

# Part 3: IIR Filters – Bilinear Transformation Method

## Tutorial ISCAS 2007

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- A procedure for the design of IIR filters that would satisfy arbitrary prescribed specifications will be described.

- A procedure for the design of IIR filters that would satisfy arbitrary prescribed specifications will be described.
- The method is based on the bilinear transformation and it can be used to design lowpass (LP), highpass (HP), bandpass (BP), and bandstop (BS), Butterworth, Chebyshev, Inverse-Chebyshev, and Elliptic filters.

**Note:** The material for this module is taken from Antoniou, *Digital Signal Processing: Signals, Systems, and Filters*, Chap. 12.)

Given an analog filter with a continuous-time transfer function  $H_A(s)$ , a digital filter with a discrete-time transfer function  $H_D(z)$  can be readily deduced by applying the bilinear transformation as follows:

$$H_D(z) = H_A(s) \Big|_{s = \frac{2}{T} \left( \frac{z-1}{z+1} \right)} \quad (\text{A})$$

The bilinear transformation method has the following important features:

- A stable analog filter gives a stable digital filter.

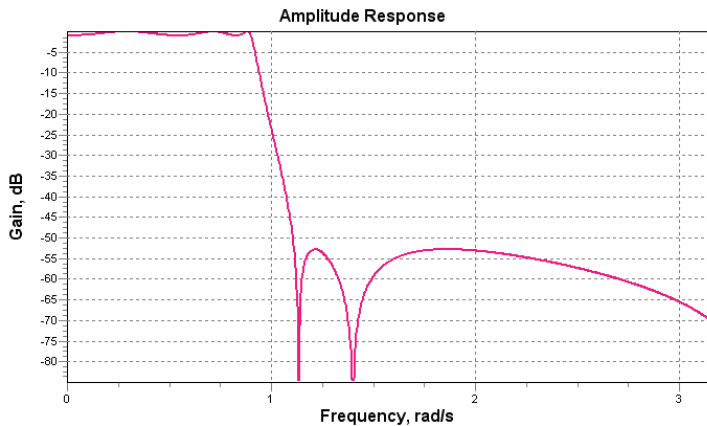
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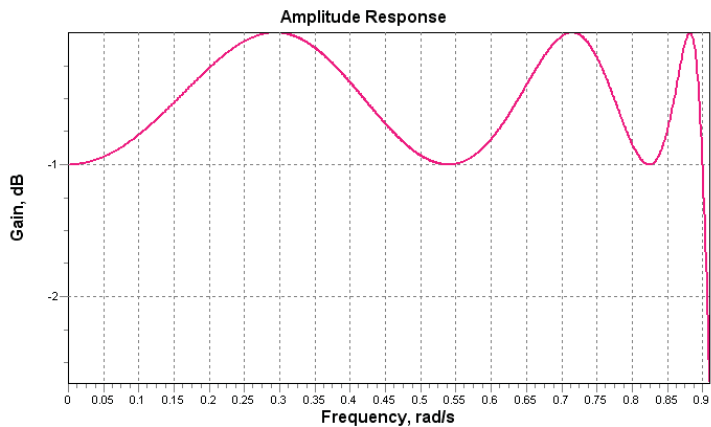
- A stable analog filter gives a stable digital filter.
- The maxima and minima of the amplitude response in the analog filter are preserved in the digital filter.

As a consequence,

- the passband ripple, and
- the minimum stopband attenuation

of the analog filter are preserved in the digital filter.







# The Warping Effect

If we let  $\omega$  and  $\Omega$  represent the frequency variable in the analog filter and the derived digital filter, respectively, then Eq. (A), i.e.,

$$H_D(z) = H_A(s) \Big|_{s=\frac{2}{T}\left(\frac{z-1}{z+1}\right)} \quad (\text{A})$$

gives the frequency response of the digital filter as a function of the frequency response of the analog filter as

$$H_D(e^{j\Omega T}) = H_A(j\omega)$$

provided that  $s = \frac{2}{T} \left( \frac{z-1}{z+1} \right)$

or 
$$j\omega = \frac{2}{T} \left( \frac{e^{j\Omega T} - 1}{e^{j\Omega T} + 1} \right) \quad \text{or} \quad \omega = \frac{2}{T} \tan \frac{\Omega T}{2} \quad (\text{B})$$

• • •

$$\omega = \frac{2}{T} \tan \frac{\Omega T}{2} \quad (\text{B})$$

- For  $\Omega < 0.3/T$

$$\omega \approx \Omega$$

and, as a result, the digital filter has the same frequency response as the analog filter over this frequency range.

