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Study on Speech Quality Improvement of Processed Signal in Reconfigurable Digital Filter used in Digital Hearing Aid Device

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Abstract

Digital filters with amplifier used in most of modern day Digital hearing aids for processing sound signals for good quality of sound and compensating hearing loss. Digital signal processing plays a vital role in current advancement of music and audio systems used in domestic and commercial applications also in medical devices. Earlier analog signal processing doesn't have that flexibility for sound equalization in hearing aid device. Audio signals covers frequency range from 0 to 8 KHz as per audiogram and realized through six subbands with help of digital filters and amplify processed signal and transfer it to the ear. Audiogram is hearing response of both ears taken for puretone audio wave signals at frequencies 250 Hz to 8000 Hz and Loss measured in dB as hearing loss and marked on scaled graph. Many Digital Hearing Aids manufacturing companies all over world are behind continuous research and development for improvement of speech quality of processed signal to improve user experience through using variety of digital signal processing algorithms, to make them more adaptive to individual patient hearing loss characteristics. Fixed subband filter design is used in most of available hearing aids designs. In this paper we have designed a reconfigurable transfer function type of single digital FIR filter to achieve a best fitting to audiogram as per specifications with IEC 60118-15 standard. Speech quality measurement of processed signal is very important to approve a filter design in digital hearing aid device, we tested processed signals with ITU-T-PESQ standards for measurement of speech quality .We used TIMIT speech corpus audio signals to test designed filter and results presented in comparison with the fixed filter banks and in next sections.

Keywords: Hearing Loss, Digital Hearing Aids, Digital Filters, Reconfigurable Filter. Speech Quality,PESQ

1.0 INTRODUCTION

In practice only 20% of patients having hearing loss use hearing aid, and about one fourth of those do not use hearing aid because of unpleasant irritating noise and whistles in their everyday life due to surrounding background noise [6]. Latest signal processing algorithms and techniques are used in upgraded digital hearing aids, but, sadly, around half of patients percentage are satisfied with performance of their hearing aids in changing noisy situations. This dissatisfaction may be algorithms used in filter designs are not currently filter out required signal and not blocking background noise effectively. Hearing aid try to match audiogram gain (amplification) and dynamic range compression to make low intensity sounds audible and keeping loud sounds comfortable. Hearing aids works fine in different noisy situations but patients still have some difficulty in acoustically complex or crowded environments. Challenge for hearing aid designers is to restore loudness or to improve understanding of speech, for this gain of present hearing aids is kept nonlinear.

Sound wave decomposed by using fixed filters and used in most of present hearing-aid systems. Fixed filter banks, cannot provide sufficient flexibility for the adaptability of different hearing impairment cases. A reconfigurable filter banks can be designed as per required number of channels and control parameters to have more adaptable hearing aid device as per user requirements in noisy situations

2.0 PRESENT THEORIES AND PRACTICES:

Digital filter banks types used in hearing aid

Fixed bands FIR filter banks

Fixed sub band filter design is used in most of available hearing aids today. Fixed filter banks, cannot provide sufficient flexibility for the adaptability of different hearing impairment cases. Two-stage filters have adaptive FIR filter and a fixed IIR filter. The fixed IIR filter use fixed coefficients, which roughly estimate the feedback in acoustic path. In offline fitting of hearing aid fixed coefficients are used in design, and hence the values of these Coefficients are totally different for various hearing-aid wearers. The adaptive FIR filter is employed to model the distinction between the IIR filter and the changing acoustical feedback path. As variation of this difference is too small than acoustical feedback path and length of adaptive FIR can be much shorter than it would be if the fixed IIR filter were included. Therefore Fa-Long luo and Arye Nehorai expressed in 2006 that short-tap adaptive FIR filtering may result in straight forward computations and conjointly a quick convergence in real-time implementation [7].

Reconfigurable bands FIR filter banks

Reconfiguarbility of the proposed filter bank enables deaf individuals to customize hearing aids supported their own specific conditions to improve their hearing ability. However their proposed filter bank can do a far better matching to the audiogram and has smaller complexity compared with the fixed filter bank at cost of processing delay of more than 20ms, which is unacceptable for any practical hearing aid device. They have mentioned it open problem to design filter banks with adjustable sub bands which will be custom-made for a personal hearing-loss case with acceptable processing delay and small work has been done in this regards [1].

