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Multiple Bandwidth FIR Filter Design with Adaptive Algorithms for Hearing Aid Systems

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Digital Filter design plays a vital role in signal processing and communication applications. This paper proposes a hearing loss system model with Variable Bandwidth FIR Filter (VBF) and adaptive algorithms for the application to listening. The tunable band filter is designed to provide an appropriate sound level. This filter has several sub-filters each of which is designed with set of selected bandwidths. The sub-bands obtained are adjusted with proper magnitude by trial and error method. Algorithms such as Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and Recursive Least Squares (RLS) are incorporated to improve the quality of the signal. The filter thus designed is examined by taking a number of audio signals. The tests on various hearing loss cases with different type of input signal suggest that this method is capable of reproducing a signal which sounds exactly the same as the original signal. The multiple bandwidth filters is analyzed with mild, moderate, profound and severe hearing loss patterns and the results are reported. The matching error is calculated between ideal response and actual response. The result show that the designed filter provides acceptable minimum matching error and it lies in the range 0 to 2.5dB. This filter design is implemented in TMS320C6711 processor and is tested for sinusoidal input signal.

Keywords: FIR filter, bandwidth, windowing, hearing losses modelling, LMS, NLMS, RLS

1 Introduction

Digital signal processing (DSP) techniques are popular in handling different types of information. The applications of DSP include sub-band coding of speech, audio and video signals, Digital audio tape, half band filtering, multicarrier data transmission, trans-multiplexing, sub-band coding, etc. Fractional delay filters find applications in telecommunications, sampling rate conversion from arbitrary sampling rates, speech signal processing, digital audio and modelling the musical instruments¹⁻⁵. Hearing is very important senses for survival and is needed for communication. Hearing aids are essential to enrich the hearing quality. Hearing aids in general give an improvement in hearing abilities and are normally used to compensate for hearing-loss. A hearing aid must cover the dynamic range of the impaired ear. Some sounds may not be audible and are not properly identified. The range of levels between the weakest sound and the strong sound that can be acceptable is less for a hearing impaired person than normal person⁶⁻⁷. A hearing aid must strengthen the required input signal to a desired level

so that the hearing impaired can hear with substantial quality. Hearing loss problems can be analyzed with an audiogram. An optimal listening device must provide less power, minimal delay and variable magnitude response and also it should have compact in size. The device must give constant delay in order to avoid noise. Digital Signal processing techniques are the appropriate choice for getting these improvements. The hearing aid mainly focusing on various bands that can be used for matching sound level of a particular user. These frequency bands are obtained using variable bandwidth FIR Filters.

Figure 1 shows the basic diagram of a listening aid; include A/D converter, DSP and D/A convertor. The heart of the hearing aid design is with DSP which focuses many functions which include compensating listening level, minimizing distortion, voice quality improvement. In DSP, hearing loss is compensated by

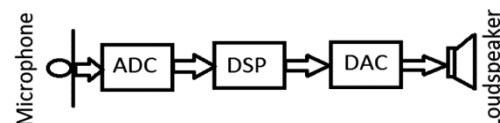


Fig. 1 — Basic Diagram

exploiting auditory compensation which is the major functionality used in hearing aid. The magnitude responses of designed filters are adjusted to compensate the poor listening ability. In this paper, non-uniform spaced sub bands are obtained with the Farrow structure based on variable bandwidth filters (VBW). Variable bandwidth FIR Filter is used to obtain uniform and non-uniform filters bands that can be used for matching the audiogram of a hearing aid. Here the filter design involves two arbitrary sampling rate conversion system and one fixed low pass FIR filter⁸. Three channel variable filter bank is designed and the output responses are used to match the audiogram of different hearing loss patterns⁹. The signal processing section of the listening device incorporates variable filter bank design with frequency response masking and half band filters with reduction in computational cost^{10, 11}. The adjustable filter bank involves frequency response masking technique with three filters and the design yield less complexity¹². Appropriate magnitude and shifting is done to get the magnitude response that matches the audiogram of a hearing aid. This is achieved by designing a Farrow structure based variable bandwidth filter¹³. Modified Farrow structure is implemented with different order sub-filters which minimize the maximum error¹⁴. A continuously variable digital delay element is used for interpolator design. This design finds applications in echo cancellation, data transfer from one carrier system to another carrier system¹⁵. The fraction delay filter part employs delay parameter that vary with time can be used to fine tune the required sampling rates¹⁶. Polynomial based fractional delay filter gives optimum amplitude and phase responses¹⁷. Delay suffered by the group of frequencies is called group delay. An FIR filter having linear phase and unconditionally stable characteristics. The group delay must be constant to achieve linear phase response. Linear phase filter structures have been employed to minimize round-off noise and to minimize complexity¹⁸. Linear programming based adjustable bandwidth filters have been designed to minimize the approximation error¹⁹. Polynomial approximation based fractional delay filters used to minimize the error. This error is significant when Taylor series approximation is considered. The Sum of powers of two coefficients is used to design an efficient Farrow structure²⁰. Genetic algorithm based canonic signed digit coefficients are used to design a trans-multiplexing system which can be exploited in multicarrier communication²¹. Farrow based approach

offers efficient design of filters²⁷. Deep Neural network technique offers good improvement in speech quality of smart hearing aid systems²⁸.

2. Hearing Loss System Model with VBF and Adaptive algorithms

In this section, FIR filter design, variable bandwidth filter and the hearing loss modeling with LMS, RLS and NLMS algorithms are discussed.

2.1 FIR Filter Design

FIR filters are unconditionally stable and has linear phase characteristics. The FIR filter design methods include windowing, frequency sampling and minimax design. The IIR (infinite impulse response) is truncated at finite number of points to get FIR which looks like passing the IIR sequence through a window of length 0 to N-1 called windowing^{2, 22-25}.

Similarly in frequency sampling, the frequency response of an FIR filter can be represented by

$$H_d(\omega) = \sum_{n=0}^{N-1} h(n) e^{-j\omega n}. \text{ The frequency sampling}$$

approach involves set of equal spaced samples which is given by:

$$w_k = \frac{2\pi}{N}(k + \alpha) \quad k = 0 \text{ to } \frac{N-1}{2}$$

$$\text{for } N = \text{odd}; k = 0 \text{ to } \frac{N}{2} - 1 \quad \text{for}$$

$$N = \text{even}; \alpha = 0 \text{ or } \frac{1}{2}$$

$$H(k + \alpha) \equiv H\left(\frac{2\pi}{N}(k + \alpha)\right)$$

hence the desired frequency response is specified as a set of equal spaced frequencies and is given in equation (1).

$$H(k + \alpha) \equiv \sum_{n=0}^{N-1} h(n) e^{-j\frac{2\pi}{N}(k+\alpha)n} \quad \text{for } k = 0 \rightarrow N-1 \quad \dots (1)$$

The filter coefficients are obtained by taking inverse transform which is given in equation (2)

$$h(n) \equiv \frac{1}{N} \sum_{k=0}^{N-1} H(k + \alpha) e^{j\frac{2\pi}{N}(k+\alpha)n} \quad \text{for } n=0 \rightarrow N-1 \quad \dots (2)$$

The basic idea behind the optimal method (minimax design) is to compute the filter coefficients

using iterative procedure so that error present in the filter response is comes to a lesser value. Parks-McClellan algorithm finds the unique set of filter coefficients that minimize the error function. The optimal design method yields equiripple solution by spreading the error over the pass band and stop band of the filter, but the design requires a very large amount of computation.

2.2 Design of Variable Bandwidth Filter

The variable delay filters are implements using Farrow structure. The important advantage of Farrow structure is that, it has fixed filter coefficients and a variable fractional delay parameter called μ . In DSP, especially, variable bandwidth filter plays a vital role in improving the hearing losses. The variable bandwidth FIR filter design employs several sub-filters with selected bandwidth. The response of FIR filter is given by the equation (3).

$$H(z, \mu) = \sum_{n=0}^L a_n(\mu) Z^{-n} \quad \dots (3)$$

where $\alpha_n(\mu)$ refers to the polynomial function in μ of degree M. This polynomial function is given in equation (4). $a_n(\mu) = \sum_{k=0}^M a_{nk} \mu^k \quad \dots (4)$

By using equation (3) and (4), the transfer function can be rewritten as in equation (5)

$$H(z, \mu) = \sum_{k=0}^M \sum_{n=0}^L a_{nk} z^{-n} \mu^k = \sum_{k=0}^M S_k(z) \mu^k, \text{ where}$$

$$S_k(z) = \sum_{n=0}^L a_{nk} z^{-n} \quad \dots (5)$$

For a variable bandwidth filter design, the fractional delay parameter is replaces with bandwidth which will vary for each and every filter to get uniform or non-uniform bandwidth. The variable bandwidth filter design is used to obtain the audiogram matching on various type of hearing losses. Sample sound level statistics is used for comparing the filter design⁶⁻⁷. These are given in Fig. 2. Using the variable bandwidth filter, a bank of digital filters is to be designed to match each of the audiograms of Fig. 2. Optimum sub-band bandwidths for matching these audiograms are decided by simulating them individually for getting minimum matching error.

2.2.1 Design Example

Consider 8-bands of filters to be used. The optimum transition bandwidth for the chosen set of audiograms is obtained by trial and error, as 311Hz. A set of 8 different bandwidths is to be obtained using the variable bandwidth filter. Where the VBF is a Type 1 low pass filter with order 301, with varying band edges. The bandwidths, frequency shifts and gains assigned to each and every band of the VBW filter, to match these audiograms, are as given in Table 1 to Table 4.