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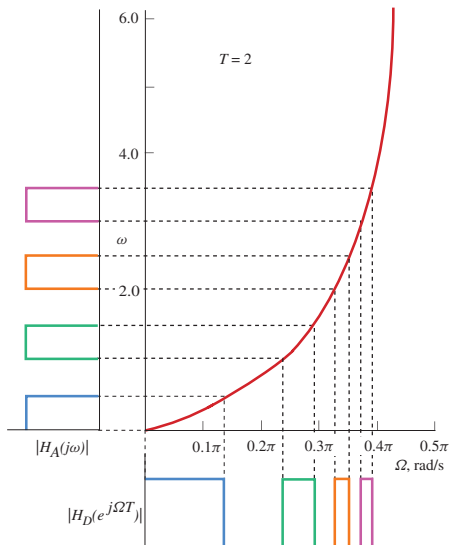
$$\omega \approx \Omega$$

and, as a result, the digital filter has the same frequency response as the analog filter over this frequency range.

- For higher frequencies, however, the relation between  $\omega$  and  $\Omega$  becomes nonlinear, and *distortion is introduced in the frequency scale of the digital filter relative to that of the analog filter.*

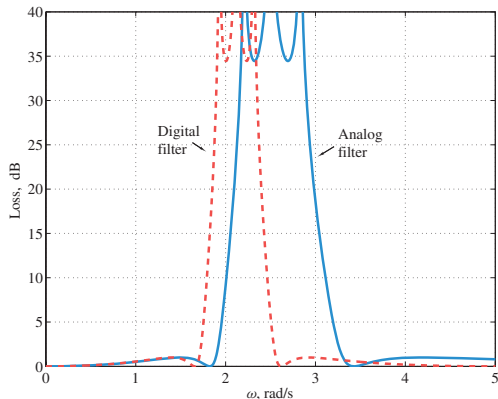
This is known as the *warping effect*.

# The Warping Effect *Cont'd*



## The Warping Effect *Cont'd*

The warping effect changes the band edges of the digital filter relative to those of the analog filter in a nonlinear way, as illustrated for the case of a BS filter:



- From Eq. (B), i.e.,

$$\omega = \frac{2}{T} \tan \frac{\Omega T}{2} \quad (\text{B})$$

a frequency  $\omega$  in the analog filter corresponds to a frequency  $\Omega$  in the digital filter and hence

$$\Omega = \frac{2}{T} \tan^{-1} \frac{\omega T}{2}$$

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$$\Omega = \frac{2}{T} \tan^{-1} \frac{\omega T}{2}$$

- If  $\omega_1, \omega_2, \dots, \omega_i, \dots$  are the passband and stopband edges in the analog filter, then the corresponding passband and stopband edges in the derived digital filter are given by

$$\Omega_i = \frac{2}{T} \tan^{-1} \frac{\omega_i T}{2} \quad i = 1, 2, \dots$$

- If prescribed passband and stopband edges  $\tilde{\Omega}_1, \tilde{\Omega}_2, \dots, \tilde{\Omega}_j, \dots$  are to be achieved, the analog filter must be prewarped before the application of the bilinear transformation to ensure that its band edges are given by

$$\omega_j = \frac{2}{T} \tan \frac{\tilde{\Omega}_j T}{2}$$



- If prescribed passband and stopband edges  $\tilde{\Omega}_1, \tilde{\Omega}_2, \dots, \tilde{\Omega}_i, \dots$  are to be achieved, the analog filter must be prewarped before the application of the bilinear transformation to ensure that its band edges are given by

$$\omega_i = \frac{2}{T} \tan \frac{\tilde{\Omega}_i T}{2}$$

- Then the band edges of the digital filter would assume their prescribed values  $\Omega_i$  since

$$\begin{aligned}\Omega_i &= \frac{2}{T} \tan^{-1} \frac{\omega_i T}{2} \\ &= \frac{2}{T} \tan^{-1} \left( \frac{T}{2} \cdot \frac{2}{T} \tan \frac{\tilde{\Omega}_i T}{2} \right) \\ &= \tilde{\Omega}_i \quad \text{for } i = 1, 2, \dots\end{aligned}$$

