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Variable Digital Filters used for Hearing aid application: A Survey

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ABSTRACT
Lack of earshot cause hearing loss problems in individuals. Hearing aid is an electroacoustic device used to aid the hearing loss problems. Digital filter bank is an important module used in digital hearing aids. Filter banks comprises of the Variable digital filters. Finite Impulse Response filters is a type of Variable digital filter which is more stable and sensitive than the Infinite Impulse Response filter. This study provides the different FIR filter design techniques used in hearing aid applications. This study also discusses the various approaches for filter banks in hearing aid with respect to the parameters such as complexity, power consumption and size.
Keywords - FIR filters, Digital filter bank, Hearing aid, VCF(Variable Cutoff Frequency)

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I. INTRODUCTION

Hearing loss or hearing impairment has been a major problem among many people, affecting their day to day life interactions. To compensate this problem, one of the solution is to make use of hearing aids. The field of digital signal processing has been making improvements in terms of size, power, complexity and various other parameters so as to have an efficient implementation of this device. For the designing of the hearing aid one of the most important module is the digital filter banks. There has been an increasing demand for the filter banks due to its emerging trends towards the use of compact systems with better quality. A typical filter bank comprises of analysis block, processing unit and synthesis block. The analysis block splits the input signals into sub bands. The sub bands pass through the processing unit and the synthesis block recovers the original signal at the end. The filter bank is a collection of the various filters called the variable digital filters. A variable digital filter can be defined as the filter whose frequency response can be changed on-the-fly. And it is also known that for these filters, all the parameters need not be changed all the time.

The variable digital filters are basically of two types-Finite Impulse Response (FIR) and Infinite Impulse Response (IIR). Finite Impulse Response is preferred over Infinite Impulse Response due to its property of linear phase. It also has the characteristics of stability and coefficient sensitivity benefit. [1]

The FIR filters can be implemented by various techniques, and these techniques are used in the designing of the digital hearing aids. Hence it becomes necessary for us to study the different techniques to design the FIR filters.

This paper is structured as follows. In Section 2 the various FIR design techniques is discussed that is followed by the various approaches used in digital hearing aid. In section 4 the comparison table is shown for the approaches in hearing aids and the methods are discussed, finally the paper is concluded in section 5.

II. FIR FILTER DESIGN TECHNIQUES:

The FIR filter can be obtained by various methods. One of the simplest method to design a FIR filter is called the variable coefficient VCF filter where the FIR filter coefficients is changed as per the desired frequency response[2]. The corresponding filter coefficients of it is stored in the memory. Based on the control over the cutoff frequency FIR filters could be designed in various methods which include the Coefficient decimation method, Modified Coefficient decimation method, Interpolation approach, Farrow Structure and two staged FRM approach.

1.1 Coefficient Decimation Method: The Coefficient decimation method for designing the FIR filters makes use selective filter coefficients by replacing them with zeros or discarding them [2] so as to get the filter response as shown in[14,15]. The coefficient decimation is basically of two types CDM I and CDM II. In CDM 1 the low pass prototype filter is decimated as a Dth factor while others are made zero as so as to obtain a multiband frequency response. While in CDM II operation every Dth coefficient is retained and all others are discarded to obtain a low pass frequency response with its cutoff frequency and transition bandwidth being D times of the low pass prototype filter. The CDM operation frequency responses is shown in [2].

1.2 Modified Coefficient Decimation method: The modified coefficient decimation method is proposed in [8]. Even MCDM has two types MCDM I and MCDM II[9]. The MCDM I operation decimates the prototype as Dth coefficient and is retained , the sign of every alternative coefficient is reversed and all other coefficients are made zeros [2]. In MCDM II every Dth coefficient is retained and all others coefficients are discarded [2]. Every alternate retained coefficient is reversed to obtain a high pass frequency response with its bandwidth being D times that of the prototype filter.

1.3 Interpolation method: The Interpolation approach is used to obtain a narrow transition bandwidth filter from an existing wider transition bandwidth filter [2]. The interpolation approach itself makes use of the FRM, where the M-1 zeros are inserted between the successive coefficients of the prototype filter and this new impulse response is interpolated by using a suitable cascaded filter [2]. The cascaded filters is used to extract the desired band by masking the other bands present [2].