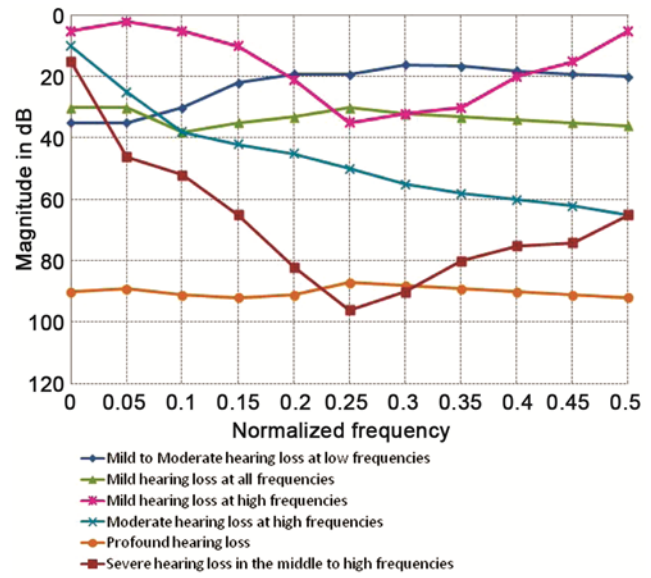


Fig. 2 — Sample Audiograms

Table 1 — Design parameters of VBF for Mild Hearing Loss

Band	Frequency Range (Hz)	Bandwidth ((Hz)	Gain (dB)	Shift (Normalized)
1	up to 1050	1500	30	0
2	800 – 2300	1500	38	0.0968750
3	2175 – 3175	1000	35	0.1640625
4	2975 – 3975	1000	33	0.2140625
5	3775 – 4775	1000	30	0.2640625
6	4550 – 6050	1500	33	0.3421875
7	5800 – 7300	1500	34	0.4203125
8	7050 – 8650	1500	36	0.4984375

Table 2 — Design parameters of VBF for Moderate Hearing Loss

Band	Frequency Range (Hz)	Bandwidth (Hz)	Gain (dB)	Shift (Normalized)
1	up to 775	1000	58	0
2	525 – 1525	1000	55	0.0640625
3	1275 – 2275	1000	60	0.1281250
4	1970 – 4470	2500	60	0.2075000
5	4275 – 5275	1000	61	0.2715625
6	5025 – 6025	1000	62	0.3356250
7	5600 – 7100	1500	63	0.4143750
8	6650 – 8150	1500	65	0.4931250

2.3 Hearing Loss System Model

Figure 3 shows the diagram of the proposed adjustable band filter with LMS, NLMS, and RLS algorithms²⁶ for hearing aid. It describes how a signal which a hearing impaired hears is modeled followed by how the same signal is processed to make the hearing impaired hear well. A signal which a person with normal hearing ability can hear is taken. This signal is split into different frequency bands and to each band an attenuation factors (a_n , where n varies from 1 to 8) is given selectively

Table 3 — Design parameters of VBF for Severe Hearing Loss

Band	Frequency Range (Hz)	Bandwidth (Hz)	Gain (dB)	Shift (Normalized)
1	up to 1100	1500	14	0
2	750 - 2250	1500	51	0.0937500
3	1975 - 2975	1000	65	0.1609375
4	2675 - 3675	1000	82	0.2281250
5	3375 - 4375	1000	86	0.2953125
6	4125 - 6025	1900	89	0.3781250
7	5625 - 7525	1900	75	0.4609375
8	7125 - 8625	1500	65	0.5046875

Table 4 — Design parameters of VBF for Profound Hearing Loss

Band	Frequency Range (Hz)	Bandwidth (Hz)	Gain (dB)	Shift (Normalized)
1	up to 1000	1500	90	0
2	750 - 2250	1500	90	0.0937500
3	1975 - 2975	1000	92	0.1609375
4	2675 - 3675	1000	91	0.2046875
5	3375 - 4375	1000	87	0.2484375
6	4125 - 6025	1900	89	0.3312500
7	5750 - 7250	1500	90	0.4250000
8	7000 - 8500	1500	92	0.5027500

depending on the degree of hearing loss. The different bands with different attenuation factor combine to give a signal, which is nothing but the signal which a person with a hearing loss hears. This signal is termed as “simulated signal” or “modeled signal”. This modeled signal is the one which is supposed to be boosted up.

As next step, the modeled signal is taken as input to the eight band variable bandwidth filter where the signal is again divided into eight different sub bands and to each band different gain factor (g_n , where n varies from 1 to 8) is provided. While designing the filter, the eight different frequency bands, the gain that is to be given to every band is found by iterative procedure until the response of the variable bandwidth filter matches the audiogram of the hearing impaired. Thus the different frequency ranges are selectively amplified so that the eight sub-bands mix up together to produce a resultant signal which is in close accordance with the original signal. This is how the modeled signal is amplified to make the signal audible to the hearing impaired. Finally on this amplified signal, noise cancellation algorithms are applied to result in a noise free signal.

2.3.1 Structure for Mild Hearing Loss

Figure 4 illustrates the diagram of the variable bandwidth filter with noise cancellation algorithms designed for mild hearing loss, the sound level of which is given in Fig. 2. The eight different frequency ranges and the attenuation factor and the gain factor for each of these eight bands are as specified in Fig. 4.

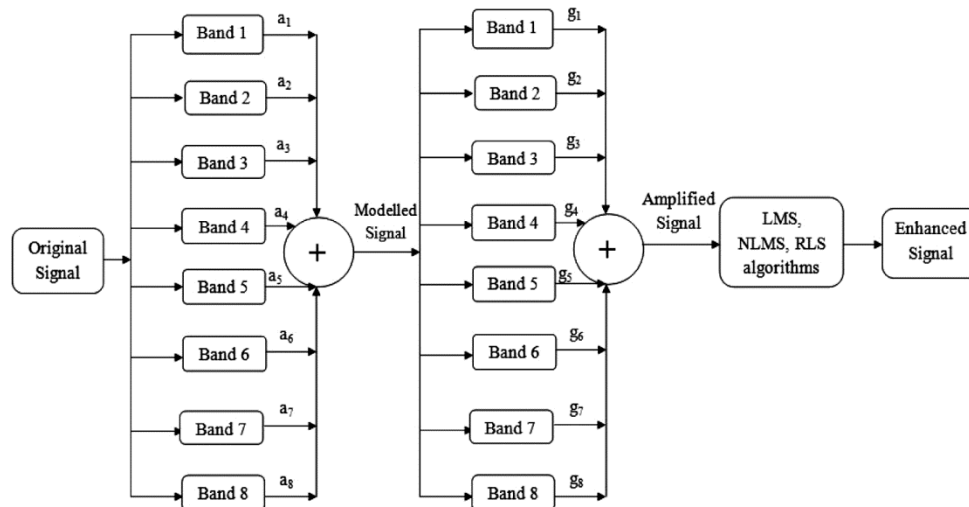


Fig. 3 — Structure of Proposed System

2.3.2 Structure for Moderate Hearing Loss

Figure 5 shows the diagram of the variable bandwidth filter with noise cancellation algorithms designed for moderate hearing loss, whose audiogram

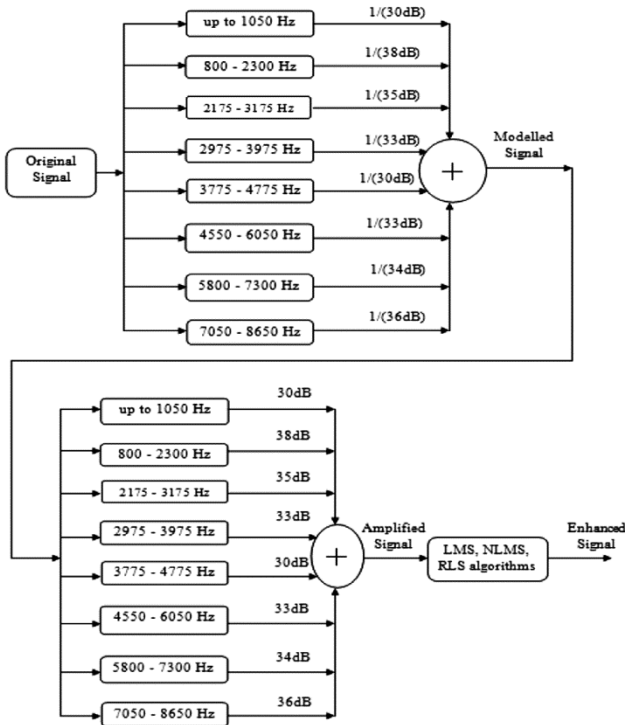


Fig. 4 — Structure for Mild hearing loss Modelling

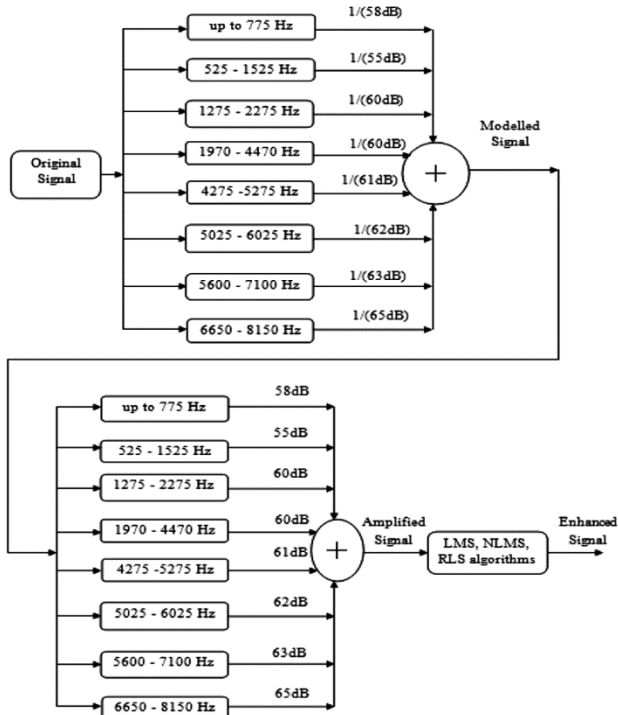


Fig. 5 — Structure for Moderate hearing loss Modelling

is given in Fig. 2. The eight different frequency ranges and the attenuation factor and the gain factor for each of these eight bands are as specified in Fig. 5.

2.3.3 Structure for Severe Hearing Loss

Figure 6 shows the block diagram of variable bandwidth filter with noise cancellation algorithms designed for severe hearing loss, the audiogram of which is shown in Fig. 2. The eight different frequency ranges and the attenuation factor and the gain factor for each of these eight bands are as specified in Fig. 6.

2.3.4 Structure for Profound Hearing Loss

Figure 7 shows the block diagram of the variable bandwidth filter with noise cancellation algorithms designed for profound hearing loss, whose audiogram is depicted in Fig. 2. The eight different frequency ranges and the attenuation factor and the gain factor for each of these eight bands are as specified in the Fig. 7.

2.3.5 Design Parameters

In this work, eighteen different signals are considered for comparing the characteristics of the proposed variable bandwidth filter design followed by adaptive noise cancellation techniques for mild, moderate, severe and profound hearing losses.

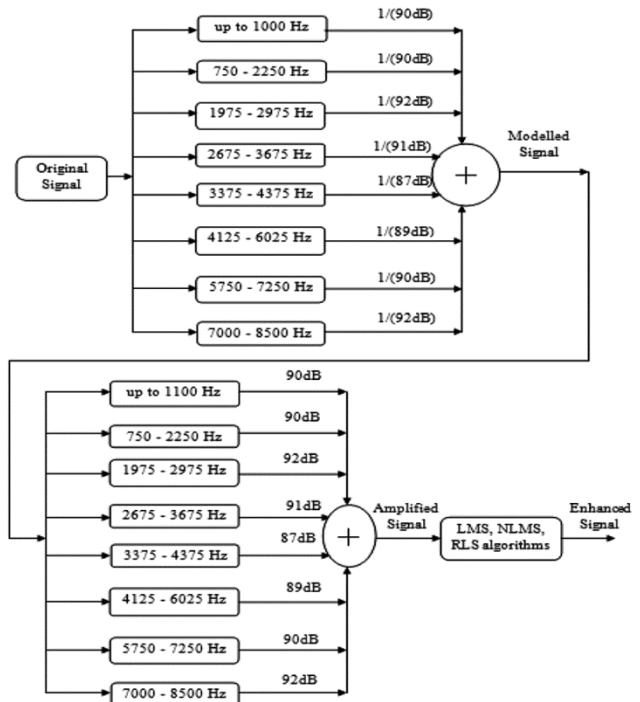


Fig. 6 — Structure for Severe hearing loss Modelling

Various signals such as dialogues in a normal conversation, announcement in a railway station, conversation taking place in a restaurant, sound of alps, Beethoven musical piece, sounds of instruments like clarinet and piano, singing of rhymes by kids, sounds of birds, ducks and frogs, industrial sound, telephone ring and sounds generated in road traffic are chosen for examining how the proposed filter design works.

