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Design FIR Filter with Signed Power of Two Terms Using MATLAB

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Abstract: The design of digital filter, using signed power of two terms are implemented. The filter are designed without multipliers. The design is based on the remez exchange algorithm for the design of low pass and filters. Linear phase FIR filter with coefficients constisting of minimum number of signed power of two terms is formulated. The aim of this thesis is first to reduce the region that contains the optimal solution in order to decrease the computation time and ,then FIR filter are designed using mixed integer linear programming. The response obtained by this technique is compared with the simply rounding technique.

Keyword: remez exchange algorithm, Linear phase FIR filter , Mixed integer linear programming, Signed power of two terms, Rounding technique.

1 INTRODUCTION

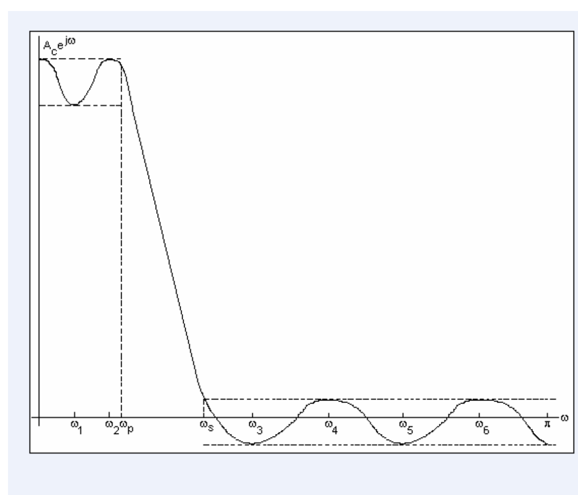
Recently numberous algorithm have been proposed for designing multiplierless finite impulse response (FIR) filters. In multiplierless digital filter multiplication are replaced with a sequence of shift and adds. Therefore only adders are required for the coefficient implementation. This leads to significant reduction in the computational complexity and power consumption.filters with such specifications can be designed by appropriate modification of the McClellan-Park algorithm. By suitable choice of the weighting function of the equirriple error.

The purpose of this paper is to advance a new technique for the design of linear phase FIR filters with equirriple stop band and with a prescribed degree of flatness in the pass band.the proposed technique is based on McClellan-Park algorithm for FIR filter design and optimization is involved here. In Section II the method is introduced along with numerical examples. For the design of narrow passband filters, a number of improved method are described in Section III, and IV based on low pass and high pass filters. In Section V we discuss certain implementation considerations.

II SATEMENT OF THE PROBLEM

The Parks-McClellan algorithm, published by James mcClellan and Thomas Parks in 1972, is an iterative algorithm for finding the optimal Chebyshev finite impulse responses(FIR) filter.. The Parks-McClellan algorithm is utilized to design and implement efficient and optimal FIR filters. It uses an indirect method for finding the optimal filter coefficients.

The goal of the algorithm is to minimize the error in the pass and stop bands by utilizing the Chebyshev approximation. The Parks-McClellan algorithm is a variation of the Remez algorithm or Remez exchange algorithm, with the change that it is specifically designed for FIR filters and has become a standard method for FIR filter design.



Pass and Stop Bands of Parks-McClellan Algorithm

The y-axis is the frequency response $H(\omega)$ and the x-axis are the various radian frequencies, ω_i. It can be noted that the two frequences marked on the x-axis, ω_p and ω_s. ω_p indicates the pass band cutoff frequency and ω_s indicates the stop band cutoff frequency. The ripple like plot on the upper left is the pass band ripple and the ripple on the bottom right is the stop band ripple. The two dashed lines on the top left of the graph indicate the δ_p and the two dashed lines on the bottom right indicate the δ_s. All other frequencies listed indicate the extremal frequencies of the frequency response plot. As a result there are six extremal frequencies, and then we add the pass band and stop band frequencies to give a total of eight extremal frequencies on the plot.

According to the IEEE Signal Processing Magazine, [1] the Parks-McClellan Algorithm is implemented using the following steps:

1. Initialization: Choose an extremal set of frequences $\{\omega_i^{(0)}\}$.
2. Finite Set Approximation: Calculate the best Chebyshev approximation on the present extremal set, giving a value $\delta^{(m)}$ for the min-max error on the present extremal set.
3. Interpolation: Calculate the error function $E(\omega)$ over the entire set of frequencies Ω using (2).
4. Look for local maxima of $|E^{(m)}(\omega)|$ on the set Ω.
5. If $\max_{(\omega\varepsilon\Omega)}|E^{(m)}(\omega)| > \delta^{(m)}$, then update the extremal set to $\{\omega_i^{(m+1)}\}$ by picking new frequencies where $|E^{(m)}(\omega)|$ has its local maxima. Make sure that the error alternates on the ordered set of frequencies as described in (4) and (5). Return to Step 2 and iterate.
6. If $\max_{(\omega\varepsilon\Omega)}|E^{(m)}(\omega)| \le \delta^{(m)}$, then the algorithm is complete. Use the set $\{\omega_i^{(0)}\}$ and the interpolation formula to compute an inverse discrete Fourier transform to obtain the filter coefficients.

According the Professor Douglas Jones of the University of Illinois, [4]the Parks-McClellan Algorithm may be implemented as the following:

1. Make an initial guess of the L+2 extremal frequencies.
2. Compute δ using the equation given.
1. Using Lagrange Interpolation, we compute the dense set of samples of $A(\omega)$ over the passband and stopband.
2. We Determine the new L+2 largest extrema.
3. If the Alternation Theorem is not satisfied, then we go back to (2) and iterate until the Alternation Theorem is satisfied.

