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A Low Complex 12-Band Non-uniform FIR Digital Filter Bank Using Frequency Response Masking Technique For Hearing Aid

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Abstract— This paper presents the low complexity design of a non-uniformly spaced digital finite impulse response (FIR) filter bank for digital hearing aid application. Frequency response masking (FRM) technique is used for the implementation of 12 non-uniformly spaced subband filters. FRM technique ensure the drastic reduction in the number of multipliers and adders in linear phase FIR filter. It employs two half-band filters as prototype filters, which leads to significant savings in terms of arithmetic operations. Further reduction can be achieved by using a masking filter with minimum order. The simulation result shows that the proposed filter bank with stop band attenuation of 80 *dB* can be achieved with 10 multipliers. The audiogram fitting with the selected audiogram gives a maximum matching error of 1.4 *dB*. From the audiogram matching result, it can be seen that the introduction of non-uniformly spaced filter bank improves the audiogram matching especially for sharp transition in hearing losses at low and high frequencies

Keywords- *non-uniform filter bank, FIR filter, FRM technique, hearing aid, half-band filters, audiogram.*

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I. INTRODUCTION

A hearing aid is an electro-acoustic device which is designed to amplify sound, with the aim of making speech more intelligible [1]. The main task of the hearing aid is to selectively amplify the audio sounds such that the processed sound matches one's audiogram. Audiogram is a graph that shows the softest sounds a person can hear at different pitches or frequencies. Hearing thresholds become high at certain frequencies causing hearing loss ie, they have low hearing sensitivity at certain frequencies. To compensate this type of hearing loss it is necessary to selectively amplify sounds at required frequencies. Figure 1 shows an example of an audiogram for normal hearing and hearing impairment by noise exposure.

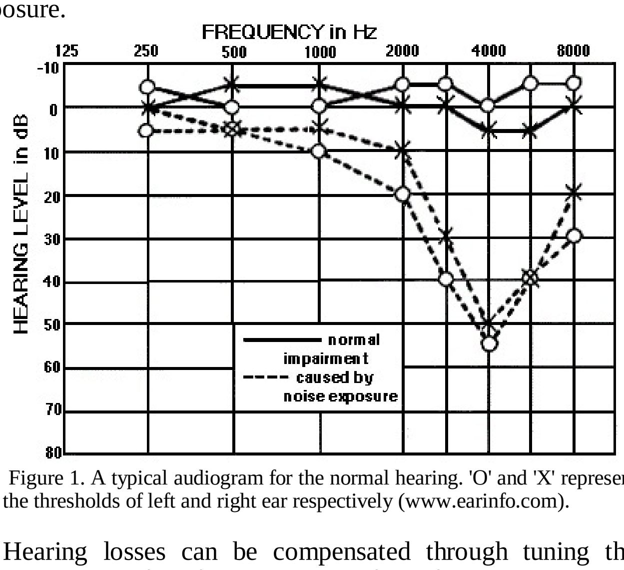

Figure 1. A typical audiogram for the normal hearing. 'O' and 'X' represent the thresholds of left and right ear respectively (www.earinfo.com).

Hearing losses can be compensated through tuning the subband gains of uniform or non-uniform filterbanks. Inorder to reduce the maximum matching error in hearing aid fitting, an increased number of frequency subbands are required because the actual hearing loss pattern (audiogram) differs from person to person [2]. So the filter-banks must have high tuning flexibility to fit various audiograms [3,4]. Based on the functionality the hearing aid can be classified into three categories: analog, programmable analog and digital hearing aid. Analog hearing aid is the least expensive type, but the drawback is that noise is also amplified without discriminating the sound. In programmable analog hearing aid a programmable control circuitry is added to the analog audio circuitry to program the gain and frequency settings. Digital hearing aid have much advantages over analog systems because of the advanced digital signal processing (DSP) algorithms contained is to compensate speech, improved intelligibility in noisy environment and echo or feedback cancellation. Researchers have investigated several techniques suitable for hearing aid applications. These techniques include uniform filter banks , non-uniform filter banks and fast fourier transform. In these techniques the frequencies of the audio signal are split into different bands and then amplification is provided according to the different levels of hearing loss [5].

