An Improved Fractional Fourier Transform Based Reconfigurable Filter Bank for Hearing Aid

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***Abstract*---The problem of reducing noise in hearing aids is one of great importance and great difficulty. The problem has been addressed in many different ways over the years. The techniques used range from relatively simple forms of filtering to advanced signal processing methods. Filter banks for digital hearing aids must use significantly different criteria than those designed for coding applications. For digital hearing aids, the filter bank channel gains must be adjustable over a large dynamic range to compensate for the hearingloss. This paper presents a filter bank designed exclusively for hearing aid applications. Here we propose a fractional Fourier transform filter for reduction of noise in hearing aids. Our method enables hearing-impaired people to customize hearing aids based on their own specific conditions to improve their hearing ability. An experimental result is evaluated and shows the proposed method has high performance when compared with existing methods. 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 other filter bank methods.**

Keywords*:* Decimation, frequency-response masking, interpolation, reconfigurable filter bank.

I. INTRODUCTION

The most common sensory disturbances in the worldare hearing impairment. An effective treatment for the problem is hearing assistive devices. The main function of a hearing-aid device is to amplify the sound selectively, and then transfer the processed signal to the ear. Much study has been invested into the design of digital filter banks for selective amplification. Most of the current designs use filter banks with fixed sub bands. One approach is to use uniform filter-banks. While the previous research aims at Realizing the band decomposition, recent research focuses on reducing the Complexity of the algorithms besides the accomplishment of the decomposition. Lattice wave digital filter banks (LWDFBs) were used to process the sound waves. LWDFBs have lower complexity than finite-impulse response (FIR) filter banks and are not sensitive to the coefficients. In, a DFT filter bank with the multidimensional logarithmic number system was realized to reduce the complexity. In, some well-known simple methods for critically sampled filter banks were extended to the over sampled case. The simplicity comes from its flexibility to generate multiple prototype filters by one method. In, a joint stereo filter bank structure was proposed to satisfy the requirement of both the audio coding and the hearing-aid application, which reduces the complexity of the system. Dividing the frequency range uniformly is straightforward yet it does not consider the non-uniform scaling of human auditory filtering.

Another approach is to use no uniform filter banks that are widely used in audio coding and audio enhancement and have increasingly gained the attention of hearing-aid researchers because of the ability to mimic the resolution characteristic of human hearing. Some types of nonuniformfilter banks have been proved to be suitable for hearing-aid systems. A tree-structured filter bank based on all-pass compensatory filters and elliptic minimal Q-factor filters was used as the analysis filter bank. An eight band frequency-response masking filter bank was proposed for hearing aids. Both the designs lower the complexity at the cost of delay. A critical band-like spaced filter bank was used. The irregularity of the sub bands increases the difficulty of the implementation. A 1/3 octave filter-bank was realized. The octave filter bank was based on an IIR structure and could not provide linear phase property. These algorithms better match the characteristics of human auditory filtering compared to algorithms based on uniform filter banks. However, the complexity of nonuniformfilter banks is generally higher than that of uniform filter banks.

All the filter banks mentioned earlier have fixed sub bands, thus they cannot provide flexible sound decomposition plans ac-cording to the characteristics of different types of hearing loss. It is attractive to design filter banks with adjustable sub bands that can be customized for an individual hearing-loss case. Little work has been done in this area. In, a programmable spectrum cut-up permitted filters' bands to be adjusted to a patient's pathology. However, the realization of the sub bands was not discussed. Recently, a three-channel variable filter bank was proposed. The variable filter bank was based on IIR filters; thus, it cannot provide linear phase property. In, a filter bank with adjustable sub bands was proposed. All the sub-bands were first generated and then some of them were selected while some of them were abandoned. This design can achieve satisfactory performance yet the complexity is comparatively high.

In order to improve the "individuality" of digital hearing aids, in this paper, an FIR reconfigurable filter bank is proposed. The proposed filter bank is expected to provide multiple band decomposition plans and has small computational complexity. To make the filter bank reconfigurable, interpolation and decimation techniques are used. To lower the complexity, frequency-response masking technique is employed.

