Design and Implementation of an Area and Power-efficient Reconfigurable Hearing Aid using Interpolated Sub-band Distribution Technique

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Abstract: A hearing aid is a compensatory device that helps in overcoming various hearing disabilities. Since different people possess different hearing problems and the requirement gets changed over time, it is necessary to design a reconfigurable hearing aid which is generic in nature such that it supports various hearing disabilities without modifying the hardware components. The objective of the paper is to implement a reconfigurable digital hearing aid which is hardware efficient and it is auto-adaptable to disabilities ranging from mild to severe intensities. Since multipliers and LUTs are power hungry elements, we have proposed a design which is multiplier less and LUT-less DA Architecture. The prototype filter is a FIR filter which is designed using LUT-less Distributed Arithmetic Algorithm that saves 64% of logic elements & memory and 76% of power utilization. As it is a multiplier-less as well as LUT- less architecture, hence this can be claimed to be area and power efficient design. The delays and matching errors are within the standard limits which are accepted globally. The input audio spectrum is divided into three regions and for each region there are four different filter banks are proposed using interpolated sub-bands distribution. Xilinx System Generator is used to implement the proposed design. The proposed design requires least manual configuration for selection of filter banks for audiogram matching which also minimizes the trial-and-error method to establish the best match with the person's audiogram.

Keywords: audiogram, reconfigurable, sub-band distribution

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1. Introduction

Hearing aid device benefits lot of people having hearing loss problems and 90% of users accept it. The hearing aid performs the function of making low intensity sounds audible and loud sounds comfortable. These devices match audiogram gains and dynamic ranges of ear characteristics. The basic function of hearing aid is amplification of sound, but designers face challenges of restoring other characteristics like loudness, speech intelligibility etc. Fixed signal bands formation plans are used in most of the present-day hearing-aid systems. Fixed filter banks not have sufficient flexibility to match audiogram response of different hearing losses. Reconfigurable filter bank can be a solution to control parameters for more adaptable hearing aid device as per user requirements in every situation using non uniform sub-band distribution. [1]

The most common type of hearing loss is Sensorineural Hearing Loss (SNHL) where inner sensory nerves are damaged and this loss can happen at any age so most of the hearing aid devices are design to compensate SNHL[4]. The technique of audiogram matching is basically adjustment of magnitude response of filters in an inverted fashion where amplification is done to the magnitude response as required . Unfortunately, only 10% of the hearing aids are able to meet up the requirement. Thus, a properly adjustable hearing aid can improve the speech matching of hearing deprived people [5].

ENT specialist or we can say audiologist perform audiogram test by the method of pure tone audio frequency signals. These signals are within the range of 250Hz to 16Khz. The mark 'O' is for the left ear and 'X' is for right Ear on Y-axis, and amplitude or loudness of hearing sensitivity is on X-Axis. The following audiogram in Fig. 1

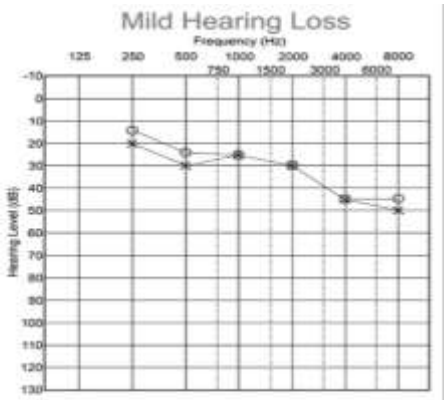
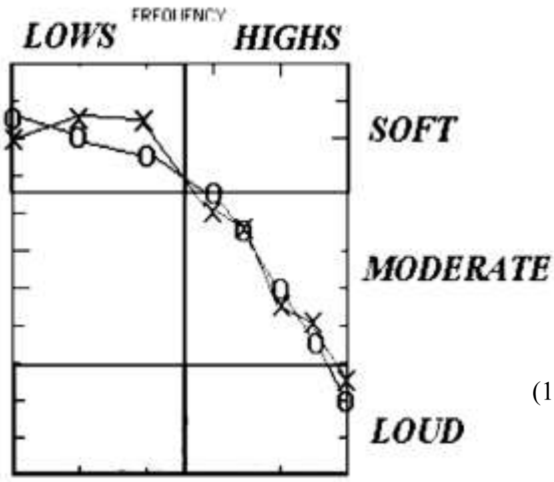


Fig. 1. Audiogram is taken from www.Earinfo.com website and this represents a hearing problem to a patient at high frequency.

is for hearing loss at high frequency. There are other audiograms as well for different range of hearing losses. The graph is measured from low to high frequencies (low to high pitches) going from left to right, and the graph is measured from soft sounds on the top to loud sounds at the bottom shown in Fig. 2. The above information is important for achieving reconfigurability. [7]

Finite impulse response (FIR) are digital filters having common DSP functions and are widely used in multiple applications like telecommunications, wireless/satellite communications, video and audio processing, biomedical signal processing and many others. On one hand, ASICs have certain restrictions related to high development costs and time-to-market factors while, on the other hand, programmable DSP processors are enable to meet desired performance due to their sequential-execution architecture [9].



