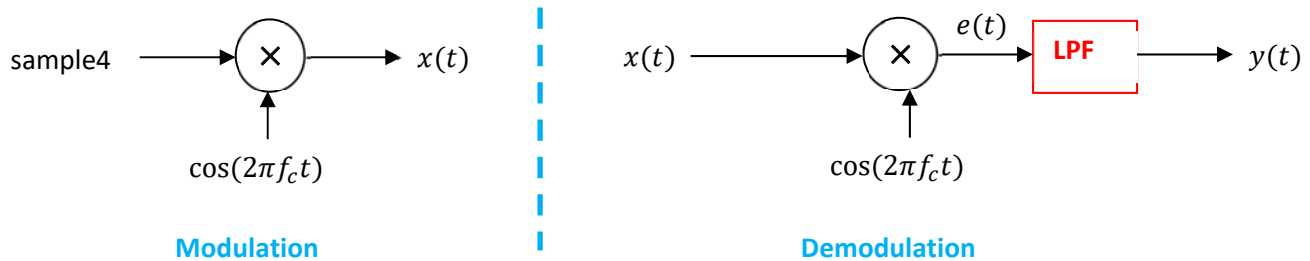


Objectives

- To be familiar with modulation and demodulation
- To be familiar with sampling

Ex.1 Modulation and Demodulation

The block diagram of modulation and demodulation is shown below.



```
[s4,fs]=audioread('sample4.wav');  
s4=s4';  
soundsc(s4,fs);
```

% Modulation

```
t=[0:length(s4)-1]/fs;  
c=cos(2*pi*10e3*t);  
x=s4.*c;  
f=[-length(s4)/2:length(s4)/2-1]*fs/length(s4);
```

% Demodulation

```
e=x.*c;
```

```
% read the audio file and sample rate
% transpose
% hear the audio signal
```

```
% time index
% carrier frequency is 10 kHz (10e3 = 10000)
% x is the modulated signal
% frequency index (from -fs/2 to fs/2)
```

% frequency shifting (back to the baseband)

Design a lowpass filter (LPF) and plot the magnitude response. Check the cutoff frequency before performing lowpass filtering. Plot figure(1).

- % Use 'butter' to design a lowpass filter (1)
- % Use 'filter' to perform lowpass filtering (1)
- % You may refer to Ex.2 and Ex.3 in Prelab 3. Set N = 18.

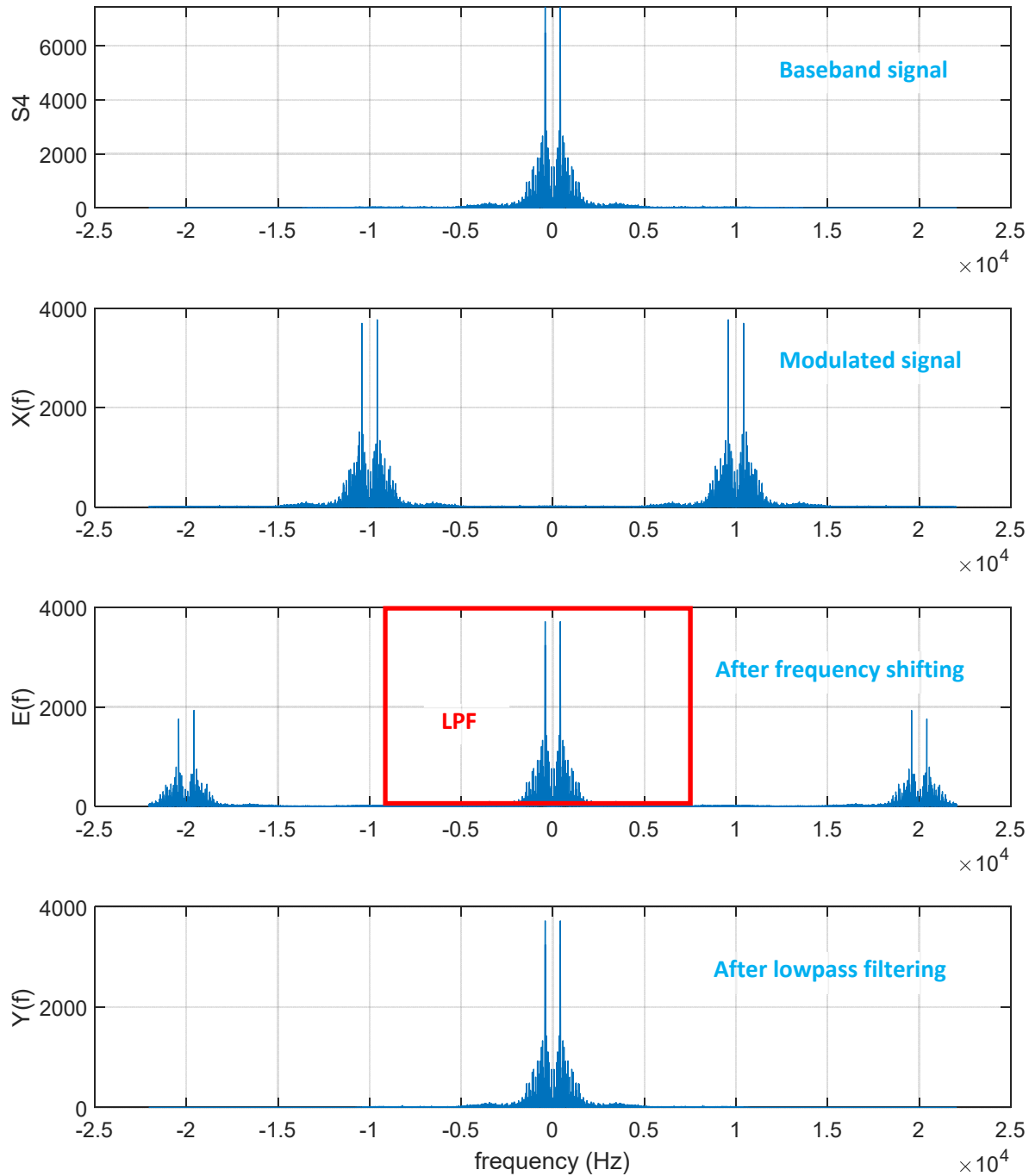
(Write your Matlab codes)

