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ANALYSIS OF MULTIBAND FIR FILTERBANK FOR SPEECH PROCESSING USING INTERPOLATION AND DECIMATION

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Abstract: All the current hearing aid devices use fix filter banks and so cannot be used for different hearing impaired people. So, a new method consisting of multiband FIR filter banks is proposed .The use of interpolation and decimation which is the added benefit of using FIR filter helps us to reduce the computational complexity .This technique helps to customize the hearing aid devices used by hearing impaired people. An extract of multiband FIR filter bank techniques is presented.
Index Terms: Decimation, frequency-response masking, interpolation.

I. INTRODUCTION

Human hearing loss is one of the most common deficiency in the world. Hearing assistive devices using digital filters and amplifiers are used to overcome this problem.[1],[2].Hearing aids have become very successful at maintaining some hearing abilities for hearing impaired people. Even though, solutions are available, patients still have problem in understanding the speech in presence of noise. According to one survey, one-eighth of population in the most developed countries suffers from considerable hearing loss that strongly affects their speech communication skills and which restricts them from leading a normal life. These hearing impaired patients have an option of using modern hearing assistive devices, but many patients avoid using so, because of irritating and unpleasant sound noise they encounter in their everyday life. Hearing-aid device mainly amplifies the selected sound and transfers this amplified signal to the ear[3].Even if the objective of these assistive devices may be to simply make sound intelligible, it has nonlinear gain . Digital hearing aid is now a widely used technique in most modern devices as it assist in easy fitting of the hearing aid characteristics to each patient. Hearing aid manufacturers have tried to minimize background sounds with directional microphones, adaptable digital noise reduction, and use of binaural hearing aids to improve localization .Many of these techniques are used in modern digital hearing aids, but, unfortunately, only some patients are satisfied with the performance of their hearing aids in noisy situations. One of the reason for this unsatisfaction may be due to the fact that its algorithm may not decide what information is important and which noise is to be abandoned. Research has been done for the design of digital filterbanks for the selective amplification. Almost all the hearing aid devices use filterbanks with fixed sub-bands. Fixed sub-bands are not enough flexible for sound decomposition according to the characteristics of different types of hearing loss. We propose filter bank with adjustable sub-bands in FIR filter bank , All the sub-bands were first generated and later some of them were selected according to the requirement and rejecting unwanted subbands. So we propose flexible filter banks , which provides multiple band decomposition with less complexity and flexible by using interpolation and decimation techniques.

II. LITERATURE REVIEW

A significant amount of literature was evaluated and good balance of these topics was a primary goal when analyzing the literature. In order to control the gain automatically, first the input sound level is detected and then the gain is changed accordingly. Hearing aid basically divide the signal into minimum two frequency bands, detects the input level, and accordingly applies the gain. But, this adjustment is not easy go thing. Because a single sample never represent the intensity of the signal accurately, the level must be detected over some time interval .Moreover, the gain is normally adjusted in order to change slowly over time and, thus decrease the distortion. Recently, the design issues of the biomedical devices are changed to how to accommodate the individual differences for the performance optimization according to the each individual user and comfortable level enhancement [3] .That is, it is required that the initial design process should consider the human factors. After the chip fabrication of digital hearing aid (DHA) chip, the human factors are normally incorporated by the external gain fitting and verification process through the hearing loss test. However the post gain fitting process has many intrinsic problems. The ANSI S1.11 1/3-octave filter bank is useful for digital hearing aids, but due to its large group delay and high computational complexity, complication arises[4]. The standard ANSI S1.11 1/3-octave bank is rarely adopted in hearing aids because of its high computation complexity and rather large group delay, even though it has the advantage of good match to human hearing characteristics . T. Schneider and R. Brennan[6] discovered a method of mapping the wide dynamic range of speech signals into the reduced dynamic range of hearing impaired listeners. It presented a multichannel compression scheme that employs an oversampled, polyphase DFT filterbank. Instead ,this technique helped in band decomposition, a current research concentrates on reducing the complexity of the algorithms besides the accomplishment of the decomposition. Y. Lian and Y.Wei [11] proposed that the filter bank is constructed by three prototype filters based on the principle of frequency-response masking technique. . In that they have used eight nonuniform spaced subbands with the help of frequency-response masking technique. And two half-band finite-impulse response (FIR) filters are employed as prototypes resulting in significant improvements in the computational efficiency . Y.-T. Kuo, T.-J.Lin, Y.-T. Li [12] proposed that ANSI S1.11 1/3-octave filter bank is used in acoustic applications, mainly in acoustic analyzers and equalizers. It is also desirable in hearing aids because of prescription formula of hearing aid. However, the high computation complexity limits its usage.. To tackle this, implementation of ANSI S1.11 filter bank for digital hearing aids is proposed. Digital signal processing in modern hearing aids is typically performed in a subband or transform domain that introduces analysis–synthesis delays in the forward path. Long forward-path delays are not acceptable because the processed sound combines with the unprocessed sound which interrupts the quality. Nevertheless, subband domain transformations for digital hearing aids is the most popular choice for hearing aids because of the associated computational simplicity. T. B. Deng [13] designed a three-channel (3-channel) variable filter-bank (VFB) that consists of variable lowpass, variable bandpass and variable highpass digital filters. The three variable digital filters are obtained from a normalised analog prototype Chebyshev type-I lowpass filter using analog frequency transformations along with a modified bilinear transformation. This three variable digital filters are adjustable independently, the 3-channel VFB is flexible, and can applied for solving various hearing loss digital hearing aids. N. Ito and T.-L. Deng [14] presented a variable filter-bank (VFB) for low-power digital hearing aids. This VFB uses digital spectral transformation. After succesfull transformations from a digital prototype lowpass filter into variable lowpass filter (LPF), variable bandpass filter (BPF), and variable highpass filter (HPF), it illustrated how to optimize the parameters of the VFB for optimally fitting a given audiogram .Y.Wei and D. Liu[15] designed a filter bank which generated 27 different subband techniques .It enables different patients to take advantage of their specific case to improve the auditive performances. The use of frequency response masking technique enables us to realize the whole system with 3 different prototype filters, which minimizes the complexity to a large extent. In [16], adjustable filterbank technique was proposed. All the subbands were first generated and then some of them were selected while some of them were abandoned. In this, FIR filter is used , in order to provide multiband decomposition and small computational complexity with the help of interpolation, decimation and frequency masking.