The variable parameter in LMS algorithm is μ , whereas in NLMS algorithm the variable parameter is β and in the case of RLS algorithm the varying parameter is λ . All these varying parameters are found by trial and error method for each of the eighteen different types of signals and for each of the different types of hearing losses mentioned above. For each signal, the varying parameters are found after enormous number of trials, until the resultant signal closely matches the noise free signal. The number of samples and sampling frequency of each signal is as mentioned in the Table.5.

3. Results and Discussion

The results of mild, moderate, profound and severe hearing loss with VBF and adaptive algorithms are discussed here. Simulation is done using MATLAB. The hardware implementation is done in TMS320C6713 DSP Starter Kit.

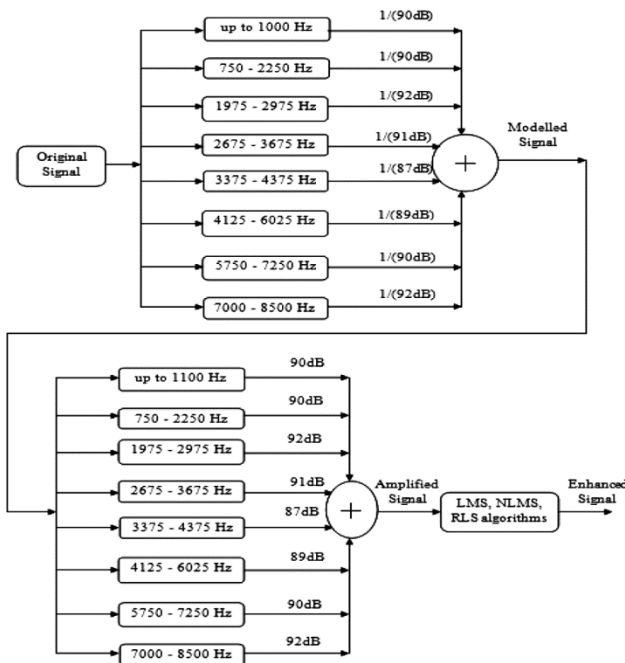


Fig. 7 — Structure for Profound hearing loss Modelling

3.1 FIR Filter Design

Various types of windows have been used to find their relative performance. The performance is measured in terms of Side Lobe Attenuation (SLA) and Main Lobe width (MLW). The performance comparison of windows is shown in Fig. 8 & Fig. 9. Always there should be a compromise between the attenuation of side lobes and main lobe width. It is observed that the hamming window and modified hamming window 3 are found to have equally good stop band attenuation and main lobe width characteristics. The frequency responses of the above mentioned windows and the filter responses of the LPFs of order 106 designed with these windows are illustrated in Figs 10-12.

The frequency responses of the LPF of order 106, designed by the three methods (windowing, frequency sampling and equiripple) are shown in Fig.13 & 14. From analyzing the results, it is seen that FIR filters designed using modified hamming window 3 and optimal design method, provide equal ripples in the stop band. Design based on window is simple, but it offers unnecessary high accuracy in the pass band. On the other hand, optimal design gives equal ripples in both the pass band and stop band i.e., it spreads the error evenly. Hence it is desirable to design the filter for digital hearing aid using optimal design method.

Two filters of order 106 designed by using optimal filter design method. First filter has a cut off

Table 5 — Specifications of Input signal

S. No	Audio Signal	Number of Samples	Sampling Frequency (Hz)
1	Speech 1	266240	44100
2	Speech 2	31600	16000
3	Speech 3	266240	44100
4	Speech 4	167616	16000
5	Speech 5	183744	16000
6	Announcement	173376	16000
7	Speaker in restaurant	184320	16000
8	Alps	167040	16000
9	Beethoven music	174528	16000
10	Birds	164160	16000
11	Clarinet	138240	16000
12	Ducks	97344	16000
13	Frogs	39168	16000
14	Industry	100224	16000
15	Kids	283392	16000
16	Piano	152064	16000
17	Telephone	52992	16000
18	Traffic	148608	16000

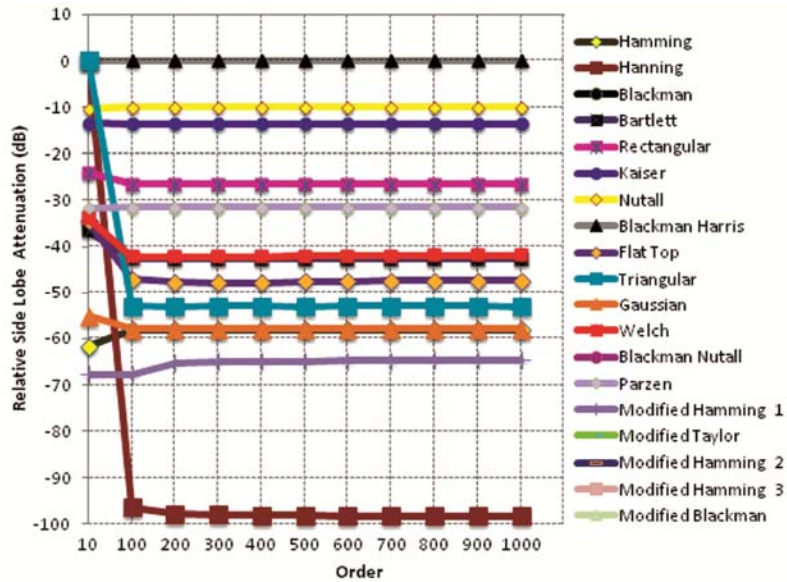


Fig. 8 — SLW of Windows for different orders

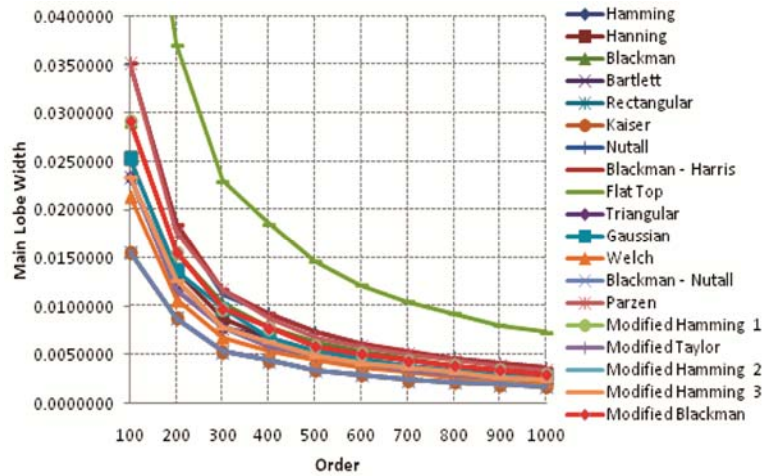


Fig. 9 — MLW of Windows for different orders

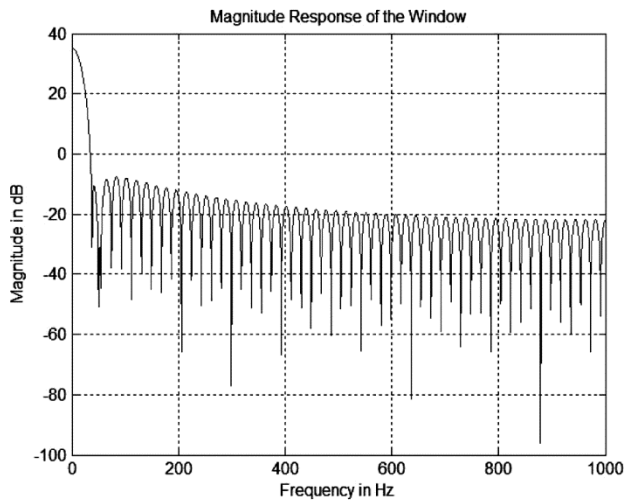


Fig. 10 — Magnitude Response

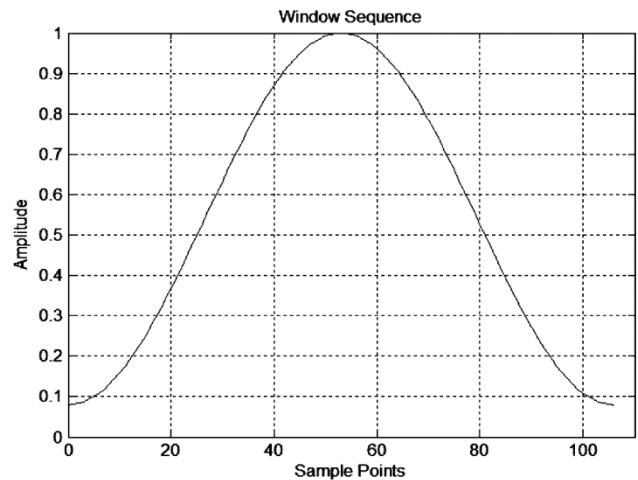


Fig. 11 — Hamming Window

frequency of 2000Hz and the second filter has a cut off frequency of 800Hz. The two filters are connected in cascade. An input signal with frequency components at 700Hz, 1200Hz and 2400Hz is passed through the filters. Since the cut off frequency of the first filter is 2000Hz, the filtered signal from LPF 1 has frequency components at 700Hz and 1200Hz. This signal is then sent into the second filter whose cut off frequency is 800Hz. The filtered output from LPF 2 consists of only the frequency component 700Hz. Thus the resultant signal has only one frequency component and the remaining two frequency components are filtered out. The frequency domain and time domain representation of input signals and the filter output are given in Figs 15-17.

