ISSN:0975-9646

Sangita Solanki et al, / (IJCSIT) International Journal of Computer Science and Information Technologies, Vol. 3 (5) , 2012,4968 - 4970

FIR Filter Design Using Rounding Technique

Sangita Solanki , Devendra Singh Mandloi

EC-Department

Indore Institute science of Technology, Indore

Abstract: **In this paper, design low pass FIR filter with rounding technique. The design is based on the remez exchange algorithm (RE) for the design of high pass filters. Linear phase FIR filter with coefficients consisting of nearest quantized value is provided. Results are obtained as a sum or difference of two power of two.**

Keywords: **remez exchange algorithm, optimal filter, power of two, rounding technique**.

1 INTRODUCTION

Last decade so many papers are published in this technique. First we design low pass filter using remez exchange (RE) algorithm. Because this algorithm is provided to minimize the error in pass and stop band s by utilizing the chebyshev approximation. The parks-McClellan algorithm is a variation of the remez exchange, with the change that it is specifically designed for FIR filters and has become a standard method for FIR filter design.[4]. The discrete coefficient space discussed in [2] is the “power-of-two” space where each coefficient value is represented as a sum or difference of several power-of-two number. Introduction of rounding method discussed in [ ] is provide nearest integer value. In Section II the optimal filter design using remez exchange algorithm. In Section III discussed the discrete programming using rounding method.

II OPTIMAL FILTER DESIGN USING REMEZ EXCHANGE

The **Remez algorithm** (sometimes also called **Remes algorithm**, **Remez/Remes exchange algorithm**, or simply **Exchange algorithm**), published by Evgeny Yakovlevich Remez in 1934 is an iterative algorithm used to find simple approximations to functions, specifically, approximations by functions in a Chebyshev space that are the best in the uniform norm L_∞ sense.

is an application of the Chebyshev alternation theorem that constructs the polynomial of best approximation to certain functions under a number of conditions. The Remez algorithm in effect goes a step beyond the minimax approximation algorithm to give a slightly finer solution to an approximation problem.

Parks and McClellan (1972) observed that a filter of a given length with minimal ripple would have a response with the same relationship to the ideal filter that a polynomial of degree $\leq n$ of best approximation has to a certain function, and so the Remez algorithm could be used to generate the coefficients.

In this application, the algorithm is an iterative procedure consisting of two steps. One step is the determination of candidate filter coefficients $h(n)$ from candidate "alternation frequencies," which involves solving a set of linear equations. The other step is the determination of candidate alternation frequencies from the candidate filter coefficients (Lim and Oppenheim 1988). Experience has shown that the algorithm converges quickly, and is widely used in practice to design filters with optimal response for a given number of taps. However, care should be used in saying "optimal" coefficients, as this is implementation dependent and also depends on fixed or floating-point implementation as well as numerical accuracy.

A typical example of a Chebyshev space is the subspace of Chebyshev polynomials of order n in the space of real continuous functions on an interval, $C[a, b]$. The polynomial of best approximation within a given subspace is defined to be the one that minimizes the maximum absolute difference between the polynomial and the function

PREVIEW

A typical FIR digital filter can be characterized by the transfer function

$$H(z) = \sum_{n=0}^{N-1} h(n) z^{-n}$$

where N is the filter length of the impulse response, $h(n)$, and its frequency response is represented by

$$H(e^{j\omega}) = \hat{H}(\omega) e^{j[L\omega/2-(N-1)\omega/2]}$$

where $H(\omega)$ is the magnitude response which is a real-valued function and L is equal to 0 or 1. According to the symmetric properties of the impulse response and the filter length, there are four cases to be taken into account [5].

Case 1: Symmetric impulse response and odd length: $L = 0$ and

$$\hat{H}(\omega) = \sum_{k=0}^{(N-1)/2} b(k) \cos(k\omega),$$

Where a(o) -- h((N-1)/2) and a(k) = 2h((N-1)2-k)
For k = 1, 2, ………………. (N-1)/2.

Case 2: Symmetric impulse response and even length: $L = 0$ and

$$\hat{H}(\omega) = \sum_{k=1}^{N/2} b(k)\cos((k-1/2)\omega),$$

where $a(k) = 2h(N\,2 - k)$ for $k = 1,2,L,\ (N-1)\ \ 2$.

Case 3: Anti-symmetric impulse response and odd length: $L = 1$ and

$$\hat{H}(\omega) = \sum_{k=1}^{(N-1)/2} b(k)\sin(k\omega),$$

where $a(k) = 2h((N-1)\ \ 2 - k)$ for $k = 1,2,L,\ (N-1)\ \ 2$.

