

## Lecture 24 (Multirate DSP)

Last class:

- Main types of IIR filters ( $B, C1, C2, E$ )

FIR vs IIR ?

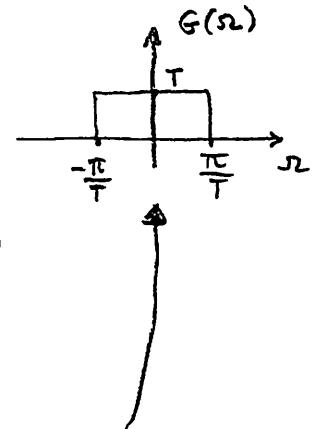
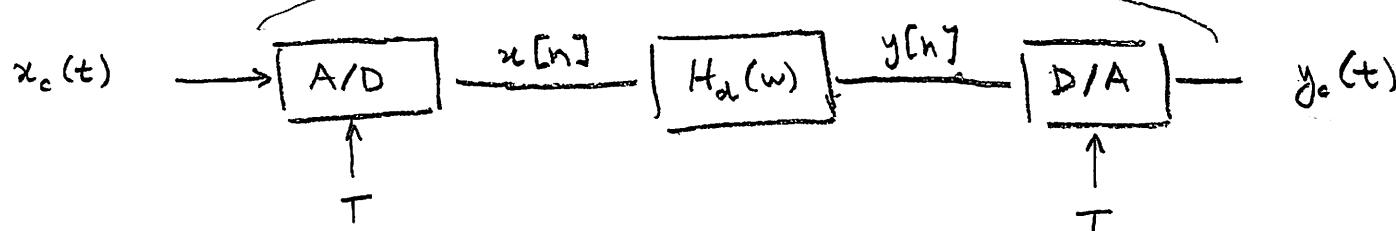
Advantages of FIR :

- Right choice if GLP is required. Only FIR has exact GLP
- Easy to design by hand with window-based approach

Disadvantages of FIR (via windowing)

- Same errors (ripples) in passband and stopband
- Longer filter lengths than IIR

• DT processing of CT signals:  $H_c^{(\text{eff})}(\omega)$



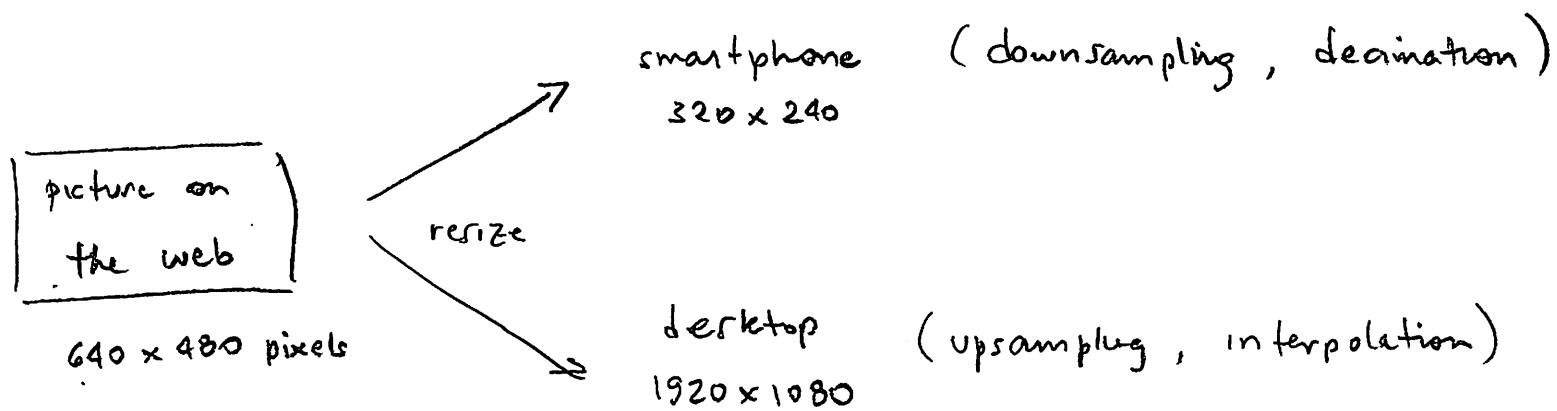
Main result: If  $x_c(t)$  is bandlimited,  $\frac{1}{T}$  is above Nyquist rate, ideal D/A:

then the effective analog system is  $H_c^{(\text{eff})}(\omega) = H_d(\omega T)$

