## 4.11 Impulse Invariant Transformation (IIT)

If  $h_a(t)$  represents the impulse response of an analog filter, then the unit sample response of discrete-time filter H(z) used in an A/D-H(z)-D/A structure is selected to be the sampled version of h(t).

That is, 
$$h(n) = h_a(nT)$$
$$= h_a(t)|_{t=nT}$$

and the discrete (digital) transfer function is

$$H(z) = \mathcal{Z}\{h(n)\}$$

Notice that the digital transfer function H(z) is the Z-transform of the unit sample respons h(n), while the analog transfer function  $H_a(s)$  is the Laplace transform of the unit impulse respons

 $h_a(t)$ . Don't be tempted to write  $H(z) = H(s)|_{s=z}$ , because it is incorrect. Let us now generalize this procedure and at the same time show that H(z) can be obtained

Consider an analog transfer function with N different poles that has the s-domain transfe directly from  $H_a(s)$  without intervening steps of finding  $h_a(t)$  and then  $h_a(nT)$ . function written in partial fraction expansion form as

$$H_a(s) = \sum_{i=1}^{N} \frac{C_i}{s - s_i}$$
 (4.49)

with the corresponding unit impulse response

$$h_a(t) = \sum_{i=1}^{N} C_i e^{s_i t}$$
 (4.50)

In the above expression,  $C_i$  is the constant associated with the partial fraction expansion of  $H_a(s)$ . If this response is sampled every T seconds (t = nT), we have the sampled response

$$h_a(nT) = h(n) = \sum_{i=1}^{N} C_i \ e^{sinT}$$
 (4.51)

Finally, we take the Z-transform of equation (4.51) to obtain the discrete transfer function of the digital IIR filter

$$H(z) = \mathcal{Z}\{h(n)\}\$$
  
=  $\sum_{n=0}^{\infty} h(n)z^{-n}$  (4.52)

The lower limit of summation in the above equation is zero because the filter is assumed to be

Substituting equation (4.51) in equation (4.52), we get

$$H(z) = \sum_{n=0}^{\infty} \sum_{i=1}^{N} C_{i} e^{i_{i}nT} z^{-n}$$

$$= \sum_{i=1}^{N} C_{i} \sum_{n=0}^{\infty} \left[ e^{i_{1}T} z^{-1} \right]^{n}$$

$$= \sum_{i=1}^{N} C_{i} \frac{1}{1 - e^{i_{1}T} z^{-1}}$$
(4.53)

Comparing equations (4.49) and (4.53), we find that

$$\frac{1}{s - s_i} \xrightarrow{\text{IIT}} \frac{1}{1 - e^{s_i T} z^{-1}} = \frac{z}{z - e^{s_i T}}$$
(4.54)

Equation (4.54) shows that the analog pole at  $s = s_i$  is mapped to a digital pole at  $z_i = e^{s_i T}$ . The transformed digital filter H(z) has the following properties: . Its order is same as that of  $H_a(s)$  because the common denominator on the right-hand side

2. Its poles are mapped according to

$$s_i \stackrel{\text{II}}{\Longrightarrow} z_i = e^{s_i T}, \quad 1 \leqslant i \leqslant N$$

That is, the analog and digital poles are related as per the equation

$$z = e^{sT}$$

Letting  $z = re^{j\omega}$  and  $s = \sigma + j\Omega$  in the above equation, we get

