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Expt.Name : Spectrum Analyser and observe Spectrum

Aim:- To Study Spectrum Analyzer and observe the Spectrum of Sinusoidal Signal and Square wave.

Apparatus:-

(i) Spectrum Analyzer (9KHZ - 3GHZ)

(ii) Function

Theory :- A Spectrum Analyser is a laboratory instrument that displays signal amplitude (Strength) as it varies by signal frequency. The frequency appears on the horizontal axis, and the amplitude is displayed on the vertical axis. To the casual observer, a spectrum analyze looks like an oscilloscope and in fact, some labs instruments can function as oscilloscope/ Spectrum analyzer.

A Spectrum analyzer can be used to determine whether (or) not a wireless transmitter is working according to federally defined Standards for Purity of emissions. output signals at frequencies other than the intended communications frequency appear as vertical lines up (Pips) on display. A analyzer can also be used to determine, by direct observation, the bandwidth of digital/ analog signal.

A Spectrum analyzer interface is a device that can be connected to a wireless receiver or Personal Computer to allow virtual detection and analysis of frequencies.

Features of Lab Instrument GSP-830 (Gwinstek)

* 5 markers with delta markers and Peak functions.
* 3 traces
* Split windows with seperate settings
* 6.4" TFT Color LCD, 640x480 resolution.
* AC/DC/ battery multi-mode Power operation.

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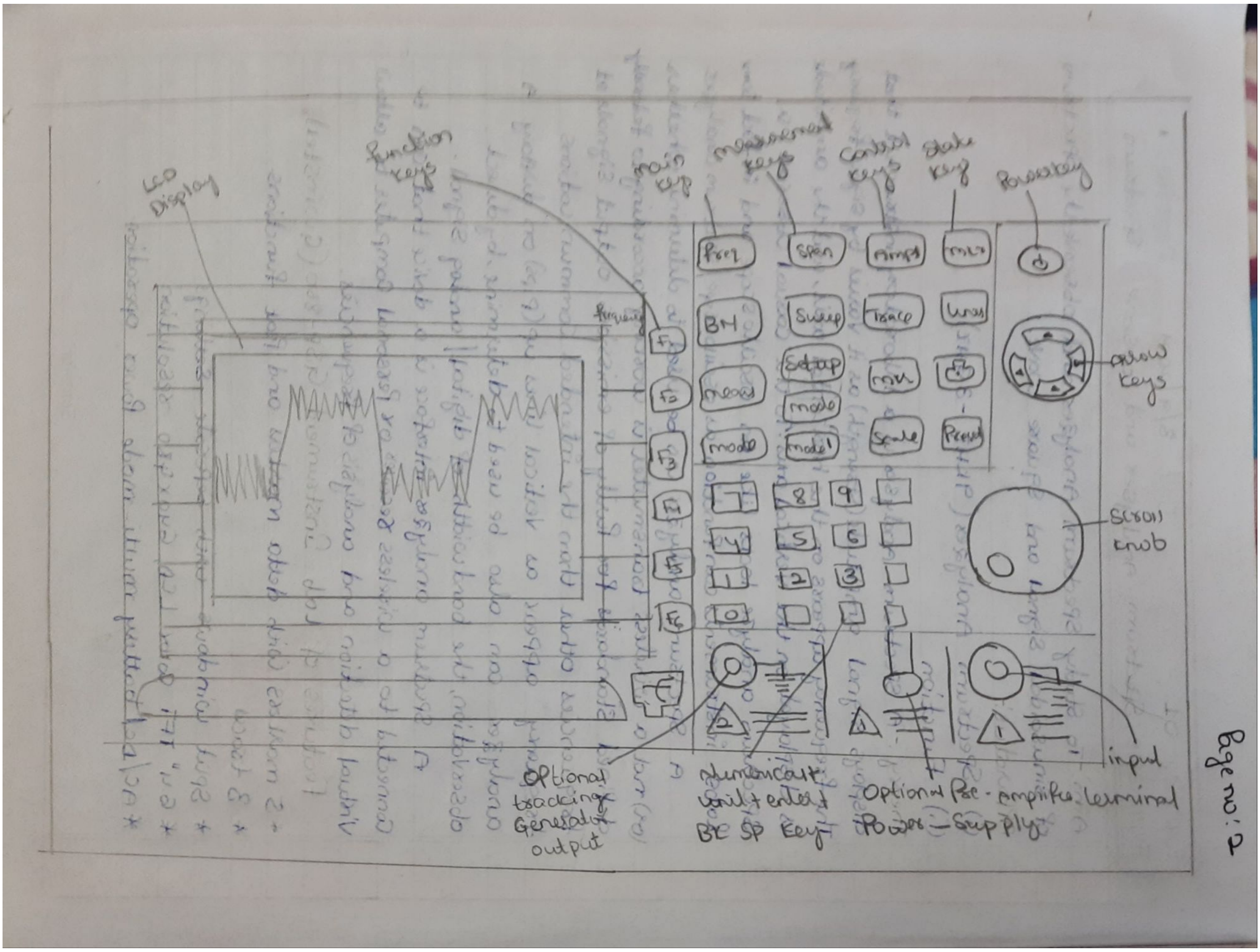
LCD Display
function keys
main key
measurement keys
control keys
state key
Power key
Freq
Span
Ampt
BW
Sweep
Trace
Meas
Setup
Mode
Scale
Preset
arrow keys
Scroll knob
input
Optional tracking Generator output
numeric & unit enter BK SP Key
Optional Pre-amplifier terminal Power Supply
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* Auto Set
* 9KHZ - 3GHZ frequency range

1) Frequency
* Frequency / Span :- Sets the frequency scale
* View signal :- defines center frequency and left / right bandwidth
* Setting frequency adjustment step :- arrow keys resolution for center, start and stop frequency

Panel operation :-
→ Press frequency keys
→ Press F_4 (step)
→ enter value using keys

2) Range :- 9KHZ to 3GHZ

3) Set center frequency :-
Panel operations :-
* Press frequency key
* Press F_1 (center)
* enter value using keys

4) Set Frequency span :-
Panel operations :-
→ Press Span key
→ Press F_1 (Span)
→ enter value using keys.

5) View Signal (Start & Stop)
→ Start and Stop define begining & end of frequency range
→ Arrow keys & Scroll keys makes resolution : 1/10.

6) Set Start Frequency
Panel operations :-

Panel operation:-

* Press frequency key
* Press F2(start)
* enter value using keys.

7) Set Stop frequency:-

Panel operation:-

* Press frequency key
* Press F2 (Stop)
* enter value using keys.

8) Full (or) zero Span:-

* full (or) Zero span sets span to extreme. Values:- 3GHZ(full) or 0KHZ(Zero).

9) Display Full Frequency Span:-

Panel operation:-

* Press Span key.
* Press F2 key
* Range : 3GHZ
* Center frequency : 1.5GHZ ; Start frequency : 0KHZ
Stop frequency : 3GHZ

10) zero Span display

* Press F3 key
* Start frequency = Stop frequency = Center frequency.

Amplitude Selection and Setting method.

1) Amplitude:- Amplitude key acts vertical attribute of display including upper limit, vertical range, vertical limit and to external gain or loss.

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3) Set Vertical Scale:-
* Vertical display scale is defined by reference amplitude range, measurement unit and external gain/loss
3) Set Vertical Reference ~~Scale~~ Amplitude:-
* Panel operation
* Press amplitude key
* Press F_1
* enter value using keys and arrows.
Range:
→ dBm - 110 to 20dBm, 0.1dB resolution
→ dBmV - 63.1 to 66.99 dBmV, 0.01dB resolution
→ dBμV - 3.01 to 126.99dBμV, 0.01dB resolution.
4) Select Amplitude Scale:-
Panel operation:-
* Press amplitude key
* Press F_2 (Scale dB/Div)
* Repeatedly to select scale
Range: 10, 5, 2, 1 dB/Div
* Press F_3 and select, Press until from F_1(dBm), F_2(dBmV), F_3(dBμV)
* F_6 to go to Previous menu.
5) Background:-
* external offset compensates amplitude gain/loss caused by an external network or device Panel operation.
* Press amplitude key.
* Press F_4
* enter value using keys ~~and~~ arrows and scrolls
Range:- -20dB to +20dB, 0.1dB resolution.

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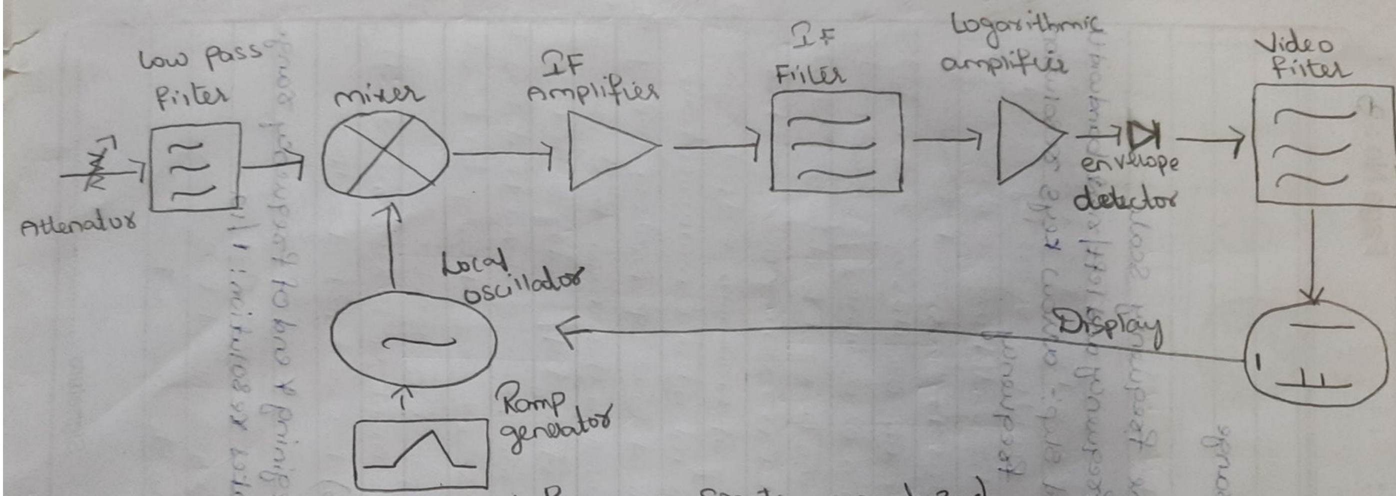
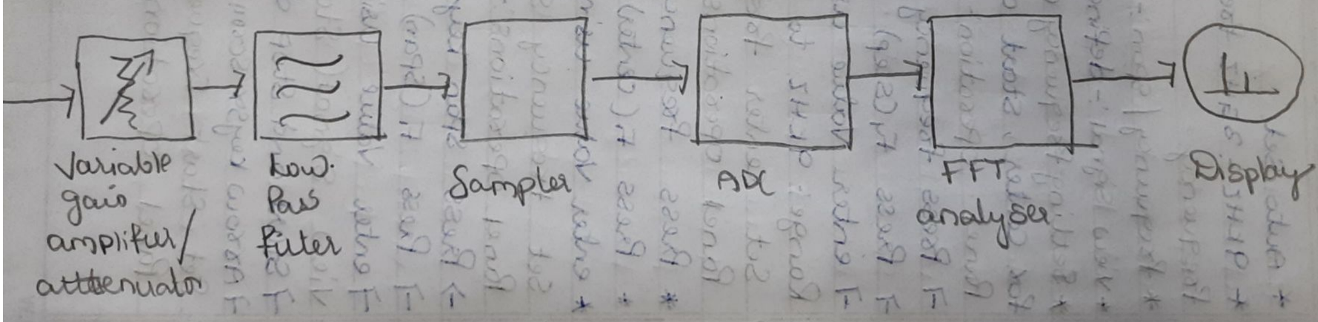
Block Diagram:-

(5)



(Superheterodyne or Swept frequency Spectrum analyzer)
Block diagram



Icon:-

* The amplitude icon appears at bottom of display when the external offset changes. To check whether spectrum analyzer working properly. Press system key, Press auxilary signal, select an option from side given menu following signal will generates 10MHZ signal with 10dB amplitude

Observation:-

waveform:- Sine

S.no	frequency(KHZ)	Amplitude(mv)
1)	2	1
2)	2.5	1.1
3)	3	1.5
4)	4	2
5)	5	2.4

Waveform:- Square

S.no	frequency(KHZ)	Amplitude(mv)
1)	2	1
2)	2.5	1.2
3)	3	1.5
4)	4	1.6
5)	5	2

Conclusion:- Hence, Successfully verified and analyzed the spectrum of sine and square wave form for different frequency and amplitude.

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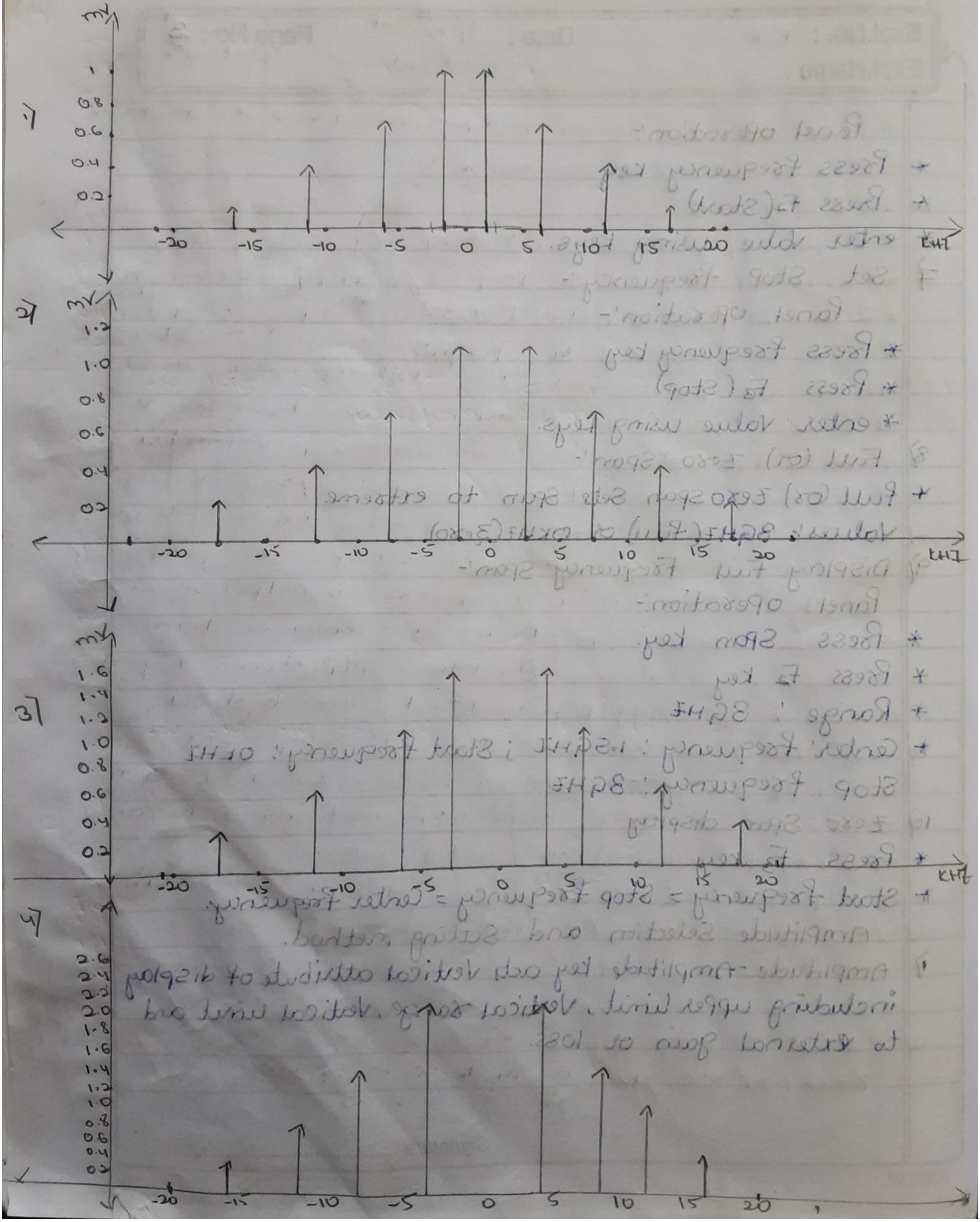
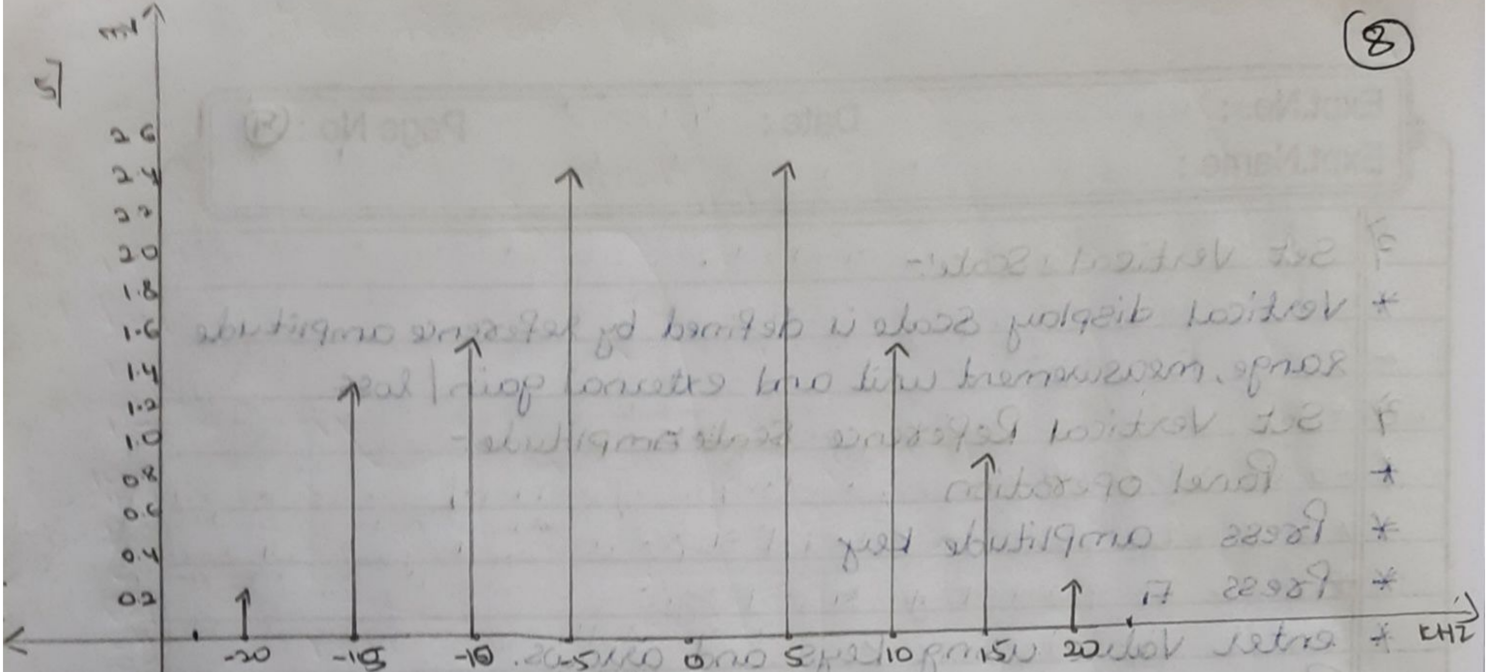
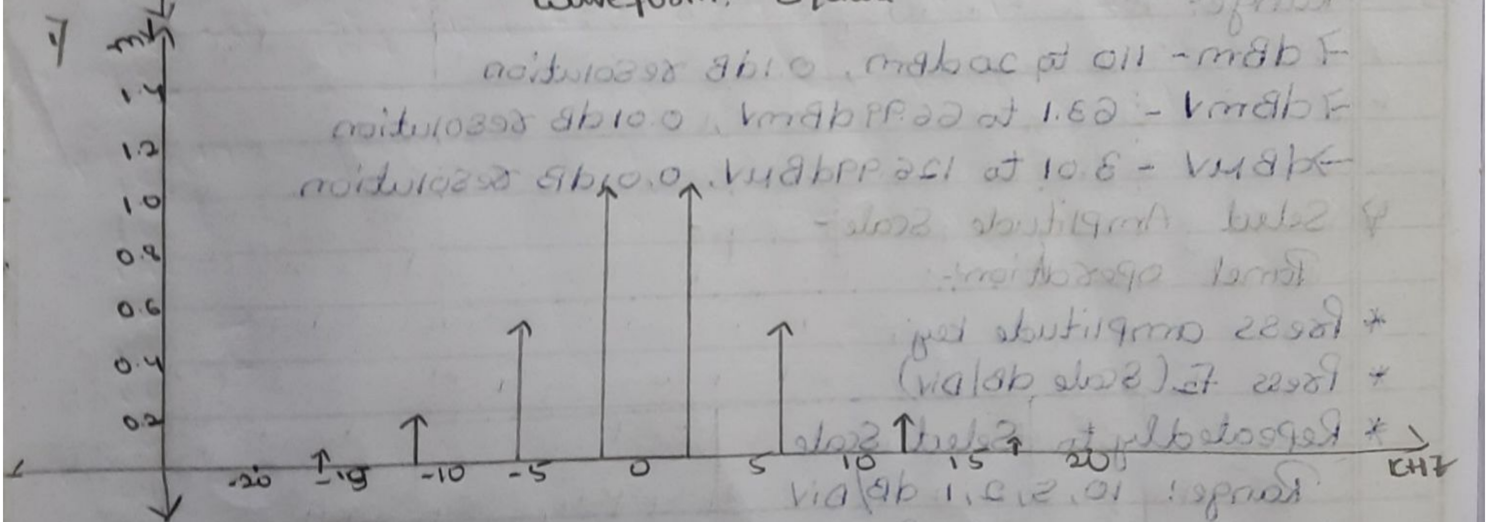
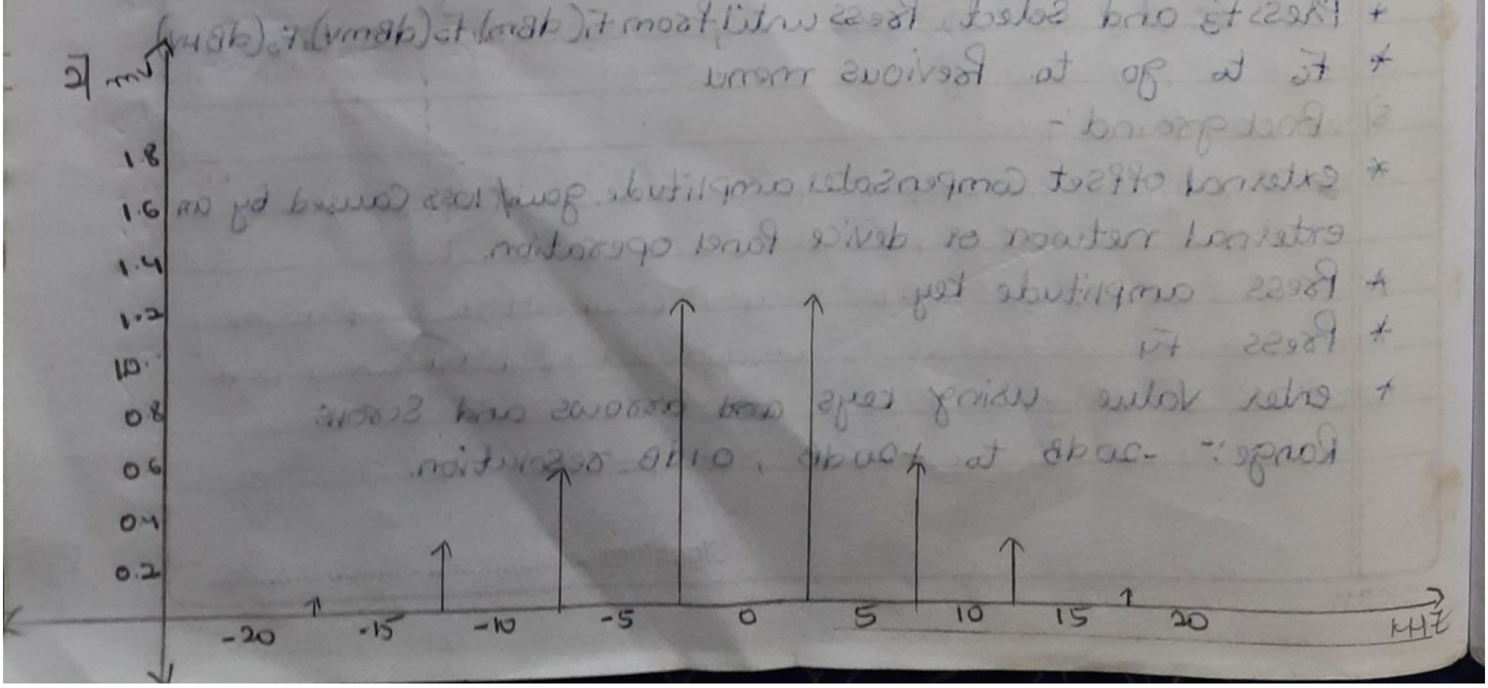
Waveform : Sine