(1)

Fig. 2. Information gathered from audiogram (Source: www.earinfo.com)

The FIR digital filter is presented as:

$$y[n] = \sum_{k=0}^{N-1} x[n-k]c_k$$

the FIR filter output, x [n-k] is the input data and ck represents the filter coefficients. Equation (1) shows the multiplier-based filter that can be implemented but it may become highly expensive in terms of area and speed. This issue has been partially resolved with the use of DA (Distributed Arithmetic) algorithm.

Initially, sub-bands are distributed uniformly which provide average audiogram matching but later non-uniform sub-bands distribution is preferred to get better matching and less matching error [18]. There are several techniques defined for sub-bands implementation but the proposed structure is implemented using multirate signal processing methods. A 17-band reconfigurable hearing aid is implemented in this paper using interpolation and decimation of the prototype filter ($H_L(z)$) which is again designed using Distributed Arithmetic (DA) Algorithm which reduces the hardware complexity. This proposed filter can adapt to optimum bands by itself for various hearing loss audiograms.

Another factor that is important in a hearing aid is the noise reduction or noise cancellation technique. It is well known that noise in background reduces the understanding of speech and that the greater the level of background noise the greater the reduction in understanding. There is less redundancy in the speech signal for a person with hearing loss since part of the speech is distorted because of the hearing loss. As a consequence, people with hearing loss have much greater difficulty than normally hearing people in understanding speech in noise [23].

In this paper, a reconfigurable hearing aid system is being proposed which is multiplier-less. The proposed design uses DA architecture which is LUT-less as well as multiplier-less design making it area and power efficient. Then the prototype filter is designed to achieve sub-band distribution using interpolation to match with the audiogram of the ailing person. Lastly, the proposed design is programmed for reconfigurability and automatic adaptability with the hearing audiogram. This paper is organised as follows: Section II describes about various theoretical considerations, Section III describes the implementation of the proposed design, Section IV constitutes experimental results, conclusion is drawn in Section V and Section VI describes the future scope of this proposed filter design.

2. Background

2.1 Distributed Arithmetic Algorithm

Distributed-Arithmetic Algorithm DA algorithm is one of the well-known multiplier-less methods which involves use of Logic elements (RAMs, ROMs) or Look-Up Tables (LUTs) to store pre- computed values of filter coefficients. It is a powerful technique for replacing the use of multiple and accumulates operations that make the design hardware-efficient. This multiplier-less architecture of DA algorithm is implemented using efficient partition of the function in partial terms using 2's complement binary representation of data. The partial terms can be pre-calculated and stored in LUTs. The use of memory/LUT capacity increases exponentially with the order of the filter, due to which DA implementations require 2K words, K being the number of taps of the filter. Assuming coefficients c_k are known constants, equation (1) can be rewritten as follows:

$$y[n] = \sum_{k=0}^{N-1} x[n]c_k \quad (2)$$

The binary decimal form of the variable x[n] can be represented as follows:

$$x[n] = \sum_{b=0}^{B-1} x_b[n]2^b \ldots\ldots\ldots x_b \in \{0\ 1\}$$

where x_b [n] represents the b^{th} bit of x[n] and B represents the input width. Finally, the inner product are as follows:

$$y[n] = \sum_{n=0}^{N-1} c[n] \sum_{b=0}^{B-1} 2^b x_b[k]$$

$= c[0](x_{B-1}[0]2^{B-1} + x_{B-2}[0]2^{B-2} + \ldots + x_0[0]2^0) +$

$c[1](x_{B-1}[1]2^{B-1} + x_{B-2}[1]2^{B-2} + \ldots + x_0[1]2^0) + \ldots + c[N-1] (x_{B-1}[N-1]2^{B-1} + x_{B-2}[N-1]2^{B-2} + \ldots + x_0[N-1]2^0)$