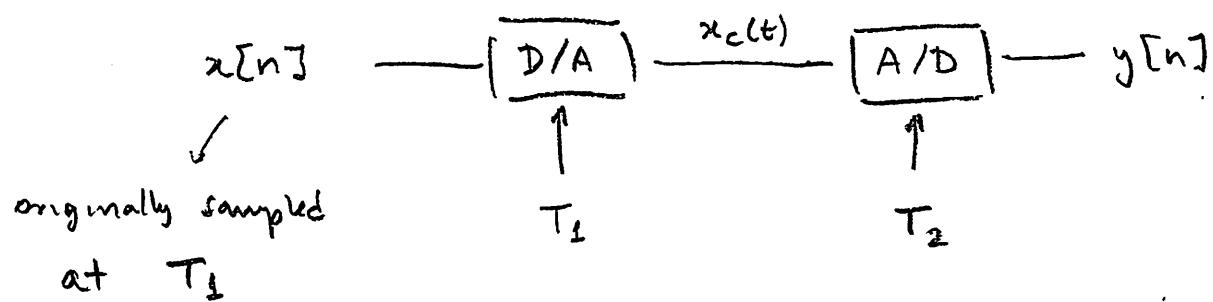
Notice that the whole system operates in the same sampling rate

Some applications may need to convert sampling rate  
from  $T_1$  to  $T_2$

- Example of multirate DSP:

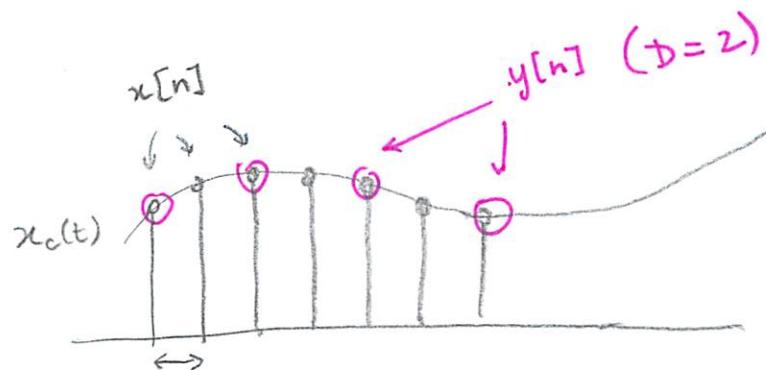


Using tools we've learned, sampling rate conversion could be implemented as follows:



We will look at sample rate conversion while staying in the digital domain

Downsampling (by an integer)



$$x[n] = x_c(nT)$$

$$\text{convert to } T_2 = DT$$

$$y[n] = x_c(nDT) = x[DN]$$

In frequency domain?

$$X_d(\omega) = \frac{1}{T} \sum_{n=-\infty}^{\infty} X_c\left(\frac{\omega + 2\pi n}{T}\right)$$

$$Y_d(\omega) = \frac{1}{DT} \sum_{m=-\infty}^{\infty} X_c\left(\frac{\omega + 2\pi m}{DT}\right) = \frac{1}{DT} \sum_m X_c\left(\frac{\frac{\omega}{D} + \frac{2\pi m}{D}}{T}\right)$$

$m = D \cdot k + l$   
where  $-\infty \leq k \leq \infty$   
and  $0 \leq l \leq D-1$

$$= \frac{1}{DT} \sum_{k=-\infty}^{\infty} \sum_{l=0}^{D-1} X_c\left(\frac{\frac{\omega}{D} + 2\pi k + \frac{2\pi l}{D}}{T}\right) = \frac{1}{D} \sum_{l=0}^{D-1} \frac{1}{T} \sum_{k=-\infty}^{\infty} X_c\left(\frac{\frac{\omega}{D} + 2\pi l}{T} + 2\pi k\right)$$

$$\Rightarrow Y_d(\omega) = \frac{1}{D} \sum_{l=0}^{D-1} X_d\left(\frac{\omega + 2\pi l}{D}\right)$$

$$X_d\left(\frac{\omega + 2\pi l}{D}\right)$$

Notation (downsampler)

