A 16-Band Nonuniform FIR Digital Filterbank for Hearing Aid

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***Abstract*—A computationally efficient 16-band non-uniformly spaced digital FIR filter bank is proposed for hearing aid applications. The filter bank is constructed by three prototype filters based on the principle of frequency-response masking technique. It requires only 34 multiplications in the implementation of a 16-band filter bank with stopband attenuation of -60dB. The simulation results show that the proposed filter bank achieves good matching between audiograms and filter magnitude responses. The filter delay is significantly reduced compared to that of the 8-band non-uniform filter bank [8].**

I. Introduction

Hearing thresholds are defined as the softest sounds one can hear and are represented by the audiograms. A typical pure tone audiogram measured on a person with normal hearing is shown in Figure 1 where all hearing thresholds are below the intensities of vowels and consonants. However, for people with impaired hearing, the hearing thresholds become high at certain frequencies causing hearing loss. A common type of hearing loss, presbycusis, is shown in Figure 2 where the hearing thresholds increase with frequencies. Such loss is caused by aging on the inner ear and related structure. To compensate this type of hearing loss, it is necessary to selectively amplify sounds at high frequencies. This can be done by means of a filter bank, $H(z)$, as illustrated in Figure 3, where the adjustable gain is used to control the magnitude response of filter to match the audiogram [1]. This approach works well especially when hearing loss is lower than 45dB. In such cases, audibility is most important. Amplification can compensate the loss effectively.

Much effort has been invested in the design of uniform digital filter banks for hearing aid applications [2]-[7]. However, hearing level measurements are done at each octave: 250 / 500 / 1k / 2k / 4k / 8k in a standard audiogram. This suggests that the uniform filter banks may face difficulties matching the audiogram at low frequencies unless a large number of bands are used. On the other hand, it is worth noting that one of the main causes of sensorineural hearing loss is aging, which is characterized by the high thresholds in high frequency in an audiogram, as shown in Figure 2. It is obvious that a good match to such an audiogram is achievable if more bands are allocated at high frequencies in a filter bank. Therefore a non-uniform spaced digital FIR filter bank becomes very attractive.

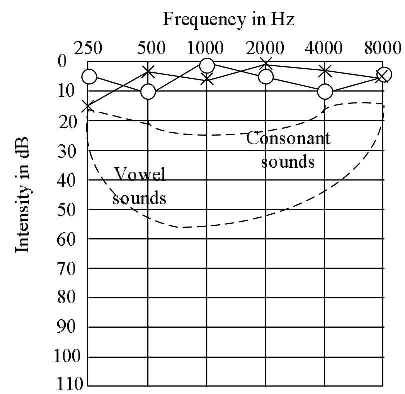

Figure 1. A typical audiogram for the normal hearing. O and X represent the thresholds of left and right ear respectively. The area in the dashed lines represents the intensities of normal voice.

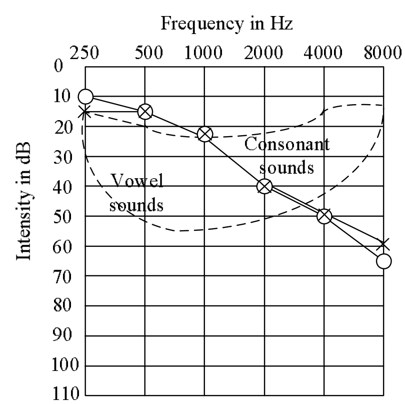

Figure 2. An audiogram for the hearing loss caused by aging.

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It is shown in [8] that a non-uniform filter bank outperforms its uniform counterpart in terms of matching errors. However, the bandwidths of two mid bands are close to 2 kHz under a 16 kHz sampling frequency in the 8-band filter bank, which makes it not very attractive for certain hearing loss cases. Another disadvantage of the 8-band filter bank is the relatively long filter delay which may affect lip-reading. In this paper, we proposed a novel filter bank structure to address the drawbacks of [8]. The proposed filter bank doubles the number of bands to 16 yet reduces the filter delay by more than 50%.

This paper is organized into four sections. In Section II, we presented the details of filter bank structure. The design of a non-uniform 16-band filter bank was given in Section III as well as the evaluation of the proposed filter bank. Conclusion was drawn in Section IV.

