# Reconfigurable Filter Design and Testing with ISTS Standard for Proposed Hearing Aid Application

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Digital Hearing Aid (DHA) device selectively filters sound signals in subbands. Gain is added as per hearing loss mentioned in audiogram. DHA uses digital filters and amplifies processed signal, and this signal is transferred to the ear. Multiple DHA manufacturing companies all over world have innovative and miniature DHA devices in their product range. They do rigorous research and development to improve product functionalities. Most of them use design method of digital filters using selective amplification by adding gain to subband where patients have hearing loss. Nowadays, DHAs are more customized to individual patient hearing loss characteristics. Most of the available hearing aid designs use filter banks with fixed subbands. The paper focuses on reducing the complexity of the algorithms improving DHA user experience in changing noise and proposes a single reconfigurable transfer function type of digital Finite Impulse Response (FIR) filter to achieve a best fitting to audiogram as per the specifications with IEC 60118-15 standard, and the processed signals are tested with ITU-T-PESQ standards. The paper uses International Speech Test Signal (ISTS) standard speech audio signal to test designed filter, and the results are found to be very satisfactory compared with the fixed filter banks. The paper discusses combined or reconfigured transfer function approach for use in DHA devices and design of reconfigurable FIR filter bank with adjusting different parameters in terms of requirements of DHA.

***Keywords***: Hearing loss, Digital Hearing Aid (DHA), Digital filters, Reconfigurable filter, Perceptual Evaluation of Speech Quality (PESQ), International Speech Test Signal (ISTS)

## Introduction

Digital Hearing Aid (DHA) is very beneficial for people having hearing loss problems. These patients can get hearing benefit using a DHA device. In real world, only 20% of hearing affected patients purchase a hearing aid and around 25% of them do not use DHA due to irritating noise and unpleasant whistles. Some DHAs have processed signal with other amplified noises due to surrounding background noise

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in their daily life (Enrique *et al.*, 2007). Advanced signal processing techniques are used in modern DHA, but only about 50% of patients are happy with the performance of their hearing aids in crowed noisy situations. This problem may be due to the fact that the algorithms used in filter designs have not yet reached the decision levels to which signal is required to preserve and what to block as background noise. DHA restores audiogram gain (amplification) by adding gain in subband frequencies. They make low intensity sounds audible and keep loud sounds comfortable. Hearing aids have been successful in different situations, but patients still have some difficulty in acoustically complex or crowded environments. Amplification and filtering of sound is a basic function of hearing aid. But the 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 a fixed filter called Fixed Wave Decomposition Plan (FWDP) is used in the most hearing-aid systems (Fa-Long and AryeNehorai, 2006). Fixed filter banks cannot provide sufficient flexibility for the adaptability of different hearing impairment cases. Reconfigurable filter banks are generated according to the number of required channels, and control parameters should have a more adaptable hearing aid device as per the user requirements in noisy situations.

The paper considers combined or reconfigured transfer function approach for use in DHA devices and presents the design of reconfigurable filter banks for use in DHAs.

## Literature Review

### Reconfigurable Bands Finite Impulse Response (FIR) Filter Banks

Reconfiguarbility of the proposed filter bank enables deaf individuals to customize hearing aids to support 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. Audiogram is a graph recorded by audiologists with standard pure tone audio frequency signals in the range of 250 to 8 kHz. Signal processing delay of more than 20 ms is unacceptable for any practical hearing aid device. Ying and Debao (2013) mentioned it as an open problem to design reconfigurable filter banks with acceptable processing delay. Adjustable subbands for a personal hearing-loss case with low processing delay are desirable.

#### *Reconfigurable FIR Bank Types*

A reconfigurable digital filter bank can be designed according to the number of channels required and control parameters to get more adaptable DHA as per user requirements in changing noise environments.

Ying and Debao (2013) defined that 'reconfigurable' means adjusting subbands with few control parameters without changing the structure of the filter bank.

Improving 'individuality' of DHA reconfigurable filters iteratively alters their parameters. They minimized the difference between desired output and DHA output. In the case of acoustic noise, the optimal output is a noised signal that accurately emulates the unwanted noise signal.

### *Uniform Filter Banks*

Uniform American National Standards Institute (ANSI) filter bank is mostly not used in hearing aids because they have computation complexity and large group delay (ITU-T P.862).

FIR filter is always stable and possesses a linear phase response if its coefficients are symmetric. Such properties are most welcome by the hearing aid devices due to the requirement of arbitrary magnitude adjustment in the different frequency bands (the DARPA TIMIT). The drawback of an FIR filter is relatively high computational cost due to the involvement of a large amount of multipliers in order to reduce the filter complexity (The DARPA TIMIT).

### *Series Parallel Filter Banks*

Digital signal processing introduces analysis-synthesis delays in the forward path in modern hearing aids. Signal processing is typically performed in a subband or transform domain. Long forward-path delays are not desirable because the processed sound combines with the unprocessed sound. Unprocessed sound arrives at the cochlea through the vent and degrades the sound quality. Subband signal processing is the most popular choice for hearing aids due to its computational simplicity. The paper presents an alternative DHA structure with low-delay characteristics (Chih-Wei *et al.*, 2013). A hearing aid system requires to simulate auditory processes ideally and include those aspects of the speech signal that are perceptually important (Min *et al.*, 2001).

# Methodology

## Reconfigurable Filter Bank Design Process

Six frequency band windows $wn_1$, $wn_2$, $wn_3$, $wn_4$, $wn_5$ and $wn_6$ for decomposing audio signal into frequency range in nonuniform pattern 250-500, 500-1000,1000-2000, 2000-4000, 4000-8000 and 8000-16000 Hz (Yong and Ying, 2005) are designed and the classical method of windowed linear-phase FIR digital filter design is used. MATLAB '$fir_1$' implements this type of digital FIR filters, where filter is normalized so that the magnitude response of the filter at the center frequency of the passband is 0 dB. It designs filter in standard lowpass, highpass, bandpass and bandstop configurations.

