# ECE 376 - Homework \#12 

Digital Filters - Due Monday, April 22nd

## Filters in the z-Plane

1) Assume $\mathrm{G}(\mathrm{s})$ is a low-pass filter with real poles:

$$
G(s)=\left(\frac{500}{(s+3)(s+7)(s+10)}\right)
$$

Design a digital filter, $\mathrm{G}(\mathrm{z})$, which has approximately the same gain vs. frequency as $\mathrm{G}(\mathrm{s})$. Assume a sampling rate of $\mathrm{T}=0.01$ second.

Convert to the z-plane

$$
\begin{array}{ll}
s=-3 & z=e^{s T}=e^{-0.03}=0.9704 \\
s=-7 & z=e^{s T}=e^{-0.07}=0.9324 \\
s=-10 & z=e^{s T}=e^{-0.1}=0.9048
\end{array}
$$

so the form of $G(z)$ is

$$
G(z)=\left(\frac{k}{(z-0.9704)(z-0.9324)(z-0.9048)}\right)
$$

To find k , match the DC gain

$$
\begin{aligned}
& D C=\left(\frac{500}{(s+3)(s+7)(s+10)}\right)_{s=0}=2.3810 \\
& D C=\left(\frac{k}{(z-0.9704)(z-0.9324)(z-0.9048)}\right)_{z=1}=2.3810 \\
& k=0.00045355
\end{aligned}
$$

so

$$
G(z)=\left(\frac{0.00045355}{(z-0.9704)(z-0.9324)(z-0.9048)}\right)
$$

You could also add three zeros at $\mathrm{z}=0$ to remove the time-delay (optional - delays don't affect the gain vs. frequency)

$$
G(z)=\left(\frac{0.00045355 z^{3}}{(z-0.9704)(z-0.9324)(z-0.9048)}\right)
$$

Plot the gain vs. frequency for both filters from 0 to $50 \mathrm{rad} / \mathrm{sec}$.

```
>> w = [0:0.01:50]';
>> s = j*w;
> Gs = 500./ ( (s+3).*(s+7).* (s+10) );
>>
>> T = 0.01;
>> z = exp(s*T);
>>Gz = 0.00045355 ./ ( (z-0.9704).*(z-0.9324).*(z-0.9048));
>>
>> plot(w,abs(Gs),'b',w,abs(Gz),'r')
>> xlabel('Frequency (rad/sec) ');
>> ylabel('Gain')
>>
```


2) Assume $G(s)$ is the following band-pass filter:

$$
G(s)=\left(\frac{10 s}{(s+5+j 20)(s+5-j 20)}\right)
$$

Design a digital filter, $\mathrm{G}(\mathrm{z})$, which has approximately the same gain vs. frequency as $\mathrm{G}(\mathrm{s})$. Assume a sampling rate of $\mathrm{T}=0.01$ second.

Same procedure - works for real poles, works for complex poles

$$
\begin{array}{ll}
s=0 & z=e^{s T}=e^{0}=1 \\
s=-5+j 20 & z=e^{s T}=0.9323+0.1890 \mathrm{i} \\
s=-5-j 20 & z=e^{s T}=0.9323-0.1890 \mathrm{i}
\end{array}
$$

so the form of $\mathrm{G}(\mathrm{z})$ is

$$
G(z)=\left(\frac{k(z-1)}{(z-0.9323+j 0.1890)(z-0.9323-j 0.1890)}\right)
$$

Pick k to match the gain at some frequency. DC doesn't work since the gain is zero. Pick some other frequency, like $20 \mathrm{rad} / \mathrm{sec}$

$$
\begin{aligned}
& \left(\frac{10 s}{(s+5+j 20)(s+5-j 20)}\right)_{s=j 20}=0.9923 \angle 7.1250^{0} \\
& \left(\frac{k(z-1)}{(z-0.9323+j 0.1890)(z-0.9323-j 0.1890)}\right)_{z=e^{j 0.2}}=10.4817 \angle 1.2997^{0} \\
& k=\left(\frac{0.9923}{10.4817}\right)=0.0947
\end{aligned}
$$

