

State University of New York

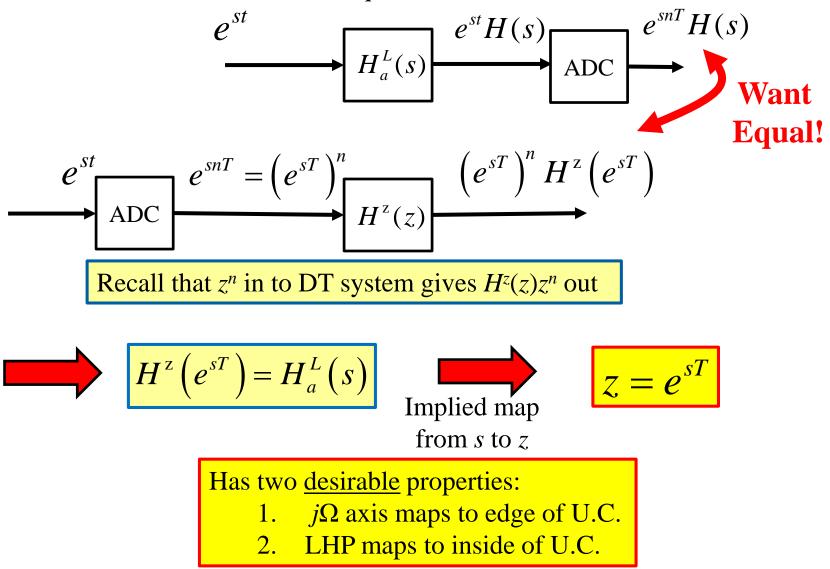
EEO 401 Digital Signal Processing Prof. Mark Fowler

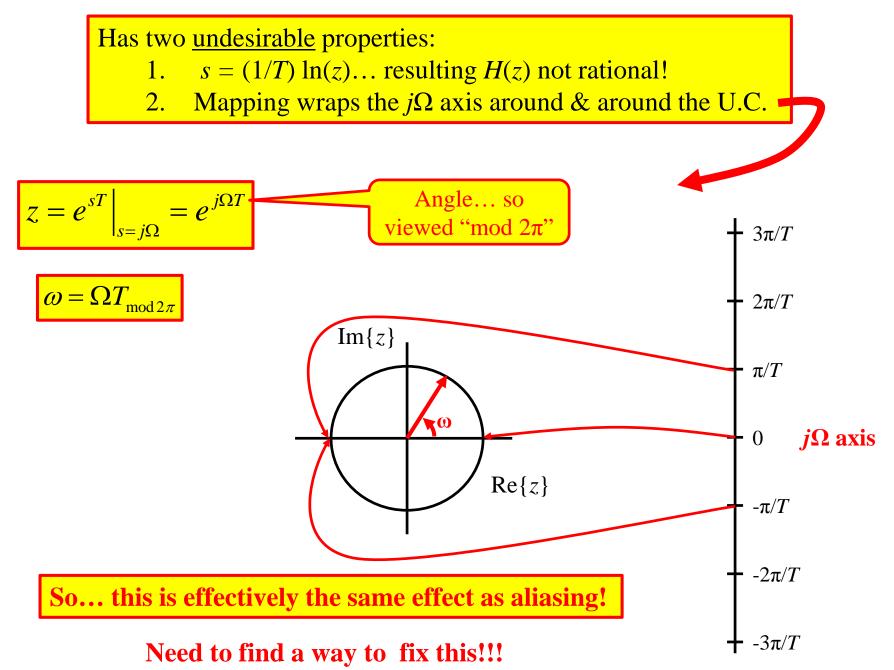
<u>Note Set #33</u>

- IIR Design Bilinear Transform
- Reading: Sect. 10.3.3 of Proakis & Manolakis

Motivation (different from the P&M textbook)

If the CT and DT systems are equivalent from a transfer function point of view then we should have this kind of equivalence:





Instead of using $z = e^{sT}$... $s = (1/T) \ln(z)$... as the mapping we look for some other mapping that:

- Still maps to the U.C. but without wrapping!!
- And ensures that the resulting H(z) is rational!!