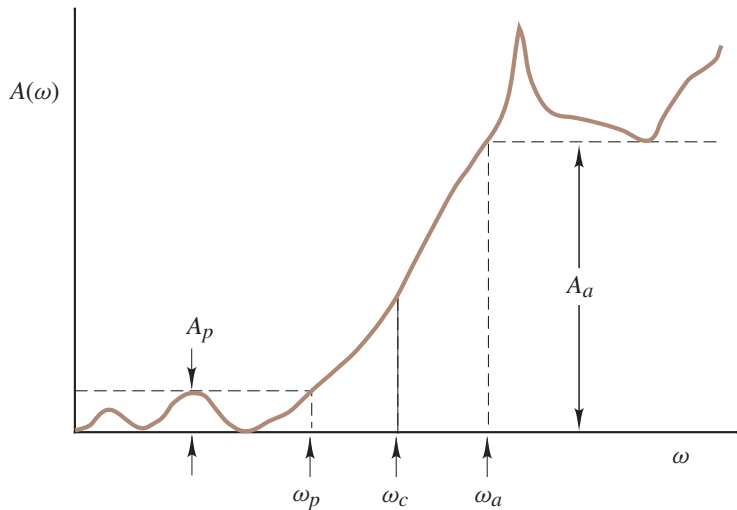
Consider a normalized analog LP filter characterized by  $H_N(s)$  with an attenuation

$$A_N(\omega) = 20 \log \frac{1}{|H_N(j\omega)|}$$

(also known as loss) and assume that

$$\begin{aligned} 0 \leq A_N(\omega) \leq A_p \quad \text{for } 0 \leq |\omega| \leq \omega_p \\ A_N(\omega) \geq A_a \quad \text{for } \omega_a \leq |\omega| \leq \infty \end{aligned}$$

**Note:** The transfer functions of analog LP filters are reported in the literature in normalized form whereby the passband edge is typically of the order of unity.



# Design Procedure

A denormalized LP, HP, BP, or BS filter that has the same passband ripple and minimum stopband attenuation as a given normalized LP filter can be derived from the normalized LP filter through the following steps:

1. Apply the transformation  $s = f_X(\bar{s})$

$$H_X(\bar{s}) = H_N(s) \Big|_{s=f_X(\bar{s})}$$

where  $f_X(\bar{s})$  is one of the four standard analog-filters transformations, given by the next slide.

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where  $f_X(\bar{s})$  is one of the four standard analog-filters transformations, given by the next slide.

2. Apply the bilinear transformation to  $H_X(\bar{s})$ , i.e.,

$$H_D(z) = H_X(\bar{s}) \Big|_{\bar{s}=\frac{2}{T}\left(\frac{z-1}{z+1}\right)}$$

Standard forms of  $f_X(\bar{s})$ 

X	$f_X(\bar{s})$
LP	$\lambda \bar{s}$
HP	$\lambda / \bar{s}$
BP	$\frac{1}{B} \left( \bar{s} + \frac{\omega_0^2}{\bar{s}} \right)$
BS	$\frac{B \bar{s}}{\bar{s}^2 + \omega_0^2}$

- The digital filter designed by this method will have the required passband and stopband edges only if the parameters  $\lambda$ ,  $\omega_0$ , and  $B$  of the analog-filter transformations and the order of the continuous-time normalized LP transfer function,  $H_N(s)$ , are chosen appropriately.

- The digital filter designed by this method will have the required passband and stopband edges only if the parameters  $\lambda$ ,  $\omega_0$ , and  $B$  of the analog-filter transformations and the order of the continuous-time normalized LP transfer function,  $H_N(s)$ , are chosen appropriately.
- This is obviously a difficult problem but general solutions are available for LP, HP, BP, and BS, Butterworth, Chebyshev, inverse-Chebyshev, and Elliptic filters.