$b = fir_1(n, Wn)$ returns row vector $b$ containing the '$n + 1$' coefficients of an order $n$ lowpass FIR filter. This is a Hamming-window based and linear-phase filter with normalized cutoff frequency '$Wn$'. The output filter coefficients '$b$' are ordered in descending powers of '$z$'.

'*Wn*' is a number between 0 and 1, where 1 corresponds to the Nyquist frequency.

If '*Wn*' is a two-element vector, $Wn = [w_1\ w_2]$, *fir*1 returns a bandpass filter with passband $w_1 < \omega < w_2$.

If '$W_n$' is a multi-element vector, $Wn = [w_1\ w_2\ w_3\ w_4\ w_5\ \dots\ wn]$, '$fir_1$' returns an order '*n*' multiband filter with bands $0 < \omega < w_1$, $w_1 < \omega < w_2$, ..., $w_n < \omega < 1$.

By default, the filter is scaled so that the center of the first passband has a magnitude of exactly 1 after windowing.

Using this, we designed six FIR filters as $h_1$, $h_2$, $h_3$, $h_4$, $h_5$ and $h_6$ as:

$h_1 = fir_1(FiltOrder,\ wn_1)$;

$h_2 = fir_1(FiltOrder,\ wn_2)$;

$h_3 = fir_1(FiltOrder,\ wn_3)$;

$h_4 = fir_1(FiltOrder,\ wn_4)$;

$h_5 = fir_1(FiltOrder,\ wn_5)$; and

$h_6 = fir_1(FiltOrder,\ wn_6)$;

The filter function is implemented as a direct form II transposed structure,

$$\begin{aligned} &a(1) * y(n)\ \ b(1) * x(n) + b(2) * x(n-1) + \dots + b(nb+1) * x(n-nb) \\ &- a(2) * y(n-1) - \dots - a(na+1) * y(n-na) \end{aligned} \qquad \dots(1)$$

where '$n - 1$' is the filter order which handles both FIR and IIR filters (Ying and Debao, 2013), '*na*' is the feedback filter order and '*nb*' is the feedforward filter order. Due to normalization, assume $a(1) = 1$.

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

Nyquist frequencies will be from 0 to 1 and adjusted bands as $wn_1$, $wn_2$, $wn_3$, $wn_4$, $wn_5$ and $wn_6$ and we use signal sampling frequency '*Fs*'.

$wn_1 = [250\ 500]/Fs$;

$wn_2 = [500\ 1000]/Fs$;

$wn_3 = [1000\ 2000]/Fs$;

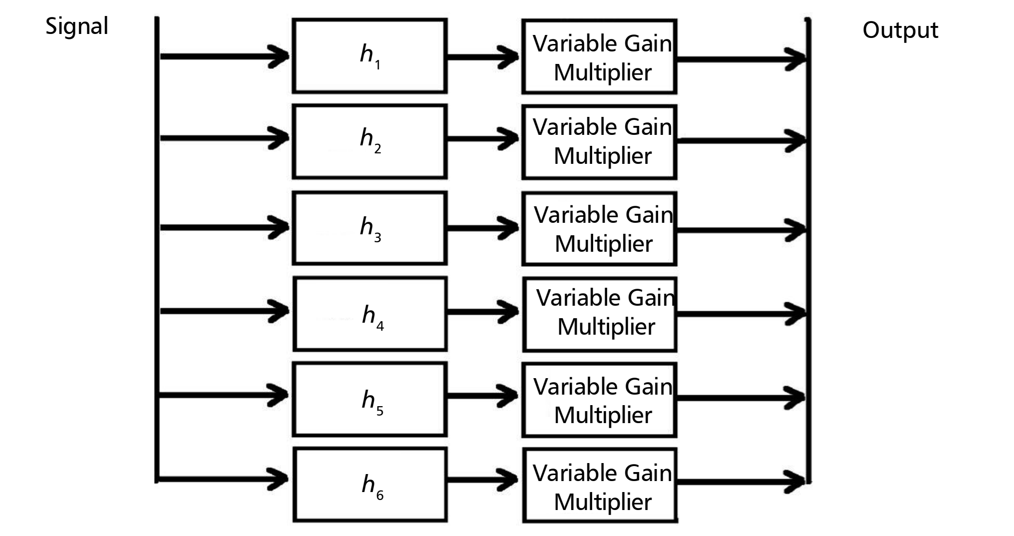
$wn_4 = [2000\ 4000]/Fs$;

$wn_5 = [4000\ 8000]/Fs$; and

$wn_6 = [8000\ 16000]/Fs$;

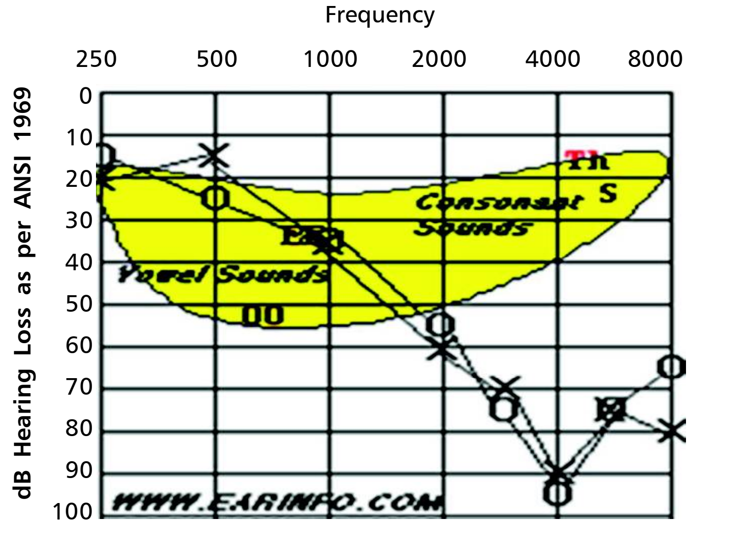
These six filters are connected in parallel and fed into the test signal simultaneously, as shown in Figure 1.