III. PROPOSED WORK

Speech processing has attracted significant research efforts in the last two decades. Speech/non-speech classification (removal of noise) is an unsolved problem in speech processing and affects diverse applications. So we design a customized filterbanks with adjustable sub bands for an individual. Flexible filters , iteratively alter their parameters in order to minimize a noise encountered by the hearing aid devices. To make the filterbank flexible, interpolation and decimation techniques are used and the complexity problem is solved using frequency masking technique[16].

The flexible filter banks consists of 2 blocks which is shown in figure 1:

- Multiband- Generation Block: This block generates multiple bands of the sound or speech given as input using interpolation and decimation.
- Sub band –Selection Block: The function of this block is to select the corresponding bands as per the requirement and make the unwanted bands abandoned using interpolation.

A. Design of a Multiband Generation Block:

The Multiband generation block is shown above in fig 1.It uses interpolation (up sampling) and decimation (down sampling) method to make the multiband generation block reconfigurable .From fig 1 , we can see that the prototype filter H(z) is first decimated by factor say Mi(decimation factor) and then interpolated by some factors say D1 and D2(interpolation factors). Thus, after decimation and interpolation we get prototype filter as H(z^ Mi /D1) and H(z^ Mi /D2) which in turn produces frequency responses having same number of pass band and different passband widths.

Frequency response shown in fig1 are given as;

$$Ai(z) = H(z^{Mi/D1}) \quad (1)$$

$$Bi(z) = H(z^{Mi/D2}) - H(z^{Mi/D1}) \quad (2)$$

$$Ci(z) = z^{-\Delta} - [H(z^{Mi/D2})] \quad (3)$$

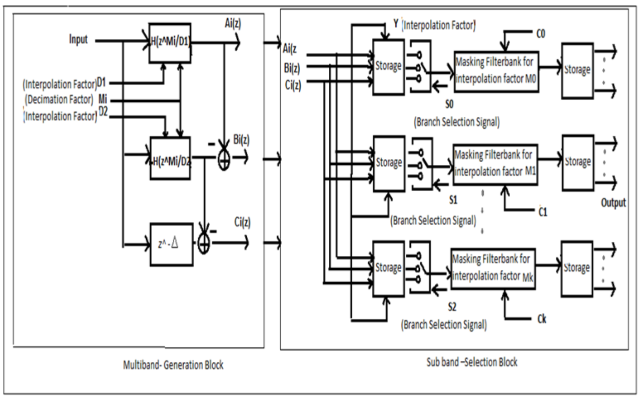

Figure 1: Block Diagram of Multiband Filter bank

B. Design of Subband-Selection Block:

The structure of the subband-selection block is presented in Fig. 1.Set of interpolation factors is represented by Y. The branches from multiband generation block is stored in one of the storage units according to the usage of interpolation factors. With the help of the branch selection signal Si, one of the branch signals is chosen as input to the masking filterbank. Masking filterbank comprises of several subfilters and switches. The status of this switches in the filterbank is determined by a control signal Ci .Therefore, these various Ci's results in different transfer functions of the masked filterbank. Thus, we can adjust the subband extraction by setting the value of Ci and Si.

In short, we can put forth a procedure from above discussion:

- Determine the control parameters such as decimation factor and interpolation factors.
- Find the prototype filter and get the frequency responses Ai(z), Bi(z), and Ci(z) as shown in figure1. Prototype filter should be band limited.
- Design the masking filterbank for each interpolation factor. Here, the prototype filter depends on the output of the multiband generation block.
- Thus, using above mentioned procedure, we can select appropriate sub-band of the output of Sub-band Selection Block with the respect to SNR value.

IV. CONCLUSION

One of the application of speech processing is hearing aid devices which needs to be reconfigured as current available aid devices uses fixed filter banks and practically not suitable as per different hearing loss case. Thus in order to overcome above mentioned drawbacks such as larger group delay, fixed filerbanks, computational complexity, we design a customized filterbanks with adjustable sub bands for an individual hearing-loss case. Flexible filters iteratively alter their parameters in order to minimize a noise encountered by the hearing aid devices .In order to make the flexible filterbank, up sampling and down sampling techniques are used and the complexity problem is solved using frequency masking technique[16].

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