

# Part 2: FIR Filters – Weighted-Chebyshev Method

Tutorial ISCAS 2007

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# Introduction

The weighted-Chebyshev method for the design of FIR filters is an iterative multi-variable optimization method based on the *Remez Exchange Algorithm*.

It can be used to design optimal FIR (nonrecursive) filters with arbitrary amplitude responses.

**Note:** The material for this module is taken from Antoniou, *Digital Signal Processing: Signals, Systems, and Filters*, Chap. 15.

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- These developments led in 1975 to the well-known McClellan-Parks-Rabiner computer program for the design of FIR filters, which has found widespread applications.
- Enhancements to the weighted-Chebyshev method were proposed by Antoniou during the early eighties.

# Problem Formulation

Consider an FIR filter characterized by the transfer function

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

and assume that

- $N$  is odd,
- the impulse response is symmetrical, and
- the sampling frequency is  $\omega_s = 2\pi$ .

# Problem Formulation

The frequency response of the filter can be expressed as

$$H(e^{j\omega}) = e^{-jc\omega} P_c(\omega)$$

where

$$P_c(\omega) = \sum_{k=0}^c a_k \cos k\omega \quad (\text{A})$$

is the *gain function* and

$$a_0 = h(c)$$

$$a_k = 2h(c - k) \quad \text{for } k = 1, 2, \dots, c$$

$$c = (N - 1)/2$$

**Note** that  $P_c(\omega)$  is the frequency response of a noncausal version of the required filter.



# Error Function

- An error function  $E(\omega)$  can be constructed as

$$E(\omega) = W(\omega)[D(\omega) - P_c(\omega)]$$

where  $e^{-jc\omega}D(\omega)$  is the idealized frequency response of the desired filter,  $W(\omega)$  is a weighting function, and

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- If  $|E(\omega)|$  is minimized such that

$$|E(\omega)| = |W(\omega)[D(\omega) - P_c(\omega)]| \leq \delta_p \quad \text{for } \omega \in \Omega \quad (\text{B})$$

with respect a set of frequencies in the interval  $[0, \pi]$ , say  $\Omega$ , a filter can be obtained in which

$$|E_0(\omega)| = |D(\omega) - P_c(\omega)| \leq \frac{\delta_p}{|W(\omega)|} \quad \text{for } \omega \in \Omega \quad (\text{C})$$

# Lowpass Filters

- In the case of a lowpass filter, the minimization of  $|E(\omega)|$  will force the inequality

$$|E_0(\omega)| = |D(\omega) - P_c(\omega)| \leq \frac{\delta_p}{|W(\omega)|} \quad \text{for } \omega \in \Omega \quad (\text{C})$$

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$$D(\omega) = \begin{cases} 1 & \text{for } 0 \leq \omega \leq \omega_p \\ 0 & \text{for } \omega_a \leq \omega \leq \pi \end{cases}$$

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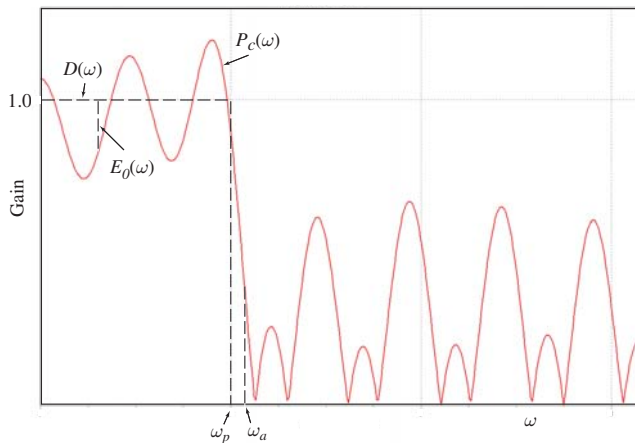
$$|E_0(\omega)| = |D(\omega) - P_c(\omega)| \leq \frac{\delta_p}{|W(\omega)|} \quad \text{for } \omega \in \Omega \quad (\text{C})$$

where

$$D(\omega) = \begin{cases} 1 & \text{for } 0 \leq \omega \leq \omega_p \\ 0 & \text{for } \omega_a \leq \omega \leq \pi \end{cases}$$

- In effect, a minimization algorithm will force the actual gain function  $P_c(\omega)$  to approach the ideal gain function  $D(\omega)$ .

# Lowpass Filters *Cont'd*



## Lowpass Filters *Cont'd*

- If we choose the weighting function

$$W(\omega) = \begin{cases} 1 & \text{for } 0 \leq \omega \leq \omega_p \\ \frac{\delta_p}{\delta_a} & \text{for } \omega_a \leq \omega \leq \pi \end{cases}$$

then from Eq. (C), i.e.,

$$|E_0(\omega)| = |D(\omega) - P_c(\omega)| \leq \frac{\delta_p}{|W(\omega)|} \quad \text{for } \omega \in \Omega \quad (\text{C})$$

we get

$$|E_0(\omega)| \leq \begin{cases} \delta_p & \text{for } 0 \leq \omega \leq \omega_p \\ \delta_a & \text{for } \omega_a \leq \omega \leq \pi \end{cases}$$

# Minimax Problem

- The most appropriate approach for the solution of the optimization problem just described is to solve the minimax problem

$$\underset{\mathbf{x}}{\text{minimize}} \{ \max_{\omega} |E(\omega)| \}$$

where

$$\mathbf{x} = [a_0 \quad a_1 \quad \cdots \quad a_c]^T$$

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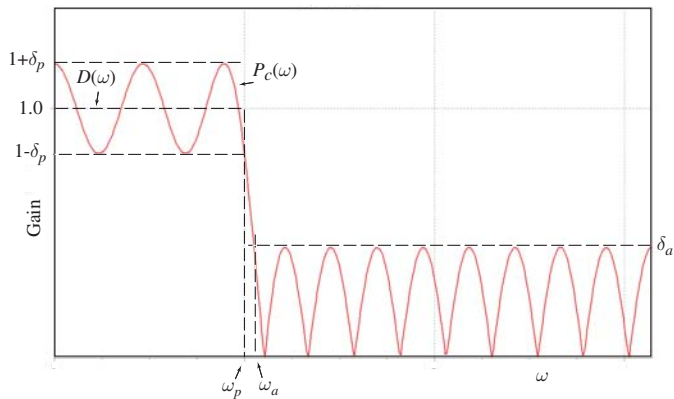
$$\underset{\mathbf{x}}{\text{minimize}} \quad \left\{ \max_{\omega} |E(\omega)| \right\}$$

where

$$\mathbf{x} = [a_0 \quad a_1 \quad \cdots \quad a_c]^T$$

- By virtue of the so-called *alternation theorem*, there is a *unique equiripple* solution of the above minimax problem.
- Note** that weighted-Chebyshev filters are so called because they have an *equiripple* amplitude response just like Chebyshev filters but are not related to Chebyshev filters in any other way.