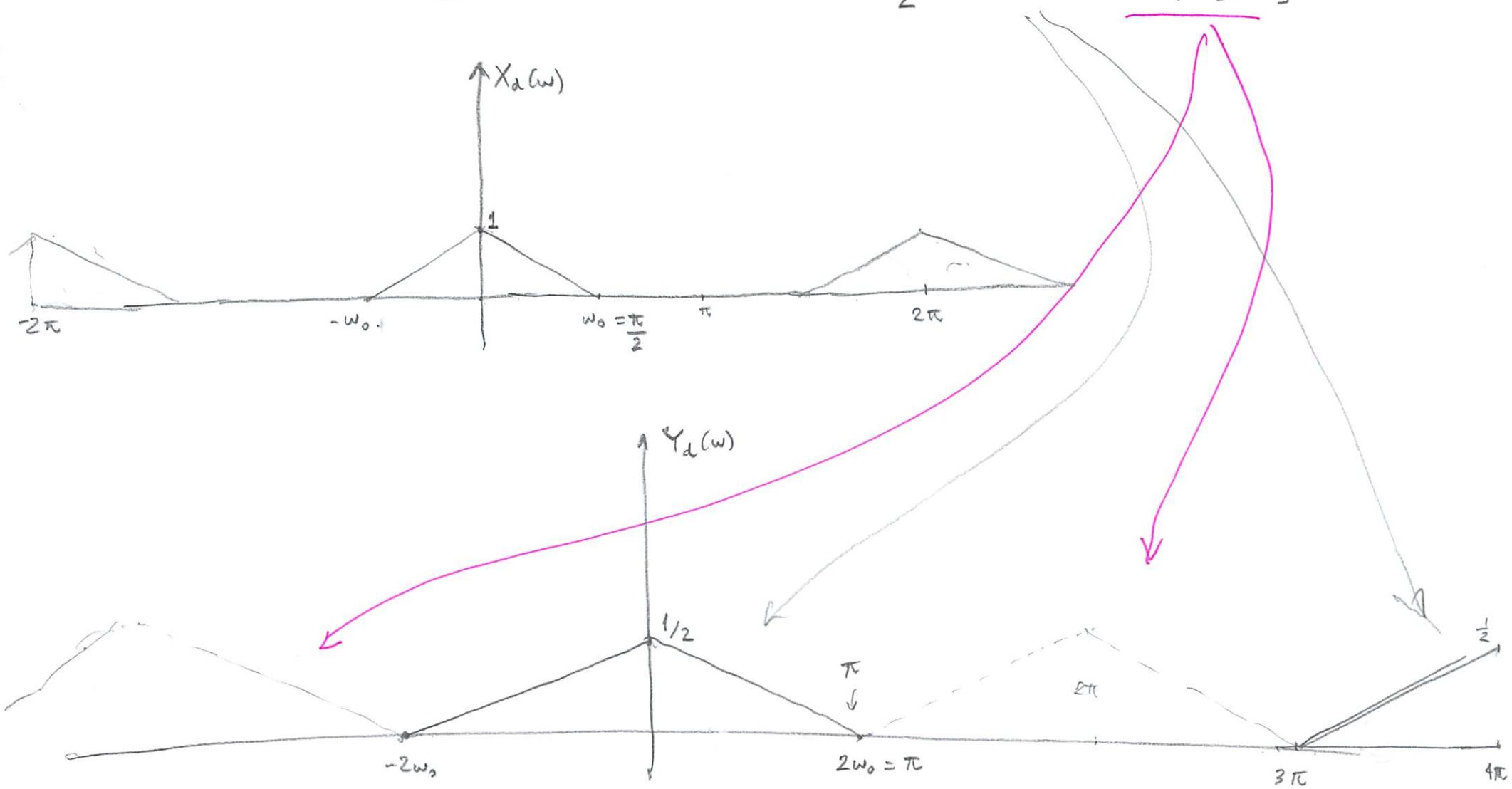


D is an integer

In words: Downsampling by D creates D shifted and scaled copies of  $X_d(\omega)$