II. PROPOSED FILTERBANK

A. FILTER BANK

A filter bank is an array of band-pass filters that separates the input signal into multiple components, each one carrying a single frequency sub-band of the original signal. One application of a filter bank is a graphic equalizer, which can attenuate the components differently and recombine them into a modified version of the original signal. The process of decomposition performed by the filter bank is called analysis (meaning analysis of the signal in terms of its components in each sub-band); the output of analysis is referred to as a sub band signal with as many sub bands as there are filters in the filter bank. The reconstruction process is called synthesis, meaning reconstitution of a complete signal resulting from the filtering process.

In digital signal processing, the term filter bank is also commonly applied to a bank of receivers. The difference is that receivers also down-convert the sub bands to a low center frequency that can be re-sampled at a reduced rate. The same result can sometimes be achieved by under sampling the band pass subbands.Another application of filter banks is signal compression, when some frequencies are more important than others. After decomposition, the important frequencies can be coded with a fine resolution. Small differences at these frequencies are significant and a coding scheme that preserves these differences must be used. On the other hand, less important frequencies do not have to be exact. A coarser coding scheme can be used, even though some of the finer (but less important) details will be lost in the coding.

III. GENERATION BLOCK

A. Determine control parameter

The function of Generation Block is to produce magnitude responses having multiple pass bands.The proposed structure for the generation block is shown in Fig.1.The prototypefilter(z) is first decimated by factor of *MI*and then interpolated by factors $D1$ and $D2$, respectively. Weknow that $H(z^{Mi}/_{D1})$ and $H(z^{Mi}/_{D2})$produce two frequency responses with the same number of pass

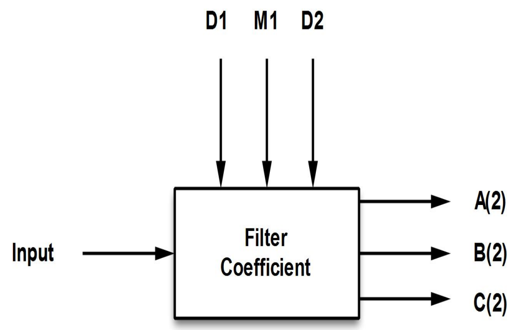

Fig.1.structure of Generation block

Suppose the cutoff frequency of H (z) is f_c,we have

$$PBWj = \frac{2(fc.Dj)}{Mi} \quad j=1,2 \quad (1)$$

The stopband width SBW_j can be calculated as

$$SBWj = \frac{2(\pi - fc.Dj)}{Mi} \quad j=1,2 \qquad (2)$$

Now, we can form the three branches of frequency responses, noted as $A_i(z)$, $B_i(z)$, and $C_i(z)$.

The magnitude responses of $A_i(z)$, $B_i(z)$, and $C_i(z)$ are shown. By changing M_i, the can be changed.

$$Ai(Z) = H(z^{Mi}/_{D1}) \quad (3)$$

$$Bi(Z) = H\left(z\frac{Mi}{D2}\right) - H\left(z\frac{Mi}{D1}\right) \quad (4)$$

$$Ci(z) = Z^{-\Delta} - H\left(z\frac{Mi}{D2}\right) \qquad (5)$$

IV. SELECTION BLOCK

A. Design mashing filter

The sub band-selection block whose functionis to extract the desired sub bands? Control information is sent toboth blocks so that the blocks can be reconfigured accordingly.

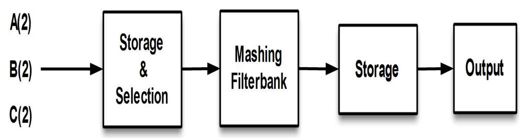

Fig.2.structure of selection block

Thefrequency-response masking technique is used to reduce the complexity of the filters. The status of the switches in the filter bank is determined by a control signal *Wi* .Therefore, different *Ws* lead to different transfer functions of the masking filter bank. We can control which sub bandto be extracted by setting the value of *Wi* and *Si*.