$$v = e^{\sigma T} e^{j\Omega T}$$

$$r = e^{\sigma T}$$

$$r = e^{\sigma T}$$

Hence,

the interval  $-\pi\leqslant\omega\leqslant\pi$  , where q is an integer. Thus, the mapping of analog frequency,  $\Omega$  in to digital frequency  $\omega$  is many-to-one, which simply reflects the effects of aliasing due to sampling. Fig. 4.13 shows the mapping from s-plane to z-plane using impulse invariant interval  $-\pi \leqslant \omega \leqslant \pi$ . In general, the interval  $(2q-1)^{\frac{\pi}{p}} \leqslant \Omega \leqslant (2q+1)^{\frac{\pi}{p}}$  also maps into one-to-one. The mapping,  $\omega = \Omega T$  implies that the interval  $-\frac{\pi}{r} \leqslant \Omega \leqslant \frac{\pi}{r}$  maps into the the right half pole in s is mapped outside the unit circle in the z-plane. Thus, a stable analog filter  $H_a(s)$  is transformed to a stable digital filter H(z). Also, the image of the  $j\Omega$  axis in the z-plane is the unit circle as indicated above. However, the mapping of the j \Omega axis is not Consequently,  $\sigma < 0$  implies that 0 < r < 1 and  $\sigma > 0$  implies that r > 1. When  $\sigma = 0$ , we have r=1. Hence, the left-half pole is mapped inside the unit circle in the z-plane and transformation.

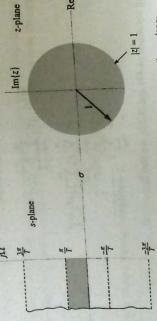


Fig. 4.13 The impulse invariant transformation from the s-plane to the z- plane: the imaginary axis maps to the unit circle, a strip of  $\frac{2\pi}{7}$  maps to the disk.

he sampling theorem,  $X_{\delta}(\mathcal{P}) = \int_{-T}^{T} \sum_{k=-\infty}^{\infty} H\left(\frac{\omega - 2\pi k}{T}\right)$   $S_{1} = -1$   $(4.55) - m J_{3}$   $s_{1} = -1$ 3. The frequency response  $H(\omega)$  of the digital filter is related to the frequency response of the analog filter  ${}^3H_a(\Omega)$  by the sampling theorem,

$$h(\omega) = \frac{1}{T} \sum_{k=-\infty}^{\infty} H\left(\frac{\omega - 2\pi k}{T}\right)$$

minimal by choosing the sampling frequency  $\frac{1}{T}$  high enough such that the fraction of the energy in the range  $|\omega| > \frac{\pi}{T}$  will be negligible. On the otherhand, if  $H_a(s)$  is highpass or Since, for a rational analog filter,  $H(\Omega)$  is never band limited, the frequency response of the bandstop, the impulse invariant method cannot be used at all. This is because the frequency digital filter is always aliased. If  $H_a(s)$  is lowpass or bandpass, aliasing can be made very response of these two filter classes does not decay to zero, so the right-hand side of equation (4.55) does not converge.

Here,

of degree (N-1) in  $z^{-1}$ . This is regardless of the degree (q) of the numerator polynomial The zeros of H(z) and  $H_a(s)$  do not share a simple relationship. When the right-hand side of equation (4.53) is brought to a common denominator, the numerator will be a polynomial

Example 4.26 A third-order Butterworth lowpass filter has the transfer function

$$H(s) = \frac{1}{(s+1)(s^2+s+1)}$$

 $^3H_a(\Omega)$  was denoted earlier by  $H_a(j\Omega)$ .

$$H(s) = \frac{1}{(s+1)(s^2+s+1)}$$

$$= \frac{(s+1)(s+0.5-j0.866)(s+0.5+j0.866)}{C_2}$$

$$= \frac{C_1}{s+1} + \frac{C_2}{s+0.5-j0.866} + \frac{C_2^*}{s+0.5+j0.866}$$

Using partial fraction expansion, we find

$$C_1 = 1$$
,  $C_2 = 0.577 e^{-j2.62}$  and  $C_2^* = 0.577 e^{j2.62}$   
 $H(s) = \frac{1}{s+1} + \frac{0.577 e^{-j2.62}}{s+0.5 - j0.866} + \frac{0.577 e^{j2.62}}{s+0.5 + j0.866}$ 

free points are 
$$s_1 = -1$$
,  $s_2 = -0.5 + j0.866$  and  $s_3 = -0.5 - j0.866$ 

