# Efficient Farrow Structure Based Bank of Variable Bandwidth Filters for Digital Hearing Aids

Nisha Haridas, Elizabeth Elias
Department of Electronics and Communication Engineering
National Institute of Technology Calicut
Calicut, Kerala 673601
Email: nisha_p120093ec@nitc.ac.in

***Abstract*—This paper proposes an efficient bank of filters derived from variable bandwidth filter structure, for audiogram matching in a digital hearing aid. The variable bandwidth filter is designed using Farrow structure, for lower hardware complexity. The magnitude and band edge frequencies of the filters are independently adjustable which allows this arrangement to successfully fit a given audiogram with as minimum number of bands as possible.**

***Index Terms*—Digital Hearing Aid, Farrow structure, Variable Bandwidth Filters, Audiogram**

## I. Introduction

A hearing aid is used to selectively amplify sounds for a person whose auditory threshold is below the normal level. Auditory thresholds are the minimum magnitude that one can perceive and are represented by the audiograms [1]. An audiogram gives hearing capability in decibels (dB) for various frequencies (usually 250 Hz to 8 kHz) and are used to understand the hearing loss patterns. Hearing aids are designed for a particular hearing loss such that there is minimum deviation, within tolerance, from that audiogram. In a digital hearing aid, the selection of the band of frequencies for amplification is done using a digital filter.

Higher flexibility, minimum hardware, low power consumption, low delay and linear phase (to prevent distortion) are the required characteristics of the digital hearing aid. Significant amount of study is available on bank of filters designed for audiogram matching. Initial approaches are based on uniform subbands. Since, humans perceive loudness on a logarithmic scale [2], non-uniform filter banks are better suited, so that the matching can be achieved with lesser number of sub-bands, if possible.

In [3], a nonuniform digital FIR filter bank for hearing aid applications, using eight subbands, which are formed with the help of frequency-response masking (FRM) technique, by using one or more interpolated masking filters, is given. This technique has relatively long filter delay which may affect lip-reading. In [4], a 16-band filter bank structure using half-band FIR filters as prototype filters, in combination with FRM technique is used. This reduces the filter delay by more than 50%. However, using fixed sub-bands makes the hearing aids less flexible. Hence, in [5] a three- channel variable filter bank structure is introduced. It considers designs that could tolerate non-linearities in phase. In [6], a system which can provide filter bank with adjustable number of subbands and adjustable bandwidths is proposed. Once the subbands and sections are designed, the filter bank can be selected from the separate sections. The best matching depends on the design of these sections and thus limits the flexibility. In [7], a variable bandwidth filter using sampling rate conversion, without altering the filter order or filter coefficients is proposed. A fixed length FIR filter is designed initially, whose characteristic bandwidth is then changed by modifying the bandwidth ratio, given as input to an interpolation filter. Using this filter structure and by varying the bandwidth ratio, a bank of filters that processes different subbands, are realised. Here, the complexity of the structure is high.

This paper proposes the use of variable bandwidth filter designed using Farrow structure, to be used as the basic unit for the hearing aid. This structure is an efficient way to realize reconfigurable systems and can result in minimum hardware complexity filters for hearing aid application.

The rest of the paper is organized as follows. In section II, variable bandwidth filter using Farrow structure is given. The proposed method is discussed in section III. Comparison of the method with that of [7] is given in section IV using a design example. Conclusion is drawn in section V.

## II. Variable Bandwidth Filter using Farrow structure

Variable bandwidth (VBW) filter can be effectively realized using Farrow structure [8], where the overall filter response is a weighted linear combination of fixed linear phase FIR subfilters, as given in Figure 1. The variability is achieved with simple update of adjustable parameters, directly determined by bandwidth. This renders minimum hardware implementation complexity [9] [10].

The filter structure shown in Figure 1 can be designed to meet the specification for each of the variable bandwidth parameter, b, such that there is complete control on the desired specification and performance. A properly chosen offset is given to the bandwidth parameter, making it $b-b_0$, which helps to reduce the round-off noise, makes the filter coefficients smaller and finds a lower bound on the filter order [10]. Here, the offset is chosen as mid-point between the highest and lowest required bandwidth. The transfer function, H(z,b) can

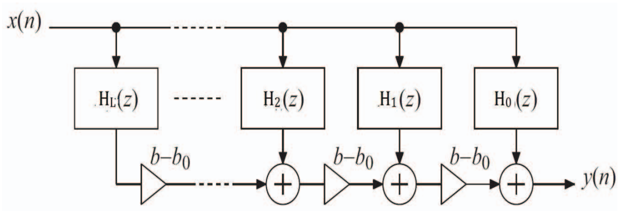


Fig. 1: Farrow structure based VBW for adjustable factor $b - b_0$

be expressed as,

$$H(z,b) = \sum_{k=1}^{P} (b - b_0)^k H_k(z) \tag{1}$$

where $H_k(z)$ is the fixed linear phase FIR subfilters of order $N_k$ each. The overall filter structure thus has linear phase [10].

