

When you speak, your voice is picked up by an analog sensor in the cell phone's microphone.

An analog-to-digital converter chip converts your voice, which is an analog signal, into digital signals, represented by 1s and 0s.

The DSP compresses the digital signals and removes any background noise.

In the listener's cell phone, a digital-to-analog converter chip changes the digital signals back to an analog voice signal.

Your voice exits the phone through the speaker.

EHM 310

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MATLAB UYGULAMALAR (2)

An LTI system is specified by the difference equation

$$y(n) = 0.8y(n-1) + x(n)$$

- a. Determine $H(e^{j\omega})$.
- b. Calculate and plot the steady-state response $y_{ss}(n)$ to

$$x(n) = \cos(0.05\pi n)u(n)$$

Rewrite the difference equation as $y(n) - 0.8y(n-1) = x(n)$.

- a. Using (3.21), we obtain

$$H(e^{j\omega}) = \frac{1}{1 - 0.8e^{-j\omega}}$$

- b. In the steady state the input is $x(n) = \cos(0.05\pi n)$ with frequency $\omega_0 = 0.05\pi$ and $\theta_0 = 0^\circ$. The response of the system is

$$H(e^{j0.05\pi}) = \frac{1}{1 - 0.8e^{j0.05\pi}} = 4.0928e^{-j0.5377}$$

Therefore

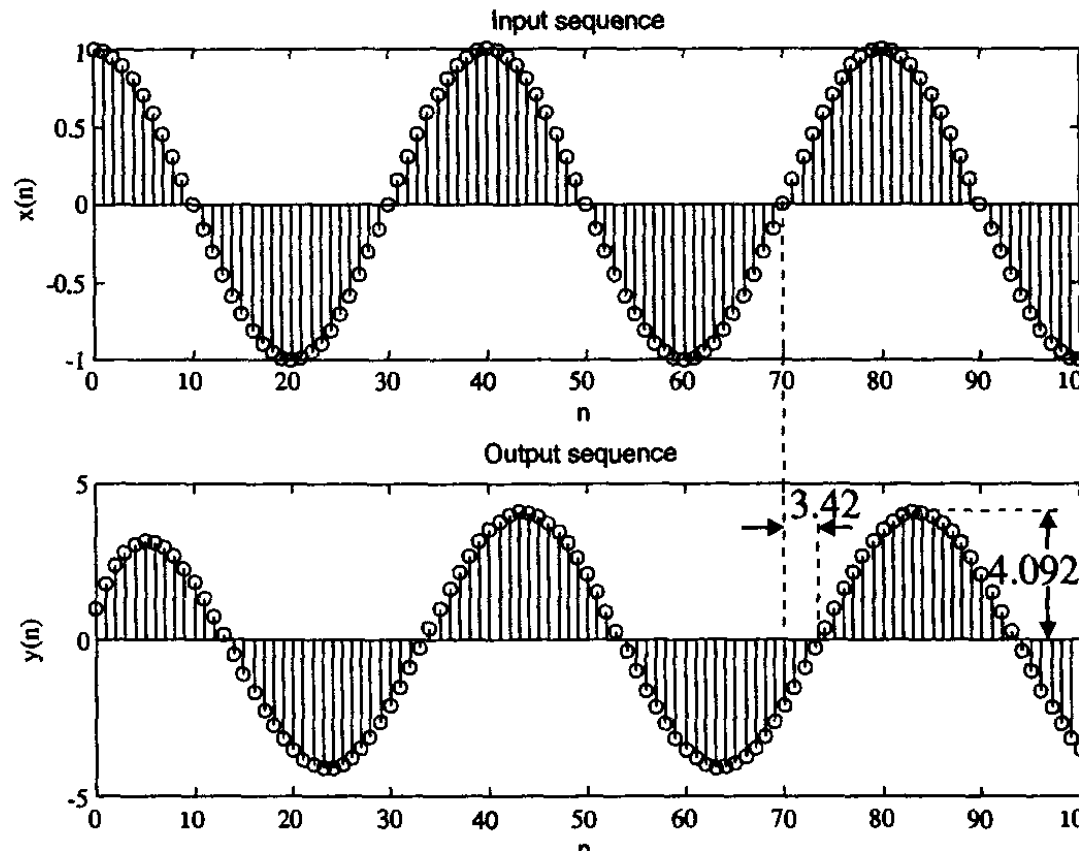
$$y_{ss}(n) = 4.0928 \cos(0.05\pi n - 0.5377) = 4.0928 \cos[0.05\pi(n - 3.42)]$$

This means that at the output the sinusoid is scaled by 4.0928 and shifted by 3.42 samples. This can be verified using MATLAB.

```

subplot(1,1,1)
b = 1; a = [1,-0.8];
n=[0:100];x = cos(0.05*pi*n);
y = filter(b,a,x);
subplot(2,1,1); stem(n,x);
xlabel('n'); ylabel('x(n)'); title('Input sequence')
subplot(2,1,2); stem(n,y);
xlabel('n'); ylabel('y(n)'); title('Output sequence')