As the number subbands (filters) increases better audiogram matching can be achieved. But this will increases the computational complexity in terms of the numbers of arithmetic operations (additions and multiplications) and in turn costs much power. Much effort was invested in the design of uniform digital filter banks for hearing aid applications [6,7]. Typical hearing loss such as sensorineural hearing loss (SNHL), conductive hearing loss occurs at high and low frequencies and hearing loss occurs at high frequencies caused by aging. Uniform filter banks may face difficulties in matching the audiogram in all frequencies. To achieve a better compensation, narrower bands need to be allocated at high and low frequencies. Therefore, a non-uniform spaced digital finite-impulse response (FIR) filter bank becomes very attractive [2].

In this paper a low complex non-uniform FIR digital filter bank using frequency response masking technique [2,9-11] is proposed, which provides better audiogram matching in both low and high frequencies. With help of FRM technique and half-band filters, drastic reduction in the number of multipliers and adders in linear phase FIR filters can be achieved.

The paper is organized as follows. Section II gives a brief overview about the FRM Technique. Section III gives the design method for the proposed filter bank. Section IV consist of experimental results and the result for the audiogram

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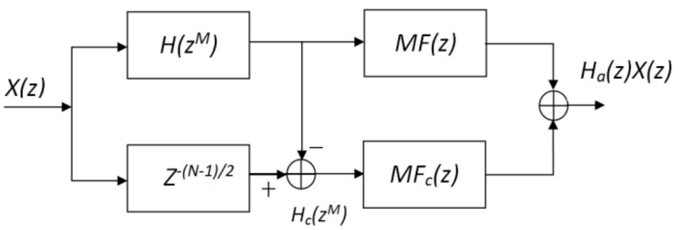
matching for different hearing loss. Section V consist of future work and Section VI concludes the paper.

II. OVERVIEW OF FRM TECHNIQUE

With FRM technique, linear phase FIR filters can be realized with guaranteed stability and phase response [12]. A realization structure for a filter using the FRM technique is illustrated in Figure 2, which comprises a band-edge shaping filter $H_a(z)$ and two masking filters $MF(z)$ and $MF_c(z)$ to synthesize narrow transition band linear phase digital FIR filters. $H_a(z^M)$ is obtained by replacing each delay element of a prototype filter, by M delay elements and therefore has very sharp transition bands and very low arithmetic complexity. Θ and $\varnothing$ are the pass band and stop band edges respectively and the transition width of $H(z^M)$ is $(\Theta - \varnothing) / M$ which is a factor of M narrower than the transition width of the prototype filter. The complementary filter of $H(z^M)$ denoted by $H_c(z^M)$ can be expressed as $(z^{-(N-1)/2}-H(z^M))$ where N the length of the impulse response of $H(z^M)$ Two masking filters are cascaded to $H(z^M)$ and $H_c(z^M)$ and added together to form the FRM filter with an overall transfer function [12]. Figure 3 shows the illustration of FRM approach. The output is given by the equation.

$$H_a(z) = H(z^M)\, MF(z) + H_c(z^M)\, MF_c(z) \qquad (1)$$



. Figure 2. Realization structure of FRM filter [12].

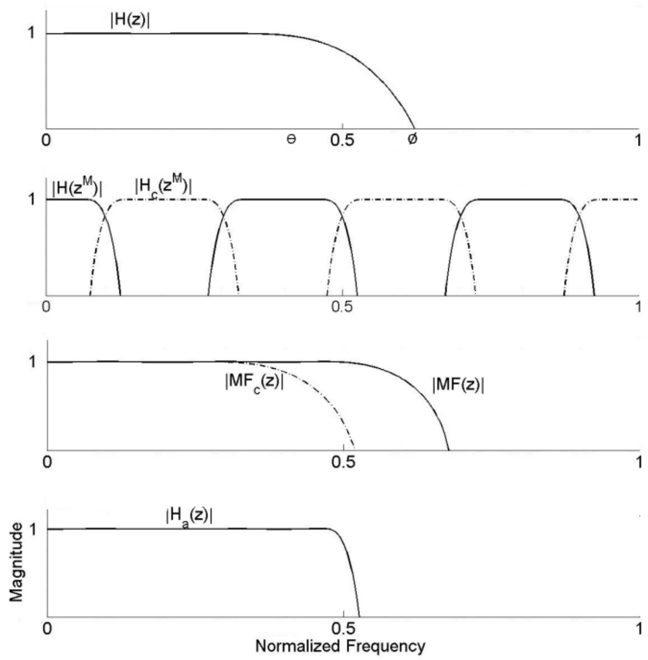


Figure 3. Illustration of FRM approach [12].