## III. Proposed Design of Filters

According to the given audiogram, the required number of bands and bandwidths of each band are chosen for the initial design. This can be later altered according to matching error, if necessary. Matching error is the overall error between the filter output and the audiogram. The unique set of bandwidths, say $b_{ss}$, are then extracted. The VBW filter is designed for the set $b_{ss}$ using Farrow structure as explained in section II.

The given specifications of the frequency response characteristics decide the minimum order, $N_k$, of the subfilters. Also, the number of unique bandwidths decides the number of subfilters, $P$, in the proposed design. The filters $H_k(z)$ are obtained by means of linear programming, such that the overall transfer function H(z,b), achieves the specifications within tolerable limits.

The bands, thus obtained using VBW filter, are to be shifted appropriately using the spectrum shifting property. Proper magnitude gain is provided to each band by trial and error approach until it matches with the given audiogram.

The advantage of the proposed method is that, the hardware overhead in realizing the non-uniform frequency bands is minimal and depends on the number of unique bands required.

## IV. Design Example

An audiogram for mild to modearte hearing loss at low frequencies [7] as given in Figure 2, is used to evaluate the performance of the proposed method.

Two sets of 16-band non-uniform subbands, used in [7], named here as band-I and II, are used for audiogram matching, as given in Table I. Hardware complexity and matching Error obtained in [7] are given in Table II. The same subbands and their shifts are used in the proposed method using Farrow structure. Hence, the required unique subband bandwidths for the first set of subbands (Band I) are 250 Hz, 500 Hz and 1000 Hz.

A variable bandwidth filter is realized as mentioned for the desired bandwidths. The required characteristics of the filter

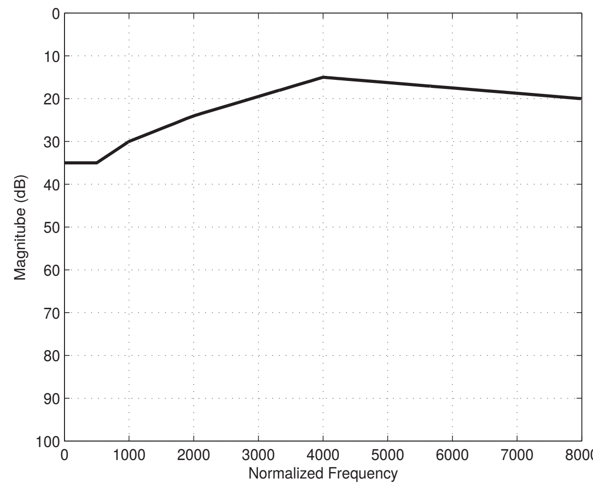


Fig. 2: Sample Audiogram- mild to moderate hearing loss at low frequencies

TABLE I: Subbands Considered in Literature [5]

| Band no. | Band-I | | Band-II | |
|---|---|---|---|---|
| | Bandwidth (Hz) | Shift (Normalised) | Bandwidth (Hz) | Shift (Normalised) |
| 1 | 250 | 0 | 400 | 0 |
| 2 | 250 | 0.0234 | 250 | 0.0328 |
| 3 | 250 | 0.0391 | 250 | 0.0484 |
| 4 | 250 | 0.0547 | 300 | 0.0656 |
| 5 | 500 | 0.0781 | 300 | 0.0844 |
| 6 | 500 | 0.1094 | 300 | 0.1031 |
| 7 | 1000 | 0.1563 | 300 | 0.1219 |
| 8 | 1000 | 0.2188 | 300 | 0.1406 |
| 9 | 1000 | 0.2813 | 300 | 0.1594 |
| 10 | 1000 | 0.3438 | 300 | 0.1781 |
| 11 | 500 | 0.3906 | 400 | 0.2 |
| 12 | 500 | 0.4219 | 500 | 0.2281 |
| 13 | 250 | 0.4453 | 1200 | 0.2813 |
| 14 | 250 | 0.4659 | 1200 | 0.3563 |
| 15 | 250 | 0.4766 | 1175 | 0.4305 |
| 16 | 250 | 0.5 | 525 | 0.5 |