# Minimax Problem *Cont'd*



## Alternation Theorem

If  $P_c(\omega)$  is a linear combination of  $r = c + 1$  cosine functions of the form

$$P_c(\omega) = \sum_{k=0}^c a_k \cos k\omega$$

then a necessary and sufficient condition that  $P_c(\omega)$  be the unique, best, weighted-Chebyshev approximation to a continuous function  $D(\omega)$  on  $\Omega$ , where  $\Omega$  is a dense and compact subset of the frequency interval  $[0, \pi]$ , is that the weighted error function  $E(\omega)$  exhibit at least  $r + 1$  extremal frequencies  $\hat{\omega}_i$  in  $\Omega$  such that

$$\hat{\omega}_0 < \hat{\omega}_1 < \dots < \hat{\omega}_r$$

$$E(\hat{\omega}_i) = -E(\hat{\omega}_{i+1}) \quad \text{for } i = 0, 1, \dots, r - 1$$

and

$$|E(\hat{\omega}_i)| = \max_{\omega \in \Omega} |E(\omega)| \quad \text{for } i = 0, 1, \dots, r$$

## Alternation Theorem *Cont'd*

- From the alternation theorem and Eq. (B), i.e.,

$$E(\omega) = W(\omega)[D(\omega) - P_c(\omega)] \quad (\text{B})$$

we can write

$$E(\hat{\omega}_i) = W(\hat{\omega}_i)[D(\hat{\omega}_i) - P_c(\hat{\omega}_i)] = (-1)^i \delta$$

for  $i = 0, 1, \dots, r$ , where  $\delta$  is a constant.

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for  $i = 0, 1, \dots, r$ , where  $\delta$  is a constant.

- The above system of equations can be put in matrix form as

$$\begin{bmatrix} 1 & \cos \hat{\omega}_0 & \cos \hat{\omega}_0 & \cdots & \cos \hat{\omega}_0 & \frac{1}{W(\hat{\omega}_0)} \\ 1 & \cos \hat{\omega}_1 & \cos \hat{\omega}_1 & \cdots & \cos \hat{\omega}_1 & \frac{-1}{W(\hat{\omega}_1)} \\ \cdots & \cdots & \cdots & \cdots & \cdots & \cdots \\ 1 & \cos \hat{\omega}_r & \cos \hat{\omega}_r & \cdots & \cos \hat{\omega}_r & \frac{(-1)^r}{W(\hat{\omega}_r)} \end{bmatrix} \begin{bmatrix} a_0 \\ a_1 \\ \vdots \\ a_c \\ \delta \end{bmatrix} = \begin{bmatrix} D(\hat{\omega}_0) \\ D(\hat{\omega}_1) \\ \vdots \\ D(\hat{\omega}_{r-1}) \\ D(\hat{\omega}_r) \end{bmatrix}$$

## Alternation Theorem *Cont'd*

- If the extremal frequencies (or extremals for short) were known, coefficients  $a_k$  and, in turn, the frequency response of the filter could be computed using Eq. (A), i.e.,

$$P_c(\omega) = \sum_{k=0}^c a_k \cos k\omega \quad (\text{A})$$

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- The solution of this system exists since the above  $(r + 1) \times (r + 1)$  matrix is known to be nonsingular.

## Remez Exchange Algorithm *Cont'd*

- The Remez exchange algorithm is an *iterative multivariable algorithm* that is naturally suited for the solution of the minimax problem just described.
- It is based on the second optimization method of Remez.



## Basic Remez Exchange Algorithm *Cont'd*

1. Initialize extremal frequencies  $\hat{\omega}_0, \hat{\omega}_1, \dots, \hat{\omega}_r$  and ensure that an extremal is assigned at each band edge.

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2. Solve the system of equations to get  $\delta$  and the coefficients  $a_0, a_1, \dots, a_c$ .
3. Using the coefficients  $a_0, a_1, \dots, a_c$ , calculate  $P_c(\omega)$  and the magnitude of the error

$$|E(\omega)| = |W(\omega)[D(\omega) - P_c(\omega)]|$$

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$$|E(\omega)| = |W(\omega)[D(\omega) - P_c(\omega)]|$$

4. Locate the frequencies  $\hat{\omega}_0, \hat{\omega}_1, \dots, \hat{\omega}_p$  at which  $|E(\omega)|$  is maximum and  $|E(\hat{\omega}_i)| \geq \delta$  (these frequencies are *potential extremals* for the next iteration).

5. Compute the convergence parameter

$$Q = \frac{\max |E(\hat{\omega}_i)| - \min |E(\hat{\omega}_i)|}{\max |E(\hat{\omega}_i)|}$$

where  $i = 0, 1, \dots, \rho$ .

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6. Reject  $\rho - r$  *superfluous potential extremals*  $\hat{\omega}_i$  according to an appropriate rejection criterion and renumber the remaining  $\hat{\omega}_i$  by setting  $\hat{\omega}_i = \hat{\omega}_i$  for  $i = 0, 1, \dots, r$ .

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7. If  $Q > \varepsilon$ , where  $\varepsilon$  is a convergence tolerance (say  $\varepsilon = 0.01$ ), repeat from step 2; otherwise continue to step 8.

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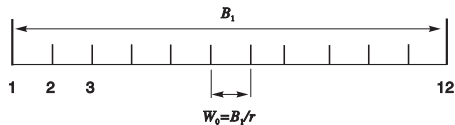
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7. If  $Q > \varepsilon$ , where  $\varepsilon$  is a convergence tolerance (say  $\varepsilon = 0.01$ ), repeat from step 2; otherwise continue to step 8.
8. Compute  $P_c(\omega)$  using the last set of extremal frequencies; then deduce  $h(n)$ , the impulse response of the required filter, and stop.

