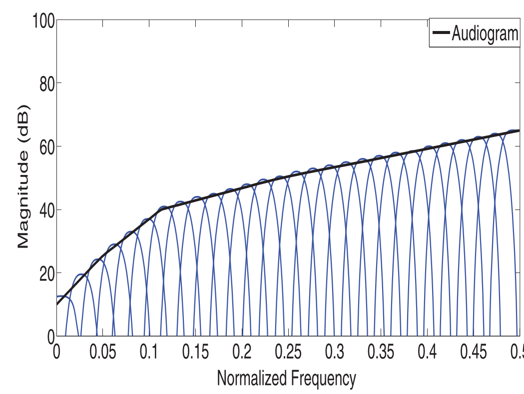

Fig. 14. Moderate hearing loss at high frequencies audiogram matching using uniform CMFB.

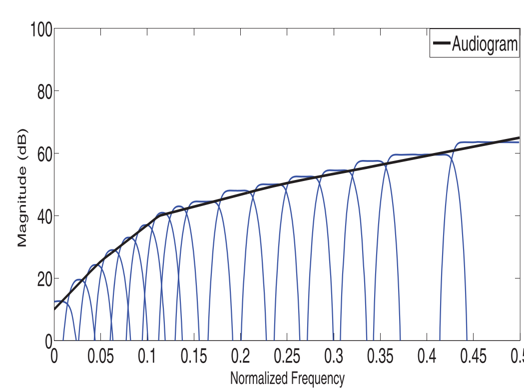

Fig. 15. Moderate hearing loss at high frequencies audiogram matching using non-uniform CMFB – merging.

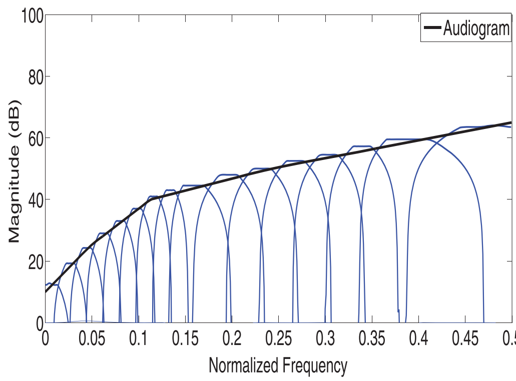

Fig. 16. Moderate hearing loss at high frequencies audiogram matching using non-uniform CMFB – transition filters.

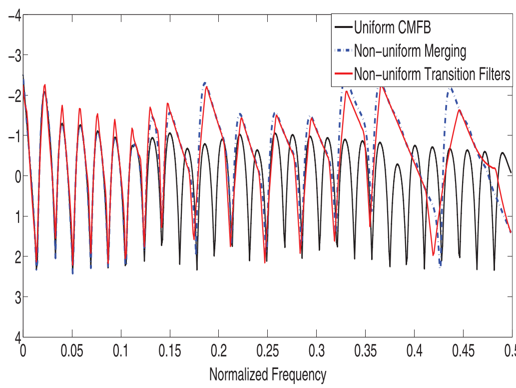

Fig. 17. Comparison of matching errors (moderate hearing loss at high frequencies).

approach gives less matching error compared to that using merging method. Hence non-uniform CMFBs can result in hearing aids with compact size (Figs. 13–17).

5. Conclusion

Non-uniform cosine modulated filter banks for hearing aid application is investigated and the performances are compared for various types of audiograms. It is observed that the proposed structures have a number of advantages compared to the existing methods. A simple design procedure with greater flexibility to tune the bandwidth to satisfy the different types of audiograms is developed. The implementation complexity of the hearing aid also reduces when non-uniform filter banks are deployed. The matching errors are also obtained within tolerable limits. The prototype filters are designed with high stopband attenuation and using a linear search to obtain the edge frequencies which minimizes the amplitude distortions.

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