A Design of IIR Based Digital Hearing Aids Using Genetic Algorithm

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Abstract— **This paper presents a design of digital filter for digital hearing aids application. The structure of filter is consist a combination in parallel form of IIR (Infinite Impulse Response) a low-pass, a band-pass and a high-pass filter. This study shows an advantage of IIR filter can gives a good result in the low complexity digital hearing aids which leads to low hardware resources requirement and low power consumption for VLSI design. The filter coefficients of there IIR filter will obtained from the optimization procedure by genetic algorithm (GA.) The error between desired magnitude response and actual magnitude response will be minimized by GA. In order to achieve a capable of the best compensation for each hearing loss pattern. Finally, the design example and simulation results will show the accuracy of hearing loss compensation and optimal coefficients.**

Keywords IIR Filter ,Digital Hearing Aids, Genetic Algorithm

I. INTRODUCTION

Recently, digital hearing aids designs and developments usually have an important role more than analog hearing aids designs and development because it has low power consumption, small size and low noise. Especially, it has high flexibility accounts for a more accurate fitting that results in increased hearing clarity for the wearer. [1-8].

Generally, Digital hearing aids consists of FIR (Finite Impulse Response) digital filter bank. [1] proposed an eight band-pass FIR filters are pre-designed and stored and three out of the eight FIR filters are selected and used to match a given audiogram. This method has only specific frequency band in each sub-circuit and each sub-circuit can adjust amplify rate for compensate hearing loss in each sub-band. So, this method must be designed high order and many sub-band for high accuracy compensate hearing loss. Also, it has large and complex circuits and high energy consumption. Consequently,[8] proposed a new approach using variable filter-bank (VFB) that consists of three channels. They are variable low-pass filter, variable band-pass filter and variable high-pass filter, respectively. Their results are very high accuracy. However, [8] use only Nelder-Mead algorithm for minimize coefficient of VFB.The Nelder-Mead algorithm is applicable to the minimization of a multivariable objective function, but other methods may be more efficient than the Nelder-Mead algorithm in achieving a global optimization. So, this paper will apply genetic algorithm to optimize the coefficients of VFB.

II. IIR DIGITAL FILTER DESIGNS FOR DIGITAL HEARING AIDS.

Digital hearing aids that use the IIR filters for realization have been proposed in [8] as shown in Fig.1.The magnitude response only three IIR filters where both of the band edge frequencies and magnitudes can be freely adjusted as shown in Fig.2. So, it was called 3-channel variable filter-bank (VFB).

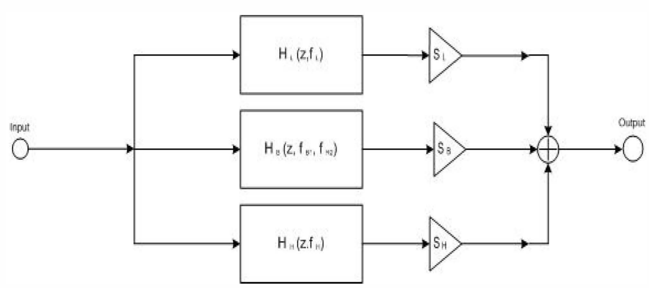

Fig. 1 Three-channel variable filter-bank (VFB).

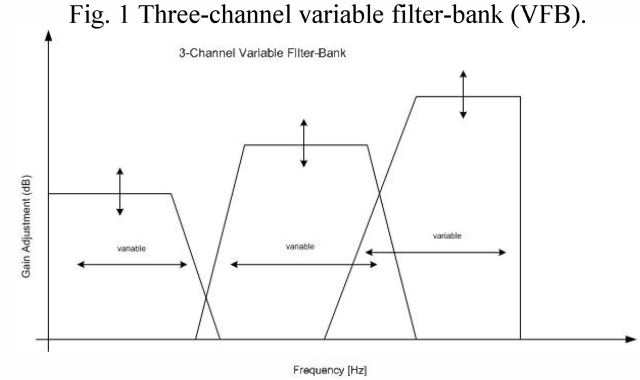

Fig 2. Tunability of 3-channel VFB

A 3-channel variable filter-bank has been constructed by parallelizing with variable low-pass, variable band-pass, and variable high-pass digital filters whose both magnitudes and bandwidths can be independently tuned for matching various hearing loss patterns. (audiograms). Consequently, it can freely adjust to the band edge frequencies f_L, f_{B1}, f_{B2} and f_H as well as the scaling factors S_L, S_B and S_H to fit a given audiogram. The transfer function of 3-channel VFB is shown [8].