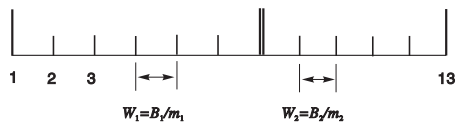


# Initialization of Extremal Frequencies

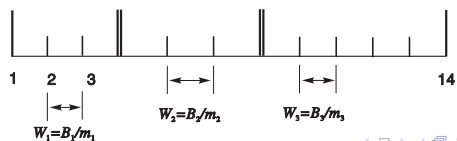
Bands: 1  
Extremals:  $r+1$  (12)  
Intervals:  $r$  (11)



Bands: 2  
Extremals:  $r+1$  (13)  
Intervals:  $r-1$  (11)



Bands: 3  
Extremals:  $r+1$  (14)  
Intervals:  $r-2$  (11)



## Initialization of Extremal Frequencies *Cont'd*

For a filter with  $J$  bands with bandwidths  $B_1, B_2, \dots, B_J$ , the number of extremals and interval between extremals for each band can be calculated by using the following formulas:

$$W_0 = \frac{1}{r+1-J} \sum_{j=1}^J B_j$$

$$m_j = \left( \frac{B_j}{W_0} + 0.5 \right) \quad \text{for } j = 1, 2, \dots, J-1$$

$$\text{and } m_J = r - \sum_{j=1}^{J-1} (m_j + 1)$$

$$W_j = \frac{B_j}{m_j} \quad \text{for } j = 1, 2, \dots, J$$

where  $r = (N+1)/2$  and  $N$  is the filter length.

# Updating of Extremals

- In each iteration, the extremals need to be updated. This is done by finding the maxima of the error function

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- This could be done by solving the system

$$\begin{bmatrix} 1 & \cos \hat{\omega}_0 & \cos \hat{\omega}_0 & \cdots & \cos \hat{\omega}_0 & \frac{1}{W(\hat{\omega}_0)} \\ 1 & \cos \hat{\omega}_1 & \cos \hat{\omega}_1 & \cdots & \cos \hat{\omega}_1 & \frac{-1}{W(\hat{\omega}_1)} \\ \cdots & \cdots & \cdots & \cdots & \cdots & \cdots \\ 1 & \cos \hat{\omega}_r & \cos \hat{\omega}_r & \cdots & \cos \hat{\omega}_r & \frac{(-1)^r}{W(\hat{\omega}_r)} \end{bmatrix} \begin{bmatrix} a_0 \\ a_1 \\ \vdots \\ a_c \\ \delta \end{bmatrix} = \begin{bmatrix} D(\hat{\omega}_0) \\ D(\hat{\omega}_1) \\ \vdots \\ D(\hat{\omega}_{r-1}) \\ D(\hat{\omega}_r) \end{bmatrix}$$

for the coefficients  $a_k$  and then calculating

$$P_c(\omega) = \sum_{k=0}^c a_k \cos k\omega$$

and in turn  $E(\omega)$ .

- This approach is inefficient and may be subject to numerical ill-conditioning, in particular if  $\delta$  is small and  $N$  is large.

*Note:* A  $50 \times 50$  matrix is quite typical.

- An alternative and more efficient approach is to deduce  $\delta$  analytically (by using Cramer's rule) and then interpolate  $P_c(\omega)$  on the  $r$  frequency points using the barycentric form of the Lagrange interpolation formula, as follows:

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- Calculate parameter  $\delta$  as

$$\delta = \sum_{k=0}^r \frac{\alpha_k D(\hat{\omega}_k)}{\frac{\sum_{k=0}^r (-1)^k \alpha_k}{W(\hat{\omega})}}$$

- With  $\delta$  known,  $P_c(\omega)$  can be obtained as

$$P_c(\omega) = \begin{cases} C_k & \text{for } \omega = \hat{\omega}_0, \hat{\omega}_1, \dots, \hat{\omega}_{r-1} \\ \frac{\sum_{k=0}^{r-1} \frac{\beta_k C_k}{x - x_k}}{\sum_{k=0}^{r-1} \frac{\beta_k}{x - x_k}} & \text{otherwise} \end{cases}$$

where  $\alpha_k = \prod_{i=0, i \neq k}^{r-1} \frac{1}{x_k - x_i}$ ,  $\beta_k = \prod_{i=0, i \neq k}^{r-1} \frac{1}{x_k - x_i}$

and  $C_k = D(\hat{\omega}_k) - (-1)^k \frac{\delta}{W(\hat{\omega}_k)}$

with  $x = \cos \omega$  and  $x_i = \cos \hat{\omega}_i$  for  $i = 0, 1, \dots, r$



# Rejection of Superfluous Potential Extremals

- The problem formulation is such that *there must be exactly  $r + 1$  extremals* in each iteration.

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- The problem formulation is such that *there must be exactly  $r + 1$  extremals* in each iteration.
- Analysis will show that  $|E(\omega)|$  can have as many as  $r + 2J - 1$  maxima where  $J$  is the number of bands.

If in any iteration the number of maxima exceeds  $r + 1$ , then the iteration is said to have generated *superfluous potential extremals*.

## Rejection of Superfluous Potential Extremals *Cont'd*

- In the standard McClellan, Rabiner, and Parks algorithm, this difficulty is circumvented by rejecting the  $\rho - r$  potential extremals  $\hat{\omega}_j$  that yield the lowest error  $|E(\omega)|$ .

# Computation of Impulse Response

- The impulse response in Step 8 of the algorithm can be determined by recalling that function  $P_c(\omega)$  is the frequency response of a noncausal version of the required filter.

# Computation of Impulse Response

- The impulse response in Step 8 of the algorithm can be determined by recalling that function  $P_c(\omega)$  is the frequency response of a noncausal version of the required filter.
- The impulse response of the noncausal filter, denoted as  $h_0(n)$  for  $-c \leq n \leq c$ , can be determined by computing  $P_c(k\Omega)$  for  $k = 0, 1, \dots, c$  where  $\Omega = 2\pi/N$ , and then using the inverse discrete Fourier transform.

## Computation of Impulse Response *Cont'd*

- It can be shown that

$$h_0(n) = h_0(-n) = \frac{1}{N} \left\{ P_c(0) + \sum_{k=1}^c 2P_c(k\Omega) \cos\left(\frac{2\pi kn}{N}\right) \right\}$$

for  $n = 0, 1, \dots, c$ .

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for  $n = 0, 1, \dots, c$ .

- The impulse response of the required causal filter is given by

$$h(n) = h_0(n - c)$$

for  $n = 0, 1, \dots, c$ .

