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Audiogram matching in hearing aid using approximate arithmetic

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Abstract
Filter banks are the major signal processing blocks that dissipate large amount of power in a portable digital hearing aid device. The power consumption can be reduced by replacing the power-hungry multipliers of the filter by power efficient approximate multipliers. This paper illustrates the application of an approximate multiplier for error tolerant hearing aid application. Frequency response masking approach is used for the development of a 10-band non-uniform approximate FIR filter bank with a minimum stop band attenuation of greater than 50 dB. Audiogram matching is done with audiograms of different types of moderate hearing loss and the matching error is computed. Simulation results show that the audiogram matching error falls within +/− 5 dB range.

Keywords Approximate multiplier · Filter bank · Hearing aid · Audiogram

1 Introduction

The inability of a person to hear sounds properly is due to various factors right from prenatal disorders to aging. The intensity of the softest frequencies one can hear is called the threshold of hearing for that frequency. The hearing thresholds are normally represented as an audiogram in which the hearing thresholds are plotted against frequency in an octave scale. The low sensitivity of the ear towards certain audio frequencies is called hearing loss. When the hearing loss falls below 25 dB appropriate corrective measures, often the use of hearing aids advised. Hearing aids, cochlear implants, and other assistive devices are popularly used in hearing loss treatment. Hearing loss can be classified as mild, moderate, severe, and profound based on the values of the hearing threshold (Deng, 2010).

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Hearing aids are preferred for the treatment of mild and moderate hearing loss. The hearing aid amplifies the audio frequencies in a selective fashion to match one's audiogram. Therefore, in hearing aid design, the arbitrary selection of frequencies or ranges of frequencies are done using a filter bank that divides the sound signal to different bands. The gain of specific bands can be adjusted to match the frequency response of the filter bank with a patient's audiogram.

Digital filters incorporated into the digital hearing aids for signal processing are the major power consuming element in wearable hearing devices. Digital Finite Impulse Response (FIR) filters are preferred over Infinite Impulse Response (IIR) filters considering their phase linearity, low coefficient sensitivity, and bounded input bounded output stability etc. (Lyons, 2011; Saramaki & Mitra, 1993). The major drawback of the FIR filter is its high computational complexity to achieve a specific value for transition bandwidth. It is suggested that the hardware complexity can be significantly reduced by the Frequency Response Masking (FRM) method (Lim, 1986; Lim & Lian, 1993; Saramaki, 1987). The FRM method substantially reduces the filter complexity by the use of cascaded sub-filters with lower complexity. The core component of FIR filter is the Multiply and Accumulate (MAC) unit. The multiplier unit decides the speed, size, and power dissipation of the MAC unit. The hardware complexity of the multiplier itself can be reduced by using approximate multipliers instead of exact multipliers.

Approximate arithmetic circuits are extensively reviewed in Mittal (2016) and Jiang et al. (2017, 2019). The important design parameters involved in approximate design are power, speed, and accuracy. Subjected to the tolerance level of the applications involved, significant improvement in power saving and computational speed can be achieved (Kulkarni et al., 2011; Liu et al., 2020). The research efforts by IBM (Nair, 2015), Intel (Mishra et al., 2014), Microsoft (Bornholt et al., 2015), Google (Jouppi et al., 2017), and other research groups (Chippa et al., 2010; Gupta et al., 2012) have shown that there are broad range of application domains exhibit significant intrinsic resilience. Error resilient applications like datamining, machine learning, wireless communication, and biomedical signal processing demands for signed approximate arithmetic algorithms and circuits. Most of the approximate multipliers proposed are focused on the design of unsigned multipliers, while studies on approximate signed multiplier have not been extensively reported. Accordingly, there is a renewed demand for further research in signed approximate arithmetic and its applications. The applications like FIR filtering are based on signed arithmetic operations and is extensively employed for audio processing in applications like hearing aids. Exact multipliers may be replaced with approximate multipliers in the design of low power high performance FIR filters by trading off accuracy with speed and power.

Sound intensity resolution of the ear is limited by the logarithmic response characteristic makes it insensitive to small relative changes in the intensity of the sound and makes the ear an error tolerant system. Filter bank design with signed approximate arithmetic is not a well explored area of research. The importance of approximate arithmetic in data processing and other evolving computing paradigms makes this area of research very important. The paper addresses the pertinence of approximate arithmetic in audio signal processing for application in hearing aid design.

In this paper, a 10-band FRM-FIR filter bank is developed using signed approximate multiplier and is used to demonstrate the efficacy of approximate multiplier in low power digital hearing aids. The paper is organized as follows. A brief description of 10-band FRM based FIR filter bank design is given in Sect. 2. The digital hearing aid, audiogram, and audiogram matching are briefly explained in Sect. 3. Section 4 describes the design of 10-band FRM-FIR filter bank using approximate multiplier. The results of the audiogram matching experiment to verify the efficacy of the approximate multiplier for digital hearing aid application are presented in Sect. 5 and in Sect. 6, the conclusions are presented.

