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

Part I **(11)**

a) Download the audio file (**m1a.wav**) . Notice that there are **TWO** modulated signals.

b) Use "**audioread**" to read the audio file and sampling frequency fs (in kHz).

(1) **The sampling frequency = 192 000 Hz**

c) Define a frequency index from $-\frac{f_s}{2}$ to $\frac{f_s}{2}$.

d) Use **"subplot(311)"**, "**fft**", "**fftshift**" and "**abs**" to plot the magnitude spectrum of the audio file versus frequency (Hz) in figure(1).

e) Observe the carrier frequency (in kHz) for each modulated signal.

(2) **Located at lower frequency band = 28 kHz** **Located at higher frequency band = 55 kHz**

f) Shift the spectrum located at **higher** frequency band back to the baseband using a **correct** carrier frequency.

g) Use **"subplot(312)"**, "**fft**", "**fftshift**" and "**abs**" to plot the magnitude spectrum after frequency shifting in figure(1).

h) Design a Butterworth lowpass filter using "**butter**" and set **N = 16**.

i) Determine the cutoff frequencies (in kHz) and write down the corresponding value of Wn. Use "Datatip" to check the width of the passband.

(1) **Cutoff = 10 kHz** **Wn = 0.1042**

j) Use "**freqz**" to generate the frequency response and "**abs**" to plot the magnitude response in figure(2).

k) Use "**filter**" to perform lowpass filtering.

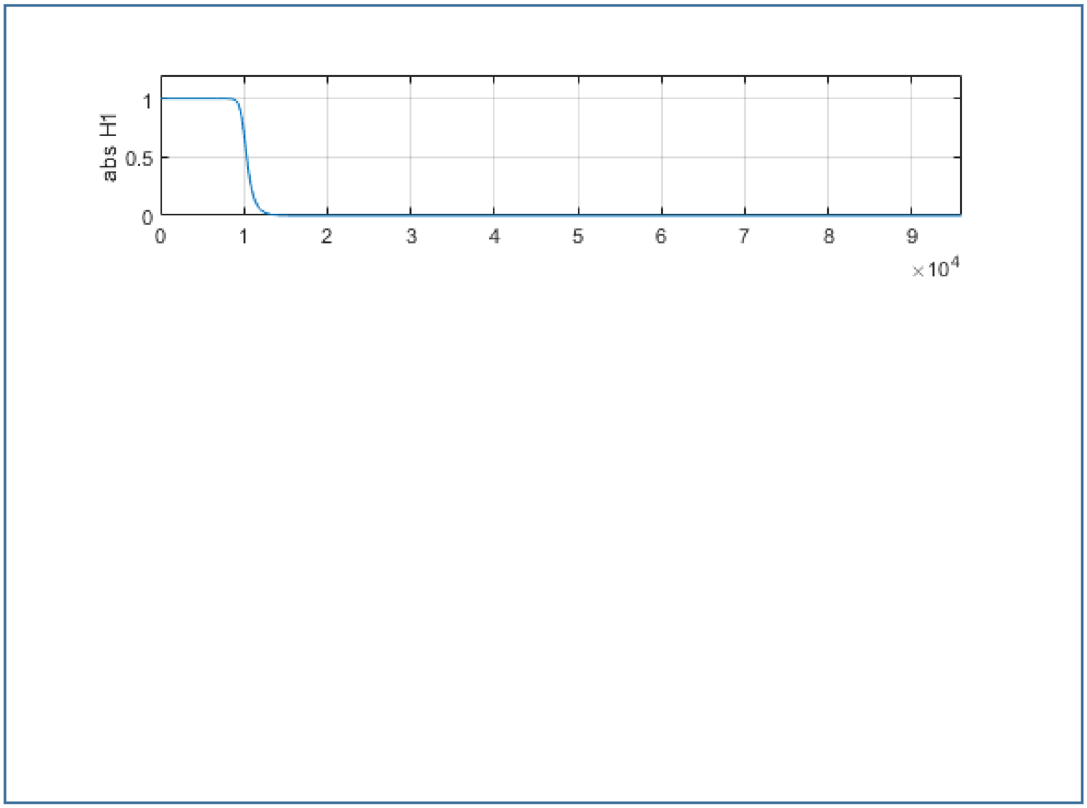
l) Use **"subplot(313)"**, "**fft**", "**fftshift**" and "**abs**" to plot the magnitude spectrum of the output in figure(1).

m) Use "**soundsc**" and the sampling frequency to hear the audio file (**m1a**) and the output.