TABLE II: Hardware complexity and Matching Error in [7]

| | Matching Error (dB) | No. of Multipliers |
|---|---|---|
| Band I | 2.1 | 395 |
| Band II | 1.08 | 395 |

response are given below:
Maximum Pass band ripple: 0.05 dB
Minimum stop band attenuation: 65 dB
Sampling frequency: 16 kHz
The variable bandwidth filter using Farrow structure is designed to meet the given filter specifications for passband and stopband errors. The band edges of the filters are decided based on the required discrete bandwidths and transition

TABLE III: New Subbands Proposed

| Band no. | Modified-Band-I | | Proposed-Band | |
|---|---|---|---|---|
| | Bandwidth (Hz) | Shift (Normalised) | Bandwidth (Hz) | Shift (Normalised) |
| 1 | 500 | 0 | 500 | 0 |
| 2 | 250 | 0.0234 | 300 | 0.029 |
| 3 | 250 | 0.0391 | 300 | 0.0484 |
| 4 | 500 | 0.0547 | 300 | 0.0656 |
| 5 | 500 | 0.0781 | 300 | 0.0844 |
| 6 | 500 | 0.1194 | 300 | 0.1031 |
| 7 | 500 | 0.166 | 300 | 0.1219 |
| 8 | 500 | 0.2088 | 300 | 0.1406 |
| 9 | 500 | 0.25 | 300 | 0.1594 |
| 10 | 1000 | 0.3138 | 300 | 0.1781 |
| 11 | 1000 | 0.3906 | 300 | 0.2 |
| 12 | 500 | 0.4219 | 300 | 0.222 |
| 13 | 500 | 0.4453 | 500 | 0.25 |
| 14 | 500 | 0.4659 | 1000 | 0.32 |
| 15 | 500 | 0.4766 | 1000 | 0.415 |
| 16 | 500 | 0.5 | 1000 | 0.5 |

widths. The subfilter order ($N_F$) and number of subfilters ($P$) for the Farrow structure are optimally selected considering all the above criteria.

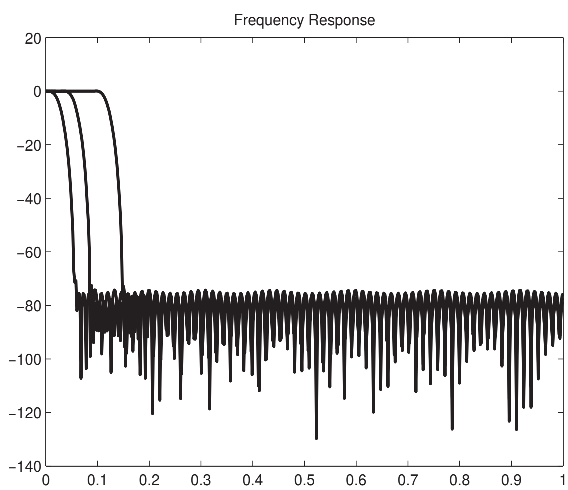


Fig. 3: VBW Filter obtained using Farrow structure designed for Band-I

The frequency response of the variable bandwidth filter is as shown in Figure 3. For band-I realization, the order of each subfilter in the Farrow structure is $N_F = 146$ and the number of subfilters is $P = 3$, for the given set of bandwidths and to obtain the desired characteristics. The total number of multipliers is given by,

$$P * (\frac{N_F}{2} + 1) + P - 1 \tag{2}$$

The performances of the bank of filters using the proposed method on bands I and II, are given in Table IV. Compared to the result in [7] (Table II), the matching error is higher for the proposed method, whereas the number of multipliers is lower. Hence, in this paper, we have optimized the band-I for the proposed design, by altering bandshifts alone, as modified-band-I in Table III. It gives a better fit of 1.8486 dB than band-I along with lower number of multipliers. But, still the matching error is higher when compared to band-II. In a similar manner band-II can be modified. But, 7 different bandwidths are selected to be realized, which is not an optimal design.

Hence, a new set of bands and bandshifts are proposed in this paper, and are given as proposed-band in Table III. The design method proposed in this paper gives a better matching and considerably lower number of multipliers when compared to the method in [7]. The 16-band filter response achieves the audiogram fitting as shown in Figure 4. Now, from Figure 4, it can be observed that there is further scope for minimization of the number of bands. Lesser the number of bands, lesser will be the implementation complexity.