# Example

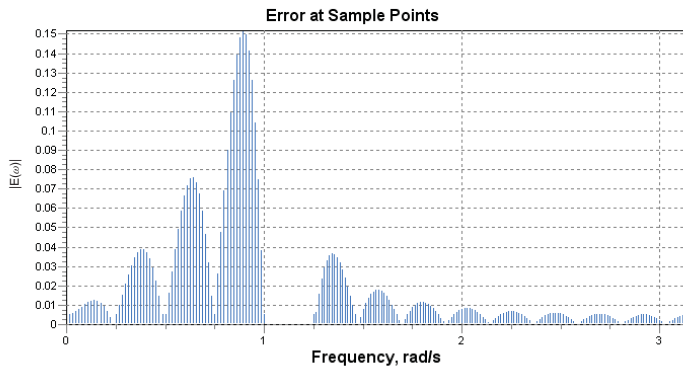
Band	$D(\omega)$	$W(\omega)$	Left band edge	Right band edge
1	1	1	0	1.0
2	0	0.4	1.25	$\pi$
Sampling frequency: $2\pi$				



# Example *Cont'd*

Filter length: 27  
Iteration no: 1

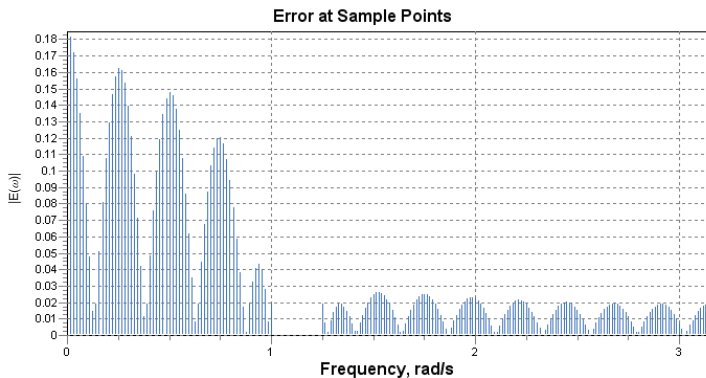
Function Evals: 0



# Example *Cont'd*

Filter length: 27  
Iteration no: 2

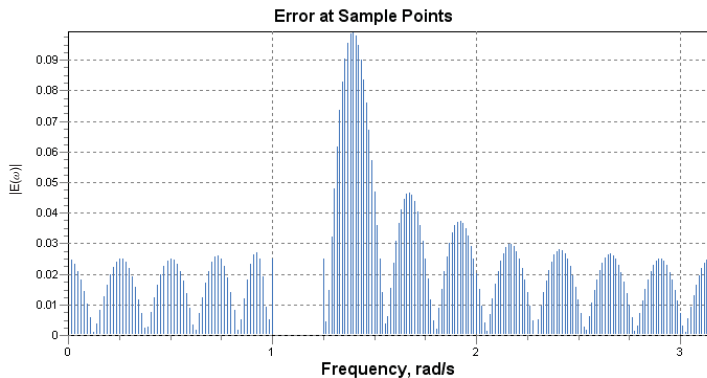
Function Evals: 199



# Example *Cont'd*

Filter length: 27  
Iteration no: 3

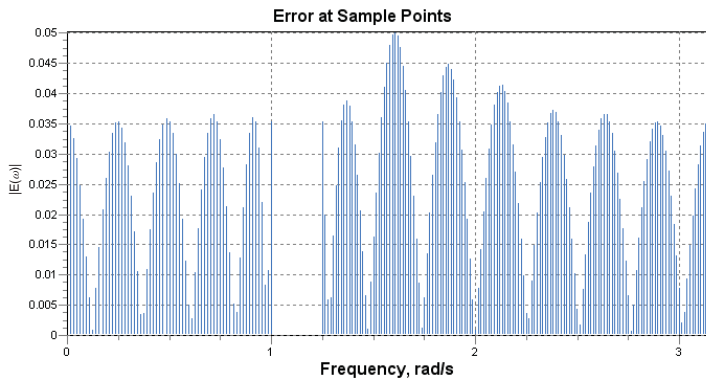
Function Evals: 398



# Example *Cont'd*

Filter length: 27  
Iteration no: 4

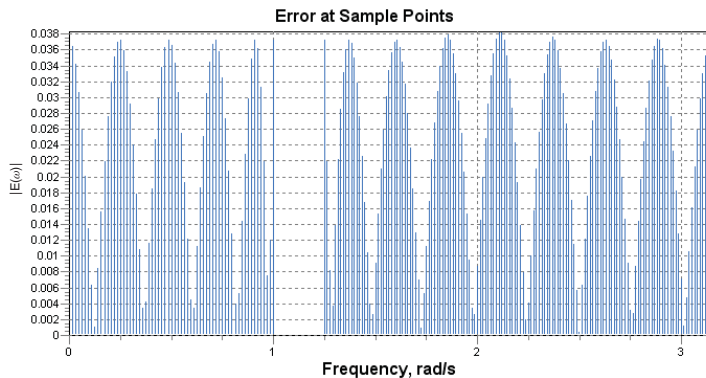
Function Evals: 597



# Example *Cont'd*

Filter length: 27  
Iteration no: 5

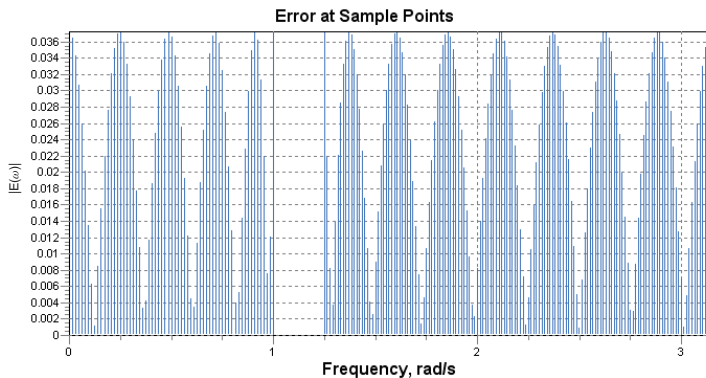
Function Evals: 796



# Example *Cont'd*

Filter length: 27  
Iteration no: 6

Function Evals: 995



# Selective Step-by-Step Search

- When the system of equations

$$\begin{bmatrix} 1 & \cos \hat{\omega}_0 & \cos \hat{\omega}_0 & \cdots & \cos \hat{\omega}_0 & \frac{1}{W(\hat{\omega}_0)} \\ 1 & \cos \hat{\omega}_1 & \cos \hat{\omega}_1 & \cdots & \cos \hat{\omega}_1 & \frac{-1}{W(\hat{\omega}_1)} \\ \cdots & \cdots & \cdots & \cdots & \cdots & \cdots \\ 1 & \cos \hat{\omega}_r & \cos \hat{\omega}_r & \cdots & \cos \hat{\omega}_r & \frac{(-1)^r}{W(\hat{\omega}_r)} \end{bmatrix} \begin{bmatrix} a_0 \\ a_1 \\ \vdots \\ a_c \\ \delta \end{bmatrix} = \begin{bmatrix} D(\hat{\omega}_0) \\ D(\hat{\omega}_1) \\ \vdots \\ D(\hat{\omega}_{r-1}) \\ D(\hat{\omega}_r) \end{bmatrix}$$

is solved, the error function  $|E(\omega)|$  is forced to satisfy the relation

$$|E(\hat{\omega}_i)| = |W(\hat{\omega}_i)[D(\hat{\omega}_i) - P_c(\hat{\omega}_i)]| = |\delta|$$

