EEE 3208

Communication Theory Lab

Experiment No: 06

Experiment Name: Study of FDM (Multiplexer and Demultiplexer)

Submitted by,

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Semester: 2nd
Section: C2

Objective: The purpose of this experiment is to gain a clear understanding of how Frequency-Division Multiplexing (FDM) and demultiplexing work in theory and design and practically implement an FDM system, including both the multiplexer and demultiplexer.

1. Draw the block diagram of transmitter and receiver.

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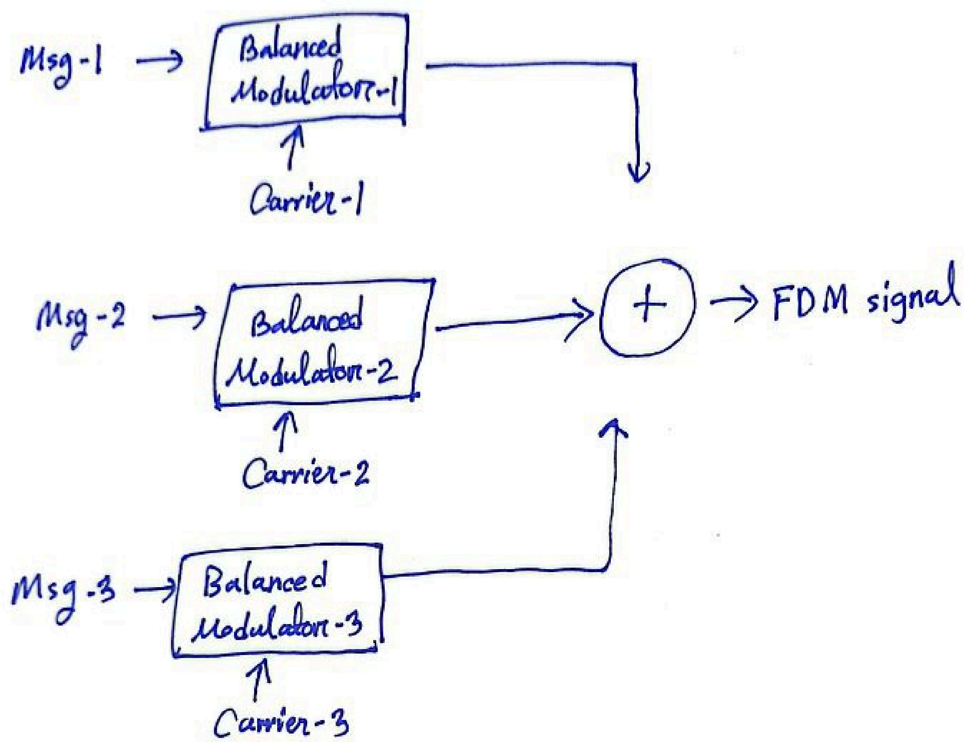
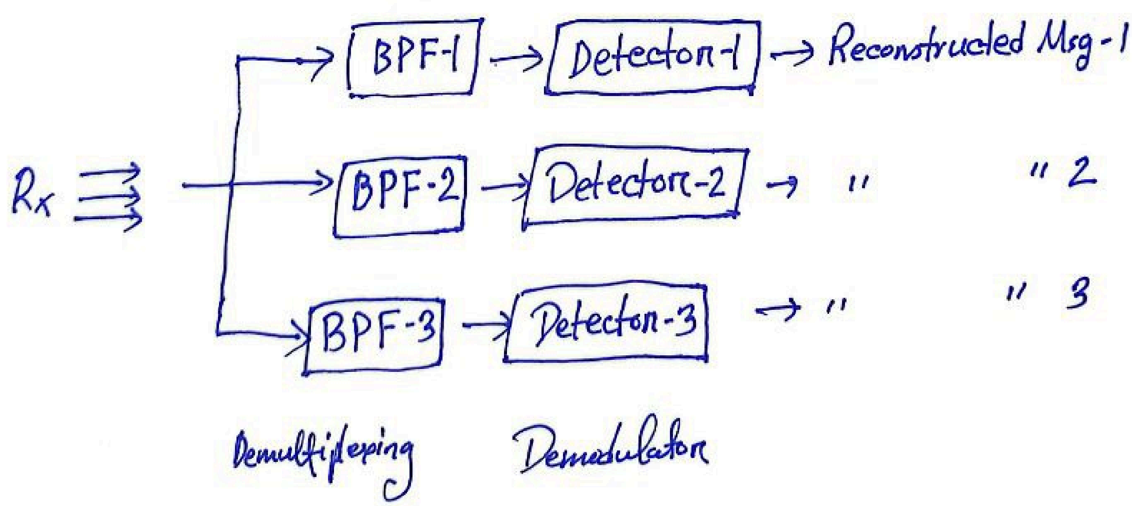
Transmitter block



Reciever block



2. Comment on the function of various blocks.

Multiplexer (FDM Transmitter) Blocks:
(a) Audio Signal Inputs:
They are actual message signals such as voice and audio, or any analog messages to transmit. All audio signals vary and require a specific carrier frequency added to them to prevent overlaps in terms of frequency.