2 Design of 10-band FRM-FIR filter bank

Frequency Response Masking (FRM) approach got wide acceptance in designing FIR filters of sharp transition band with less hardware complexity (Gustafsson et al., 2000, 2001; Lim, 1986; Lim & Lian, 1993; Saramaki, 1987). A combination of wide transition band filters with less complexity are generally identified as model filters and masking filters are used to derive the required sharp transition bands. An upsampling of the model filter will compress the frequency response of the filter. The desired sharpness of the transition band can be achieved by choosing an appropriate upsampling factor. This results in extra frequency bands called images and are removed by cascading the masking filters with the model filters in the FRM method. The FRM approach allows the implementation of arbitrary band pass FIR filters with less hardware complexity since many of the filter coefficients are zero.

The FRM method utilizes a model filter G(z) and masking filters $F_0(z)$ and $F_1(z)$. The periodic model filters $G(z^L)$ are generated from G(z) by upsampling by a factor L. In addition, the complement of the periodic model filter is given by

$$G_C\left(z^L\right) = z^{-K} - G\left(z^L\right) \tag{1}$$

where $K=(N-1)/2$, N is the length of the filter $G(z^L)$.

The transfer function of the FIR filter thus designed is,

$$H(z) = G\left(z^L\right)F_0(z) + G_C\left(z^L\right)F_1(z) \tag{2}$$

In FRM method, one or several pass bands of the periodic model filters are extracted by the two masking filters, $F_0(z)$ and $F_1(z)$. In the present implementation, the model filter G(z) is upsampled by different values i.e., 2, 4, 8, 16 to generate the periodic model filters. The model filter is G(z) and the interpolated filters $G(z^2)$, $G(z^4)$, $G(z^8)$, and $G(z^{16})$ are the periodic model filters required for the design of a 10-band filter bank. The masking filters are selected as $F_0(z)=G(z)$ and $F_1(z)=G_C(z)$. This restricts the filter coefficients of the masking filters to be the same as that of the model filter. The higher order filters are appropriately cascaded to generate the desired frequency response of the sub filters of the filter bank when masked with G(z) or $G_C(z)$. The transfer functions of the different sub-filters thus generated are listed in Table 1.

The transfer function for the lower sub filters is given by

$$B_i(z) = \begin{cases} P_i(z) & i = 1 \\ P_i(z) - P_{i-1}(z) & i = 2,3,4,5 \end{cases} \tag{3}$$

The transfer function of the upper sub filters is given by

$$B_i(z) = \begin{cases} P_i(z) & i = 10 \\ P_i(z) - P_{i+1}(z) & i = 6,7,8,9 \end{cases} \tag{4}$$

The structure of the 10-band filter bank is illustrated in Fig. 1. In the filter bank structure, the lower and upper five sub-bands are symmetric. The lower bands are complementary

Table 1 Transfer function of different sub-filters

Sub-filter	Transfer function
$P_1(z)$	$G(z^{16})G(z^8)G(z^4)G(z^2)G(z)$
$P_2(z)$	$G(z^8)G(z^4)G(z^2)G(z)$
$P_3(z)$	$G(z^4)G(z^2)G(z)$
$P_4(z)$	$G(z^2)G(z)$
$P_5(z)$	$G(z)$
$P_6(z)$	$G_C(z)$
$P_7(z)$	$G(z^2)\ G_C(z)$
$P_8(z)$	$G(z^4)G(z^2)G_C(z)$
$P_9(z)$	$G(z^8)G(z^4)G(z^2)G_C(z)$
$P_{10}(z)$	$G(z^{16})G(z^8)G(z^4)G(z^2)G_C(z)$

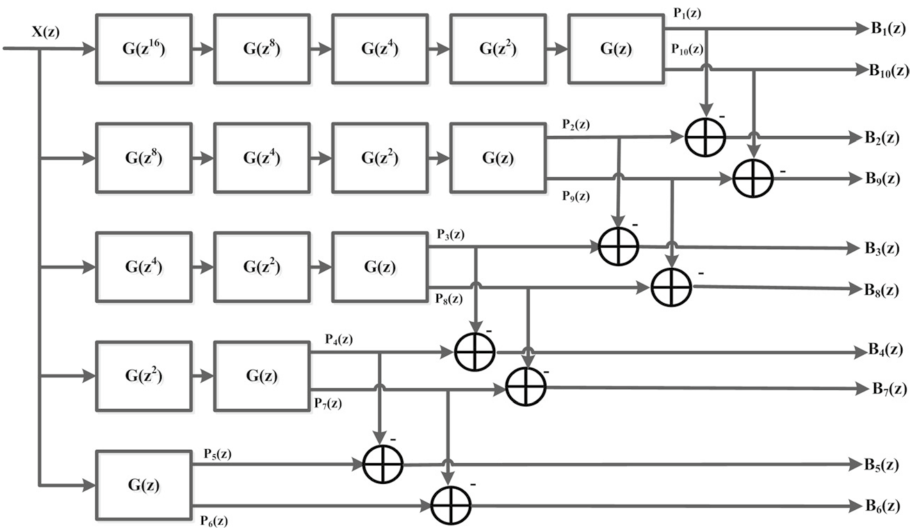


Fig. 1 Structure of 10-Band FRM-FIR filter bank

bands of the upper bands formed by replacing G(z) with its complement $G_C(z)$ as the masking filter. The outputs of the sub-bands are termed as $B_i(z)$, $i=1, 2, \ldots, 10$, where $B_1(z)$ to $B_5(z)$ are formed by the outputs $P_1(z)$ to $P_5(z)$ selected by the masking filter G(z) and $B_6(z)$ to $B_{10}(z)$ are formed by the complimentary outputs $P_6(z)$ to $P_{10}(z)$ selected by the masking filter $G_C(z)$.