We know that  $H(z) = \sum_{i=1}^{3} \frac{C_i}{1 - e^{s_i T} z^{-1}}$ 

$$H(z) = \sum_{i=1}^{C} \frac{C_i}{1 - e^{s_1 T} z^{-1}}$$

$$= \frac{C_1}{1 - e^{s_1 T} z^{-1}} + \frac{C_2}{1 - e^{s_2 T} z^{-1}} + \frac{C_3}{1 - e^{s_3 T} z^{-1}}$$

$$C_3 = C_2^*$$

Hence,  $H(z) = \frac{C_1}{1 - e^{s_1}T_z^{-1}} + \frac{C_2}{1 - e^{s_2}T_z^{-1}} + \frac{C_2^*}{1 - e^{s_3}T_z^{-1}}$ 

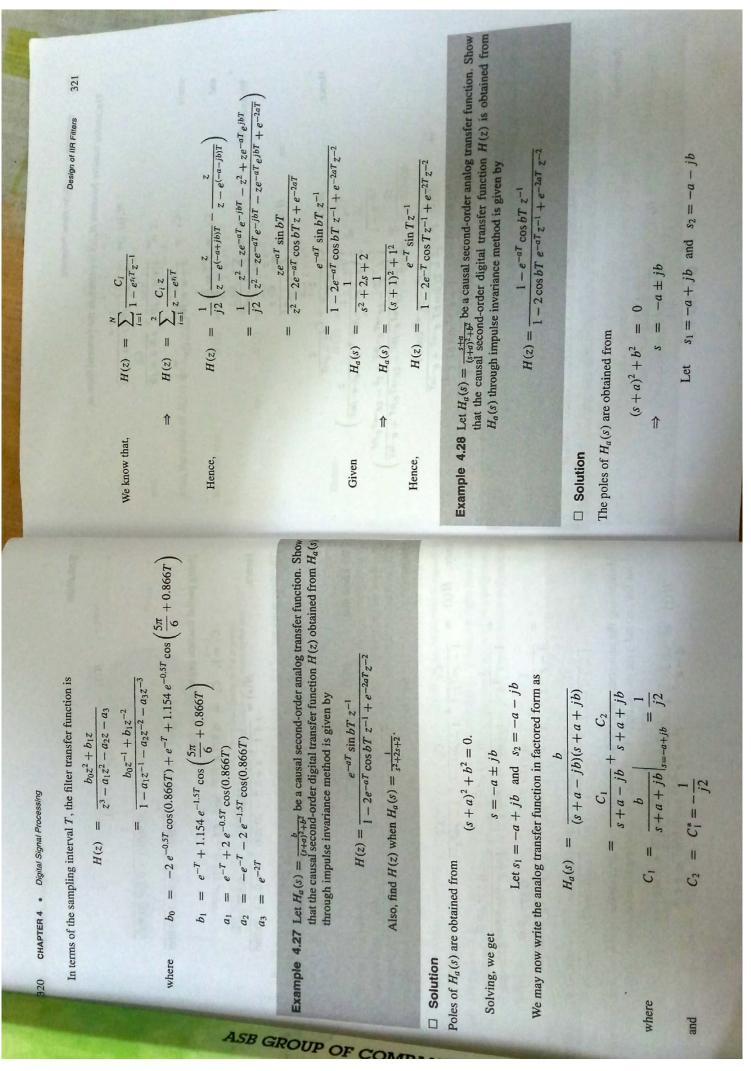
$$= \frac{1}{1 - e^{-T}z^{-1}} + \frac{0.577 e^{-j2.62}}{1 - e^{(-0.5 + j0.866)T}z^{-1}} + \frac{0.577 e^{j2.62}}{1 - e^{(-0.5 - j0.866)T}z^{-1}}$$

$$= \frac{1}{1 - e^{-T}z^{-1}} + \frac{0.577 e^{-j2.62}}{1 - e^{-0.5T}e^{j0.866T}z^{-1}} + \frac{0.577 e^{j2.62}}{1 - e^{-0.5T}e^{-j0.866T}z^{-1}}$$