1.4 Farrow Structure: In the Fractional delay structure based VCF FIR, the cutoff frequency is inversely proportional to the interpolation factor. The cut off frequency can be varied continuously by varying the interpolation factor [2]. This means that the unit delay of the filter structure needs to be replaced by a fractional delay[2]. This is achieved by a second order modified Farrow structure [10].

1.5 Two staged FRM approach: FRM based filter have a large group delay [2].The two stage FRM approach in [6,7] is used to get a multistage design. The FRM technique is widely used in the digital hearing aid design of filter banks.
In the fast filter bank, from the second stage onwards interpolation as well as the frequency shifted masking filters is used to extract the bands from the interpolated subbands from the first stage [2].

III. DIFFERENT APPROACHES USED IN DIGITAL HEARING AID:

The general working procedure of the digital hearing aid is as follows. The microphone takes the input from the environment in analog form. The A/D converter converts the analog signal into digital form. Then the digital filter bank separates the input signal into multiple channels each having a particular band of frequency. After adjusting the gain for that required band, the D/A converter converts the processed signal into analog form.

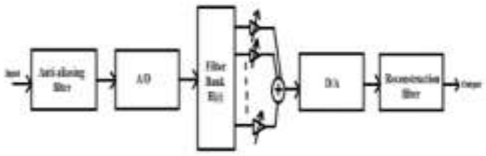

Figure 1: Block diagram of a Digital Hearing Aid

In the low complexity design of non-uniformly spaced digital FIR filter banks, the proposed technique makes use of 10 non-uniformly spaced sub-band filters. So as to have reduced multipliers and adders. The technique used here is the FRM (Frequency Response Masking) technique. The FRM technique makes use of 2 half band as prototype filters so that the arithmetic operations can be reduced. The synthesis shown is for 80 db using 10 multipliers[3].

With the 2 half band filters as the prototype, symmetricity occurs at the mid frequency point. It makes use of the masking filter with minimum order. The lower and upper bands are made symmetric. Efficient implementation is obtained by making all the coefficients non zero and only the central coefficient is zero. The frequency response masking is done by repetitive use of Mf(z). This proposed filter bank can be used only for the audiogram with sharp transition hearing loss at low and high frequency regions. It is not applicable for mid frequency ranges this can be achieved by redesigning the filter bank or making use of additional subbands[3].
FRM technique which implements 8 non uniformly spaced sub band filters by Arun Sebastian et.al,in [4] is similar to that proposed in 2014. The simulation results were shown for filter bank giving 120 db attenuation making use of 13 multipliers only. The proposed technique involves single prototype filter and six sub filters in total using 13 multipliers only. The proposed filter bank gives a maximum matching error in low and high frequency losses, it has low complexity compared to the existing methodologies which could be used for various hearing loss. The 8 band non uniform

linear phase digital filter, is good for the variations occurring in the mid frequency regions[4].

IFIR (Interpolated Finite Impulse Response) filter ,it is used for designing the narrow band filters, with less complexity compared to existing filter design methods. The design of low complex 17 band non uniform IFIR filter bank is used for the digital hearing aid applications. Frequency transformations are applied on the prototype low pass filter, then the high pass filter and the band pass filters have been created. Interpolations has also been performed on these filters so as to get various bands. Audiogram matching is done on this technique and good results have been obtained.The FIR filter used is obtained by Parks Mc-clean algorithm. This technique gives equiripple passband and stopband. This filter bank requires less number of multipliers and hence the hearing aid would be small in size. It has also given better results in audiogram matching experiments. Good matching of audiogram can be achieved by making use of less number of bands and adjusting their widths. The design requires a minimum order filter for better response.[5]