**Figure 1: Subband Signal Processing Filter Bank Structure**




We multiply by a gain factor to increase the amplitude of voice signal in particular frequency band as per patients hearing requirement and to match audiogram. Figure 2 shows the audiogram and frequency response of digital filter bank to this audiogram.

**Figure 2: Audiogram for High Frequency Hearing Loss**




Audiogram is the test taken by hearing aid doctor with pure tone audio frequency signals with a range of 250 to 8 KHz and mark O-zero for left ear and X-cross for right ear on *Y*-axis and mark magnitude/amplitude of hearing sensitivity on *X*-axis.

The audiogram, as shown in Figure 2, is taken from www.Earinfo.com website and this represents hearing problem to patient at high frequency. We matched the frequency response of digital filter bank to the audiogram by adjusting gains to these different bands.

$hd_1 = h_1*20;$

$hd_2 = h_2*15;$

$hd_3 = h_3*35;$

$hd_4 = h_4*60;$

$hd_5 = h_5*90;$ and

$hd_6 = h_6*65$

This filter bank can be used in a traditional way, but its hardware implementation cost is high and requires more area, number of multipliers and delay units to implement on FPGA chip. To get reconfiguarbility of filter bank, we 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.

$tt_1 = tf(hd_1);$

$tt_2 = tf(hd_2);$

$tt_3 = tf(hd_3);$

$tt_4 = tf(hd_4);$

$tt_5 = tf(hd_5);$ and

$tt_6 = tf(hd_6)$

We used *tf* to create real or complex-valued transfer function models (TF objects) or to convert state-space or zero-pole-gain models to transfer function form. We also used *tf* to create generalized state-space (genss) models or uncertain state-space (uss) models. We calculated transfer function of each subband filter with added gain as per audiogram frequency response prescription formula (Figure 3).

After computing each individual transfer function, we combined all these transfer functions into single transfer function using parallel command (Figure 4).

$tt_a = parallel(tt_1, tt_2);$

$tt_b = parallel(tt_3, tt_4);$

$tt_c = parallel(tt_5, tt_6);$

$tt_e = parallel(tt_a, tt_b);$ and

$tt_f = parallel(tt_e, tt_c)$

'$tt_f$' is a final combined transfer function for complete filter bank. Parallel command connects two model objects in parallel. This function accepts any type of model. The two systems must be either both continuous or both

Figure 3: Transfer Function for Subband

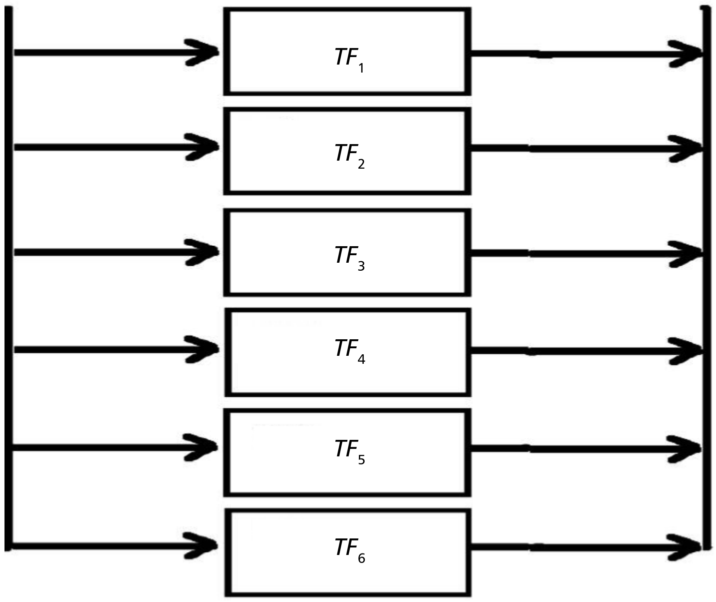
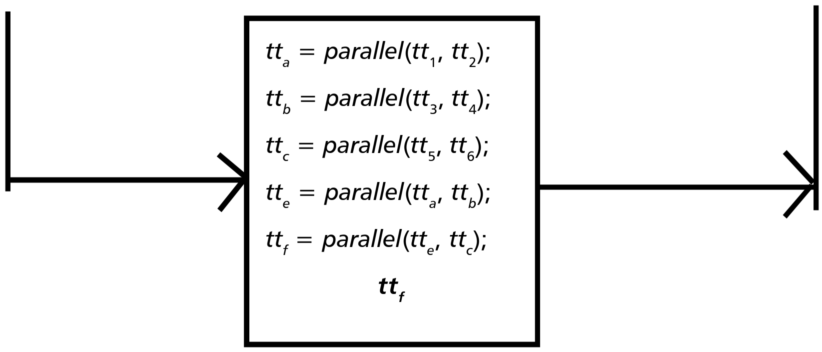



Figure 4: Combined Transfer Function




discrete with identical sample time. Static gains are neutral and can be specified as regular matrices.

We have transfer function '$tt_f$' and its coefficients; these coefficients can be used to build our filter to use in DHA device, so numerator $a_m = 1$ is used, and for calculating the values of denominator $b_m$, *$cell_2mat$* function is used.