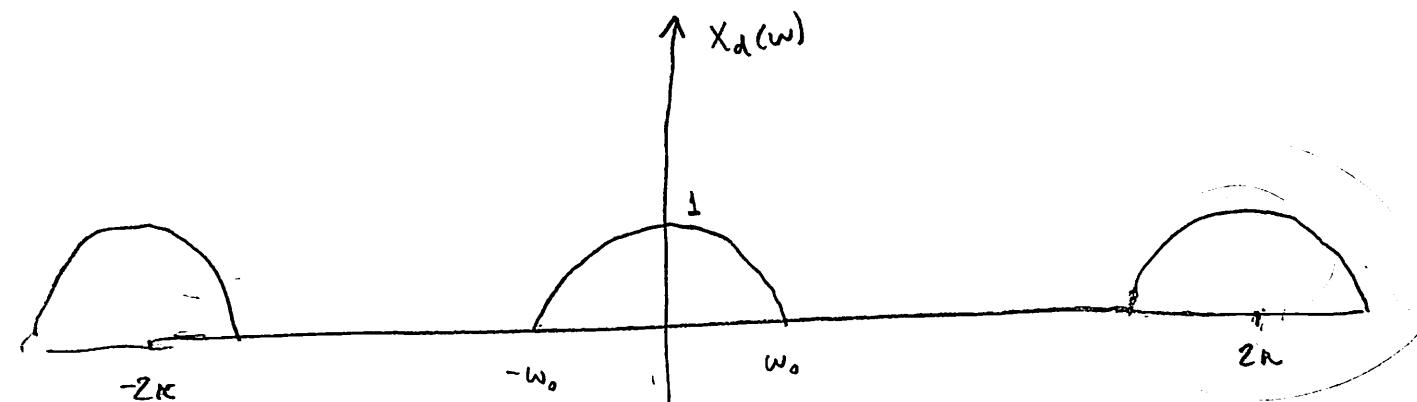
Ex:  $D=2$   $x[n] \rightarrow \text{Downsample by 2} \rightarrow y[n]$

$$Y_d(\omega) = \frac{1}{2} \left[ X_d\left(\frac{\omega}{2}\right) + X_d\left(\frac{\omega+2\pi}{2}\right) \right]$$