(b) Balanced Modulator:
A balanced modulator mixes each audio signal with a carrier signal. It employs double sideband suppressed carrier (DSB-SC) modulation, generating two sidebands and no carrier portion. It conserves power and keeps carriers apart during transmission, not interfering with each other. Output of each balanced modulator is a modulated form of audio signal translated to a high range of frequencies.

(c) Carrier Signals:
Every audio signal is mixed with its individual carrier frequency. All carrier signals are continuous waves with disparate carrier frequencies for each modulated signal. With disparate carrier frequencies, we ensure each signal occupies its individual bandwidth in the composite output. Carrier frequencies have to be spaced apart such that no two collide with each other during transmission.

(d) Linearity Adder:
The output of many balanced modulers is fed into a linear adder, a summation circuit in its own right. All modulated signals combine in a linear adder and form a single composite signal with all individual modulated signals, each in its individual bandwidth. The FDM signal is then prepared for transmission via a channel such as coaxial cable, optical fiber, or radio waves (wireless channel).

Demultiplexer (FDM Receiver) Blocks:
(a) Band-Pass Filters (BPF):
The received FDM signal carries many modulated signals, each in its individual bandwidth. Band-pass filters (BPFs) pass one specific modulated signal and reject all others.
Each band-pass filter is tuned to a specific carrier frequency utilized in the stage of multiplexing. It ensures only the proper modulated signal progresses to the following stage, rejecting unwanted nearby channel signals.

(b) LPFs: Despite passing through a band-pass filter, the signals retain some high-frequency segments of the modulating process. To remove such high-frequency segments and restore audio signal to its original state, an LPF is utilized. LPF permits only baseband (audio) through and rejects high-frequency segments. LPF output is a cleaner form of the initial audio signal dispatched outwards.

(c) Audio Signal Outcomes: After passing through the LPF, recovered signals become identical to original audio signals having been subjected to multiplexing. Recovered signals can then proceed to audio speakers, recorders, and audio processing circuits for audio output and storing.

3. Draw the wave shapes at various test points:

Fig: Audio input signal 1 (yellow) , Audio input signal 2 (green)

Fig: Audio input signal 1 (yellow) , Audio input signal 3 (green)

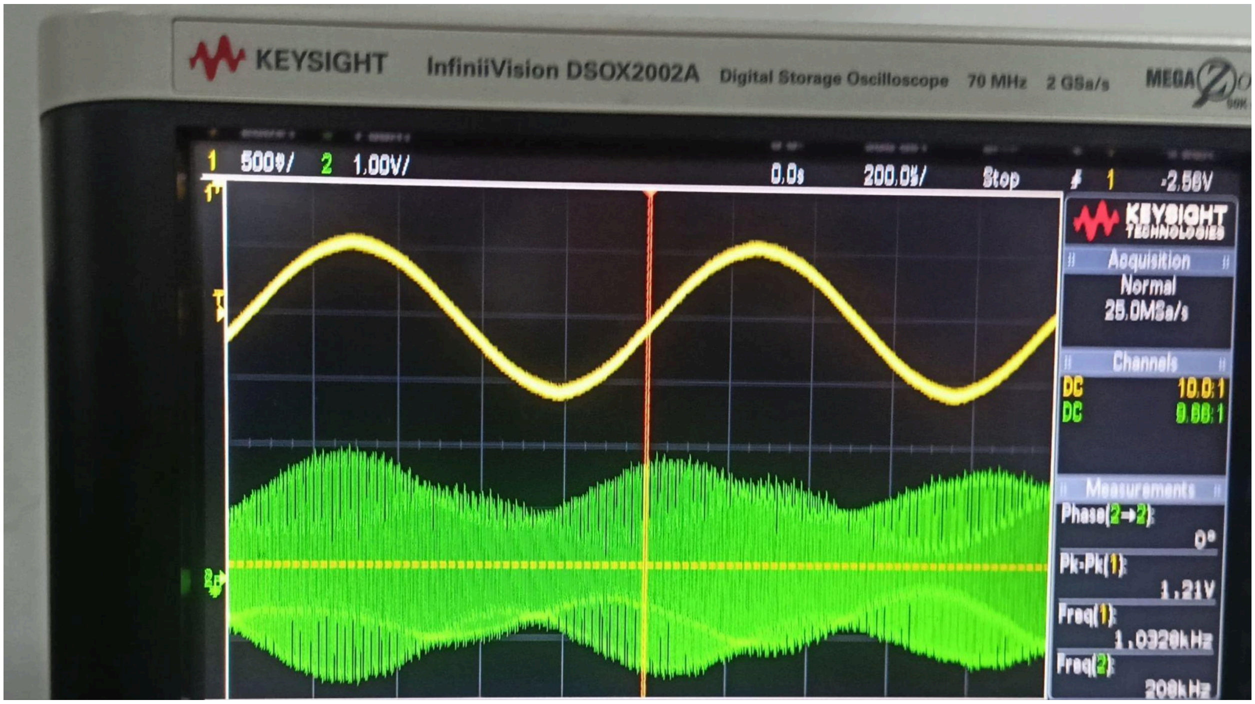


Fig: Audio input signal 1 (yellow) , FDM output signal (green)

Fig: Audio input signal 3 (yellow) , Reconstructed signal 3 (green) both have same 1.3kHz freq.