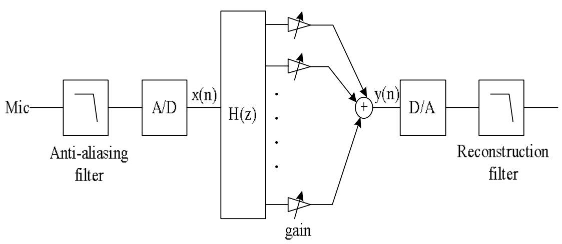

Figure 3. Block diagram of filterbank based hearing aid.

II. THE STRUCTURE OF PROPOSED FILTER BANK

The proposed filter bank is designed in such a way that lower and upper eight subbands are symmetric. This is aimed at improving the matching performance at both low and high frequencies compared with the uniform filter bank. The construction of proposed filter bank is based on the frequency-response masking technique [9] and its structure is presented in Figure 4. There are eight branches in the filter bank. Each branch consists of either one or several filters connected in series. $H(z)$, $G(z)$ and $F_m(z)$ are linear phase lowpass FIR filters acting as prototypes, which are interpolated by different factors to form the bases of all subbands. $F_m(z)$ plays additional role as masking filter to remove the unwanted passbands in interpolated prototypes. The right most filters $F_m(z)$ in each branch produce a pair of complementary outputs, i.e. a lowpass and a highpass. The highpass output is given by

$$F_{mc}(z) = z^{-(N_F-1)/2} - F_m(z) \tag{1}$$

where N_F is the filter length of F_m(z). The branch that contains only one filter, $G(z)$, has slightly different arrangement, e.g. its highpass output is generated by replacing z^{-1} with ($-z^{-1}$) in the original lowpass filter.

The hardware cost for generating the complementary outputs could be minimized by obtaining the required delays $z^{-(N_F-1)/2}$ from the original filter. Additionally, the multipliers can be shared among interpolated filters such as $G(z)$, $G(z^2)$ and $G(z^4)$. And the implementation of cascaded filters such as $F_m(z^2)F_m(z)$ can be effectively done by sharing the multipliers and adders.

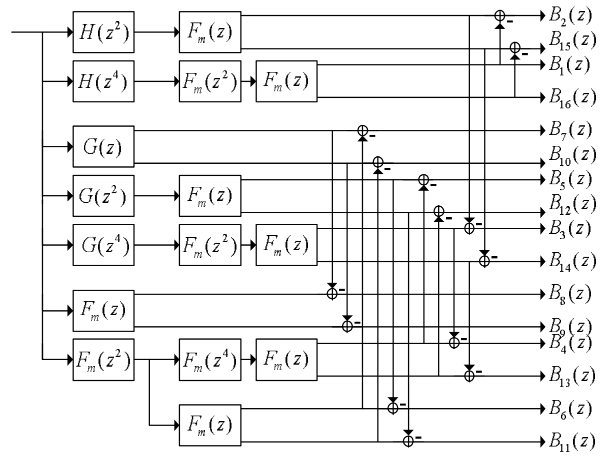

Figure 4 . The block diagram of the 16-band non-uniform filter bank.

TABLE I. THE *Z*-TRANSFORM TANSFER FUNCTIONS FOR EACH SUBBANDS

Band	Transfer Function
1	$B_1(z)=H(z^4)\ F_m(z^2)F_m(z)$
2	$B_2(z)=[H(z^2) - H(z^4)F_m(z^2)]F_m(z)$
3	$B_3(z)=[G(z^4)F_m(z^2) - H(z^2)]F_m(z)$
4	$B_4(z)=[F_m(z^4) - G(z^4)]F_m(z^2)F_m(z)$
5	$B_5(z)=[G(z^2) - F_m(z^4)\ F_m(z^2)\ ]F_m(z)$
6	$B_6(z)=[F_m(z^2)- G(z^2)]\ F_m(z)$
7	$B_7(z)= G(z) - F_m(z^2)F_m(z)$
8	$B_8(z)= F_m(z)- G(z)$
9	$B_9(z)= F_{mc}(z) – G_h(z)$
10	$B_{10}(z)= G_h(z) - F_m(z^2)F_{mc}(z)$
11	$B_{11}(z)= [F_m(z^2)- G(z^2)]\ F_{mc}(z)$
12	$B_{12}(z)= [G(z^2) - F_m(z^4)\ F_m(z^2)\ ]F_{mc}(z)$
13	$B_{13}(z)= [F_m(z^4) - G(z^4)]F_m(z^2)F_{mc}(z)$
14	$B_{14}(z)= [G(z^4)F_m(z^2) - H(z^2)]F_{mc}(z)$
15	$B_{15}(z)= [H(z^2) - H(z^4)F_m(z^2)]F_{mc}(z)$
16	$B_{16}(z)= H(z^4)\ F_m(z^2)F_{mc}(z)$