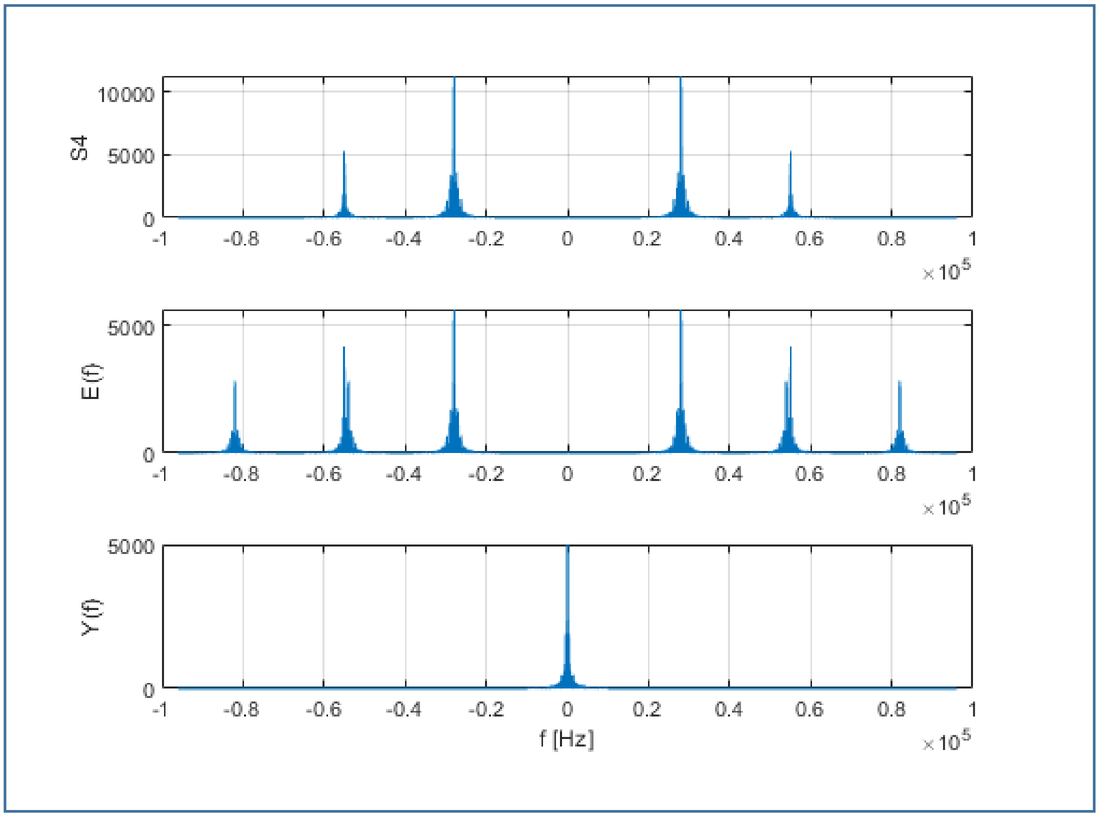
n) Describe the difference.