```



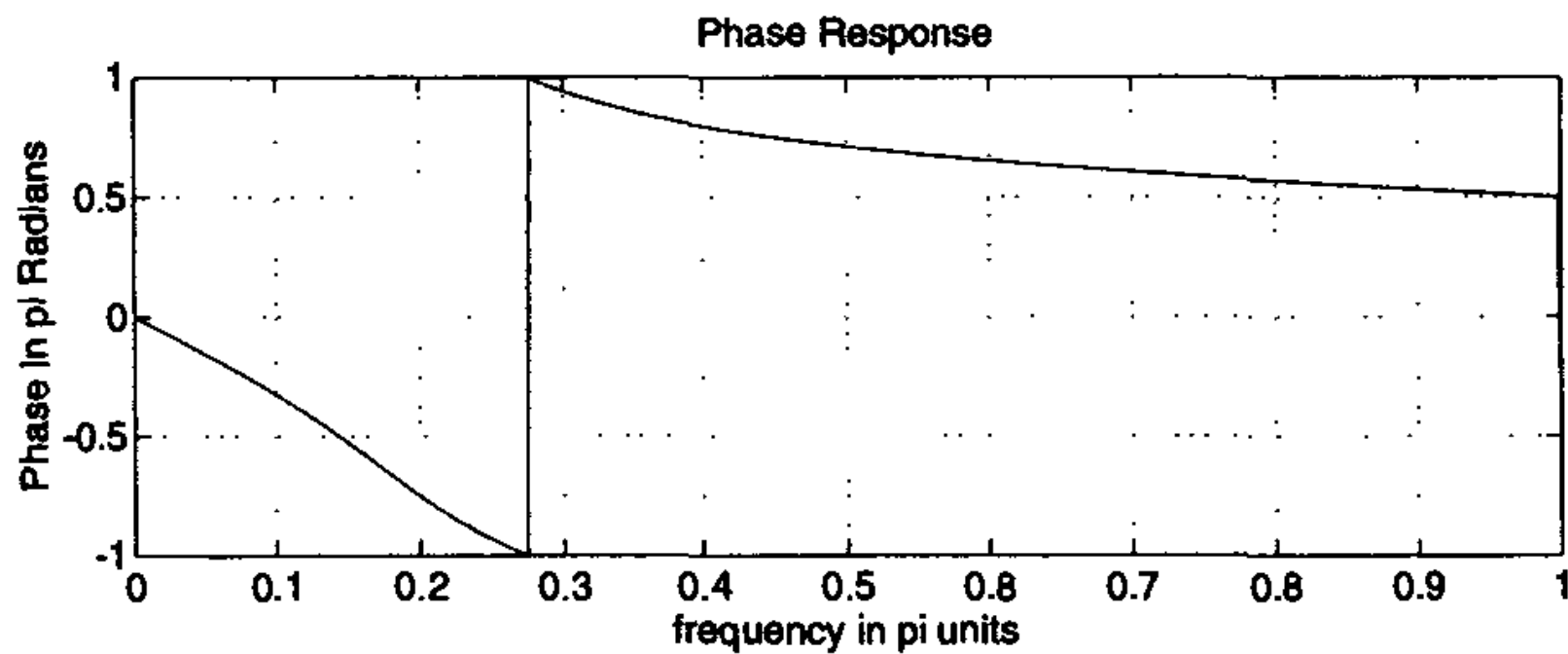
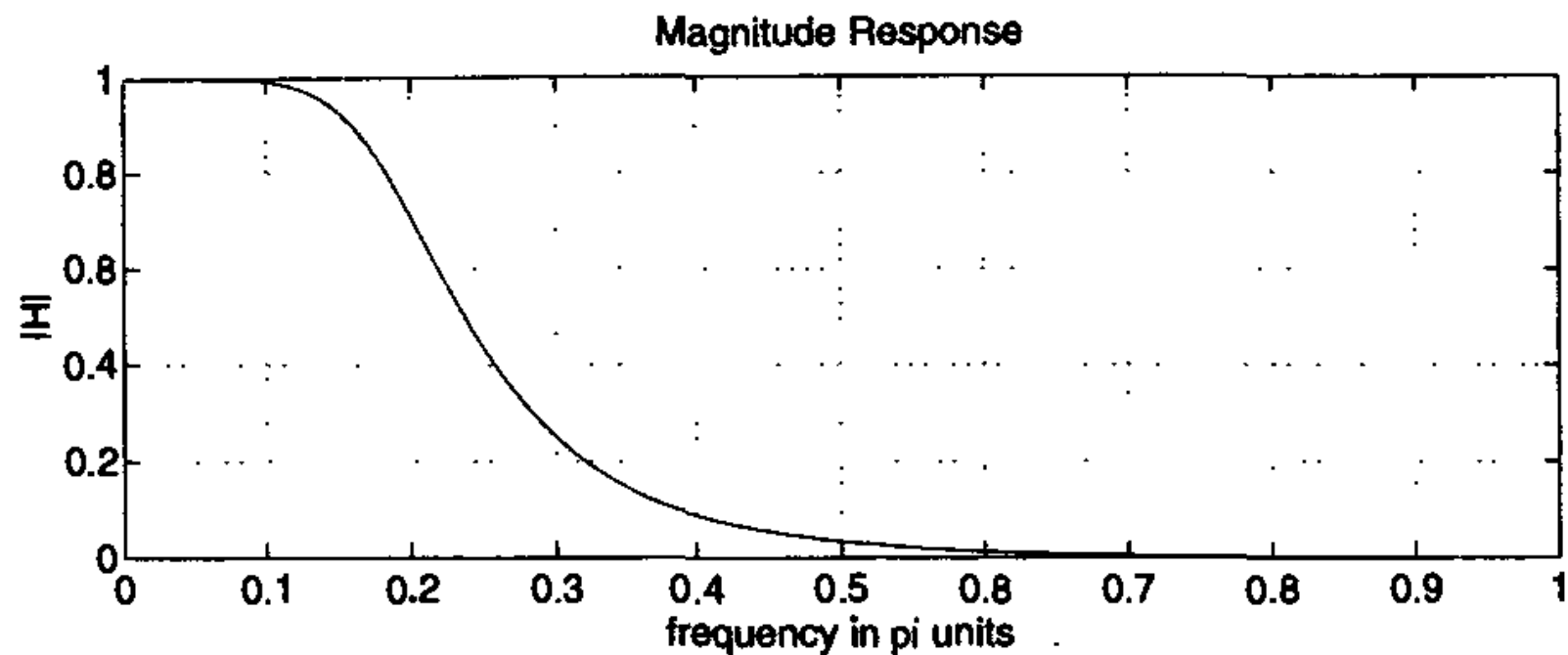
A 3rd-order lowpass filter is described by the difference equation

$$y(n) = 0.0181x(n) + 0.0543x(n-1) + 0.0543x(n-2) + 0.0181x(n-3) \\ + 1.76y(n-1) - 1.1829y(n-2) + 0.2781y(n-3)$$

Plot the magnitude and the phase response of this filter and verify that it is a lowpass filter.

```
b = [0.0181, 0.0543, 0.0543, 0.0181]; % filter coefficient array b
a = [1.0000, -1.7600, 1.1829, -0.2781]; % filter coefficient array a
m = 0:length(b)-1; l = 0:length(a)-1; % index arrays m and l
K = 500; k = 0:1:K; % index array k for frequencies
w = pi*k/K; % [0, pi] axis divided into 501 points.
num = b * exp(-j*m'*w); % Numerator calculations

den = a * exp(-j*l'*w); % Denominator calculations
H = num ./ den; % Frequency response
magH = abs(H); angH = angle(H); % mag and phase responses
subplot(1,1,1);
subplot(2,1,1); plot(w/pi,magH); grid; axis([0,1,0,1])
xlabel('frequency in pi units'); ylabel('|H|');
title('Magnitude Response');
subplot(2,1,2); plot(w/pi,angH/pi); grid
xlabel('frequency in pi units'); ylabel('Phase in pi Radians');
title('Phase Response');
```



$x_a(t) = e^{-1000|t|}$. Determine and plot its Fourier transform.