$$H(z,x) = S_L \cdot H_L(z,f_L) + S_B \cdot H_B(z,f_{B1},f_{B2}) + S_H \cdot H_H(z,f_H) \quad (1)$$

Where the parameter vector

$$x = [f_L \; f_{B1} \; f_{B2} \; f_H \; S_L \; S_B \; S_H] \quad (2)$$

The x vector consists of seven adjustable parameters. [8] is showed many matching examples that use only second-order Chebyshev type-I prototype analog low-pass filter could be successfully applied to obtain a 3-channel VFB. It could match various audiograms with very high-accuracy fitting.

The procedures of the VFB design were as follows [8] and assume that an audiogram was given, which expresses the threshold hearing levels $M_d(\hat{f}_i)$ in decibels (dB) measured at a set of octave frequencies $\hat{f}_i$, when $\hat{f}_i$=250/500/1k/2k/4k /8k Hz. Then, the points $M_d(\hat{f}_i)$ were uniformly (linearly) interpolated at a set of denser uniform frequency points in the logarithmic scale (base 10) points, which led to a set of denser grids $M_d(f_i)$, where

$$M_d(f_i) = M_d(\hat{f}_i), \text{ if } f_i = \hat{f}_i \quad (3)$$

The maximum absolute error

$$E_{\max}(x) = \max\{|M(f_i, x) - M_d(f_i)|\} \quad (4)$$

where $M(f_i, x)$ denotes the actual magnitude response of the 3-chanel VFB at frequency f_i. In this paper, the error function (4) is minimized using the Genetic Algorithm (GA) optimization method. On the other hand, the three pass-bands shows in Fig. 2 may overlap, it depends upon the actual audiograms to be matched. So, we should be scaling the parameter vector as shown [8].

$$x = [x_1 \; x_2 \; x_3 \; x_4 \; S_L \; S_B \; S_H] \quad (5)$$

Equation (5) is optimized for the maximum absolute error (4) that is minimized

III. GA FOR SEARCHING THE FILTER COEFFICIENTS

Genetic algorithm (GA) is a directed random search technique that is modeled on the natural evolution/selection process toward the survival of the fittest. Generally, GA consists of initialization, evaluation, reproduction (selection) crossover and mutation as depicted in Fig. 3 [9]

In this paper, we use GA for searching the filter coefficient in (5). It is optimized such that the maximum absolute error (4) is minimized. Follow this procedure. [9]

Step 1. Assigns the initial value

$$x_0 = [x_{01} \; x_{02} \; x_{03} \; x_{04} \; S_{0L} \; S_{0B} \; S_{0H}] \quad (6)$$

Assigned lower and upper bound, population size N_P, the vector N_b consisting of the numbers of the bits assigned for the representation of each variables x_i, the probability of crossover P_c, the probability of mutation P_m, the learning rate η and the maximum number of iterations $k_{\max}$

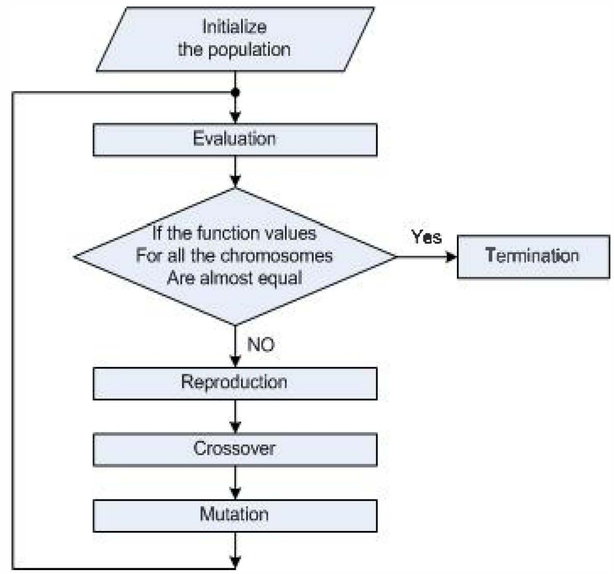

Fig. 3 Flowchart for a genetic algorithm

Step 2. Random Generation of Initial Population

Set $x^0 = x_0$, $f^0 = f(x^0)$ and construct in a random way the initial population array X_1 that consists of N_P states including the initial state x_0 , by (20) and (21)