# Selective Step-by-Step Search

- When the system of equations

$$\begin{bmatrix} 1 & \cos \hat{\omega}_0 & \cos \hat{\omega}_0 & \cdots & \cos \hat{\omega}_0 & \frac{1}{W(\hat{\omega}_0)} \\ 1 & \cos \hat{\omega}_1 & \cos \hat{\omega}_1 & \cdots & \cos \hat{\omega}_1 & \frac{-1}{W(\hat{\omega}_1)} \\ \cdots & \cdots & \cdots & \cdots & \cdots & \cdots \\ 1 & \cos \hat{\omega}_r & \cos \hat{\omega}_r & \cdots & \cos \hat{\omega}_r & \frac{(-1)^r}{W(\hat{\omega}_r)} \end{bmatrix} \begin{bmatrix} a_0 \\ a_1 \\ \vdots \\ a_c \\ \delta \end{bmatrix} = \begin{bmatrix} D(\hat{\omega}_0) \\ D(\hat{\omega}_1) \\ \vdots \\ D(\hat{\omega}_{r-1}) \\ D(\hat{\omega}_r) \end{bmatrix}$$

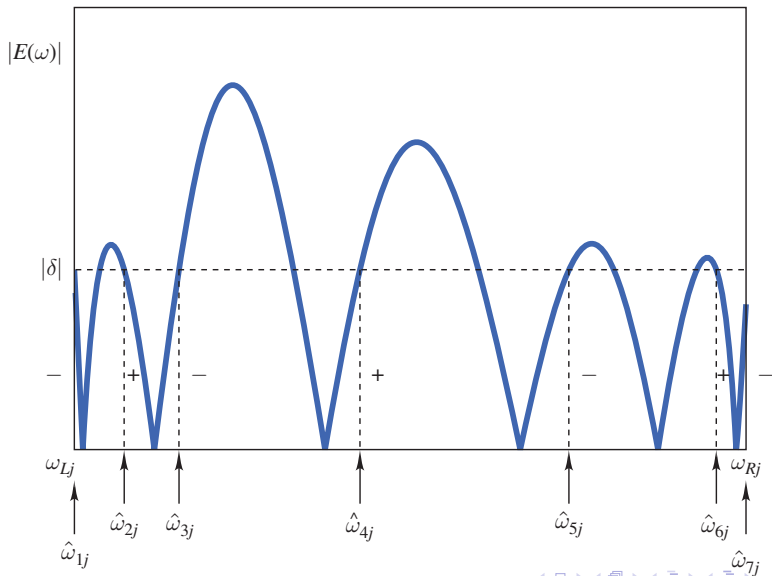
is solved, the error function  $|E(\omega)|$  is forced to satisfy the relation

$$|E(\hat{\omega}_i)| = |W(\hat{\omega}_i)[D(\hat{\omega}_i) - P_c(\hat{\omega}_i)]| = |\delta|$$

- This relation can be satisfied in a number of ways but the most likely possibility for the  $j$ th band is illustrated in the next slide where  $\omega_{Lj}$  and  $\omega_{Rj}$  are the left-hand and right-hand edges, respectively.



# Selective Step-by-Step Search *Cont'd*

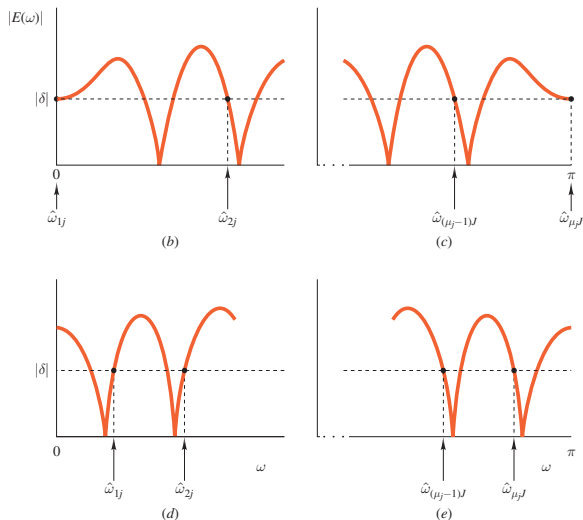


Because of the special nature of the error function

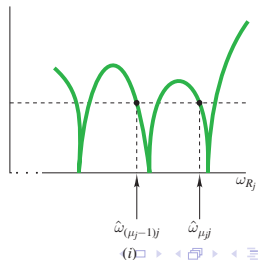
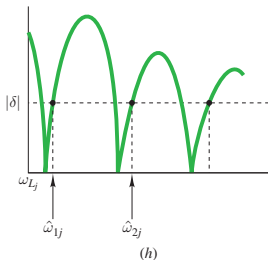
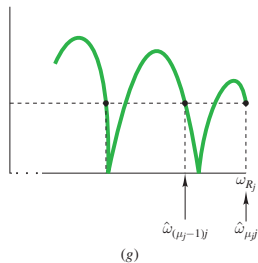
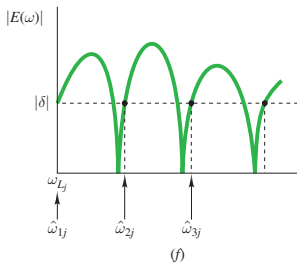
- (a) the maxima of  $|E(\omega)|$  can be easily found by searching in the vicinity of the extremals;
- (b) gradient information can be used to expedite the search for the maxima of  $|E(\omega)|$ ; and
- (c) the closer we get to the solution, the closer are the maxima of the error function to the extremals.

By using a *selective step-by-step search*, a large amount of computation can be eliminated.

Extra ripples can arise in the first and last bands.:



Also in interior bands:



# Cubic Interpolation Search

- Increased computational efficiency can be achieved by using a search based on cubic interpolation.