3 Digital hearing aid and audiogram matching

The function of the hearing aid is to improve the functionality of a defective ear to match with that of normal ear. If the hearing threshold falls in the normal hearing levels, the hearing is considered to be normal. However, for people with impaired hearing, the hearing thresholds become high at certain frequencies causing hearing loss i.e., they have low sensitivity towards certain frequencies. Therefore, the input sound signals have to be amplified according to the sensitivity of the ear so as to bring all the frequencies within the hearing thresholds of respective frequencies. The hearing loss of most of the people occurs at specific range or bands of audio frequencies depending on the cause of hearing loss. For example, hearing loss due to old age and noise pollution appears as the slow response of the ear to high frequencies of the consonant sounds. For this reason, the audio signal is separated into many distinct bands, often octave bands, and processed by a filter bank approach that can meet different types of hearing loss with acceptable delay.

With extensive signal processing and with a desire to have tiny un-obstructive devices, the hearing aid needs to address the longevity of its battery. Many of these portable devices run on 1.3 V battery drawing 2 mA and have battery life span of about 100 h. The filter banks proposed for hearing aid applications (Deng, 2010; Haridas & Elias, 2016; Lian & Wei, 2005; Ma et al., 2019; Sebastian & James, 2015; Wei & Lian, 2006; Wei et al., 2018) follows exact arithmetic. However, audio signal processing is an error tolerant signal processing application where approximate arithmetic may deliver satisfactory results because audio signal processing mainly concerns with perceived sound quality rather than absolutely precise numerical results. In the numerically intensive domain of audio signal processing, the substitution of approximate multipliers in place of exact multipliers can provide significant power savings over the full precision counterparts. In this process, the floating-point filter coefficients are converted into signed integers first and stored for subsequent processing. Both the incoming data and the filter coefficients are approximated and multiplied in the approximate MAC unit to perform the filtering operation.

Audiogram is a graph that shows the softest sounds a person can hear at different pitches or frequencies. Audiogram represents the hearing thresholds at different audio frequencies marked in an octave scale. This hearing threshold varies from person to person. Generally, the hearing levels are measured at octave frequencies such as 250 Hz/ 500 Hz/1000 Hz/ 2000 Hz/ 4000 Hz/ 8000 Hz. The illustration of audiograms was shown in Fig. 2. The symbol "o" is used to represent the responses for the right ear and "x" is used to show the responses for the left ear.

The region inside the dashed curve represents an area in which all the phonemes of all languages can be represented and is called the speech banana. The audiogram for normal hearing is shown in Fig. 2a. This audiogram shows the normal hearing ability of the left and the right ears. Figure 2b shows the audiogram of hearing impaired who suffers from hearing loss at high frequencies. The hearing threshold of high frequency sound is higher than the normal speech intensities.

To compensate for this hearing loss, it is necessary to selectively amplify sounds at high frequencies. Table 2 shows the classification of hearing loss according to the degree of hearing loss. It should also be noted that a hearing loss beyond the moderate loss amounts

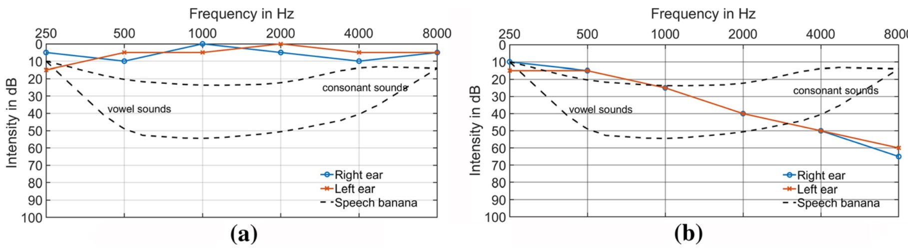


Fig. 2 Examples of audiograms. **a** Audiogram of normal hearing. **b** Audiogram of high frequency hearing loss

Table 2 Various degrees of hearing loss

Type of hearing loss	Hearing loss range
Mild	20–40 dB
Moderate	41–55 dB
Moderately severe	56–70 dB
Severe	71–90 dB
Profound	>90 dB

to 100% loss of speech information. Digital hearing aids can bridge the gap between hearing impairment and one's ability to recognize the sounds and conversation they hear. Hearing aids amplify the sound signals so that one can understand what is being said to him. The hearing aids are tuned in such a way the gain of each frequency band is adjusted to match one's audiogram as closely as possible. That is the hearing aid compensates for the hearing loss and makes one's hearing ability as that of a normal person as closely as possible. Hearing aids have a large beneficial effect in helping adults with mild to moderate hearing loss to take part in everyday life. The mild and moderate hearing losses can easily be compensated with simple hearing aids and other losses preferably use cochlear implants for treatment.