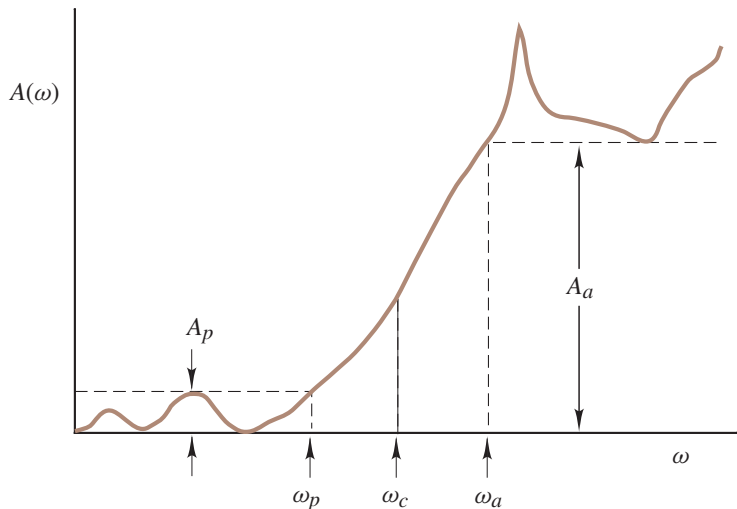


# General Design Procedure for LP Filters

An outline of the methodology for the derivation of general solutions for LP filters is as follows:

1. Assume that a continuous-time normalized LP transfer function,  $H_N(s)$ , is available that would give the required passband ripple,  $A_p$ , and minimum stopband attenuation (loss),  $A_a$ .

Let the passband and stopband edges of the analog filter be  $\omega_p$  and  $\omega_a$ , respectively.



Attenuation characteristic of continuous-time normalized LP filter

2. Apply the LP-to-LP analog-filter transformation to  $H_N(s)$  to obtain a denormalized discrete-time transfer function  $H_{LP}(\bar{s})$ .

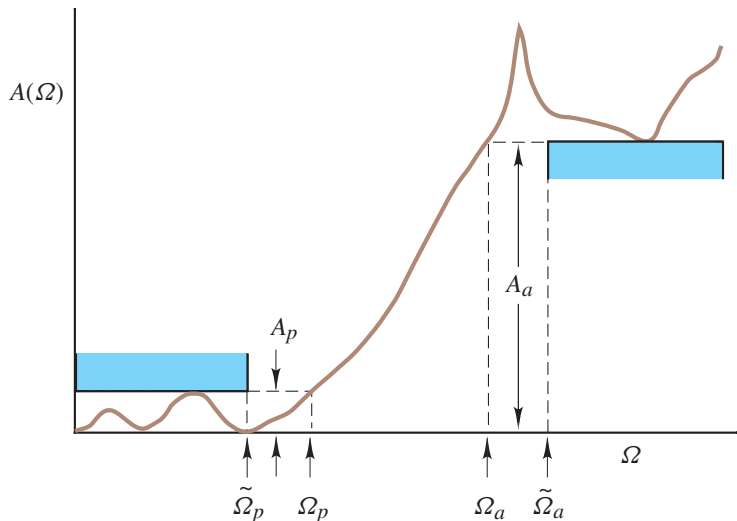
2. Apply the LP-to-LP analog-filter transformation to  $H_N(s)$  to obtain a denormalized discrete-time transfer function  $H_{LP}(\bar{s})$ .
3. Apply the bilinear transformation to  $H_{LP}(\bar{s})$  to obtain a discrete-time transfer function  $H_D(z)$ .

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3. Apply the bilinear transformation to  $H_{LP}(\bar{s})$  to obtain a discrete-time transfer function  $H_D(z)$ .
4. At this point, assume that the derived discrete-time transfer function has passband and stopband edges that satisfy the relations

$$\tilde{\Omega}_p \leq \Omega_p \quad \text{and} \quad \Omega_a \leq \tilde{\Omega}_a$$

where  $\tilde{\Omega}_p$  and  $\tilde{\Omega}_a$  are the *prescribed* passband and stopband edges, respectively.

In effect, we assume that the digital filter has passband and stopband edges that *satisfy* or *oversatisfy* the required specifications.



Attenuation characteristic of required LP digital filter

5. Solve for  $\lambda$ , the parameter of the LP-to-LP analog-filter transformation.

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6. Find the minimum value of the ratio  $\omega_p/\omega_a$  for the continuous-time normalized LP transfer function.

The ratio  $\omega_p/\omega_a$  is a fraction less than unity and it is a measure of the steepness of the transition characteristic. It is often referred to as the *selectivity* of the filter.