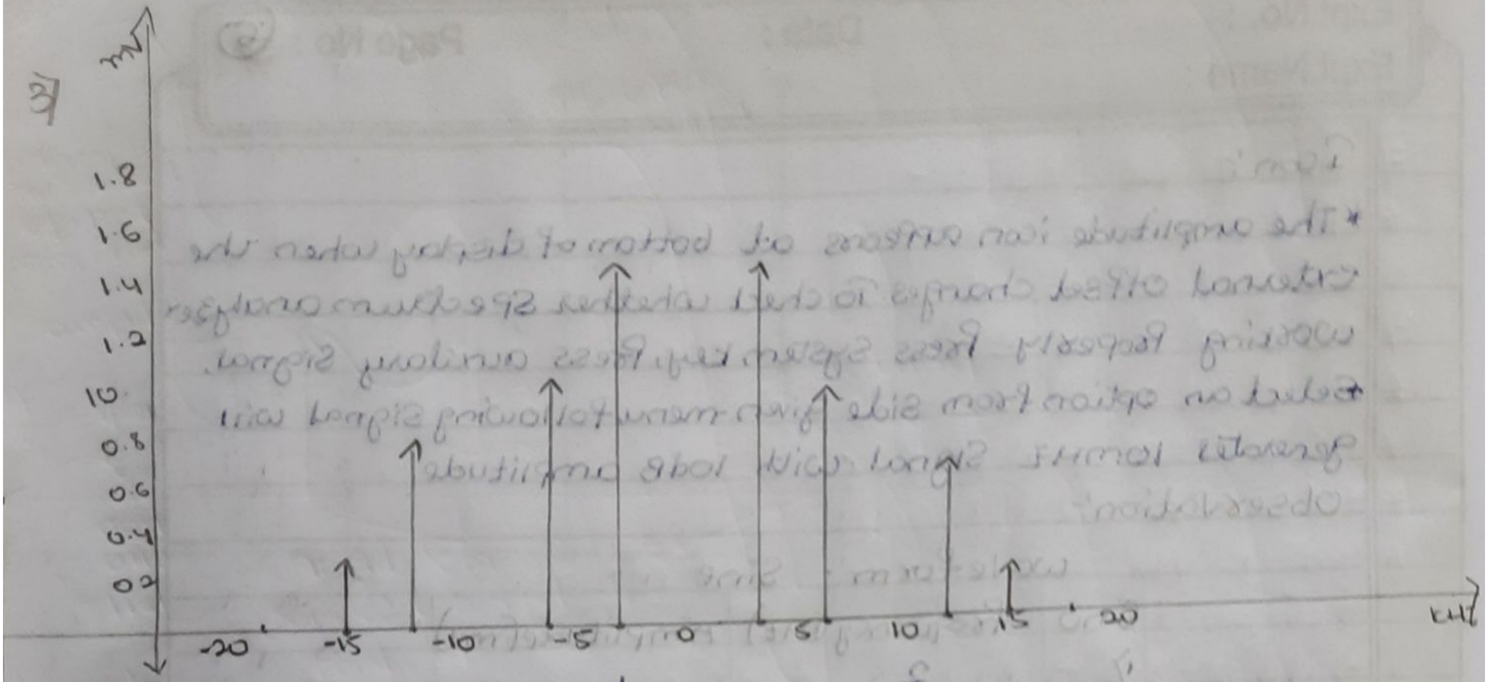


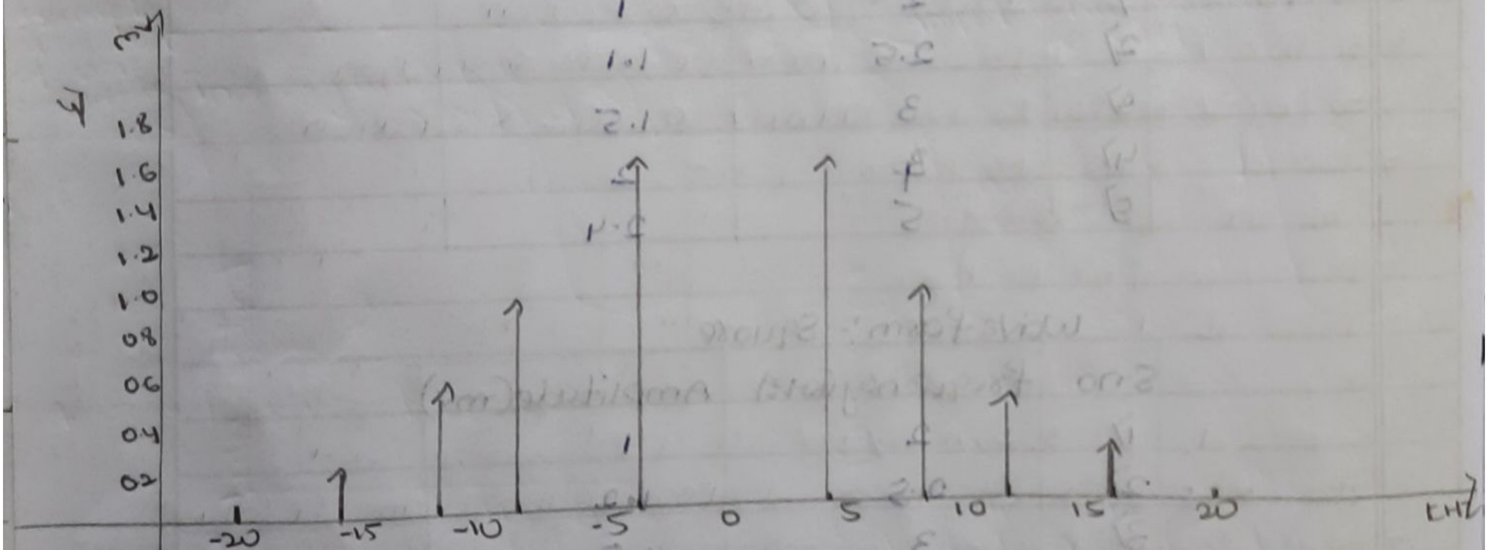


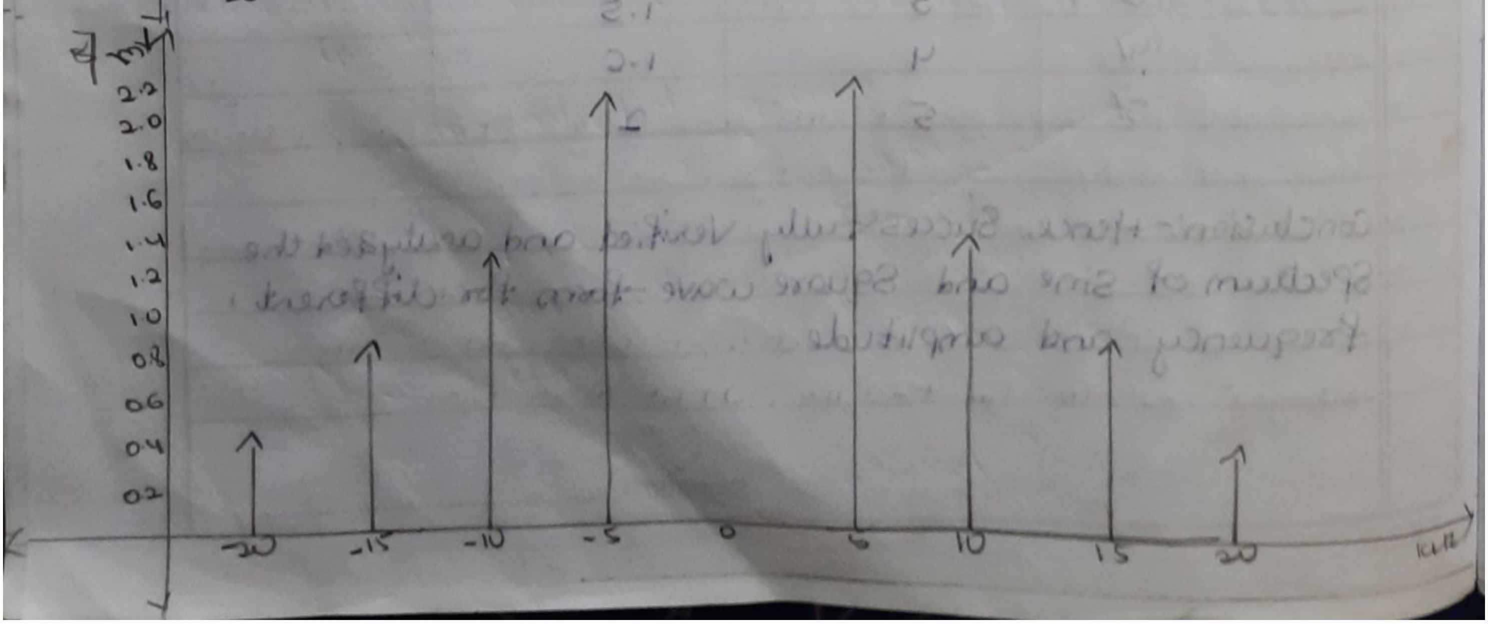
wave form: Square






mV
1.8
1.6
1.4
1.2
1.0
0.8
0.6
0.4
0.2
-20
-15
-10
-5
0
5
10
15
20
KHz


mV
1.8
1.6
1.4
1.2
1.0
0.8
0.6
0.4
0.2
-20
-15
-10
-5
0
5
10
15
20
KHz


mV
2.2
2.0
1.8
1.6
1.4
1.2
1.0
0.8
0.6
0.4
0.2
-20
-15
-10
-5
0
5
10
15
20
KHz

Aim:- To examine Sampling and reconstruction of Signal, verify the Nyquist criteria by varying Sampling frequency. Draw Sampled version of waveform. For conditions i) $f_s < 2f_m$ (ii) $f_s > 2f_m$ (iii) $f_s = 2f_m$ where f_s - Sampling frequency; f_m - max baseband frequency and represent output responses for different order low Pass filter

Apparatus:- Nyquist Applet

Theory:- A continuous thin Signal can be Stored in a digital computer in form of equidistant discrete Points or Sample. The higher the Sampling rate(f_s), the more accruate would be Stored information and Signal reconstruction from its Samples. However, high Sampling rates Produces a large volume of data to be Stored.

Analog signal:- continuous time varying feature of Signal.
Digital Signal:- data as a Sequence of discrete values.
Technique used for Analog-to-digital convertor → Pulse code modulation. It has following

* → Sampling :-
* Quantization ⇒ Digitalization
* Encoding
* Sampling is Process of measuring instantenous values of continuous-time Signal in descrete form. Sample is a Piece of data taken from whole data which is continuous in time domain.
* when a source generates a analog Signal and if that has to be discretized in time. This discretization of analog Signal is called Sampling.

It is obvious that a much higher Sampling rate is needed for Sampling a Signal which is rich in high frequency Component Such as Sound of music compared to sampling freq needed for

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Sampling a slowly varying signal, such as the output of a gas-chromatograph detector or the Potential of a glass-electrode during acid-base titration.

The minimum Sampling frequency of a signal that it will not distort its underlying information, should be double the frequency of its heightest frequency compod this is Nyquist sampling theorem.

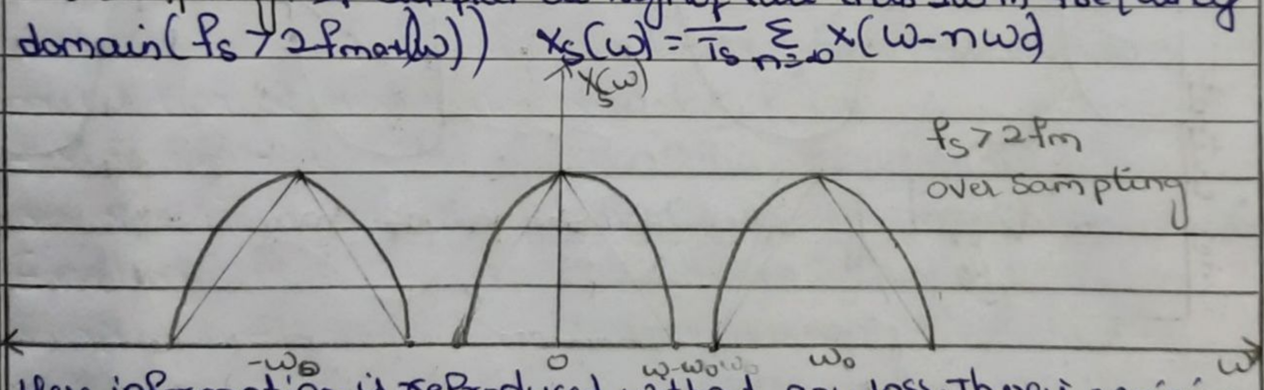
Nyquist Rate:-

Suppose that a signal is band-limited and w is highest frequency Therefore for effective reproduction of original signal the sampling rate should be twice highest frequency i.e $f_s = 2f_{max}$

* This is Nyquist rate of Sampling

Condition 1:-

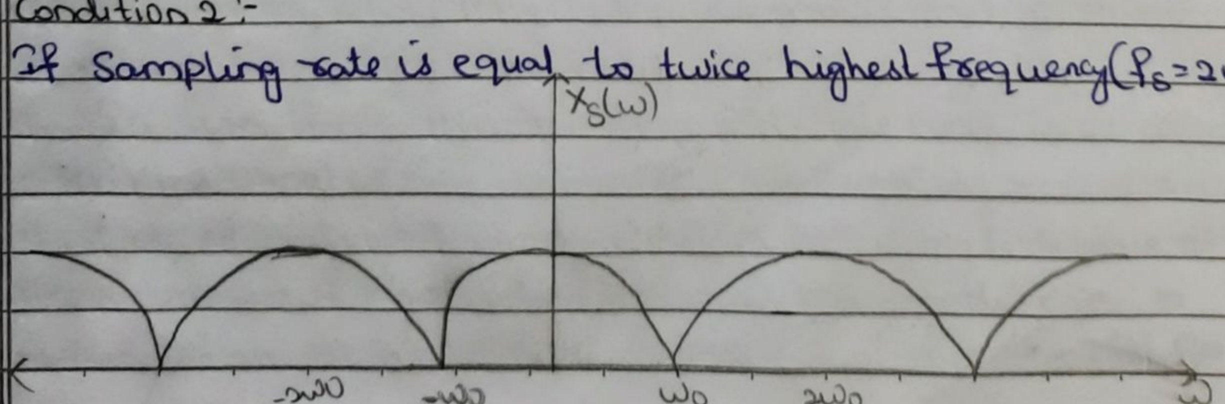
over Sampling:- If Sampled at higher rate than 2w in frequency domain ($f_s > 2f_{max}(w)$) $X_s(\omega) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} X(\omega - n\omega_s)$



Here information is reproduced without any loss. There is no mixing

Condition 2:-

If sampling rate is equal to twice highest frequency ($f_s = 2\omega$)