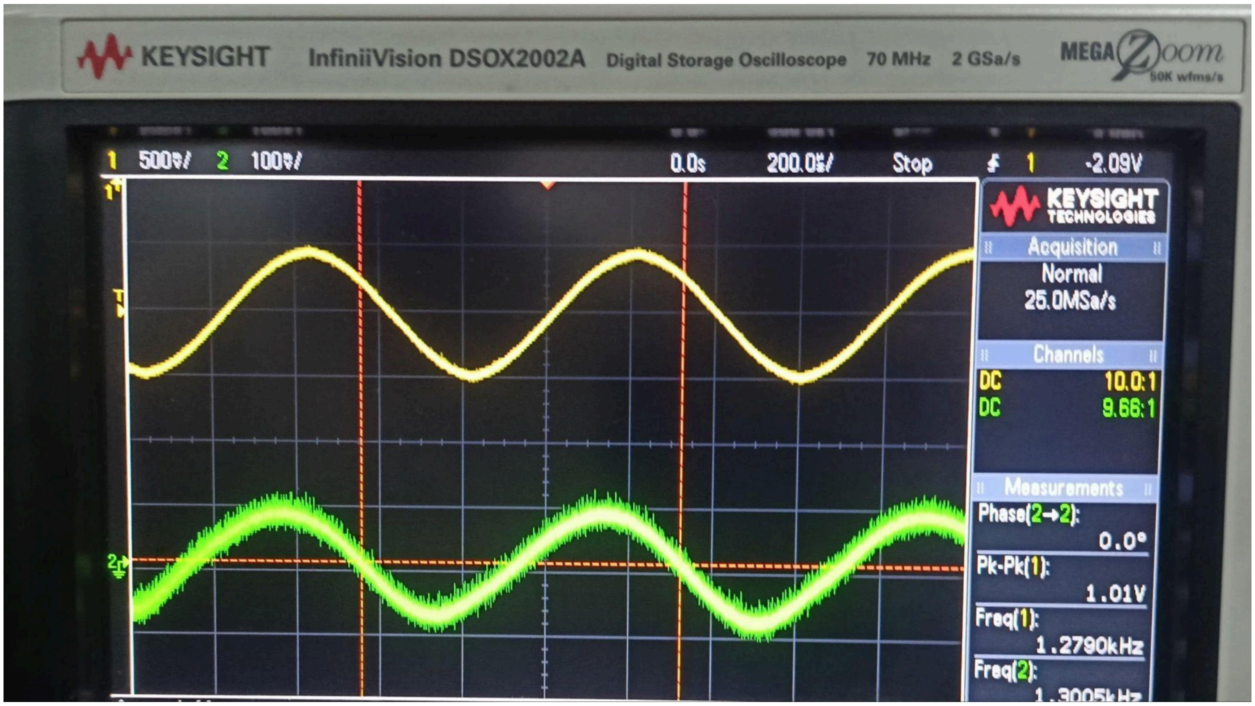


Fig: Audio input signal 2 (yellow) , Reconstructed signal 2 (green)

4. Draw the circuit diagram of audio signal generator, DSB-SC modulator by utilizing MC1496 and linearity adder. Comment on the operating principle of each circuit.

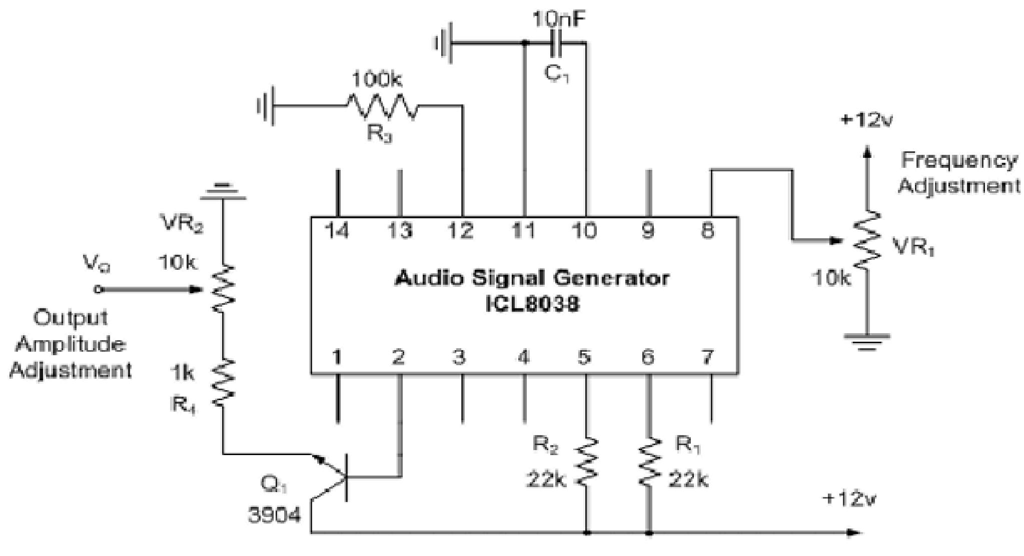


Fig. : Audio signal generator.