III. STRUCTURE OF PROPOSED FILTER BANK.

Linear phase digital filters have many advantages such as guaranteed stability, free of phase distortion, and low coefficient sensitivity under certain conditions. But the main disadvantage of linear phase FIR filter is its complexity due to the involvement of large amount of multipliers. In order to reduce the filter complexity, the proposed non-uniform FIR filter bank uses two simple half-band filters $H(z)$ and $MF(z)$ as prototypes and the subbands are designed with symmetry at the mid-frequency point. The resulting filter has very sparse coefficients. Since only a very small fraction of its coefficient values are non-zero, its complexity is very much lower than compared to other filters. More reduction in complexity can be achieved by using a masking filter with minimum order since masking filter need not require very high transitional bandwidth.

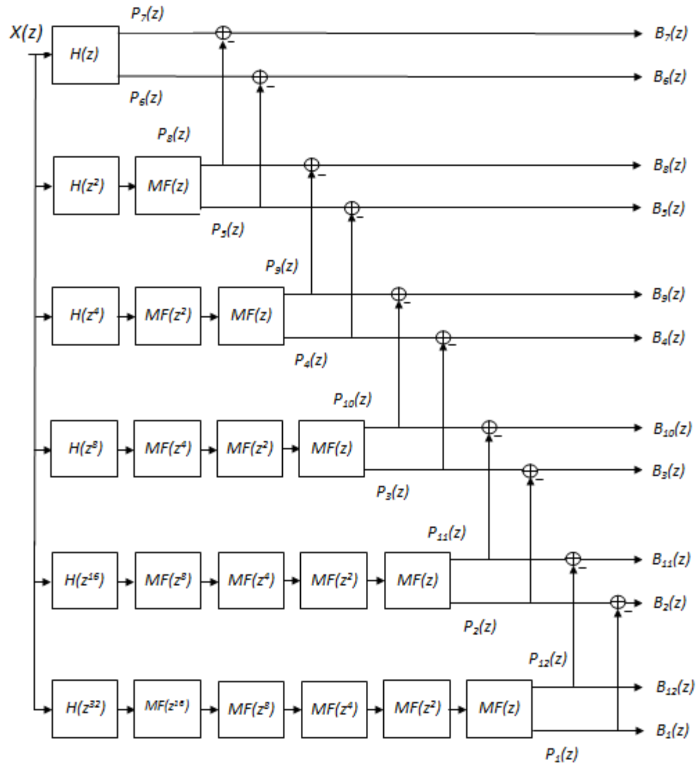


Figure 4. Proposed filter bank.

The proposed filter bank shown in Figure 4 is designed in such a way that lower and upper six subbands are symmetric. This is aimed at improving the matching performance at both low and high frequencies compared with the uniform filter bank. The right-most filters in the figure of each branch provide a pair of outputs, e.g., $P_6(z)$ and $P_7(z)$, $P_5(z)$ and $P_8(z)$ etc. At each branch, the top output comes from the original filter and the bottom one is from the complementary filter. This is formed with the help of a complementary filter pairs $MF(z)$ and $MF_c(z)$. $MF_c(z)$ is a complement of an original filter $MF(z)$, where $MF_c(z)$ can be implemented by subtracting the output of $MF(z)$ from the delayed version of the input as shown in Figure 6. $MF_c(z)$ is given by

$$MF_c(z) = z^{-(N-1)/2} - MF(z) \quad (2)$$

where N is the length of $MF(z)$. The extra delays for deriving $MF_c(z)$ from $MF(z)$ need not be implemented explicitly since the delays in $MF(z)$ can be used for this purpose as shown in Figure 6 and Figure 7. Thus the hardware cost for producing the complementary output is minimized.