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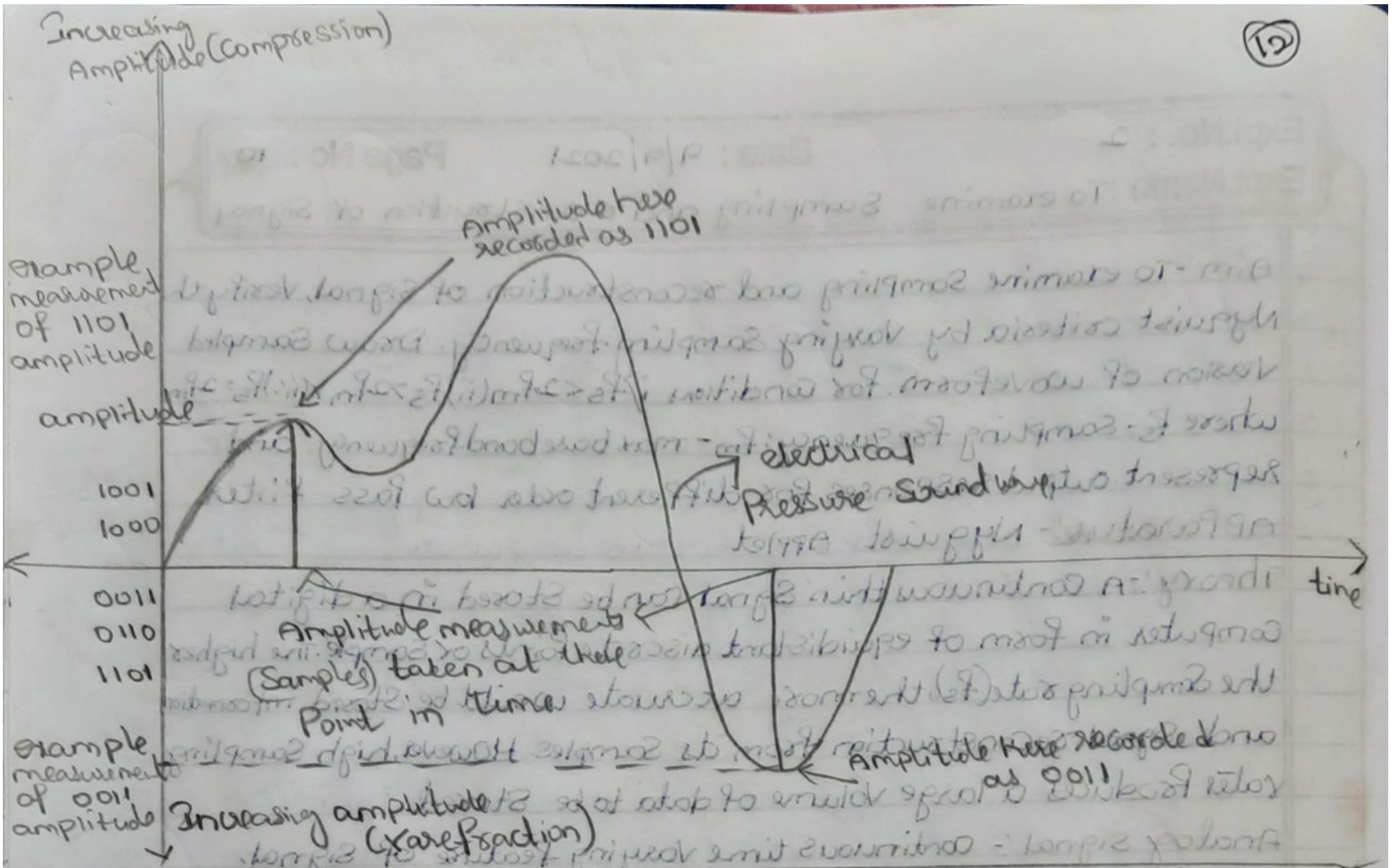
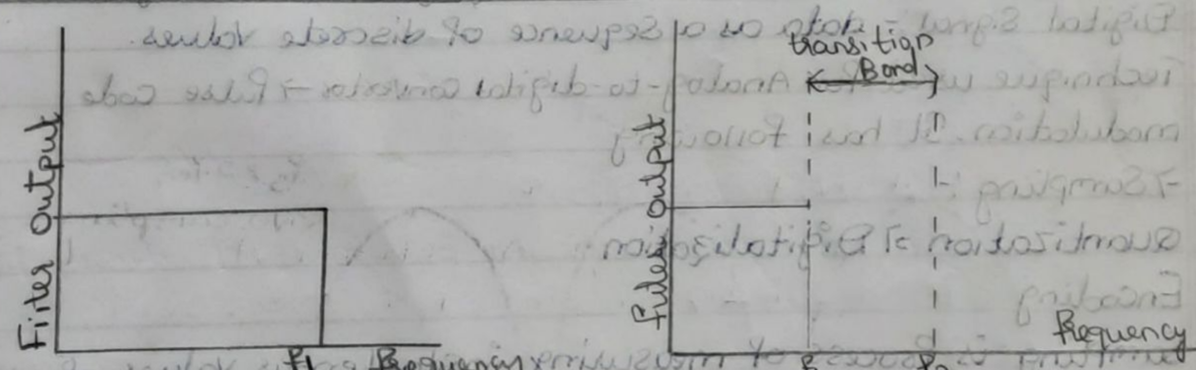
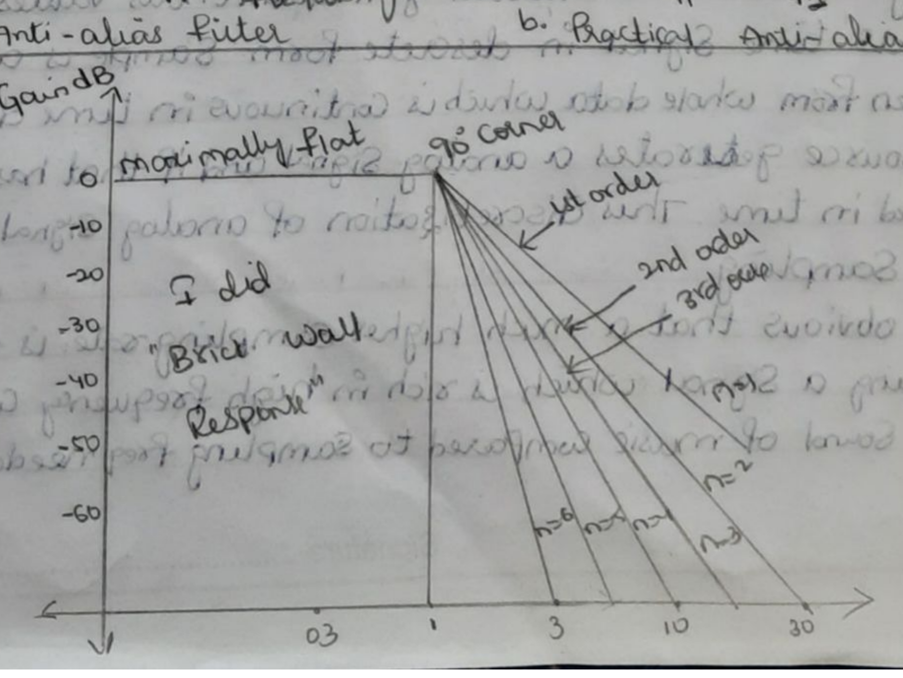
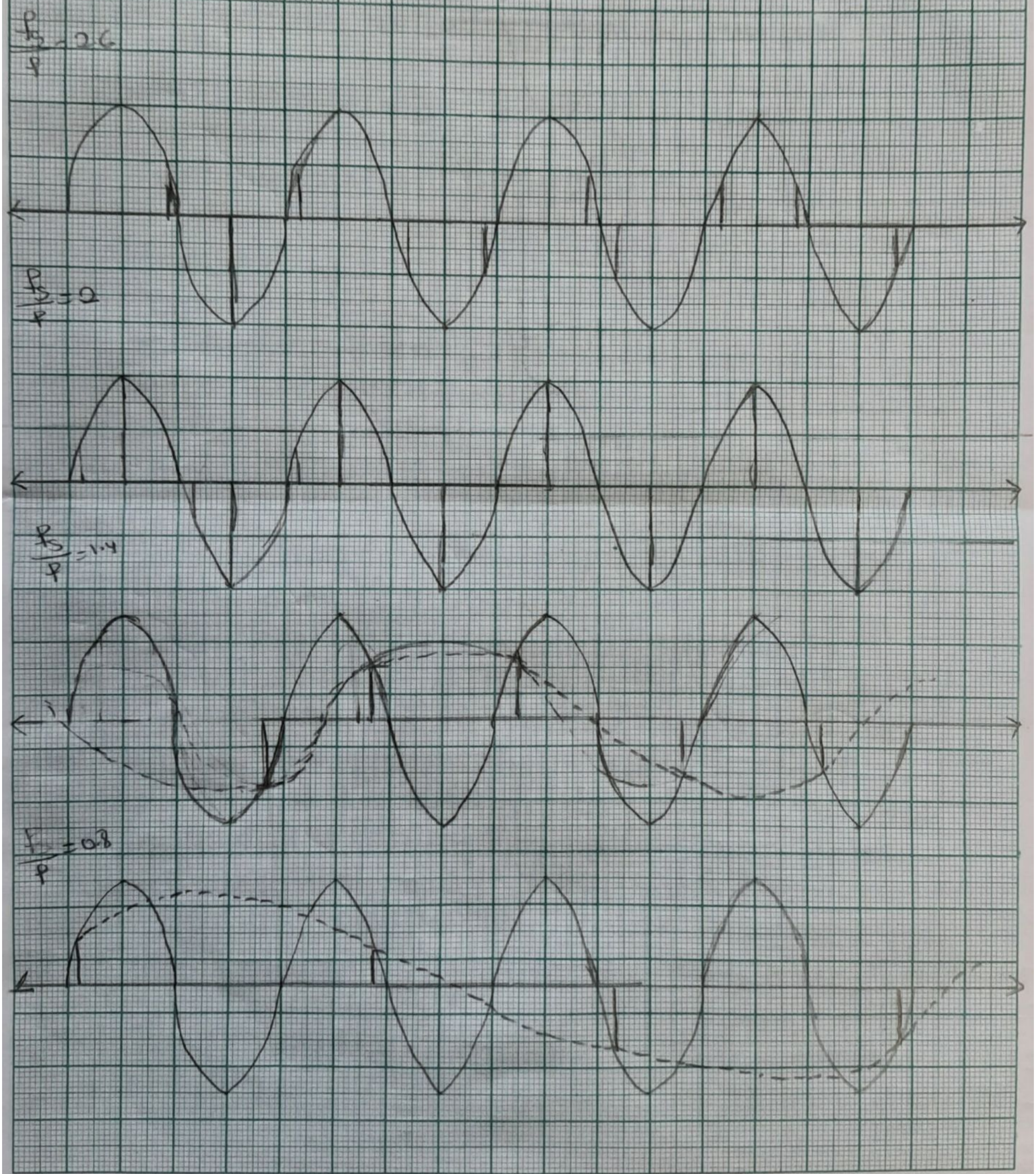




a. Ideal Anti-alias Filter b. Practical Anti-alias Filter



Sampling of a Sinusoidal Signal of frequency f at different Sampling rates f_s with dashed lines are shown the alias frequency when $\frac{f_s}{f} < 2$



* Inform is retrived without any loss. Hence it is also good.

Quantizes:

* method of sampling chuoses a few points on the analog signal and then these points are joined to roundoff the value to a near stabilized value. such a process is called quantization.
* Quantization of a signal produces the closest representable value.

Encoding:

* The digitalisation of analog signal is done by encoder
* After each sample is quantized, the no. of bits per sample is decided
* Each sample is changed to an n-bit code.
* Encoding is also used to minimize bandwidth
* Designing this filter is to determine bandwidth requied.
* max frequency of input should be less(or) equal to half of sampling rate
* This sets cutoff freq of low-pass filter
* If f_s is sampling frequency, then critical frequency(f_N) = $f_s/2$
* Any sinusoidal component of signal of frequency f higher than f_N is not only lost, but it is reintroduced them in sampled signal by folding at frequency f_N as alias frequency $F' = f_N - \Delta F$

This effect is known as aliasing. Aliasing is demonstrated. A sinusoidal signal of frequency f is sampled at four different sampling frequencies $f_s = 2.6F$, $2.0F$, $1.4F$ and $0.8F$

In two first cases sampling rates are enough for reconstruction of original signal whereas in last two subsampling occurs and the the collected points may be considered as belonging to signals of lower frequencies. The alias frequencies due to subsampling can be calculated by following Alias frequency ; $f' = |f - kf_s|$ $k=1,2$

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when $k=1.4$ the $F = |f - 1.4f| = 0.4f$

when $k=0.8$ then $F = |f - 0.8f| = 0.2f$

Problems arising due to aliasing:- The effect of the sampling frequency on spectrum of signal consisting of infinite number of components. In fig the Nyquist frequency is sufficiently higher from the max frequency (f_{max}) component of signal and stored signal is not distorted.

The opposite occurs in fig where $f_{max} > f_N$ and all frequency components higher than F_N are not only lost, but are folded at f_N and they are added to other sinusoidal components and thus signal stored due to aliasing.

A typical application for digitilaization of sound a sampling rate of about 6KHZ is sufficient for telephony since normal human voice doesnot contain an appreciable amount of freq components higher than 2-5KHZ. However, a sampling rate of about 40KHZ is needed for the digitalisation of music, since frequency components of about 15-20KHZ are common & needed for achieving sound reconstruction.

Notes:- (1):- In practice the sampling rate of f_s is in range of $2.5 \times f_{max} - 3 \times f_{max}$. For digital recording of music $f_s = 44KHZ$ used.

(2):- Prior to sampling, signal must pass through a low-pass filter which will remove all unnecessary components lighter than f_{max}. Preventing thus the "contamination" of stored signal by their biased frequencies.

Procedure:- Applet

with this easy to use applet, we can observe effect of sampling rate on a specified signal using two horizontal sliders you can select a) freq of signal b) sampling freq (0 - 50HZ), independently of each other

on the topmost grapling window, the freq spectrum is shown. The blue line indicates freq of sample signal freq by red line and indicates

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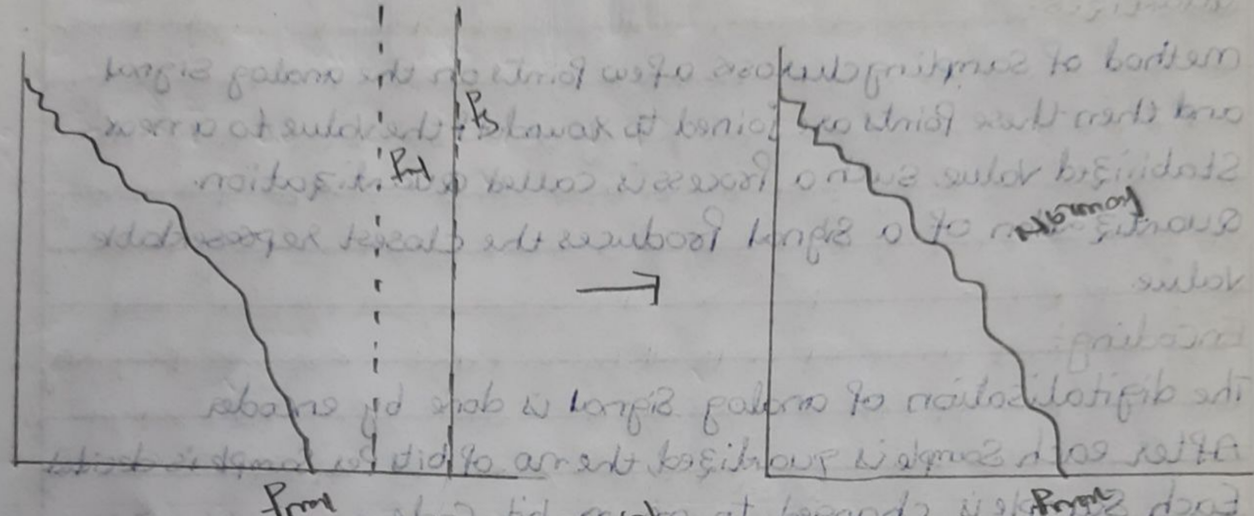
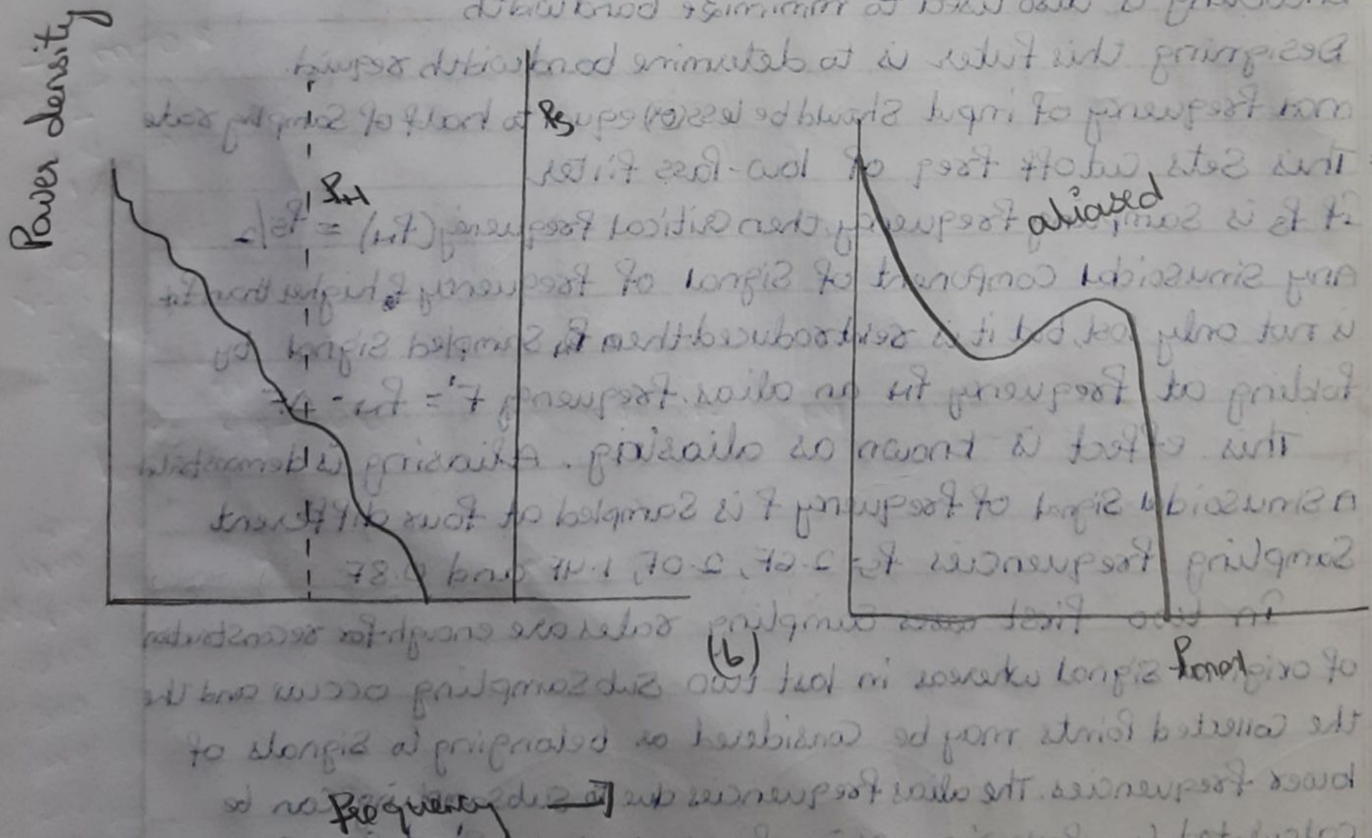
A Signal with frequency Spectrum shown of maximum useful frequency f_{max} is Sampled for digitalisation at frequency f_s (16)

a) The sampled frequency is sufficiently higher than f_{max} & no aliasing occurs

b) Sampling frequency is insufficient, aliasing takes place and signal is distorted



(a)



(b)

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black colour appears the critical frequency. The dashed green line indicates freq of signal resulting due to aliasing and it appear whenever critical freq less than signal frequency

observation:-

1)

Signal frequency = 10HZ	
Sampling frequency (HZ)	Alias frequency (HZ)
7	3
10	0
15	5
20	-
22	-

2)

Signal frequency = 20HZ	
Sampling frequency (HZ)	Alias frequency (HZ)
19	1
20.1	0.1
30	10
40	7
42	-

Conclusion:-

Successfully examined sampling and reconstruction of a signal and verified the Nyquist criteria by varying the sampling frequency.