Reconfigurable FIR banks types

A reconfigurable filter banks can be designed as per number of channels required and control parameters to have more adaptable hearing aid device as per user requirements in noisy situations.

Ying Wei and Debao Liu mentioned in details for these hearing aid reconfiguration issues in their paper in which “Reconfigurable” implies that the subbands are adjustable in line with some control parameters without changing the structure of the filterbank, in order to improve the “individuality” of digital hearing aids. Reconfigurable filters are a type of filters that change filter parameters to minimize a difference between a desired filter output and ideal output. Acoustic noise is a noised signal that simulates the noise signal

Uniform Filters bank

Uniform ANSI Filter bank is rarely used in digital hearing aid device because of its high computation complexity and rather large group delay, even though it has the advantage of good match to human hearing characteristics. [8]

FIR filter is always stable and linear phase characteristics, if their coefficients are symmetric. Such properties are most welcome for hearing aid devices due to the requirement of arbitrary magnitude adjustment in the different frequency bands [9]

The drawback of an FIR filter is high computational cost as requirement of large number of multipliers. In order to reduce the filter complexity [9]

Series Parallel Filter Banks

In modern hearing aids signal processing is performed in a subband domain which adds analysis–synthesis delays. Long forward-path delays are not desirable because the processed sound combines with the unprocessed sound that arrives at the cochlea through the vent and changes the sound quality. But subband signal processing is the most popular choice for hearing aids because simplicity. We present an alternative digital hearing aid structure with low-delay characteristics in this research paper[2]. A hearing aid device should match ear response with all aspects of the speech signal that are important to hearing aid device user.[10]

3.0 RECONFIGURABLE FILTER BANK DESIGN PROCESS

<u>Step 1 to 3.</u>

1. Divide hearing frequency band in six frequency band as per audiogram
2. Assign nyquist frequency windows wn1,wn2,wn3,wn4,wn5,wn6 in non uniform pattern
3. Frequency range in non uniform pattern 250-500Hz,500-1000Hz,1000-2000Hz,2000-4000Hz,4000-8000Hz,8000-16000Hz bands.

Hearing aid frequency response is nonlinear and tried to match ear response of hearing aid user by conducting a test by ENT doctor called as audiogram. while designing filter bank we divide hearing frequency band in six frequency band windows wn1,wn2,wn3,wn4,wn5,wn6 for decomposing audio signal into frequency range in non uniform pattern 250-500Hz,500-1000Hz,1000-2000Hz,2000-4000Hz,4000-8000Hz,8000-16000Hz bands. We will use windowed linear-phase FIR digital filter designed by classical method. In MATLAB fir1 implements digital FIR filters where filter magnitude response is normalized to 0 dB at the center frequency of the passband . by using fir1 command we can designs standard low, high, bandpass, and bandstop filter configurations.

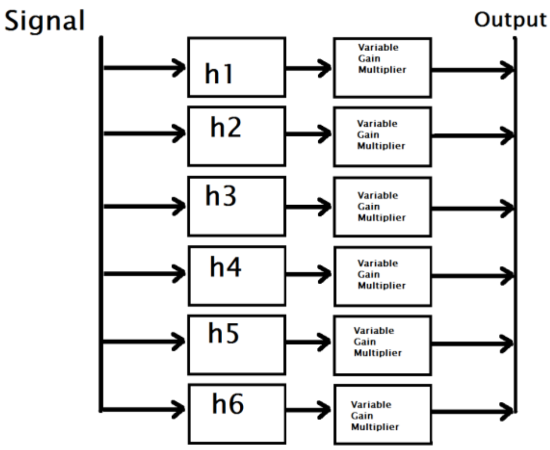

Figure 1.0

b = fir1(n,Wn) This operation with fir1 command gives output ‘b’ which contains the n+1 coefficients where order of lowpass FIR filter is ‘n’. We will use Hamming-window filter which have linear-phase properties and ‘Wn’ normalized cutoff frequency. filter coefficients ‘b’ have descending order powers of z.

Nyquist frequency ‘Wn’ has range between 0 to 1,

Wn have two-elements, as [w1 w2], fir1 gives a bandpass filter having passband of $w1 < \omega < w2$.