4 Approximate FRM-FIR filter bank for hearing aid

4.1 Selection of approximate multiplier for filter design

There are two major approaches in the design of approximate multipliers. The first approach is the wordlength reduction of the input operands to use with predefined fixed width multiplier (Hashemi et al., 2015; Narayanamoorthy et al., 2014; Ramya & Moorthi, 2019; Zendegani et al., 2016). The second approach is the application of approximate methods like truncation, deletion, rounding, and compression at the generation or at the processing stage (Garofalo et al., 2008; Jiang et al., 2015; Kulkarni et al., 2011; Leon et al., 2017; Liu et al., 2017; Momeni et al., 2014; Petra et al., 2009; Stine & Duverne, 2003; Venkatachalam & Ko, 2017). Truncation offers better power saving compared to the other approaches and also prevents the growth of word length. Booth multipliers forms the major building block of many signed multipliers. Booth encoded multipliers generate partial products using a Booth encoder and are accumulated using compressors and the resultant numbers are added in a fast adder to generate the final product. The approximation may be done at the generation stage using an approximate Booth encoder (Liu et al., 2017). The additional delay incurring in the operation of odd multiples of the multiplicand is the disadvantage of radix-8 Booth multiplier. An approximate 2-bit adder is used in Jiang et al. (2015) to generate the triple multiplicand without carry propagation in a radix-8 multiplier. In Leon et al. (2017), approximation is carried out using a hybrid radix Booth encoding at the partial product generation stage.

For most of the truncation multipliers, unsigned implementations are unavailable in literature. Recently, Ramya and Moorthi (2019) proposed a signed implementation of their unsigned approximate multiplier. The modified shift derive logic adopted in the signed design makes use of sign-extension encoder and simple inverter based control logic. This avoids the use of complex Leading- One- Detector and N-bit encoder circuits used elsewhere which gives more area and power reduction. RoBA architecture proposes rounding of the input operand to a power of 2 such that multiplier is replaced by shifter and adder units. Two signed implementations of RoBA architecture are proposed in Zendegani et al. (2016). Due to the better circuit performance offered by the multipliers based on wordlength reduction of input operands, Signed RoBA (S-RoBA), Approximate Signed RoBA (AS-RoBA), and the signed multiplier proposed by Ramya and Moorthi are considered here for the implementation of FIR filter bank.

The performance of the approximate multipliers is evaluated based on their error and circuit characteristics. The error metrics, normalized mean error distance (NMED) and mean relative error distance (MRED) are evaluated by simulating the multipliers in MATLAB with two sets of hundred thousand random numbers with uniform probability as the input of the multipliers. For the verification of circuit characteristics, the designs which are considered here are synthesized in Cadence RTL Compiler using gPDK 90-nm CMOS technology with typical library settings. The post-synthesis circuit performance characteristics such as power, area, and critical-path delay are measured with respect to the maximum achievable frequency of the exact multiplier as reference. Since leakage currents are also significant in contributing to the total power in 90-nm technology, combined design metrics for characterizing circuit performance are Energy or Power-Delay Product (PDP), Energy-Delay Product (EDP), and Power-Delay-Area product (PDA) (Jiang et al., 2017; Sengupta & Saleh, 2007). The circuit metrics PDP, EDP, and PDA are calculated from the measured circuit metrics and tabulated in Table 3 along with the error metrics.

The results tabulated in Table 3 show that the signed multiplier proposed by Ramya and Moorthi improves the Energy (PDP), EDP, and PDA up to 37%, 26%, and 62% respectively, compared to those of the exact Baugh Wooley multiplier. Based on the overall performance, the approximate multiplier proposed by Ramya and Moorthi is selected for the implementation of the FRM-FIR filter bank.

4.2 Selection of the filter length for 10-band FRM-FIR filter bank

In approximate filtering, the approximation may manifest as reduced attenuation in the stop band of the filter. Therefore, the number of approximated filter coefficients used in the filter design becomes an important design factor. Therefore, it is important to find out the minimum filter length which satisfies the design criteria of minimum attenuation required in the stop band.

The minimum attenuation in the stop-band is calculated for the exact and approximate filter bank for different filter lengths and is plotted in Fig. 3. Results show that as the number of filter coefficients increases, the attenuation in the stop band decreases. It is because of the increased noise resulting from a large number of approximate filter coefficients.

Table 3 Error and circuit characteristics of signed approximate multiplier designs

Signed multiplier designs	NMED (%)	MRED (%)	PDP (pJ)	EDP (pJ. ns)	PDA (pJ. μm^2)
Ramya and Moorthi (2019)	0.032	0.52	2.21	11.80	8867
S-RoBA (Zendegani et al., 2016)	0.172	2.88	2.83	14.82	15,961
AS-RoBA (Zendegani et al., 2016)	0.173	2.89	2.76	14.21	14,380
Baugh and Wooley (1973)	–	–	3.51	16.005	23,443

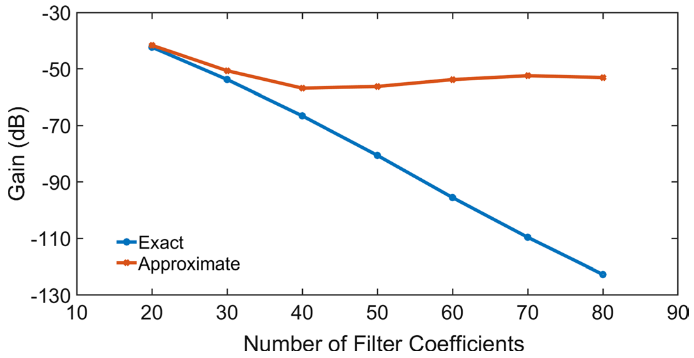


Fig. 3 Variation of attenuation with respect to filter length

From Fig. 3, it can be seen that 40 tap for the filter is very preferable since it provides less number of taps and high attenuation in the stop band.