This circuit performs a DSB-SC modulation-a principal technique of FDM that allows the transport of multiple signals over a common communication channel. The MC1496 balanced modulator is employed for this type of modulation.

Carrier Signal Input: The high-frequency carrier signal is fed through C1 to pin 10, which is the DC blocking capacitor. It is further subjected to conditioning through resistors R3 and R4 to make it compatible at the input stage of the IC.
Audio (Message) Signal Input: The lower-frequency audio signal, the modulating signal, is applied to pins 1 and 4 via C2, R1, and VR1-adjustable resistor for signal scaling. These inputs are differential in nature and thus provide for balanced modulation.

Modulation Process: The MC1496 acts as a four-quadrant multiplier. It multiplies the modulating signal with the carrier signal. The output contains two frequency components:
The output contains two frequencies:
Carrier Suppression: Since the MC1496 is a balanced modulator, the carrier component at Wc is cancelled out, with only the two sidebands coming out of pin 6 in the output. The output undergoes further filtration and buffering before transmission.

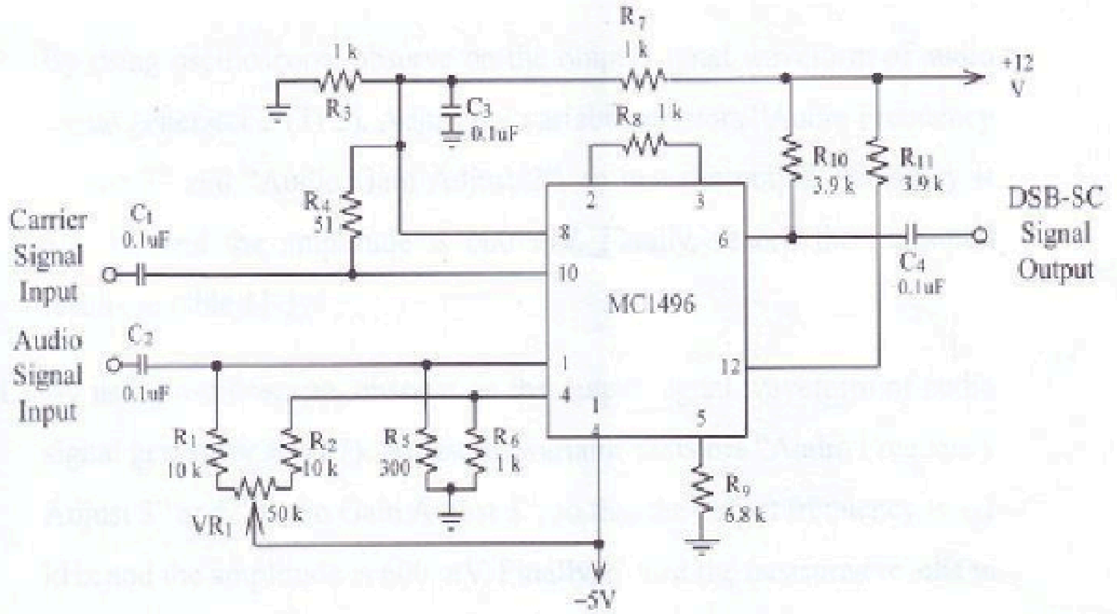


Fig. : Circuit diagram of DSB-SC modulation by utilizing MC 1496.

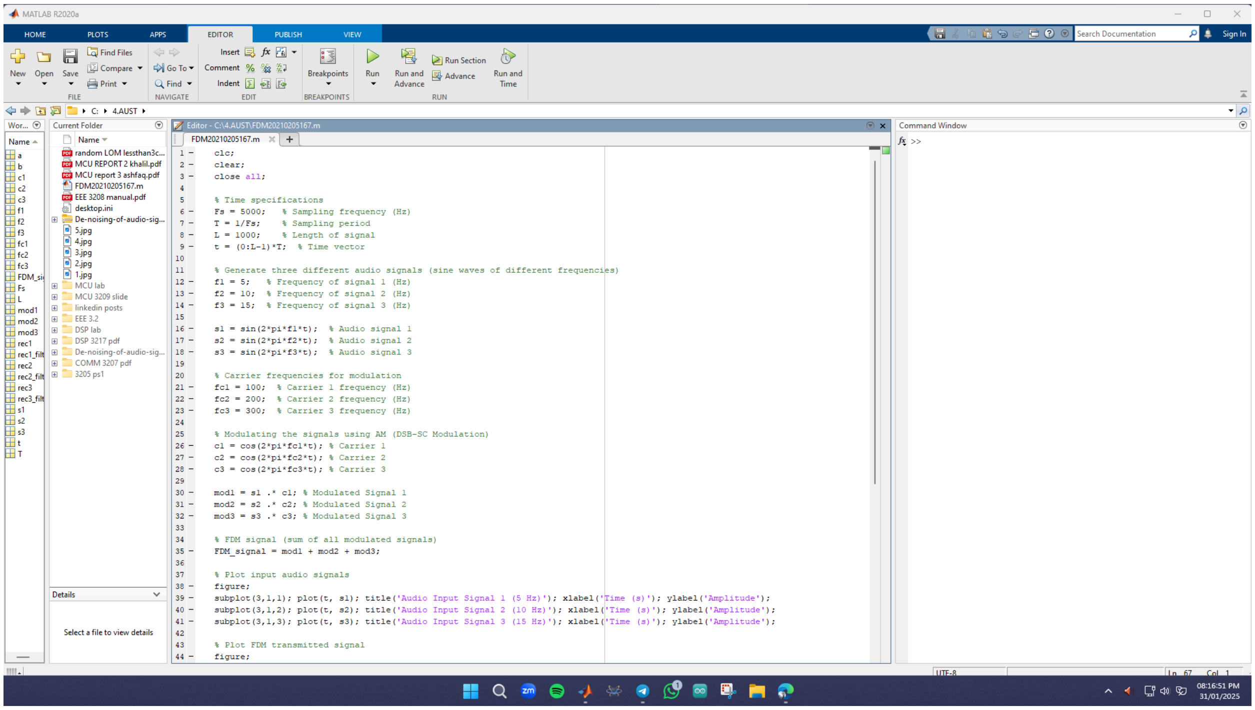
The ICL8038 Waveform Generator generates sine, square, and triangular waveforms; here, for the low frequency audio tone in FDM experiments, it provides the modulating signal.