TABLE IV: Hardware complexity and Matching Error using proposed method

| | Matching Error (dB) | No. of Multipliers |
|---|---|---|
| Band I | 3.4 | 224 |
| Band II | 3.58 | 300 |
| Modified Band I | 1.8486 | 224 |
| Proposed-Band | 1.0521 | 263 |

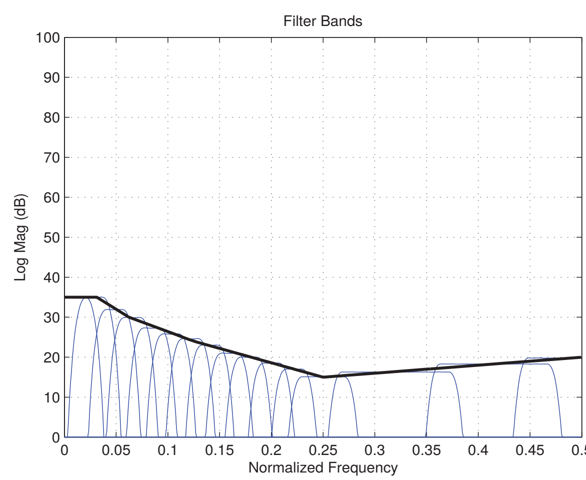


Fig. 4: Best Audigram Fitting with 16-Band using proposed method

Next, the number of bands is chosen as 8 and the new set of bands is selected as given in Table V.

The order of each subfilter for this case is, $N_F = 132$ and number of subfilters is P=3, that makes the passband ripple to be 0.0579 dB and stoband attenuation as -72.863 dB. The

TABLE V: Minimum number of subbands

| Band no. | Bandwidth | Shift |
|---|---|---|
| 1 | 750 | 0 |
| 2 | 500 | 0.0547 |
| 3 | 500 | 0.1 |
| 4 | 500 | 0.147 |
| 5 | 500 | 0.19 |
| 6 | 1000 | 0.2676 |
| 7 | 1000 | 0.375 |
| 8 | 1000 | 0.4759 |

total number of multipliers is 203 and the Matching error = 2.1327 dB. The audiogram fitting can be viewed in Figure 5.

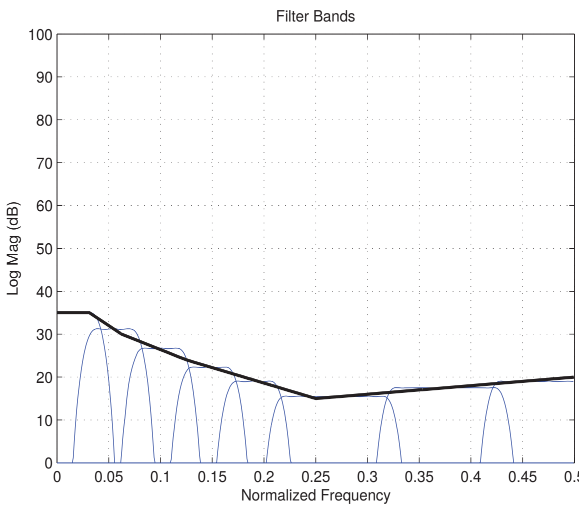


Fig. 5: Audigram Fitting with minimum number of bands

It can be noted that, as the number of bands increases, the matching error gets better, for the same amount of hardware complexity. The number of bands were varied from 8 to 16, in an experiment and the resulting matching error graph is as shown in Figure 6.

Since, a matching error below 3 dB is insignificant for human auditory senses [5], the matching result of a bank of filters of 8 bands, each realized using VBW based Farrow structure is satisfactory for practical purposes. Thus, an optimum minimum band design is possible for a digital hearing aid using the proposed approach, with each filter having minimum hardware complexity compared to previous approaches [7].

## V. Experiment on a real patient audiogram

The proposed method is also applied for real data of some patients, collected from National Institute of Speech and Hearing (NISH), Trivandrum, India. The right and left ear audiogram of a particular patient are provided in Figure 7 and

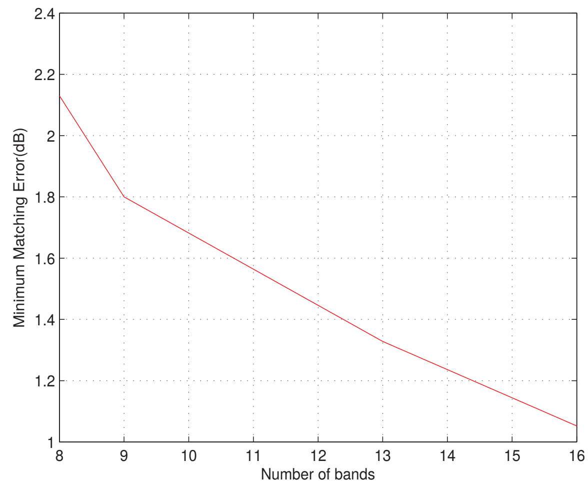


Fig. 6: Matching Error variation with number of bands

is classified as severe and profound sensorineural hearing loss (HL) respectively. The right ear audiogram has comparatively larger slope. The minimum number of bands and matching errors obtained are provided in Table VI along with the hardware complexity for single subband implementation. Due to the large transition width used for the filter design, the number of multipliers is found to be less. Thus, it can be seen to have large amount of saving in terms of hardware.

