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

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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

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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 $\leq$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]

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

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 $\tilde{H}_k(z)$,

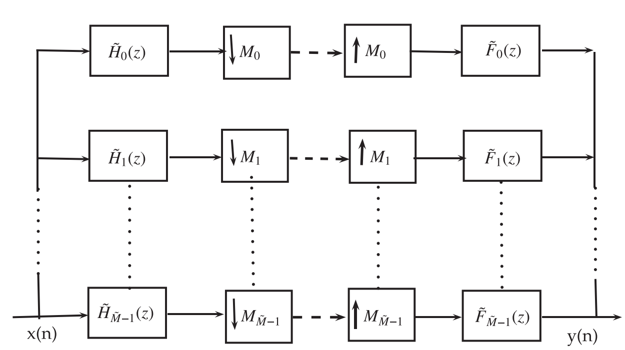

Fig. 1. Cosine modulated non-uniform filter bank.