$$\begin{aligned} X_a(j\Omega) &= \int_{-\infty}^{\infty} x_a(t)e^{-j\Omega t} dt = \int_{-\infty}^0 e^{1000t} e^{-j\Omega t} dt + \int_0^{\infty} e^{-1000t} e^{-j\Omega t} dt \\ &= \frac{0.002}{1 + \left(\frac{\Omega}{1000}\right)^2} \end{aligned}$$

```
% Analog Signal
```

```
>> Dt = 0.00005; t = -0.005:Dt:0.005; xa = exp(-1000*abs(t));
```

```
% Continuous-time Fourier Transform
```

```
>> Wmax = 2*pi*2000; K = 500; k = 0:1:K; W = k*Wmax/K;
```

```
>> Xa = xa * exp(-j*t'*W) * Dt; Xa = real(Xa);
```

```
>> W = [-fliplr(W), W(2:501)]; % Omega from -Wmax to Wmax
```

```
>> Xa = [fliplr(Xa), Xa(2:501)]; % Xa over -Wmax to Wmax interval
```

```
>> subplot(1,1,1)
```

```
>> subplot(2,1,1); plot(t*1000, xa);
```

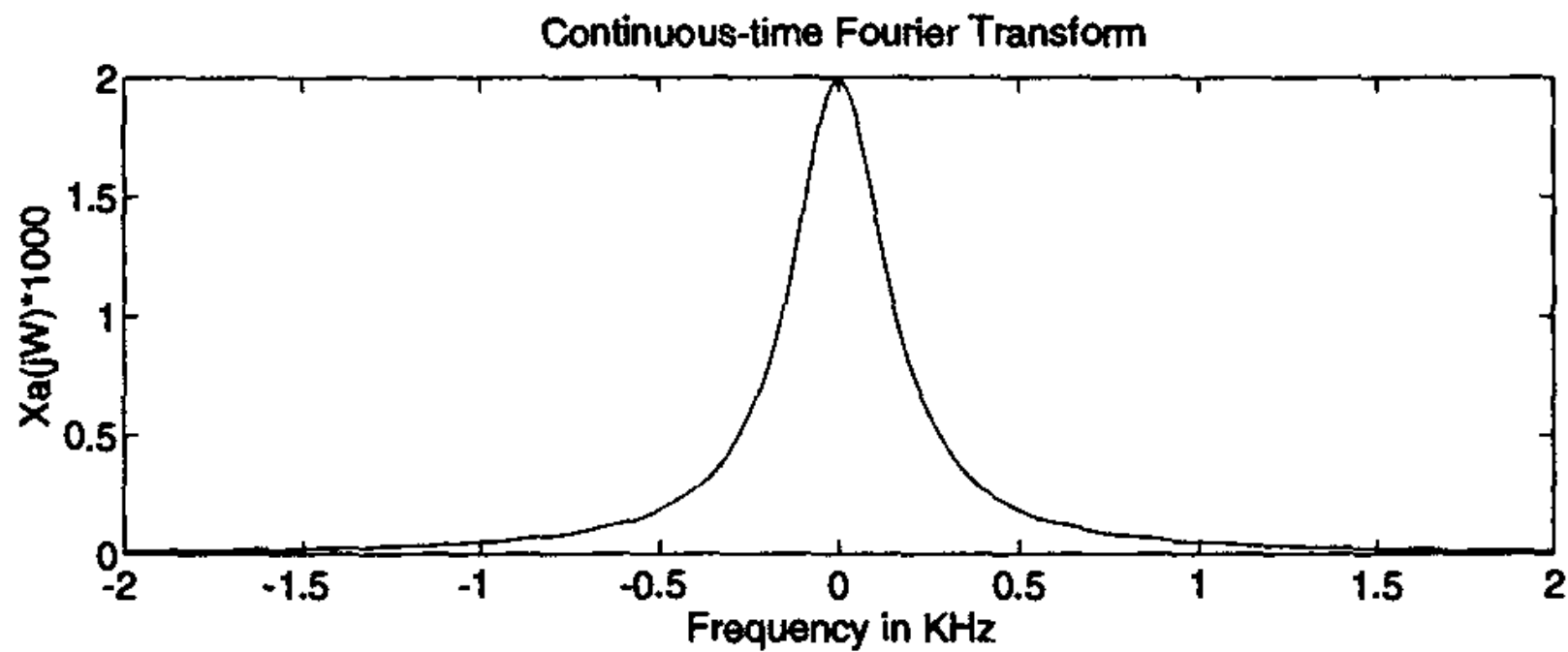
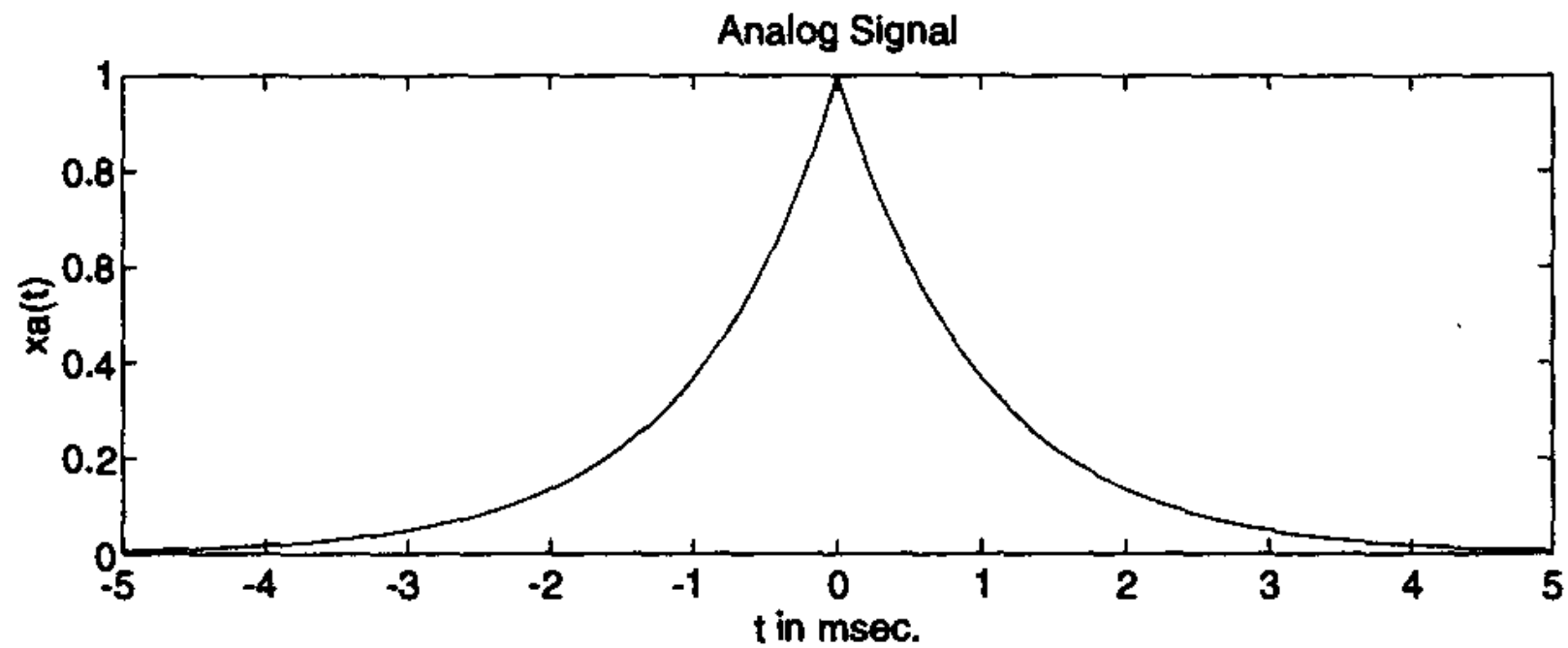
```
>> xlabel('t in msec. '); ylabel('xa(t)')
```

```
>> title('Analog Signal')
```

```
>> subplot(2,1,2); plot(W/(2*pi*1000), Xa*1000);
```

```
>> xlabel('Frequency in KHz'); ylabel('Xa(jW)*1000')
```