$= (c[0]x_{B-1}[0] + c[1]x_{B-1}[1] + \ldots + c[N-1]x_{B-1}[N-1])\, 2^{B-1} + (c[0]x_{B-2}[0] + c[1]x_{B-2}[1] + \ldots + c[N-1]x_{B-2}[N-1])\, 2^{B-2} + \ldots + (c[0]x_0[0] + c[1]x_0[1] + \ldots + c[N-1]x_0[N-1])2^0$

$$y[n] = \sum_{n=0}^{N-1} c[n]\, x_b[k] \sum_{b=0}^{B-1} 2^b \quad (3)$$

The coefficients in most cases for the multiply accumulate operation are constants. The partial products are achieved by multiplying the coefficients c[i] in one bit of data x[i] at a time using AND operation. These partial products are then added and the result depends only on the outputs of the input shift registers. Further, the AND functions and adders can be replaced by Look Up Tables (LUTs) whose outputs are the partial products. Input in sequence manner is fed to the shift register at the input sampling rate. The serial output is presented to the RAM based shift registers at the bit clock rate which is n+1 times (n is number of bits in a data input sample) the sample rate. The data is stored in particular address in RAM based shift registers. The scaling accumulator loads the results from LUTs from LSB to MSB and the filter output will be accumulated over the time. For a 'n' bit input, n+1 clock cycles will be required for a symmetrical filter to generate the output [15].

One disadvantage with original DA architecture is that its LUT size (2K-words) grows exponentially with increasing the filter order K. DA offset binary coding (DA-OBC) can be used to overcome exponentially increasing memory burden of DA. For a K-tap FIR filter, DA-OBC requires a 2K−1-word LUT. The modified DA-OBC can reduce the LUT size from 2K−2 to as low as 2 by exploiting the observation that if the single term inside the LUT can be relocated outside the LUT, then the mirrored version in the lower half of the LUT is the upper half of the LUT where the signs are reversed. The LUT size decreased by half at every iteration and at last the LUT-less DA architecture can be achieved. Thus, the LUT-less DA architecture enables more high-order FIR filter implementation on a given FPGA platform.

TABLE 1 Filter coefficients as they are defined in LUT for traditional DA Architecture

B3B2B1B0	LUT data
0000	0
0001	h[0]
0010	h[1]
0011	h[1] + h[0]
0100	h[2]
0101	h[2] + h[0]
0110	h[2] + h[1]
0111	h[2] + h[1] + h[0]
1000	h[3]
1001	h[3] + h[0]
1010	h[3] + h[1]
1011	h[3] + h[1] +h[0]
1100	h[3] + h[2]
1101	h[3] + h[2] + h[0]
1110	h[3] + h[2] + h[1]
1111	h[3] + h[2] + h[1] + h[0]

2.2 Subband Distribution

Considering both the complexity and the performance, non-uniformly spaced filter banks are mainly preferred in digital hearing aids. For sub-band distribution, two factors are needed to be taken into account. One fact is the hearing resolution is more at lower frequencies than in higher frequencies. The other fact is associated with the transmission process of sound signal in ears (cochlea). Sound waves create vibration which travels from base to apex when waves enter cochlea. In this process, all the high frequency components of the signal pass through the base but only few of the low frequency components of the signal reach the apex, so the screen near the base suffers more damage than the screen near the apex. Based on these two facts, better adaptability can be obtained if more sub-bands in low frequency range as well as in high frequency range are created. The common problem of existing non- uniform filter banks is that is to achieve both low complexity and low delay at the same time.

To maintain low complexity, the number of sub-bands is usually kept as small as possible, which will minimize the matching performance. Additionally, it is observed that low complexity is usually achieved with the disadvantage of long delays. Large interpolation factor results in long processing delays in the filter bank. This is not a good thing as long delays can cause mismatch in speech and lip-reading.

3. Implementation

In all the reconfigurable hearing aid devices that were mentioned in the literature are highly complex and the coefficient multiplier which is the most resource consuming in the hardware is used by the other filter bank methods. Thus, the Distributed Arithmetic Algorithm is implemented using modified technique where multipliers as well as LUTs are not used which reduce the design complexity and make it area and power efficient. The diagram of the proposed reconfigurable hearing aid system is shown in Fig 3. Each filter bank shown in the figure below is a combination of interpolated FIR filters and each filter is designed using DA