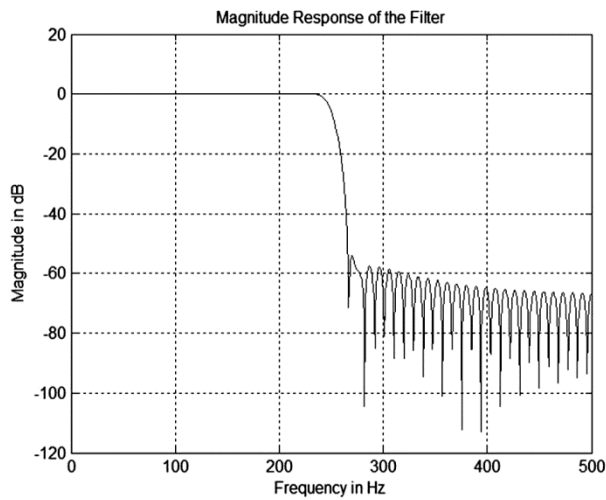


Fig. 12 — LPF with Hamming Window

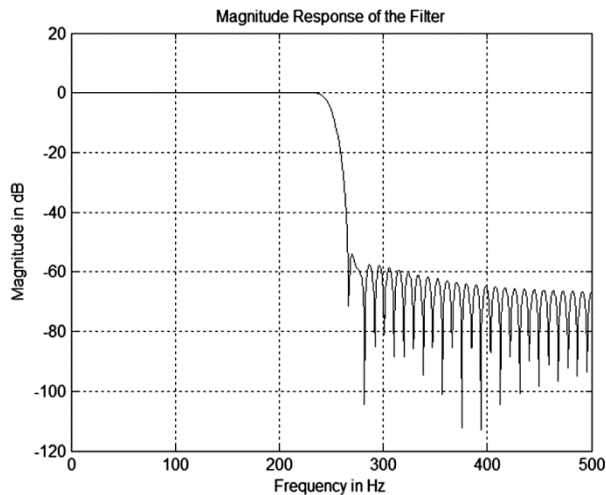


Fig. 13 — Magnitude Response in dB

3.2 Variable Bandwidth FIR Filter

Two low pass filters are used to get variable bandwidth filter by applying appropriate frequency shifting. The responses of the variable bandwidth filters designed with 10, 8, 6 and 4 bands are shown in Figs.18-21.

3.3 Audiograms

The designed filter response is matched with the set of audiograms^{6, 7} in Fig. 2. The audiogram and the matching result for mild to moderate hearing loss at low frequencies is shown in the

Fig. 22. The sound level and the comparison result for mild hearing loss at all frequencies is shown in the Fig. 23. The sound level and the comparison result mild hearing loss in high frequencies is shown in the Fig. 24. The sound level and the comparison result for moderate hearing loss at high frequencies is shown in the Fig. 25. The sound level and the comparison result of profound hearing loss is shown in the Fig. 26. The sound level and the comparison result of severe hearing loss is shown in the Fig. 27.