4.3 Selection of transition bandwidth

Transition bandwidth is also a deciding factor in the design of filter banks. Minimum transition band width is preferred to ensure the sharp band edges for the different band pass filters in order to avoid overlapping of the constituent filters of the filter bank. The variation of attenuation with transition bandwidth is shown in Fig. 4. It is seen that the minimum normalized transition bandwidth that can be selected for the design is 0.2 in order to maximize the difference between stop band and pass band attenuation.

4.4 Filter bank specifications

The prototype filter G(z) is designed in MATLAB using the least square method. The normalized transition band is selected as 0.4–0.6. The normalized transition bandwidth of the

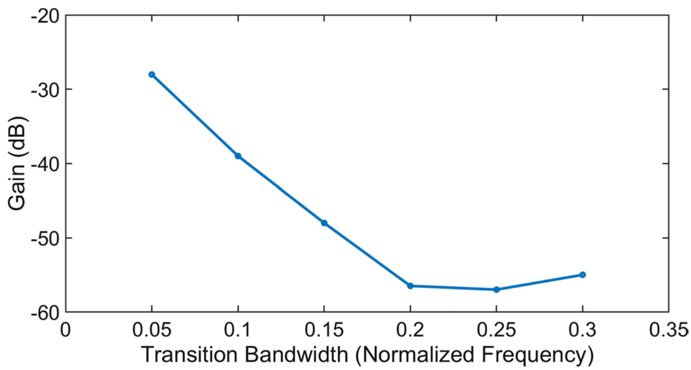


Fig. 4 Variation of attenuation with transition bandwidth

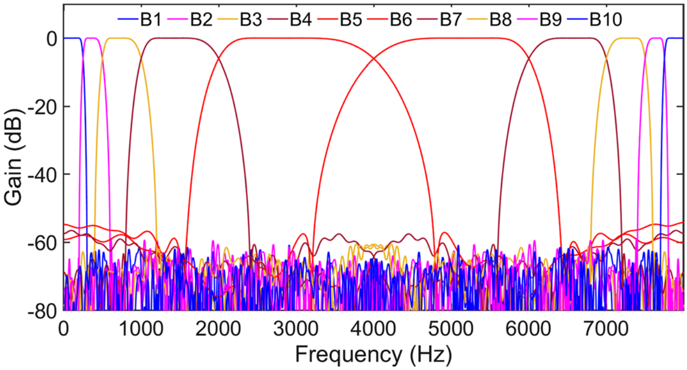


Fig. 5 Frequency response of 10-band approximate FRM-FIR filter bank

Table 4 Hearing threshold for various frequency ranges

Types of Hearing Loss	Frequency (Hz)							
	250	500	1000	2000	3000	4000	6000	8000
	Hearing threshold (dB)							
Case (i)	15	15	25	40	–	50	–	60
Case (ii)	25	25	25	35		25		30
Case (iii)	35	35	30	25	–	15	–	20
Case (iv)	50	55	60	55	–	55	–	70
Case (v)	20	20	20	20	30	40	30	20

prototype G(z) is fixed as 0.2 and the order of the filter is selected as 40 to ensure stop band attenuation greater than 50 dB. The proposed filter bank covers the frequency ranges from 0 to 8 kHz with 10 non-uniform bands. The sampling frequency selected as 16 kHz. The frequency response of the 10-band approximate filter bank with integer coefficients designed in MATLAB is shown in Fig. 5.

5 Audiogram matching for 10-band FRM-FIR approximate filter bank

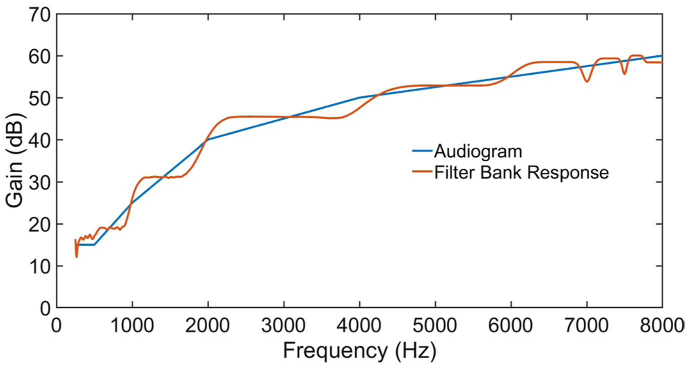
The mild and moderate hearing losses are very common and require adequate compensation with hearing aids. Considering the magnitude of the moderate hearing loss approximate methods can be utilized in realizing the required filter bank. A 50 dB to 60 dB compensation is the maximum requirement over the audible range. The 10-band approximate filter bank is used to demonstrate the efficacy of the approximate multipliers in hearing aid applications. For presenting the results, audiograms for the mild to moderate hearing loss pattern are considered. The audiograms are downloaded from Independent Hearing Aid Information, a public service by Hearing Alliance of America (http://www.earinfo.com, http://www.asha.org). Typical audiograms for mild and moderate hearing loss are selected for the study and the matching errors have been calculated to find the adequacy of application of approximate methods in audio processing in hearing aid applications. For the purpose of demonstration, five typical audiograms have been selected and their hearing thresholds are listed in Table 4 for different frequency ranges.