(1) **Before modulating and demodulating (sample freq), we cannot hear the audio, while the output result is hearable.**

(1) **figure(2) with one "Datatip" to show the width of the passband**



(2) **figure(1) including (311), (312) and (313)**



(3) Screenshot of Matlab code for Part I

lab4_1.m   lab4_2.m

```
[s4,fs]=audioread('m1a.wav'); % read the audio file and sample rate
s4=s4'; % transpose

% Modulation
t=[0:length(s4)-1]/fs; % time index
h_carrier_f = 55000;
c=cos(2*pi*h_carrier_f*t); % carrier frequency is 10 kHz (10e3 = 10000)
x=s4.*c; % x is the modulated signal
f=[-length(s4)/2:length(s4)/2-1]*fs/length(s4); % frequency index (from - fs/2 to fs/2)
% Demodulation
e=x.*c; % frequency shifting (back to the baseband)

N=16;

W_n = 2 * 10000 / fs;
[B1, A1] = butter(N, W_n);

y = filter(B1, A1, x);
[H1, fh] = freqz(B1, A1, 1e3, fs);

figure(1);
subplot(311); plot(f, abs(fftshift(fft(s4)))); ylabel('S4'); grid; % spectrum of baseband signal
subplot(312); plot(f, abs(fftshift(fft(e)))); ylabel('E(f)'); grid; % after frequency shifting
subplot(313); plot(f, abs(fftshift(fft(y)))); ylabel('Y(f)'); grid; % after lowpass filtering
xlabel("f [Hz]");

figure(2);
subplot(211); plot(fh, abs(H1)); axis([0 fs/2 0 1.2]); grid; ylabel('abs H1');

soundsc(s4, fs);
soundsc(y, fs);

```

Part II (9)

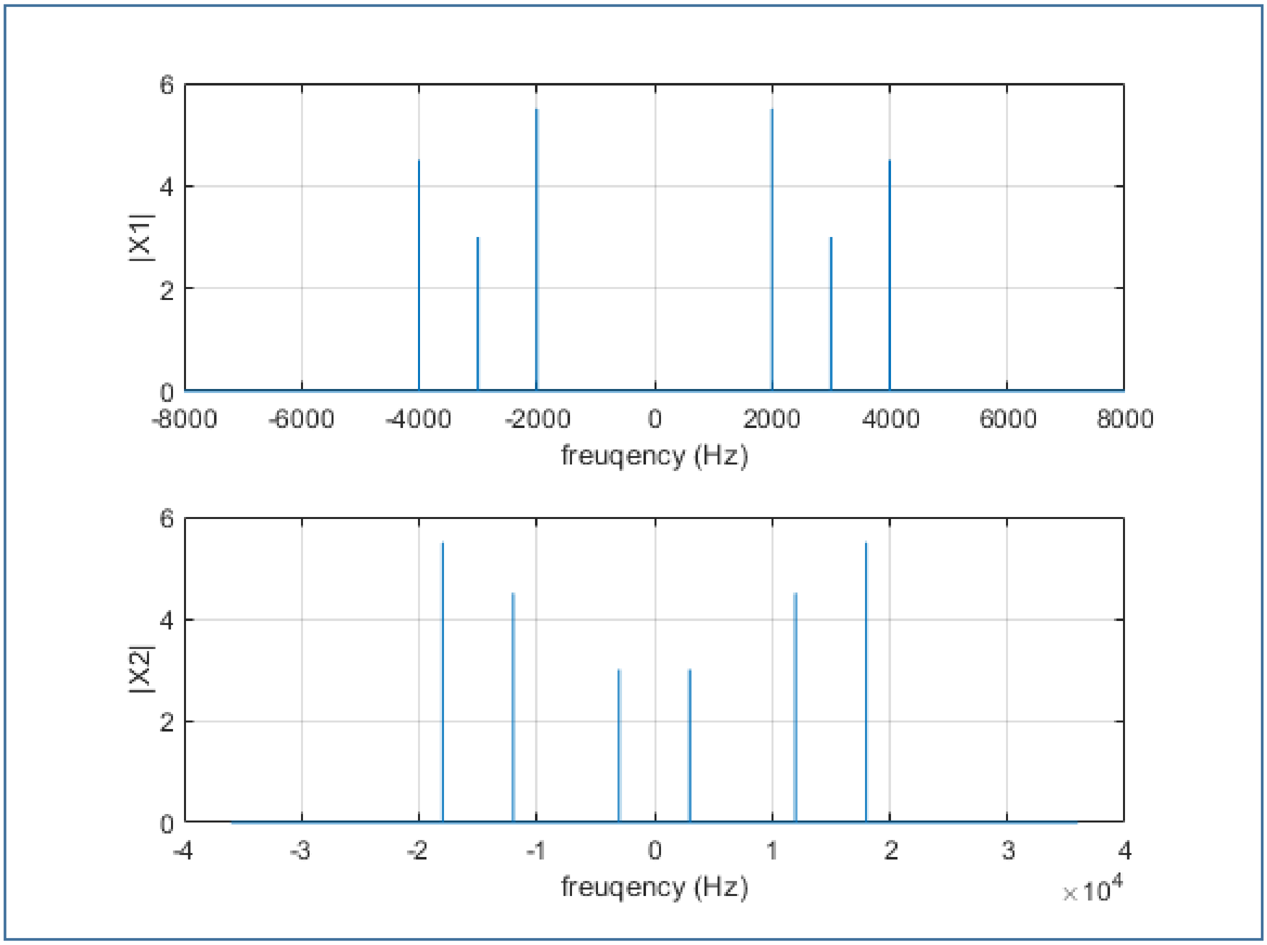
A CT signal is given as $x(t) = 6\cos(6\pi \times 10^3 t) + 9\cos(24\pi \times 10^3 t) + 11\cos(36\pi \times 10^3 t)$.