The *z*-transform transfer functions for each subband are given in Table I. The working principle of proposed filter bank can be best illustrated by an example. Suppose we want to form the third subband $B_3(z)$, four filters, $G(z^4)$, $H(z^2)$, $F_m(z^2)$ and $F_m(z)$, are involved. Figure 5 shows the magnitude responses of various subfilters. The interpolated prototype $G(z^4)$ produces an output with three passbands, as shown by the solid line in Figure 5 (a). In order to form the required lowpass filter as shown in Figure 5(b), the unwanted passbands at mid and high frequencies are removed by masking filters $F_m(z^2)$ and $F_m(z)$. Similarly, $H(z^2)$ is filtered by $F_m(z)$ to produce another lowpass filter. The subband $B_3(z)$ is generated by subtracting the output in Figure 5 (d) from that in Figure 5(b). Note that leading delays should be added to subfilters to make sure that they have the same group delay when connected in parallel. This is necessary to avoid frequency dependant delay [10].

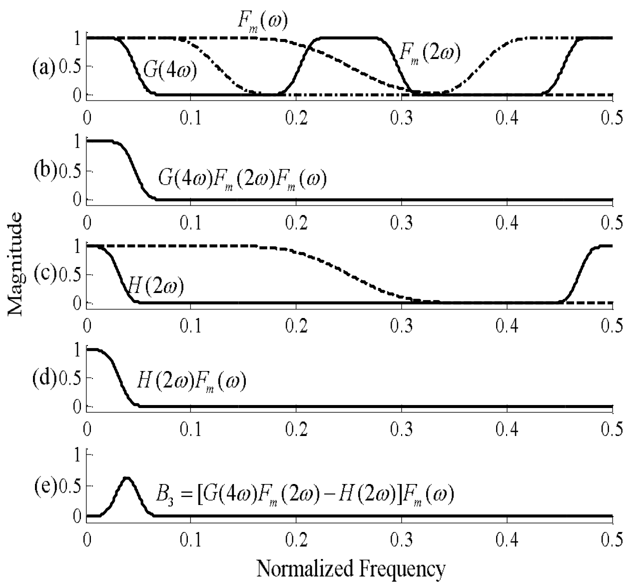

Figure 5. The formation of subbands $B_3(z)$.

III. The Design of Proposed Filter Bank and Performance Evaluation

A. The Design of Proposed Filter Bank

Let us consider the design of a 16-band non-uniform filter bank covering frequencies up to 8 kHz at sampling frequency of 16 kHz. The 3-dB cutoff frequencies of each band are listed in Table II. The design starts with the selection of bandedges of three prototype filters, $H(z)$, $G(z)$ and $F_m(z)$ as they determine the complexity of overall filter bank and affect the matching errors. Suppose that the transition bandwidth of $G(z)$ is ΔB , the transition bandwidths of $H(z)$ and $F_m(z)$ are set to be $\Delta B/2$ and ΔB, respectively. To find out the influence of the transition bandwidth, several non-uniform filter banks with normalized transition bandwidths ranging from 0.05 to 0.20 were designed. The hearing loss caused by aging shown in Figure 2 is selected as the objective curve to measure the matching errors. Matching errors and filter complexities are listed in Table III. It can be seen that as the transition bandwidth increases, the matching curves become smoother which leads to smaller matching errors. ΔB of 0.20 is selected for the prototype filters. Figure 6 shows the frequency response of the filter bank.