5.1 Case (i): Moderate loss at high frequencies

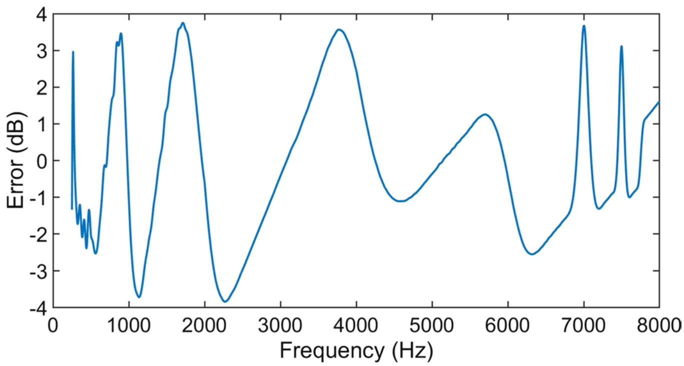
The moderate hearing loss at high frequencies is known as Presbycusis, occurs due to the effects of aging on the inner ear and related structures. The hearing sensitivity is better at low frequencies, and falls at high frequencies. People with this kind of loss may miss so many consonants and they have difficulty in distinguishing one word from another. Figure 6a shows audiogram matching for moderate hearing loss at high frequencies. The audiogram matching error plot is shown in Fig. 6b. The maximum audiogram matching error obtained is within +/− 4 dB.

5.2 Case (ii): Mild loss at high frequencies

The hearing loss dips mainly in the consonant letters. People with this kind of hearing loss have difficulties in hearing s's, z's, th's, v's, and other soft, high frequency consonants and will have trouble in noisy conditions. Figure 7a shows the audiogram matching for mild



(a)



(b)

Fig. 6 **a** Audiogram matching—moderate loss at high frequencies. **b** Plot of matching error loss at high frequencies. The corresponding matching error plot is shown in Fig. 7b. The maximum matching error is found to be within + /− 5 dB.

5.3 Case (iii): Mild to moderate hearing loss at low frequencies

An overall loss of loudness with difficulty in hearing the vowel sounds makes the conversation unintelligible at low frequencies. Thus, very close distance conversations are necessary. People with this kind of hearing loss will have normal hearing at high frequencies. Hearing-aids are needed if the hearing threshold in both the ears at 500 Hz is around 35 dB or more. Figure 8a shows the audiogram matching for mild to moderate hearing loss at low frequencies. The corresponding matching error plot is shown in Fig. 8b. The maximum matching error is within + /− 3 dB range.

5.4 Case (iv): Moderate loss at all frequencies

The hearing loss will have two effects. First, because the hearing sensitivity is worse than the volumes used in normal conversation, one will lose much of the loudness

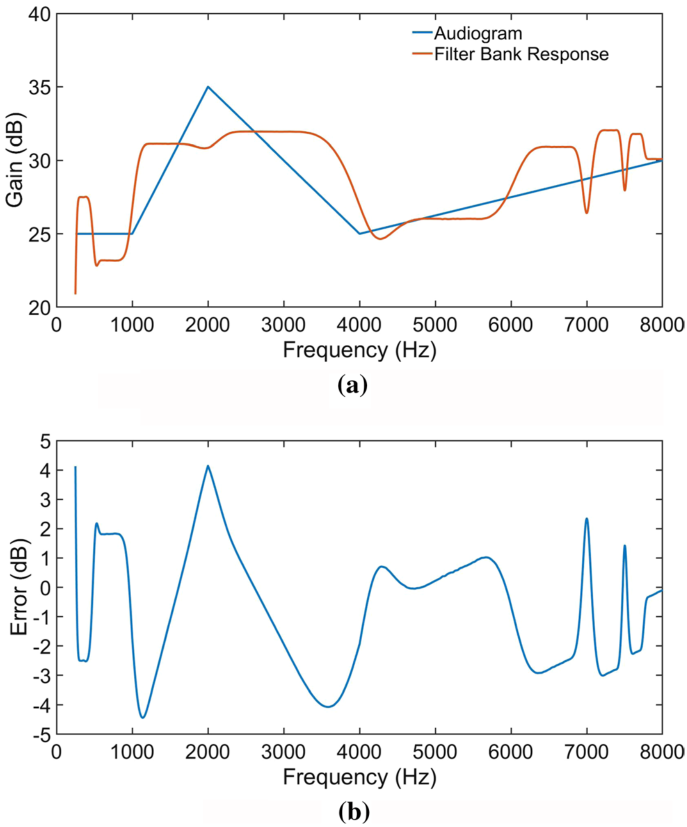


Fig. 7 **a** Audiogram matching—mild loss at high frequencies. **b** Plot of matching error