(Write your Matlab codes)

```
figure(1);
subplot(411); plot(f, abs(fftshift(fft(s4)))); ylabel('S4'); grid;
subplot(412); plot(f, abs(fftshift(fft(x)))); ylabel('X(f)'); grid;
subplot(413); plot(f, abs(fftshift(fft(e)))); ylabel('E(f)'); grid;
subplot(414); plot(f, abs(fftshift(fft(y)))); ylabel('Y(f)'); grid;
xlabel('frequency (Hz)');
```

- % spectrum of baseband signal
- % spectrum of modulated signal
- % after frequency shifting
- % after lowpass filtering

figure(1)



Self-check :

- What is the unilateral bandwidth of sample4 ?
- How to decide the cutoff frequency of LPF ?
- Hear the modulated signal (x).
- Hear the demodulated signal (y).
- What is the difference between the modulated signal and demodulated signal ?
- What do you hear if the carrier frequency used in the demodulation is changed to 20 kHz ?
i.e. The carrier frequencies used in the modulation and the demodulation are different from each other.

Ex.2 Sampling

Convert a CT signal $x(t) = \cos(40000\pi t)$ into DT sequence using two different sampling frequencies.
Plot the sequences in the time domain.
Plot the magnitude spectrums versus the **actual** frequency (in Hz).

<code>N=1000;</code>	% number of points
<code>n=0:N-1;</code>	% n index
<code>fs1=25e3;</code>	% sampling frequency 1 (25 kHz)
<code>x1=cos(2*pi*20e3*n/fs1);</code>	% x is sampled using fs1 = x1[n]
<code>f1=[-N/2:N/2-1]*(fs1/N);</code>	% frequency index for x1
<code>figure(2)</code>	
<code>subplot(211); stem(n,x1,'. '); ylabel('x1[n]'); xlabel('n');</code>	% plot the DT signal x1
<code>grid; axis([0 20 -1 1]);</code>	
<code>subplot(212); plot(f1, abs(fftshift(fft(x1))/length(x1)));</code>	% plot magnitude spectrum of x1
<code>grid; ylabel(' X1 '); xlabel('frequency (Hz)');</code>	
<code>fs2=200e3;</code>	% sampling frequency 2 (200 kHz)
<code>x2=cos(2*pi*20e3*n/fs2);</code>	% x is sampled using fs2 = x2[n]
<code>f2=[-N/2:N/2-1]*(fs2/N);</code>	% frequency index for x2
<code>figure(3)</code>	
<code>subplot(211); stem(n,x2,'. '); ylabel('x2[n]'); xlabel('n');</code>	% plot the DT signal x2
<code>grid; axis([0 20 -1 1]);</code>	
<code>subplot(212); plot(f2, abs(fftshift(fft(x2))/length(x2)));</code>	% plot magnitude spectrum of x2
<code>grid; ylabel(' X2 '); xlabel('frequency (Hz)');</code>	

Self-check :

- What is the signal frequency (in Hz) from the mathematical expression ?
- What is the signal frequency (in Hz) shown in figure(2) and figure(3) ?
- Can you explain the result observed in figure(2) and figure(3) using sampling theorem ?
- Can you write down the mathematical expressions of sampled sequence $x1[n]$ and $x2[n]$?
- What is the normalized frequency of $x1[n]$?
- What is the normalized frequency of $x2[n]$?
- What is the relationship between actual frequency and normalized frequency ?