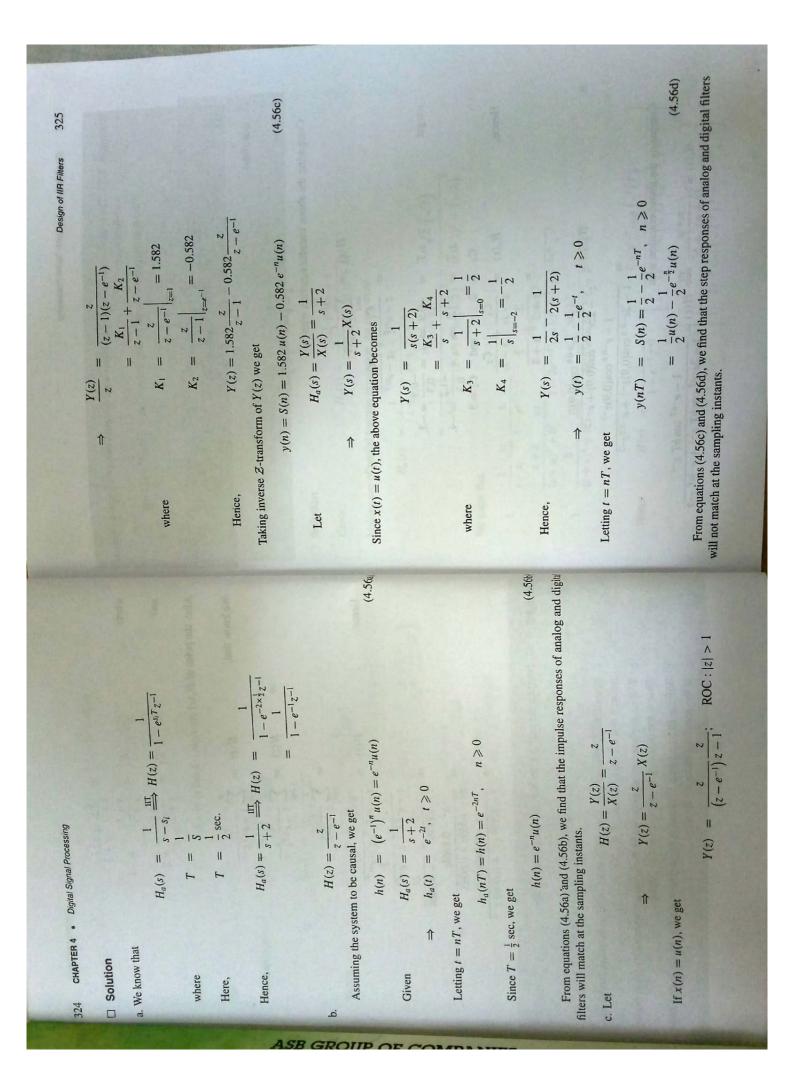
$$= \frac{1}{1 - e^{-T}z^{-1}} + \frac{2(0.577)\cos(-2.62) - 2(0.577)e^{-0.57}z^{-1}\cos(-2.62 - 0.866T)}{1 - 2 e^{-0.57}\cos(0.866T)z^{-1} + e^{-T}z^{-2}}$$