# Cubic Interpolation Search

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- Assuming that the error function shown in the figure can be represented by the third-order polynomial

$$|E(\omega)| = M = a + b\omega + c\omega^2 + d\omega^3$$

where  $a$ ,  $b$ ,  $c$ , and  $d$  are constants then

$$\frac{dM}{d\omega} = G = b + 2c\omega + 3d\omega^2$$

Hence, the frequencies at which  $M$  has stationary points are given by

$$\bar{\omega} = \frac{1}{3d} \left[ -c \pm \sqrt{(c^2 - 3bd)} \right]$$

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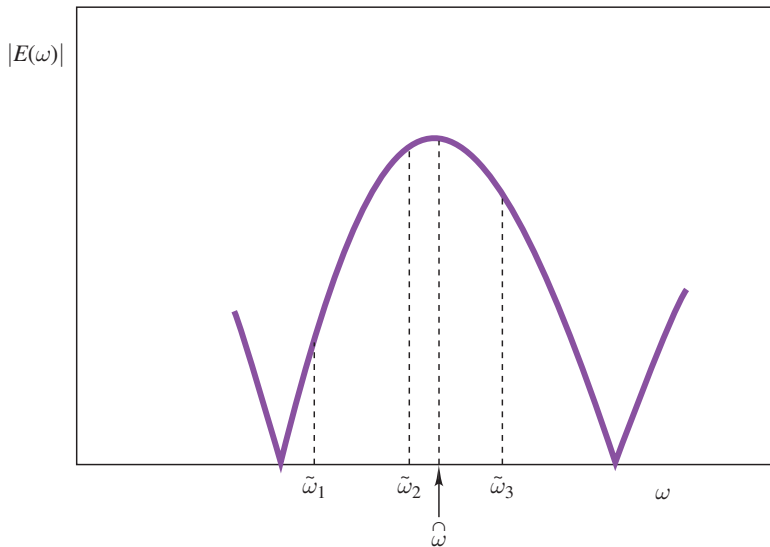
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$$\bar{\omega} = \frac{1}{3d} \left[ -c \pm \sqrt{(c^2 - 3bd)} \right]$$

- Therefore,  $|E(\omega)|$  has a maximum if

$$\frac{d^2M}{d\omega^2} = 2c + 6d\hat{\omega} < 0 \quad \text{or} \quad \hat{\omega} < -\frac{c}{3d}$$

# Cubic Interpolation Search *Cont'd*





## Cubic Interpolation Search *Cont'd*

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## Cubic Interpolation Search *Cont'd*

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- By using the cubic interpolation to start with and then switching over to the step-by-step search, an very efficient algorithm can be constructed.
- The decision to switch from cubic to selective can be based on the value of the convergence parameter  $Q$  (see Step 5).

Switching from the cubic to the selective when  $Q$  is reduced below 0.65 works well.

# Improved Rejection Scheme for Superfluous Potential Extremals

- If an extremal does not move from one iteration to the next, then the minimum value of  $E(\hat{\omega}_i)$  is simply  $\delta$ , as can be easily shown, and this happens quite often even in the first or second iteration of the Remez algorithm.

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- As a consequence, rejecting potential extremals on the basis of the individual values of  $E(\hat{\omega}_i)$  tends to become random and this can slow the Remez algorithm quite significantly particularly for multiband filters.
- An improved scheme for the rejection of superfluous extremals based the rejection on the lowest average band error as well as the individual values of  $E(\hat{\omega}_i)$  is described in the next transparency.

- Compute the average band errors

$$E_j = \frac{1}{\nu_j} \sum_{\hat{\omega}_i \in \Omega_j} |E(\hat{\omega}_i)| \quad \text{for } j = 1, 2, \dots, J$$

where  $\Omega_j$  is the set of extremals in band  $j$  given by

$$\Omega_j = \{\hat{\omega}_i : \omega_{Lj} \leq \hat{\omega}_i \leq \omega_{Rj}\}$$

$\nu_j$  is the number of potential extremals in band  $j$ , and  $J$  is the number of bands.



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$v_j$  is the number of potential extremals in band  $j$ , and  $J$  is the number of bands.

- Rank the  $J$  bands in the order of lowest average error and let  $l_1, l_2, \dots, l_J$  be the ranked list obtained, i.e.,  $l_1$  and  $l_J$  are the bands with the lowest and highest average error, respectively.

## Improved Rejection Scheme *Cont'd*

- Reject one  $\hat{\omega}_j$  in each of bands  $I_1, I_2, \dots, I_{J-1}, I_1, I_2, \dots$  until  $\rho - r$  superfluous  $\hat{\omega}_j$  are rejected.

In each case, reject the  $\hat{\omega}_j$ , other than a band edge, that yields the lowest  $|E(\hat{\omega}_j)|$  in the band.

### Example:

If  $J = 3$ ,  $\rho - r = 3$ , and the average errors for bands 1, 2, and 3 are 0.05, 0.08, and 0.02, then  $\hat{\omega}_j$  are rejected in bands 3, 1, and 3.

**Note:** The potential extremals are not rejected in band 2 which is the band of highest average error.

# Example

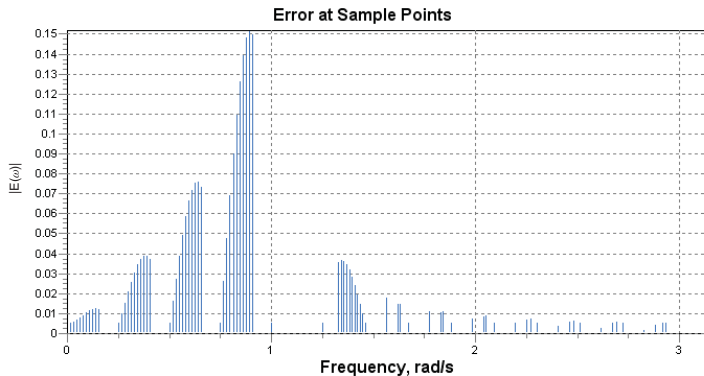
Band	$D(\omega)$	$W(\omega)$	Left band edge	Right band edge
1	1	1	0	1.0
2	0	0.4	1.25	$\pi$
Sampling frequency: $2\pi$				