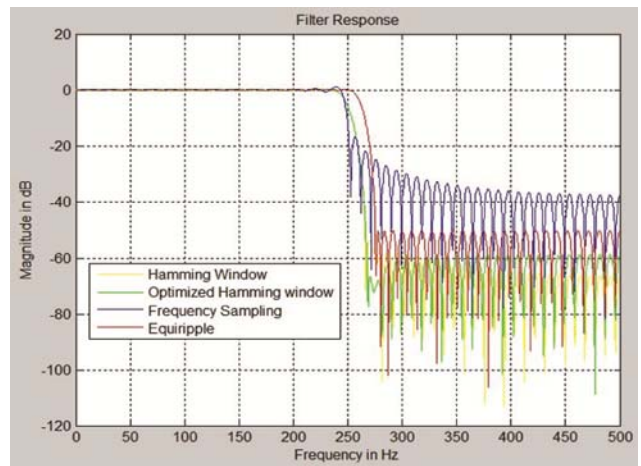


Fig. 14 — Magnitude Response

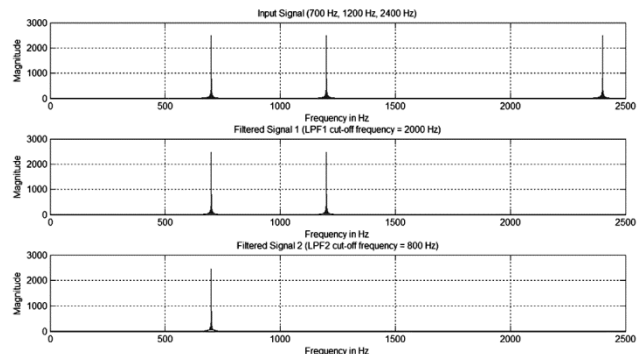


Fig. 15 — Frequency domain

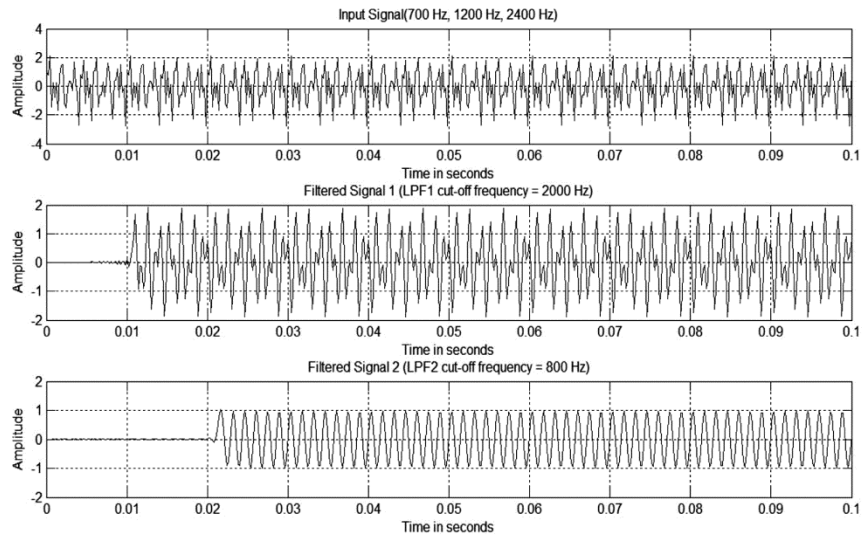


Fig. 16 — Time domain

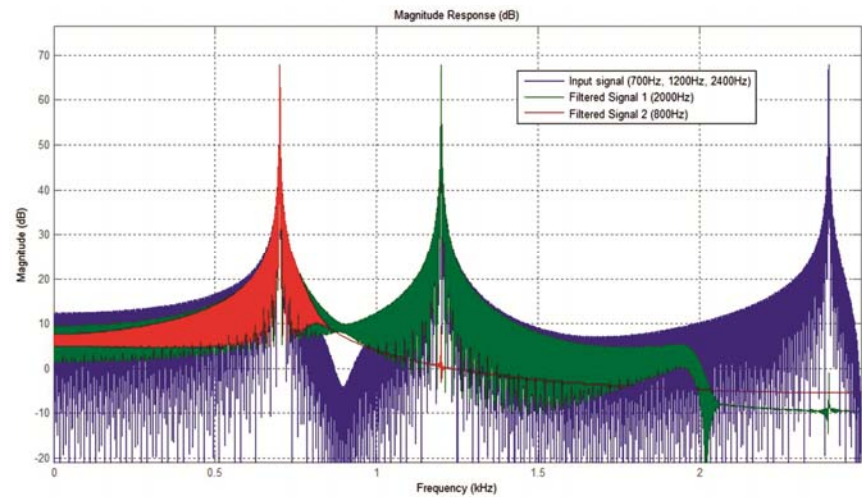


Fig. 17 — Filtering by optimal design

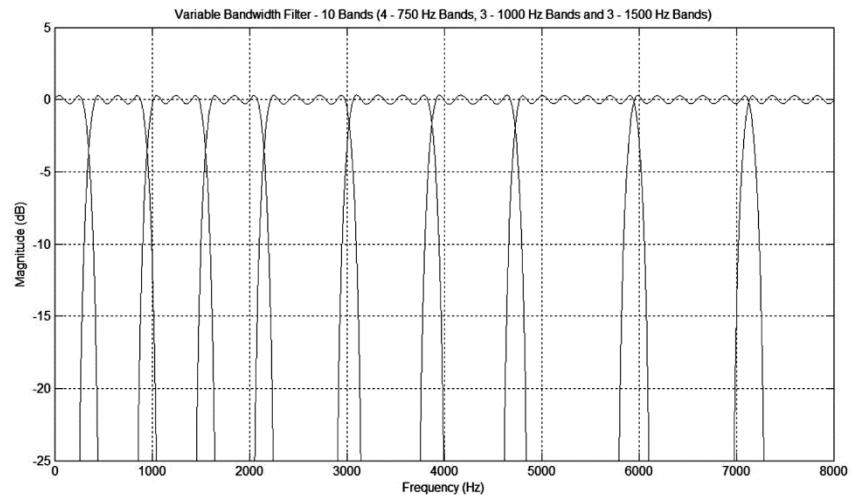


Fig. 18 — Variable bandwidth filter with 10 bands

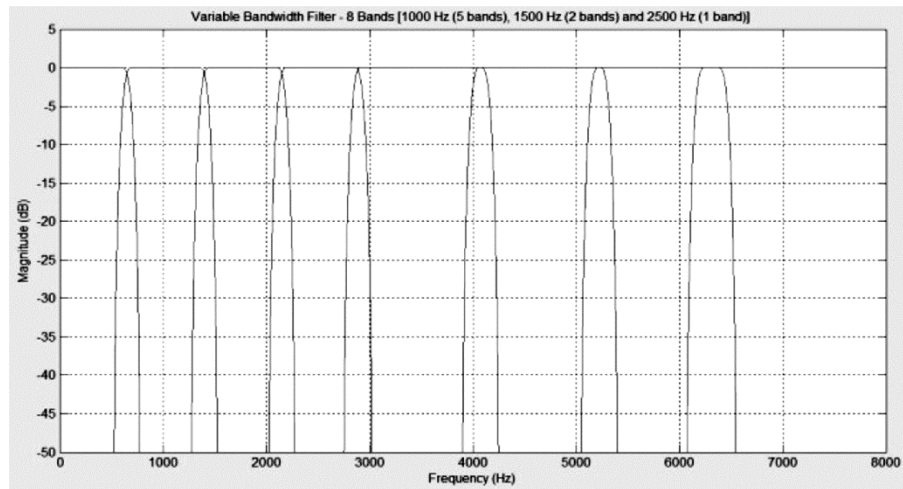


Fig. 19 — Variable bandwidth filter with 8 bands

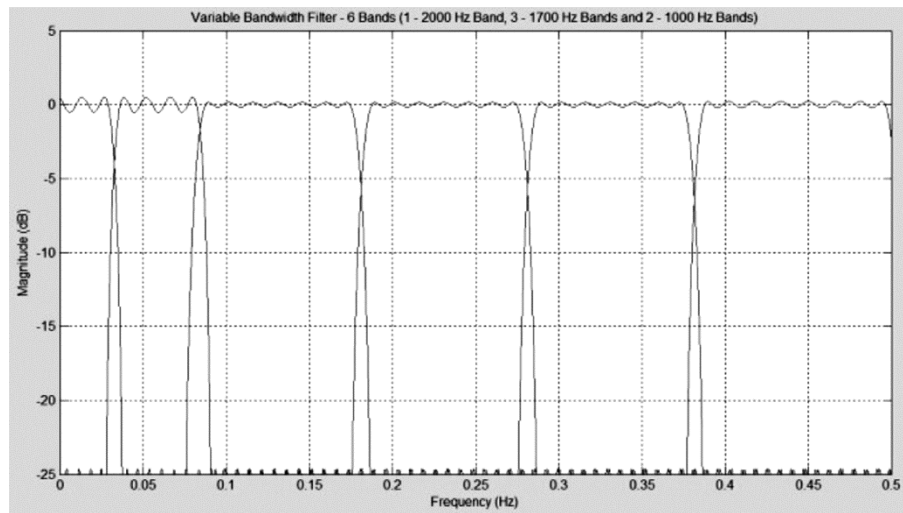


Fig. 20 — Variable bandwidth filter with 6 bands

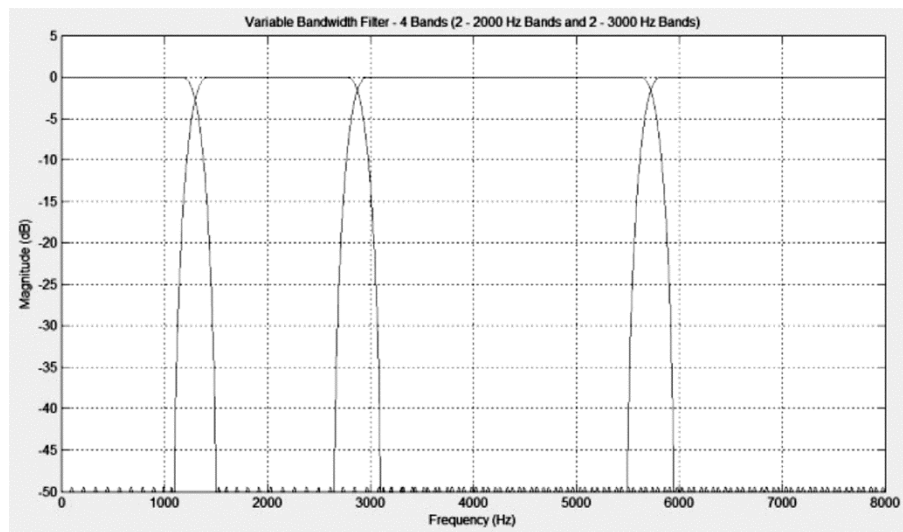


Fig. 21 — Variable bandwidth filter with 4 bands

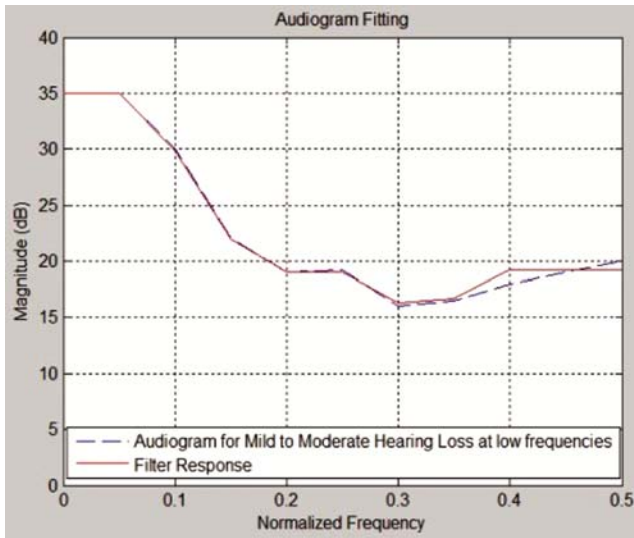


Fig. 22 — Mild to Moderate

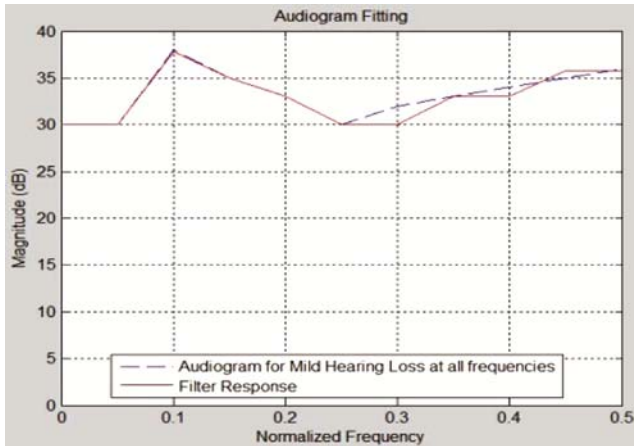


Fig. 23 — Mild at all frequencies

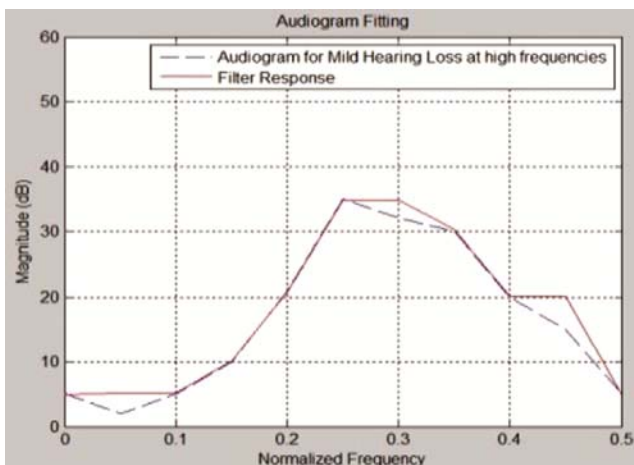


Fig. 24 — Mild at high frequencies

The matching error between the sample audiogram and the 8 band VBW filter response is given in Table.6. The audiograms have been analyzed with two cases. The first case considers six sample audiograms and the second case considers

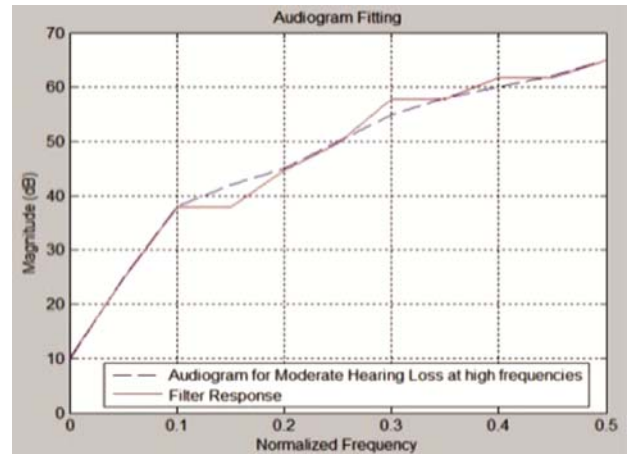


Fig. 25 — Moderate at high frequencies

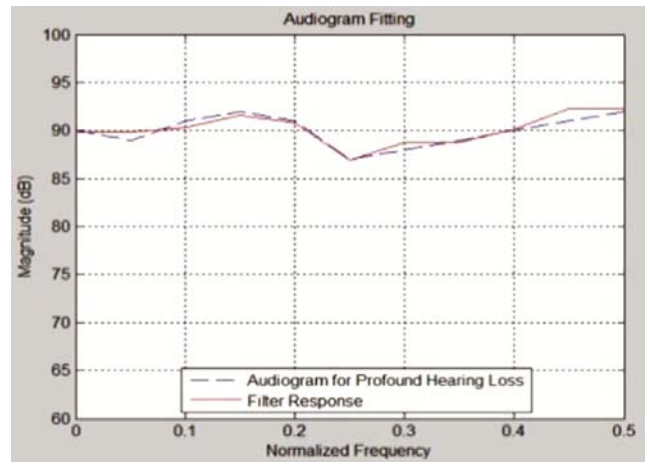


Fig. 26 — Profound hearing loss

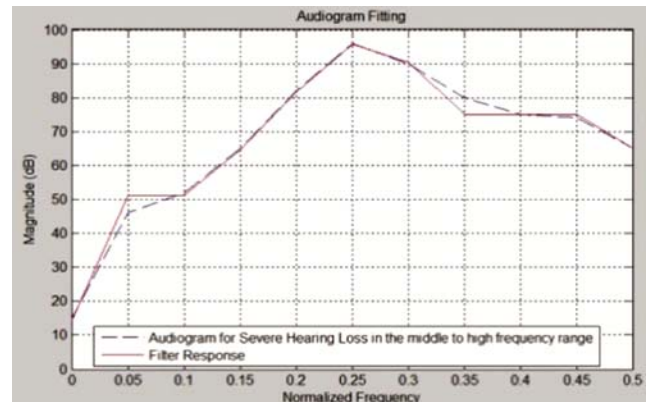


Fig. 27 — Severe in middle to high frequency

Table 6 — Matching Errors for sample audiograms

S. No	Signal	Mild Hearing Loss			Moderate Hearing Loss			Severe Hearing Loss			Profound Hearing Loss		
		LMS	NLMS	RLS	LMS	NLMS	RLS	LMS	NLMS	RLS	LMS	NLMS	RLS
		- μ	- β	- λ	- μ	- β	- λ	- μ	- β	- λ	- μ	- β	- λ
1	Speech 1	0.6	0.5	0.97999995	0.6	0.5	0.97999995	0.5	0.25	0.989999	0.5	0.25	0.989999