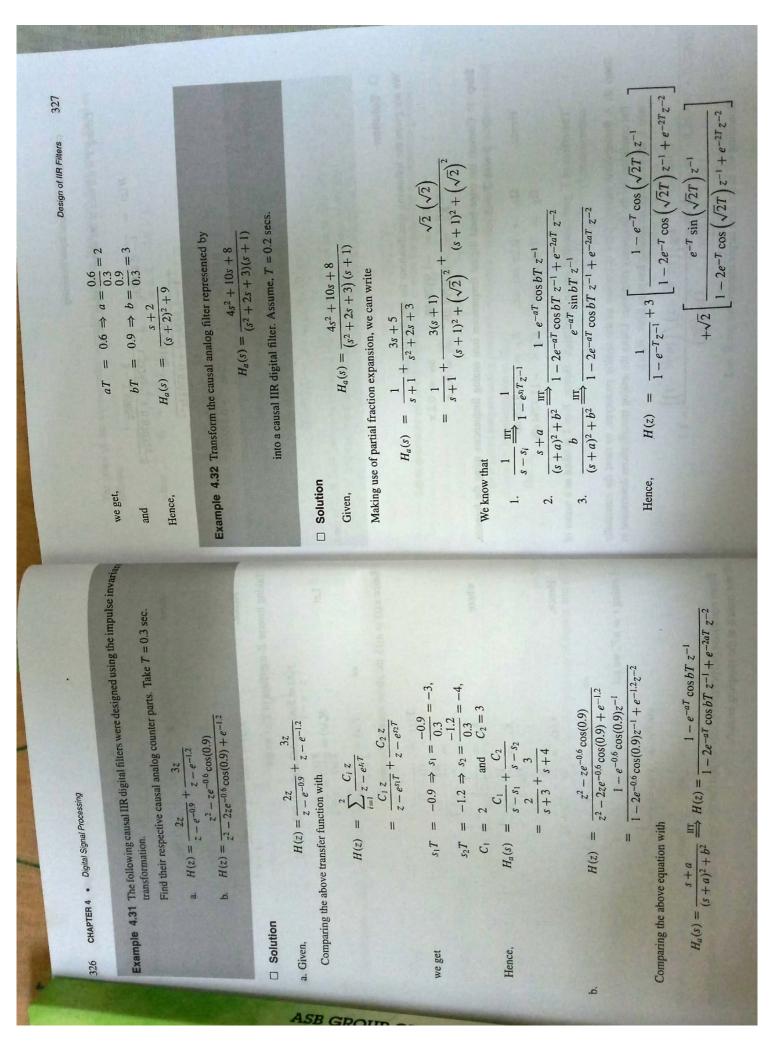
Multiplying the numerator and the denominator of first term on the right-hand side by z and by z² for the second term on the right- hand side, the above equation becomes

$$H(z) = \frac{z}{z - e^{-T}} + \frac{-z^2 - 1.154e^{-0.5T}\cos\left(\frac{3\pi}{6} + 0.866T\right)z}{z^2 - 2e^{-0.5T}\cos(0.866T)z + e^{-T}}$$



 $\frac{s+1}{(s+2)(s+3)}$ 





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Letting 
$$T = 0.2$$
 secs in the above equation, we get 
$$\frac{3 - 2.3585z^{-1}}{3 - 0.3585z^{-1}} + \frac{1}{1 - 0.51873z^{-1}} + \frac{1}{1 - 0.51873z^{-1}} + \frac{3 - 2.3585z^{-1}}{1 - 0.5134z^{-1}}$$

$$+\frac{0.32314z^{-1}}{1-1.5724z^{-1}+0.67032z^{-2}}$$

$$= \frac{1}{1 - 0.81873z^{-1}} + \frac{3 - 2.03536z^{-1}}{1 - 1.5724z^{-1} + 0.67032z^{-2}}$$

Example 4.33 A digital lowpass filter is required to meet the following specifications:

$$20 \log |H(\omega)|_{\omega=0.2\pi} \geqslant -1.9328 \, dB$$
  
 $20 \log |H(\omega)|_{\omega=0.6\pi} \leqslant -13.9794 \, dB$ 

The filter must have a maximally flat frequency response. Find H(z) to me. the above specifications using impulse invariant transformation.

Solution

We are given the following specifications for the digital filter:

$$K_P = -1.9328 \, \text{dB}, \qquad \omega_P = 0.2 \, \pi$$
  
 $K_S = -13.9794 \, \text{dB}, \quad \omega_S = 0.6 \, \pi$ 

Step 1: Convert the edge-band digital frequencies into analog frequencies using the formula  $\Omega = \frac{\omega}{T}$  with T = 1 sec.

