Standard approach

- (1) Convert the digital filter specifications into an analogue prototype lowpass filter specifications
- (2) Determine the analogue lowpass filter transfer function $H_a(s)$
- (3) Transform $H_a(s)$ by replacing the complex variable to the digital transfer function

• This approach has been widely used for the following reasons:

(1) Analogue approximation techniques are highly advanced

(2) They usually yield closed-form solutions

(3) Extensive tables are available for analogue filter design

(4) Very often applications require digital simulation of analogue systems

• Let an analogue transfer function be

$$H_a(s) = \frac{P_a(s)}{D_a(s)}$$

- where the subscript "a" indicates the analogue domain
- A digital transfer function derived from this is denoted as

$$G(z) = \frac{P(z)}{D(z)}$$

- Basic idea behind the conversion of $H_a(s)$ into G(z) is to apply a mapping from the *s*-domain to the *z*-domain so that essential properties of the analogue frequency response are preserved
- Thus mapping function should be such that
 - Imaginary $(j\Omega)$ axis in the *s*-plane be mapped onto the unit circle of the *z*-plane
 - A stable analogue transfer function be mapped into a stable digital transfer function

IIR Digital Filter: The bilinear transformation

- To obtain G(z) replace s by f(z) in H(s)
- Start with requirements on G(z)

<u>G(z)</u>	<u>Available H(s)</u>
Stable	Stable
Real and Rational in z	Real and Rational in <i>s</i>
Order n	Order n
L.P. (lowpass) cutoff $\mathbf{\Omega}_c$	L.P. cutoff $\omega_c T$

IIR Digital Filter

• Hence f(z) is real and rational in z of order <u>one</u>

• i.e.
$$f(z) = \frac{az+b}{cz+d}$$

• For LP to LP transformation we require

$$s = 0 \rightarrow z = 1 \quad f(1) = 0 \rightarrow a + b = 0$$

$$s = \pm j \infty \rightarrow z = -1 \quad f(-1) = \pm j \infty \rightarrow c - d = 0$$

Thus
$$(a) = z = 1$$

$$f(z) = \left(\frac{a}{c}\right) \cdot \frac{z-1}{z+1}$$

• The quantity $\left(\frac{a}{c}\right)$ is fixed from $\omega_c T \leftrightarrow \Omega_c$

• ie on C:|z|=1 $f(z)|_c = \left(\frac{a}{c}\right) \cdot j \tan \frac{\omega T}{2}$

Or

$$j\Omega_{c} = \left(\frac{a}{c}, j \tan \frac{\omega_{c}T}{2}\right)$$
and

$$s = \left(\frac{\Omega_{c}}{\tan\left(\frac{\omega_{c}T}{2}\right)}\right) \cdot \frac{1-z^{-1}}{1+z^{-1}}$$

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 Transformation is unaffected by scaling. Consider inverse transformation with scale factor equal to unity



• Mapping of *s*-plane into the *z*-plane



• For $z = e^{j\omega}$ with unity scalar we have

$$j\Omega = \frac{1 - e^{-j\omega}}{1 + e^{-j\omega}} = j\tan(\omega/2)$$

or $\Omega = \tan(\omega/2)$



- Mapping is highly nonlinear
- Complete negative imaginary axis in the *s*plane from $\Omega = -\infty$ to $\Omega = 0$ is mapped into the lower half of the unit circle in the *z*-plane from z = -1 to z = 1
- Complete positive imaginary axis in the *s*plane from $\Omega = 0$ to $\Omega = \infty$ is mapped into the upper half of the unit circle in the *z*-plane from z = 1 to z = -1

Spectral Transformations

- To transform $G_L(z)$ a given lowpass transfer function to another transfer function $G_D(\hat{z})$ that may be a lowpass, highpass, bandpass or bandstop filter (solutions given by Constantinides)
- z^{-1} has been used to denote the unit delay in the prototype lowpass filter $G_L(z)$ and \hat{z}^{-1} to denote the unit delay in the transformed filter $G_D(\hat{z})$ to avoid confusion

Spectral Transformations

• Unit circles in z- and \hat{z} -planes defined by

$$z = e^{j\omega_{\rm l}} \qquad \hat{z} = e^{j\hat{\omega}}$$

- Transformation from z-domain to
 - \hat{z} -domain given by

$$z = F(\hat{z})$$

• Then

$$G_D(\hat{z}) = G_L\{F(\hat{z})\}$$