$$X_1(1) = x_0 \quad (7)$$

$$X_1(k) = l + rand.*(u - l) \quad (8)$$

Encode each number of population arrays into a binary string by (9)

$$P_1(n, 1 + \sum_{i=1}^{m-1} N_{bi} : \sum_{i=1}^{m} N_{bi}) = (2^{N_{bm}} - 1)\frac{X_1(n,m) - l(m)}{u(m) - l(m)} \quad (9)$$

For n = 1:Np and m = 1:N where N

Step 3. For k = 1 to $k_{\max}$, do the following

1. Decode each number in the pool into a decimal number by (10)

$$X_k(n,m) = P_k(n,1)\frac{u(m) - l(m)}{2^{N_{bm}} - 1} + l(m) \quad (10)$$

For n = 1:Np and m = 1:N and evaluate the value $f(n)$ of function for every row $X_k(n,:) = x(n)$ corresponding to each chromosome and find the minimum $f_{\min} = f(n_b)$ for $X_k(n_b,:) = x(n_b)$

2. If $f_{\min} = f(n_b) < f^0$, set $f^0 = f(n_b)$ and $x^0 = x(n_b)$

3. Convert the function values into the values of fitness by (11)

$$f_1(n) = Max_{n=1}^{N_p}\{f(n)\} - f(n) \quad (11)$$

4. If $f_1(n) = Max_{n=1}^{N_p}\{f(n)\} - f(n) \approx 0$, then terminate this procedure, declaring x^0 as the best. Otherwise, in order to make more chromosomes around the best point $x(n_b)$ in the next generation, use the reproduction rule (12) to get a new population X_{k+1} with $X_{k+1}(n,:) = x(n)$ and encode it to reconstruct a new pool array P_{k+1} by (9)

$$x(n) \leftarrow x(n) + \eta \frac{f_1(n_b) - f_1(n)}{f_1(n_b)}(x(n_b) - x(n)) \quad (12)$$

5. Shuffle the row indices of the pool array for random mating of the chromosomes.

6. With the crossover probability P_c, exchange the tail part starting from some random bit of the numbers in two randomly paired chromosomes (rows of P_{k+1}) with each other's to get a new pool array P'_{k+1}

7. With the mutation probability P_m, reverse a random bit of each number represented by chromosomes (row of P'_{k+1}) to make a new pool array P_{k+1}

IV. DESIGNS EXAMPLES AND SIMULATIONS RESULTS

In this section, example designs and simulations of match various audiograms present. The 3-channel VFB with second order or fourth-order variable filters can obtain high-accuracy audiogram match. The audiograms were down-loaded from the Independent Hearing Aid Information, a public service by Hearing Alliance of America. The prototype filter is the second-order Chebyshev type-I analog low-pass filter as shown in (13)

$$H_a(s) = \frac{1}{1.6013 + 1.4841s + s^2} \quad (13)$$

An audiogram illustrates shown in Fig. 4 which mild hearing loss (20-39 dB) in the whole frequency band. People with kind of hearing loss have difficulties hearing most vowels and consonants that will have more trouble in noisy conditions. The results show in Fig. 5 and Fig. 6 plot the errors. The maximum error is 1.26 dB less than result of [8] 0.42 dB.

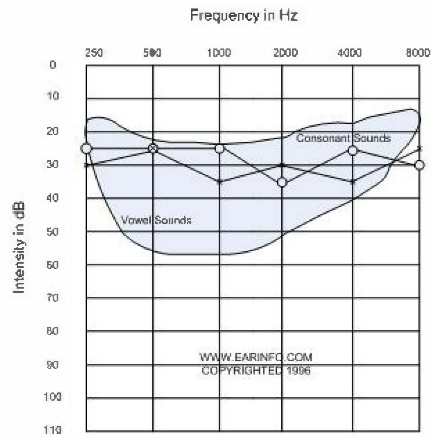

Fig. 4 An audiogram of mild hearing loss in the whole frequency band.