We start with a known expression for the hyperbolic tangent!

$$\tanh\left(\frac{sT}{2}\right) = \frac{e^{sT/2} - e^{-sT/2}}{e^{sT/2} + e^{-sT/2}}$$

Bilinear

And its power series : $\tanh\left(\frac{sT}{2}\right) = \frac{sT}{2} - \frac{1}{3}\left(\frac{sT}{2}\right)^3 + \frac{2}{15}\left(\frac{sT}{2}\right)^3 \cdots$

Equating these two and ignoring these terms (negligible for small T)

$$s \approx \left(\frac{2}{T}\right) \frac{e^{sT/2} - e^{-sT/2}}{e^{sT/2} + e^{-sT/2}} = \left(\frac{2}{T}\right) \frac{e^{sT} - 1}{e^{sT} + 1}$$

$$S \approx \left(\frac{2}{T}\right) \frac{e^{sT/2} - e^{-sT/2}}{e^{sT} + 1} = \left(\frac{2}{T}\right) \frac{e^{sT} - 1}{e^{sT} + 1}$$

$$S \approx \left(\frac{2}{T}\right) \frac{z - 1}{z + 1}$$

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

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

$$S \approx \left(\frac{2}{T}\right) \frac{z - 1}{z + 1}$$

Bilinear Transform

$$s = \left(\frac{2}{T}\right) \left(\frac{z-1}{z+1}\right) \dots OR \dots s = \left(\frac{2}{T}\right) \left(\frac{1-z^{-1}}{1+z^{-1}}\right) \longrightarrow H^{f}(z) = H^{L}_{a}\left(\frac{2z-1}{Tz+1}\right)$$

Investigate characteristics using $z = re^{j\omega}$ & $s = \sigma + j\Omega$

$$s = \left(\frac{2}{T}\right)\frac{z-1}{z+1} = \left(\frac{2}{T}\right)\frac{re^{j\omega}-1}{re^{j\omega}+1}$$
$$= \left(\frac{2}{T}\right)\frac{r^2-1}{1+r^2+2r\cos(\omega)} + j\left(\frac{2}{T}\right)\frac{2r\sin(\omega)}{1+r^2+2r\cos(\omega)}$$
$$= \sigma$$
$$= \sigma$$
Note: $r \stackrel{\leq}{=} 1 \implies \sigma \stackrel{\leq}{=} 0 \implies \dots \text{ stability is preserved!!}$
$$\dots \text{ stability is preserved!!}$$
$$\dots \text{ stability is preserved!!}$$

Need to see if we've fixed the "wrapped around & around" problem...

Bilinear Transform: Mapping between CT & DT Frequencies

Setting r = 1 gives $\sigma = 0$ so that is the condition for mapping between CT & DT frequencies:

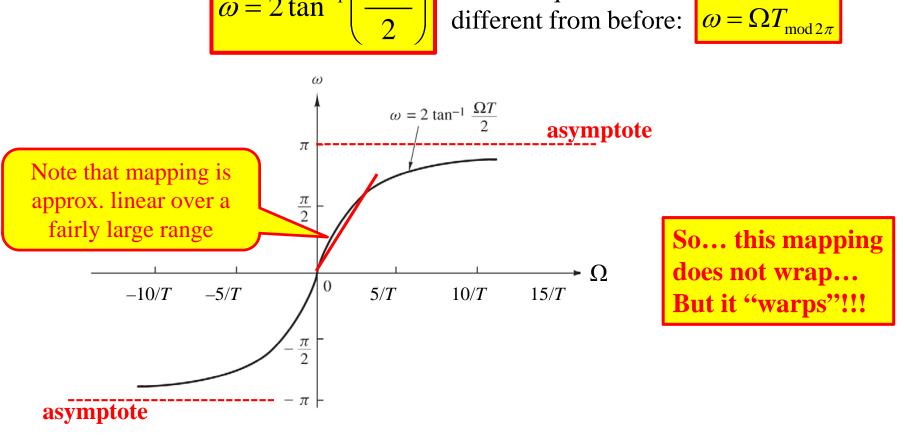
 $\Omega 7$

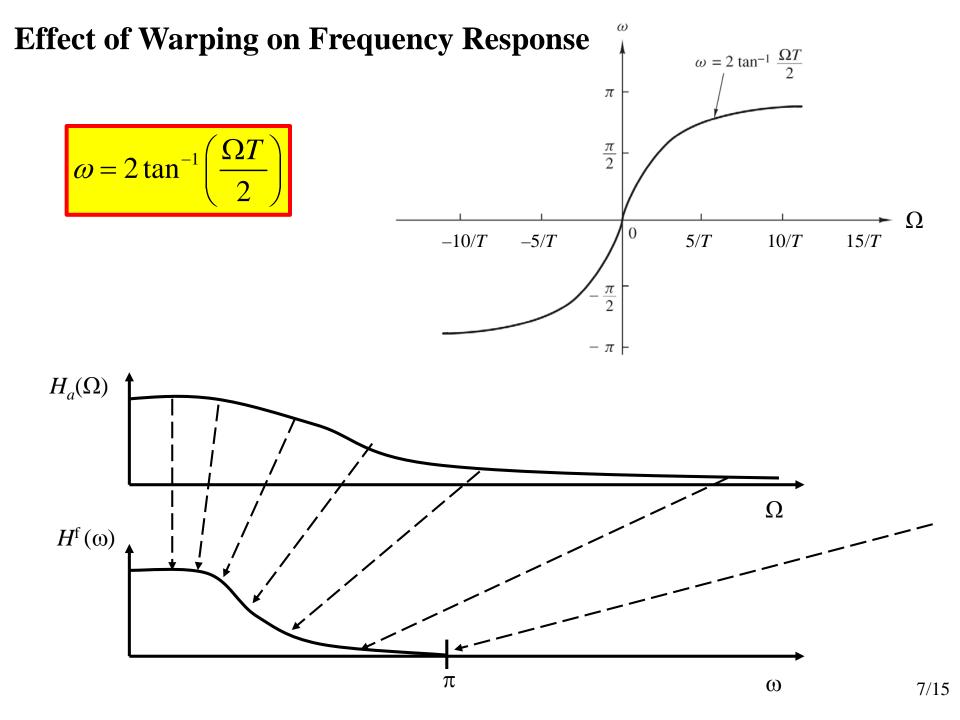
 $\omega = 2 \tan^{-1}$

$$\Omega = \left(\frac{2}{T}\right) \frac{2r\sin(\omega)}{1 + r^2 + 2r\cos(\omega)} \bigg|_{r=1} = \left(\frac{2}{T}\right) \frac{\sin(\omega)}{1 + \cos(\omega)} = \left(\frac{2}{T}\right) \tan\left(\frac{\omega}{2}\right)$$

which is quite

Or...





Matlab Command for Bilinear Transform

[NUMd,DENd] = bilinear(NUM,DEN,Fs),

$$F_s = 1/T$$

where NUM and DEN are row vectors containing numerator and denominator transfer function coefficients, NUM(s)/DEN(s), in descending powers of s, transforms to z-transform coefficients NUMd(z)/DENd(z).

Choosing Value of T

Two different approaches:

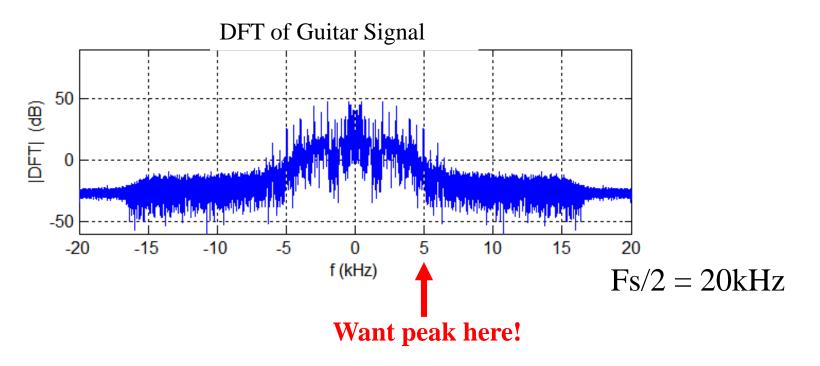
- 1. Pick *T* for convenience in the filter design
 - This requires picking a different sampling rate for the signal that matches its bandwidth
 - This approach requires determining frequency cutoffs in the digital domain and mapping them to some CT filter
- 2. Pick *T* such that $F_s = 1/T$ is appropriate for the signal sampling
 - Then deal with its impact in the filter design
 - This approach requires an analog filter design with cutoff frequencies spec'd according to the signal's CT requirements

Example of Choosing *T* via Method #1

Based on Example 10.3.4 in textbook

- Pick *T* for convenience in the filter design
 - This requires picking a different sampling rate for the signal that matches its bandwidth
 - This approach requires determining frequency cutoffs in the digital domain and mapping them to some CT filter
- Suppose we have a signal that is appropriately sampled at some F_s .
- With that sampling... suppose we determine we want an IIR filter with a resonant peak at DT frequency $\omega = \pi/4$ (which corresponds to $CT f = F_s/8$)

1.

