Modeling Frequency Division Multiplexing/DE-multiplexing

Lab # 09

Fall 2023

CSE-402L Digital Signal Processing Lab

Submitted by: **Ali Asghar**

Registration No.: **21PWCSE2059**

Class Section: **C**

"On my honor, as student of University of Engineering and Technology, I have neither given nor received unauthorized assistance on this academic work."

Submitted to:

Dr. Yasir Saleem Afridi

Date:

8th January 2023

Department of Computer Systems Engineering

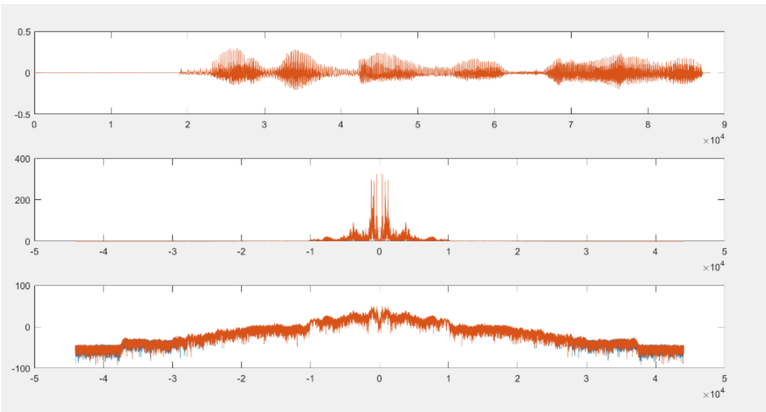
University of Engineering and Technology, Peshawar

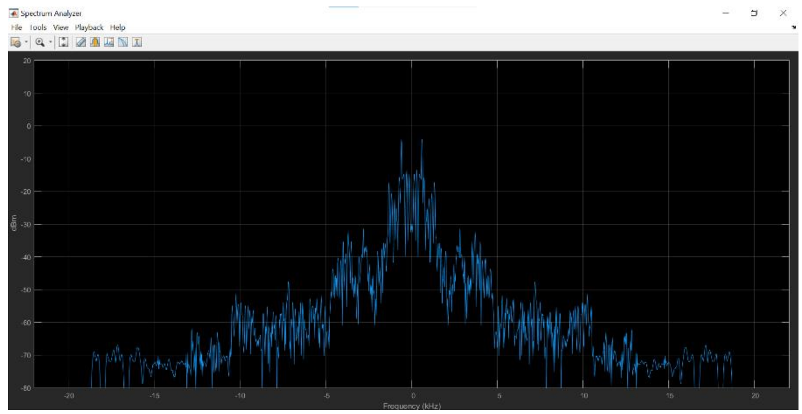
Demonstration of Concepts	**Poor (Does not meet expectation (1))** The student failed to demonstrate a clear understanding of the assignment concepts	**Fair (Meet Expectation (2-3))** The student demonstrated a clear understanding of some of the assignment concepts	**Good (Exceeds Expectation (4-5)** The student demonstrated a clear understanding of the assignment concepts	**Score** **30%**
Accuracy	The student completed (<50%) tasks and provided MATLAB code and/or Simulink models with errors. Outputs shown are not correct in form of graphs (no labels) and/or tables along with incorrect analysis or remarks.	The student completed partial tasks (50% - <90%) with accurate MATLAB code and/or Simulink models. Correct outputs are shown in form of graphs (without labels) and/or tables along with correct analysis or remarks.	The student completed all required tasks (90%-100%) with accurate MATLAB code and/or Simulink models. Correct outputs are shown in form of labeled graphs and/or tables along with correct analysis or remarks.	**30%**
Following Directions	The student clearly failed to follow the verbal and written instructions to successfully complete the lab	The student failed to follow the some of the verbal and written instructions to successfully complete all requirements of the lab	The student followed the verbal and written instructions to successfully complete requirements of the lab	**20%**
Time Utilization	The student failed to complete even part of the lab in the allotted amount of time	The student failed to complete the entire lab in the allotted amount of time	The student completed the lab in its entirety in the allotted amount of time	**20%**

Tasks:

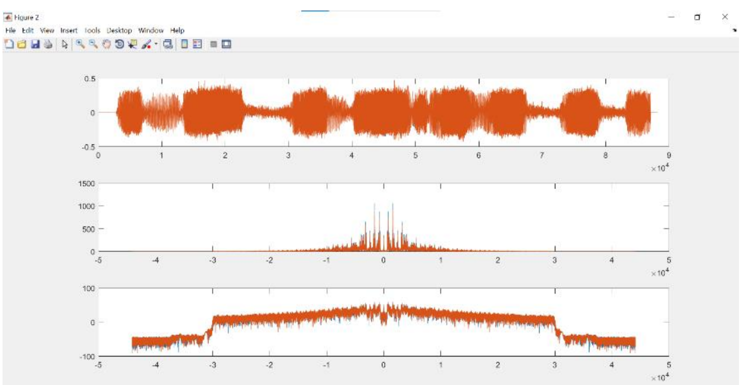
Step 1 & 2:

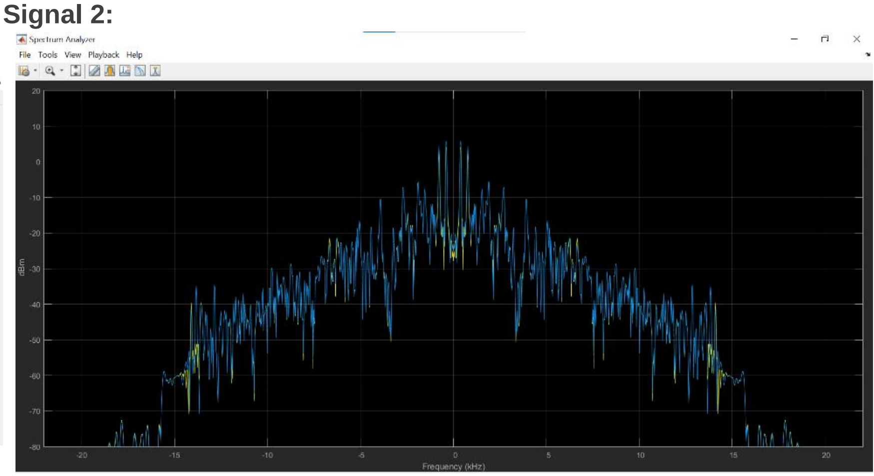
Signal 1:



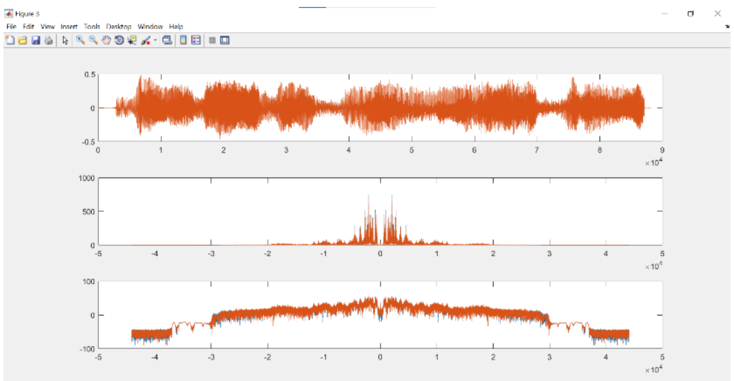


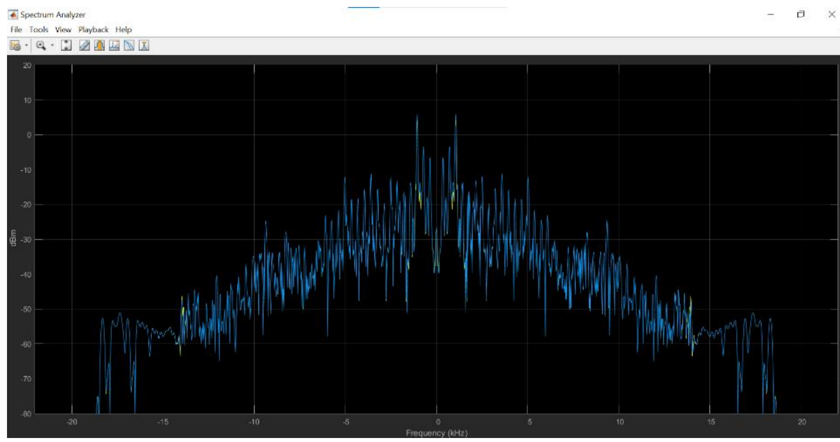
Signal 2:





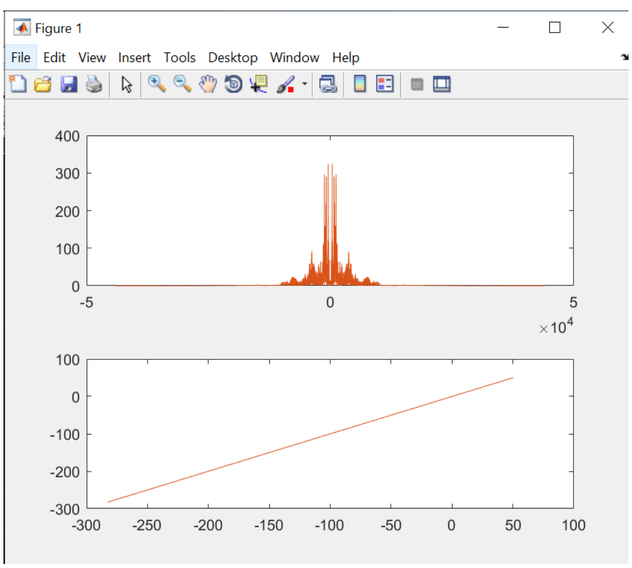
Signal 3:

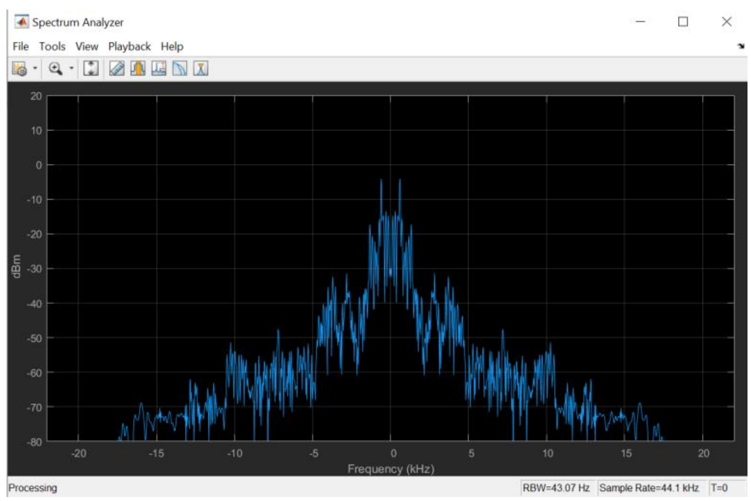




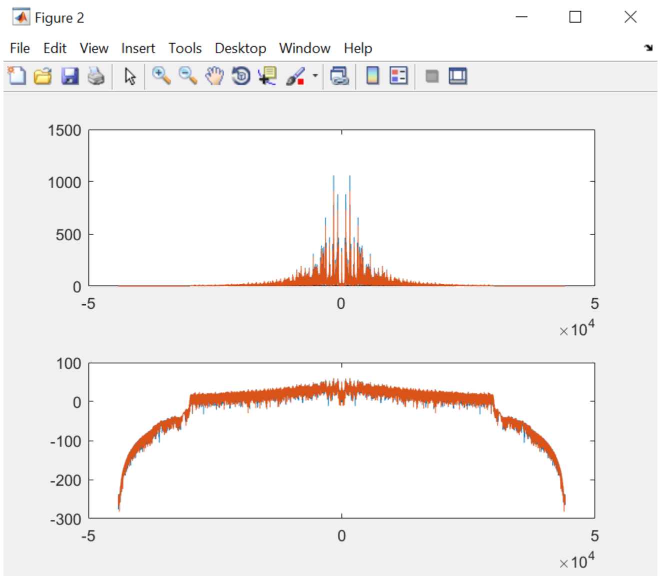
Step 3 & 4:

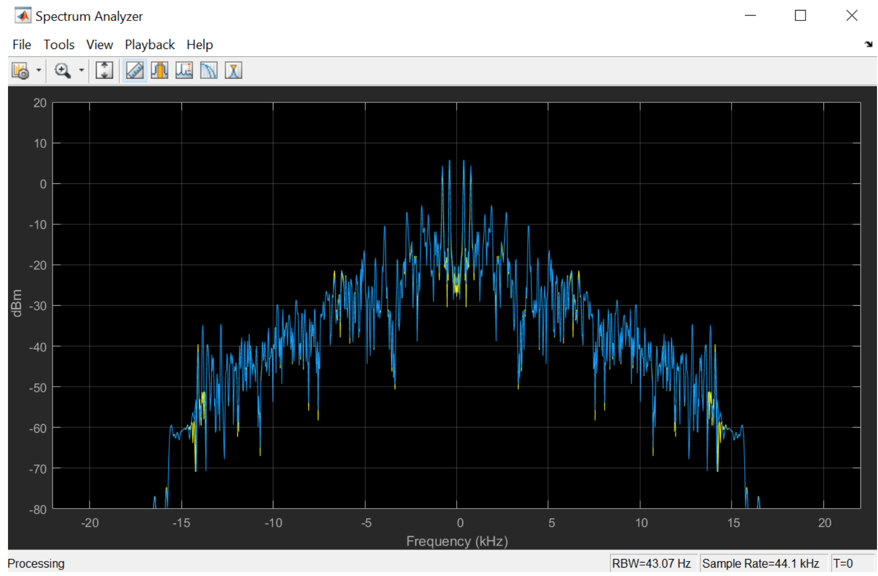
Signal 1:



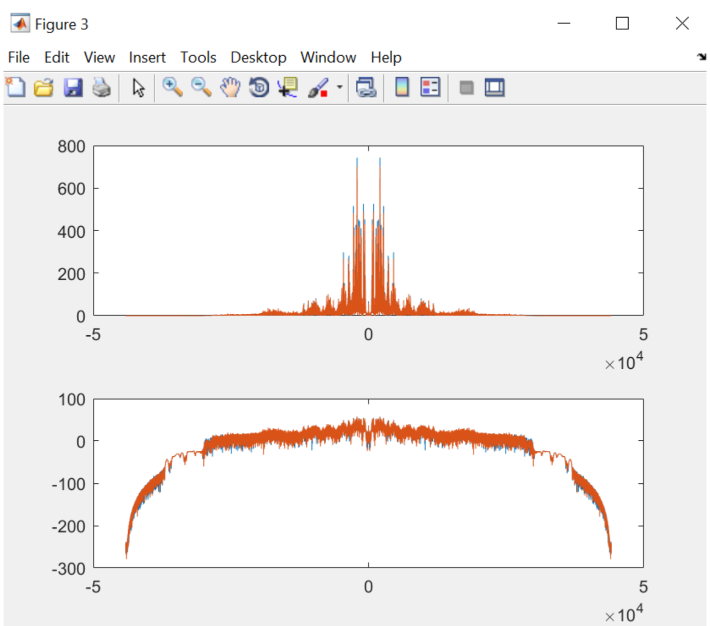


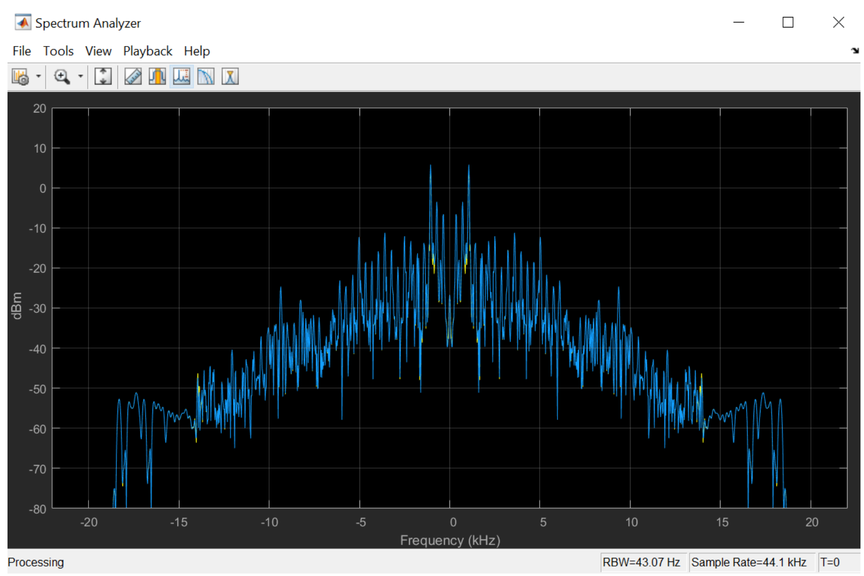
Signal 2:





Signal 3:

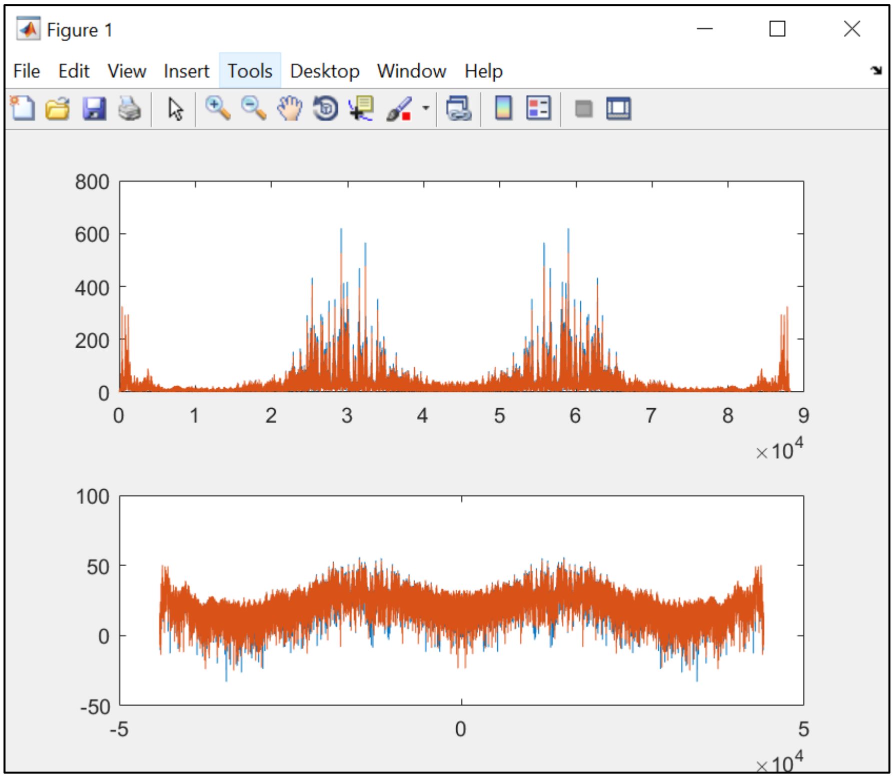




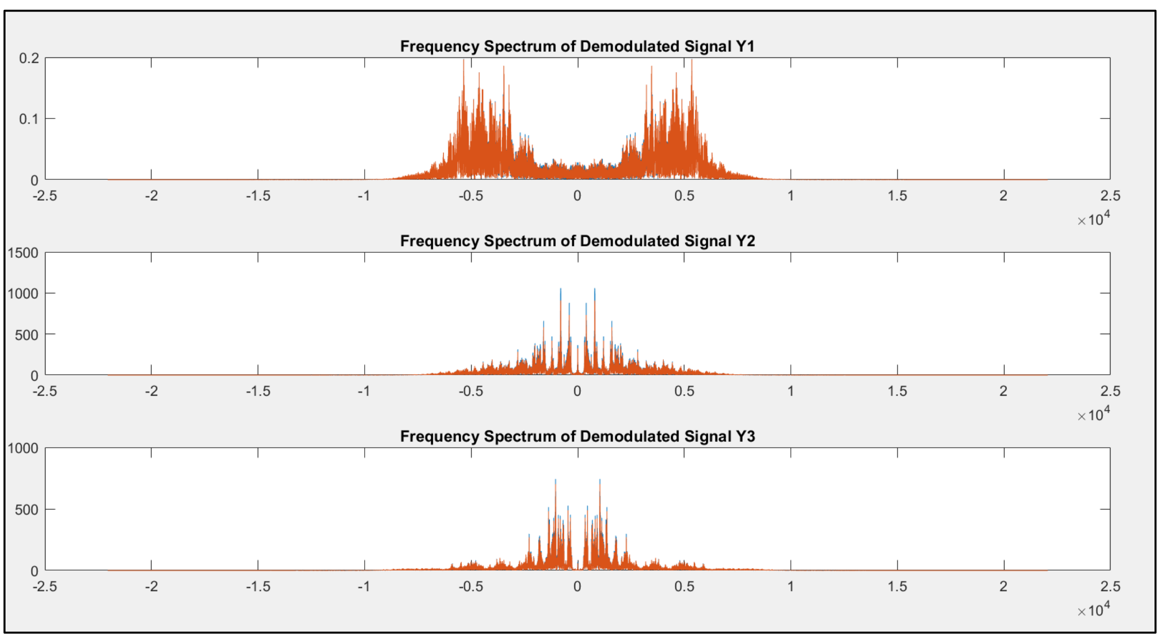
Listen the signals again and observe the difference in quality. Why?

Answer: Because we have attenuated some of the high frequency components(Mostly Causing Noise). That is the reason we’re hearing it differently.