$b_m = cell_2mat\,(tt_f\,(1: end).num);$

'$cell_2mat$' function converts cell array into numeric array, once we get '$a_m$' and '$b_m$' values, we can use them to filter test voice signal using filter command.

$out\ 3x = filter\ (b_m, a_m, signal);$

The filter function filters a data sequence using a digital filter, which works for both real and complex inputs. The filter is a direct form II transposed implementation of the standard difference equation.

$y$ = filter ($b, a, X$) filters the data in vector '$X$' with the filter described by numerator coefficient vector '$b$' and denominator coefficient vector '$a$'.

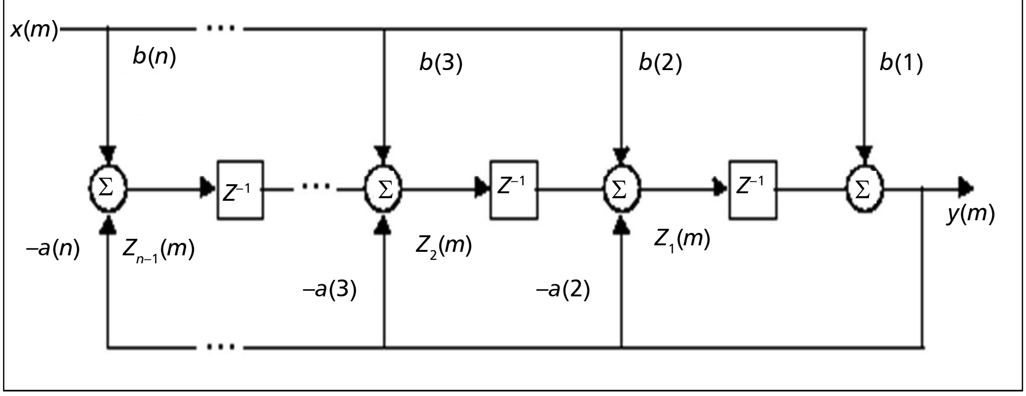
If '$X$' is a matrix, filter operates on the columns of '$X$'. If '$X$' is a multidimensional array, filter operates on the first non-singleton dimension.

The filter function is implemented as a direct form II transposed structure,

$$\begin{aligned}\alpha(1) * y(n) &= b\ (1) * x(n) + b\ (2) * x(n-1) + \ldots + b\ (nb+1) * x(n-nb) \\ &\quad - a\ (2) * y(n-1) - \ \ldots - a(na+1) * y(n-na)\end{aligned} \qquad \ldots(2)$$

where '$n - 1$' is the filter order, which handles both FIR and IIR filters (Ying and Debao, 2013), '$na$' is the feedback filter order and '$nb$' is the feedforward filter order. Due to normalization, assume $a(1) = 1$ (Figure 5).

**Figure 5: FIR Filter Bank**




The operation of filter at sample '$m$' is given by the time domain difference equations.

$$\begin{aligned}&y(m) = b\ (1)\ x\ (m) + z1(m-1) \\ &z1(m) = b(2)\, x(m) + z2(m-1) - a(2)\, y(m) \\ &: \quad = \quad : \quad : \\ &z_{n-2}\ (m) = b(n-1)\, x(m) + z_{n-1}(m-1) - a(n-1)\, y(m) \\ &z_{n-1}\ (m) = b(n)\, x(m) - a(n)\, y(m)\end{aligned} \qquad \ldots(3)$$

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

$$Y(z) = \frac{b(1) + b(2)z^{-1} + \ldots + b(nb+1)z^{-nb}}{1 + a(2)z^{-1} + \ldots + a(na+1)z^{-na}} X(z) \qquad \ldots(4)$$

The designed single transfer function digital filter can be used in DHA with low hardware requirement. Due to less hardware implementation, cost of adders and multipliers results in a low delay filter, with very low power requirement and superior hearing performance than individual subband filters processing digital filter bank.

We tested this filter for the performance with different audiogram standards specified by IEC 60118-15. Table 1 shows six audiogram vectors chosen for the flat and moderately sloping group specified in IEC 60118-15.

**Table 1: Flat and Moderately Sloping Audiograms IEC 60118-15**

| No. | ID | Rank | Category | 250 | 500 | 2.000 | 3.000 | 4.000 | 6.000 |
|---|---|---|---|---|---|---|---|---|---|
| $N_1$ | A4 | 36 | Very Mild | 10 | 10 | 10 | 20 | 35 | 40 |
| $N_2$ | A31 | 6 | Mild | 20 | 20 | 35 | 40 | 50 | 50 |
| $N_3$ | A23 | 2 | Moderate | 35 | 35 | 50 | 55 | 60 | 70 |
| $N_4$ | A48 | 4 | Moderate/Severe | 55 | 55 | 65 | 65 | 75 | 80 |
| $N_5$ | A21 | 19 | Severe | 65 | 70 | 80 | 75 | 80 | 80 |
| $N_6$ | A22 | 16 | Severe | 75 | 80 | 90 | 95 | 100 | 100 |
| $N_7$ | A17 | 34 | Profound | 90 | 95 | 105 | 105 | 105 | 105 |

ISMADHA working group under European Hearing Instrument Manufacturing Association (EHIMA) initiated International Speech Test Signal (ISTS). The ISTS test signal comprises natural recordings. This signal is segmented and remixed, hence it is largely non-intelligible. This test signal was promoted by EHIMA with a new measurement method for hearing aid standard.