Implementation of a Low pass Finite Impulse Response(FIR) filter of order 10 in [10] are used for the hearing aid. For better optimization different multiplications have been used which include vedic multipliers booth multiplier, modified booth multiplier to multiply filter coefficients with the input sequence. And for adding the product terms Ripple carry adder, carry look ahead adder and carry save adder have been used. For the FIR realisation there are basically 2 techniques involved Transpose form and Direct form. The synthesis result for the FIR designed in MATLAB using Xilinx 14.7, Spartan 3E XC3S500E.Booth multipliers is used for multiplying signed binary numbers. Vedic multiplier is one of the fastest approach to perform multiplication.8 bit vedic multiplier is shown in the figure. The modified booth multiplier is to perform high speed multiplications. The constant multipliers is performed by using shift and add operation corresponding to the non-zero bit position.. Different combinations had been made as shown in the table in [10] , it was observed that ripple carry adder and constant multiplier uses minimum number of slices, while the filter having a combination of ripple carry adder and modified booth has more number of slices. From the synthesis results it has been concluded that the Transpose form structure is better than the direct form structure. The constant multiplier has highest maximum frequency in direct form structure and lowest in transpose structure. While in the transpose form modified booth gives maximum frequency.[10]

Frequency response masking (FRM) technique is a computationally efficient method for the design of sharp FIR filters. FRM can save more than 98% of multipliers in sharp FRM filters, which leads to significant power and area savings.[1]

FRM Filters classified as:
1. Single stage
2. Multi stage

In single stage FRM filters the prototype filters are replaced by M delays and seen, after replacing each delay in the prototype filter with M delays, the bandwidth of transition band and passband is reduced by 1/M in [1]. By carefully choosing the band edges for the two masking filters and cascading them with the two multi-band filters, unwanted passbands can be eliminated. Finally, the resulting band- pass filter with narrower transition band can be obtained by adding the two masked multi-band filters together. In multi stage filter the transition bandwidth is extremely narrow, multi-stage FRM filter is applied to reduce the cost of hardware and power further. The FRM filters are the best for implementing hearing aid in hardware and the delays used can be reused[1]. It helps to save more than 70% of the taps, while in the filtering of hearing aid, multiple sub-filters are needed in order to cover all the frequency ranges, and FRM technique also allows significant savings in terms of number of multipliers.[1]

The Digital hearing aids are self-adjusting and flexible. The functionality and technical performance of DSP hearing aids are more efficient compared to analog hearing aids. Adaptive filtering is used which effectively cancels out the additive noise by changing the filter coefficients over time and adapts to the changing signal characteristics according to an optimization algorithm. Adaptive filtering is used which effectively cancels out the additive noise by changing the filter coefficients over time and adapts to the changing signal characteristics according to an optimization algorithm in [12]. The problem is high frequency sounds on the hearing curve looks like a ski slope in an audiogram and is a special kind of sensorineural hearing loss. So it will be difficult to hear high pitch voices to overcome this, they have used Spectrogram to show the filtering process and the outcome of the same. So using adaptive filtering technique the noise can be removed and amplification of different filter banks can be done.[12]

The Farrow structure are used to design variable digital filter. The hearing aid characteristics are higher flexibility, minimum hardware, low power consumption, low delay and linear phase (to prevent distortion) . The non-uniform filters are used for better filter outcome. Non uniform filters uses FRM i.e. frequency response masking . Variable bandwidth (VBW) filter can be effectively realized using Farrow structure. A bank of filters is designed based on variable bandwidth filters that can be used as part of a digital hearing aid for the purpose of audiogram fitting [11].

The digital hearing aid simulated on MATLAB in [13]. The implementation of this configurable digital hearing aid system includes noise reduction filter, frequency-dependent amplification and amplitude compression.
Block diagram representation for MATLAB implementation of Digital Hearing Aid System is given below:

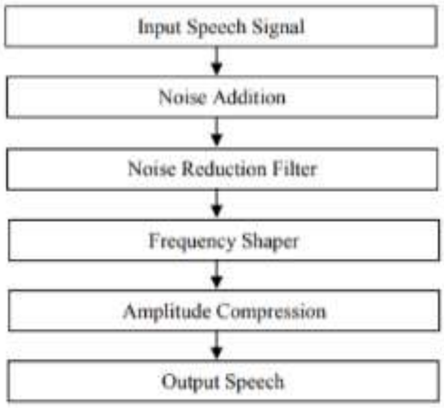

Fig 2: Block diagram of a hearing aid system

An input speech signal is passed through various functions i.e., noise addition, noise reduction filter, frequency shaper and amplitude compression before producing an adjusted output speech signal which is audible to the hearing impaired person.