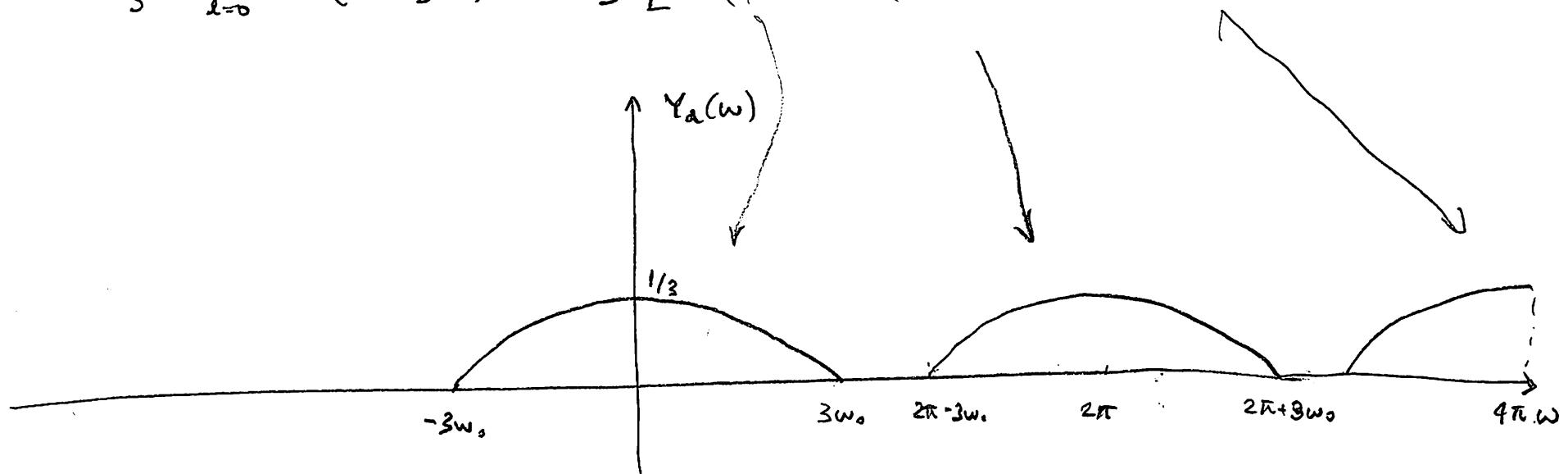


What if  $w_0 > \frac{\pi}{2}$ ? Aliasing.

Ex:  $D = 3$ :



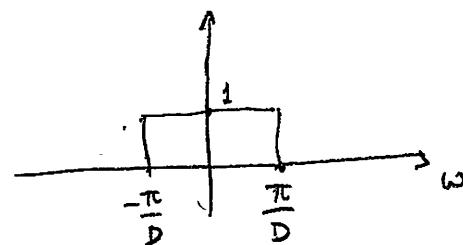
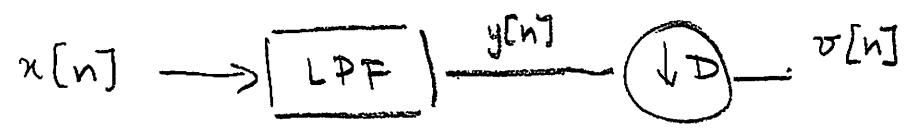
$$Y_d(w) = \frac{1}{3} \cdot \sum_{l=0}^2 X_d\left(\frac{w+2\pi l}{3}\right) = \frac{1}{3} \left[ X_d\left(\frac{w}{3}\right) + X_d\left(\frac{(w+2\pi)}{3}\right) + X_d\left(\frac{(w+4\pi)}{3}\right) \right]$$



Aliasing if  $w_0 > \frac{\pi}{3}$

In general, aliasing if  $w_0 > \frac{\pi}{D}$

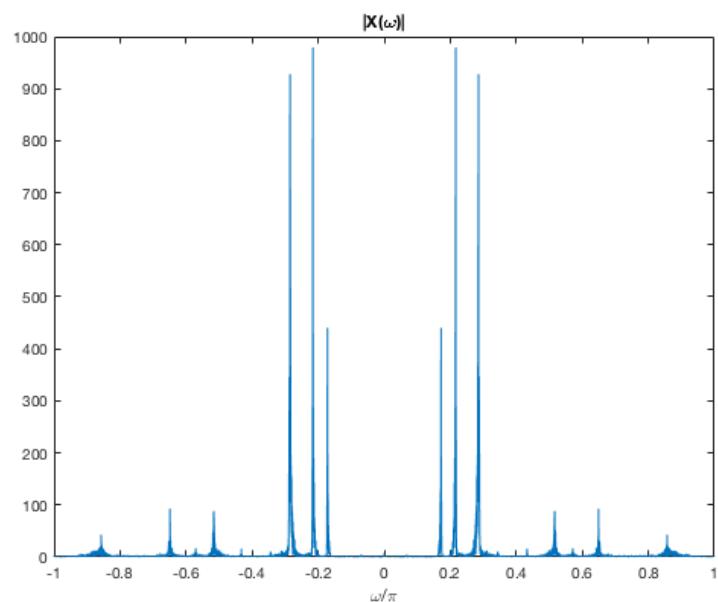
Solution: To avoid aliasing when down sampling by  $D$ , we apply a lowpass filter with  $\omega_c = \frac{\pi}{D}$  before down sampling



```
load train; x = y;
```

```
sound(x, Fs); % Listen to the audio signal
```