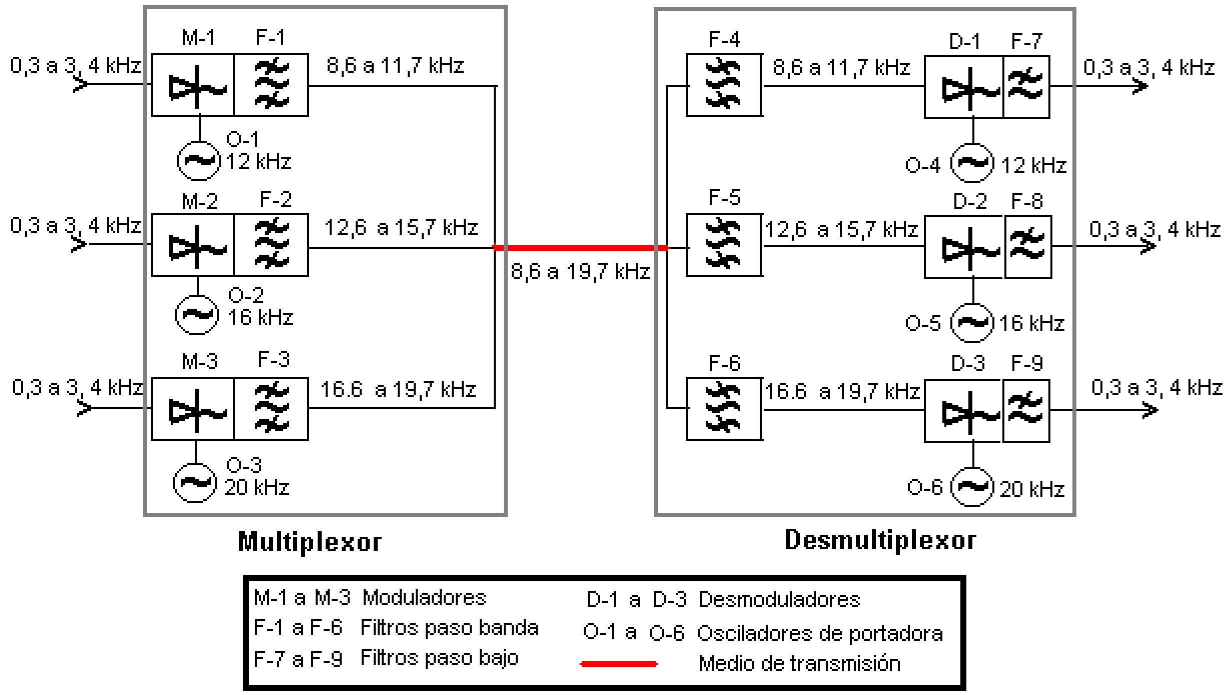
Step 5,6,7,8,9 & 10:



Modulated + Multiplexed Signals



Frequency Spectrum of Demodulated signals


M-1
F-1
0,3 a 3, 4 kHz
8,6 a 11,7 kHz
O-1
12 kHz
M-2
F-2
0,3 a 3, 4 kHz
12,6 a 15,7 kHz
O-2
16 kHz
M-3
F-3
0,3 a 3, 4 kHz
16.6 a 19,7 kHz
O-3
20 kHz
8,6 a 19,7 kHz
F-4
8,6 a 11,7 kHz
D-1
F-7
0,3 a 3, 4 kHz
O-4
12 kHz
F-5
12,6 a 15,7 kHz
D-2
F-8
0,3 a 3, 4 kHz
O-5
16 kHz
F-6
16.6 a 19,7 kHz
D-3
F-9
0,3 a 3, 4 kHz
O-6
20 kHz
Multiplexor
Desmultiplexor
M-1 a M-3 Moduladores
F-1 a F-6 Filtros paso banda
F-7 a F-9 Filtros paso bajo
D-1 a D-3 Desmoduladores
O-1 a O-6 Osciladores de portadora
Medio de transmisión

Code:

Lab9_Loading.m

```
%%
[Y1, Fs1] = audioread('recording1(Male).mp3');
[Y2, Fs2] = audioread('recording2(Female).mp3');
[Y3, Fs3] = audioread('recording3(Male).mp3');

%%
figure(1);
subplot(3,1,1);
plot(Y1);

Y1_w = fftshift(fft(Y1));
w1 = -length(Y1_w)/2 : (length(Y1_w) - 1)/2;

% Convert the magnitude of Y_w to dBm
Y1_w_dbm = mag2db(abs(Y1_w));

subplot(3,1,2);
plot(w1, abs(Y1_w));

subplot(3,1,3);
plot(w1, Y1_w_dbm);

hsa1 = dsp.SpectrumAnalyzer('SampleRate',Fs1);
step(hsa1, Y1);
%%
```

```
figure(2);
subplot(3,1,1);
plot(Y2);

Y2_w = fftshift(fft(Y2));
w2 = -length(Y2_w)/2 : (length(Y2_w) - 1)/2;

% Convert the magnitude of Y_w to dBm
Y2_w_dbm = mag2db(abs(Y2_w));

subplot(3,1,2);
plot(w2, abs(Y2_w));

subplot(3,1,3);
plot(w2, Y2_w_dbm);

hsa2 = dsp.SpectrumAnalyzer('SampleRate',Fs2);
step(hsa2, Y2);

%%
figure(3);
subplot(3,1,1);
plot(Y3);

Y3_w = fftshift(fft(Y3));
```

```
hsa2 = dsp.SpectrumAnalyzer('SampleRate',Fs2);
step(hsa2, Y2);

%%
figure(3);
subplot(3,1,1);
plot(Y3);

Y3_w = fftshift(fft(Y3));
w3 = -length(Y3_w)/2 : (length(Y3_w) - 1)/2;

% Convert the magnitude of Y_w to dBm
Y3_w_dbm = mag2db(abs(Y3_w));

subplot(3,1,2);
plot(w3, abs(Y3_w));

subplot(3,1,3);
plot(w3, Y3_w_dbm);

hsa3 = dsp.SpectrumAnalyzer('SampleRate',Fs3);
step(hsa3, Y3);

save('Data.mat');
```