Step 4- Design linear-phase windowed FIR filter for each window for filter order 80

By using this we will design six FIR filters as h1,h2,h3,h4,h5,h6 ass below

```
h1=fir1(FiltOrder,wn1);
h2=fir1(FiltOrder,wn2);
h3=fir1(FiltOrder,wn3);
h4=fir1(FiltOrder,wn4);
h5=fir1(FiltOrder,wn5);
h6=fir1(FiltOrder,wn6);
```

This type of filter function is direct form II transposed structure as below,

a(1)*y(n) = b(1)*x(n) + b(2)*x(n-1) + ... + b(nb+1)*x(n-nb) - a(2)*y(n-1) - ... - a(na+1)*y(n-na)

where 'na' is the feedback, and 'nb' is the feed forward filter order.

This filtering operation can be expressed in the z-transform as transfer function,

Initially we will keep filter order 40 and will vary up to 80.Nyquist frequencies will be from 0 to 1 and adjusted bands as wn1,wn2,wn3,wn4,wn5,wn6 and for this we will use signal sampling frequency Fs.

```
wn1=[250 500]/Fs;
wn2=[500 1000]/Fs;
wn3=[1000 2000]/Fs;
wn4=[2000 4000]/Fs;
wn5=[4000 8000]/Fs;
wn6=[8000 16000]/Fs;
```

These Six Filters we can connect in parallel and can fed our test signal simultaneously as in Figure 1.0.

We will multiply by a gain factor to increase amplitude of voice signal in particular frequency band as per patients hearing requirement and match to Audiogram.

We now design digital filter with frequency response identical with below audiogram .

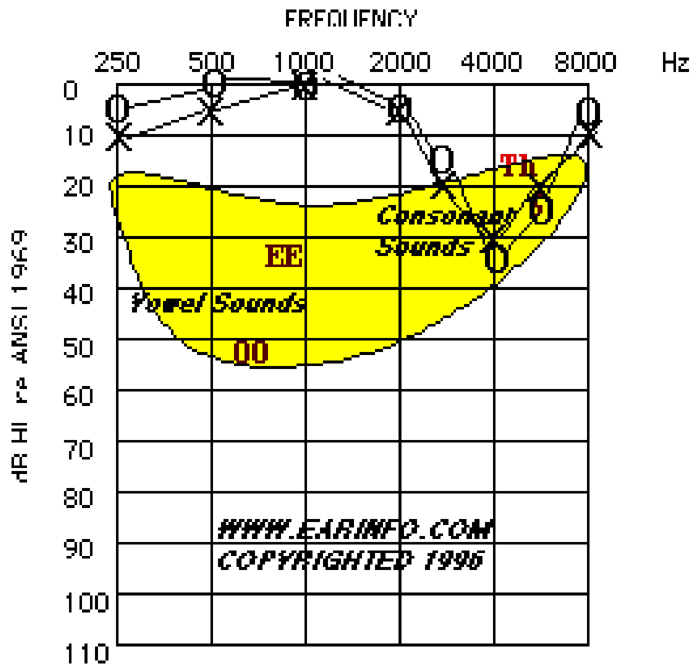

Figure 1.1

Audiogram is test taken by hearing aid doctor with pure tone audio frequency signals in range 250 to 8Khz and mark O-zero for left ear and X-cross for right Ear on Y axis and They mark magnitude/Amplitude of hearing sensitivity on X-Axis.

Step 5 - Multiply by a gain factor to match audiogram hearing loss in dB to each .

Above Audiogram is taken from www.Earinfo.com website and this represent hearing problem to patient at high frequency. We will match our filter frequency response to above audiogram by adjusting gains to these different bands.

hd1=h1*5;

hd2=h2*0;

hd3=h3*0;

hd4=h4*5;

hd5=h5*35;

hd6=h6*5;

This filter bank we can use traditional way but its hardware implementation cost is more requires more area and numbers of multipliers and delay units to implement on FPGA chip.

<u>Step 6 - Calculate transfer function of each subband filter</u>

To get Reconfiguarbility of filter bank, we will compute single transfer function for entire bank including added gains and use this single transfer function as filter to voice signal and check its frequency response to match with audiogram.

```
tt1=tf(hd1);
tt2=tf(hd2);
tt3=tf(hd3);
tt4=tf(hd4);
tt5=tf(hd5);
tt6=tf(hd6);
```

We use tf to create transfer function models to transfer function form.