so

$$
G(z)=\left(\frac{k(z-1)}{(z-0.9323+j 0.1890)(z-0.9323-j 0.1890)}\right)
$$

Plot the gain vs. frequency for both filters from 0 to $50 \mathrm{rad} / \mathrm{sec}$.

```
>> w = [0:0.01:50]';
>> s = j*W;
> Gs = 10*s./ ( (s+5+j*20).*(s+5-j*20) );
>> T = 0.01;
>> p1 = exp( (-5+j*20)*T );
>> p2 = exp( (-5-j*20)*T );
>> z = exp(s*T);
>> Gz = 0.0947*(z-1) ./ ( (z-p1).*(z-p2) );
>> plot(w,abs(Gs),'b',w,abs(Gz),'r')
>> xlabel('Frequency (rad/sec) ');
>> ylabel('Gain')
>> ylim([0,1.2])
```


3) Write a $C$ program to implement the digital filter, $G(z)$

$$
Y=\left(\frac{0.0947(z-1)}{(z-0.5403+j 0.8415)(z-0.5403-j 0.8415)}\right) X
$$

multiply out

$$
Y=\left(\frac{0.0947(z-1)}{z^{2}-1.8645 z+0.9048}\right) X
$$

cross multiply

$$
\left(z^{2}-1.8645 z+0.9048\right) Y=0.0947(z-1) X
$$

meaning

$$
y(k+2)-1.8645 y(k+1)+0.9048 y(k)=0.0947(x(k+1)-x(k))
$$

time shift

$$
\mathrm{y}(\mathrm{k})-1.8645 \mathrm{y}(\mathrm{k}-1)+0.9048 \mathrm{y}(\mathrm{k}-2)=0.0947(\mathrm{x}(\mathrm{k}-1)-\mathrm{x}(\mathrm{k}-2))
$$

Solve for $y(k)$

$$
y(k)=1.8645 y(k-1)-0.9048 y(k-2)+0.0947(x(k-1)-x(k-2))
$$

That's your program

```
while(1) {
    x2 = x1;
    x1 = x0;
    x0 = A2D_Read(0);
    y2 = y1;
    y1 = y0;
    y0 = 1.8645*y1 - 0.9048*y2 + 0.0947 * ( x1 - x2 );
    D2A(y0);
    wait_ms(10);
    }
```


## FIR Filters

4) Find the impulse response of a filter with the following gain vs. frequency:

- hint: Approximate the waveform by adding up ideal low-pass filters


$$
\mathrm{G}(\mathrm{~s})=0.2 * \mathrm{LPF}(6 \mathrm{rad} / \mathrm{sec})+0.4 * \mathrm{LPF}(4 \mathrm{rad} / \mathrm{sec})+0.4 * \mathrm{LPF}(2 \mathrm{rad} / \mathrm{sec})
$$

$$
H(t)=0.2\left(\frac{\sin (6 t)}{t}\right)+0.4\left(\frac{\sin (4 t)}{t}\right)+0.4\left(\frac{\sin (2 t)}{t}\right)
$$



Impulse Response of Filter
5) Design a FIR filter to approximate this impulse reaponse. Include in your design

The sampling rate

- 100 points for 10 seconds
- $\mathrm{T}=0.1$ second

The length of the window

- clip the signal from -5 s to +5 s
- delay the signal by 5 seconds

The impulse response of your FIR fitler.

```
>> t = [-5:0.1:5]';
>> t = [-5:0.1:5]' + 1e-9;
>> H = 0.2*sin(6*t)./t + 0.4*sin(4*t)./t + 0.4*sin(2*t)./t;
>> H = H / sum(H);
>> DC = sum(H)
DC = 1.0000
>> plot(t,H,'.')
>> plot(t+5,H,'.')
>> xlim([0,10])
>> xlabel('Time (seconds)')
```



Note: The DC gain is the sum of all terms.

- (the sum should be 1.000 for a DC gain of 1.000 )

6) Plot the gain vs. frequency of your filter
```
>> w = [0:0.01:10]';
>> s = j*w;
>> T = 0.1;
>> z = exp(s*T);
>> Gz = 0*S;
>> for i=1:length(t)
    Gz = Gz + H(i) * z.^(-i);
    end
>> plot(w, abs(Gz))
```



It's not perfect, but pretty good. Adding more terms would improve the frequency response