The selectivity of a filter *dictates the minimum order* to achieve the required specifications.

**Note:** As the selectivity approaches unity, the filter-order tends to infinity!



7. The same methodology is applicable for HP filters, except that the LP-HP analog-filter transformation is used in Step 2.

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8. The application of this methodology yields the formulas summarized by the table shown in the next slide.

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$$\frac{\omega_p}{\omega_a} \geq K_0$$

LP

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$$\lambda = \frac{\omega_p T}{2 \tan(\tilde{\Omega}_p T/2)}$$

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$$\frac{\omega_p}{\omega_a} \geq \frac{1}{K_0}$$

HP

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$$\lambda = \frac{2\omega_p \tan(\tilde{\Omega}_p T/2)}{T}$$

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where  $K_0 = \frac{\tan(\tilde{\Omega}_p T/2)}{\tan(\tilde{\Omega}_a T/2)}$

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- The table of formulas presented is applicable to all the classical types of analog filters, namely,
  - Butterworth
  - Chebyshev
  - Inverse-Chebyshev
  - Elliptic

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  - Butterworth
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  - Elliptic
- Formulas that can be used to design digital versions of these filters will be presented later.

# General Design Procedure for BP Filters

An outline of the methodology for the derivation of general solutions for BP filters is as follows:

1. Assume that a continuous-time normalized LP transfer function,  $H_N(s)$ , is available that would give the required passband ripple,  $A_p$ , and minimum stopband attenuation,  $A_a$ .

Let the passband and stopband edges of the analog filter be  $\omega_p$  and  $\omega_a$ , respectively.

2. Apply the LP-to-BP analog-filter transformation to  $H_N(s)$  to obtain a denormalized discrete-time transfer function  $H_{BP}(\bar{s})$ .

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3. Apply the bilinear transformation to  $H_{BP}(\bar{s})$  to obtain a discrete-time transfer function  $H_D(z)$ .



4. At this point, assume that the derived discrete-time transfer function has passband and stopband edges that satisfy the relations

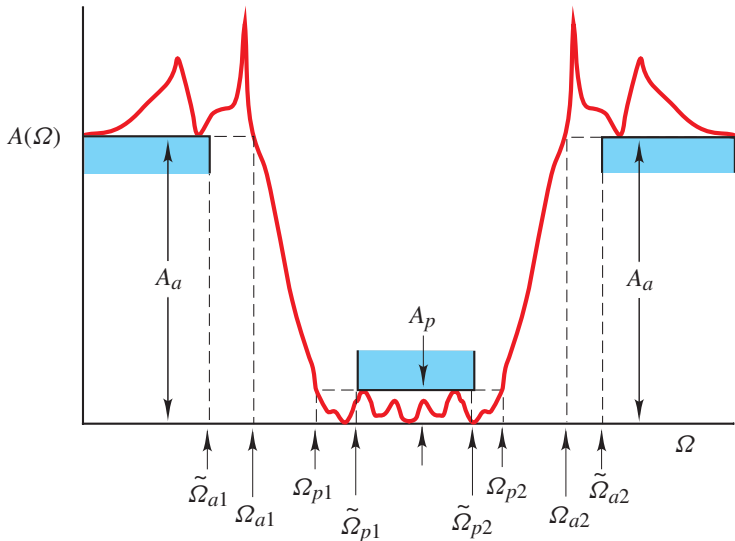
$$\Omega_{p1} \leq \tilde{\Omega}_{p1} \quad \Omega_{p2} \geq \tilde{\Omega}_{p2}$$

and

$$\Omega_{a1} \geq \tilde{\Omega}_{a1} \quad \Omega_{a2} \leq \tilde{\Omega}_{a2}$$

where

- $\Omega_{p1}$  and  $\Omega_{p2}$  are the actual lower and upper passband edges,
- $\tilde{\Omega}_{p1}$  and  $\tilde{\Omega}_{p2}$  are the *prescribed* lower and upper passband edges,
- $\Omega_{a1}$  and  $\Omega_{a2}$  are the actual lower and upper stopband edges,
- $\tilde{\Omega}_{a1}$  and  $\tilde{\Omega}_{a2}$  are the *prescribed* lower and upper stopband edges, respectively.