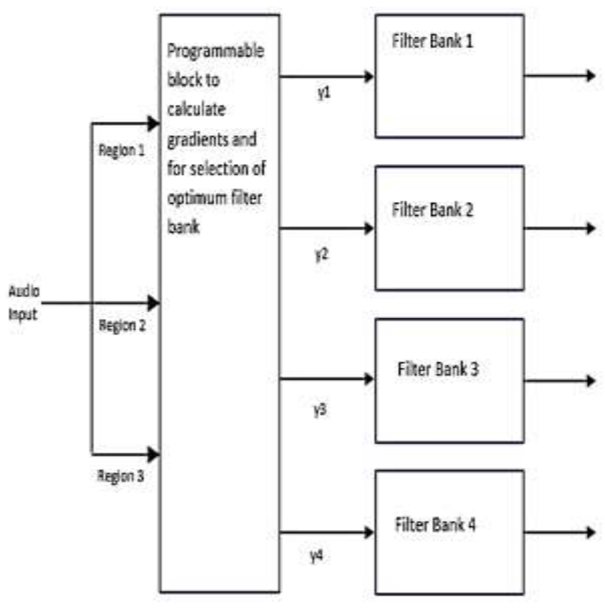


Fig. 3 Structure of the proposed reconfigurable structure

algorithm, it is explained in the upcoming sub-sections.

3.1 FIR filter design using multiplier-less DA algorithm

Basically, selection of filter coefficients with respect to the input signal is the main idea taken behind implementation of the LUT-less DA architecture. The filter coefficients as shown in Table 1 is for a 4-bit filter order where, B3 B2 B1 B0 are the input bits and h[0] h[1] h[2] h[3] are the filter coefficients obtained from FDA tool in MATLAB. This LUT coefficient data is important factor to implement a LUT-less DA architecture. Instead of LUT, Multiplexer and slice blocks replace the LUTs. As it is seen in Table 1 with different values of B3 B2 B1 B0 filter coefficients are selected and added to get the desired results of the Filter. Therefore, the following diagram is shown in Fig 4 which is implemented in this paper in place of traditional DA architecture.

The structure shown in Fig 4 is considered to be basic design for FIR filter, $H_L(z)$. $H_L(z)$ is a low pass filter which will be used further to design reconfigurable hearing aid system. One of the advantages of FIR filter is the symmetricity of its coefficients due to which FIR filter $H_L(z)$ is used for interpolation and decimation operations.

3.2 Band formation in FIR filter

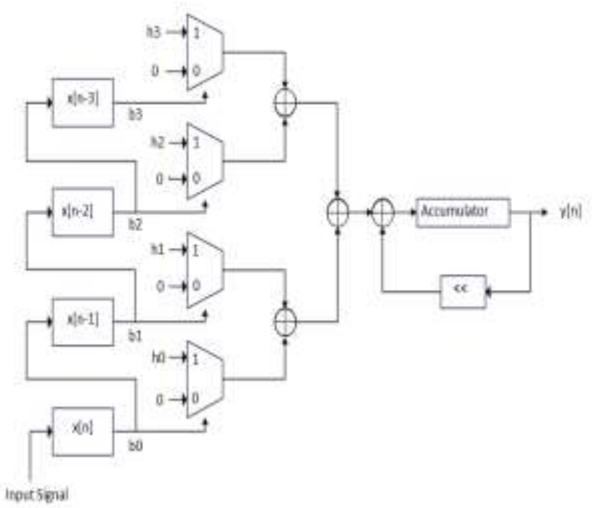


Fig. 4 A 4-Tap FIR Filter using Distributed Arithmetic Algorithm

Frequency Response Masking is considered in this proposed design. Multirate signal processing is used in these systems where up-sampling and down-sampling is performed to get variation in the width of the passbands. An interpolation by M will increase the sampling rate in time domain and Decimation by N will decrease the sampling rate in the time domain. $H_L(z)$ is the basic prototype filter with a bandwidth of $\pi/3$ using Distributed Arithmetic Algorithm. Further, a high pass filter $H_H(z)$ and mid frequency range filter $H_M(z)$ is designed from $H_L(z)$ as given in the equations (4) and (5).

$$H_H(z) = z^{-N/2} - H_L(z) \qquad (4)$$

$$H_M(z) = 1 - H_L(z) - H_H(z) \qquad (5)$$

where N is the length of the filter. The equations used for generating sub-filters from the prototype filters are given in Table 2.