Hence, 
$$\Omega_P = 0.2\pi \, \mathrm{rad/sec}$$
,  $K_P = -1.9328 \, \mathrm{dB}$   
 $\Omega_S = 0.6\pi \, \mathrm{rad/sec}$ ,  $K_S = -13.9794 \, \mathrm{dB}$ 

The effect of T gets cancelled out in the design. Hence, T=1 is taken as a matter of

Step 2: A Butterworth analog filter is chosen as the analog prototype, to meet the maximally flat condition for the frequency response. Using the analog specifications determined in step 1, let us design an analog lowpass filter,  $H_a(s)$ .

$$N = \frac{\log\left[\left(10^{-\frac{K_{p}}{10}} - 1\right) / \left(10^{-\frac{K_{s}}{10}} - 1\right)\right]}{2\log\left(\frac{\Omega_{p}}{\Omega_{s}}\right)} = 1.7$$

Rounding off to the next larger integer, we get N=2.

The poles of the second-order normalized lowpass Butterworth filter are as found as

$$s_k = 1 \frac{l\theta_k}{N}$$
 where 
$$\theta_k = \frac{\pi}{N} k + \frac{\pi}{2N} + \frac{\pi}{2}, \quad k = 0, 1, \dots 2N - 1$$

Hence, 
$$H_2(s) = \frac{1}{\prod_{\substack{\text{LHP} \\ \text{only}}}} = \frac{1}{(s-s_0)(s-s_1)}$$

$$= \frac{1}{(s+0.707-j0.707)(s+0.707+j0.707)}$$

$$= \frac{1}{(s+0.707)^2+(0.707)^2}$$

Let us determine the cutoff frequency  $\Omega_{\mathcal{C}}$  to meet the passband requirement precisely.

 $=\frac{s^2+1.414s+1}{s^2+1.414s+1}$ 

$$\Omega_C = \frac{\Omega_P}{\left[10^{\frac{-K_P}{10}} - 1\right]^{\frac{1}{2N}}} = 0.7255$$

The required lowpass analog protype is obtained by applying lowpass-to-lowpass analog frequency transformation to  $H_2(s)$ .

That is,

$$H_a(s) = H_2(s)|_{s \to \frac{s}{\alpha_C}}$$
  $\Rightarrow$   $H_a(s) = \frac{1}{s^2 + 1.414s + 1}|_{s \to \frac{s}{0.725s}}$ 

$$= \frac{\left(\frac{s}{0.7255}\right)^2 + 1.414\left(\frac{s}{0.7255}\right) + 1}{0.52635}$$

$$= \frac{s^2 + 1.02586s + 0.52635}{0.52635}$$

$$= \frac{s^2 + 1.02586s + 0.263097 + 0.52635}{0.52635}$$

 $(s + 0.5129298)^2 + (0.513082)^2$ 

 $1.0259 \times 0.513082$ 

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Step 3: Let us design H(z) using  $\Pi T$  with T=1 sec.

We know that,

$$\stackrel{b}{\underset{(s+a)^2+b^2}{\longrightarrow}} \xrightarrow{\text{IIT}} \stackrel{e^{-aT} \sin bTz^{-1}}{\underset{1-2e^{-aT}\cos bTz^{-1}+e^{-2aTz^{-2}}}{\longrightarrow}$$

ce, 
$$H(z) = 1.0259 \left[ \frac{e^{-0.5129298} \sin(0.513082)z^{-1}}{1 - 2e^{-0.5129298} \cos(0.513082)z^{-1} + e^{-2 \times 0.5129298} z^{-2}} \right]$$
  

$$= \frac{0.301512z^{-1}}{1 - 1.0433z^{-1} + 0.3585z^{-2}}$$

Verification

$$H(e^{j\omega}) = H(\omega) = \frac{0.301512e^{-j\omega}}{1 - 1.0433e^{-j\omega} + 0.3585e^{-j2\omega}}$$

$$\Rightarrow |H(\omega)| = \frac{0.301512\sqrt{\cos^2\omega + \sin^2\omega}}{\sqrt{(1 - 1.0433\cos\omega + 0.3585\cos2\omega)^2 + (1.0433\sin\omega - 0.3585\sin2\omega)^2}}$$