ISTS test signal is designed for analyzing the process of speech by a hearing aid. This standard test signal is necessary which can be used for reproducible measurement conditions in hearing aid device applications. It has all features and properties of natural speech. Modulation spectrum and the fundamental frequency properties and harmonics are well included to test hearing aid device for all types of speech. Artificial signals have limitations and some inadequately represent recordings from natural speakers in only one language and are therefore not internationally applicable.

ISTS recording includes speech from female speakers from six different mother tongues, viz., American English, Chinese, Arabic, German, French and Spanish, of reading sentence "The north wind and the sun".

ISTS is designed as per LTASS (Long-Term Average Speech Spectrum) standards. We tested our designed reconfigured filter for the above-mentioned six IEC 60118-15 standard audiograms by passing ISTS speech signal and checked quality of speech processed signal by filter using PESQ (Perceptual Evaluation of Speech Quality) standard. PESQ is an objective method for end-to-end speech quality assessment of narrow and wide band telephone networks and speech codecs and is a standard method for objective and subjective assessment of quality (Holube *et al.*, 2010).

The perceptual model of PESQ is used to calculate a distance between the original and degraded speech signal (PESQ score) (Ashutosh and John, 2011). The PESQ score is mapped to a Mean Opinion Score (MOS) like scale, a single number in the range of –0.5 to 4.5, although for most cases, the output range is between 1.0 and 4.5, the normal range of MOS values is found in an ACR listening quality experiment. 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 (ITU-T P.862).

## Results and Discussion

Tables 2 to 8 show audiogram and WB MOS LQO-PESQ scores for ISTS speech signal.

**Table 2: PESQ Scores for Audiogram $N_1$**

| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
|---|---|---|---|
| 10 | $N_1$ | 2.529 | Very Mild |
| 20 | $N_1$ | 1.322 | Very Mild |
| 40 | $N_1$ | 4.154 | Very Mild |
| 80 | $N_1$ | 3.674 | Very Mild |
| 100 | $N_1$ | 3.652 | Very Mild |
| 150 | $N_1$ | 3.779 | Very Mild |
| 200 | $N_1$ | 3.405 | Very Mild |

**Table 3: PESQ Scores for Audiogram $N_2$**

| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
|---|---|---|---|
| 10 | $N_2$ | 4.221 | Mild |
| 20 | $N_2$ | 4.186 | Mild |
| 40 | $N_2$ | 3.973 | Mild |
| 80 | $N_2$ | 4.066 | Mild |
| 100 | $N_2$ | 4.236 | Mild |
| 150 | $N_2$ | 4.224 | Mild |
| 200 | $N_2$ | 4.093 | Mild |

| Table 4: PESQ Scores for Audiogram $N_3$ | | | |
|---|---|---|---|
| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
| 10 | $N_3$ | 3.709 | Moderate |
| 20 | $N_3$ | 3.894 | Moderate |
| 40 | $N_3$ | 3.829 | Moderate |
| 80 | $N_3$ | 4.011 | Moderate |
| 100 | $N_3$ | 4.109 | Moderate |
| 150 | $N_3$ | 4.152 | Moderate |
| 200 | $N_3$ | 4.000 | Moderate |

| Table 5: PESQ Scores for Audiogram $N_4$ | | | |
|---|---|---|---|
| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
| 10 | $N_4$ | 4.328 | Moderate/Severe |
| 20 | $N_4$ | 4.186 | Moderate/Severe |
| 40 | $N_4$ | 3.754 | Moderate/Severe |
| 80 | $N_4$ | 3.697 | Moderate/Severe |
| 100 | $N_4$ | 3.857 | Moderate/Severe |
| 150 | $N_4$ | 1.142 | Moderate/Severe |
| 200 | $N_4$ | 3.879 | Moderate/Severe |

| Table 6: PESQ Scores for Audiogram $N_5$ | | | |
|---|---|---|---|
| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
| 10 | $N_5$ | 4.446 | Severe |
| 20 | $N_5$ | 3.876 | Severe |
| 40 | $N_5$ | 4.061 | Severe |
| 80 | $N_5$ | 4.170 | Severe |
| 100 | $N_5$ | 4.366 | Severe |
| 150 | $N_5$ | 2.096 | Severe |
| 200 | $N_5$ | 4.247 | Severe |

| Table 7: PESQ Scores for Audiogram $N_6$ | | | |
|---|---|---|---|
| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
| 10 | $N_6$ | 3.797 | Severe |
| 20 | $N_6$ | 3.949 | Severe |
| 40 | $N_6$ | 4.081 | Severe |
| 80 | $N_6$ | 4.208 | Severe |
| 100 | $N_6$ | 4.307 | Severe |
| 150 | $N_6$ | NA | Severe |
| 200 | $N_6$ | 4.175 | Severe |

| Table 8: PESQ Scores for Audiogram $N_7$ | | | |
|---|---|---|---|
| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
| 10 | $N_7$ | NA | Profound |
| 20 | $N_7$ | NA | Profound |
| 40 | $N_7$ | NA | Profound |
| 80 | $N_7$ | 4.191 | Profound |
| 100 | $N_7$ | NA | Profound |
| 150 | $N_7$ | NA | Profound |
| 200 | $N_7$ | NA | Profound |

We tested our reconfigured filter structure system with ISTS signal on audiogram '$N_1$' which represents 'very mild' hearing loss for patient, as mentioned in IEC 60118-15 standard (Nikolai *et al*., 2010).

Filter is tested with filter order 10 to 200 for FIR filter design and output speech signal is tested on PESQ standard. We found PESQ-WB MOS LQO values varying between 2.529 and 4.154, as given in Table 2, and found maximum PESQ score 4.154 with filter order 40.