a) What is the unilateral bandwidth (fm in kHz) of $x(t)$?

(1) **Unilateral bandwidth = 18 kHz**

b) Define a DT sequence *x*1 if the sampling frequency (*f*s1) is 16 kHz and number of points is 14400.
c) Define a DT sequence *x*2 if the sampling frequency (*f*s2) is 72 kHz and number of points is 14400.
d) Define **actual** frequency index *f*1 according to the sampling frequency *f*s1.
e) Define **actual** frequency index *f*2 according to the sampling frequency *f*s2.
f) Use **"subplot"**, "**fft**", "**fftshift**" and "**abs**" to plot the magnitude spectrum of *X*1 versus *f*1 and the magnitude spectrum of X2 versus *f*2 in figure(3).

(1) **figure(3)**



g) Fill in the following tables by looking at the positive frequency axis.

(2)

Spectrum of $x_1[n]$	Frequency (Hz)	Magnitude
1st component	2000	5.5
2nd component	3000	3
3rd component	5000	4.5

Spectrum of $x_2[n]$	Frequency (Hz)	Magnitude
1st component	3000	3
2nd component	12000	4.5
3rd component	18000	5.5

h) Which spectrum ($X1$ or $X2$) is the correct spectrum of $x(t)$?

(1) **X2**

i) Explain your answer using **sampling theorem**.

(2) **The sampling theorem states that sampling frequency (fs) must be at least 2 times more than the highest frequency component so that Nyquist frequency is high enough to capture all relevant info in the signal. Only 2nd case works here, 72kHz > 2*18 kHz.**

(2) **Screenshot of Matlab code for Part II**

lab4_1.m | lab4_2.m

```
clear
N=14400; % number of points
n=0:N-1; % n index

fs1=16e3; % sampling frequency 1 (16 kHz)
x1 = 6*cos(6*pi*1000*n/fs1)+9*cos(24*pi*1000*n/fs1)+11*cos(36*pi*1000*n/fs1);
f1=[-N/2:N/2-1]*(fs1/N); % frequency index for x1

fs2=72e3; % sampling frequency 2 (72 kHz)
x2 = 6*cos(6*pi*1000*n/fs2)+9*cos(24*pi*1000*n/fs2)+11*cos(36*pi*1000*n/fs2); % x is sampled using fs2 = x2[n]
f2=[-N/2:N/2-1]*(fs2/N); % frequency index for x2

figure(3)
subplot(211); plot(f1, abs(fftshift(fft(x1))/length(x1))); % plot magnitude spectrum of x1
grid; ylabel('|X1|'); xlabel('freuqency (Hz)');

subplot(212); plot(f2, abs(fftshift(fft(x2))/length(x2))); % plot magnitude spectrum of x2
grid; ylabel('|X2|'); xlabel('freuqency (Hz)');
```