Frequency Control: The output frequency is determined by VR1, a 10kΩ potentiometer that provides a variable resistance to the internal oscillator timing. The base frequency is set by capacitor C1 (10nF) and resistors R1/R2 (22kΩ each).

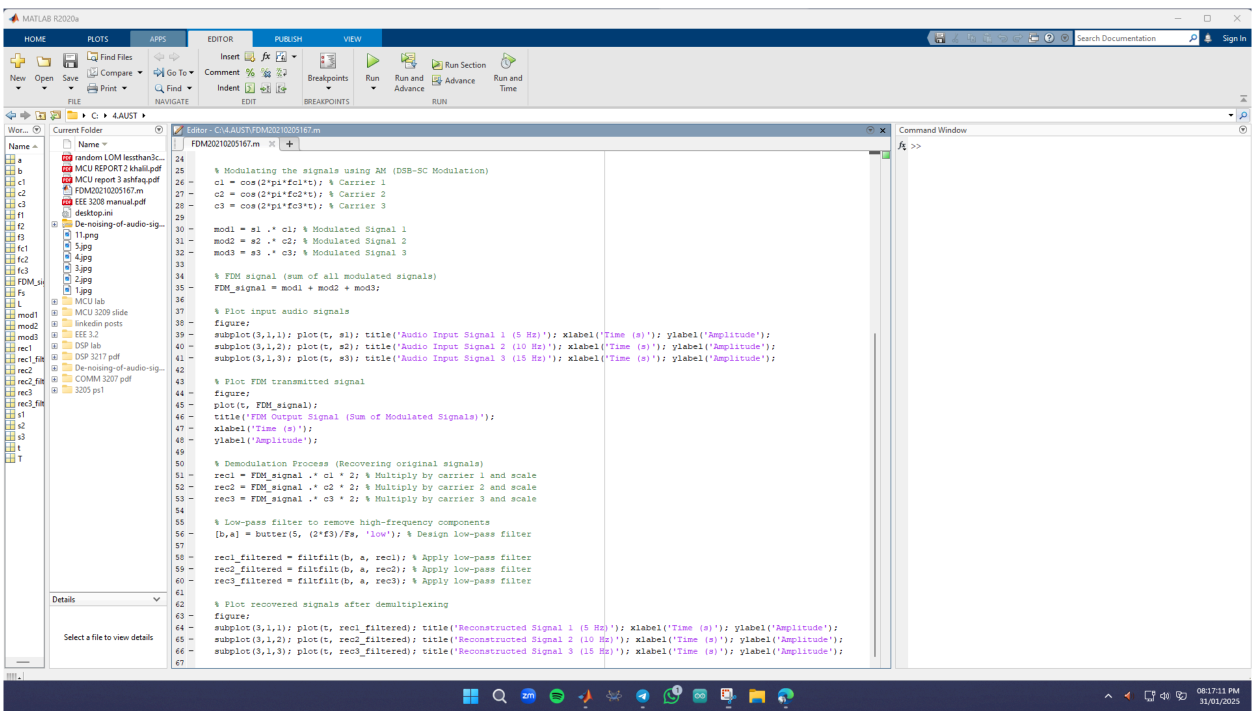
Amplitude Control: The output amplitude is controlled through VR2 (10kΩ) along with transistor Q1 acting as a variable gain stage (3904). In this way, the strength of the modulating signal can be finely adjusted before being fed to the DSB-SC modulator.

Waveform Generation: The ICL8038 provides a stable sine wave output that is to be used as the audio input signal in the MC1496 modulator circuit.