- **Noise Addition**: The input signal is noiseless. Hence to simulate a real world scenario an additional noise i.e., Additive White Gaussian Noise (AWGN) and random noise are added using MATLAB function.
- **Noise Reduction Filter**: Sometimes it is difficult for the hearing aid users to differentiate noise and intended speech signal .Therefore to eliminate the noise, a reduction filter function is used in this design.
- **Frequency Shaper**: One major problem that the hearing aid users face is that the hearing aid amplifies all signals rather than the significant signal that they desire to hear. Most of the patients have difficulty to hear high frequency signals. Therefore, the frequency shaper is designed to correct for the loss of hearing at certain frequencies.
- **Amplitude Compression:** Amplitude compression function is the task of controlling the overall gain of a speech amplification system. Amplitude compression will ensure that the amplified signal will not exceed saturation power [13].

IV. RESULT AND DISCUSSION:

The filter banks used in digital hearing aids make use of various techniques in order to give better results with respect to complexity, delay, power consumption or hardware implementation**.** The below table gives the comparison of the various approaches used for the digital hearing aids:

Table Comparison:

Author Name	**Methods used**	**Proposed filter bank name**	**Advantages**
Arun Sebastian, Ragesh M. N and James T. G (2014)	Frequency Response Masking Technique (FRM)	Low Complex 10 band non uniform filter bank.	This method is applicable for high and low frequency range.
Arun Sebastian and James T. G. (2015)	Frequency response Masking Technique (FRM)	8 non uniformly spaced sub band filters	This method is applicable for mid frequency range as well.
Tomson Devis, Manju Manuel	Interpolated Finite Impulse Response Technique(IFIR)	17-Band Non-uniform Interpolated FIR Filter Bank	This method gives less chip area.
Kazi Nikhat Parvin, Md. Zakir Hussain	Survey on various multipliers and adders.	1.Booth Multiplier 2.Vedic Multiplier 3.Constant Multiplier	The constant multiplier with ripple carry adder has highest frequency in direct form structure.
Zhongxia Shang	FRM technique	Low Power FIR Filter Design	-Significant number of saving in

, Yang Zhao, Yong Lian, Fellow			multipliers -The technique gives reduced cost and low power.
Nisha Haridas, Elizabeth Elias	Farrow Structure	Efficient Farrow Structure Based Bank of Variable Bandwidth Filters	-Efficient to implement Variable Bandwidth Structure. -Low complexity
Rithwik Dhawan Et al.	Noise reduction, amplitude compression and frequency shaping	Digital Filtering in hearing aid system for impaired hearing	-MATLAB implementation shown by removing the noise factor -Problems faced by hearing aids related to high frequency signals is corrected.

The FRM technique can be applied for high to low frequency and mid range as well. FRM makes use of the sub band filters which is similar to that of the interpolation technique. It reduced the number of multipliers, gives low power consumption based on the increase of the sub filters at the required frequency ranges. Farrow structure is another efficient method which uses the variable bandwidth filter which gives a low complex hardware . Adaptive filtering and noise reduction techniques are used at the implementation stage for the digital hearing aids. The most widely used technique is the Frequency response masking technique which involves masking as it gives good audiogram matching results and makes use of less number of multipliers and also gives low power consumption by adding sub bands at each stage. While the Interpolation FIR filter bank occupies less chip area. Farrow structure is not used much in the audio applications fields.

V. CONCLUSION

We have presented a survey on the various FIR design techniques which could be used in hearing aid applications. The scope of this paper is limited to the filter bank techniques which have widely been used for the hearing aids. A comparison based on these proposed filter bank technique is drawn. Among the various FIR filter design methods Interpolation approach, FRM(Frequency Response Masking), Farrow structure is used in designing the digital hearing aid. Among all these techniques the most widely used is the FRM technique along with interpolation method. Since there are different parameters, each approach has been achieving better results for the DHA's.

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