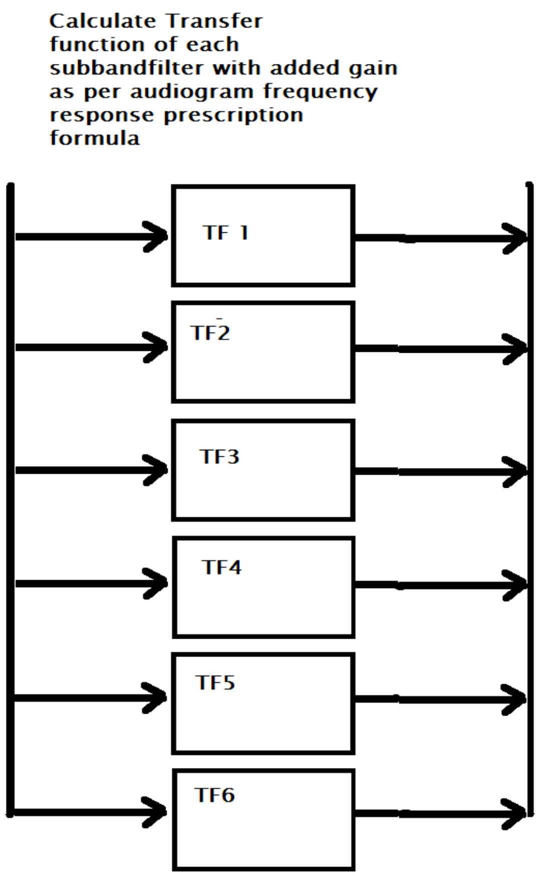

Figure 1.2

Step 7 - Combine all six transfer function in single transfer function by parallel operation

After computing each individual transfer function we will combine all these transfer functions into single transfer function by using parallel command.

```
tta =parallel(tt1,tt2);
ttb =parallel(tt3,tt4);
ttc =parallel(tt5,tt6);
tte =parallel(tta,ttb);
ttf =parallel(tte,ttc);
```

'ttf' is final combined transfer function for complete filter bank. parallel connects two model objects in parallel. This function accepts any type of model. Two models must be continuous or discrete and with same sample time.

We will combine all filters tranfer function by parallel command and combine all six filters transfer function into single transfer function by MATLAB parallel command

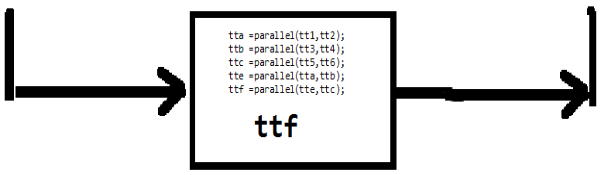

Figure 1.3

Step 8 - By taking denominator coefficients of this reconfigured transfer function design filter

Now we have transfer function 'ttf' and its coefficients, we can use these coefficients to build our filter to use in digital hearing aid device, so we will keep numerator am =1 and to calculate values of denominator bm use cell2mat function .

```
bm=cell2mat(ttf(1:end).num);
```

cell2mat function convert cell array to numeric array, once we get 'am' and 'bm' values

we can use them to filter test voice signal by using filter command.

Step 9 - Test this filter different audiograms from IEC standards with TIMIT database

Reconfigured digital filter by using transfer function coefficients

out3x=filter(bm,am,signal);

The filter function filters a data sequence using a digital filter which works for both real and complex inputs. Direct form II transposed Filter implementation have standard difference equation.

y = filter(b,a,X) data in vector X is filtered with the filter having transfer function numerator coefficient as 'b' and denominator coefficient 'a'.

filter operates on the columns of X.

a(1)*y(n) = b(1)*x(n) + b(2)*x(n-1) + ... + b(nb+1)*x(n-nb)- a(2)*y(n-1) - ... - a(na+1)*y(n-na)

where n-1 is the filter order [1], 'na' is the feedback, and 'nb' is the feedforward filter order.