TABLE 2 Equations for generating interpolated filters

$H_H(z^2) = z^{-N/2} - H_L(z^2)$
$H_M(z_2) = 1 - H_L(z^2) - H_H(z^2)$
$H_H(z^4) = z^{-N/2} - H_L(z^4)$
$H_M(z^4) = 1 - H_L(z^4) - H_H(z^4)$
$H_H(z^8) = z^{-N/2} - H_L(z^8)$
$H_M(z^8) = 1 - H_L(z^8) - H_H(z^8)$
$H_C(z^{2/3}) = z^{-N/2} - H_L(z^{2/3})$
$H_C(z^{4/3}) = z^{-N/2} - H_L(z^{4/3})$

Using the above equations, a 16-band FIR filter can be constructed as shown in the Fig. 5. The above diagram can be also be modified to implement 3-band/8-band/9-band/17-band FIR Filter system. The bandwidths of different sub-bands are $\pi/3$, $\pi/6$, $\pi/12$, $\pi/24$ and these bandwidths are selected differently as per design requirements in Table 3.

In the proposed design a comparison is done by implementing different sub-band system in order to check the minimum matching error which should be within the standard limit ± 3 dB for different Audiograms. Eight types of audiograms are taken into consideration in this paper. A comparison table is shown in the Table 4 from which it can be considered which filter is optimum for different Audiograms depending on their matching errors and delays. The maximum delay should be limited to 20ms otherwise it will cause a mismatch in synchronizing the visual lip reading and the audio being processed. If the audiogram of any human is within the range of 20dB then the person is not suffering from any hearing losses. But different person can have different audiograms for different hearing losses. Filter banks having lesser sub-bands shows more matching errors as seen in Table 4.

3.3 Implementation of the complete proposed system design

In order to achieve automatic reconfigurability study of the audiogram is necessary. The values achieved in different Audiograms are considered for auto- reconfigurabilty. The Audiogram can be divided into 3 parts based on frequency ranges 0-2.5 kHz, 2.5-5.5 kHz and 5.5 – 8 kHz. So, changes in graph in these regions can be taken care of by the allotted filter banks. When the graph is almost flat then filter banks with lower sub-bands are selected and when the graph shows sharp variation then filter banks with higher sub-bands are selected. An audiogram describes the mildest sound that can be heard at test frequencies of 250 Hz, 500Hz, 1kHz, 2kHz, 4kHz and 8kHz by the hearing impaired. The audiograms have six distinct hearing threshold values which are represented as ‘t_i’ at different octaves. The gradients are calculated as ‘$g_i = t_{i+1} - t_i$’ and the maximum gradient in any frequency range is considered for optimum filter bank selection. The maximum value of gradients in k^{th} region can be termed as slope, ‘S_k’. There will be 5 gradients which can be further divided (considering the normalised frequency ranges) as-

For Region 1, frequency range - 0 – 2.5 kHz
Slope Value = $S_1 = \max(|g_1|,|g_2|,|g_3|)$
For Region 2, frequency range - 2.5 – 5.5 kHz
Slope Value = $S_2 = \max(|g_4|,|g_5/3|)$
For Region 3, frequency range - 5.5 – 8 kHz
Slope Value = $S_3 = \max(|2g_5/3|)$

Four type of filter banks are chosen based on Table 4 looking at their audiogram matching performances, while 1st type of filter bank is a combination of $H_L(z)$, $H_H(z)$ and $H_M(z)$

which gives 3 sub-bands. The optimum filter bank selection is done from the slope values S_k of maximum compounded gradients. If S_k is within the upper threshold of 5dB, filter bank 1 is suggested in the region 'k'. Similarly, when S_k is within the upper threshold of 10dB, filter bank 2 is suggested in the corresponding region. Again, an upper threshold of 15dB suggests filter bank 3 and that of 30dB suggests filter bank 4 in the respective regions. The following are the design specifications –

TABLE 3 Design Specifications

Design Specifications	Value
Sampling Frequency	16 KHz
Passband ripples	0.05 dB
Stopband Attenuation	50 dB
Transition Bandwidth	0.175
Filter Order of the prototype	35
Bandwidth of subbands	$\pi/3$, $\pi/6$, $\pi/12$, $\pi/24$
Passband edges	0.2459
Stopband edges	0.4209

When the variations in the graph is sharp then most of the time higher sub-band system is considered while for the graphs with minimum variation lower sub-band systems are preferred. So, depending on the slope values, different filter bank will be selected for the respective regions. The whole concept mentioned above is programmed in the MATLAB. In MATLAB data is described in the floating-point form while described in the fixed- point form in this FPGA system. After quantizing the filter coefficients using 12-bit-width signed binary[23], we can obtain the final coefficients.