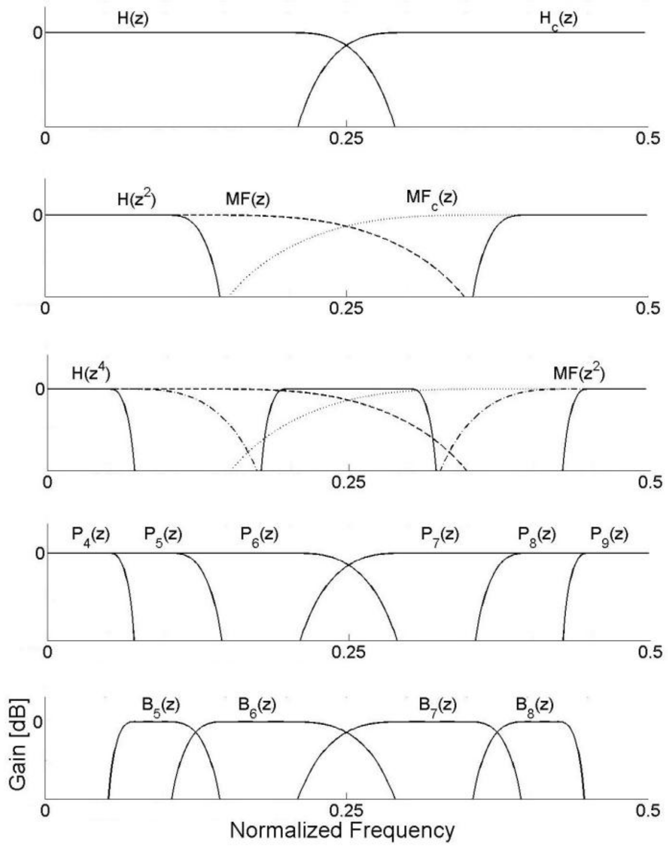


Figure 5. Magnitude response of 12 band non-uniform filter bank.

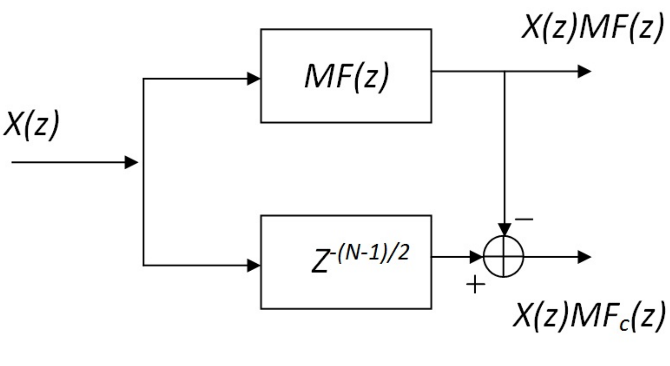


Figure 6. Complementary filter.

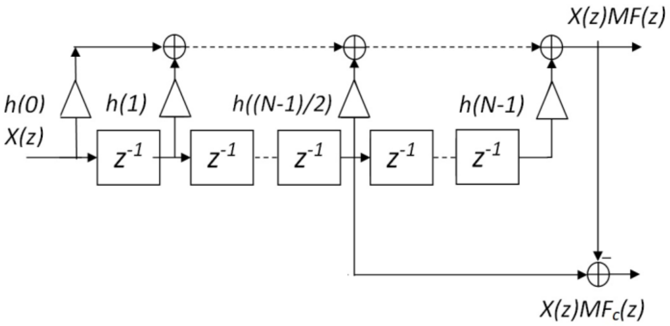


Figure 7. Architecture of complementary filter.

First step is to design two half-band filters $H(z)$ and $MF(z)$ as prototype filters. In half-band filter case, all odd-indexed coefficients except for the central coefficient are zero valued making the implementation very attractive. An important advantage of a linear-phase half-band filter is the efficient implementation, which follows from two favorable properties of the filter impulse response. The first is that the number of non-zero valued coefficients is nearly half of the filter length. The second is the non-zero coefficients exhibit symmetry property. Also by designing a masking filter $MF(z)$ with minimum order other than in [2], an additional multiplier less method can be achieved with much lesser complexity.