Attenuation characteristic of required BP digital filter

5. Solve for  $B$  and  $\omega_0$ , the parameters of the LP-to-BP analog-filter transformation.

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7. The same methodology can be used for the design of BS filters except that the LP-to-BS transformation is used in Step 2.

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7. The same methodology can be used for the design of BS filters except that the LP-to-BS transformation is used in Step 2.
8. The application of this methodology yields the formulas summarized in the next two slides.

# Formulas for the design of BP Filters

$$\omega_0 = \frac{2\sqrt{K_B}}{T}$$

$$\text{BP} \quad \frac{\omega_p}{\omega_a} \geq \begin{cases} K_1 & \text{if } K_C \geq K_B \\ K_2 & \text{if } K_C < K_B \end{cases}$$

$$B = \frac{2K_A}{T\omega_p}$$

$$\text{where} \quad K_A = \tan \frac{\tilde{\Omega}_{p2} T}{2} - \tan \frac{\tilde{\Omega}_{p1} T}{2} \quad K_B = \tan \frac{\tilde{\Omega}_{p1} T}{2} \tan \frac{\tilde{\Omega}_{p2} T}{2}$$
$$K_C = \tan \frac{\tilde{\Omega}_{a1} T}{2} \tan \frac{\tilde{\Omega}_{a2} T}{2} \quad K_1 = \frac{K_A \tan(\tilde{\Omega}_{a1} T/2)}{K_B - \tan^2(\tilde{\Omega}_{a1} T/2)}$$
$$K_2 = \frac{K_A \tan(\tilde{\Omega}_{a2} T/2)}{\tan^2(\tilde{\Omega}_{a2} T/2) - K_B}$$

# Formulas for the Design of BS Filters

$$\omega_0 = \frac{2\sqrt{K_B}}{T}$$

$$\text{BS} \quad \frac{\omega_p}{\omega_a} \geq \begin{cases} \frac{1}{K_2} & \text{if } K_C \geq K_B \\ \frac{1}{K_1} & \text{if } K_C < K_B \end{cases}$$

$$B = \frac{2K_A\omega_p}{T}$$

$$\text{where} \quad K_A = \tan \frac{\tilde{\Omega}_{p2} T}{2} - \tan \frac{\tilde{\Omega}_{p1} T}{2} \quad K_B = \tan \frac{\tilde{\Omega}_{p1} T}{2} \tan \frac{\tilde{\Omega}_{p2} T}{2}$$
$$K_C = \tan \frac{\tilde{\Omega}_{a1} T}{2} \tan \frac{\tilde{\Omega}_{a2} T}{2} \quad K_1 = \frac{K_A \tan(\tilde{\Omega}_{a1} T/2)}{K_B - \tan^2(\tilde{\Omega}_{a1} T/2)}$$
$$K_2 = \frac{K_A \tan(\tilde{\Omega}_{a2} T/2)}{\tan^2(\tilde{\Omega}_{a2} T/2) - K_B}$$



## Formulas for $\omega_p$ and $n$

- The formulas presented apply to any type of normalized analog LP filter with an attenuation that would satisfy the following conditions:

$$0 \leq A_N(\omega) \leq A_p \quad \text{for } 0 \leq |\omega| \leq \omega_p$$
$$A_N(\omega) \geq A_a \quad \text{for } \omega_a \leq |\omega| \leq \infty$$

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- However, the values of the normalized passband edge,  $\omega_p$ , and the required filter order,  $n$ , depend on the type of filter.

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- However, the values of the normalized passband edge,  $\omega_p$ , and the required filter order,  $n$ , depend on the type of filter.
- Formulas for these parameters for Butterworth, Chebyshev, and Elliptic filters are presented in the next three slides.