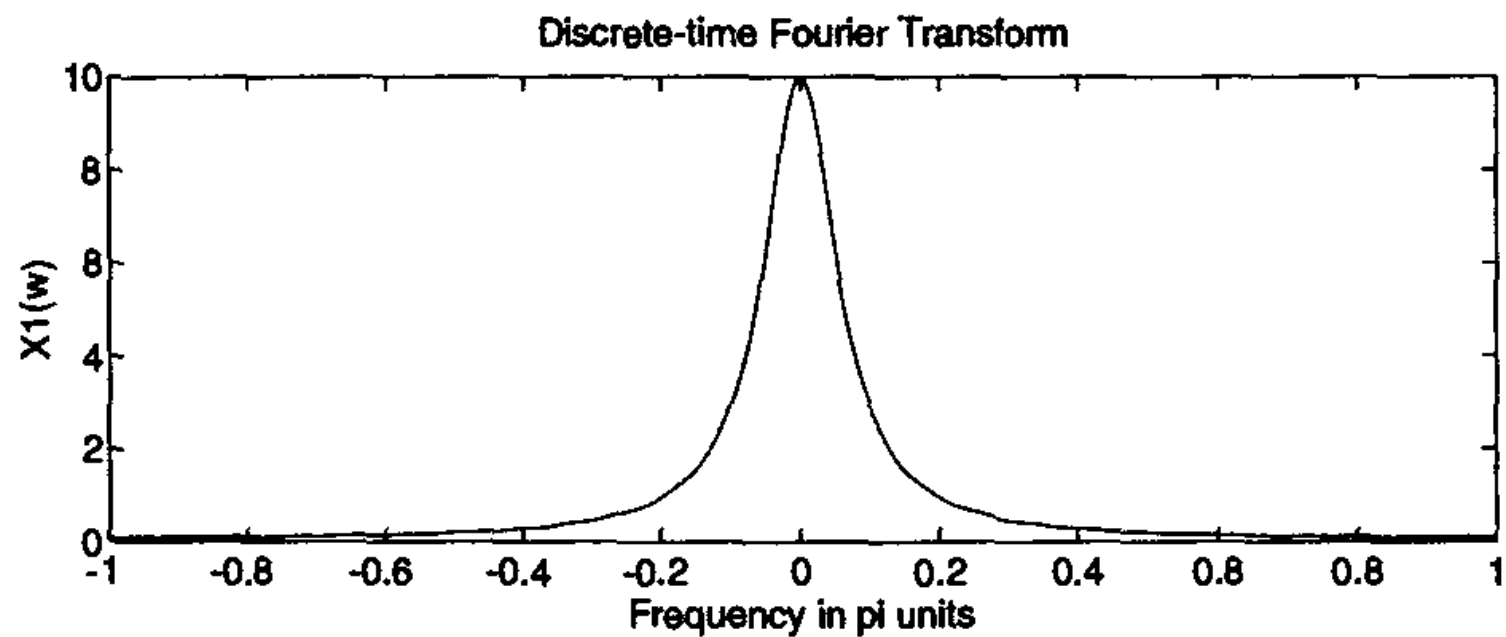
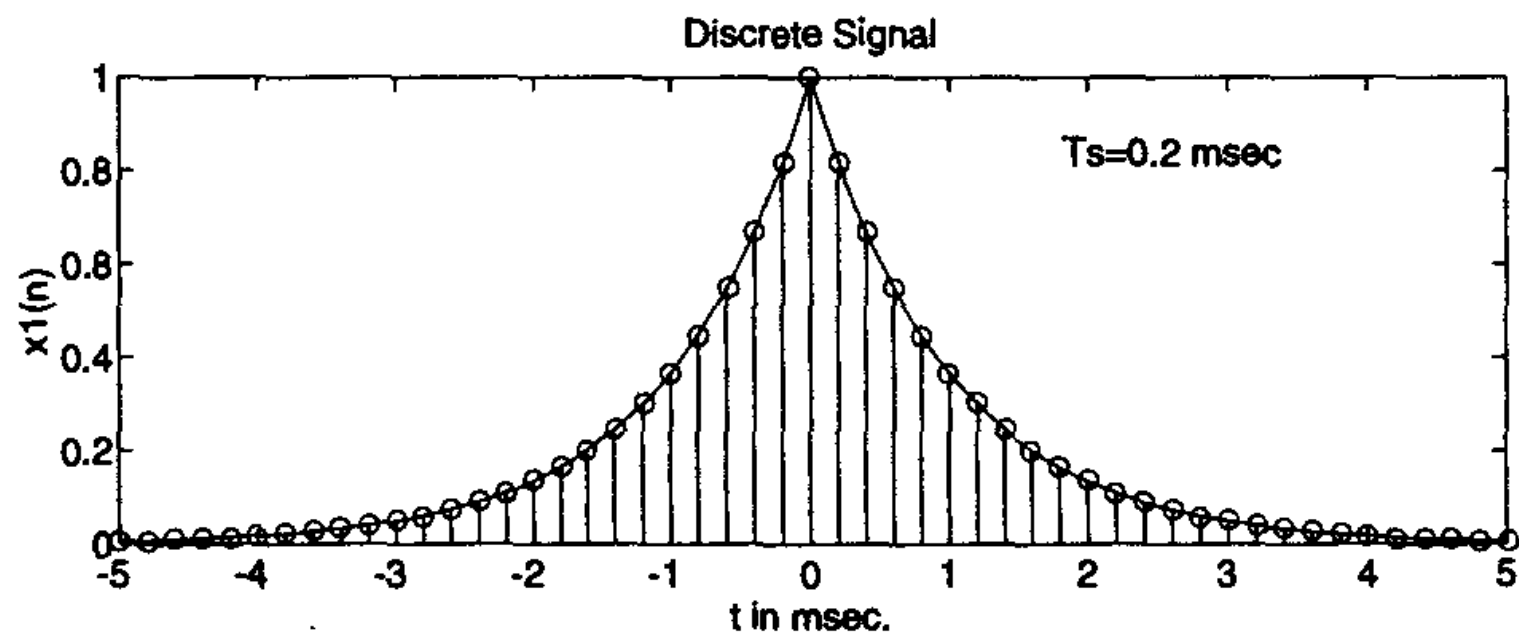
```
>> title('Continuous-time Fourier Transform')
```



