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Design and implementation of reconfigurable filter bank structure for low complexity hearing aids using 2-level sound wave decomposition

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ABSTRACT

This paper proposes a reconfigurable digital filter bank structure, which is suitable for designing hearing aids for most types of hearing losses. The proposed structure exploits the fractional interpolation and symmetry property of linear phase filters. The structure has two stages; the first one is called masking stage and the second one is called multiple passbands generation stage. The second stage i.e., multiple passband generation stage has 2 levels. By adjusting a 7-bit control signal, different sub-bands generated by the two stages can be obtained for audiogram matching. The number of sub-bands can be increased by increasing the number of fractional interpolated filters in level 2 of the multiple passbands generation block. Using the proposed structure, various types of audiograms can be matched with acceptable delay and matching error. The merits of the proposed structure are low hardware complexity and good audiogram matching with tolerable matching error and acceptable delay, when compared to the state of the art techniques for audiogram matching. Moreover, it is a reconfigurable structure. FPGA implementation of the proposed structure is also done to supplement the theoretical claim for low hardware complexity and power.

1. Introduction

Hearing is a very important and sensitive feature of human beings. There are various factors which can affect the hearing ability of a person such as presbycusis (aging), exposure to loud noise and in some cases, hereditary. This hearing loss experienced by hearing impaired people can be compensated by the use of a hearing aid. The block diagram of a commonly used digital hearing aid is shown in Fig. 1. An analog to digital converter shown as A/D, converts the analog output of the microphone into digital form. This digitized signal is fed to a bank of filters. This filter bank splits the input signal frequency spectrum into different sub-bands with different frequency ranges. This is followed by gain blocks that will be used to adjust the amplitude of each sub-band, where adjustment of gains can be done in linear or nonlinear approach. After amplification, the sub-bands are combined and then converted back to analog signal by a digital to analog converter shown as D/A in Fig. 1. There are many practical aspects of hearing aids in which one aspect is amplification in multiple frequency bands, which is mainly done in the filter bank block. Other practical aspects of hearing aids are signal pre-processing (noise reduction, beam forming and auditory scene analysis) and acoustic components, which come outside these blocks. In this paper, emphasis is given on the design of a reconfigurable low complexity digital filter bank block, which is an essential part of a hearing aid. Since the emphasis of this paper is on the aspect of amplification in multiple frequency bands, the other aspects mentioned above are not considered. Digital filters are less sensitive to noise and they can be designed to have linear phase under certain conditions [1,2]. Hence, digital filters are used in this work.

The hearing capability of a person at different frequencies can be represented with the help of a graph called audiogram. Fig. 2 shows a typical audiogram that is obtained from an audiometer. The hearing level at different frequencies are shown in dB. From left to right of an audiogram, frequencies vary from low to high values and from bottom to top of an audiogram, intensity decreases, which represent the hearing capability of a person and appears as hearing loss (HL) in the audiogram. The value of the intensity categorizes hearing losses into normal (0 dB to 20 dB), mild (20 dB to 40 dB), moderate (40 dB to 70 dB), severe (70 dB to 90 dB) and profound ($\geqslant$90 dB) [3]. In an audiogram, intensity measures are done at fre-

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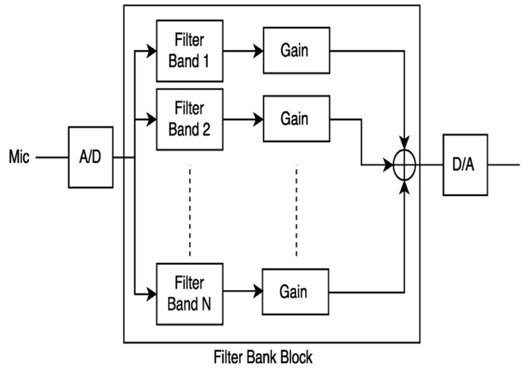

Fig. 1. Block diagram of digital hearing aid.

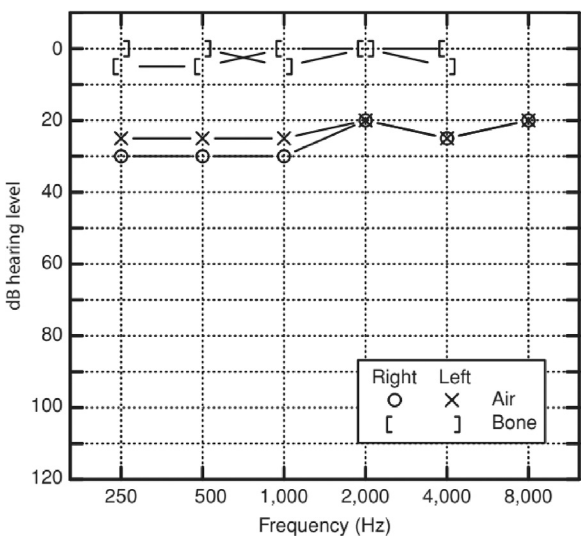

Fig. 2. Bilateral conductive hearing loss [4].