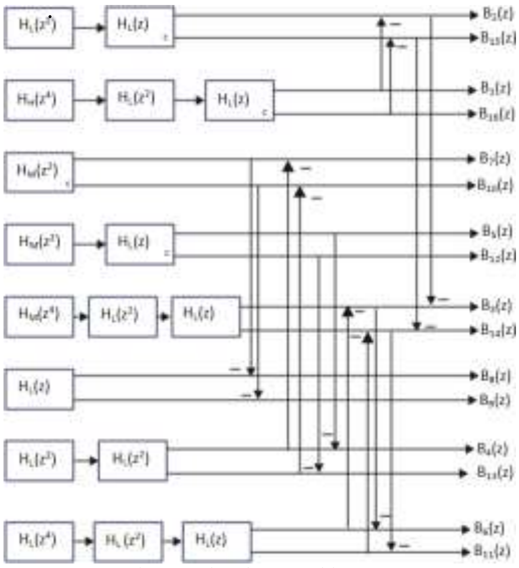


Fig 5. 16-band FIR architecture

TABLE 4. Comparison table of matching errors for different sub- band system

Hearing loss Type	6 band non uniform	8 band non uniform	9 band non uniform	16 band non uniform	17 band non uniform
	MME	MME	MME	MME	MME
Type 1	3.85	1.71	6.14	7	2.7
Type 2	4.14	1	2	2.52	2.71
Type 3	1.28	1	2.53	2.9	2.28
Type 4	3.52	1	1.28	1.42	1.28
Type 5	1.56	0.42	0.71	1.28	1.71
Type 6	5.52	0.71	1.57	1	1.28
Type 7	2.42	2.5	2	3.28	0.71
Type 8	1.14	1	1.42	2.57	1.42
Delay (ms)	15.37	7.67	12.88	6	11.78

In Table 4, TYPE 1 mild hearing loss in the high frequencies, TYPE 2 mild to moderate hearing loss in low frequencies, TYPE 3 mild hearing loss in all frequencies. TYPE 4 Most common type of hearing loss caused due to ageing specially in the consonant areas. TYPE 5 Common type of hearing loss seen in older workers working in noisy industries. TYPE 6 Moderate type of hearing loss. Patients lost much of loudness in speech. TYPE 7 High hearing loss in low frequencies, severe hearing loss in middle frequency and total hearing loss in high frequencies. TYPE 8 Profound or severe hearing loss

4. Results and Observations

In all the existing hearing aid automatic adoption to optimum bank selection was lacking, most of the hearing devices depend on manual interventions to achieve the best hearing bank. The proposed design can automatically reconfigure to the best-suited bank. The frequency responses and matching error results of some audiograms are shown in Fig. 6 (a) – Fig. 6 (l). The best filter bank is assigned for a particular audiogram for different regions of the signal without manual interference. Even the audiograms with sharp variations in different regions can also be matched successfully. The programmable block in Fig. 3 is a MATLAB function block which is an interface between MATLAB coding script and Xilinx environment. The logic used in this programmable block is Implementation section.

The proposed reconfigurable filter has very less complexity as the most power and area component i.e., multiplier is not used in the implementation. The proposed FIR filter with filter order ranging to 1024 (35 in this case) can be implemented using zero multiplier. The hardware complexity, delay and maximum matching error of different filter bank implementation method is listed in Table 6.

TABLE 5. Comparison results for all the Audiogram of the proposed system

Type	MME (in dB)	Filter bank selection			Delay (in ms)
		R1	R2	R3	
Audiogram 1	2.90	1	2	1	4.00
Audiogram 2	1.51	1	2	2	3.89
Audiogram 3	2.00	1	4	3	12.89
Audiogram 4	1.53	1	3	2	5.62
Audiogram 5	2.28	2	2	4	8.46
Audiogram 6	2.70	2	2	3	8.51
Audiogram 7	1.76	2	3	4	18.64
Audiogram 8	1.29	3	4	2	17.77

The Audiogram matching of the proposed system for different hearing losses and their matching error are shown in the previous figures, also in Table 5 the detailed results are shown where R1, R2, R3 denotes Region 1, Region 2, Region 3 of the input signal and the corresponding filter bank selected for that particular region. MME denotes

Maximum Matching Error in Table 5. The design and analysis of the proposed structure are done using the Xilinx System Generator in MATLAB R2017b where software simulation is done. The device utilization and power dissipation factors are evaluated using Xilinx Vivado 2018.3 software and it is shown in Fig 7 and Fig 8.