# Example *Cont'd*

Filter length: 27

Function Evals: 0

Iteration no: 1

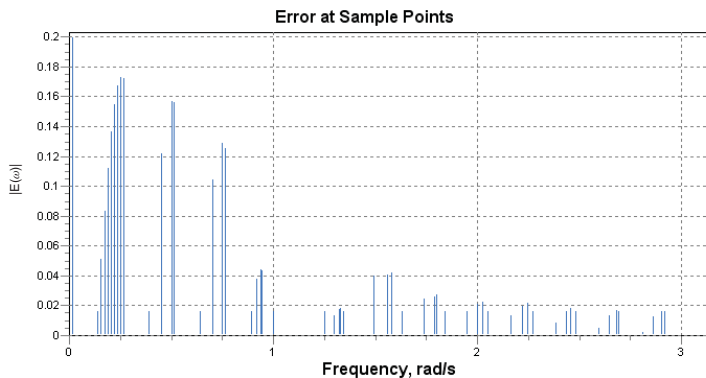


# Example *Cont'd*

Filter length: 27

Iteration no: 2

Function Evals: 87

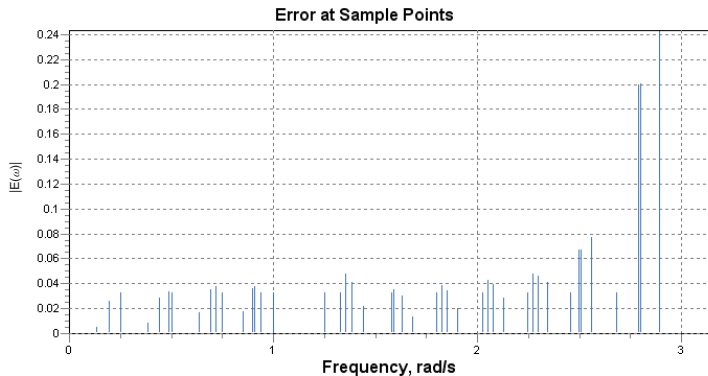


# Example *Cont'd*

Filter length: 27

Iteration no: 3

Function Evals: 134

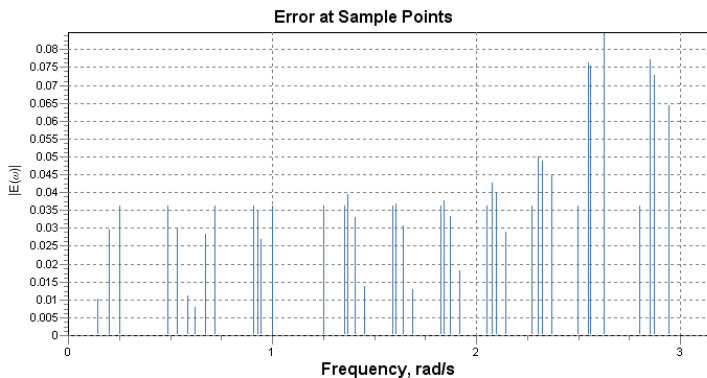


# Example *Cont'd*

Filter length: 27

Iteration no: 4

Function Evals: 171

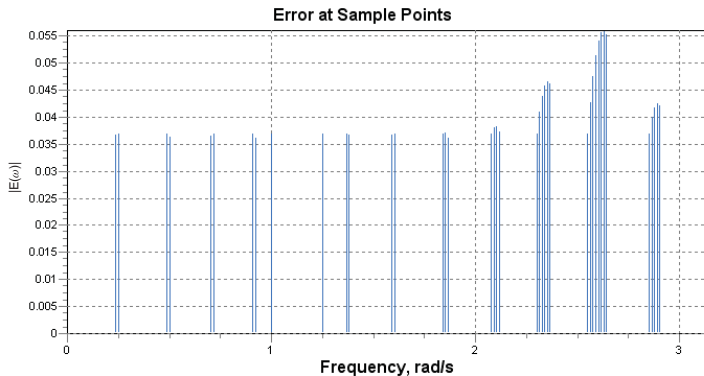


# Example *Cont'd*

Filter length: 27

Iteration no: 5

Function Evals: 208



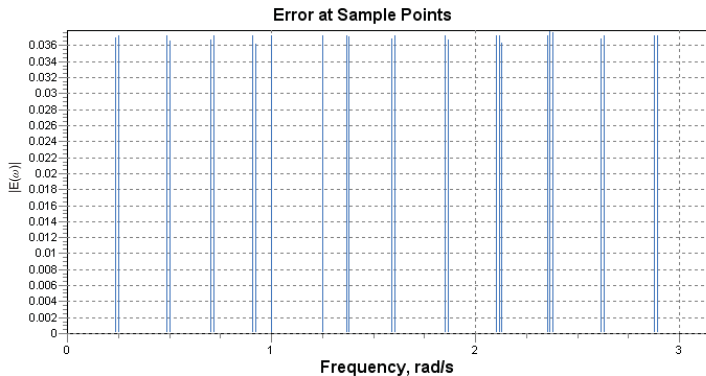


# Example *Cont'd*

Filter length: 27

Function Evals: 250

Iteration no: 6

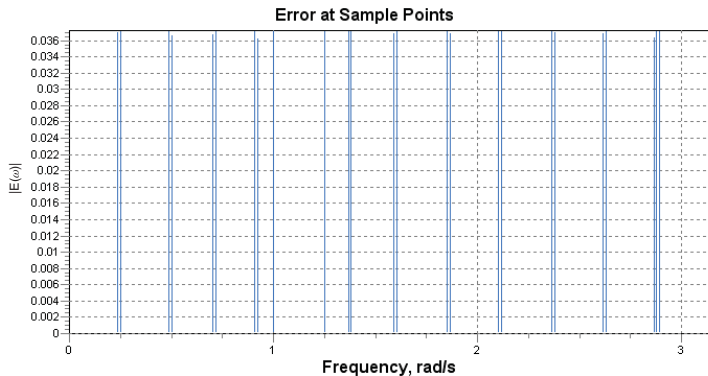


# Example *Cont'd*

Filter length: 27

Iteration no: 7

Function Evals: 278



# Comparisons — Amount of Computation

Type of Filter	No. of Examples	Range of $N$	Ave. Funct. Evals.			Saving, %	
			A	B	C	C v B	C v A
LP	45	9-101	2691	722	372	48.9	86.3
HP	42	9-101	2774	710	356	49.9	87.2
BP	44	21-89	2777	667	338	49.3	87.8
BS	35	21-91	2720	639	336	47.4	87.6

A: Exhaustive search

B: Selective search

C: Selective plus cubic search

# Comparisons — Robustness

Type of Filter	No. of Examples	No. Failures		
		A	B	C
LP	46	1	0	0
HP	43	1	0	0
BP	50	3	2	5
BS	45	6	8	8

A: Exhaustive search

B: Selective search

C: Selective plus cubic search

# Prescribed Specifications

- Given a filter length  $N$ , a set of passband and stopband edges, and a ratio  $\delta_p/\delta_a$ , an FIR filter with approximately piecewise-constant amplitude-response specifications can be readily designed.

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- While the filter obtained will have passband and stopband edges at the correct locations and the ratio  $\delta_p/\delta_a$  will be exactly as required, the amplitudes of the passband and stopband ripples are highly unlikely to have the specified values.
- An acceptable design can be obtained by predicting the value of  $N$  on the basis of the required specifications and then designing filters for increasing or decreasing values of  $N$  until the lowest value of  $N$  that satisfies the specifications is found.