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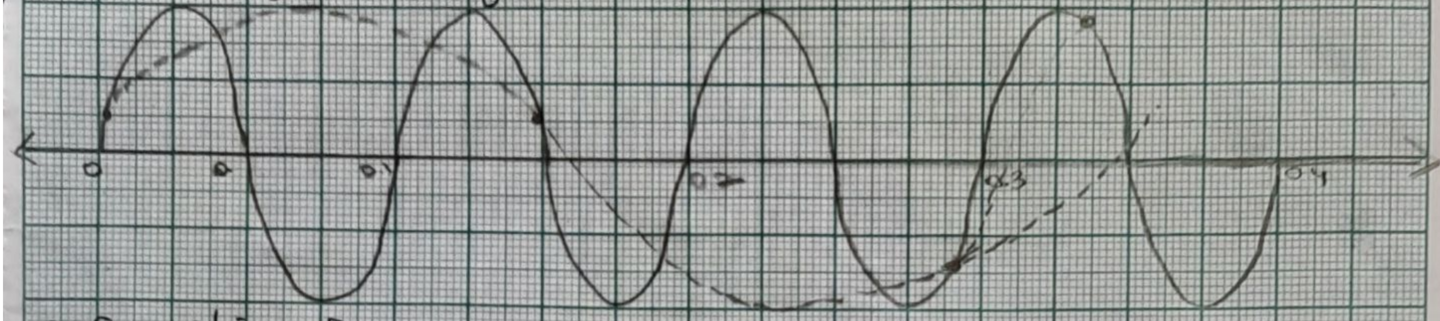
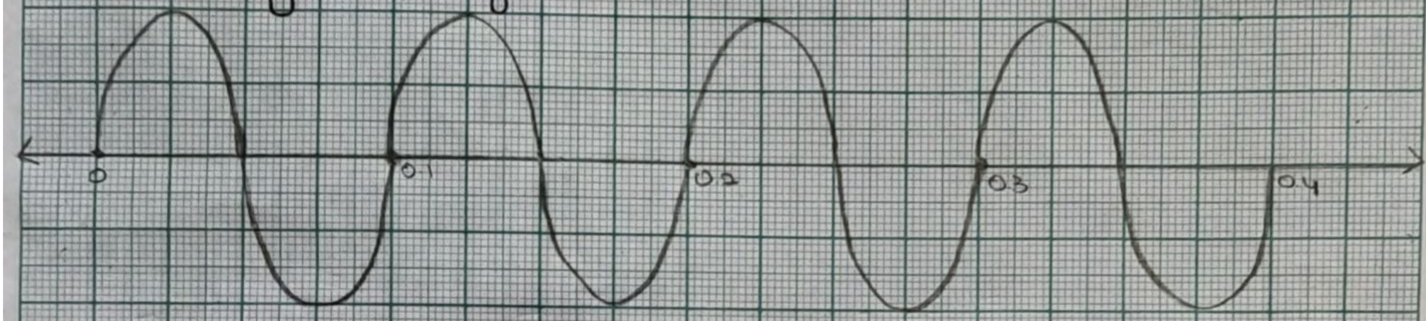
Signal frequency = 10HZ

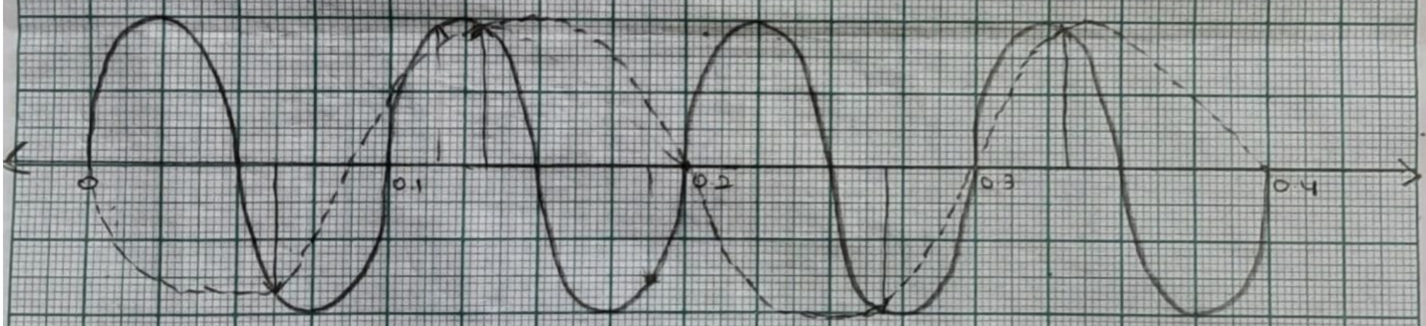
1) Sampling frequency = 7HZ



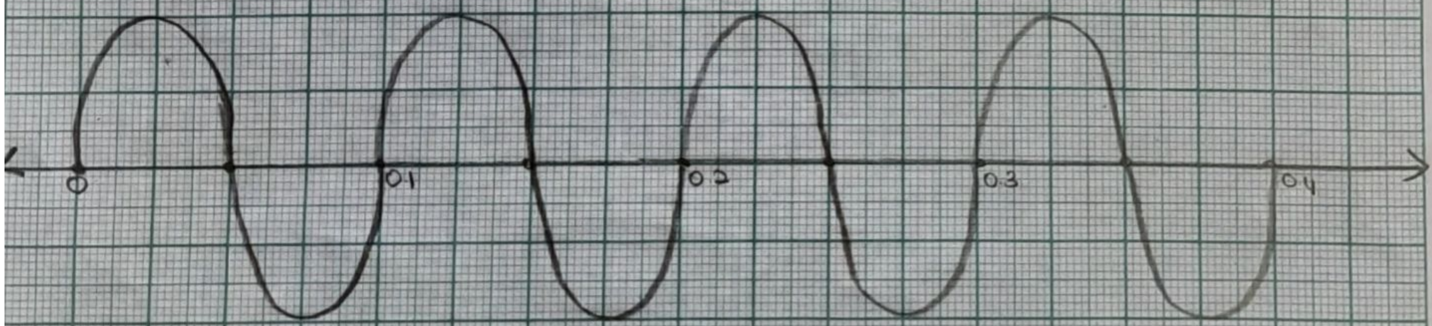
2) Sampling frequency = 10HZ



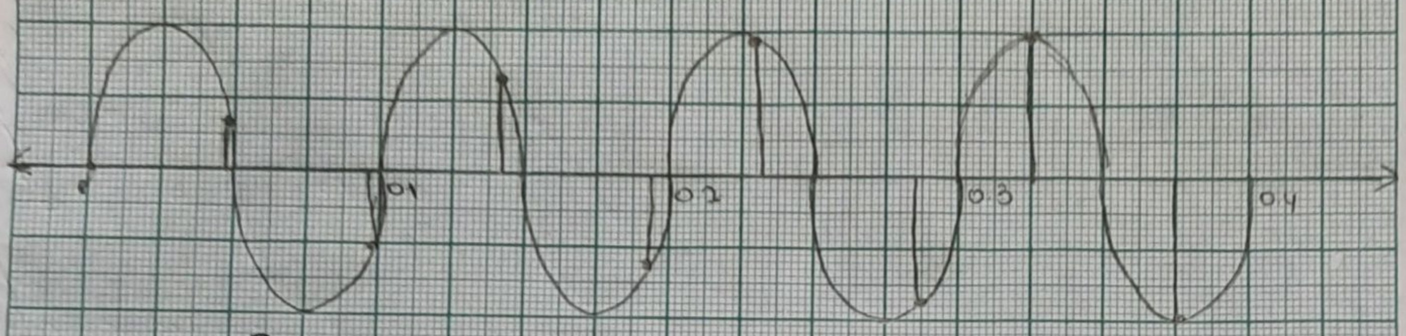
3) Sampling frequency = 15HZ



4) Sampling frequency = 20HZ

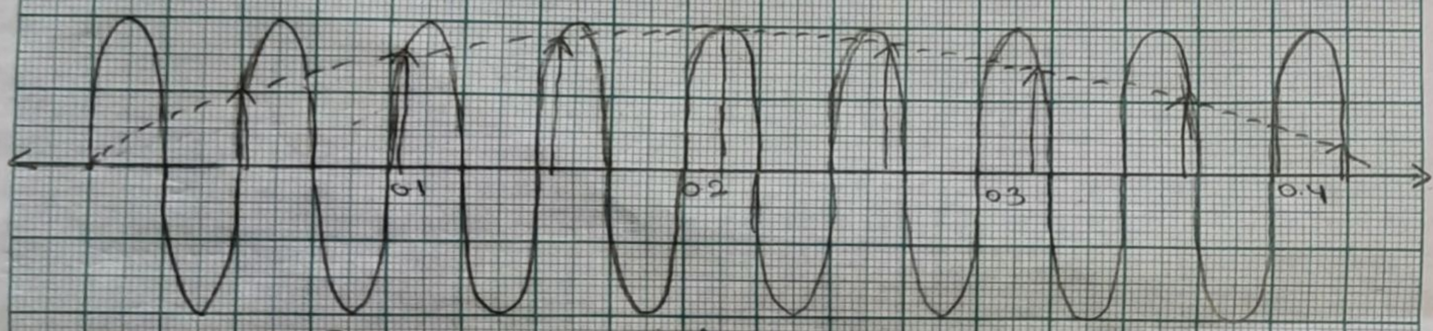


5) Sampling frequency = 22HZ

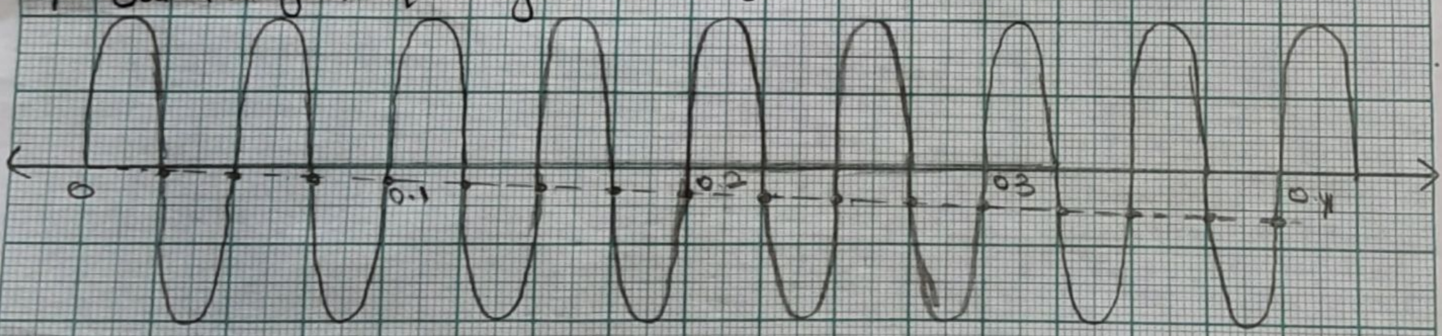


Signal frequency = 20HZ

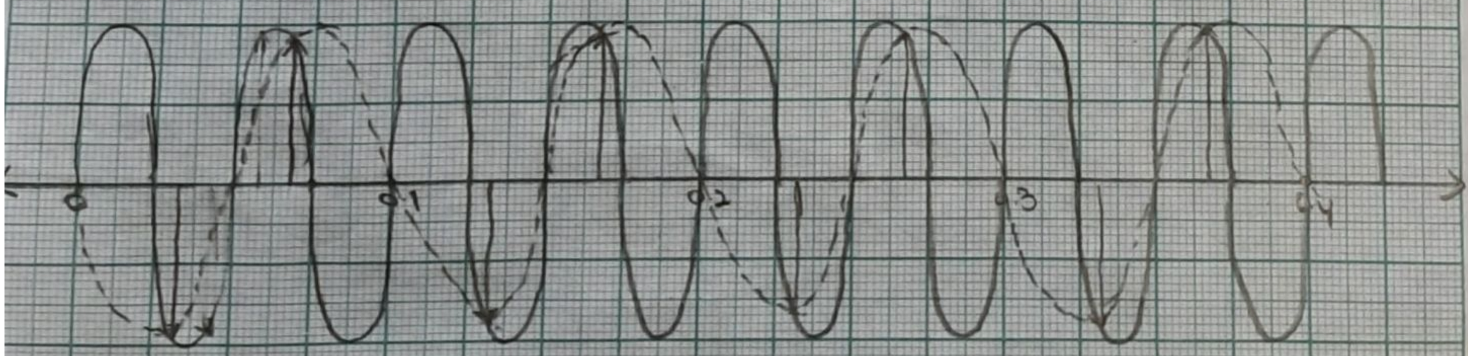
1) Sampling frequency = 19.0HZ



2) Sampling frequency = 20.1HZ

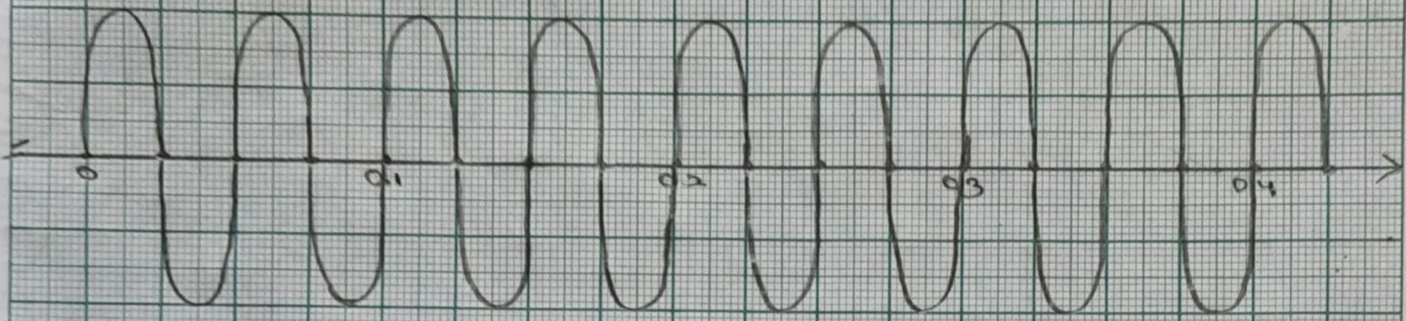


3) Sampling frequency = 30HZ

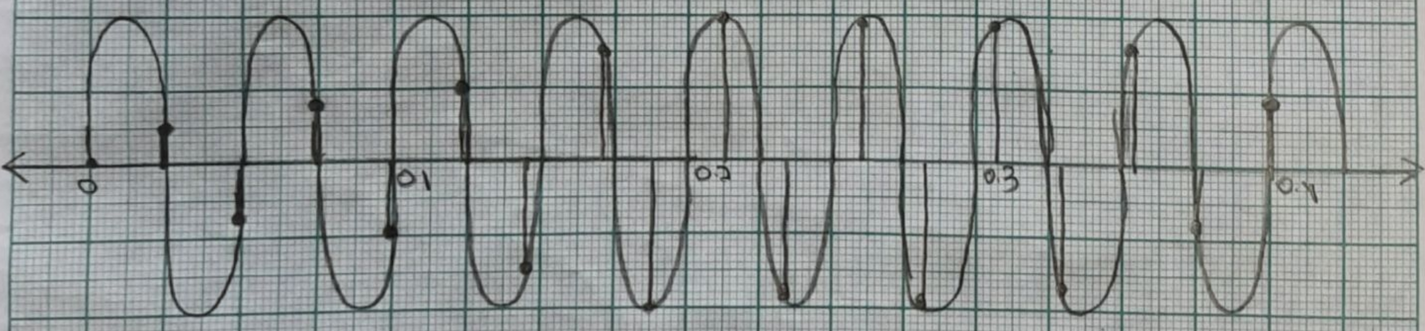


(19)

4) Sampling Frequency = 40Hz



5) Sampling Frequency :- 42Hz



Expt.Name : Amplitude modulation

Aim :- To study Amplitude modulation (Am) technique, modulation-Index (m), draw waveforms, spectra and trapezoidal display. Illustrate the observed Am signals for double sideband with and without carrier by changing m as :- $m>1$, $m<1$ and $m=1$ & draw. use virtual mode with appropriate software.

Apparatus/Software :- Labalive

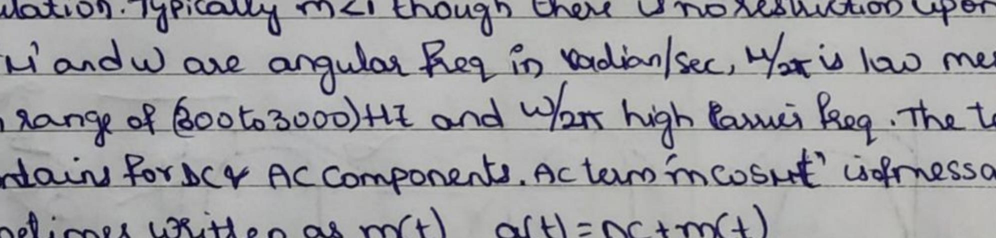
Theory :- Amplitude modulation is a modulation technique used in electronic communication, most commonly for transmitting messages with a radio carrier wave. In this amplitude of carrier wave is varied in proportion to that of a message signal, as the audio signal. The amplitude modulated signal is defined as:

$A_m = E(1+m\cos Mt)\cos wt$ — ① — (low freq term $a(t)$) x (high freq $c(t)$) $(A \cdot B = E)$

E is Am signal amplitude from ①, m is constant & defines depth of modulation. Typically $m<1$ though there is no restriction upon 'm' in eqn. 'M' and w are angular freq in radian/sec, $M/2\pi$ is low message freq in range of (300 to 3000) Hz and $w/2\pi$ high carrier freq. The term $a(t)$ contains for DC & AC components. AC term '$m\cos Mt$' is of message and is sometimes written as $m(t)$ $a(t) = DC + m(t)$

Depth of modulation :- 100% modulation is defined when $m=1$. It requires the amplitude of DC(A), part of $a(t)$ is equal to amplitude of AC(Am)

measurement of M :- The magnitude of 'm' can be measured directly from AM display itself $m = \frac{P-Q}{P+Q}$



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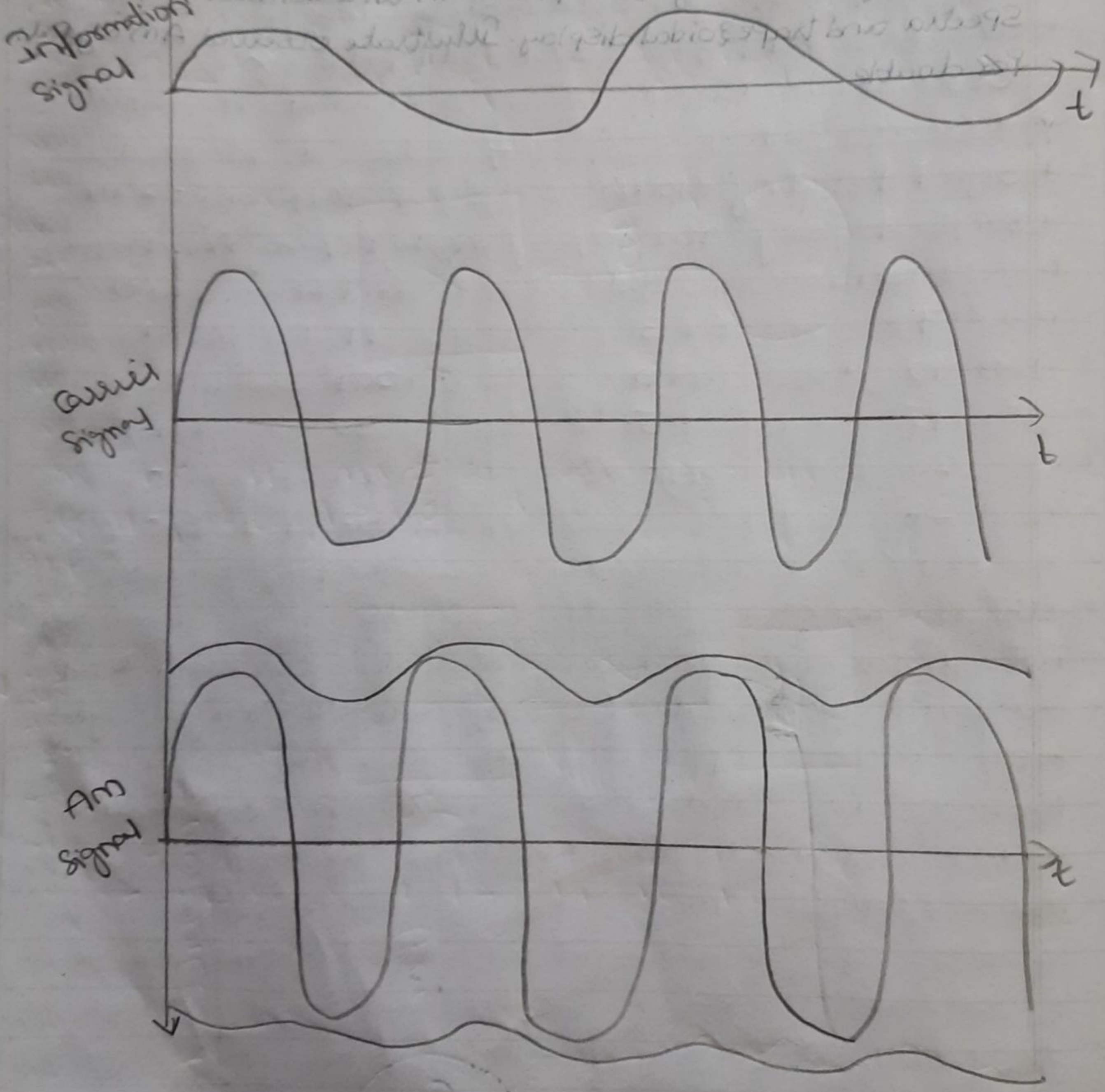
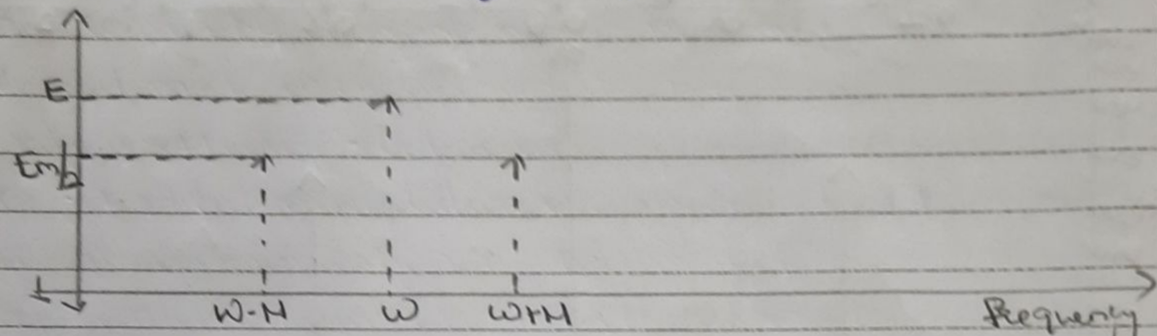

Information signal
t
carrier signal
t
Am signal
t

Spectrum:- Analysis shows that sidebands of AM, when derived from a message of frequency M rad/s are located either side of carrier frequency, spaced from it by M rad/s.