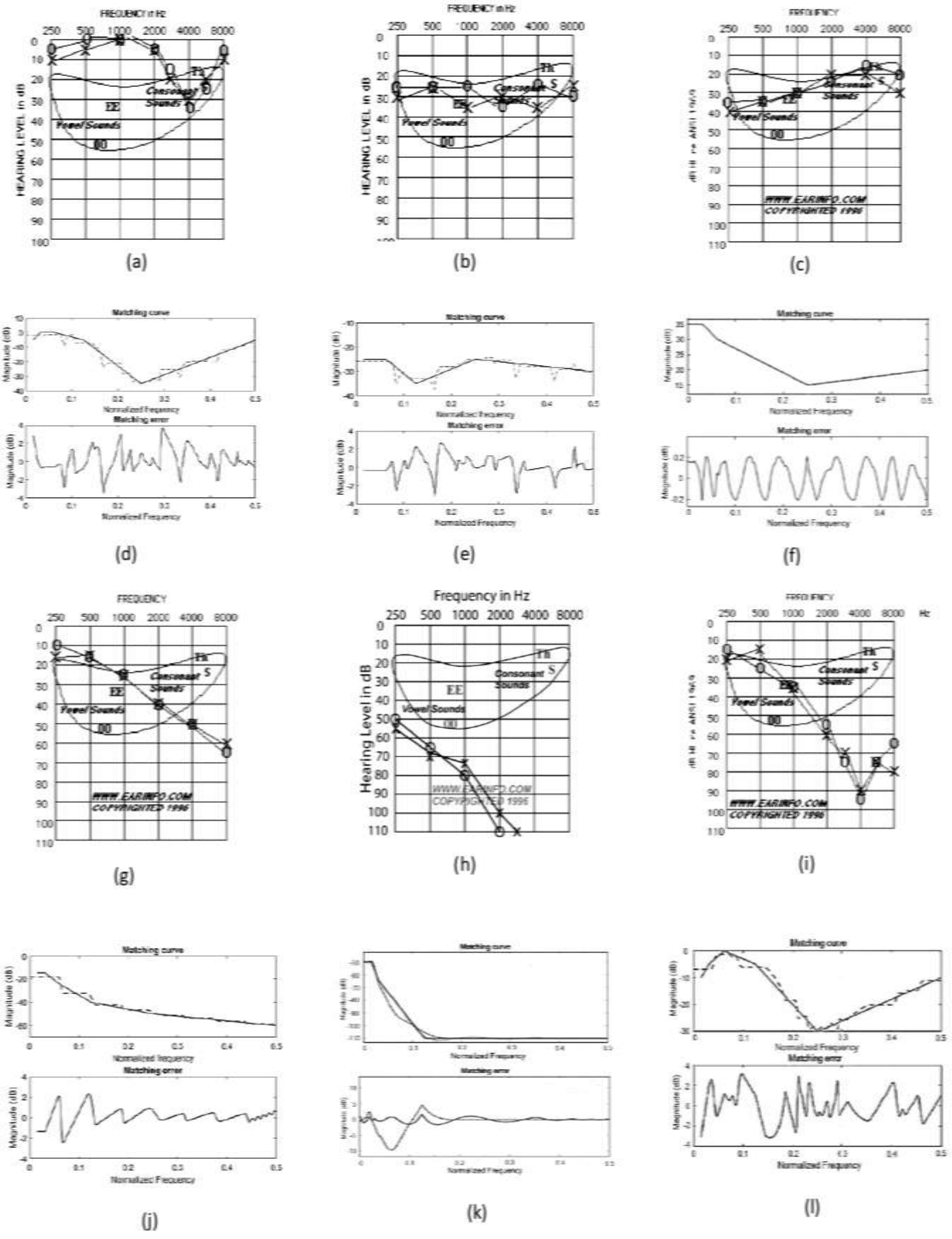


Fig 6. (a) – (d) TYPE 1 Audiogram, Matching curve & Matching error, (b) – (e) TYPE 3 Audiogram, Matching curve & Matching error, (c) – (f) TYPE 4 Audiogram, Matching curve & Matching error, (g) – (j) TYPE 5 Audiogram, Matching curve & Matching error, (h) – (k) TYPE 7 Audiogram, Matching curve & Matching error, (i) – (l) TYPE 8 Audiogram, Matching curve & Matching error.