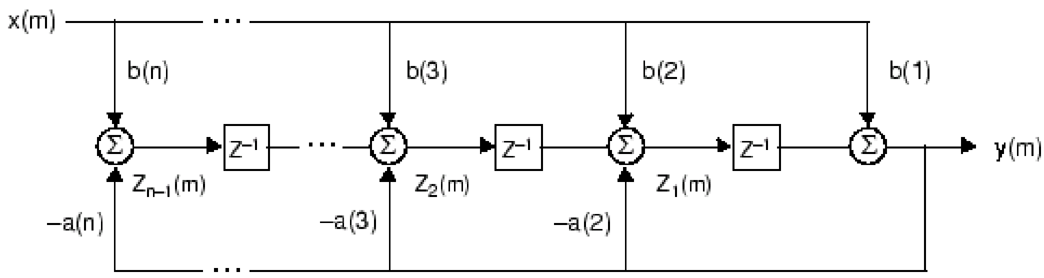

Figure 1.4

The filter operation for sample 'm' is mentioned by time domain difference equations as below

$$
\begin{aligned}
y(m) &= b(1)x(m) + z_1(m-1) \\
z_1(m) &= b(2)x(m) + z_2(m-1) - a(2)y(m) \\
\vdots \;\; &= \;\; \vdots \qquad\qquad\quad \vdots \\
z_{n-2}(m) &= b(n-1)x(m) + z_{n-1}(m-1) - a(n-1)y(m) \\
z_{n-1}(m) &= b(n)x(m) - a(n)y(m)
\end{aligned}
$$

The input-output description of this filtering operation in the z-transform domain is a rational transfer function

$$Y(z) = \frac{b(1) + b(2)z^{-1} + \ldots + b(nb+1)z^{-nb}}{1 + a(2)z^{-1} + \ldots + a(na+1)z^{-na}} X(z)$$

Above designed single transfer function digital filter can be used in digital hearing aid with low hardware requirement .Due to less hardware implementation cost of adders and multipliers will result in a low delay filter, which will have require low power and superior hearing performance than individual sub band filters processing digital filter bank.

Step 10 - Check signal quality of processed signal using ITU-T-PESQ standards

Now we will test this filter for their performance with different Audiograms standards specified by IEC 60118-15. We will test with six Audiogram Vectors Chosen for the Flat and Moderately Sloping Group specified in IEC 60118-15.Also we will use standard speech test signals by TIMIT database. For development of automatic speech recognition and acoustic-phonetic information TIMIT corpus is created by DARPA-ISTO (Defense Advanced Research Projects Agency - Information Science and Technology Office). TIMIT includes 6300 sentences, by 630 users in which 10 sentences spoken by each of them. These users are from 8 different dialect regions of the United States. We will use dialect region of New England and we will use SA1 speech signal spoken by 9 users in which 4 female and 5 Male sound having recorded at different frequencies are used [5].

We tested our designed reconfigured filter for above mentioned six IEC 60118-15 standard audiograms by passing TIMIT speech signals.

After this we have checked quality of speech processed signal by filter by using PESQ(Perceptual evaluation of speech quality) standard. PESQ is method for speech quality assessment used for telephone networks and speech processing devices .PESQ have speech codecs and standard methods for quality measurement.

The perceptual model of PESQ calculates distance between the original and degraded speech signal (PESQ score). The PESQ score is in the range of –0.5 to 4.5 and expressed to a MOS scale single number where output range will be between 1.0 and 4.5, in an ACR listening quality experiment the normal range of MOS values found. PESQ score of near to 4.5 is treated as good quality signal and signal less than or near to 1.5 is considered as poor quality or degraded signal [4].Below table shows audiogram and PESQ score for TIMIT DR1 database of speech signal SA1 With filter order 80.

Step 11

Table 1.0

No.	ID	Rank	Category	250	500	2000	3000	4000	6000
N1	A4	36	Very Mild	10	10	10	20	35	40
N2	A31	6	Mild	20	20	35	40	50	50
N3	A23	2	Moderate	35	35	50	55	60	70
N4	A48	4	Moderate/severe	55	55	65	65	75	80
N5	A21	19	Severe	65	70	80	75	80	80
N6	A22	16	Severe	75	80	90	95	100	100
N7	A17	34	Profound	90	95	105	105	105	105