In the case of Periodic modulating signal, it is easy to indentify max and min amplitudes of modulated wave. with a non-Periodic signal such as speech waveform, these quantities will vary, hence modulation index also varies. The modulation index must be <1 if exceeds 1 then negative Peak of moduilty waveform is clipped. This is bad enough in itself, but in addition, which is Potential source of interferense.

Significance of 'm':- The shape of outline envelope of AM wave is exactly that of message of the waveform. Examing the varying of 'm'.

→ For all values of m<1, envelope of AM is same shape of the message

→ For all values of m≥1, the envelope is not copy of message shape.

It is important to note that for m>1, it shouldnt be considered that there is envelop distortion, since resulting shape not that of message is shape that theory Predicts.

The modulation trapezium:- For sinuosidal message, it is easy to set the depth of modulation to any value of 'm'. But it is not ~~mo~~ convinied for other messages like speech. The trapezoidal display is useful alternative for more complex messages.

→ we can calculate m in time domain using oscilloscope and trapezoid method.

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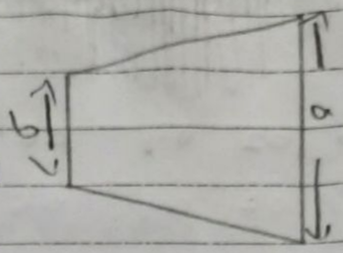
→ The scope is placed in xy mode

x: moduating signal

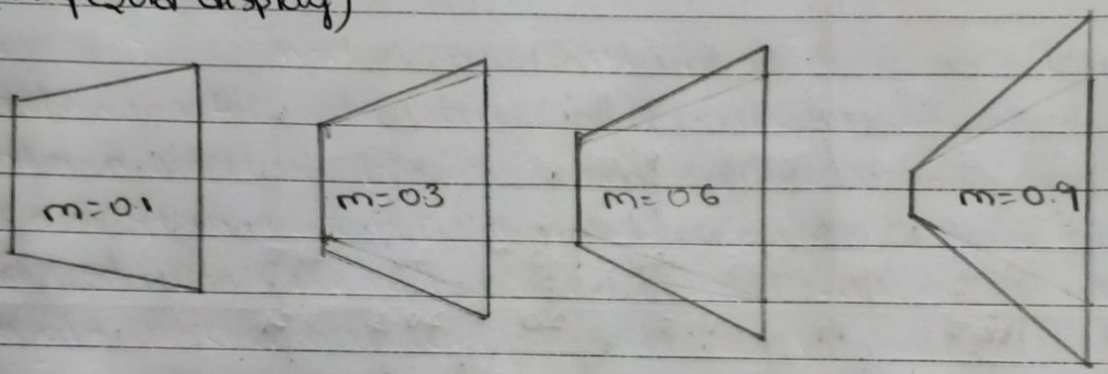
y: modulated signal

modulation index calculated from vertical edge length by: $m = \frac{a-b}{a+b}$



→ As modulation index (m) increases the ratio between vertical trapezoid edges increases. The trapezoid width unaffected by modulation depth.

(Trapezoid display)



observation

→ Double side band with carrier (DC offset ~~ON~~)

DC offset = ON

	m (modulation index)
m < 1	0.5
	0.8
m = 1	1
m > 1	1.2
	1.5

DC offset = off

	m (modulation index
m < 1	0.5
	0.8
m = 1	1
m > 1	1.2
	1.5

Signature ....

S.K.B.W

Advantages of Am:

* It is simple to implement
* Am waves can travel long distances
* Am waves have low-band width
* Am transmitters are less Complex. Am receivers are cheap as no Specialised Components are needed.

Disadvantages of Am:-

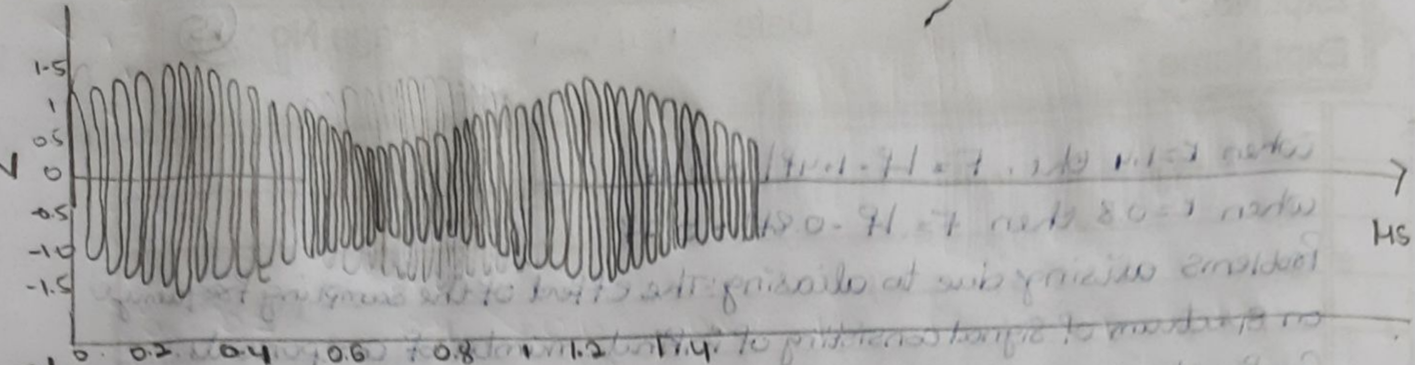
* Am Signal is not efficient in terms of its Power usage
* It is not efficient in terms of its use of band width. It requires a bandwidth equal to twice that of highest audio frequency.
* Am detectors are Sensitive to noise hence an Am is Prone to noise.
* Reproduction is not high fidelity.

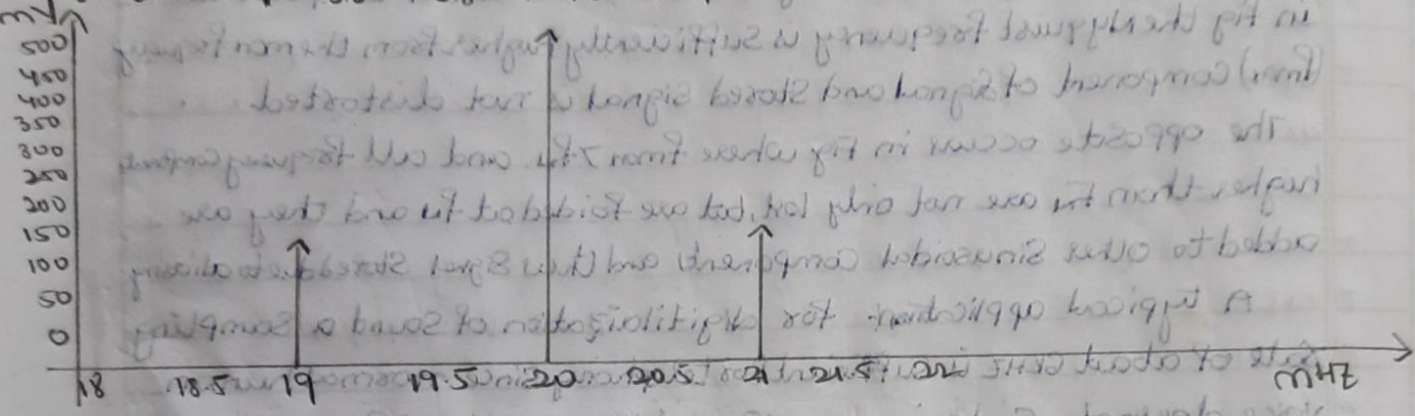
Conclusion:- Successfully observed and verified Am signals for double side band width with and without Carrier by changing m as m>1, m=1, m<1 using labalive Software.

Signature ....

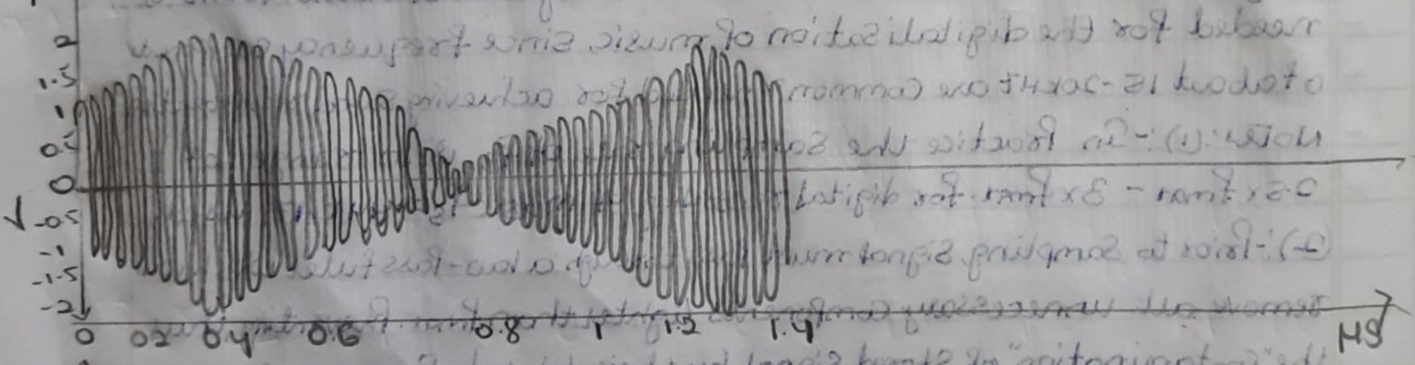
Double Sideband with carrier:

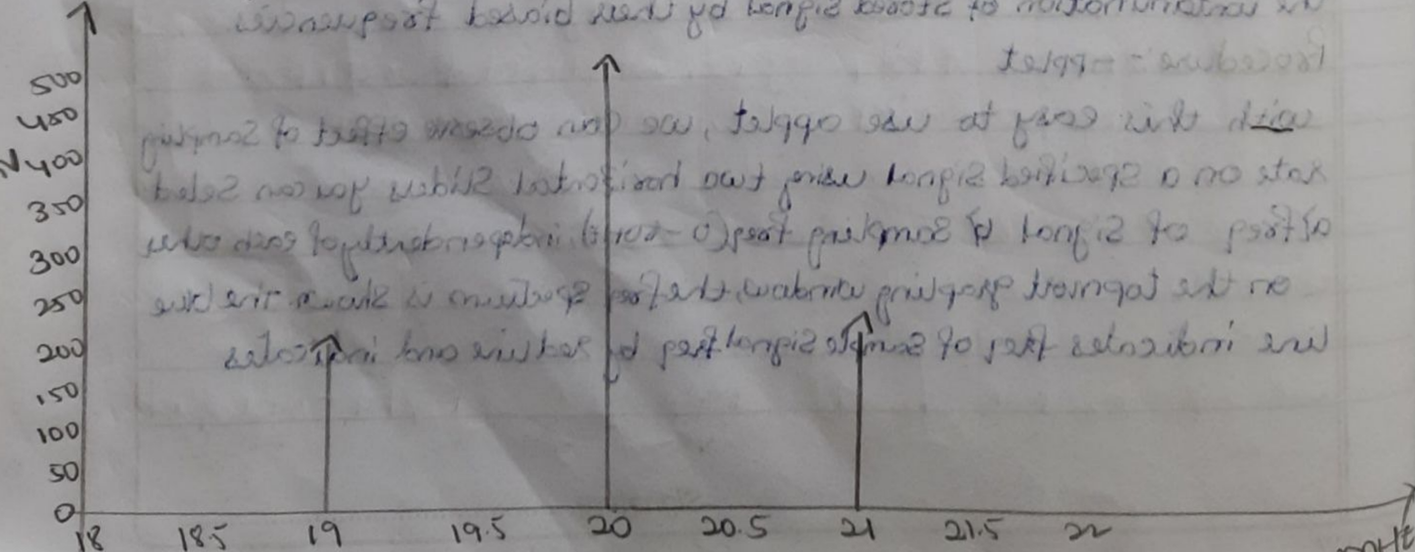
1) m= 0.5



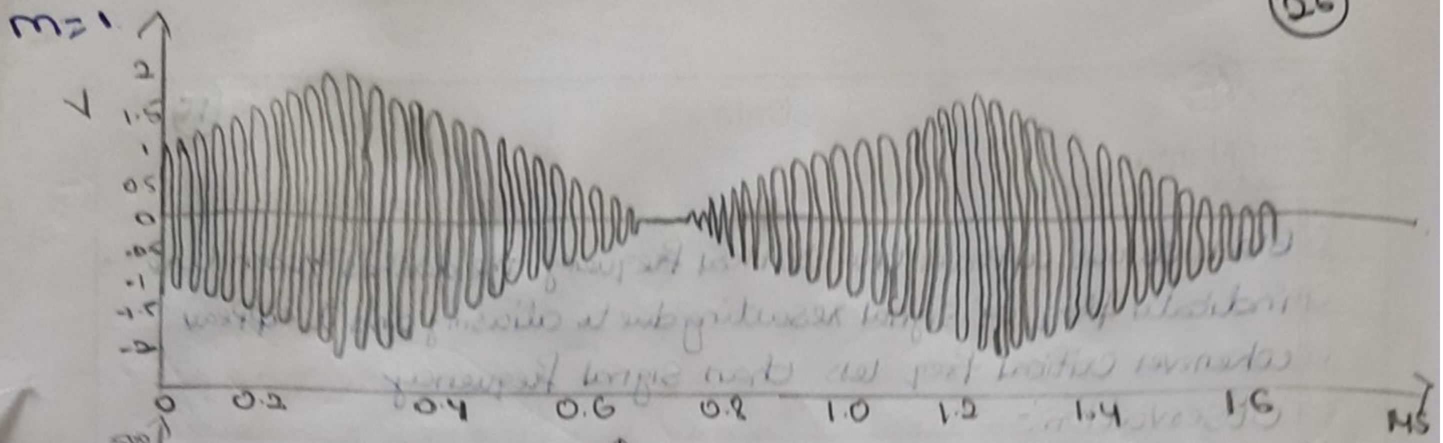


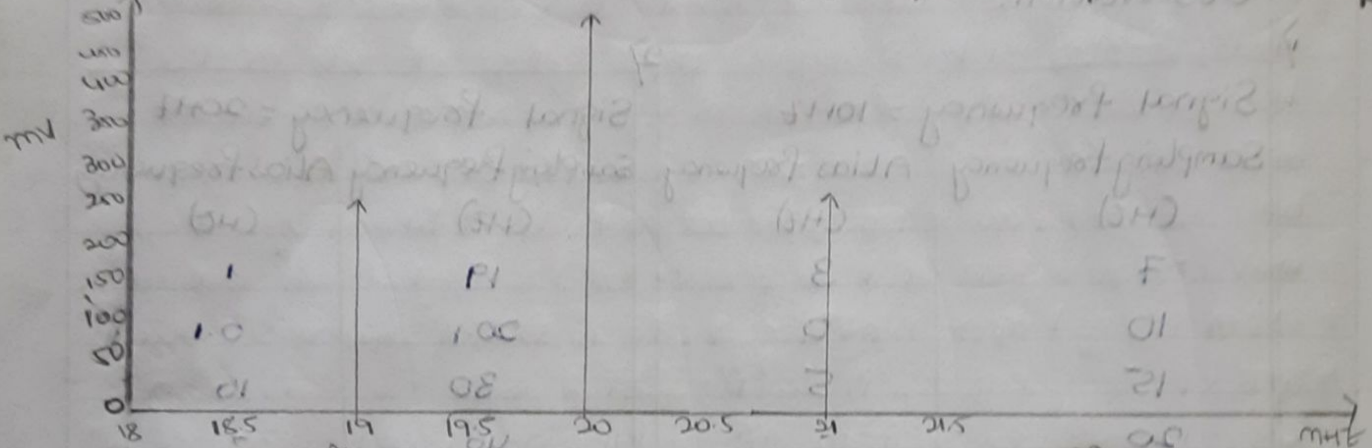
2) m=0.8



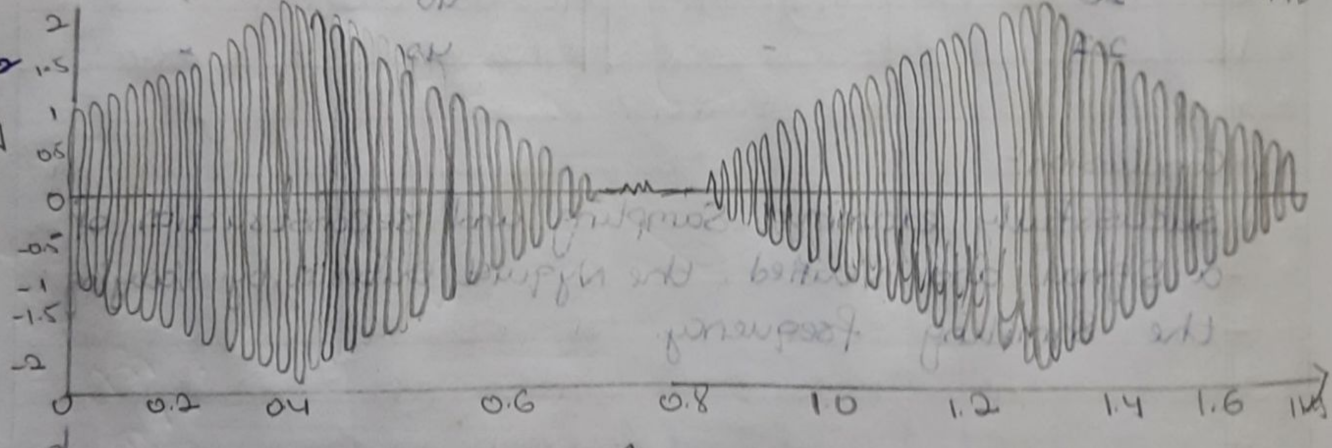


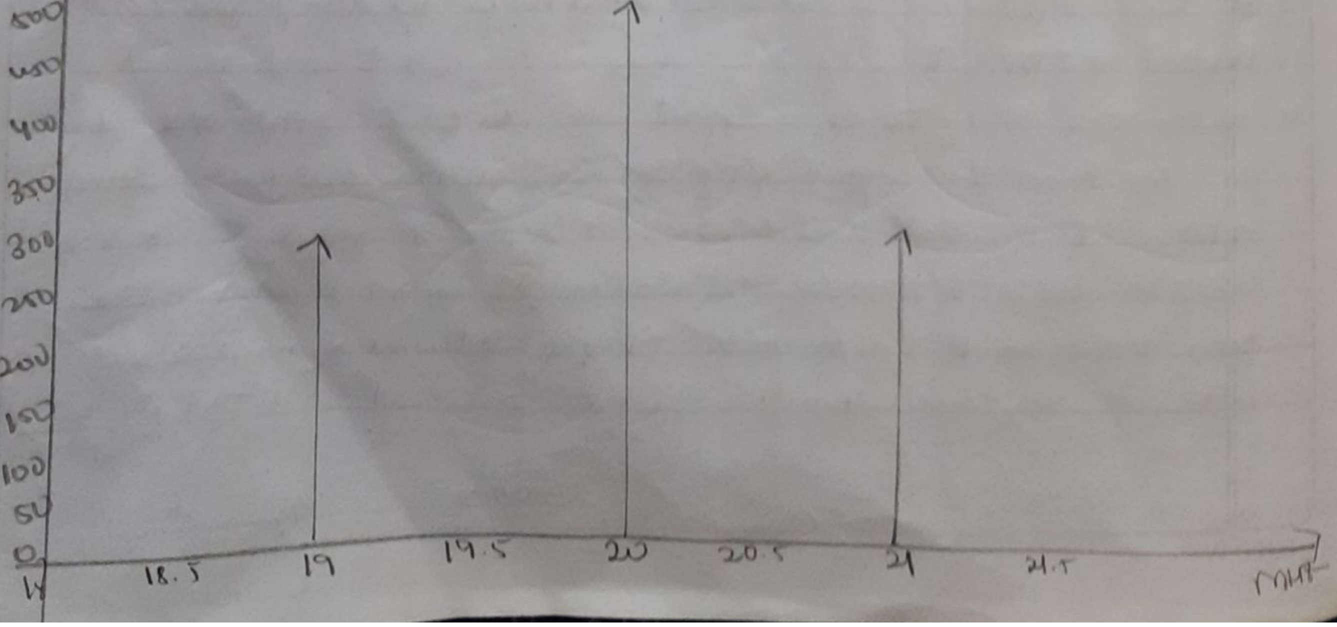
m=1


V
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
0.2
0.4
0.6
0.8
1.0
1.2
1.4
1.6
MS