# Formulas for Butterworth Filters

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$$\text{LP} \quad K = K_0$$

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$$\text{HP} \quad K = \frac{1}{K_0}$$

---

$$\text{BP} \quad K = \begin{cases} K_1 & \text{if } K_C \geq K_B \\ K_2 & \text{if } K_C < K_B \end{cases}$$

---

$$\text{BS} \quad K = \begin{cases} \frac{1}{K_2} & \text{if } K_C \geq K_B \\ \frac{1}{K_1} & \text{if } K_C < K_B \end{cases}$$

---

$$n \geq \frac{\log D}{2 \log(1/K)}, \quad D = \frac{10^{0.1A_a} - 1}{10^{0.1A_p} - 1}$$

$$\omega_p = (10^{0.1A_p} - 1)^{1/2n}$$

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# Formulas for Chebyshev Filters

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$$\text{LP} \quad K = K_0$$

---

$$\text{HP} \quad K = \frac{1}{K_0}$$

---

$$\text{BP} \quad K = \begin{cases} K_1 & \text{if } K_C \geq K_B \\ K_2 & \text{if } K_C < K_B \end{cases}$$

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$$\text{BS} \quad K = \begin{cases} \frac{1}{K_2} & \text{if } K_C \geq K_B \\ \frac{1}{K_1} & \text{if } K_C < K_B \end{cases}$$

---

$$n \geq \frac{\cosh^{-1} \sqrt{D}}{\cosh^{-1}(1/K)}, \quad D = \frac{10^{0.1A_a} - 1}{10^{0.1A_p} - 1}$$

$$\omega_p = 1$$

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# Formulas for Elliptic Filters

	$k$	$\omega_p$
LP	$K = K_0$	$\sqrt{K_0}$
HP	$K = \frac{1}{K_0}$	$\frac{1}{\sqrt{K_0}}$
BP	$K = \begin{cases} K_1 & \text{if } K_C \geq K_B \\ K_2 & \text{if } K_C < K_B \end{cases}$	$\begin{cases} \sqrt{K_1} \\ \sqrt{K_2} \end{cases}$
BS	$K = \begin{cases} \frac{1}{K_2} & \text{if } K_C \geq K_B \\ \frac{1}{K_1} & \text{if } K_C < K_B \end{cases}$	$\begin{cases} \frac{1}{\sqrt{K_2}} \\ \frac{1}{\sqrt{K_1}} \end{cases}$
$n \geq \frac{\cosh^{-1} \sqrt{D}}{\cosh^{-1}(1/K)}, \quad D = \frac{10^{0.1A_a} - 1}{10^{0.1A_p} - 1}$		

## Example – HP Filter

An HP filter that would satisfy the following specifications is required:

$$A_p = 1 \text{ dB}, \quad A_a = 45 \text{ dB}, \quad \tilde{\Omega}_p = 3.5 \text{ rad/s},$$

$$\tilde{\Omega}_a = 1.5 \text{ rad/s}, \quad \omega_s = 10 \text{ rad/s}.$$

Design a Butterworth, a Chebyshev, and then an Elliptic digital filter.

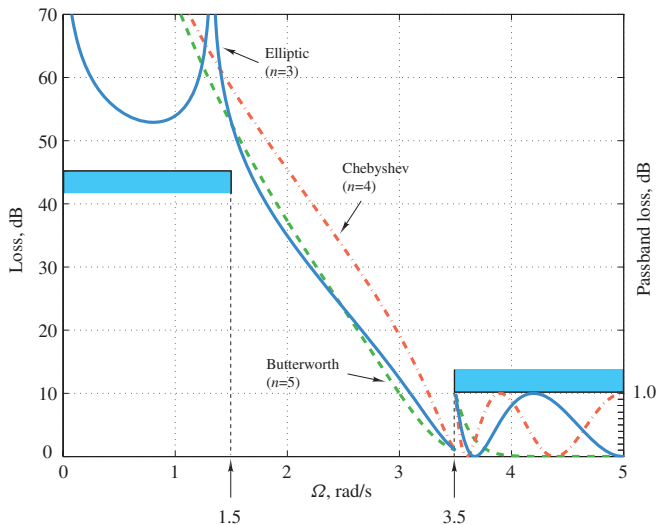
## Example – HP Filter *Cont'd*

### Solution

Filter type	$n$	$\omega_p$	$\lambda$
Butterworth	5	0.873610	5.457600
Chebyshev	4	1.0	6.247183
Elliptic	3	0.509526	3.183099