The outputs of each subband are termed as $B_i(z)$, i=1,2,..,12 where $B_1(z)$ to $B_6(z)$ are formed by outputs $P_1(z)$ to $P_6(z)$ from the original filters $MF(z)$ and $B_7(z)$ to $B_{12}(z)$ are based on the complementary outputs $P_7(z)$ to $P_{12}(z)$ from $MF_c(z)$. Leading delays should be added to each filter in branches other than the top one to ensure that all branches have the same phase shift in order to achieve the desired frequency response and avoid frequency-dependant delay [2]. The basic idea is that firstly prototype filter $H(z)$ is interpolate by different values ie, 2, 4, 8, 16 and 32 and interpolate masking filter $MF(z)$ by 2, 4, 8 and 16. Then cascading by different combination of filters each subband is formed. For example $P_5(z)$ is produced when $MF(z)$ is cascaded with $H(z^2)$. Similarly, $P_8(z)$ is a result of connecting $H(z^2)$ with $MF_c(z)$, the complement of $MF(z)$. The mid bands $B_6(z)$ and $B_7(z)$ are formed by subtracting $P_5(z)$ from $P_6(z)$ and $P_8(z)$ from $P_7(z)$ respectively as shown in Figure 5. The transfer function of the filter for lower six bands can be written as

$$B_i(z) = \begin{cases} P_i(z), & i = 1 \\ P_i(z) - P_{i-1}(z), & i = 2,3,4,5,6 \end{cases}$$

For the higher six bands, the transfer function is given by

$$B_i(z) = \begin{cases} P_i(z), & i = 12 \\ P_i(z) - P_{i+1}(z), & i = 7,8,9,10,11 \end{cases}$$

where $P_i(z)$, i=1,2,..,12 are shown in TABLE I.

TABLE I. TRANSFER FUNCTION OF EACH BANDS.

Band	Transfer function
1	$P_1 = H(z^{32})MF(z1^{6})MF(z^8)MF(z^4)MF(z^2)MF(z)$
2	$P_2 = H(z^{16})MF(z^8)MF(z^4)MF(z^2)MF(z)$
3	$P_3 = H(z^8)MF(z^4)MF(z^2)MF(z)$
4	$P_4 = H(z^4)MF(z^2)MF(z)$
5	$P_5 = H(z^2)MF(z)$
6	$P_6 = H(z)$
7	$P_7 = H_c(z)$
8	$P_8 = H(z^2)MF_c(z)$
9	$P_9 = H(z^4)MF(z^2)MF_c(z)$
10	$P_{10} = H(z^8)MF(z^4)MF(z^2)MF_c(z)$
11	$P_{11} = H(z^{16})MF(z^8)MF(z^4)MF(z^2)MF(z)$
12	$P_{12} = H(z^{32})MF(z1^{6})MF(z^8)MF(z^4)MF(z^2)MF_c(z)$

The implementation of a 12-band filter bank needs twenty one subfilters. However, the multipliers can be shared among interpolated $H(z)$ and $MF(z)$ same as that in Figure 7. The frequency response masking is achieved by repeatedly use of $MF(z)$.

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IV. EXPERIMENTAL RESULTS

The complexity of the proposed filter bank largely depends on the lengths of the two prototype half-band filters. The prototype filter *H(z)* determines the transition bandwidth of each subband. Here the masking filter *MF(z)* is designed with a length of 11 coefficients, since sharp transition bandwidth is not required for masking filter. As the transition bandwidth of *H(z)* is increased, there is reduction in filter length. The selected specifications of the prototype filter for a twelve band hearing aid is given below

Pass band frequency = 4 *kHz.*
Sampling frequency = 16 *kHz.*
Maximum pass band ripple = 0.0001.
Maximum stop band attenuation = 80 *dB.*

Further the matching between audiogram and magnitude response of the filter bank is closely related to transition bandwidth. In order to illustrate the effect of transition bandwidth non-uniform filter banks with normalized transition bandwidths ranging from 0.025 to 0.2 are simulated. Audiogram for presbycusis is selected as an objective curve for simulation. Presbycusis is a type of hearing loss, which is generally seen in people beyond the age of 60 year. The hearing sensitivity has progressively worsened over the years, and this will be reflected in the audiogram especially in the higher frequencies. The maximum matching error corresponding to the transition bandwidth and required filter length is given in the TABLE II.