4 If the Alternation Theorem is satisfied, then we Compute h(n) and we are done.

To gain a basic understanding of the Parks-McClellan Algorithm mentioned above, we can rewrite the algorithm above in a simpler form as:

1. Guess the positions of the extrema are evenly spaced in the pass and stop band.
2. Perform polynomial interpolation and re-estimate positions of the local extrema.
3. Move extrema to new positions and iterate until the extrema stop shifting.

III PROBLEM ANAYSIS

Example 1: Design and plot the equirriple linear phase FIR low pass filter with order 36.using signed power of two terms. Normalized frequency of pass band and stop band are 0.15, 0.25. also design this filter using rounding method.

TABLE 1

Cofficients of H(z) in Example 1

Filter length =36

----

h(0) = -0.0088 = h(36)

h(1) = -0.0069 = h(35)

h(2) = -0.0044 = h(34)

h(3) = -0.0004 = h(33)

h(4) = 0.0074 = h(32)

h(5) = 0.0131 = h(31)

h(6) = 0.0162 = h(30)

h(7) = 0.0111 = h(29)

h(8) = 0.0003 = h(28)

h(9) = -0.0166 = h(27)

h(10) = -0.0307 = h(26)

h(11) = -0.0375 = h(25)

h(12) = -0.0275 = h(24)

h(13) = -0.0002 = h(23)

h(14) = 0.0449 = h(22)

h(15) = 0.0979 = h(21)

h(16) = 0.1498 = h(20)

h(17) = 0.1861 = h(19)

h(18) = 0.2002

IV IMPLEMENTATION CONSIDERATION

FIR digital filter designed over the signed power of two (SPT) discrete spaces were first proposed by Lim and Constantinides. This section briefly describes the SPT number characteristics and exiting optimization techniques for the design of digital filter subject to SPT coefficient.

(a) Signed Power of Two

In mathematics, a power of two is any of the integer powers of the number two.because two is the base of the binary system, power of two are important to computer science.

(b) Signed Digit Representation

The radix-2 signed digit format is a 3-valued representation of a radix-2 number and employs three digit values 0,1 and -1.A simple algorithm representation which convert radix-2 binary number to equivalent SD representation is as follow.

$C_{-i} = a_{-i-1} - a_{-i}$, i = b , b-1, …………………..1,

After that decimal number is represent the binary number using this formula

$$h(n) = \sum_{i=1}^{b} a_{-i}\, 2^{-1}$$

TABLE II

Implemented Result

Cofficients of H(z) in Example 1

length =36

----

h(0) = 0 = h(36)

h(1) = 0 = h(35)

h(2) = 0 = h(34)

h(3) = 0 = h(33)

h(4) = 0 = h(32)

h(5) = 0 = h(31)

h(6) = 0 = h(30)

h(7) = 0 = h(29)

h(8) = 0 = h(28)

h(9) = 0 = h(27)

h(10) = 0 = h(26)

h(11) = -2^{-5} = h(25)

h(12) = 0 = h(24)

h(13) = 0 = h(23)

h(14) = -2^{-5} = h(22)

h(15) = $2^{-4}+2^{-5}$ = h(21)

h(16) = 2^{-3} = h(20)

h(17) = $2^{-3}+2^{-5}$ = h(19)

h(18) = $2^{-3}+2^{-5}$

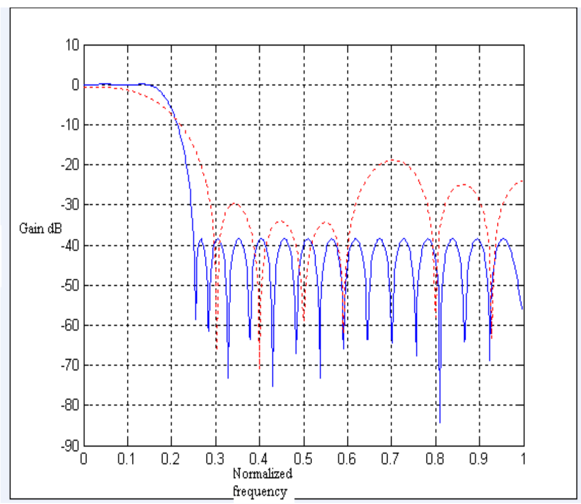


Fig. 1 Frequency response optimized with length N= 36 obtained , rounding the coefficient value

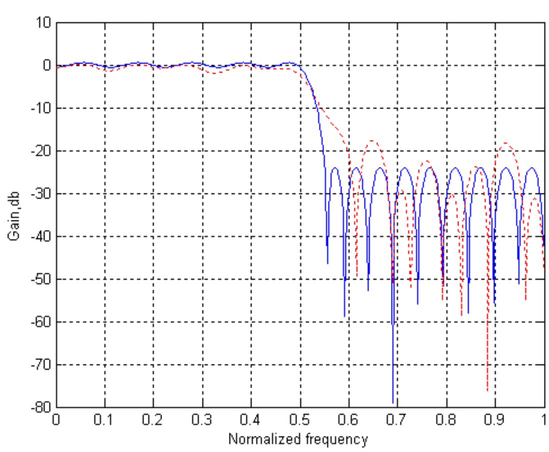


Fig. 2 Frequency response optimized with length N= 36 obtained , rounding the coefficient value

V CONCLUSION

In this paper the methods of integer linear programming are particularly useful for designing FIR filters with the power of two coefficients. The result obtained is significant when compared to simple rounding of coefficient value.the aim of optimization is only the minimization of the number of SPT terms.Extensive research has shown that the complexity of an FIR filter can be reduced by implementation its coefficient as sum of SPT terms and faster hardware implementation of the multiplication operation.

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