Lab9_Filtering.m

```
load('Data.mat');

fc = 3000; % Cut off frequency
fs = 8000; % Sampling rate
[b,a] = butter(6,fc/(fs/2)); % Butterworth filter of order 6

Y1_LPF = filter(b, a, Y1);
Y2_LPF= filter(b, a, Y2);
Y3_LPF = filter(b, a, Y3);

%%
figure(1);
Y1_w_LPF = fftshift(fft(Y1_LPF));
w1_LPF = -length(Y1_w_LPF)/2 : (length(Y1_w_LPF) - 1)/2;

% Convert the magnitude of Y_w to dBm
Y1_w_dbm_LPF = mag2db(abs(Y1_w_LPF));

subplot(2,1,1);
plot(w1_LPF, abs(Y1_w_LPF));

subplot(2,1,2);
plot(Y1_w_dbm_LPF, Y1_w_dbm_LPF);

hsa1 = dsp.SpectrumAnalyzer('SampleRate',Fs1);
```

Lab8.m × task1.m × practice.m × modulation.m × dsp.m × Lab9_Loading.m × Lab9_Filtering.m × Lab9_Multiplexing.m × m_file.m ×

```matlab
hsa1 = dsp.SpectrumAnalyzer('SampleRate',Fs1);
step(hsa1, Y1_LPF);
%%
 figure(2);

 Y2_w_LPF = fftshift(fft(Y2_LPF));
 w2_LPF = -length(Y2_w_LPF)/2 : (length(Y2_w_LPF) - 1)/2;

 % Convert the magnitude of Y_w to dBm
 Y2_w_dbm_LPF = mag2db(abs(Y2_w_LPF));

 subplot(2,1,1);
 plot(w2_LPF, abs(Y2_w_LPF));

 subplot(2,1,2);
 plot(w2_LPF, Y2_w_dbm_LPF);

hsa2 = dsp.SpectrumAnalyzer('SampleRate',Fs2);
step(hsa2, Y2_LPF);

 %%
 figure(3);

 Y3_w_LPF = fftshift(fft(Y3_LPF));
 w3_LPF = -length(Y3_w_LPF)/2 : (length(Y3_w_LPF) - 1)/2;
```

Lab8.m × task1.m × practice.m × modulation.m × dsp.m × Lab9_Loading.m × Lab9_Filtering.m × Lab9_Multiplexing.m

```matlab

 % Convert the magnitude of Y_w to dBm
 Y3_w_dbm_LPF = mag2db(abs(Y3_w_LPF));

 subplot(2,1,1);
 plot(w3_LPF, abs(Y3_w_LPF));

 subplot(2,1,2);
 plot(w3_LPF, Y3_w_dbm_LPF);

hsa3 = dsp.SpectrumAnalyzer('SampleRate',Fs3);
step(hsa3, Y3);

%%
sound(Y1_LPF, Fs1);
pause(2);
sound(Y1, Fs1);

pause(3);

sound(Y2_LPF, Fs2);
pause(2);
sound(Y2, Fs2);

pause(3);
```

Lab9_Multiplexing.m

Lab8.m | task1.m | practice.m | modulation.m | dsp.m | Lab9_Loading.m | Lab9_Filtering.m | Lab9_Multiplexing.m | m_file.m

```matlab
load('Data.mat');

Fc1 = 15000;
Fc2 = 20000;
Fc3 = 25000;

Y1_Mod = ammod(Y1_LPF, Fc1, 30000);
Y2_Mod = ammod(Y2_LPF, Fc2, 3*Fs2);
Y3_Mod = ammod(Y3_LPF, Fc3, 3*Fs3);

Y_Mux = Y1_Mod + Y2_Mod + Y3_Mod;

Noise = awgn(Y_Mux, 10);

S1 = Y3_Mod + Y2_Mod;
demux_s1 = Y_Mux - S1;

S2 = Y3_Mod + Y1_Mod;
demux_s2 = Y_Mux - S2;

S3 = Y1_Mod + Y2_Mod;
demux_s3 = Y_Mux - S3;

%%
```

```matlab
pause(2);
sound(Y2, Fs2);

pause(3);

sound(Y2_LPF, Fs2);
pause(2);
sound(Y2, Fs2);

save('Data.mat');
```