2	Speech 2	0.5	0.5	0.98999995	0.2	0.4	0.94999995	0.5	0.45	0.949999	0.5	0.45	0.949999
3	Speech 3	0.5	0.05	0.97999995	0.5	0.5	0.97999995	0.5	0.45	0.949999	0.5	0.45	0.949999
4	Speech 4	0.9	0.5	0.999	0.5	0.5	0.97999995	0.5	0.5	0.989999	0.5	0.5	0.989999
5	Speech 5	0.9	0.7	0.97999999	0.5	0.5	0.97999999	0.9	0.9	0.999999	0.9	0.9	0.999999
6	Announcement	0.999	0.5	0.99	0.7	0.7	0.99999995	0.7	0.3	0.999999	0.7	0.3	0.999999
7	Speaker in restaurant	0.9	0.5	0.999	0.9	0.5	0.99	0.5	0.5	0.989999	0.5	0.5	0.989999
8	Alps	0.9	0.9	0.999999	0.9	0.5	0.979999	0.9	0.9	0.999999	0.9	0.9	0.999999
9	Beethoven music	0.98	0.78	0.97999995	0.98	0.78	0.98999995	0.5	0.25	0.979999	0.5	0.25	0.979999
10	Birds	0.5	0.5	0.98999995	0.9	0.9	0.98999995	0.7	0.7	0.999999	0.7	0.7	0.999999
11	Clarinet	0.5	0.5	0.98999995	0.5	0.5	0.999	0.5	0.5	0.979999	0.9	0.9	0.989999
12	Ducks	0.9	0.9	0.97999995	0.9	0.9	0.97999995	0.8	0.5	0.9	0.8	0.5	0.979999
13	Frogs	0.5	0.5	0.999999	0.9	0.6	0.98999995	0.5	0.5	0.979999	0.5	0.5	0.9899
14	Industry	0.9	0.9	0.97999995	0.9	0.7	0.999	0.5	0.5	0.979999	0.5	0.5	0.979999
15	Kids	0.999	0.9	0.99	0.999	0.4	0.99	0.9	0.3	0.979999	0.9	0.3	0.979999
16	Piano	0.9	0.9	0.98999995	0.9	0.9	0.98999995	0.5	0.5	0.989999	0.5	0.5	0.989999
17	Telephone	0.5	0.5	0.98999995	0.5	0.5	0.98999995	0.5	0.5	0.989999	0.5	0.5	0.989999
18	Traffic	0.9	0.9	0.98	0.9	0.9	0.98	0.5	0.5	0.989999	0.5	0.5	0.989999

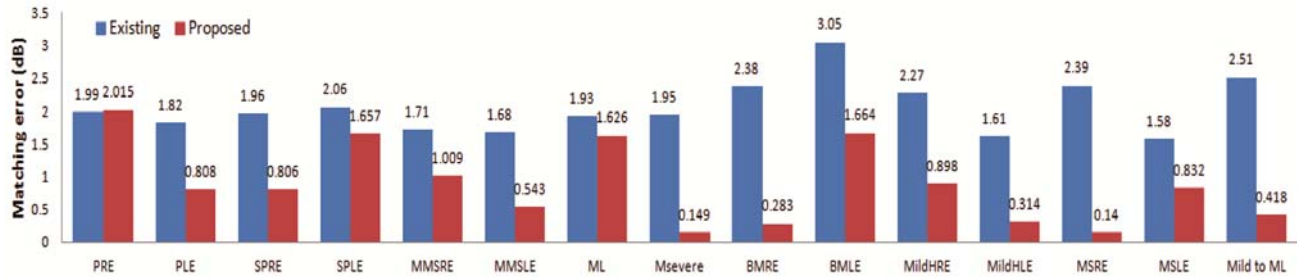


Fig. 28 — Matching error for referring 15 audiograms

fifteen audiograms. The second case is illustrated in Fig. 28. The designed filter provides minimal error. The filter responses are compared with the sample audiograms and matching error has been found. The matching error is reduced considerably and lies between 0 to 2.5dB.