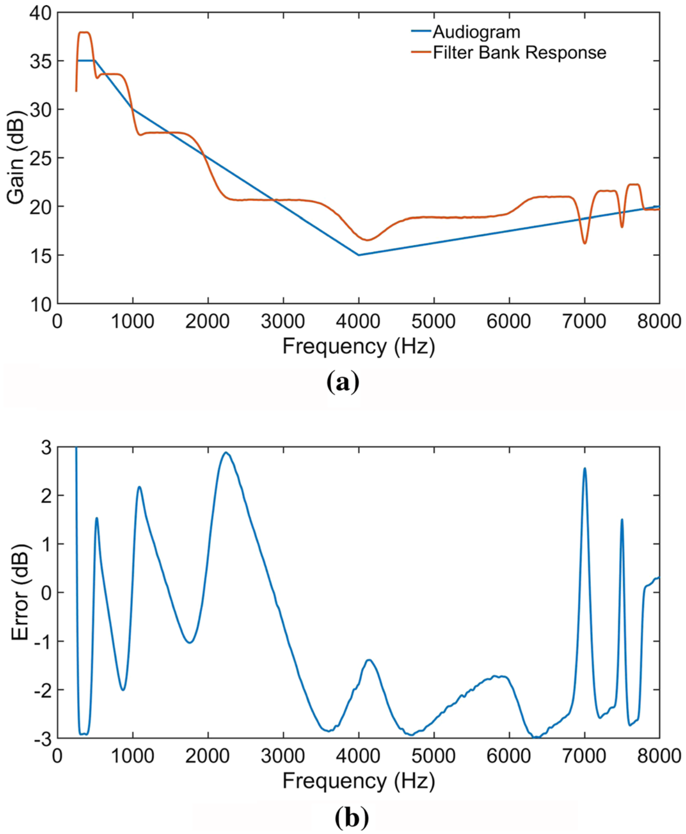


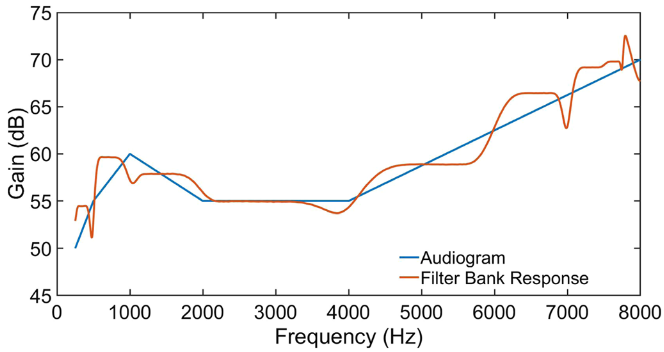
Fig. 8 **a** Audiogram matching—mild to moderate hearing loss at low frequencies. **b** Plot of matching error

of speech. The normal conversational volume is a bare whisper in this case. Second, because of the missing of so much consonant information, it is going to confuse those words that one does hear with this type of impairment. Figure 9a and b shows the audiogram matching and matching error plot respectively for moderate hearing loss at all frequencies. In this study, the maximum matching error is within + /− 4 dB range.

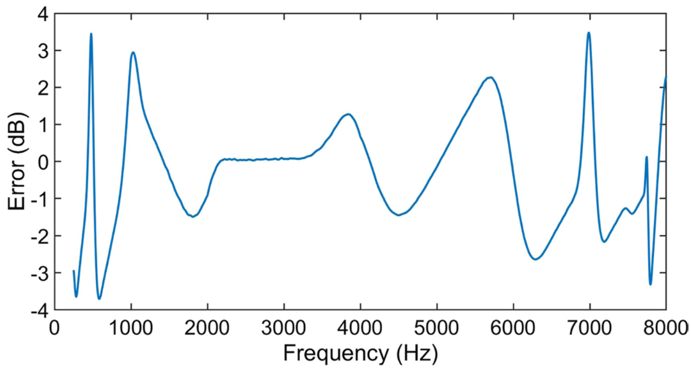
5.5 Case (v): Hearing loss at mid frequencies

Noise Induced Hearing Loss (NIHL) is the most common type of hearing loss seen in old age workers working in noisy industries. It occurs due to the effects of too much of noise for too many years on the inner ear and related structures. This type of hearing loss mostly occurs at 4 kHz. For this kind of hearing loss, the sounds have to be amplified at mid frequencies. Figure 10a and b shows the audiogram matching and the matching error plot respectively for hearing loss at mid frequencies. The maximum matching error obtained is within + /− 8 dB range.

A comparison of matching errors computed with exact multiplier and that with an approximate multiplier was done in order to examine the possible deterioration in the case of approximate method. The absolute values of maximum matching error measured with



(a)

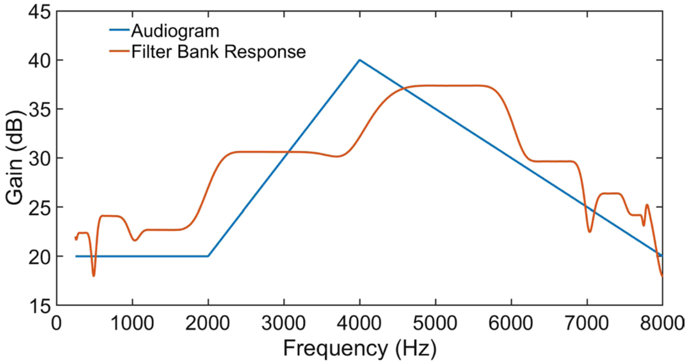


(b)

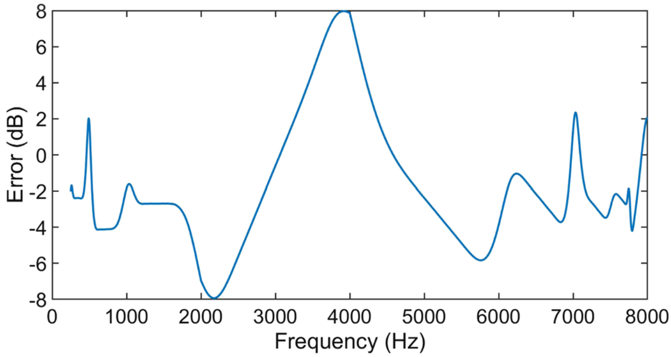
Fig. 9 **a** Audiogram matching—moderate hearing loss at all frequencies. **b** Plot of matching error

both the methods were tabulated in Table 5 for all the audiograms considered under the study. It is found that the values of matching error for the approximate method are equal to the exact method. To demonstrate the same, the matching error computed by both the exact and approximate method at different frequencies for the audiogram with mid band loss (case (v)) is plotted in Fig. 11.