quencies 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz and 8000 Hz. Fig. 2 shows an audiogram of a person suffering from bilateral conductive hearing loss [4]. In Fig. 2 the hearing ability of left and right ears is measured with the help of the curves marked by the symbols X and O, respectively. The audiogram in Fig. 2 is bilateral conductive hearing loss, since the bone conductive threshold is in normal range and air conductive threshold of both the ears is not in normal range. The portions above the curve represent the sounds which a patient cannot hear. In all the frequency regions shown in Fig. 2, the patient suffers hearing loss for both the ears, hence the audiogram is bilateral hearing loss. Otherwise it is referred to as unilateral hearing loss. In digital hearing aid, the gain of each sub-band shown in Fig. 1 is the optimized gain, which is adjusted by trial and error method, such that the difference between the gain and hearing loss, which is called matching error, is minimum. The maximum of the overall response gives an approximation of the audiogram. Maximum acceptable matching error value is ±3 **dB** [5,6].

There are many articles in the literature which discuss about the generation of reconfigurable sub-bands using digital filter banks [7–15]. In [7], a variable filter, based on sampling rate conversion method, is used to generate the sub-bands. However, this structure is seen to have very high hardware complexity when compared to the structures used in [8,9]. A digital hearing aid which uses Farrow structure to generate non-uniform sub-bands is discussed in [8,9]. Here, a significant reduction in hardware complexity is reported to have achieved, compared to that in [7].

In [10,11], the method of frequency response masking (FRM) [9] is used for non-uniform sub-band generation. However, the matching error in both the cases is found to be higher than ±3 dB [6]. Moreover, the delay using the structure in [10] is greater than the usually acceptable value of 20 ms [17]. Here, the sub-bands generated are fixed. The problem of high matching error in [10] and [11] is overcome in [12], where a cosine modulated filter bank is used to generate the non uniform sub-bands.

For many reconfigurable filter structures for audiogram matching, the number of filters is found to be equal to the number of uniform or non-uniform bands used to cover the entire spectrum of frequencies, which is from 125 Hz to 8000 Hz. However, the filter bank structure suggested in [13], shows acceptable matching error, with low hardware complexity. This structure is realized using only three prototype filters. However this structure is seen to have delays higher than 20 ms [17]. This problem is solved in [14] by using a structure with only two prototype filters. This leads to a reduction in the complexity. However, it can be seen that these structures are able to generate only limited number of sub-bands and only limited number of audiograms can be matched effectively. More number of audiograms can be matched by using the structure proposed by as in [18]. But the disadvantage of the structure in [18] is that the device utilization of this structure is more and delay for some cases exceeds 20 ms [17].

Most of the designs of the filter bank structure for hearing aids discussed in [7–15,18,25] follow selective amplification in multiple frequency. So, for comparison purpose, in this paper the same method of selective amplification is done. Modern hearing aids do not simply invert the hearing loss function to provide linear gains across the audible frequency range. Fitting gains in different frequency bands involves more sophisticated formulas [3]. So here, it is to be noted that that recruitment-phenomenon [21] is not considered. But in real application, this phenomenon should be considered. NAL-NL2, CAM2 and DSLv5, which are gain formulas should also be used. Since linear filter processing cannot be used for patients with sensorineural deafness, such audiograms are not considered here.

In this paper, a reconfigurable structure is used which increases the number of sub-bands by using fractional interpolation filters, which are included as additional levels in the structure given in [14]. The proposed structure has two stages for the generation of sub-bands. The first stage called masking stage splits the entire frequency range into three sections and the second stage called the multiple passband generation stage has two levels for the generation of more number of sub-bands. Selection of these sub-bands and levels is done by some control signals. The matching error, delay and hardware complexity obtained using the proposed structure are lower when compared to other existing structures for audiogram matching.

The organization of the paper is as follows. The theory required to understand the proposed filter bank structure is given in Section 2. Section 3 describes the proposed structure. In Section 4, the methods of selecting optimum transition width and gain for the proposed structure are detailed. Section 5 analyzes the results. This Section also demonstrates the advantages of the proposed structure. Section 6 gives the conclusion of the paper.

2. Basic idea for the design of the proposed structure

Fractional interpolation given in [14,19,20], is the basic idea used to create sub-bands for the proposed filter bank structure. Let the frequency responses of the prototype filter and its fractional interpolated version be represented as $H(z)$ and $H(z^{M/D})$ respectively. The fractional interpolation changes the sampling rate by a factor of M/D. Here, both M and D are integers, where M is the interpolation factor and D is the decimation factor. For $H(z^{M/D})$, every Dth coefficient of the filter $H(z)$ is removed and then $(M-1)$