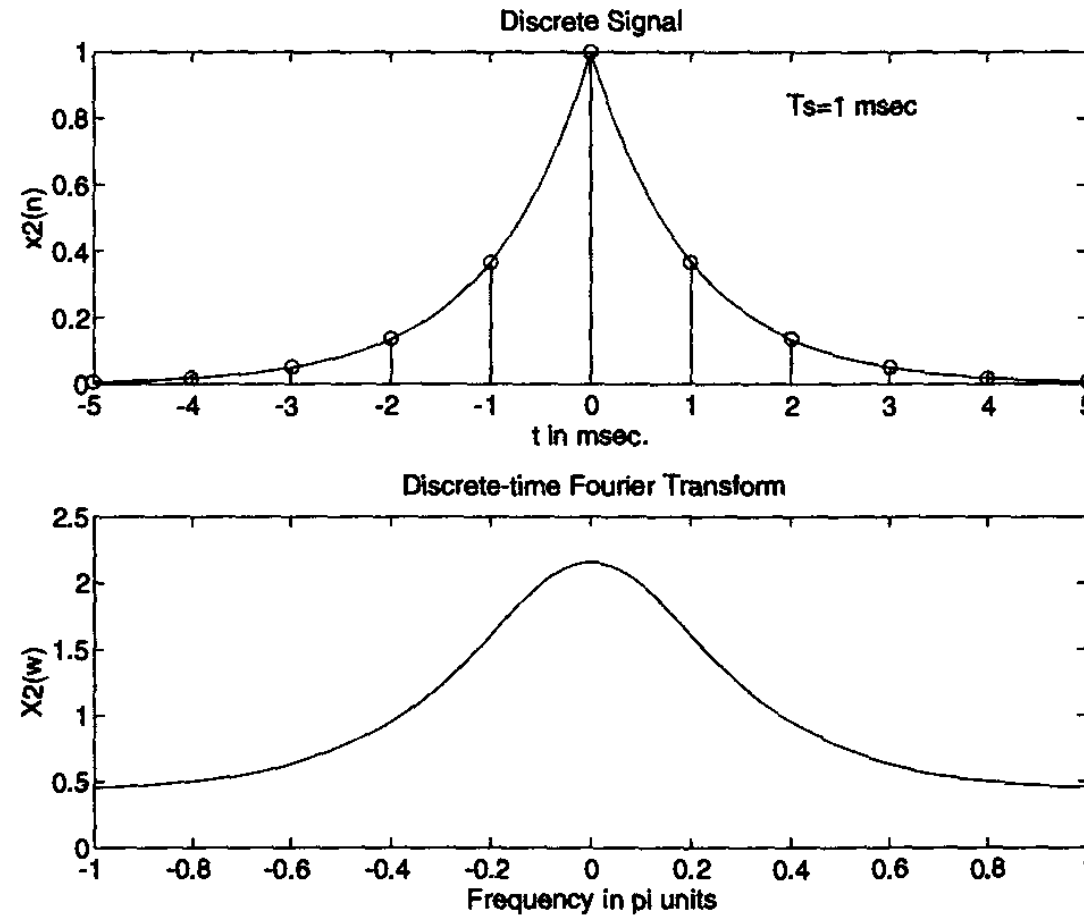
- a. Sample $x_a(t)$ at $F_s = 5000$ sam/sec to obtain $x_1(n)$. Determine and plot $X_1(e^{j\omega})$.
- b. Sample $x_a(t)$ at $F_s = 1000$ sam/sec to obtain $x_2(n)$. Determine and plot $X_2(e^{j\omega})$.

a. Since the bandwidth of $x_a(t)$ is 2KHz, the Nyquist rate is 4000 sam/sec, which is less than the given F_s . Therefore aliasing will be (almost) nonexistent.

```
% Analog Signal
>> Dt = 0.00005; t = -0.005:Dt:0.005; xa = exp(-1000*abs(t));
% Discrete-time Signal
>> Ts = 0.0002; n = -25:1:25; x = exp(-1000*abs(n*Ts));
% Discrete-time Fourier transform
>> K = 500; k = 0:1:K; w = pi*k/K;
>> X = x * exp(-j*n'*w); X = real(X);
>> w = [-fliplr(w), w(2:K+1)];
>> X = [fliplr(X), X(2:K+1)];
>> subplot(1,1,1)
>> subplot(2,1,1);plot(t*1000,xa);
>> xlabel('t in msec. '); ylabel('x1(n)')
>> title('Discrete Signal'); hold on
>> stem(n*Ts*1000,x); gtext('Ts=0.2 msec'); hold off
>> subplot(2,1,2);plot(w/pi,X);
>> xlabel('Frequency in pi units'); ylabel('X1(w)')
>> title('Discrete-time Fourier Transform')
```

b. Here $F_s = 1000 < 4000$. Hence there will be a considerable amount of aliasing. This is evident from Figure 3.13, in which the shape of $X(e^{j\omega})$ is different from that of $X_a(j\Omega)$ and can be seen to be a result of adding overlapping replicas of $X_a(j\Omega)$. \square