The observations show that the approximate method illustrated achieves high computational accuracy apart from the area and power efficiency. The large matching error observed in case (v) is due to the fact that the fifth band of the filter which covers the frequency band 2 kHz to 4 kHz is unable to match the large loss in this range. Since the filter is designed based on uniform sub-sampling of division by 2, redesigning the filter with increased or decreased number of bands may not give satisfactory results. In addition, increasing the number of bands is not a good solution in the case of low power design. The option left is that of adjusting the band edge of the fifth band to align as closely as possible with the sharp transition part of the audiogram as shown in Fig. 12a. This can be done by redesigning the filter with a new transition band with an offset from the current design. The matching error can be brought to within ± 5 dB by changing the transition band of G(z) from 0.4–0.6 to 0.45–0.65. It can be seen that the maximum matching error is brought within ± 4 dB as shown in Fig. 12b.



(a)



(b)

Fig. 10 **a** Audiogram matching—hearing loss at mid frequencies. **b** Plot of matching error

Table 5 Comparison of maximum matching error-exact and approximate methods

Audiogram	10-band (Exact) (dB)	10-band (Approximate) (dB)
Case (i)	3.78	3.79
Case (ii)	4.20	4.24
Case (iii)	2.85	2.85
Case (iv)	3.65	3.64
Case (v)	7.94	7.96

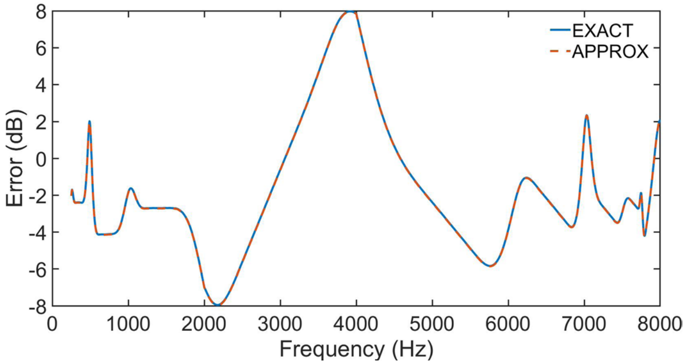


Fig. 11 Matching error comparison-exact and approximate methods

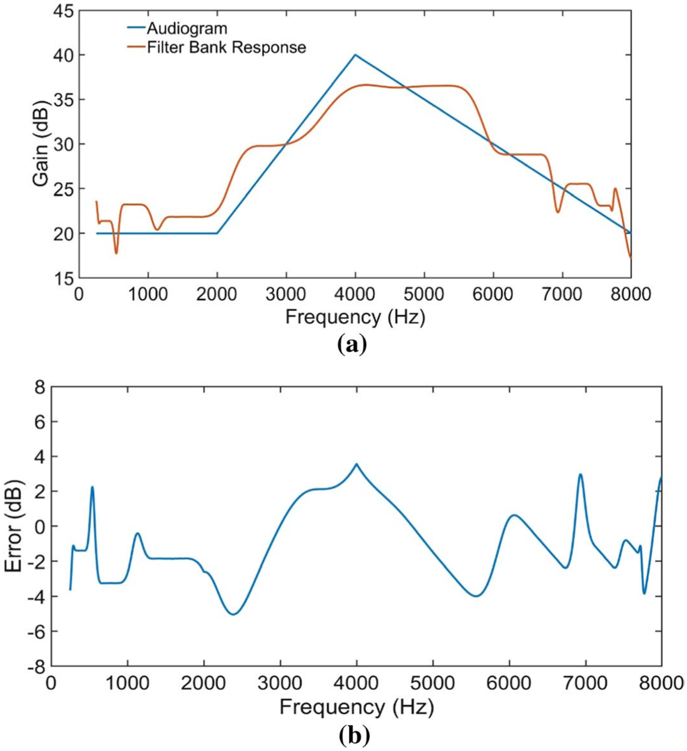


Fig. 12 **a** Audiogram matching—hearing loss at mid frequencies with transition band 0.45–0.65. **b** Matching error plot

6 Conclusions

It is demonstrated that approximate arithmetic can be effectively used to devise low cost hearing aids. By adopting frequency response masking approach, a 10-band FIR filter bank is developed using approximate multiplier to selectively amplify the audio frequencies. Audiogram matching is performed to ascertain the adequacy of approximate arithmetic for speech processing in hearing aid design. It is found that the design is perfectly suitable and found to offer comparable matching error performance with other reported works. The approximate technique is a good option for implementing a low cost and low power hearing aid for most common types of hearing loss. However, the approximation method is hampered by the low levels of attenuation in the stop band. The filter length limits the application of the approximate method to a limited range of audiograms with less than 55 dB or less difference between the maximum and minimum threshold values. This sets a limit on the approximate approach to compensate for severe and profound hearing losses in hearing aid design.

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