B. Performance Evaluation

For comparison with a 16-band uniform filter bank, we select a computationally efficient approach proposed in [5]. Suppose the transition bandwidths are the same across all subbands and the normalized transition bandwidth equals to that of the smallest subband in the non-uniform one. Four linear phase FIR filters with lengths 131, 67, 35 and 19, respectively, are used to produce the 16 subbands with attenuation of -60dB. Audiograms with hearing loss in low frequency are most suitable for comparing the performance of uniform with non-uniform filter banks. Figure 7 shows matching error of uniform and non-uniform filter bank for mild to moderate hearing loss in low frequencies. It is clear that the proposed filter bank outperforms the uniform one in low frequency range.

Table II Bandedges of Subbands

Band	Lower 3dB frequency	Upper 3dB frequency
1	-------	250
2	250	500
3	500	750
4	750	1000
5	1000	1500
6	1500	2000
7	2000	3000
8	3000	4000
9	4000	5000
10	5000	6000
11	6000	6500
12	6500	7000
13	7000	7250
14	7250	7500
15	7500	7750
16	7750	-------

Table III Impacts of The Transition Bandwidths

Transition Bandwidth (normalized)	Number of multiplications ($G(z)$+ $H(z)$+$F_m(z)$) (attenuation 60dB)	Maximum Matching Error (dB)
0.05	38+75+19=132	2.1044
0.10	19+38+10=67	0.5587
0.15	13+25+7=45	0.3961
0.18	11+21+6=39	0.3974
0.20	10+19+5=34	0.3837

An important feature of the proposed filter bank is the low delay compared with the one in [8]. The total delay of proposed filter bank is 12.8ms under the sampling frequency of 16 kHz. The one in [8] has a delay of 26.6ms under the same sampling frequency. The reduction in delay is significant as it improves the quality of hearing aid. It was reported in [11] that delays longer than 20ms may cause interference between speech and visual integration.

IV. Conclusion

A 16-band non-uniformly spaced digital FIR filter bank has been proposed. The use of half-band FIR filters as prototype filters and the combination of frequency-response masking technique lead to significant savings in terms of number of multiplications. The overall filter delay is significantly reduced as the result of novel filter structure which reduces the interpolation factor for the prototype filters. The performance is enhanced, especially in low and high frequencies, due to the increase of number of filter bands.

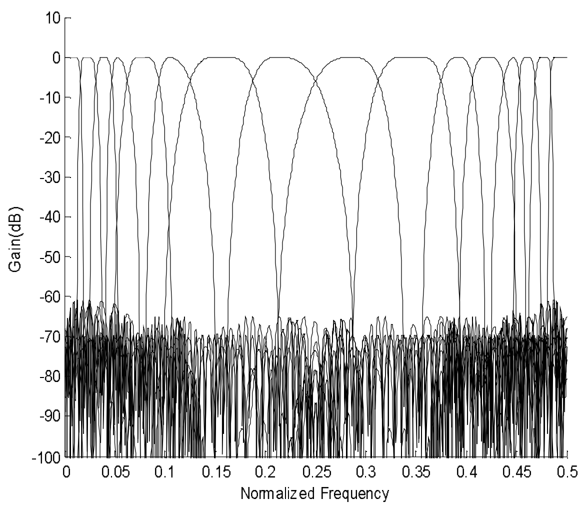

Figure 6. The frequency responses of the proposed filter bnak.

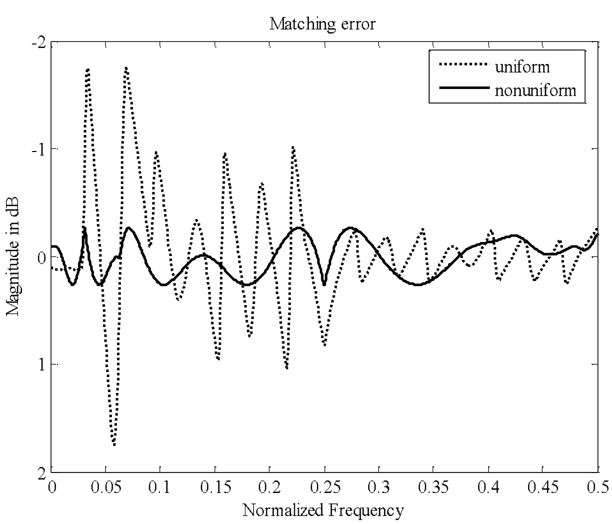

Figure 7. Matching errors of uniform and nonuniform filter banks for mild to moderate hearing loss in low frequencies.

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