4.0 PERFORMANCE PARAMETERS OF DESIGNED RECONFIGURED FILTER BANK

Table 1.1 shows PESQ scores for audiogram N1 with our designed filter with filter order 80 .We tested this filter structure by giving input test signal from TIMIT database directory DR1 and filter processed audio signal is measured for it quality degradation by using ITU-T PESQ standard. We have calculated average values of PESQ MOS= 4.472, NB MOS LQO = 4.532, WB MOS LQO = 4.130 results and found good matching characteristics of audiogram and speech quality. PESQ indicates speech quality rating number of a speech signal between 0.5 and 4.5. The highest PESQ score indicates speech signal don’t have any audible distortions and virtually identical to original input speech signal. If PESQ scores is between 0.5 and 1 means speech signal have very high distortions and residual noise and sound have unacceptable quality. The ratings of 4 is for “good quality” Rating 3 is for “slightly annoying” and Rating 2 considered “annoying” .

Table 1.1

Sr.No.	**Signal Number**	**Filter Order**	**Audiogram Type Number**	**PESQ MOS**	**NB MOS LQO**	**WB MOS LQO**
1	FAKS0	80	N_1	4.480	4.538	4.090
2	FDAC1	80	N_1	4.475	4.534	4.154
3	FELC0	80	N_1	4.470	4.531	3.898
4	FJEM0	80	N_1	4.475	4.534	4.302
5	MDAB0	80	N_1	4.473	4.532	4.007
6	MJSW0	80	N_1	4.482	4.538	4.198
7	MREB0	80	N_1	4.470	4.530	4.204
8	MRJO0	80	N_1	4.453	4.520	4.240
9	MSJS1	80	N_1	4.473	4.532	4.077
	Average Values			4.472	4.532	4.130

Table 1.2

Sr.No.	**Signal Number**	**Filter Order**	**Audiogram Type Number**	**PESQ MOS**	**NB MOS LQO**	**WB MOS LQO**
1	FAKS0	80	N_2	4.453	4.520	3.830
2	FDAC1	80	N_2	4.427	4.503	4.036
3	FELC0	80	N_2	4.406	4.490	3.303
4	FJEM0	80	N_2	4.426	4.502	3.963
5	MDAB0	80	N_2	4.434	4.508	3.635
6	MJSW0	80	N_2	4.455	4.521	4.043
7	MREB0	80	N_2	4.423	4.500	4.113
8	MRJO0	80	N_2	4.394	4.481	4.158
9	MSJS1	80	N_2	4.431	4.506	3.390
	Average Values			4.428	4.503	3.830

Table 1.3

Sr.No.	**Signal Number**	**Filter Order**	**Audiogram Type Number**	**PESQ MOS**	**NB MOS LQO**	**WB MOS LQO**
1	FAKS0	80	N_3	4.453	4.520	4.064
2	FDAC1	80	N_3	4.424	4.501	4.146
3	FELC0	80	N_3	4.428	4.504	3.801
4	FJEM0	80	N_3	4.430	4.505	4.277
5	MDAB0	80	N_3	4.436	4.509	3.985
6	MJSW0	80	N_3	4.453	4.520	4.197
7	MREB0	80	N_3	4.428	4.504	4.185
8	MRJO0	80	N_3	4.395	4.482	4.246
9	MSJS1	80	N_3	4.430	4.505	3.968
	Average Values			4.431	4.506	4.097

Table 1.4

Sr.No.	**Signal Number**	**Filter Order**	**Audiogram Type Number**	**PESQ MOS**	**NB MOS LQO**	**WB MOS LQO**
1	FAKS0	80	N_4	4.483	4.539	4.143
2	FDAC1	80	N_4	4.470	4.530	4.201
3	FELC0	80	N_4	4.475	4.533	3.951
4	FJEM0	80	N_4	4.438	4.510	4.358
5	MDAB0	80	N_4	4.472	4.531	4.055
6	MJSW0	80	N_4	4.476	4.534	4.217
7	MREB0	80	N_4	4.464	4.527	4.277
8	MRJO0	80	N_4	4.435	4.508	4.319
9	MSJS1	80	N_4	4.475	4.534	4.207
	Average Values			4.465	4.527	4.192