B. Design parameters

Suppose the bandwidths of the passbands of $A_i(z)$, $B_i(z)$, and $C_i(z)$ are equal, the decimation factor D_1 and D_2 as well as fc should satisfy some constraints. They are

{SBW2=PBW1(6)

{PBW2=3PBW1 (7)

From the aforementioned constraints, we have

$$fC = \frac{\pi}{4.D1} \quad (8)$$

$$D2 = 3D1 \quad (9)$$

C. Design procedure

The design procedure of the proposed filterbank can be de-scribed as follows.

1) Determine the control parameters M, D1 and D2. The elements of Mshould satisfy. The values of D1 and D2 should satisfy.

2) Design the prototype filter H (z) and form the three branches of frequency responses Ai(z), Bi(z), and Ci(z).H (z) should be band limited. The cutoff frequency ofH (z) can be obtained. $A_i(z)$, $B_i(z)$, and $C_i(z)$ can be generated.

3) Design the masking filterbank for each Mi. The com-plementary filter and the highpass symmetric filter of the prototype filter can be used to simplify the design.

The pro-totype filter is dependent on the output of the multiband-generation block, thus should be discussed case by case.

V. RESULTS AND DISCUSSIONS

The proposed filterbank enables us to use different plans to divide the input sound waves based on the patients' own char-acteristics of the audiograms. The gain for each subband is adjustable to suit the needs of the hearing impaired, i.e., the am-plitude response of the filter bank should equalize or "match" the audiogram. It is important to note that in a real application, the recruitment phenomenon must be taken into consideration.

The gain formu-las such as NAL-NL2, DSLv5, and CAM2 should be used. The reconfigurable filterbank is used to match differ-ent audiograms.We can see that the proposedfilter bank provides reasonable matching between the audiogram and magnitude response of thefilterbank. The maximum matching error compared to the uniform filter bank andnon-uniformfilter bank.

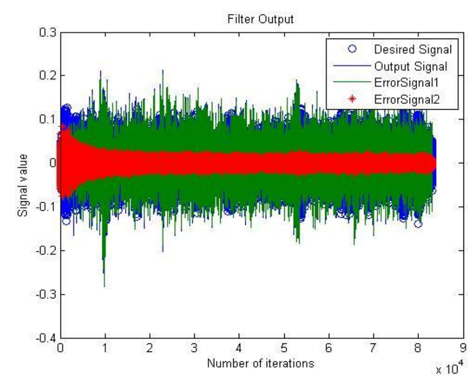

Fig 3. Filter output

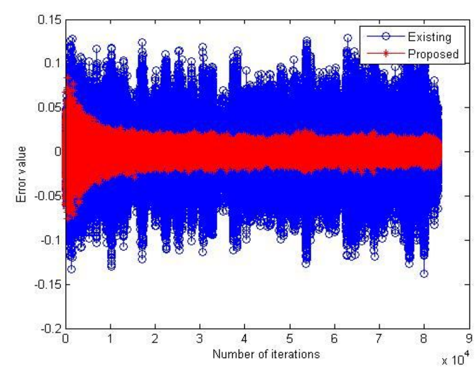

Fig 4. Comparesence between existing and proposed method.

VI. CONCLUSIONS

Tuning of filter parameter such as filter coefficient and filter type and mode selection is complex. So it's not efficient for nonlinear noise in speech signal. So a fractional Fourier transform filter for reduction of noise in hearing aids. The use of interpolation, decimation, and frequency-response masking enables us to reduce the computational complexity by realizing the entire system.

By changing the value of the control signals, the transfer function of the filter bank can be changed, thus providing different sound decomposition methods, thus the complexity of the masking stage is reduced greatly. Only three prototype filters are needed for the whole system. Simulation shows that this design can obtain better matching to the audiogram compared with traditional methods.

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