3.4 Hearing Loss System Model

The parameters that are used for processing each of the signals belonging to various losses are given in Table.7. All these varying parameters are found by trial and error method for each of the eighteen different types of signals and for each of the different types of hearing losses listed in Table.7. In this section the results obtained for the variable bandwidth filter designed for mild hearing loss are shown with all the inputs. In each figure, the original

signal in which a human with normal listening ability, simulated signal in which a human with mild listening loss hears, the amplified output from the VBF, LMS equalized signal, NLMS equalized signal and RLS equalized signal are plotted.

Figures 29-32 indicates, how well the proposed hearing loss model works for mild hearing loss (MildHL). From the above results it is conveyed that the proposed model is capable of reproducing the original signal as such i.e., the output of the filter closely matches the signal which the normal hearing person hears. Thus it can be said that the designed filter is able to deliver an intelligible and audible signal, so that the hearing impaired can hear well. The presence of background noise is invisible in the pictorial representation, as it is of small measure, but

Table 7 — Parameters used for the adaptive algorithms

S. No	Type of hearing loss	Matching Error in dB [13]	Matching Error in dB
Referring six audiograms			
1	Mild to moderate hearing loss at low frequencies	1.35	0.827
2	Mild hearing loss at all frequencies	1.27	0.694
3	Mild hearing loss at high frequencies	2.00	1.566
4	Moderate hearing loss at high frequencies	2.57	1.633
5	Profound Hearing loss	2.51	0.572
6	Severe hearing loss in the middle to high frequencies	2.9	2.219
Referring fifteen audiograms			
1	Profound loss Right Ear (PRE)	1.99	2.015
2	Profound loss Left Ear (PLE)	1.82	0.808
3	Severe to Profound loss Right Ear(SPRE)	1.96	0.806
4	Severe to Profound loss Left Ear(SPLE)	2.06	1.657
5	Moderate to Moderately Severe Right Ear(MMSRE)	1.71	1.009
6	Moderate to Moderately Severe Left Ear(MMSLE)	1.68	0.543
7	Moderate laterized 500, 2k(ML)	1.93	1.626
8	Moderately severe, laterized at 2k (MSevere)	1.95	0.149
9	Bilateral Moderate loss Right Ear(BMRE)	2.38	0.283
10	Bilateral Moderate loss Left Ear(BMLE)	3.05	1.664
11	Mild Hearing loss Right Ear(MildHRE)	2.27	0.898
12	Mild Hearing loss Left Ear (Mild HLE)	1.61	0.314
13	Moderate Sensorineural loss Right Ear(MSRE)	2.39	0.140
14	Moderate Sensorineural loss Left Ear(MSLE)	1.58	0.832
15	Mild to Moderate loss(Mild to ML)	2.51	0.418

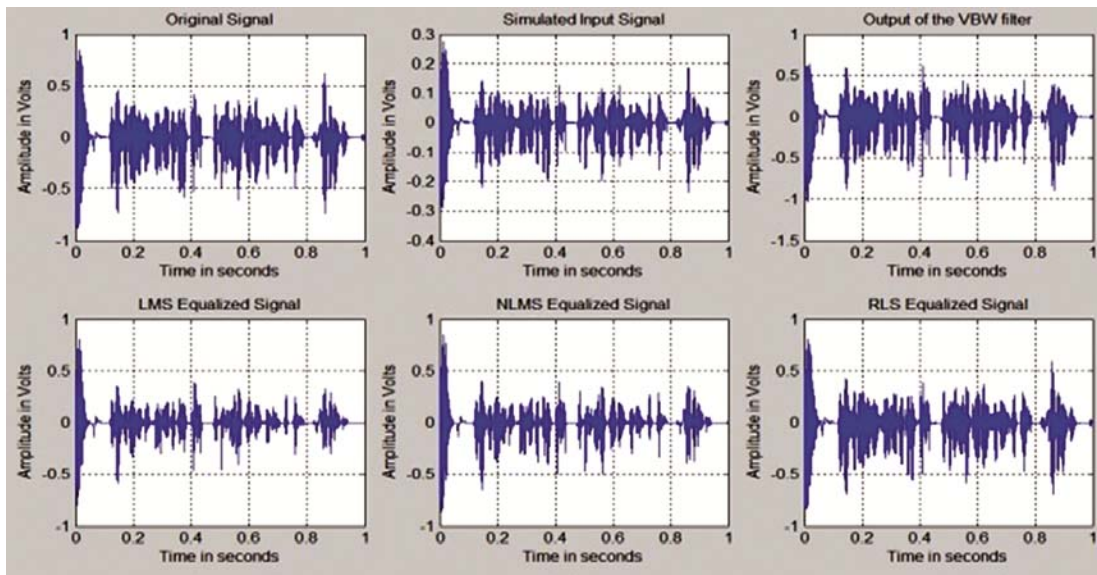


Fig. 29 — “Speech 1” as input to Mild HL model

while listening to the audio clip of the amplified signal, it is found that some noise has added up due to several reasons. From all the figures it is seen that, among the three noise cancellation techniques, RLS outperforms LMS and NLMS and the resultant RLS equalized signal is realized to be noise free. Also the

filter should be able to produce a signal whose power is equivalent to the power of the original signal. Hence the performance of the designed filter is analyzed by taking power as a metric for quantitative assessment. Power is nothing but the square of the root mean square value divided by the length of the

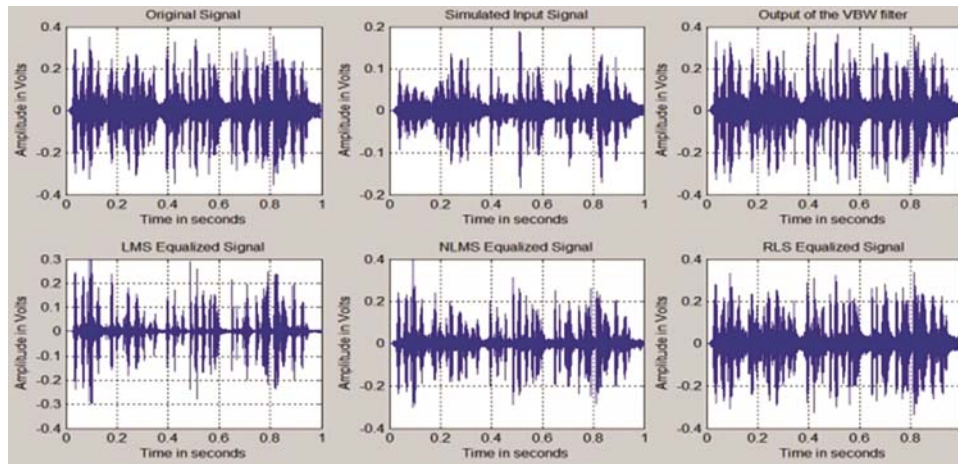


Fig. 30 — “Announcement” as input to Mild HL model

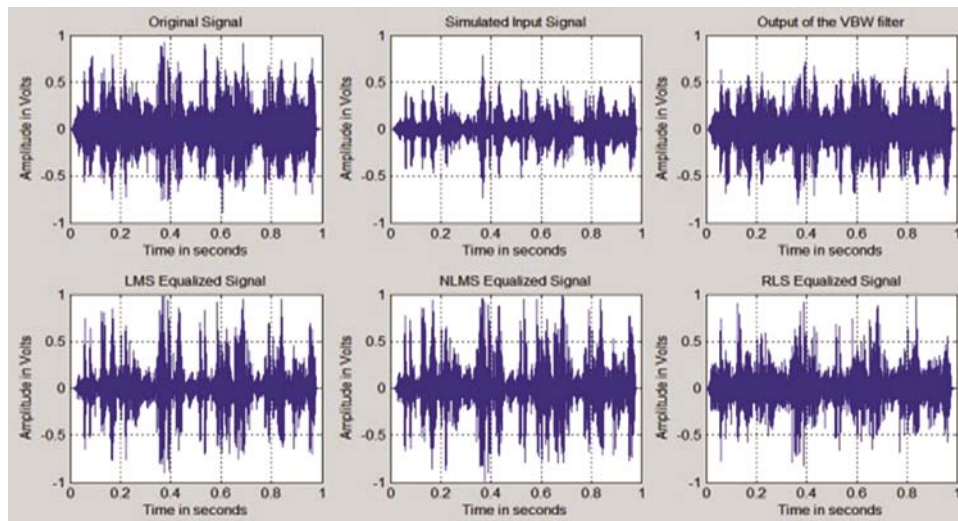


Fig. 31 — “Conversation in restaurant” as input

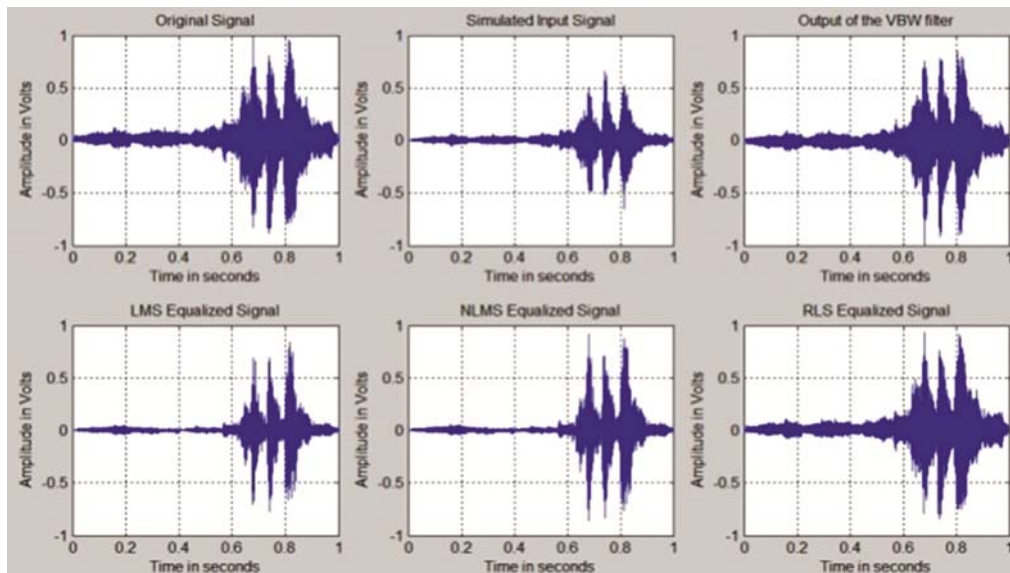


Fig. 32 — “Beethoven Music” as input

signal. The power thus computed is converted into decibels. The normalized power of mild, moderate, severe and profound hearing loss VBF is calculated and is given in Figs. 33-44. The power for the severe hearing loss analysis is given in Table.8. From the

tabulated values it is evident that the designed model has achieved reasonably good matching among the power of the amplified signal, RLS signal and the original signal.