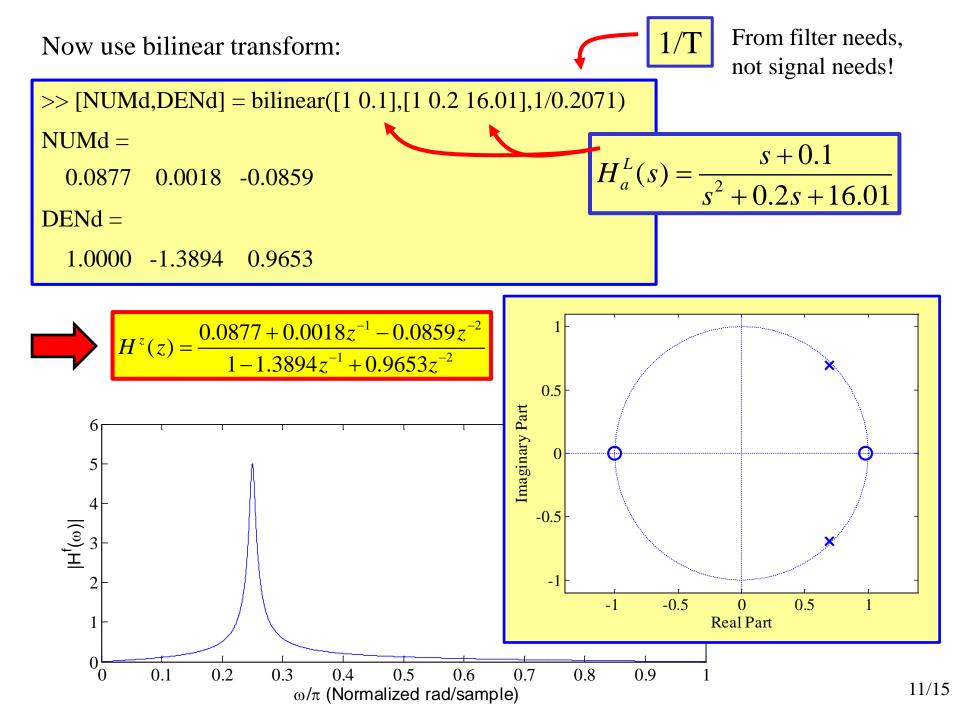


Now we pick an analog filter with resonance... and it does not have to be designed to have a peak at 5kHz!!! We pick the *T* as needed to map THIS filter's peak frequency to DT frequency of $\pi/4$ rad/sample

$$H_{a}^{L}(s) = \frac{s+0.1}{(s+0.1)^{2}+16} = \frac{s+0.1}{s^{2}+0.2s+16.01} = \frac{s+0.1}{(s+0.1+j4)(s+0.1-j4)}$$

This has a resonant peak at $\Omega = 4$ rad/sec Far cry from the "real" $2\pi \times 5000$
>> w=0:0.001:16; H=freqs([1 0.1],[1 0.2 16.01],w); plot(w,abs(H))
 $= \frac{1}{2} + \frac{$

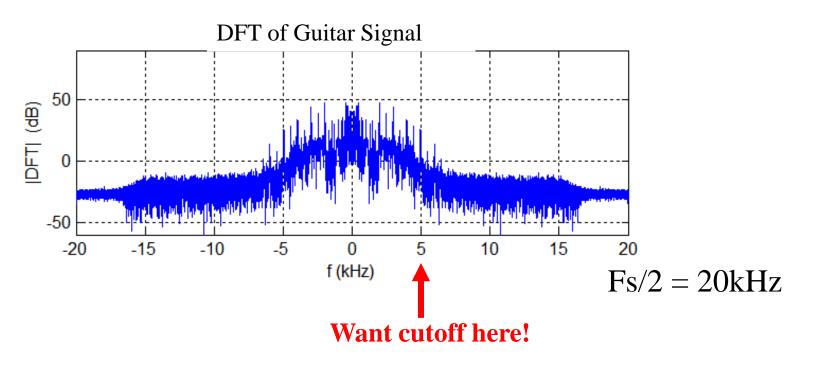
Satisfactory location of "Folding Freq" 10/15



Example of Choosing *T* via Method #2².