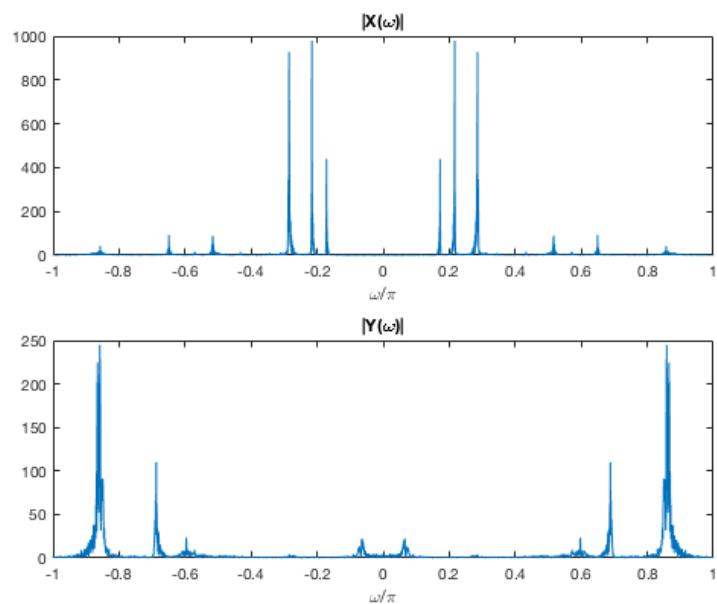
```
% magnitude of DTFT (via fft)
figure(1)
w = linspace(-pi, pi, length(x));
subplot(1,1,1), plot(w/pi, abs((fftshift(fft(x)))); ...
title ('|X(\omega)|'), xlabel ('\omega/\pi')
```



```
% DownSampling by D

D = 4;
y = x(1:D:end);

figure(2)
subplot(2,1,1), plot(w/pi, abs((fftshift(fft(x)))); ...
    title ('|X(\omega)|'), xlabel ('\omega/\pi')
subplot(2,1,2), plot(linspace(-pi,pi,length(y))/pi, ...
    abs((fftshift(fft(y)))); title ('|Y(\omega)|'), xlabel ('\omega/\pi')
```

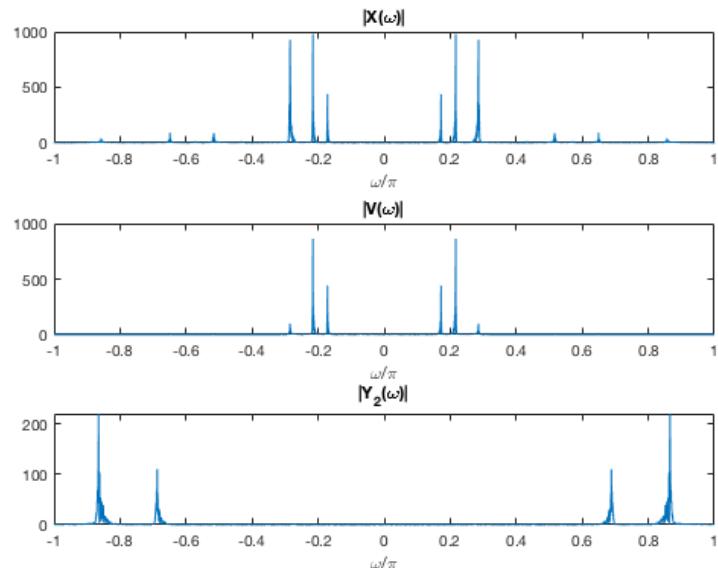


```
sound(y,Fs/D);
```

```
% Filter before downsampling
```

```
h = fir1(50, 1/D);  
v = filter(h, 1, x);  
y2 = v(1:D:end);
```

```
figure(3)  
subplot(3,1,1), plot(w/pi, abs((fftshift(fft(x)))); ...  
    title ('|X(\omega)|'), xlabel ('\omega/\pi')  
subplot(3,1,2), plot(linspace(-pi,pi,length(v))/pi, ...  
    abs((fftshift(fft(v)))); title ('|V(\omega)|'), xlabel ('\omega/\pi')  
subplot(3,1,3), plot(linspace(-pi,pi,length(y2))/pi, ...  
    abs((fftshift(fft(y2)))); title ('|Y_2(\omega)|'), xlabel ('\omega/\pi')
```



```
sound(y, Fs/D);
```

```
sound(y2, Fs/D);
```