TABLE II. Comparison of matching errors.

Normalized transition bandwidth	**Total length of filter *(H(z) +MF(z))***	**Maximum matching error *(dB)***
0.20	11+11=22	2.9
0.175	15+11=26	2.5
0.15	19+11=30	1.4
0.125	23+11=34	1.7
0.10	31+11=42	2.5
0.05	47+11=58	4.00
0.025	95+11=106	5.5

From the TABLE II it is observed that matching error decreases with increase of transition bandwidth upto 0.15. Further increase in transition bandwidth worsens the matching error due to the overlap among different bands, especially in the low and high frequencies where the subbands are narrow. Taking into considerations of maximum matching error and complexity, 0.15 is a reasonable transition bandwidth. The required length of filters *H(z)* and *MF(z)* together is 30 and can be realized with 10 multipliers only. Figure 8 shows the magnitude response of the proposed filter bank.

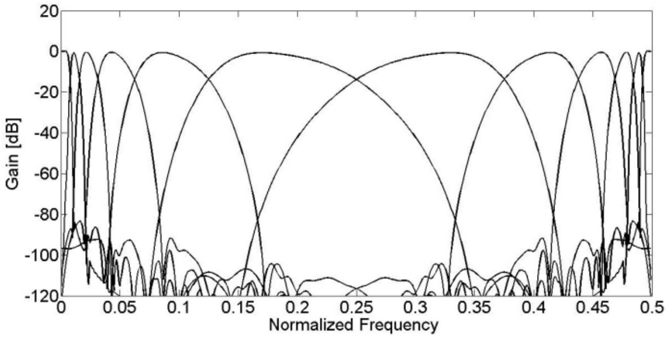

Figure 8. Magnitude response of the proposed filter bank.

A. *Audiogram Matching*

In order to verify the effectiveness of the proposed filter bank for the hearing aid application, audiogram matching with hearing loss pattern are simulated and verified. For presenting the results, the audiograms for the hearing loss types presbycusis, SNHL, bilateral conductive hearing loss and hearing loss due to middle menieres are considered [13,14]. Audiogram matching result for various hearing loss cases are shown in Figure 9 to Figure 12.

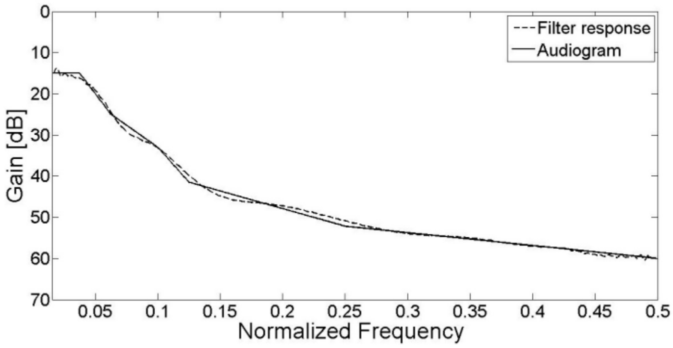

Figure 9. Audiogram matching for presbycusis.

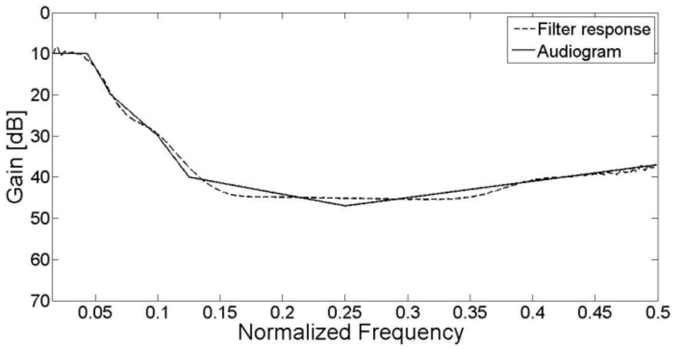

Figure 10. Audiogram matching for SNHL.