Therefore  $\sqrt{(1-1.0455\cos\omega+0.5385\cos\omega)^2 + (1.0455\sin\omega-0.556)^2}$  and  $20\log|H(\omega)|_{\omega=0.6\pi} = -2 \text{ dB}$ 

It may be noted that the passband specification is slightly exceeded and this is due to aliasing This will not be the case when H(z) is designed using bilinear transformation. If the resulting H(z) designed using IIT fails to meet the given specifications because of aliasing, there is no alternative with impulse invariance but to try again with a higher-order filter or with a different adjustment of the filter parameter, holding N fixed.

Example 4.34 Apply impulse invariant technique to the analog transfer function given by

$$H_a(s) = \frac{s^2 + 4.525}{s^2 + 0.692s + 0.504}$$

with T = 1 sec.

□ Solution

We can write the given analog transfer function as

$$H_a(s) = \frac{s^2 + 4.525 + 0.692s - 0.692s + 0.504 - 0.504}{s^2 + 0.692s + 0.504}$$
$$= \frac{(s^2 + 0.692s + 0.504) - 0.692s + 4.021}{s^2 + 0.692s + 0.504}$$

$$= 1 + \left[ \frac{-0.692(s+0.346) + 4.021 + 0.692 \times 0.346}{(s+0.346)^2} \right]$$

$$= 1 - \frac{0.692(s+0.346)^2 + 0.(6199)^2}{(s+0.346)^2 + (0.6199)^2} + \frac{4.26}{(s+0.346)^2 + (0.6199)^2}$$

$$= 1 - \frac{0.692(s+0.346)}{(s+0.346)^2 + (0.6199)^2} + \frac{0.692(s+0.346)}{(s+0.346)^2 + (0.6199)^2}$$

We know that

$$\frac{s+a}{(s+a)^2+b^2} \xrightarrow{\text{III}} \frac{1-e^{-aT}\cos bT \ z^{-1}}{1-2e^{-aT}\cos bT \ z^{-1}}$$

$$\frac{b}{(s+a)^2+b^2} \xrightarrow{\text{ITT}} \frac{e^{-aT} \sin bT z^{-1}}{1-2e^{-aT}\cos bTz^{-1} + e^{-2aT}z^{-2}}$$

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(please note that  $\mathcal{L}\{\delta(t)\}=1$  and  $\mathcal{Z}\{\delta(n)\}=1$ )

Hence, 
$$H(z) = 1 - \frac{0.692 \left[1 - e^{-0.346} \cos(0.6199)z^{-1}\right]}{1 - 2 e^{-0.346} \cos(0.6199)z^{-1} + e^{-0.692}z^{-2}} + \frac{6.872 e^{-0.346} \sin(0.6199)z^{-1} + e^{-0.692}z^{-2}}{1 - 2 e^{-0.346} \cos(0.6199)z^{-1} + e^{-0.692}z^{-2}} = 1 - \left[\frac{0.692 - 3.2234z^{-1}}{1 - 1.1517z^{-1} + 0.5006z^{-2}}\right]$$

Example 4.35 Design a digital Chebyshev I filter that satisfies the following constraints.

$$0.8 \le |H(\omega)| \le 1, \quad 0 \le \omega \le 0.2\pi$$
  
 $|H(\omega)| \le 0.2, \quad 0.6\pi \le \omega \le \pi$ 

Use impluse invariant transformation.

□ Solution

We are given the following digital specifications:

Passband ripple:  $\delta_P = 1 - 0.8 = 0.2$ . Passband-edge frequency:  $\omega_P = 0.2\pi$ .

Stopband tolerance:  $\delta_S = 0.2$ .

Stopband-edge frequency:  $\omega_S = 0.6\pi$ .