5. MATLAB CODE:



```matlab
clc;
clear;
close all;

% Time specifications
Fs = 5000;   % Sampling frequency (Hz)
T = 1/Fs;    % Sampling period
L = 1000;    % Length of signal
t = (0:L-1)*T;  % Time vector

% Generate three different audio signals (sine waves of different frequencies)
f1 = 5;   % Frequency of signal 1 (Hz)
f2 = 10;  % Frequency of signal 2 (Hz)
f3 = 15;  % Frequency of signal 3 (Hz)

s1 = sin(2*pi*f1*t);  % Audio signal 1
s2 = sin(2*pi*f2*t);  % Audio signal 2
s3 = sin(2*pi*f3*t);  % Audio signal 3

% Carrier frequencies for modulation
fc1 = 100;  % Carrier 1 frequency (Hz)
fc2 = 200;  % Carrier 2 frequency (Hz)
fc3 = 300;  % Carrier 3 frequency (Hz)

% Modulating the signals using AM (DSB-SC Modulation)
c1 = cos(2*pi*fc1*t); % Carrier 1
c2 = cos(2*pi*fc2*t); % Carrier 2
c3 = cos(2*pi*fc3*t); % Carrier 3

mod1 = s1 .* c1; % Modulated Signal 1
mod2 = s2 .* c2; % Modulated Signal 2
mod3 = s3 .* c3; % Modulated Signal 3

% FDM signal (sum of all modulated signals)
FDM_signal = mod1 + mod2 + mod3;

% Plot input audio signals
figure;
subplot(3,1,1); plot(t, s1); title('Audio Input Signal 1 (5 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');
subplot(3,1,2); plot(t, s2); title('Audio Input Signal 2 (10 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');
subplot(3,1,3); plot(t, s3); title('Audio Input Signal 3 (15 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');

% Plot FDM transmitted signal
figure;
```




```matlab

% Modulating the signals using AM (DSB-SC Modulation)
c1 = cos(2*pi*fc1*t); % Carrier 1
c2 = cos(2*pi*fc2*t); % Carrier 2
c3 = cos(2*pi*fc3*t); % Carrier 3

mod1 = s1 .* c1; % Modulated Signal 1
mod2 = s2 .* c2; % Modulated Signal 2
mod3 = s3 .* c3; % Modulated Signal 3

% FDM signal (sum of all modulated signals)
FDM_signal = mod1 + mod2 + mod3;

% Plot input audio signals
figure;
subplot(3,1,1); plot(t, s1); title('Audio Input Signal 1 (5 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');
subplot(3,1,2); plot(t, s2); title('Audio Input Signal 2 (10 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');
subplot(3,1,3); plot(t, s3); title('Audio Input Signal 3 (15 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');

% Plot FDM transmitted signal
figure;
```


```matlab
plot(t, FDM_signal);
title('FDM Output Signal (Sum of Modulated Signals)');
xlabel('Time (s)');
ylabel('Amplitude');

% Demodulation Process (Recovering original signals)
rec1 = FDM_signal .* c1 * 2; % Multiply by carrier 1 and scale
rec2 = FDM_signal .* c2 * 2; % Multiply by carrier 2 and scale
rec3 = FDM_signal .* c3 * 2; % Multiply by carrier 3 and scale

% Low-pass filter to remove high-frequency components
[b,a] = butter(5, (2*f3)/Fs, 'low'); % Design low-pass filter

rec1_filtered = filtfilt(b, a, rec1); % Apply low-pass filter
rec2_filtered = filtfilt(b, a, rec2); % Apply low-pass filter
rec3_filtered = filtfilt(b, a, rec3); % Apply low-pass filter

% Plot recovered signals after demultiplexing
figure;
subplot(3,1,1); plot(t, rec1_filtered); title('Reconstructed Signal 1 (5 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');
subplot(3,1,2); plot(t, rec2_filtered); title('Reconstructed Signal 2 (10 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');
subplot(3,1,3); plot(t, rec3_filtered); title('Reconstructed Signal 3 (15 Hz)'); xlabel('Time (s)'); ylabel('Amplitude');