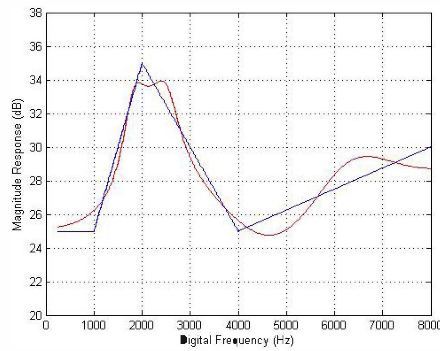

Fig. 5 Matching result.

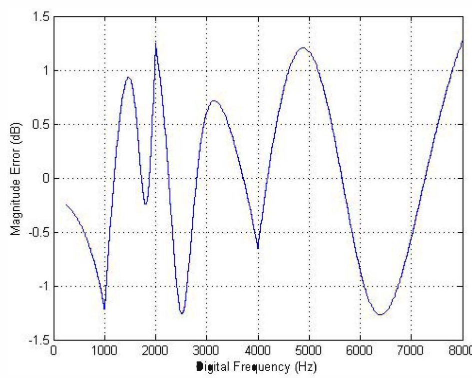

Fig. 6 Matching error.

Another, audiogram show in Fig. 7 is a mild hearing loss (20-39 dB) in high frequencies. The results are shown in Fig. 8 and Fig. 9 plot the errors. The maximum error is 1.18 dB less than result of [8] 0.14 dB.

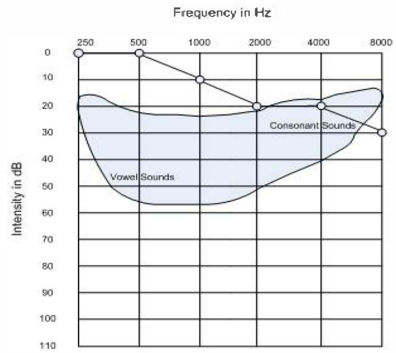

Fig. 7 An audiogram of mild hearing loss in high frequencies.

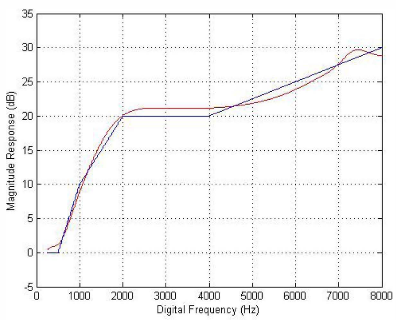

Fig. 8 Matching result.

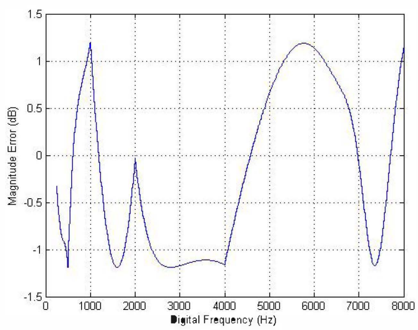

Fig. 9 Matching error.

The results show that an IIR based digital hearing aid design using GA. it less than (smaller) matching error between output value of VFB and audiogram than the method proposed by [8] that used Nelder-Mead algorithm. Although, these different values are non-significant because of value below 3 dB. It is not sensitive by people. However, GA. algorithm may be better when sampling frequency is higher. That has effect to implementation.

V. Conclusion

An IIR based digital hearing aid design using genetic algorithm is presented in this paper. The genetic algorithm is used to optimize the coefficient values of variable filter bank. Results have shown that the matching error between output value of VFB and audiogram is smaller than the method proposed by [8] that used Nelder-Mead algorithm for optimizing the coefficient values of VFB.

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Table 1. Scaling factors and Denominator coefficients

Audiogram	Scaling Factors		Denominator Coefficients				
			b_0	b_1	b_2	b_3	b_4
Figure 5	S_L	4.6866	1	-0.2391	0.2691		
	S_B	0.6546	1	-2.3760	3.0150	-1.9455	0.6755
	S_H	2.9186	1	0.9805	0.4077		
Figure 8	S_L	0.0048	1	-1.8075	0.8305		
	S_B	4.5284	1	0.2964	-0.3868	0.0291	0.2864
	S_H	0.6003	1	1.6006	0.6871		