- Pick T such that Fs = 1/T is appropriate for the signal sampling
 - Then deal with impact in the filter design
 - This approach requires an analog filter design with cutoff frequencies spec'd according to the signal's CT requirements
- Suppose we have a signal like that shown below and we want to lowpass filter it with a cutoff of 5kHz
- Fs = 40,000 is an appropriate sampling rate

$$T = 1/F_s = 1/40,000 = 2.5 \times 10^{-5}$$



Suppose we choose as our analog prototype a first-order LPF:



However, we can <u>**not**</u> use $\Omega_c = 2\pi \times 5000 = 31,415.93$ rad/sec because of the warping of the bilinear transform!!

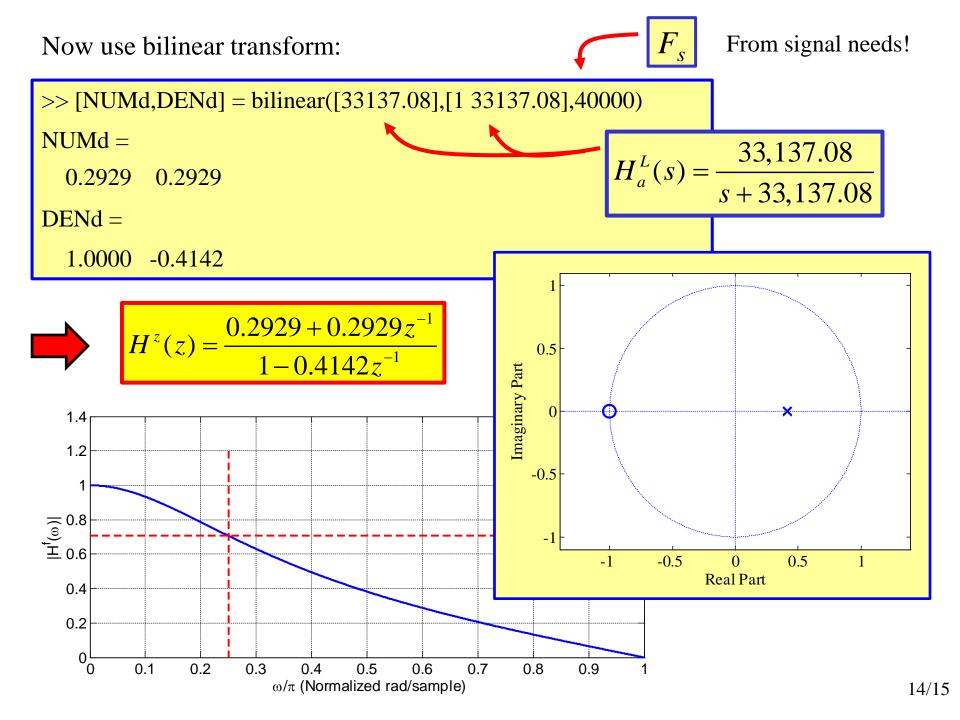
Signal frequencies get mapped linearly according to F_s :

5000 Hz \rightarrow 5000 × $(2\pi/F_s)$ = 5000 × $(2\pi/40000)$ = $\pi/4$ rad/sample

But... CT filter frequencies get mapped by the bilinear's warp: $\omega = 2 \tan^{-1} \left(\frac{\Omega T}{2} \right)$

So we "pre-warp" the 5000 Hz requirement using the inverse of this -

 $\Omega_c = 33,137.08 \text{ rad/sec}$ "Pre-warped" higher than this



<u>Using "pre-warp" feature of bilinear.m command</u> [NUMd,DENd] = bilinear(NUM,DEN,Fs,Fp),

applies prewarping before the bilinear transformation so that the frequency responses before and after mapping match exactly at frequency point Fp (match point Fp is **specified in Hz**).

where NUM and DEN are row vectors containing numerator and denominator transfer function coefficients, NUM(s)/DEN(s), in descending powers of s, transforms to z-transform coefficients NUMd(z)/DENd(z).

