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>	Available H(s)
Stable	Stable
Real and Rational in z	Real and Rational in <i>s</i>
Order <i>n</i>	Order <i>n</i>
L.P. (lowpass) cutoff Ω_c	L.P. cutoff $\omega_c T$

IIR Digital Filter

- Hence f(z) is real and rational in z of order one
- i.e. $f(z) = \frac{az+b}{cz+d}$
- For LP to LP transformation we require

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

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

IIR Digital Filter

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

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

Or

$$j\Omega_c = \left(\frac{a}{c}\right).j\tan\frac{\omega_c T}{2}$$

and

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

 Transformation is unaffected by scaling. Consider inverse transformation with scale factor equal to unity $z = \frac{1+s}{1-s}$

$$s = \sigma_o + j\Omega_o$$

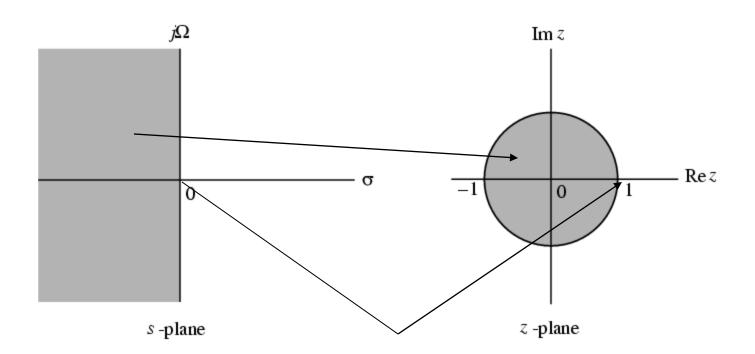
$$z = \frac{(1+\sigma_o)+j\Omega_o}{(1-\sigma_o)-j\Omega_o} \Rightarrow |z|^2 = \frac{(1+\sigma_o)^2+\Omega_o^2}{(1-\sigma_o)^2+\Omega_o^2}$$
• and so

$$\sigma_o = 0 \rightarrow |z| = 1$$

$$\sigma_o < 0 \rightarrow |z| < 1$$

$$\sigma_o > 0 \rightarrow |z| > 1$$

Mapping of s-plane into the z-plane

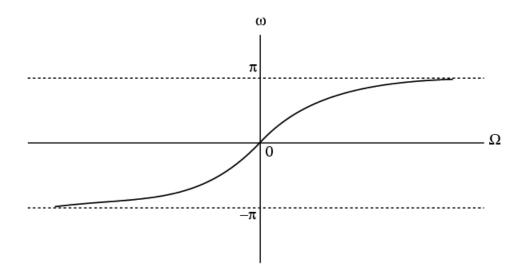


• For $z = e^{j\omega}$ with unity scalar we have

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

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

$$z = e^{j\omega}$$
 $\hat{z} = e^{j\hat{\omega}}$

Transformation from z-domain to

 \hat{z} -domain given by

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

Then

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