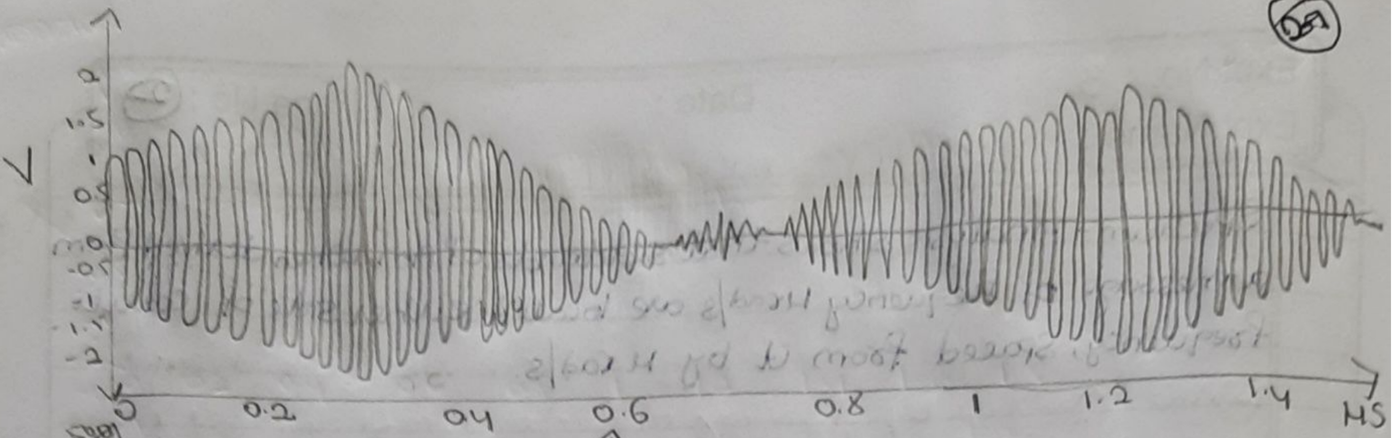

mV
500
450
400
350
300
250
200
150
100
50
0
18
18.5
19
19.5
20
20.5
21
21.5
MHz

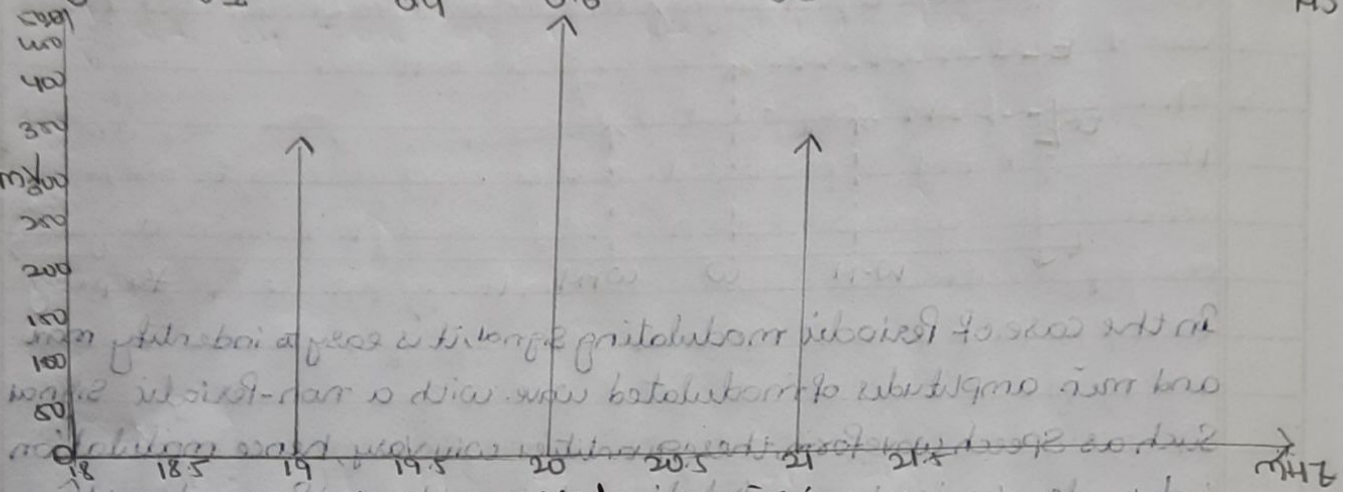
m=1.2


V
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
0.2
0.4
0.6
0.8
1.0
1.2
1.4
1.6
MS