Table 6 Comparison of the proposed system with existing methods

Method	Type of Filter	Number of Bands	MME	Max. Delay (ms)	No of Multipliers
Variable Bandwidth Filters [8]	Reconfigurable	3 to 8	2.47 dB	1.1	216
Interpolation [9]	Reconfigurable	17	5.62 dB	21.6	76
Frequency Response Masking [4]	Reconfigurable	16	2.26 dB	6	30
Quasi-ANSI Filter [18]	Fixed	18	3.26 dB	10	226
Sub-band Distribution [1]	Reconfigurable	3 to 8	4.56 dB	5	120
3-level octave interpolation [2]	Reconfigurable	3 to 17	2.39 dB	17.8	18
Proposed System	Reconfigurable	3 to 16	2.71 dB	18.6	-Nil-

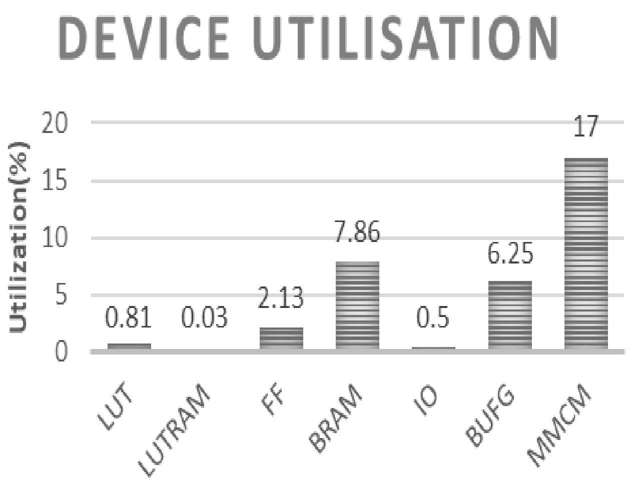


Fig 7. Device Resource Utilization of the proposed system

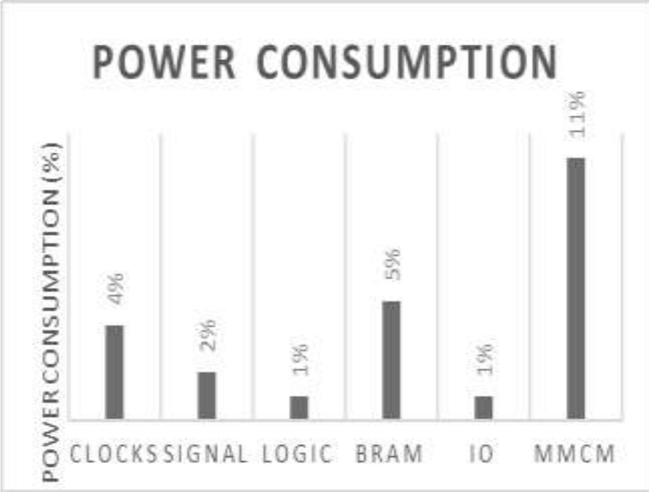


Fig 8. Power Utilization of the proposed system

5. Conclusion

A multi-filter reconfigurable hearing aid system is implemented using frequency masking response based on interpolation and decimation techniques which further divide the audio signal into several sub-bands. The prototype filter is implemented using Distributed Arithmetic Algorithm which is a multiplier-less technique making the system area- efficient and power-efficient. This reconfigurable filter is also automatic where the audio spectrum is divided into three regions in order to study the variations in the audiogram more properly and the best-suited filter bank is assigned to the hearing profile with maximum matching error 2.71 dB which is within the standard limit (± 3 dB). The delay analysis of the proposed filter is done and it is seen that the maximum delay is 18.6ms. A general comparison with existing filter bank designs is shown in Table 6 which depicts the efficient working of the proposed filter compared to the previously reported filter designs. Finally a multiplier-less filter bank for hearing aid is successfully implemented with sub-bands ranging from 3 to 16. This proposed design saves the device resource utilization upto 64% as seen in Fig 7 and power consumption upto 24% as presented in Fig 8. Thus, a simplified automatic reconfigurable filter is successfully implemented which avoids the tiresome manual matching of bands with hearing profiles. This method not only reduces hardware replacement with change in hearing profile but also proves to be efficient in terms of hardware utilization.

6. Future Scope

Further, in this system noise cancellation feature can also be introduced by studying accurate estimate of the noise spectrum as it varies over time. Various filtering methods like spatial filtering, adaptive noise cancellation, weiner filtering can be used to avoid the effect of noise on the hearing aid. One such method that can be effective in this proposed system is adaptive noise cancellation technique. Adaptive noise cancellation requires at least two directional microphones and, under ideal conditions, at least one microphone must be placed at the noise source. However, both microphones are placed near the head with one microphone picking up more speech than noise and the other microphone picking up more noise than speech. The output of the adaptive filter which is the noise signal will be subtracted from the combined signal where speech signal and noise signal are added. This will reduce the effect of noise to some extent and an improved speech-to-noise ratio will result within the upper range of 20dB, with improved intelligibility [23].

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