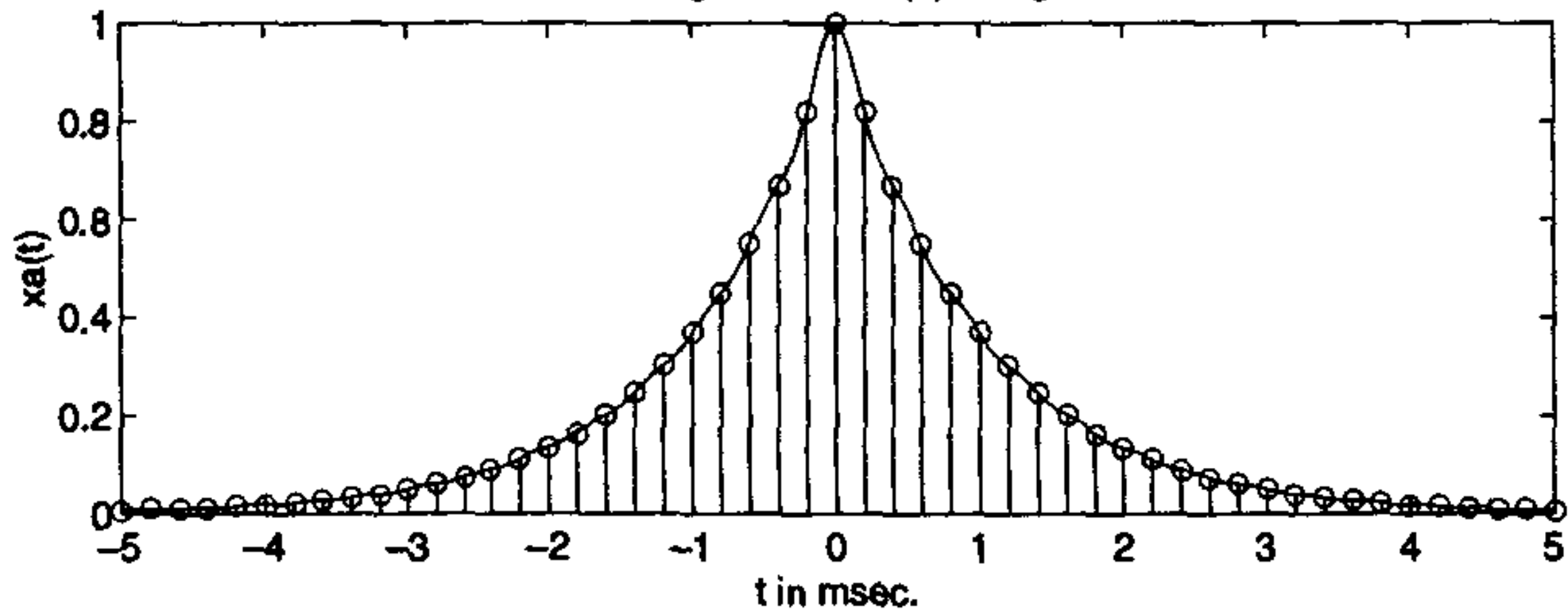
$$x_a(t) = e^{-1000|t|}, \quad \text{Reconstruct (Fs=5000 samp/sec)}$$

Note that $x_1(n)$ was obtained by sampling $x_a(t)$ at $T_s = 1/F_s = 0.0002$ sec. We will use the grid spacing of 0.00005 sec over $-0.005 \leq t \leq 0.005$, which gives $x(n)$ over $-25 \leq n \leq 25$.

```
% Discrete-time Signal x1(n)
>> Ts = 0.0002; n = -25:1:25; nTs = n*Ts;
>> x = exp(-1000*abs(nTs));
% Analog Signal reconstruction
>> Dt = 0.00005; t = -0.005:Dt:0.005;
>> xa = x * sinc(Fs*(ones(length(n),1)*t-nTs'*ones(1,length(t))));
% check
>> error = max(abs(xa - exp(-1000*abs(t))))
error =
    0.0363
```

The maximum error between the reconstructed and the actual analog signal is 0.0363, which is due to the fact that $x_a(t)$ is not strictly band-limited (and also we have a finite number of samples). From Figure 3.15 we note that visually the reconstruction is excellent.

Reconstructed Signal from $x_1(n)$ using sinc function



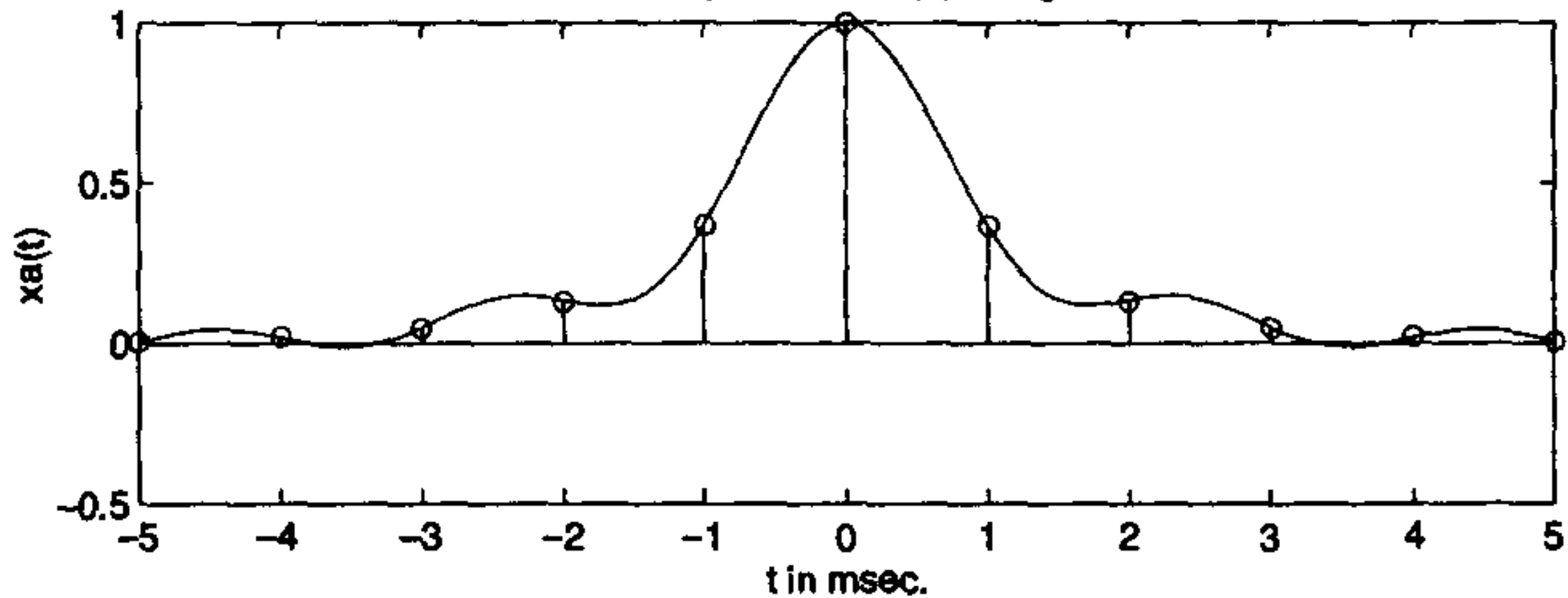
Reconstruct ($F_s=1000$ samp/sec) $x_a(t) = e^{-1000|t|}$.