```

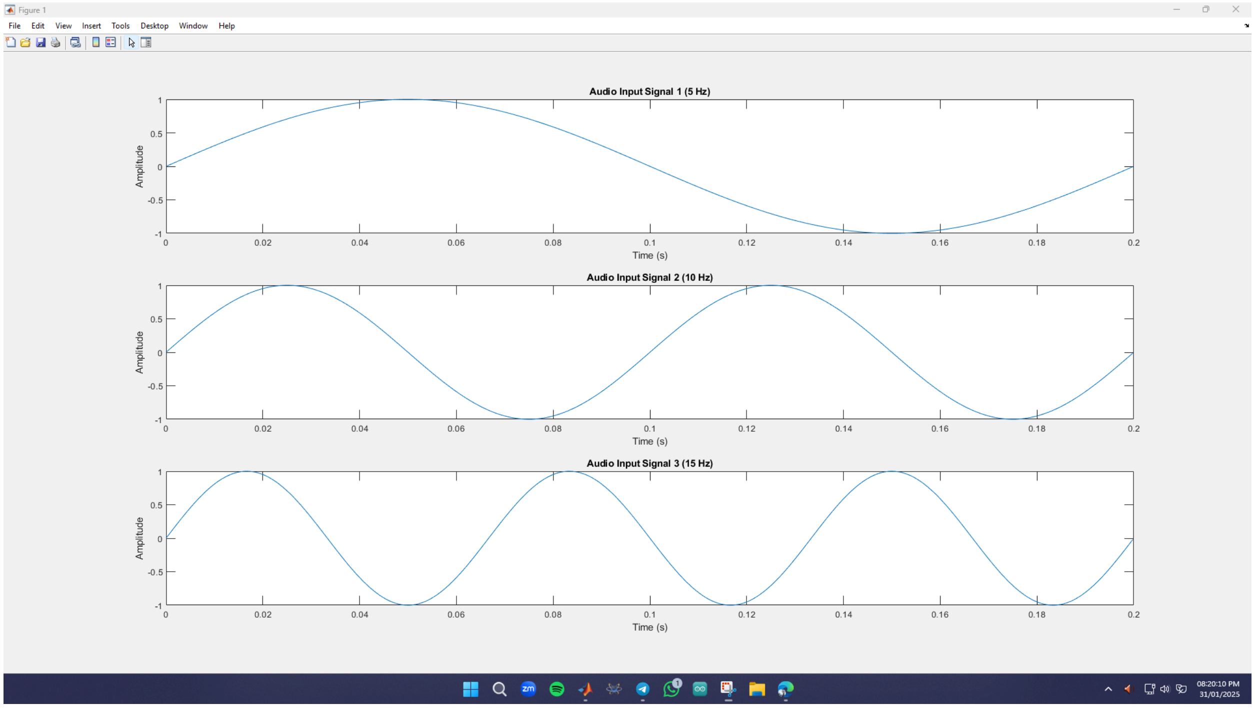


Figure 1
File Edit View Insert Tools Desktop Window Help
Audio Input Signal 1 (5 Hz)
Amplitude
Time (s)
Audio Input Signal 2 (10 Hz)
Amplitude
Time (s)
Audio Input Signal 3 (15 Hz)
Amplitude
Time (s)
08:20:10 PM
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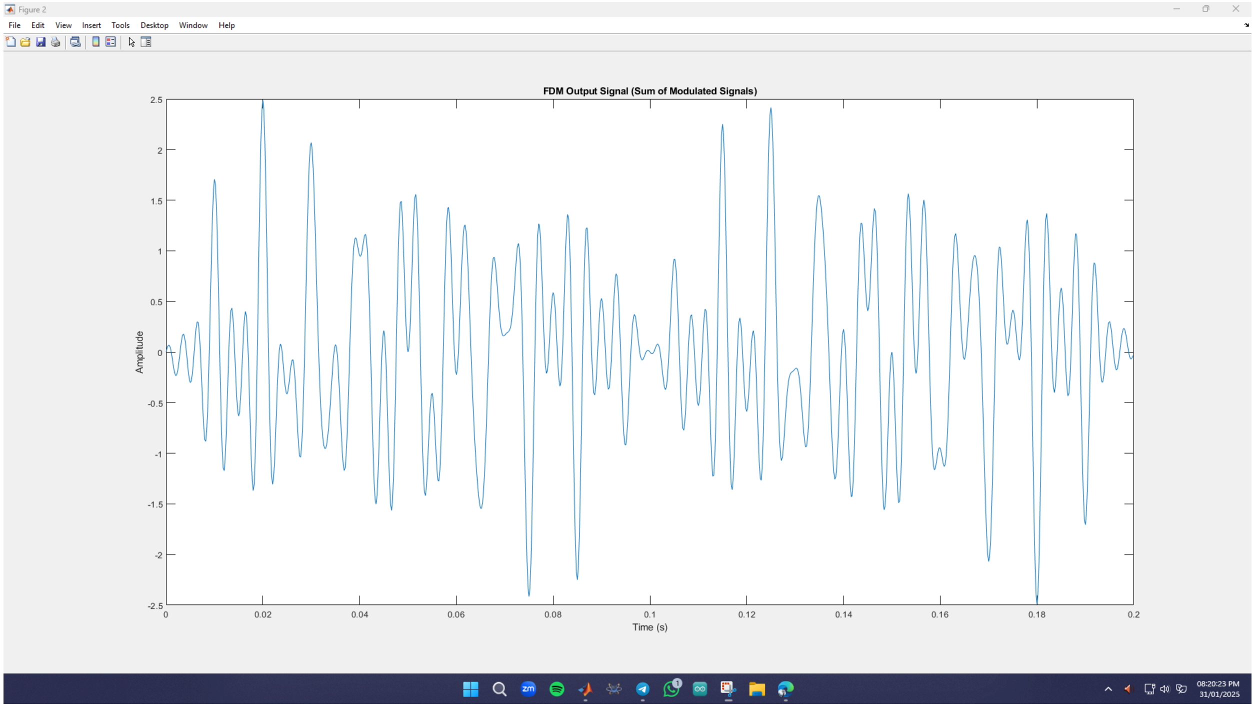
Figure 2
File Edit View Insert Tools Desktop Window Help
FDM Output Signal (Sum of Modulated Signals)
Amplitude
Time (s)
08:20:23 PM
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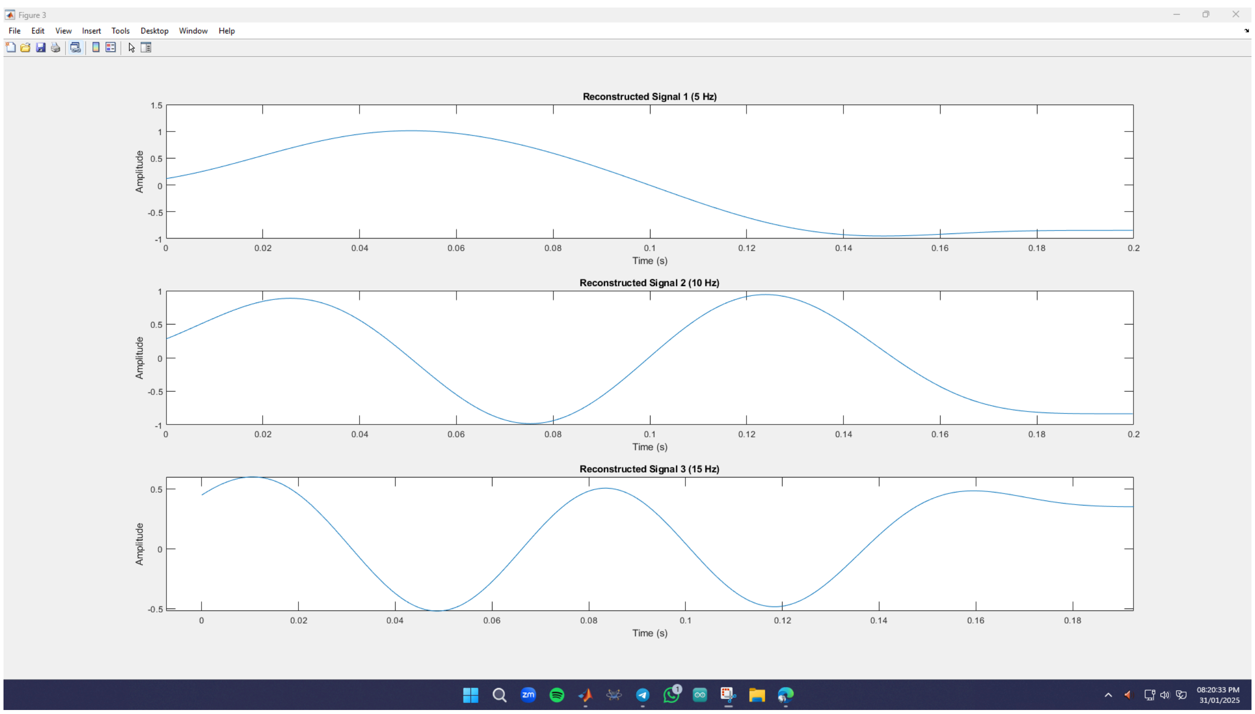
Figure 3
File Edit View Insert Tools Desktop Window Help
Reconstructed Signal 1 (5 Hz)
Reconstructed Signal 2 (10 Hz)
Reconstructed Signal 3 (15 Hz)
Amplitude
Time (s)
08:20:33 PM
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Discussion:

The FDM system is a simple method for sending numerous signals via a single channel. Several signals are placed onto a variety of carriers through a multiplexer (MUX), and a receiver separates them with specific filters through a demultiplexer (DEMUX). Only frequency is devided here. We used Amplitude modulation (DSB-SC) here because suppressed carrier frequencies consume less power, faster, better SNR, reduced noise and easier to operate. Input signals must be different in frequencies so that they don't overlap. We used operational amplifier to add the input signals. We used guard band so the signals remain reconstructable and don't overlap. Bandpass filter is used to seperate the original signal and it is designed based on the centre frequency of Fc. In demodulation we used Synchronous (Product) detector. The process is called Homodyne process because receiver and sending end carriers are same. After Mixer output we get a frequency component in 0 and other in 2Fc1 and we take the 0 one because working on high frequency requires complex circuit. We used DC blocker capacitor so that it can block the operational amplifier bias voltage to mix in the reconstructed signal.

Strengths of FDM:

Efficient bandwidth use: Several signals can simply use one channel.

Simultaneous conversation: Enables numerous individuals to converse at a single time.

Simple for analog systems: Serves analog signals well in contrast with digital multiplexing.

Disadvantages of FDM:

Problems with interference: In case carriers have a narrow bandwidth, noise and distortion can occur.

High bandwidth requirements: Needs a lot of bandwidth for sending several signals.

Synchronization problem: Assigning proper carrier frequencies is important such that signals don't interfere with one another.

Applications of FDM: Radio broadcasting: Various radio channels transmit signals through several channels.

Telephony: Applied in analog phone networks for numerous calls over a single channel.
Television Transmission: Several television channels transmit signals through a single channel