Lab8.m | task1.m | practice.m | modulation.m | dsp.m | Lab9_Loading.m | Lab9_Filtering.m | Lab9_Multiplexing.m | m_file.m

```matlab
load('Data.mat');

Fc1 = 15000;
Fc2 = 20000;
Fc3 = 25000;

Y1_Mod = ammod(Y1_LPF, Fc1, 30000);
Y2_Mod = ammod(Y2_LPF, Fc2, 3*Fs2);
Y3_Mod = ammod(Y3_LPF, Fc3, 3*Fs3);

Y_Mux = Y1_Mod + Y2_Mod + Y3_Mod;

Noise = awgn(Y_Mux, 10);

S1 = Y3_Mod + Y2_Mod;
demux_s1 = Y_Mux - S1;

S2 = Y3_Mod + Y1_Mod;
demux_s2 = Y_Mux - S2;

S3 = Y1_Mod + Y2_Mod;
demux_s3 = Y_Mux - S3;

%%
```

Editor - D:\Uni\DSP Lab\Lab 09\Lab9_Multiplexing.m

Lab8.m | task1.m | practice.m | modulation.m | dsp.m | Lab9_Loading.m | Lab9_Filtering.m | Lab9_Multiplexing.m | m_file.m

```matlab
figure(1);
Y_Mux_w = fftshift(fft(Y_Mux));
w_Y_Mux = -length(Y_Mux_w)/2 : (length(Y_Mux_w) - 1)/2;

% Convert the magnitude of Y_w to dBm
Y_Mux_w_dbm_LPF = mag2db(abs(Y_Mux_w));

subplot(2,1,1);
plot(abs(Y_Mux_w));

subplot(2,1,2);
plot(w_Y_Mux, Y_Mux_w_dbm_LPF);

hsa1 = dsp.SpectrumAnalyzer('SampleRate',Fs1);
step(hsa1, Y_Mux_w);


%%
% Demodulate the signals
Y1_Demod = amdemod(demux_s1, Fc1, 3*Fs1);
Y2_Demod = amdemod(demux_s2, Fc2, 3*Fs2);
Y3_Demod = amdemod(demux_s3, Fc3, 3*Fs3);

% Compute FFT of the demodulated signals
```

```matlab
Y1_Demod_fft = fftshift(fft(Y1_Demod));
Y2_Demod_fft = fftshift(fft(Y2_Demod));
Y3_Demod_fft = fftshift(fft(Y3_Demod));

% Compute the frequency axis
f = (-Fs1/2 : Fs1/length(Y1_Demod) : Fs1/2 - Fs1/length(Y1_Demod));

% Plot the magnitude of the FFT of the demodulated signals
figure(3);
subplot(3,1,1);
plot(f, abs(Y1_Demod_fft));
title('Frequency Spectrum of Demodulated Signal Y1');

subplot(3,1,2);
plot(f, abs(Y2_Demod_fft));
title('Frequency Spectrum of Demodulated Signal Y2');

subplot(3,1,3);
plot(f, abs(Y3_Demod_fft));
title('Frequency Spectrum of Demodulated Signal Y3');

% Design a low-pass filter
lpf = designfilt('lowpassfir', 'PassbandFrequency', 0.45, 'StopbandFrequency', 0.55, 'PassbandRipple', 1, 'StopbandAttenuation', 60, 'DesignMethod'
```

Lab8.m task1.m practice.m modulation.m dsp.m Lab9_Loading.m Lab9_Filtering.m Lab9_Multiplexing.m m_file.m

```matlab

subplot(3,1,3);
plot(f, abs(Y3_Demod_fft));
title('Frequency Spectrum of Demodulated Signal Y3');

% Design a low-pass filter
lpf = designfilt('lowpassfir', 'PassbandFrequency', 0.45, 'StopbandFrequency', 0.55, 'PassbandRipple', 1, 'StopbandAttenuation', 60, 'DesignMethod',

% Apply the low-pass filter to the demodulated signals
Y1_Demod_filtered = filter(lpf, Y1_Demod);
Y2_Demod_filtered = filter(lpf, Y2_Demod);
Y3_Demod_filtered = filter(lpf, Y3_Demod);

%%
% Play the filtered signals
sound(Y1_Demod_filtered, Fs1);
pause(1);

sound(Y2_Demod_filtered, Fs2);
pause(1);

sound(Y3_Demod_filtered, Fs3);
```

Conclusion:

We modulated three speech signals and add noise to it to simulate the behavior of channel and demodulated the signal to obtain the original signal.