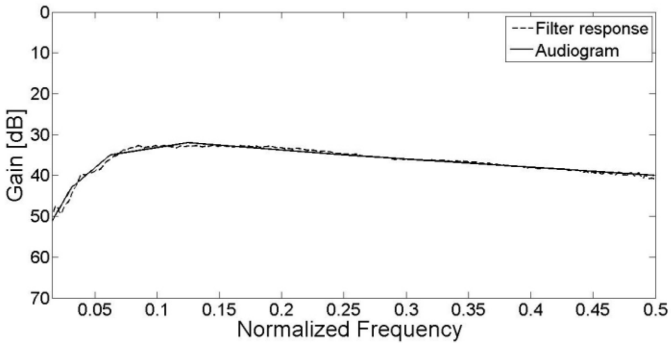

Figure 11. Audiogram matching for bilateral conductive loss.

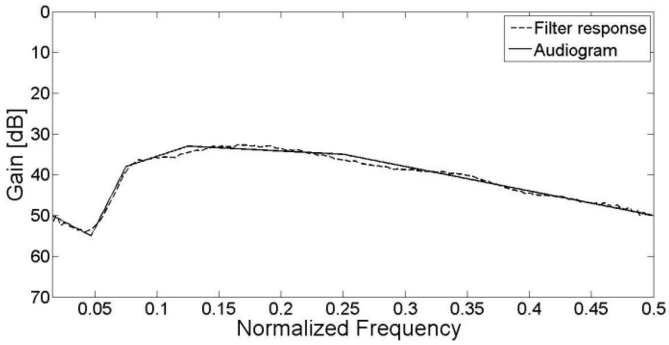

Figure 12. Audiogram matching for middle menieres.

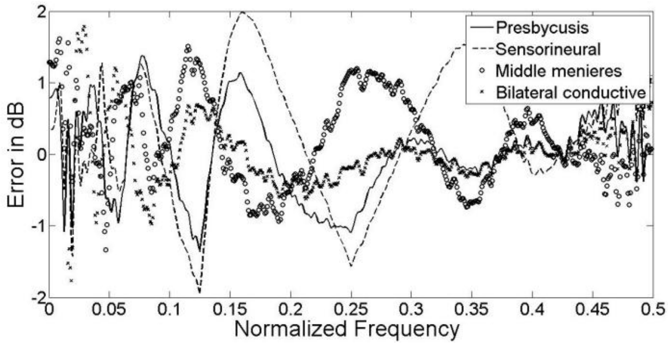

Figure 13. Comparison of matching error.

TABLE III. Maximum matching error for each hearing loss.

Hearing loss	**Maximum matching error (*dB*)**
Presbycusis	1.4
SNHL	2
Bilateral conductive	1.5
Middle menieres	1.75

Figure 13 shows the error plot of audiogram fitting for various hearing loss cases. TABLE III shows the comparison of maximum matching error corresponding to each hearing loss. From the TABLE III, it can be seen that, the proposed filter bank design is highly suitable for the audiogram with sharp transition at low and high frequency regions. If there is a sharp transition of hearing loss at mid- frequencies, matching error will be high. In this case maximum matching error can be decreased by introducing new bands at middle frequencies.

V. FUTURE WORK

If the number of bands are increased good matching of audiogram can be achieved. The proposed filter bank design is mainly applicable for the audiogram with sharp transition hearing loss at low and high frequency regions. But sharp transition of hearing loss occurs at middle frequency range, the proposed design may not be applicable. This type of hearing loss can be effectively implemented either by redesigning the filter bank structure for getting low bandwidth at mid-frequencies or additional bands can be introduced in the mid frequency range, then the maximum matching error can be reduced.

VI. CONCLUSION

A low complex twelve band non-uniform linear phase digital FIR filter bank has been developed based on the frequency masking technique for hearing aid applications. Two half-band FIR filter is taken as prototype filters. FRM technique is used for the design of filters. This reduces the number of multipliers to a minimum value with low audiogram matching error. Simulation results shows that, a twelve band filter bank can be implemented with 10 multipliers only and it gives a stop band attenuation of 80 *dB* and maximum matching error less than 1.4 *dB*. Various audiograms for the common types of hearing loss are used for evaluating the effectiveness of the filter bank. Audiogram matching shows that proposed method is good for sharp transition of hearing loss at low and high frequency regions. The matching error can be decreased by using more number of bands.

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