500
450
400
350
300
250
200
150
100
50
0
18
18.5
19
19.5
20
20.5
21
21.5
MHz

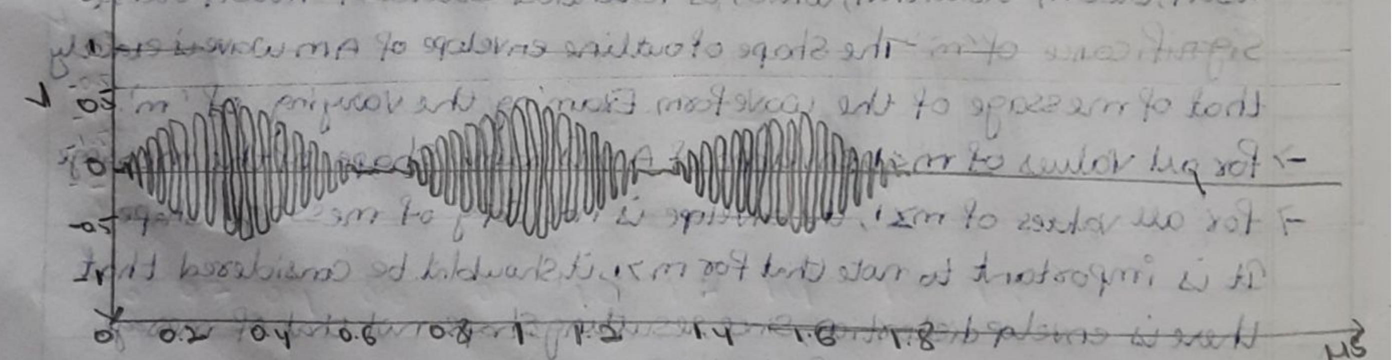
m=1.5

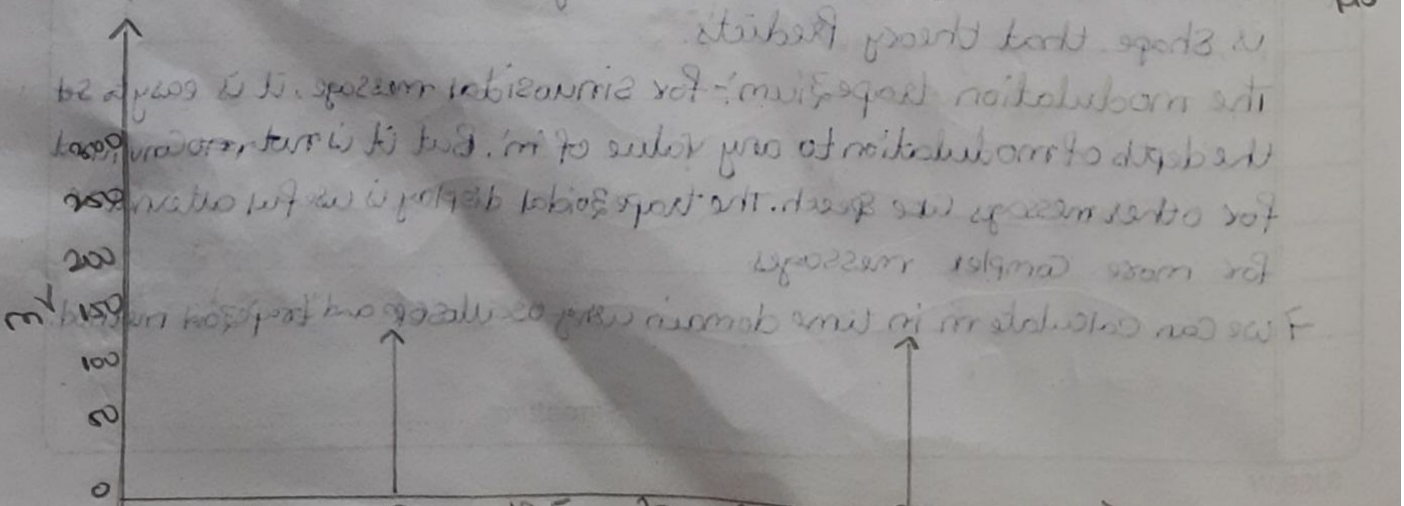




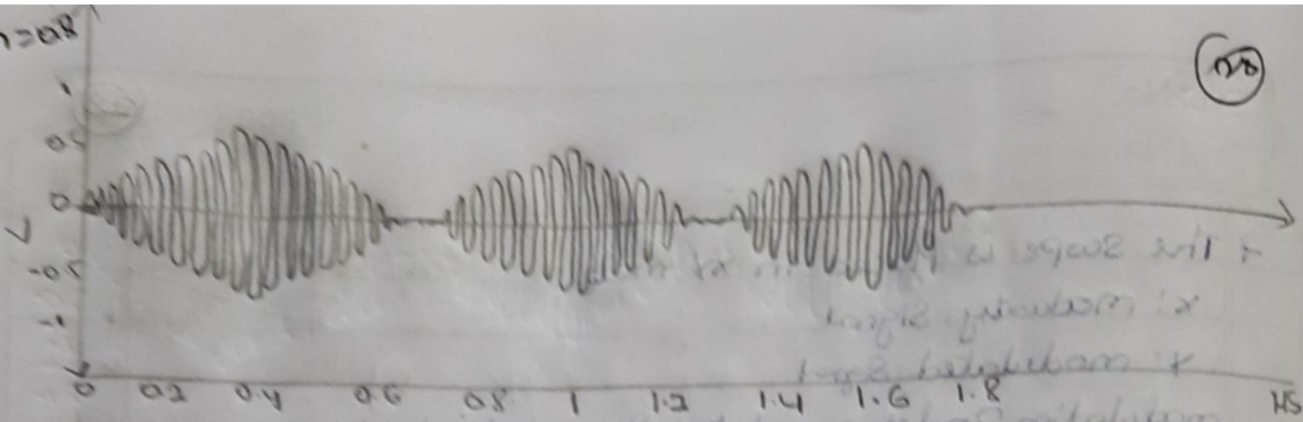
Double sideband suppressed carrier:-

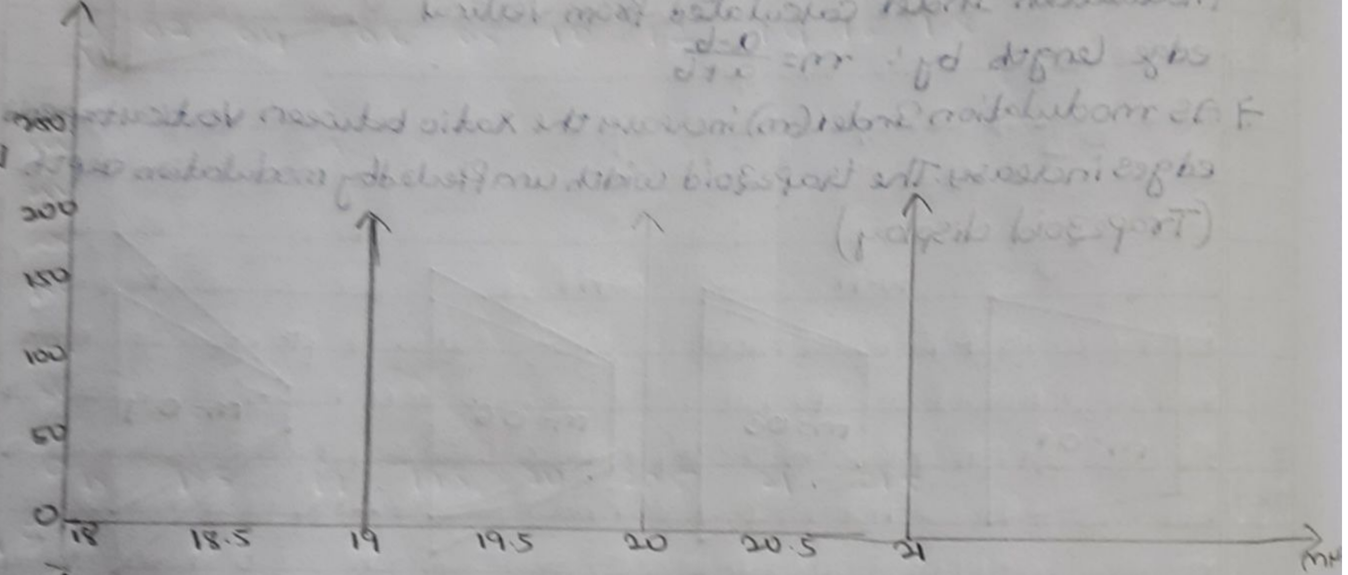
i) m=0.5



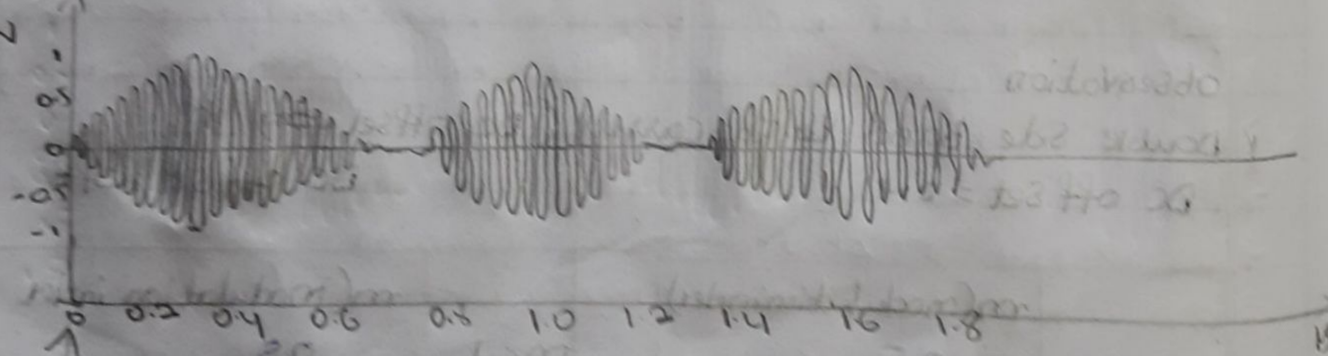


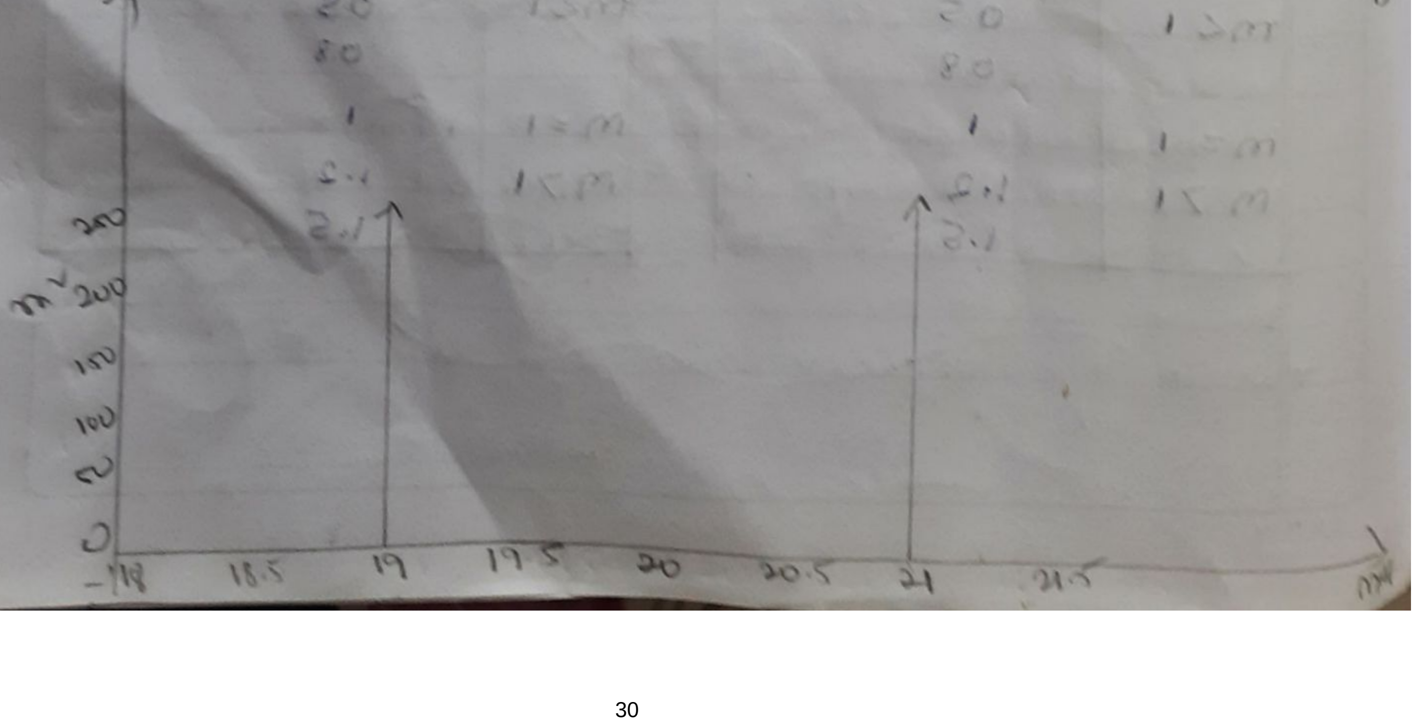
c) m=0.8


V
1
0.5
0
-0.5
-1
0
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
ms

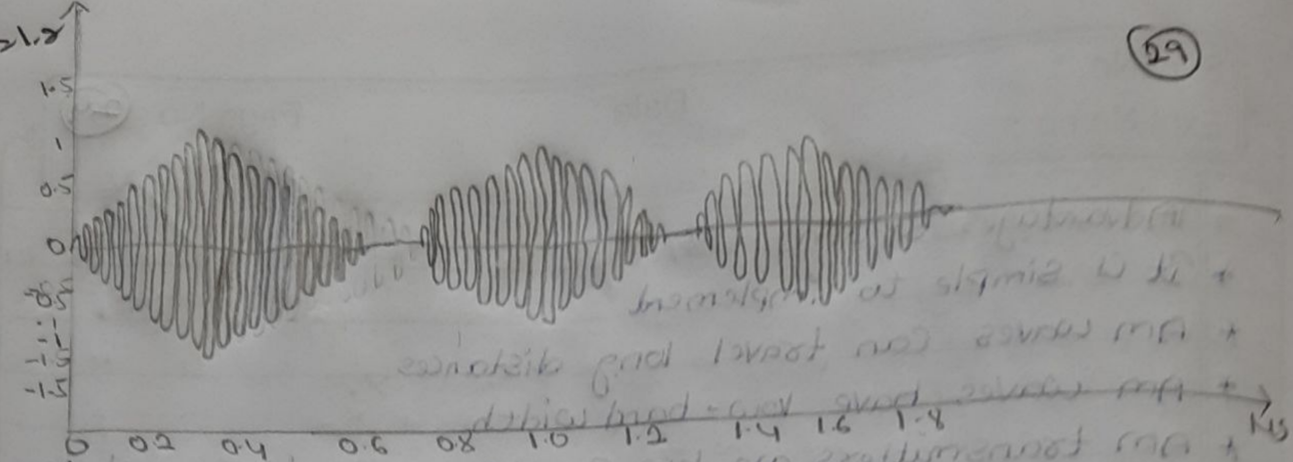

mV
200
150
100
50
0
18
18.5
19
19.5
20
20.5
21
mHz

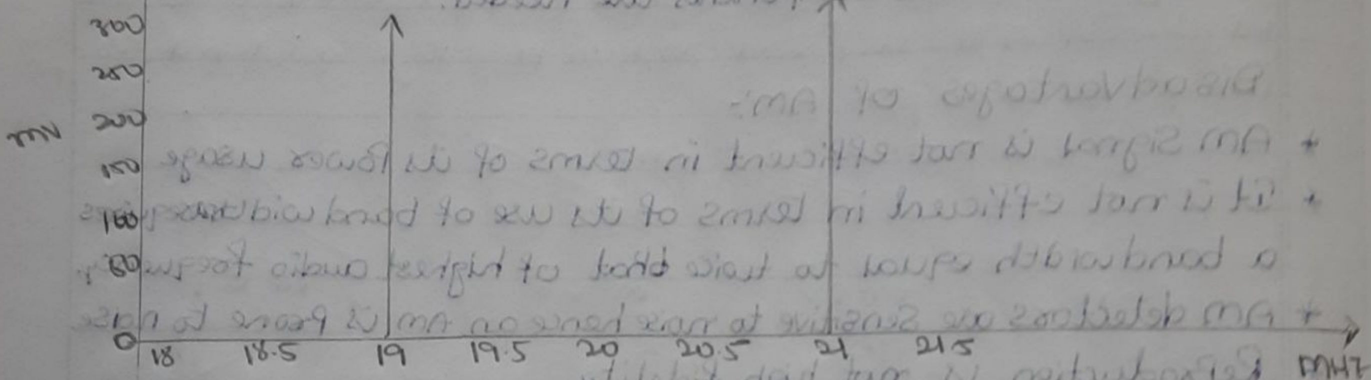
d) m=1


V
0.5
0
-0.5
-1
0
0.2
0.4
0.6
0.8
1.0
1.2
1.4
1.6
1.8
ms

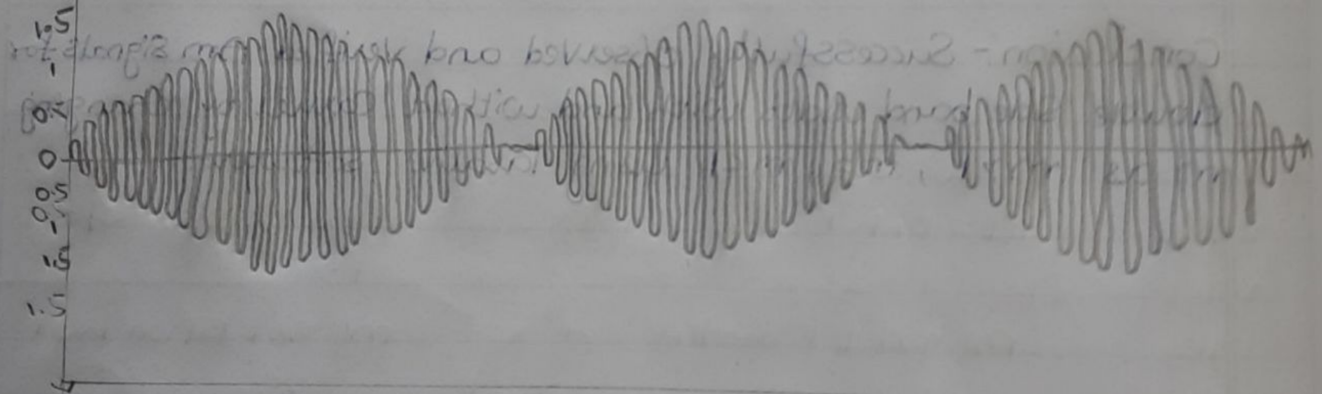

mV
250
200
150
100
50
0
18
18.5
19
19.5
20
20.5
21
21.5
mHz

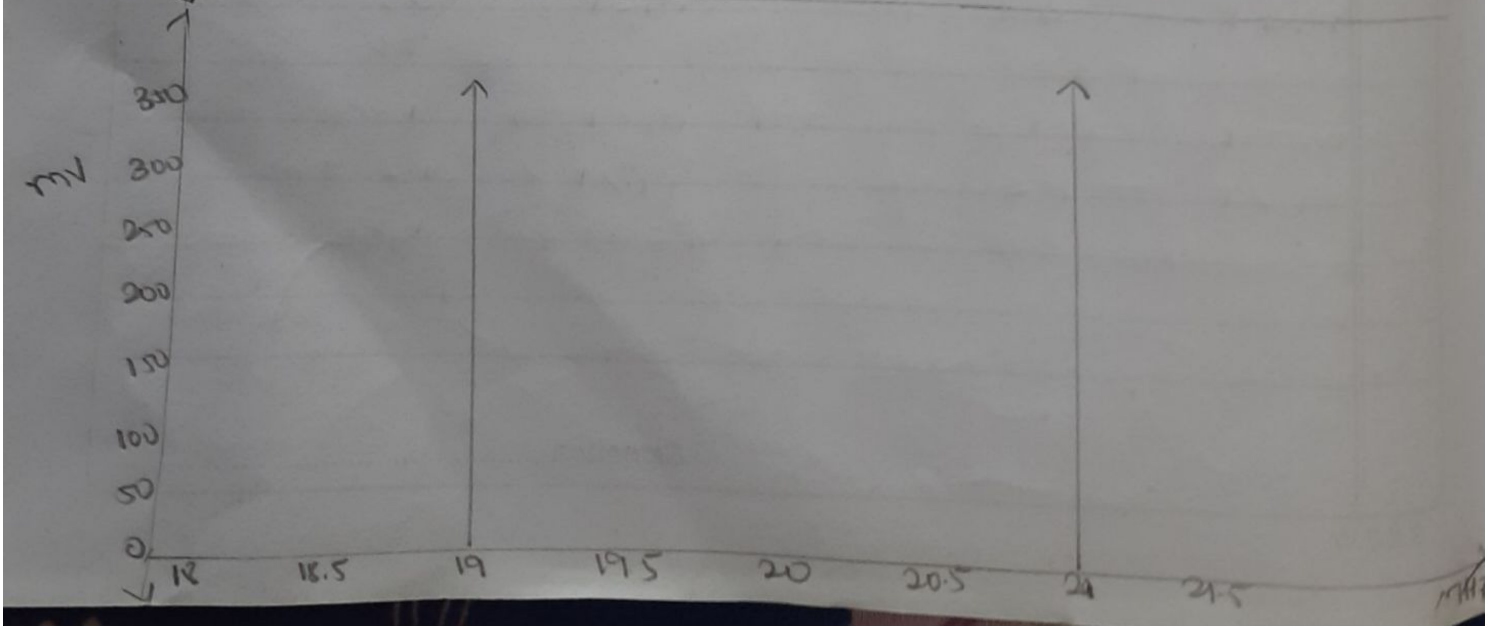
4] m=1.2


V
1.5
1
0.5
0
-0.5
-1
-1.5
0
0.2
0.4
0.6
0.8
1.0
1.2
1.4
1.6
1.8


mV
300
250
200
150
100
50
0
18
18.5
19
19.5
20
20.5
21
21.5
MHz

5] m=1.5


1.5
1
0.5
0
-0.5
-1
-1.5


mV
350
300
250
200
150
100
50
0
18
18.5
19
19.5
20
20.5
21
21.5

Expt.Name : Frequency Modulation and Demodulation

Aim:- To demonstrate Frequency modulation (FM) and demodulation Process by observing waveforms, spectra in time domain and spectra in frequency domain by varying the Parameters of message signal. Draw waveform x spectra. use virtual mode with appropriate software

Apparatus/softwares:- MATLAB, Labalive.

Theory:-

Theoretical Background:- Amplitude modulation was first modulation type to be considered in analog combination system. Althoug it had obvious advadage of being simple and relatively band width efficient. It has some disadvantages as well.

→ Since message is embedded in amplitude of carrier, the cost Performance and the size of amplifiers are difficult to accomplish for obtaining fair Performance in AM systems.

→ when message goes through a quiet Period in Double sideband or single side Band systems, very small carrier signals are transmitted. the absence of signal tends to accentate the noise.

→ The Passband width is small compared to other modulation schemes ie wifi

A major improvement in Performance in transmission is acheived with angle modulation. In this modulation amplitude of carrier is kept constant. Angle modulation Provides improvised noise Performance. Frequency modulation technique is analysed under ~~freq~~ angle modulation.

Frequency Modulation:- It is the Process of varying frequency of carrier signal linearly with message signal.

* The FM-modulated signal has its instantaneous frequency that varies linearly with amplitude of signal. Now we can get the FM-modulation by following:- $\phi(t) = \cos(2\pi f_c t + 2\pi k_f \int_0 m(t)dt)$

where k_f is sensitivity factor and represents frequency deviation

Signature ....

rate as a result of message amplitud change. The instantaneous frequency is $\omega_i = 2\pi f_c + 2\pi k_c m(t)$

modulation Index :- amount of change in carrier frequency produced by amplitude of input signal, is called frequency

modulating signal : $m(t) = E_m \cos(2\pi f_m t)$

carrier signal : $E_c \sin(2\pi f_c t)$

Freq of FM signal :- $F(t) = f_c + k_f m(t) \Rightarrow F(t) = f_c + k_f \cos(2\pi f_m t) E_m$

$\Delta f = k_f E_m$

$F(t) = f_c + \Delta f \cos(2\pi f_m t) \Rightarrow \Delta f$ = frequency deviation

max frequency deviation $\Rightarrow f_c = \Delta f$

FM Signal :-

$$y_{FM}(t) = E_c \sin(\omega_c t + k_f 2\pi \int_0^t m(t) dt)$$

$$= E_c \sin(\omega_c t + k_f 2\pi \int E_m \cos(2\pi f_m t) dt$$

$$= E_c \sin(\omega_c t + k_f 2\pi / 2\pi f_m \times E_m \sin(2\pi f_m t)$$

$$= E_c \sin(\omega_c t + k_f E_m / f_m \sin(2\pi f_m t)$$

$$= E_c \sin(\omega_c t + \frac{\Delta f}{f_m} \times \sin(2\pi f_m t))$$

modulation Index of FM $= \frac{\Delta f}{f_m}$; $m_f = \frac{\Delta f}{f_m}$

Frequency modulation equation $\Rightarrow y_{FM}(t) = E_c \sin(\omega_c t + m_f \sin(2\pi f_m t))$

Band width :-

FM signal spectrum is quite complex and will have infinite number of side bands as shown in figure. This figure gives idea, how spectrum expands as the modulation index increases. Side bands are seperated from the carrier by $f_c \pm f_m$, $f_c \pm 2f_m$, $f_c \pm 3f_m$ and so on

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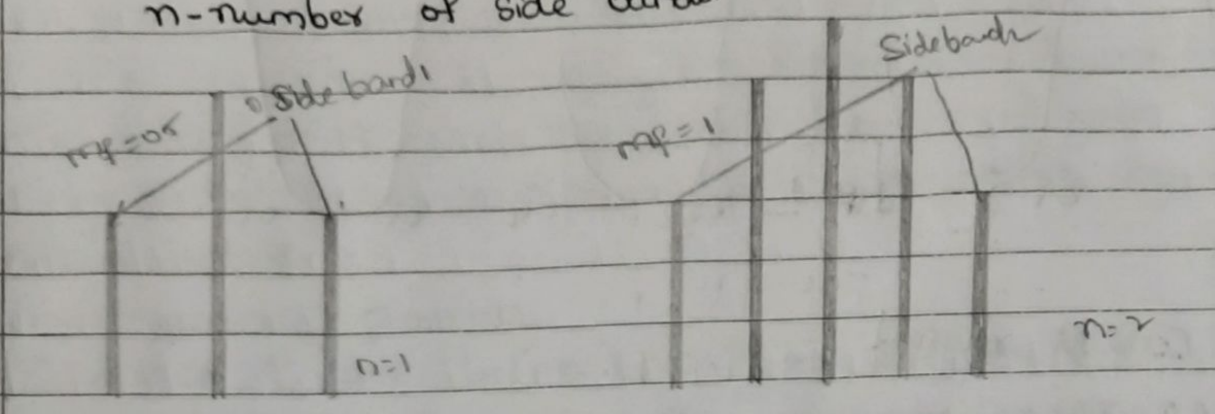
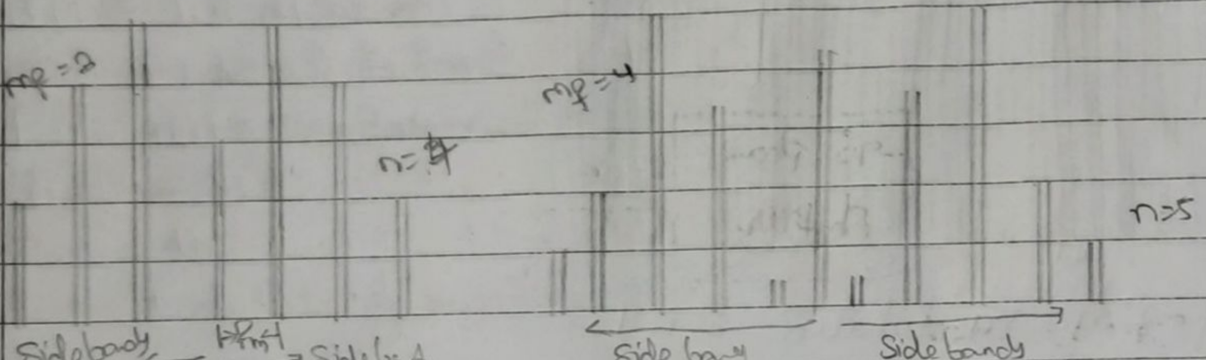
n-number of side bands.





FM Demodulation:- It is the key process in the reception of a frequency modulated signal. once signal has been received, filtered and amplified, it is necessary to recover the original modulation from the carrier. It is this process called demodulation
Demodulation using differentiation → A frequency discrimination theoretically extracts message from received FM signal.

Received FM signal → d/dt → Envelope Detection → DC Blocker → m(t)

Difference between AM and FM:-
→ In FM, carrier amplitude is constant, therefore transmitted power is constant
→ Transmitted power doesn't depend on the modulation index.

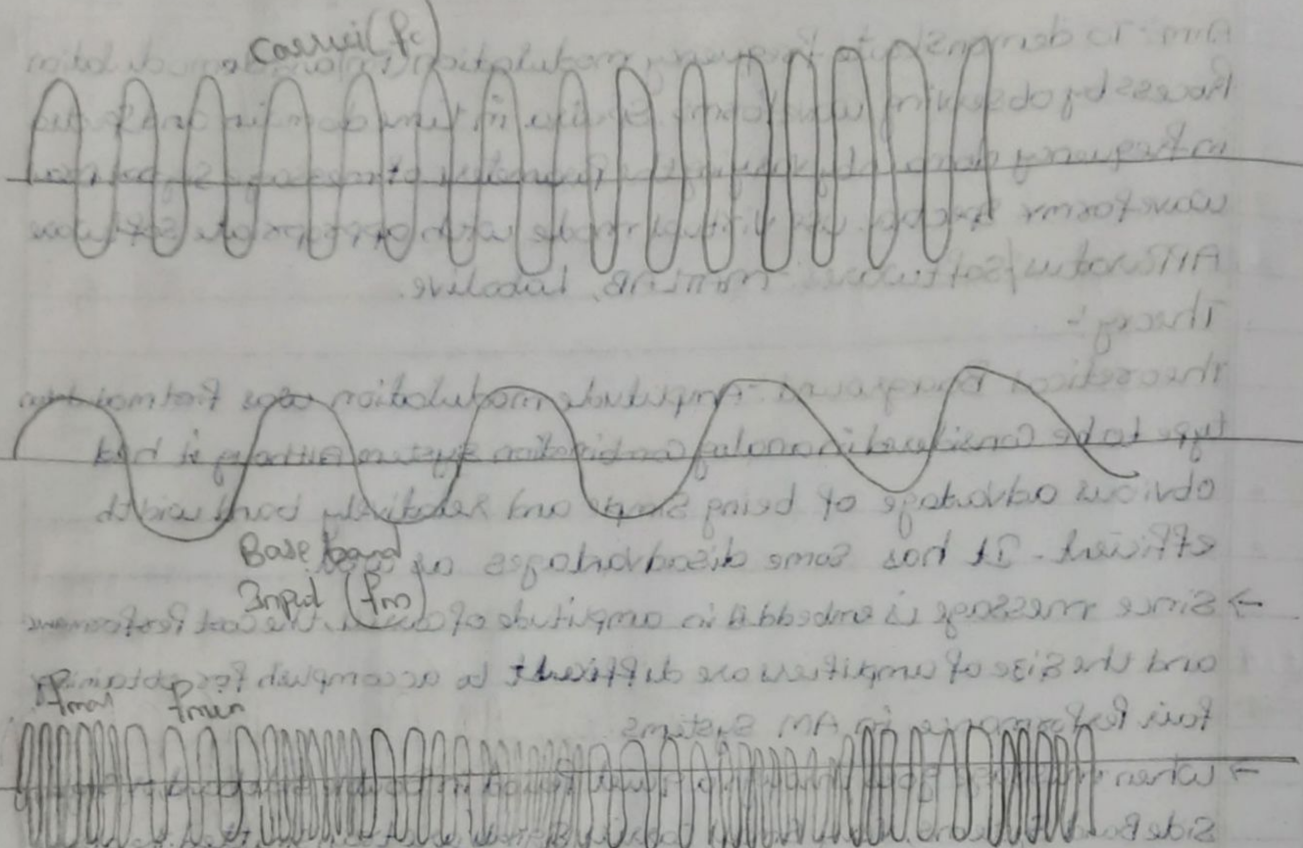
Signature ....