In this case $x_2(n)$ was obtained by sampling $x_a(t)$ at $T_s = 1/F_s = 0.001$ sec. We will again use the grid spacing of 0.00005 sec over $-0.005 \leq t \leq 0.005$, which gives $x(n)$ over $-5 \leq n \leq 5$.

```
% Discrete-time Signal x2(n)
>> Ts = 0.001; n = -5:1:5; nTs = n*Ts;
>> x = exp(-1000*abs(nTs));
% Analog Signal reconstruction
>> Dt = 0.00005; t = -0.005:Dt:0.005;
>> xa = x * sinc(Fs*(ones(length(n),1)*t-nTs'*ones(1,length(t))));
% check
>> error = max(abs(xa - exp(-1000*abs(t))))
error =
    0.1852
```

The maximum error between the reconstructed and the actual analog signal is 0.1852, which is significant and cannot be attributed to the nonband-limitedness of $x_a(t)$ alone. From Figure 3.16 observe that the reconstructed signal differs from the actual one in many places over the interpolated regions. This is the visual demonstration of aliasing in the time domain. \square

Reconstructed Signal from $x_2(n)$ using sinc function



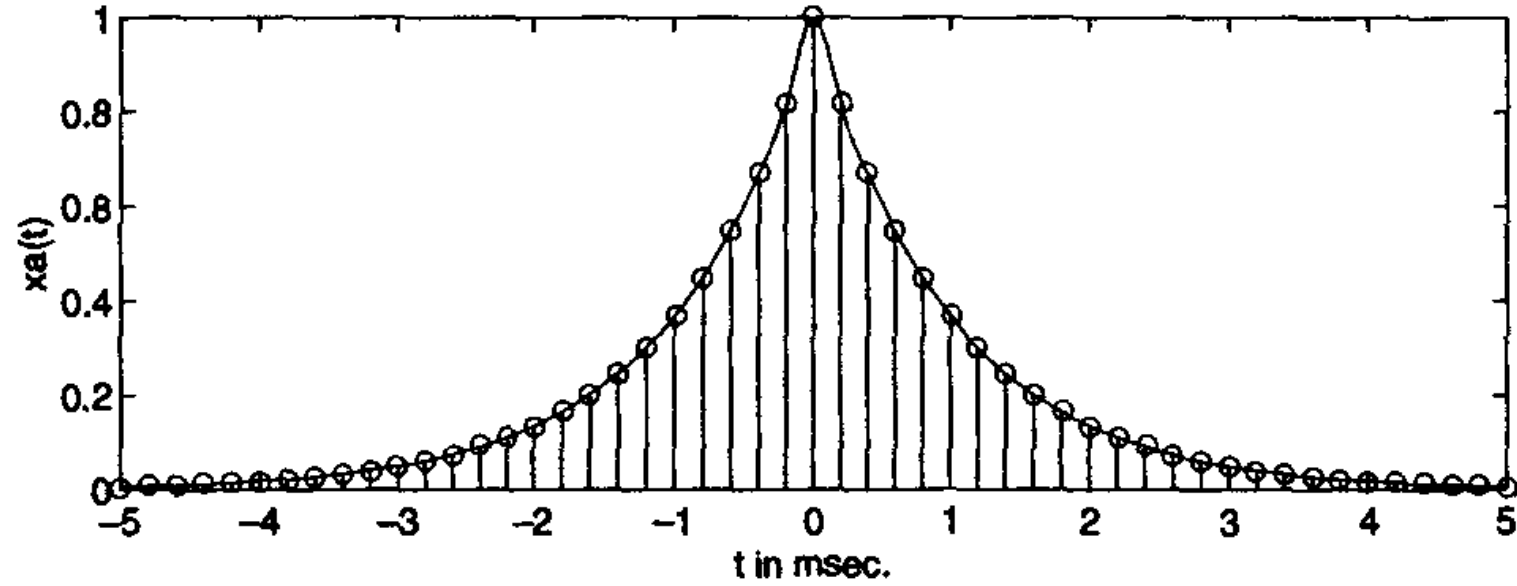
```

% a) Discrete-time Signal x1(n): Ts = 0.0002
>> Ts = 0.0002; n = -25:1:25; nTs = n*Ts;
>> x = exp(-1000*abs(nTs));
% Analog Signal reconstruction
>> Dt = 0.00005; t = -0.005:Dt:0.005;
>> xa = spline(nTs,x,t);
% check
>> error = max(abs(xa - exp(-1000*abs(t))))
error = 0.0317

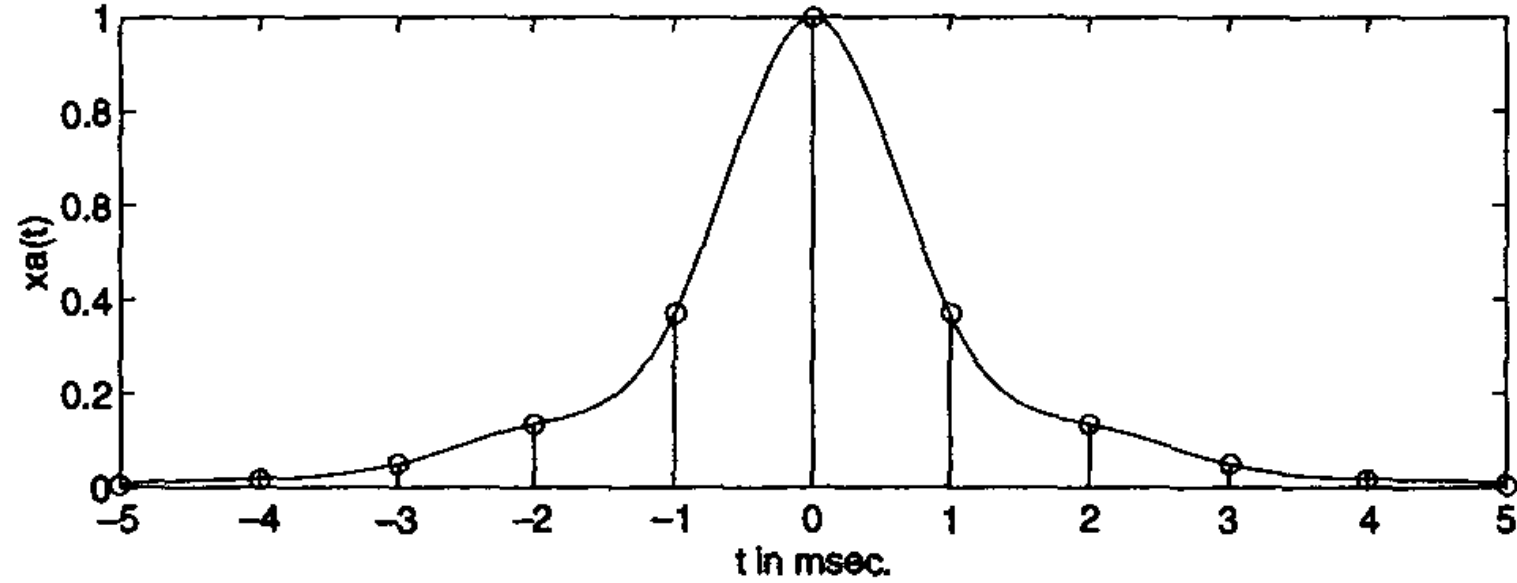
% Discrete-time Signal x2(n): Ts = 0.001
>> Ts = 0.001; n = -5:1:5; nTs = n*Ts;
>> x = exp(-1000*abs(nTs));
% Analog Signal reconstruction
>> Dt = 0.00005; t = -0.005:Dt:0.005;
>> xa = spline(nTs,x,t);
% check
>> error = max(abs(xa - exp(-1000*abs(t))))
error = 0.1679