Case 4: Anti-symmetric impulse response and even length: $L = 1$ and

$$\hat{H}(\omega) = \sum_{k=1}^{N/2} b(k)\sin((k-1/2)\omega),$$

where $a(k) = 2h(N\,2 - k)$ for $k = 1,2,L,\ (N-1)\ \ 2$.

The least-squares approach to these filters design is to formulate an objective error function, fitness, as below

$$E = \int_R \left| D(\omega) - \hat{H}(\omega) \right|^2 d\omega$$

where $D(\omega)$ is the desired magnitude response and R represents the region of design bands.

III ROUNDING

We use the result proposed in (Bartolo at al.1988) for the impulse response rounding given as

$$g(n) = \alpha \cdot g_I(n) = \alpha \cdot round(h(n)/\alpha)$$

Where h(n) is an equiripple type FIR filter which satisfies given specification, $g_I(n)$ is the new impulse response derived by rounding all coefficients of h(n) to the nearest integer, and round(.) means the round operation. The rounded impulse response $g_I(n)$ is scaled by α in order that gain in dB of the rounded filter has the value($0 \pm R_P$)dB.

in the passband , where R_P is the passband ripple. The rounding constant α determines the precision of the approximation of g(n) to h(n). Considering that the integer coefficient multiplication can be accomplished with only shift-and-add operation, the rounded impulse response filter is multiplier-free. Besides the rounding constant is chosen to be in the form $\alpha = 2^{-N}$. where N is an integer.

IV PROBLEM

Design and plot the equirriple linear phase FIR low pass filter with order 36 and Normalized frequency of pass band and stop band are 0.15, 0.25. Using rounding technique.

TABLE 1

Cofficients of H(z) in Example
Filter length =36

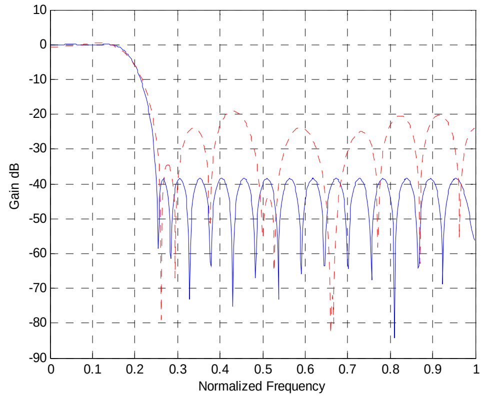
h(0) =	-0.0088 =	h(36)
h(1) =	-0.0069 =	h(35)
h(2) =	-0.0044 =	h(34)
h(3) =	0.0004 =	h(33)
h(4) =	0.0074 =	h(32)
h(5) =	0.0131 =	h(31)
h(6) =	0.0162 =	h(30)
h(7) =	0.0111 =	h(29)
h(8) =	0.0003 =	h(28)
h(9) =	-0.0166 =	h(27)
h(10) =	-0.0307 =	h(26)
h(11) =	-0.0375 =	h(25)
h(12) =	-0.0275 =	h(24)
h(13) =	-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	

TABLE II

Implemented Result
Cofficients of H(z) in Example
Filter 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) =	2^{-5} =	h(29)
h(8) =	0 =	h(28)
h(9) =	0 =	h(27)
h(10) =	-2^{-5} =	h(26)
h(11) =	-2^{-5} =	h(25)
h(12) =	-2^{-5} =	h(24)
h(13) =	-2^{-5} =	h(23)
h(14) =	0 =	h(22)
h(15) =	-2^{-5} =	h(21)
h(16) =	$2^{-3}+2^{-5}$ =	h(20)
h(17) =	$2^{-3}+2^{-5}$ =	h(19)
h(18) =	$2^{-3}+2^{-4}$	

Solid line are original filter and dashed line are rounding



Magnitude response and N = 36 & rounding

V CONCLUSION

Design FIR filters with rounding technique. 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.

REFRRENCES

[1] J.H.McClellan, T.W.Park and L.R.Rabiner, “FIR linear phase filter design program,” in program for digital signal processing, New York:IEEEPress,1979, pp.5.1.1-5.1.13.
[2] Y.C.Lim, and S.R.Parker, “FIR filter design over a discrete power two coefficient space,”IEEE Trans. Acoust. Speech, signal processing ling, Vol. ASSP-31, June 1983.
[3] Y.C.Lim, S.R.Parker, and A.G.Constanntinides, “Finite word length FIR filter design using integer programming over a discrete coefficient space,”IEEE Trans. Acoust. Speech, signal processing ling, Vol. ASSP-30,pp. 661-664, Agu. 1982.
[4] Mitra, Kaiser, Handbook for digital processing, john wiley &sons, 1993.
[5] http://www.ie.ncsu.edu/kay/matlab