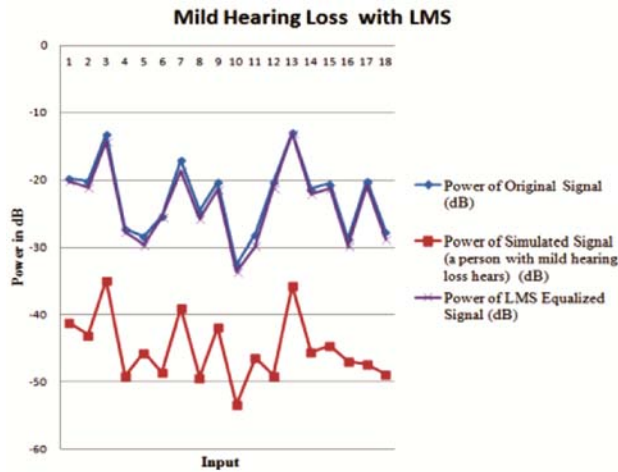


Fig. 33 — Mild HL with LMS

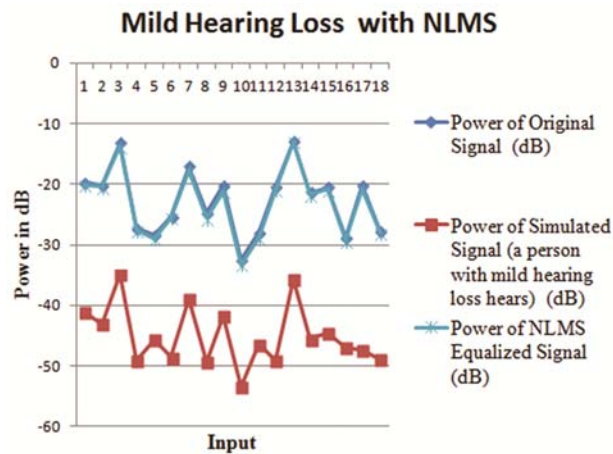


Fig. 34 — Mild HL with NLMS

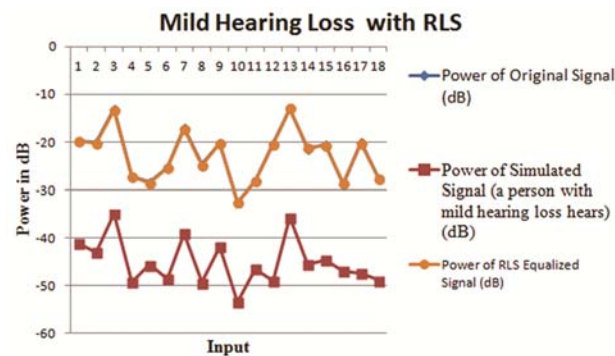


Fig. 35 — Mild HL with RLS

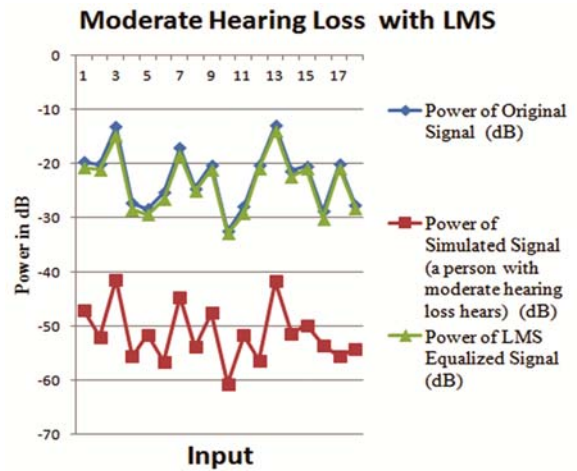


Fig. 36 — Moderate HL with LMS

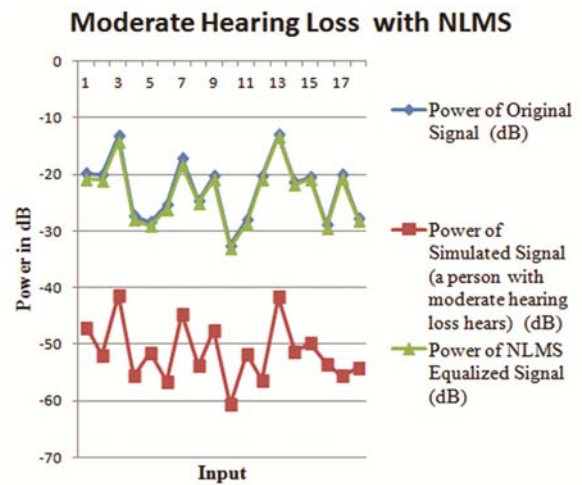


Fig. 37 — Moderate HL with NLMS

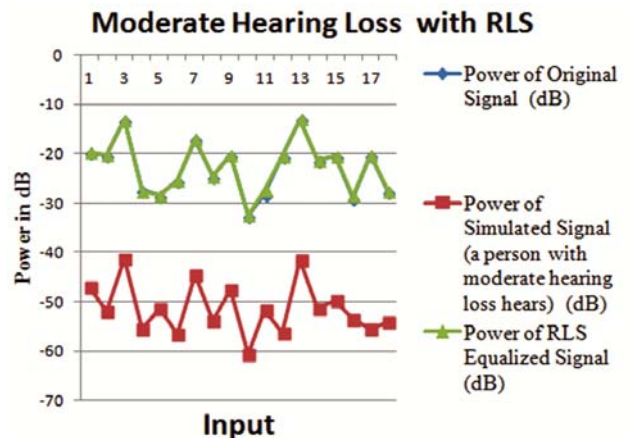


Fig. 38 — Moderate HL with RLS

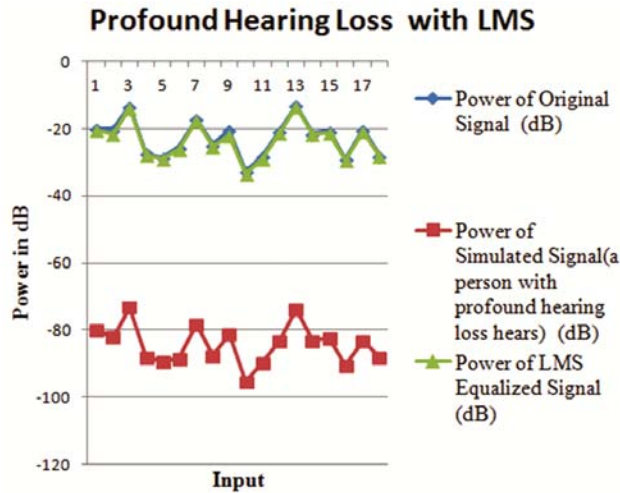


Fig. 39 — Profound HL with LMS

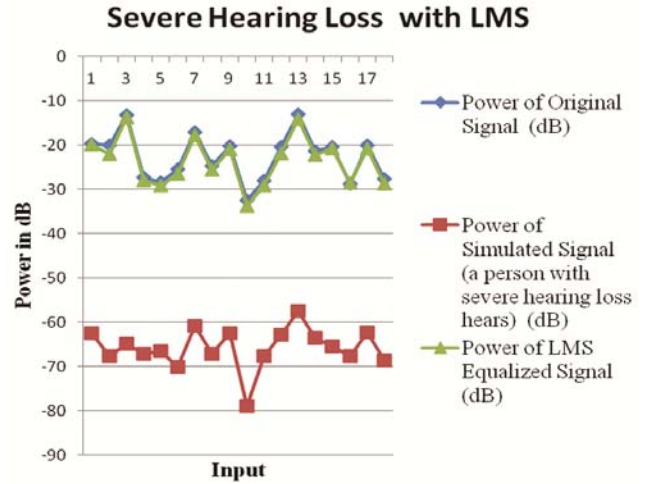


Fig. 42 — Severe HL with LMS

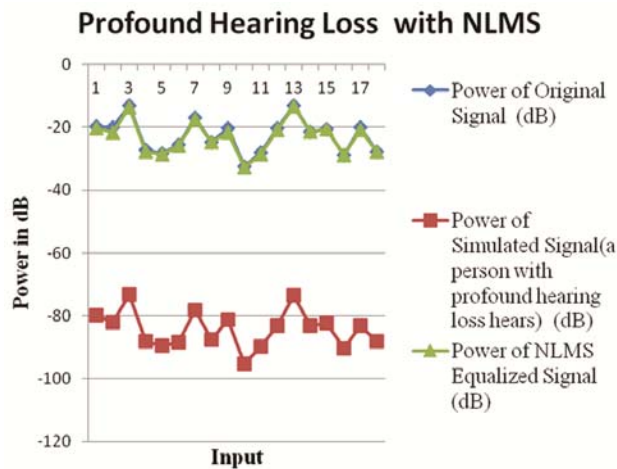


Fig. 40 — Profound HL with NLMS

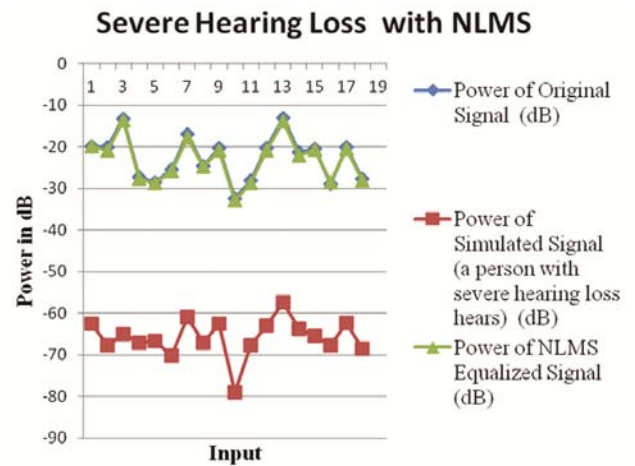


Fig. 43 — Severe HL with NLMS

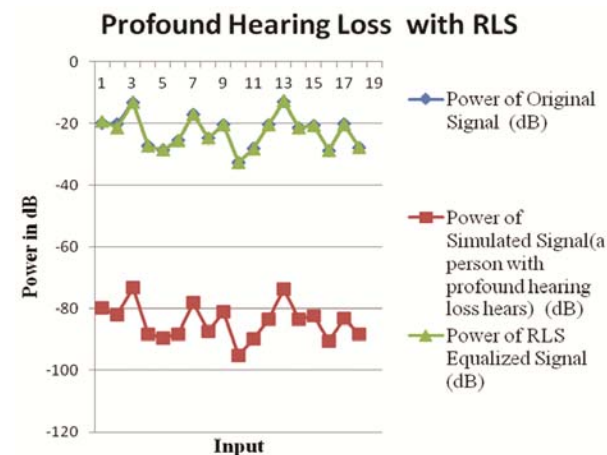


Fig. 41 — Profound HLwith RLS

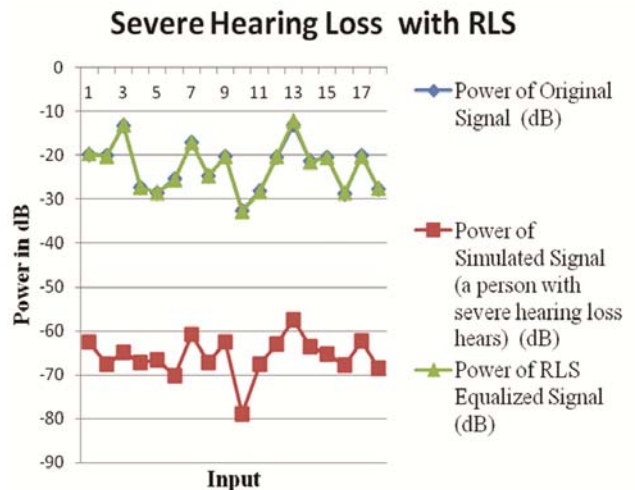


Fig. 44 — Severe HL with RLS

Table 8 — Normalized Power of Sever hearing loss VBF

S. No	Signal	Power of Original Signal (dB)	Power of Simulated Signal (a person with severe hearing loss hears) (dB)	Power of Output Signal (dB)	Power of LMS Equalized Signal (dB)	Power of NLMS Equalized Signal (dB)	Power of RLS Equalized Signal (dB)
1	Speech 1	-19.787562	-62.554392	-19.666768	-19.902600	-19.801700	-19.713900
2	Speech 2	-20.136094	-67.701276	-20.203600	-22.185589	-20.890034	-20.193296
3	Speech 3	-13.171253	-64.90525	-13.092257	-13.757300	-13.697800	-12.946099
4	Speech 4	-27.288818	-67.117450	-27.404544	-28.011791	-27.779100	-27.276100
5	Speech 5	-28.450658	-66.648259	-28.585071	-29.074100	-28.685258	-28.480500
6	Announcement	-25.428958	-70.153854	-25.359169	-26.507877	-25.748596	-25.590300
7	Speaker in restaurant	-17.070912	-60.786571	-17.209845	-17.873212	-17.737500	-17.157755
8	Alps	-24.650606	-67.133540	-24.766217	-25.614308	-24.831571	-24.649951
9	Beethoven music	-20.269421	-62.566076	-20.985257	-21.043645	-20.823886	-20.193001
10	Birds	-32.530408	-78.996189	-32.567565	-33.846777	-32.900600	-32.695100
11	Clarinet	-28.049320	-67.672582	-28.234622	-29.201400	-28.624500	-28.331700
12	Ducks	-20.365259	-62.929201	-20.501900	-21.836263	-20.919044	-20.253044
13	Frogs	-12.972983	-57.474718	-12.213907	-14.210078	-13.943784	-12.129924
14	Industry	-21.358332	-63.667748	-21.064843	-22.374200	-22.150000	-21.517400
15	Kids	-20.523119	-65.375028	-20.481218	-20.856409	-20.619672	-20.548016
16	Piano	-28.851278	-67.740306	-28.863750	-28.671449	-28.606863	-28.462583
17	Telephone	-20.102272	-62.374870	-21.038123	-20.809100	-20.436600	-20.328200
18	Traffic	-27.741911	-68.588052	-27.640632	-28.770424	-28.021024	-27.576535

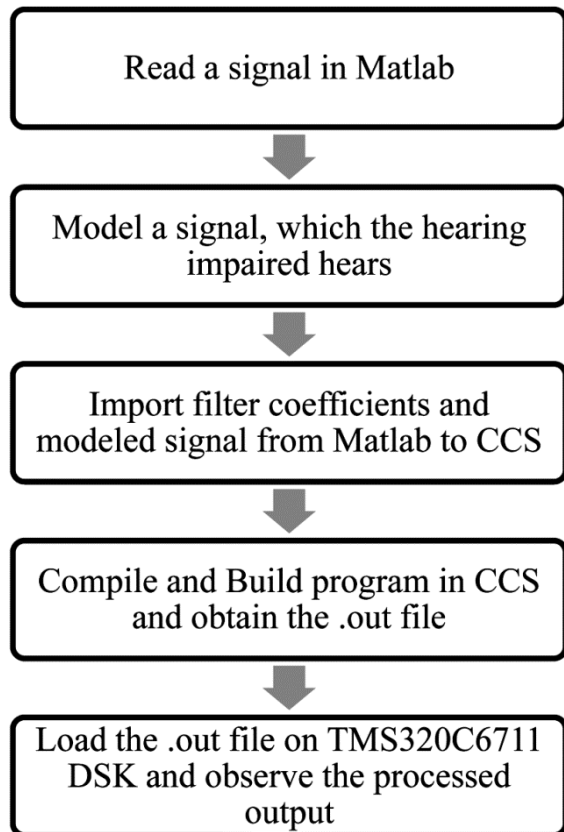


Fig. 45 — Flow Chart for hardware implementation

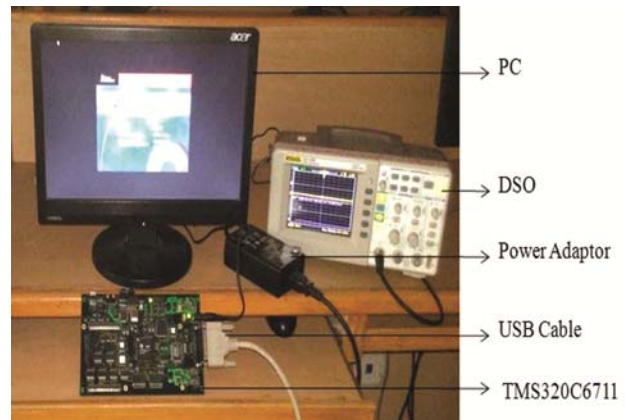


Fig. 46 — Experimental Setup

3.5. Implementation in DSP Kit

The design procedure for DSP kit implementation is given in Fig. 45. The experimental setup for the variable bandwidth filter in DSP kit is shown in Fig. 46. In this section the results obtained using CCS are discussed. A sinusoidal signal that comprises eight frequency components at 700Hz, 1500Hz, 2800Hz, 3600Hz, 4900Hz, 5800Hz, 6900Hz and 7400Hz is taken and from this sinusoidal signal, signals which a mild, moderate, severe and profound hearing loss patient hears are modeled. The simulated signals and the amplified outputs of the VBFs designed for the moderate hearing loss is given in Fig. 47.

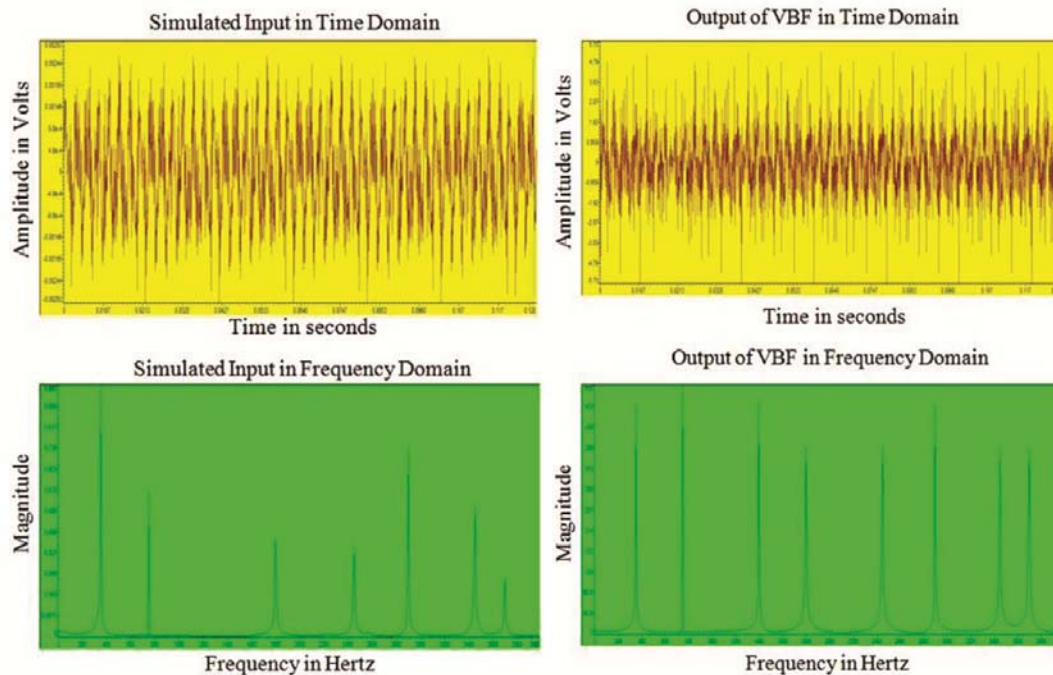


Fig. 47 — Moderate Hearing Loss (a) Simulated Input (b) Output of VBF

4. Conclusion

This paper also proposes hearing loss system model which incorporates variable bandwidth FIR filter and adaptive algorithms. With reference to the sample audiogram, the magnitude responses of the designed variable band filters are adjusted by trial and error and the appropriate magnitude is preferred for each band. An eight band variable bandwidth filter suitable for digital hearing aid is designed. Eight bands are generated by means of spectrum shifting property and providing sufficient gains to each band. The performance of the designed filter is analyzed by taking a number of real time signals. For quantitative assessment the power of the original signal and the power of the amplified output are compared. From the results obtained it is observed that the hearing aid modelling system is capable of reproducing the original signal as such without any reduction in power. Further adaptive algorithms LMS, NLMS and RLS are applied to enhance the quality of the resultant signal by removing noise. Several cases of hearing loss patterns are considered here for the analysis. However the quality of speech signal is further enhanced by applying deep neural networks. Further work is to apply deep neural networks to improve the quality of hearing aid systems.

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