# Filter Length Prediction

- A reasonably accurate empirical formula for the prediction of the required filter length,  $N$ , for the case of lowpass and highpass filters, due to Herrmann, Rabiner, and Chan, is

$$N = \text{int} \left[ \frac{(D - FB^2)}{B} + 1.5 \right]$$

where

$$B = |\omega_a - \omega_p|/2\pi$$

$$D = [0.005309(\log_{10} \delta_p)^2 + 0.07114 \log_{10} \delta_p - 0.4761] \log_{10} \delta_a \\ - [0.00266(\log_{10} \delta_p)^2 + 0.5941 \log_{10} \delta_p + 0.4278]$$

$$F = 0.51244(\log_{10} \delta_p - \log_{10} \delta_a) + 11.012$$



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- The formula of Herrmann et al. can also be used to predict the filter length in the design of bandpass, bandstop, and multiband filters in general.

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- The formula of Herrmann et al. can also be used to predict the filter length in the design of bandpass, bandstop, and multiband filters in general.
- In these filters, a value of  $N$  is computed for each transition band between a passband and stopband or a stopband and passband and the largest value of  $N$  so obtained is taken to be the predicted filter length.

# Algorithm

1. Compute  $N$  using the prediction formula of Herrmann et al.; if  $N$  is even, set  $N = N + 1$ .
2. Design a filter of length  $N$  using the Remez algorithm and determine the minimum value of  $\delta$ , say  $\bar{\delta}$ .
  - (A) If  $\bar{\delta} > \delta_p$ , then do:
    - (a) Set  $N = N + 2$ , design a filter of length  $N$  using the Remez algorithm, and find  $\bar{\delta}$ ;
    - (b) If  $\bar{\delta} \leq \delta_p$ , then go to step 3; else, go to step 2(A)(a).
  - (B) If  $\bar{\delta} < \delta_p$ , then do:
    - (a) Set  $N = N - 2$ , design a filter of length  $N$  using the Remez algorithm, and find  $\bar{\delta}$ ;
    - (b) If  $\bar{\delta} > \delta_p$ , then go to step 4; else, go to step 2(B)(a).

3. If part A of the algorithm was executed, use the last set of extremals and the corresponding value of  $N$  to obtain the impulse response of the required filter and stop.
4. If part B of the algorithm was executed, use the last but one set of extremals and the corresponding value of  $N$  to obtain the impulse response of the required filter and stop.

## Example

In an application, an FIR equiripple bandstop filter is required which should satisfy the following specifications:

- Odd filter length
- Maximum passband ripple  $A_p$ : 0.5 dB
- Minimum stopband attenuation  $A_a$ : 50.0 dB
- Lower passband edge  $\omega_{p1}$ : 0.8 rad/s
- Upper passband edge  $\omega_{p2}$ : 2.2 rad/s
- Lower stopband edge  $\omega_{a1}$ : 1.2 rad/s
- Upper stopband edge  $\omega_{a2}$ : 1.8 rad/s
- Sampling frequency  $\omega_s$ :  $2\pi$  rad/s

Design the lowest-order filter that will satisfy the specifications.

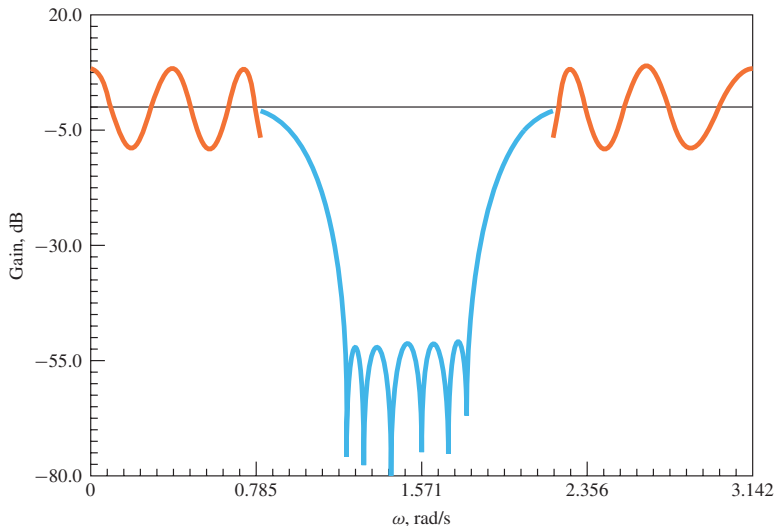
The design algorithm gave a filter with the following specifications:

- Passband ripple: 0.4342 dB
- Minimum stopband attenuation: 51.23 dB

### Progress of Algorithm

$N$	Iters.	FE's	$A_p$ , dB	$A_a$ , dB
31	10	582	0.5055	49.91
33	7	376	0.5037	49.94
35	9	545	0.4342	51.23

## Example *Cont'd*



**Note:** Passband errors multiplied by a factor of 40.



# D-Filter

A DSP software package that incorporates the design techniques described in this presentation is *D-Filter*. Please see

<http://www.d-filter.ece.uvic.ca>

for more information.



# Summary

- Three design techniques that bring about substantial improvements in the efficiency of the Remez algorithm have been described:
  - A step-by-step exhaustive search
  - A cubic interpolation search
  - An improved scheme for the rejection of superfluous potential extremals

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- Three design techniques that bring about substantial improvements in the efficiency of the Remez algorithm have been described:
  - A step-by-step exhaustive search
  - A cubic interpolation search
  - An improved scheme for the rejection of superfluous potential extremals
- These techniques are implemented in a DSP software package known as D-Filter.
- Extensive experimentation has shown that the selective and cubic interpolation searches reduce the amount of computation required by the Remez algorithm by almost 90% without degrading its robustness.

- The rejection scheme described increases the efficiency and robustness of the Remez algorithm further but the scheme has not been compared with the original method of McClellan, Rabiner, and Parks.

## Summary *Cont'd*

- The rejection scheme described increases the efficiency and robustness of the Remez algorithm further but the scheme has not been compared with the original method of McClellan, Rabiner, and Parks.
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## Summary *Cont'd*

- The rejection scheme described increases the efficiency and robustness of the Remez algorithm further but the scheme has not been compared with the original method of McClellan, Rabiner, and Parks.
- By using a prediction technique for the required filter length proposed by Herrmann, Rabiner, and Chan, filters that satisfy prescribed specifications can be designed.
- For off-line applications, the Remez algorithm continues to be the method of choice for the design of linear-phase filters, multiband filters, differentiators, Hilbert transformers.

## Summary *Cont'd*

- Despite the improvements described, the Remez algorithm continues to require a large amount of computation.

For applications that need the filter to be designed on-line in real or quasi-real time, the window method is preferred although the filters obtained are suboptimal.

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*This slide concludes the presentation.  
Thank you for your attention.*