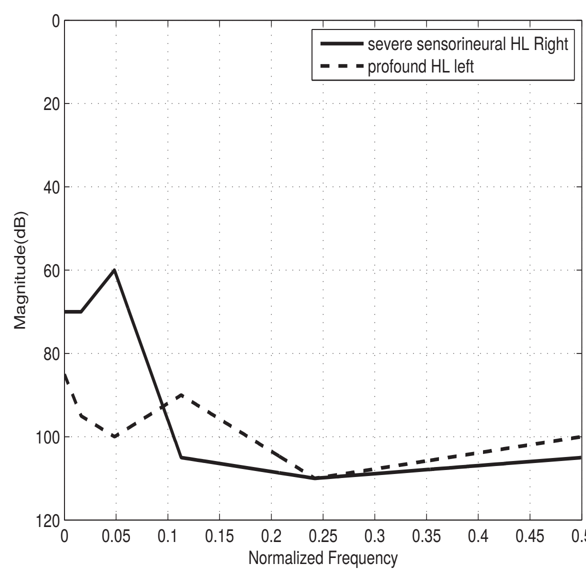


Fig. 7: Severe and Profound Loss - Audiogram collected from NISH, Trivandrum

TABLE VI: Selected Minimum Matching Errors for low hardware complexity for real patient data

| Sl. No. | Diagnosis | No.of Bands | Complexity for single filter | | Maximum Matching Error |
|---|---|---|---|---|---|
| | | | Multipliers | Adders | |
| 1 | Severe Hearing Loss (Right Ear) | 6 | 47 | 91 | 1.97 |
| 2 | Profound Hearing Loss (Left Ear) | 7 | 55 | 107 | 1.97 |

## VI. Conclusion

A bank of filters is designed based on variable bandwidth filters, that can be used as part of a digital hearing aid for the purpose of audiogram fitting. The design is based on an efficient Farrow structure design, which uses a weighted linear combination of linear phase subfilters, where the weights are directly related to the bandwidths. An audiogram matching for mild to moderate hearing loss at low frequencies, using the proposed method, reveals minimum hardware complexity in terms of multipliers for each filter. Also, the matching error that was previously achieved can be obtained using lower number of bands, using the method proposed in this paper.

## References


[1] A. Schaub, *Digital hearing aids*. Thieme, 2008.

[2] J. M. Kates, *Digital hearing aids*. Plural Pub., 2008.

[3] Y. Lian and Y. Wei, "A computationally efficient nonuniform FIR digital filter bank for hearing aids," *IEEE Trans. Circuits Syst. II*, vol. 52, no. 12, pp. 2754–2762, Dec 2005.

[4] Y. Wei and Y. Lian, "A 16-band nonuniform FIR digital filterbank for hearing aid," in *Biomedical Circuits and Systems Conference, 2006. BioCAS 2006. IEEE*. IEEE, Nov 2006, pp. 186–189.

[5] T.-B. Deng, "Three-channel variable filter-bank for digital hearing aids," *Signal Processing, IET*, vol. 4, no. 2, pp. 181–196, Apr 2010.

[6] Y. Wei and D. Liu, "A design of digital FIR filter banks with adjustable subband distribution for hearing aids," in *Information, Communications and Signal Processing (ICICS) 2011 8th International Conference on*. IEEE, Dec 2011, pp. 1–5.

[7] J. T. George, , and E. Elias, "A 16-band reconfigurable hearing aid using variable bandwidth filters," *Global Journal of Researches In Engineering*, vol. 14, no. 1, 2014.

[8] J. Vesma and T. Saramaki, "Optimization and efficient implementation of FIR filters with adjustable fractional delay," in *Circuits and Systems, 1997. ISCAS'97., Proc. IEEE International Symposium on*, vol. 4. IEEE, Jun 1997, pp. 2256–2259.

[9] H. Johansson and P. Lowenborg, "On linear-phase FIR filters with variable bandwidth," *IEEE Trans. Circuits Syst. II*, vol. 51, no. 4, pp. 181–184, Apr 2004.

[10] P. Lowenborg and H. Johansson, "Minimax design of adjustable-bandwidth linear-phase FIR filters," *IEEE Trans. Circuits Syst. I*, vol. 53, no. 2, pp. 431–439, Feb 2006.