# Example *Cont'd*



## Example – BP Filter

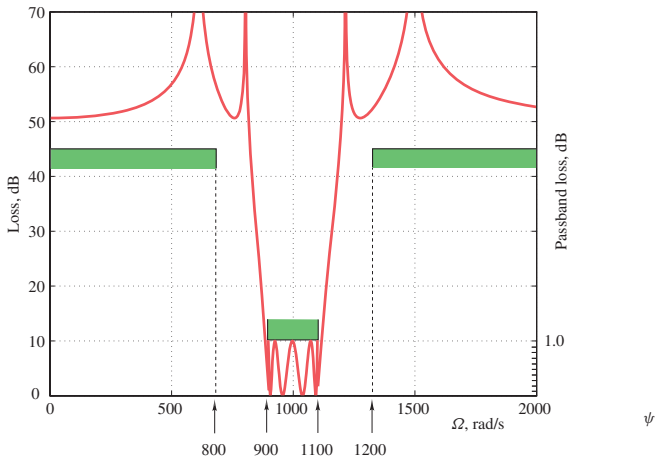
Design an Elliptic BP filter that would satisfy the following specifications:

$$A_p = 1 \text{ dB}, \quad A_a = 45 \text{ dB}, \quad \tilde{\Omega}_{p1} = 900 \text{ rad/s}, \quad \tilde{\Omega}_{p2} = 1100 \text{ rad/s}, \\ \tilde{\Omega}_{a1} = 800 \text{ rad/s}, \quad \tilde{\Omega}_{a2} = 1200 \text{ rad/s}, \quad \omega_s = 6000 \text{ rad/s}.$$

### Solution

$$k = 0.515957 \\ \omega_p = 0.718302 \\ n = 4 \\ \omega_0 = 1,098.609 \\ B = 371.9263$$

# Example – BP Filter *Cont'd*



## Example – BS Filter

Design a Chebyshev BS filter that would satisfy the following specifications:

$$A_p = 0.5 \text{ dB}, \quad A_a = 40 \text{ dB}, \quad \tilde{\Omega}_{p1} = 350 \text{ rad/s}, \quad \tilde{\Omega}_{p2} = 700 \text{ rad/s},$$

$$\tilde{\Omega}_{a1} = 430 \text{ rad/s}, \quad \tilde{\Omega}_{a2} = 600 \text{ rad/s}, \quad \omega_s = 3000 \text{ rad/s}.$$

### Solution

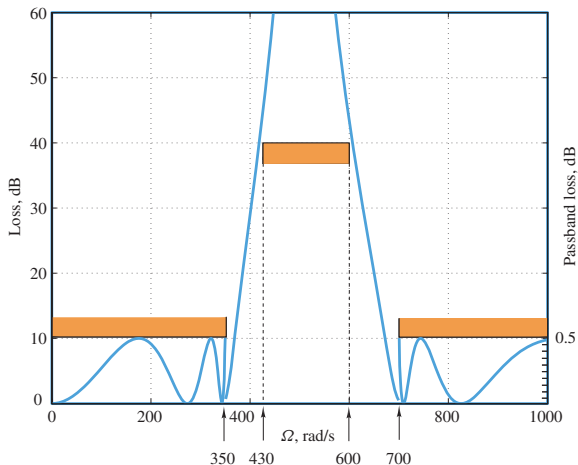
$$\omega_p = 1.0$$

$$n = 5$$

$$\omega_0 = 561,4083$$

$$B = 493,2594$$

# Example – BS Filter *Cont'd*



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

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- The method requires *very little computation* and leads to very precise *optimal* designs.



# Summary

- A design method for IIR filters that leads to a complete description of the transfer function *in closed form* either in terms of its zeros and poles or its coefficients has been described.
- The method requires *very little computation* and leads to very precise *optimal* designs.
- It can be used to design LP, HP, BP, and BS filters of the Butterworth, Chebyshev, Inverse-Chebyshev, Elliptic types.

# Summary

- A design method for IIR filters that leads to a complete description of the transfer function *in closed form* either in terms of its zeros and poles or its coefficients has been described.
- The method requires *very little computation* and leads to very precise *optimal* designs.
- It can be used to design LP, HP, BP, and BS filters of the Butterworth, Chebyshev, Inverse-Chebyshev, Elliptic types.
- All these designs can be carried out by using DSP software package D-Filter.

# References

- A. Antoniou, *Digital Signal Processing: Signals, Systems, and Filters*, Chap. 15, McGraw-Hill, 2005.
- A. Antoniou, *Digital Signal Processing: Signals, Systems, and Filters*, Chap. 15, McGraw-Hill, 2005.

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