Reconfigured filter performance with ISTS signal on audiogram '$N_2$' which represents 'mild' hearing loss is tested with filter order 10 to 200 for FIR filter design and found PESQ-WB MOS LQO values varying between 3.973 and 4.236, as given in Table 3. We found maximum PESQ score 4.236 with filter order 100.

Audiogram '$N_3$' as 'moderate' hearing loss as per IEC 60118-15 standard is tested (Nikolai *et al*., 2010) with different filter orders, as given in Table 4 for FIR filter design, and found PESQ-WB MOS LQO values varying between 3.709 and 4.152. We found maximum PESQ score 4.152 with filter order 150.

When ISTS signal is passed on reconfigured digital FIR filter with audiogram '$N_4$' which represents 'moderate/severe' hearing loss and tested with filter order 10 to

200 for FIR filter design, the PESQ-WB MOS LQO values varied between 1.142 and 4.328, as given in Table 5. We found maximum PESQ score 4.328 with filter order 10.

Audiogram '$N_5$' as 'severe' hearing loss as per IEC 60118-15 standard is tested with different filter orders, as given in Table 6 for FIR filter design and found PESQ-WB MOS LQO values varying between 2.096 and 4.446. We found maximum PESQ score 4.446 with filter order 10.

Reconfigured filter performance with ISTS signal on audiogram '$N_6$' which represents 'severe' hearing loss tested with filter order 10 to 200 for FIR filter design and found PESQ-WB MOS LQO values varying between 3.797 and 4.307 as given in Table 7. We found maximum PESQ score 4.307 with filter order 100.

Audiogram '$N_7$' as 'profound' hearing loss as per IEC 60118-15 standard is tested with different filter orders but found PESQ-WB MOS LQO value only for filter order 80 as 4.191 (Table 8).

We summarized all PESQ WB MOS LQO values for all audiograms $N_1$ to $N_7$ for filter order 10 to 200 as given in Table 9 and calculated average values for PESQ scores. We found good audiogram matching to all types of audiograms with filter order 80.

**Table 9: PESQ WB MOS LQO Values for All Audiograms $N_1$ to $N_7$**

| Filter Order | $N_1$ | $N_2$ | $N_3$ | $N_4$ | $N_5$ | $N_6$ | $N_7$ | PESQ WB MOS LQO Average |
|---|---|---|---|---|---|---|---|---|
| 10 | 2.529 | 4.221 | 3.709 | 4.328 | 4.446 | 3.797 | 0 | 3.290 |
| 20 | 1.322 | 4.186 | 3.894 | 4.186 | 3.876 | 3.949 | 0 | 3.059 |
| 40 | 4.154 | 3.973 | 3.829 | 3.754 | 4.061 | 4.081 | 0 | 3.407 |
| 80 | 3.674 | 4.066 | 4.011 | 3.697 | 4.17 | 4.208 | 4.191 | 4.002 |
| 100 | 3.652 | 4.236 | 4.109 | 3.857 | 4.366 | 4.307 | 0 | 4.080 |
| 150 | 3.779 | 4.224 | 4.152 | 1.142 | 2.096 | 0 | 0 | 2.199 |
| 200 | 3.405 | 4.093 | 4.0 | 3.879 | 4.247 | 4.175 | 0 | 3.399 |

## Performance Parameters of Filter Banks

Table 3 shows PESQ scores for audiogram $N_2$ with our designed filter with filter order 10 to 200. We tested this filter structure by giving ISTS test signal, and filter processed audio signal is measured for its quality degradation using ITU-T PESQ standard. We calculated average values of WB MOS LQO = 4.142 and found good matching characteristics of audiogram and speech quality at filter order 80 with WB MOSLQO score value of 4.066. PESQ provides perceptual quality rating of a speech signal between 0.5 and 4.5. The highest PESQ score indicates speech signal contains no audible distortions and virtually identical to original input speech signal. The PESQ scores between 0.5 and 1 indicate that the distortions and residual noise in the speech signal are very high and the segment sound unacceptably annoying. The ratings of 4, 3 and 2 can be considered as 'good quality', 'slightly annoying' and 'annoying', respectively (ITU-T P.862).

Table 10 shows audiogram $N_1$ to $N_7$ frequency response parameters on designed filter structure for filter order 80 by giving input test signal from ISTS test signal. Average PESQ score is 4.002 which is acceptable for hearing aid application.

**Table 10: PESQ Scores for Filter Order 80**

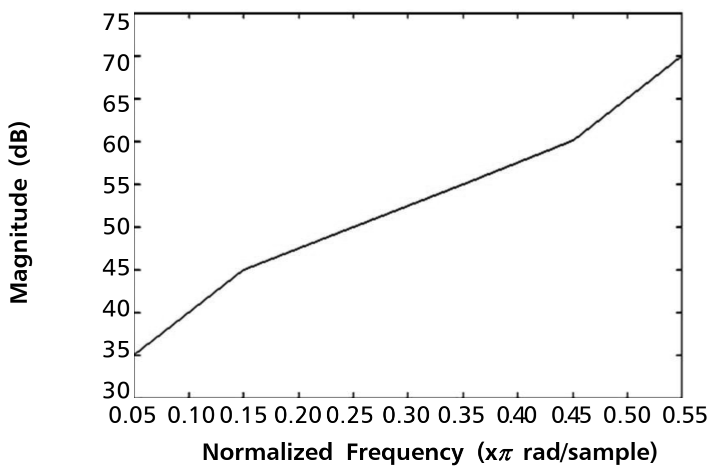
| Filter Order | Audiogram Type Number | WB MOS LQO | Audio Environment |
|---|---|---|---|
| 80 | $N_1$ | 3.674 | Very Mild |
| 80 | $N_2$ | 4.066 | Mild |
| 80 | $N_3$ | 4.011 | Moderate |
| 80 | $N_4$ | 3.697 | Moderate/Severe |
| 80 | $N_5$ | 4.17 | Severe |
| 80 | $N_6$ | 4.208 | Severe |
| 80 | $N_7$ | 4.191 | Profound |
| Average Values | | 4.002 | |

## Reconfiguration Results As Per IEC Standards

$N_3$ audiogram is considered as IEC standard for moderate hearing loss, and we used this audiogram as an example of reconfiguration.