Table 1.5

Sr.No.	Signal Number	Filter Order	Audiogram Type Number	PESQ MOS	NB MOS LQO	WB MOS LQO
1	FAKS0	80	N_5	4.470	4.530	4.111
2	FDAC1	80	N_5	4.417	4.497	4.250
3	FELC0	80	N_5	4.474	4.533	3.980
4	FJEM0	80	N_5	4.463	4.526	4.389
5	MDAB0	80	N_5	4.443	4.514	4.071
6	MJSW0	80	N_5	4.453	4.520	4.107
7	MREB0	80	N_5	4.414	4.494	4.361
8	MRJO0	80	N_5	4.395	4.482	4.362
9	MSJS1	80	N_5	4.463	4.526	4.276
	Average Values			4.444	4.514	4.212

Table 1.6

Sr.No.	Signal Number	Filter Order	Audiogram Type Number	PESQ MOS	NB MOS LQO	WB MOS LQO
1	FAKS0	80	N_6	4.461	4.525	4.125
2	FDAC1	80	N_6	4.421	4.499	4.198
3	FELC0	80	N_6	4.440	4.511	3.847
4	FJEM0	80	N_6	4.467	4.529	4.315
5	MDAB0	80	N_6	4.428	4.504	3.988
6	MJSW0	80	N_6	4.449	4.517	4.200
7	MREB0	80	N_6	4.407	4.490	4.261
8	MRJO0	80	N_6	4.382	4.473	4.192
9	MSJS1	80	N_6	4.441	4.512	4.093
	Average Values			4.433	4.507	4.135

Table 1.7

Sr.No.	Signal Number	Filter Order	Audiogram Type Number	PESQ MOS	NB MOS LQO	WB MOS LQO
1	FAKS0	80	N_7	4.467	4.529	4.110
2	FDAC1	80	N_7	4.420	4.499	4.251
3	FELC0	80	N_7	4.464	4.527	3.966
4	FJEM0	80	N_7	4.467	4.528	4.382
5	MDAB0	80	N_7	4.440	4.511	4.043
6	MJSW0	80	N_7	4.452	4.519	4.127
7	MREB0	80	N_7	4.411	4.493	4.354
8	MRJO0	80	N_7	4.393	4.480	4.350
9	MSJS1	80	N_7	4.460	4.524	4.252
	Average Values			4.442	4.512	4.204

5.0 RESULTS DISCUSSION

By observing results in Table 1.8 we found that as we goes on decreasing filter order from 80 to 40 we found improvement in PESQ WB MOS LQO score from 3.830 to 4.035 but audiogram matching to ear response slightly degraded. Below Filter order 40 audiogram matching is not possible, user can change filter parameters by pressing switch multiple times as per his desired audio quality of speech signal in changing listening environments.

we have tested audiogram N_2 to N_7 frequency response parameters on designed filter structure for filter order 80 by giving input test signal from TIMIT database directory DR1 and filter processed audio signal is measured for it quality degradation by using ITU-T PESQ standard as mentioned average PESQ score in Table 1.2 from which all audiograms found best PESQ scores

Table 1.8

Filter Order	Audiogram Type Number	PESQ MOS	NB MOS LQO	WB MOS LQO	Audio Environment
80	N_1	4.472	4.532	4.130	Very Mild
80	N_2	4.428	4.503	3.830	Mild
80	N_3	4.431	4.506	4.097	Moderate
80	N_4	4.465	4.527	4.192	Moderate/severe
80	N_5	4.444	4.514	4.212	Severe
80	N_6	4.433	4.507	4.135	Severe
80	N_7	4.442	4.512	4.204	Profound
Average Values		4.445	4.514	4.114	**4.358**

6.0 CONCLUSION

Designing different subbands for each frequency band and adding gain to each subbands as per audiogram frequency response of hearing aid user requires at least six individual digital FIR filters which leads to more processing time results in delay also increases computational complexity and power dissipation of hearing aid device. Our proposed method requires only one FIR digital filter which includes same gain addition to each frequency band as per audiogram. We have computed single transfer function for complete frequency response as per Audiogram and used coefficients of this transfer function to design a single digital FIR filter with filter order 80 gives perfect audiogram frequency response matching to filter. We have tested this filter with international standard audiograms IEC 60118-15 and tested standard speech signals from TIMIT speech corpus database to check audio quality of these processed signals with ITU-T-PESQ standards, we found our reconfigured filter bank results satisfactory and can use in hearing aid device

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