```

Reconstructed Signal from $x_1(n)$ using cubic spline function



Reconstructed Signal from $x_2(n)$ using cubic spline function



Z Dönüşümü

$$X_1(z) = 2 + 3z^{-1} + 4z^{-2} \text{ and } X_2(z) = 3 + 4z^{-1} + 5z^{-2} + 6z^{-3}.$$

$$X_3(z) = X_1(z)X_2(z).$$

$$x_1(n) = \underset{\uparrow}{\{2, 3, 4\}} \quad \text{and} \quad x_2(n) = \underset{\uparrow}{\{3, 4, 5, 6\}}$$

```
>> x1 = [2,3,4]; x2 = [3,4,5,6];
```

```
>> x3 = conv(x1,x2)
```

```
x3 =      6      17      34      43      38      24
```

$$X_3(z) = 6 + 17z^{-1} + 34z^{-2} + 43z^{-3} + 38z^{-4} + 24z^{-5}$$

$$X(z) = \frac{z}{3z^2 - 4z + 1}$$

$$X(z) = \frac{z^{-1}}{3 - 4z^{-1} + z^{-2}} = \frac{0 + z^{-1}}{3 - 4z^{-1} + z^{-2}}$$

```
>> b = [0,1]; a = [3,-4,1];
```

```
>> [R,p,C] = residuez(b,a)
```

```
R =
```

```
    0.5000
```

```
   -0.5000
```

```
p =
```

```
    1.0000
```

```
    0.3333
```

```
c =
```

```
    []
```

$$X(z) = \frac{\frac{1}{2}}{1 - z^{-1}} - \frac{\frac{1}{2}}{1 - \frac{1}{3}z^{-1}}$$

```
>> [b,a] = residuez(R,p,C)
```

```
b =
```

```
0.0000
```

```
0.3333
```

```
a =
```

```
1.0000
```

```
-1.3333
```

```
0.3333
```

$$X(z) = \frac{0 + \frac{1}{3}z^{-1}}{1 - \frac{4}{3}z^{-1} + \frac{1}{3}z^{-2}} = \frac{z^{-1}}{3 - 4z^{-1} + z^{-2}} = \frac{z}{3z^2 - 4z + 1}$$

$$X(z) = \frac{1}{(1 - 0.9z^{-1})^2 (1 + 0.9z^{-1})}, \quad |z| > 0.9$$

```
>> b = 1; a = poly([0.9,0.9,-0.9])
a =
    1.0000   -0.9000   -0.8100    0.7290
>> [R,p,C]=residuez(b,a)
R =
    0.2500
    0.5000
    0.2500
p =
    0.9000
    0.9000
   -0.9000
c =
    []
```

$$X(z) = \frac{0.25}{1 - 0.9z^{-1}} + \frac{0.5}{(1 - 0.9z^{-1})^2} + \frac{0.25}{1 + 0.9z^{-1}}, \quad |z| > 0.9$$

$$= \frac{0.25}{1 - 0.9z^{-1}} + \frac{0.5}{0.9} z \frac{(0.9z^{-1})}{(1 - 0.9z^{-1})^2} + \frac{0.25}{1 + 0.9z^{-1}}, \quad |z| > 0.9$$

$$x(n) = 0.25 (0.9)^n u(n) + \frac{5}{9} (n+1) (0.9)^{n+1} u(n+1) + 0.25 (-0.9)^n u(n)$$

$$x(n) = 0.75 (0.9)^n u(n) + 0.5n (0.9)^n u(n) + 0.25 (-0.9)^n u(n)$$

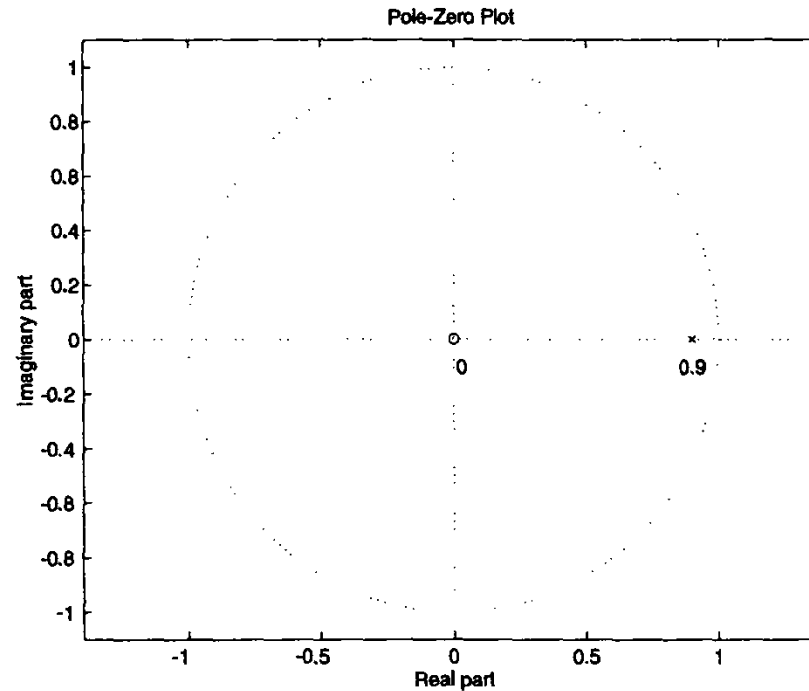
Given a causal system

$$y(n) = 0.9y(n-1) + x(n)$$

- Find $H(z)$ and sketch its pole-zero plot.
- Plot $|H(e^{j\omega})|$ and $\angle H(e^{j\omega})$.
- Determine the impulse response $h(n)$.

$$y(n) - 0.9y(n-1) = x(n) \qquad H(z) = \frac{1}{1 - 0.9z^{-1}}; \quad |z| > 0.9$$

```
b = [1, 0]; a = [1, -0.9];  
zplane(b,a)
```



```

[H,w] = freqz(b,a,100);
magH = abs(H); phaH = angle(H);
subplot(2,1,1);plot(w/pi,magH);grid
xlabel('frequency in pi units'); ylabel('Magnitude');
title('Magnitude Response')
subplot(2,1,2);plot(w/pi,phaH/pi);grid
xlabel('frequency in pi units'); ylabel('Phase in pi units');
title('Phase Response')

```

$$h(n) = \mathcal{Z}^{-1} \left[\frac{1}{1 - 0.9z^{-1}}, |z| > 0.9 \right] = (0.9)^n u(n)$$