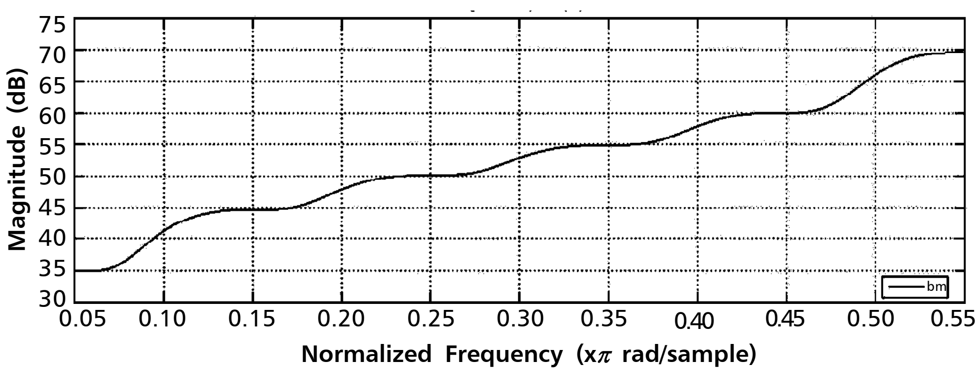
Figure 6 shows normalized frequency versus magnitude in dB for ideal audiogram response given in Table 4 as per IEC standard for $N_3$, we have taken parameters from standards and applied gain to our filter.

**Figure 6: Ideal Audiogram for $N_3$**



We applied the above-mentioned parameters to our reconfigurable filter and achieved audiogram frequency response for best matching with ideal audiogram of filter order 80, as shown in Figure 7.

**Figure 7: Achieved Audiogram for $N_3$ for Filter Order 80**




To get reconfiguarbility of our filter in different listening environment of hearing aid user, we kept filter bank structure same, but changed filter orders from 40 to 80 and checked filters performance by measuring PESQ score for ISTS test signal (Figure 8). We observed good matching of ear response for filter order 80 but WB MOS LQO score was as 4.011. To reconfigure filter parameter for audiogram $N_3$, we changed filter order from 40 to 80 in steps of 10 and checked PESQ scores. Frequency response for different order filters for $N_3$ audiogram is plotted in Figures 9 to 12.

**Figure 8: Filter Bank with Reconfigurable Filters**

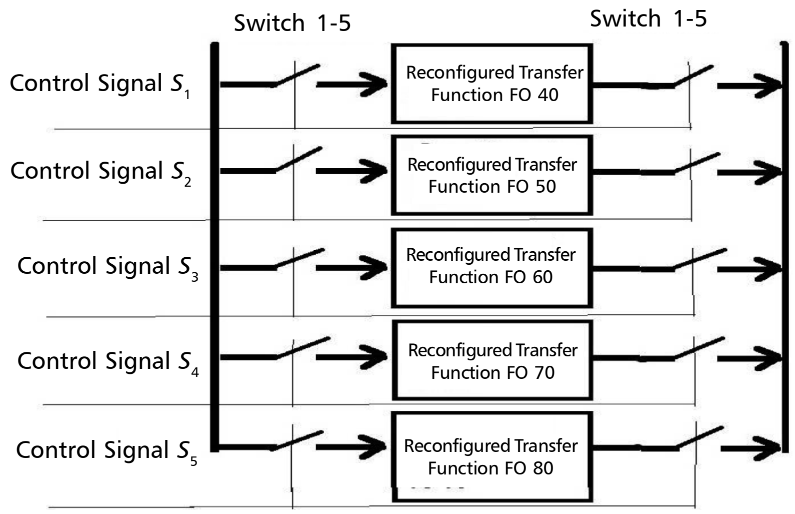


Figure 9: Achieved Audiogram for $N_3$ for Filter Order 70

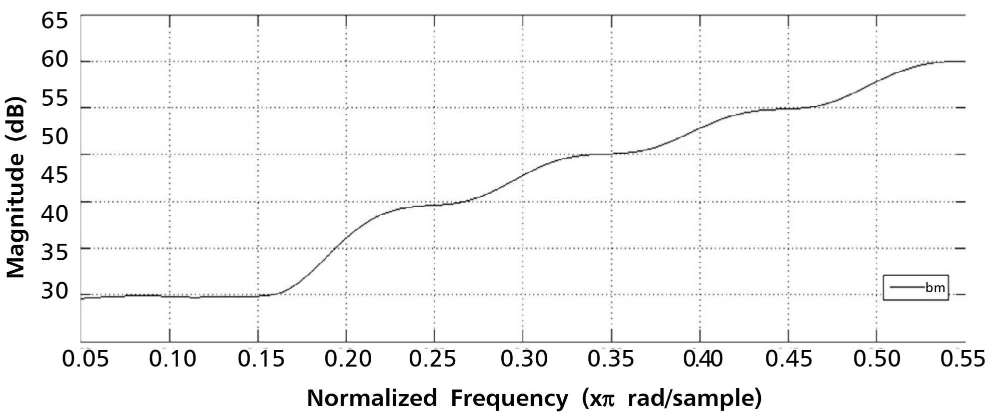



Figure 10: Achieved Audiogram for $N_3$ for Filter Order 60

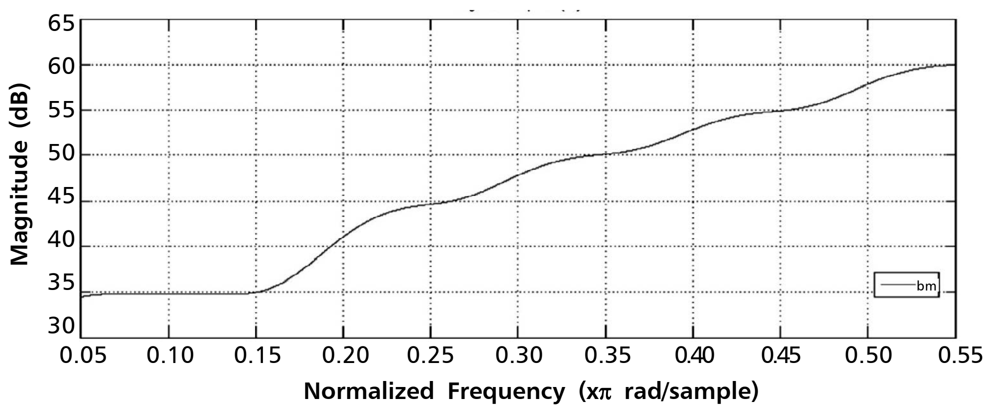



Figure 11: Achieved Audiogram for $N_3$ for Filter Order 50

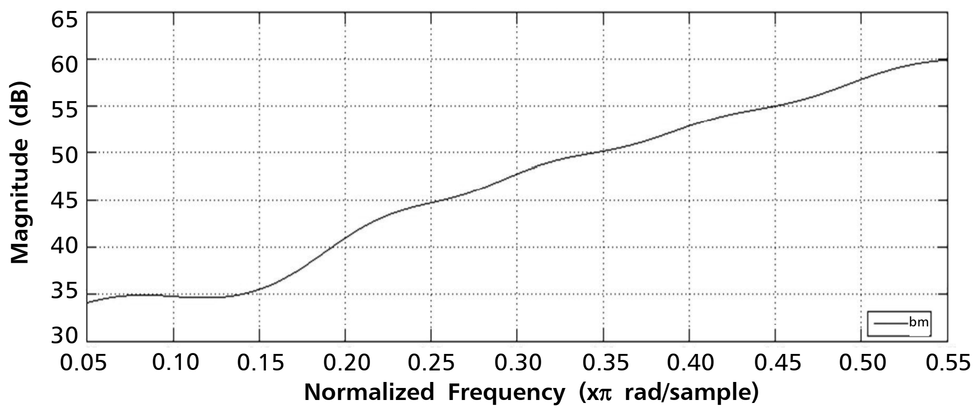
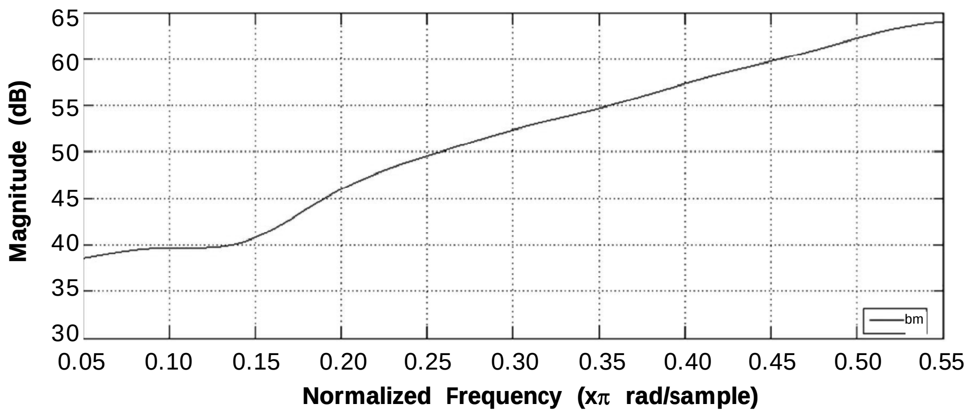


Figure 12: Achieved Audiogram for $N_3$ for Filter Order 40




Control signal controls two switches in a branch and connects reconfigured filter to input and output. This reconfiguration is achieved by only one filter and we do not have to change filter structure in real time. This gives user a choice of clear sound hearing in noisy situations and different types of listening environments. Table 11 shows PESQ score improvement by changing filter order and without much compromising filter response matching with ear response.

**Table 11: PESQ Score Improvement with Filter Order**

| Filter Order | Audiogram Type Number | PESQ WB MOS LQO | Audio Environment |
|---|---|---|---|
| 40 | $N_3$ | 3.829 | Moderate |
| 50 | $N_3$ | 3.873 | Moderate |
| 60 | $N_3$ | 3.888 | Moderate |
| 70 | $N_3$ | 3.940 | Moderate |
| 80 | $N_3$ | 4.011 | Moderate |

## Conclusion

Different subbands for each frequency band with gain addition to each subband as per audiogram frequency response of hearing aid user require at least six individual digital FIR filters. This leads to more computational complexity and power dissipation of hearing aid device. The proposed method requires only one FIR digital filter which includes same gain addition to each frequency band as per patient hearing loss curve as per audiogram. Single transfer function for complete frequency response as per audiogram can be designed. We can use these coefficients to design a single digital FIR filter and filter order 80 gives perfect audiogram frequency response matching to filter. We have tested this filter with International Standard audiograms IEC 60118-15, and standard speech signal is tested with ISTS standards and ITU-T-PESQ standards to check the audio quality of these processed signals. We found that our reconfigured filter bank results are satisfactory and can be used in hearing aid device.

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