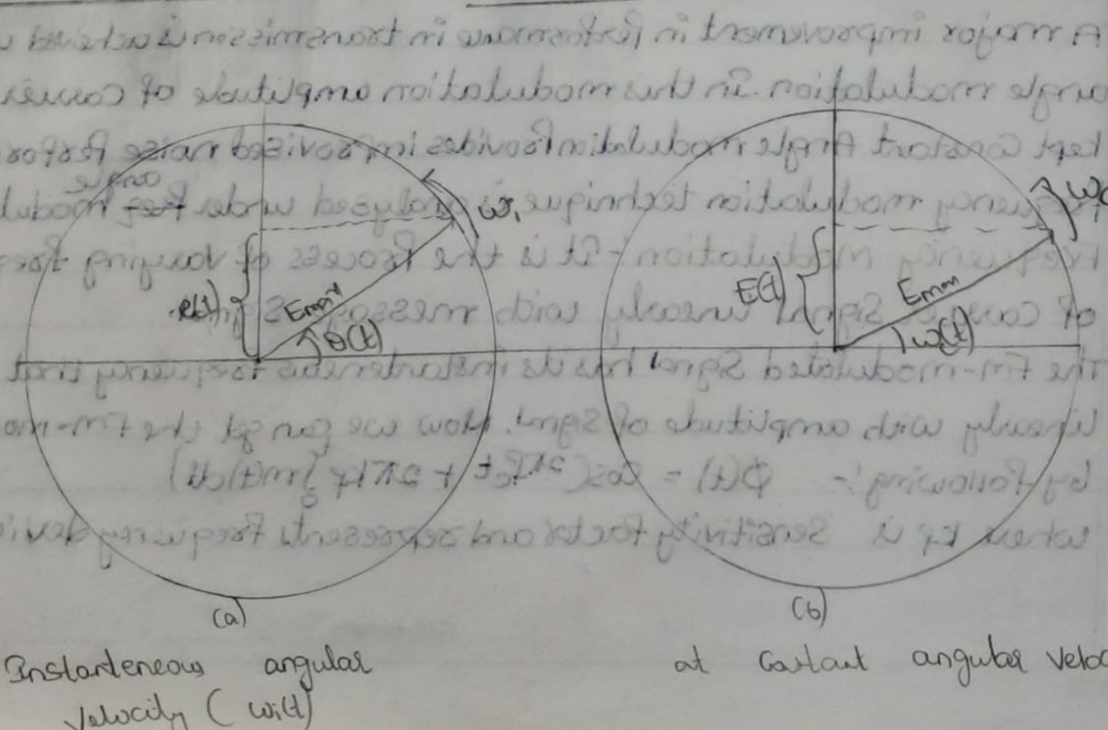
→ The number of Significant Sidebands in FM is large.
→ FM has better noise immunity. FM is rugged/robust against noise
The quality of FM will be good even in Presence of noise
→ circuits for FM transmitter are very complet and very expensive.

Matlab Code:-

```
clc; clear all; close all;
fs = 5000; fc = 200; t=(0:1/fs:04)'; m=sin(2*pi*10*t);
%+sin(2*pi*30*t); fDev = 100; % Frequency Deviation value
%% FM Modulation
y= fmmod(m,fc,fs,fDev);
%% Plotting the Baseband signal
Subplot(311);
Plot(t,m);
title('Modulating/Baseband signal');
xlabel('Time-->');
ylabel('Amplitude-->');
%% Plotting the FM Signal
Subplot(312);
Plot(t,y);
title('FM modulated Signal');
xlabel('Time-->');
ylabel('Amplitude-->');
%% Frequency Demodulation
z=fmdemod(y,fc,fs,fDev); Subplot(313); Plot(t,z,'r');
title('Received Signal(originally Transmitted via FM)');
x label ('Time-->');
Y label ('Amplitude-->');
```

Signature ....



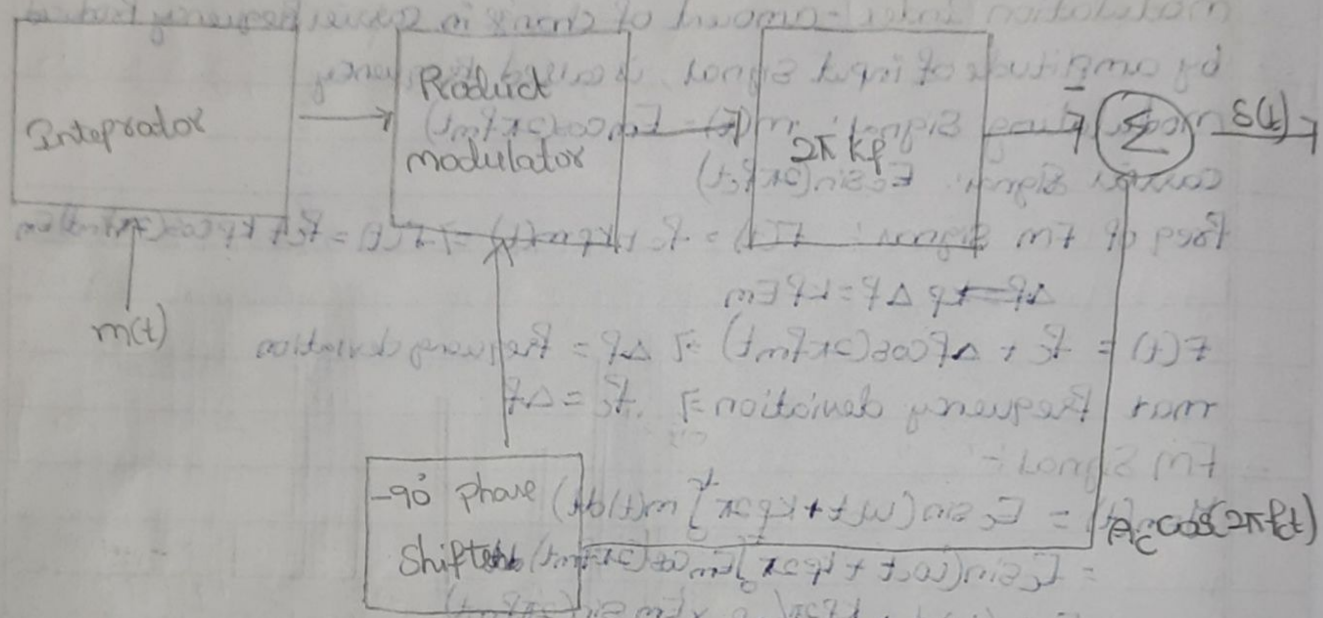


(a) Instantaneous angular Velocity ($\omega_i(t)$)

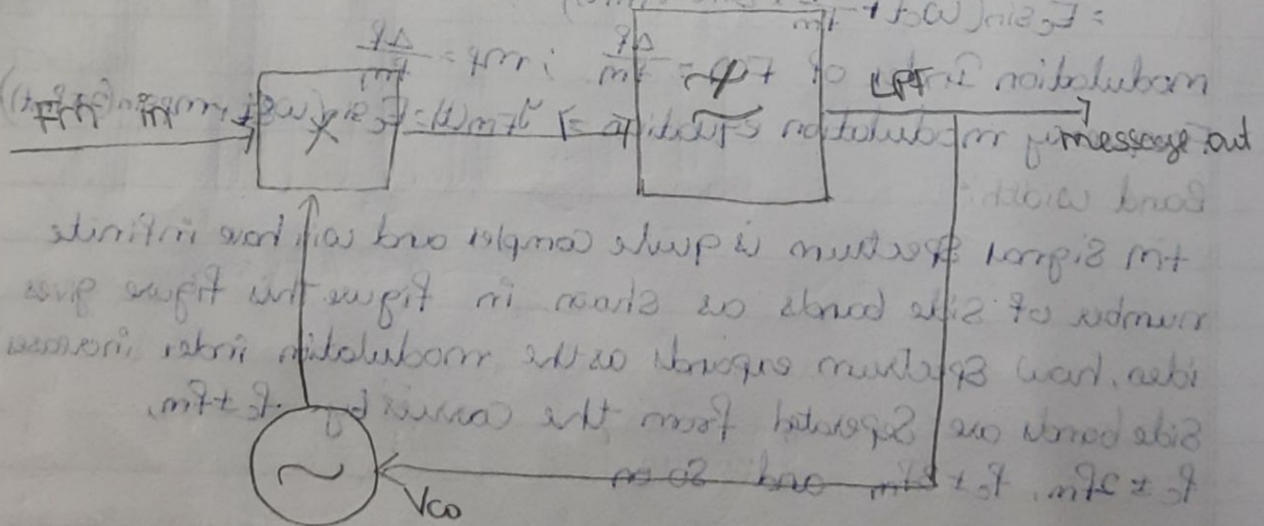
(b) at Constant angular Velocity (ω_c)

Block Diagram:

FM modulator :-

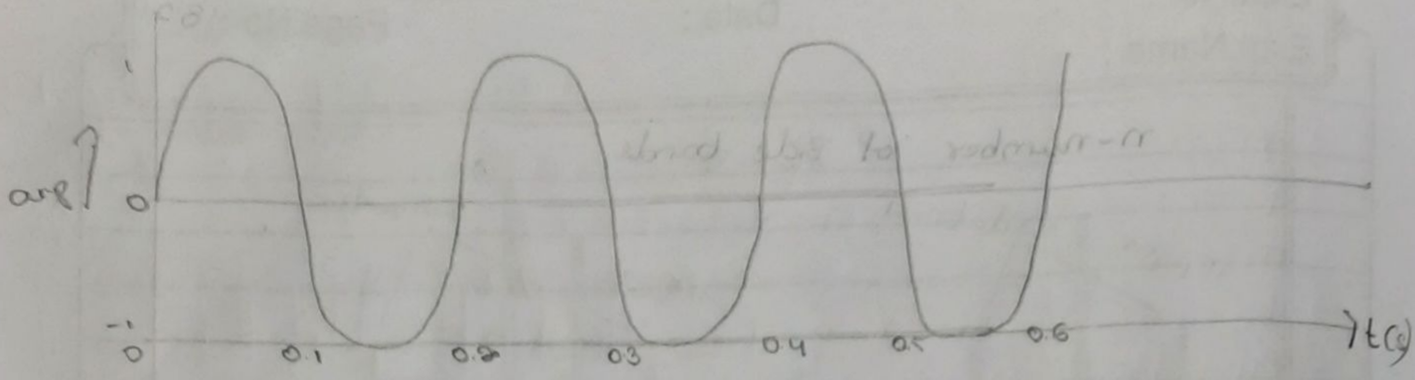


FM Demodulator :-

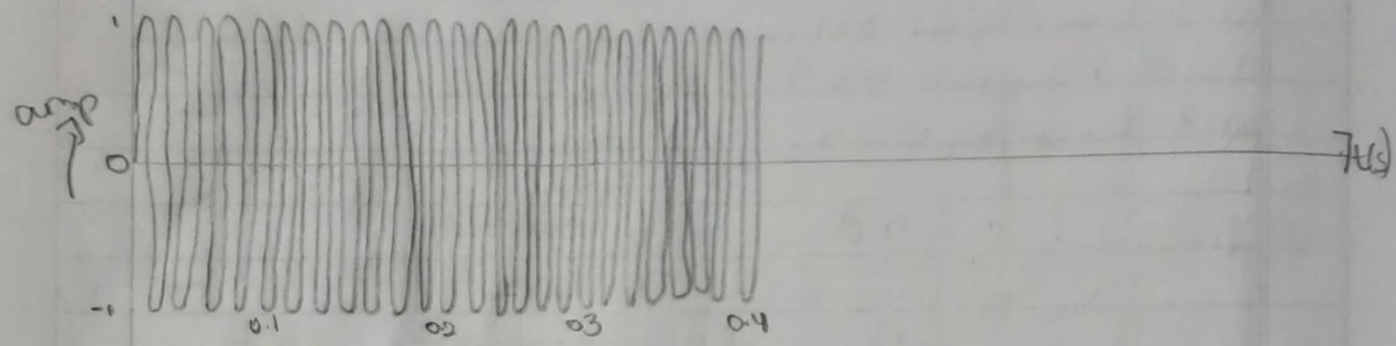


matlab observation :-

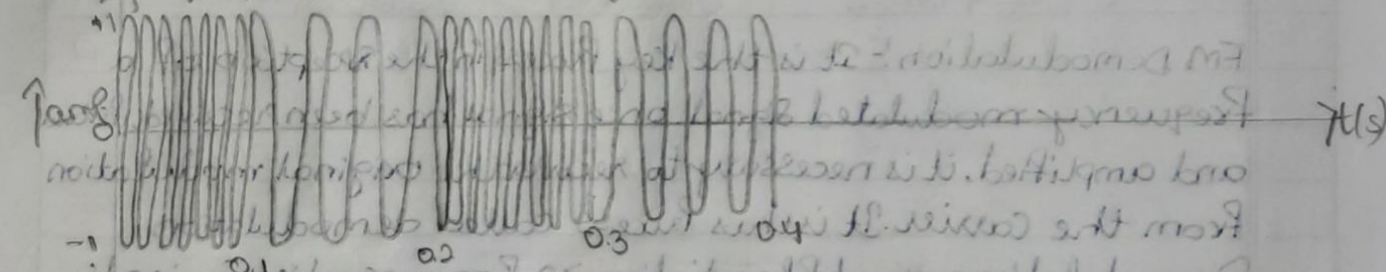
Input Signal →



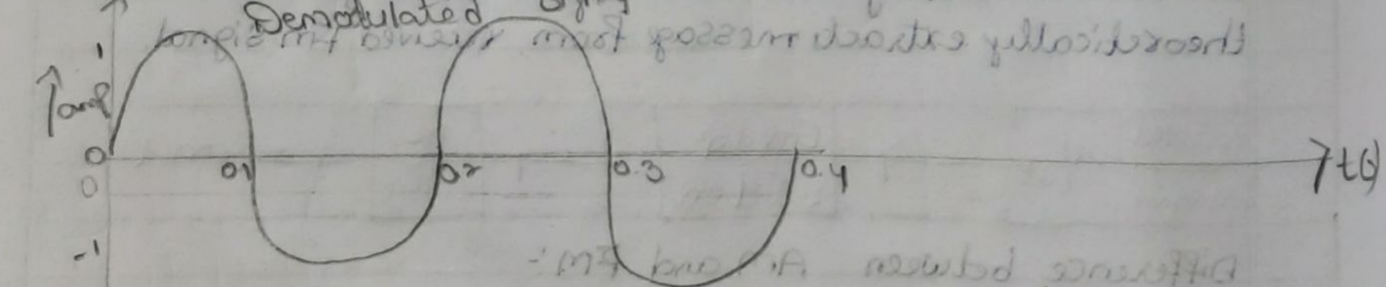
carrier Signal



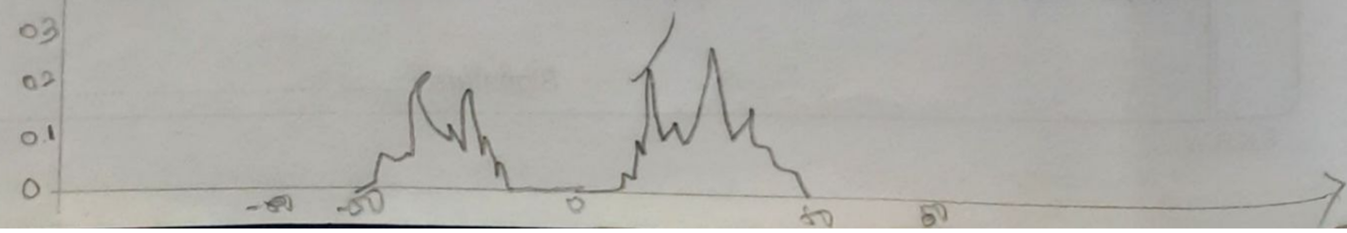
FM modulated signal



Demodulated Signal



magnitude of Spectrum of FM signal

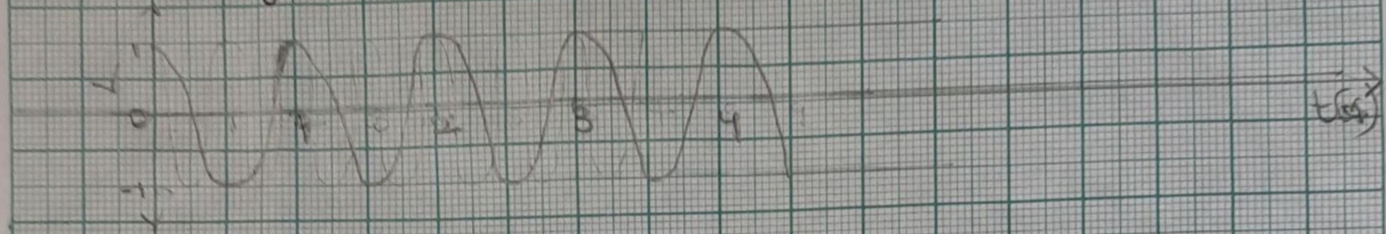


Labalive observations

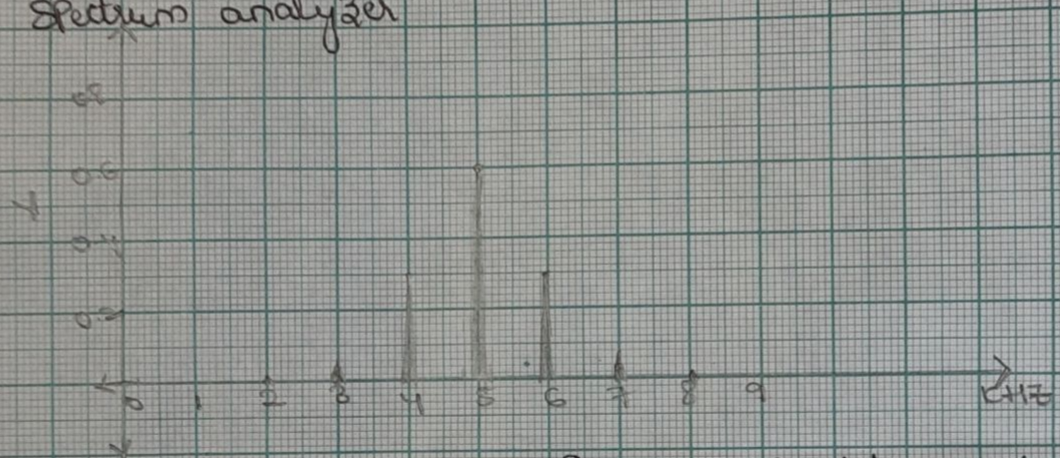
1) carrier f_c = 5KHz AC = 1V modulating signal f_m = 1KHz Am = 1V

modulation Constant = 5KHz/V

modulating signal

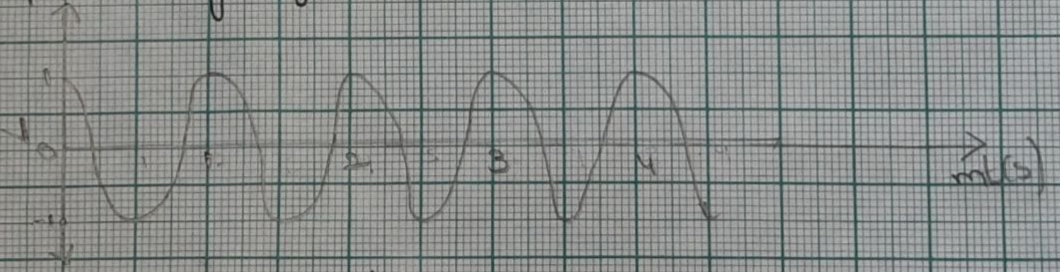


Spectrum analyzer



2) carrier frequency f_c = 5KHz AC = 1V f_m = 1KHz Am = 1V modulation Constant = 7.5KHz/V

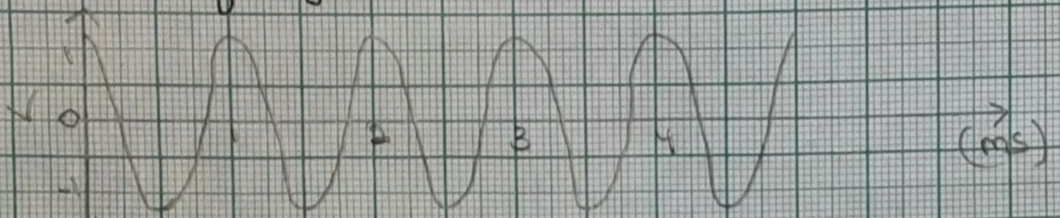
modulating signal



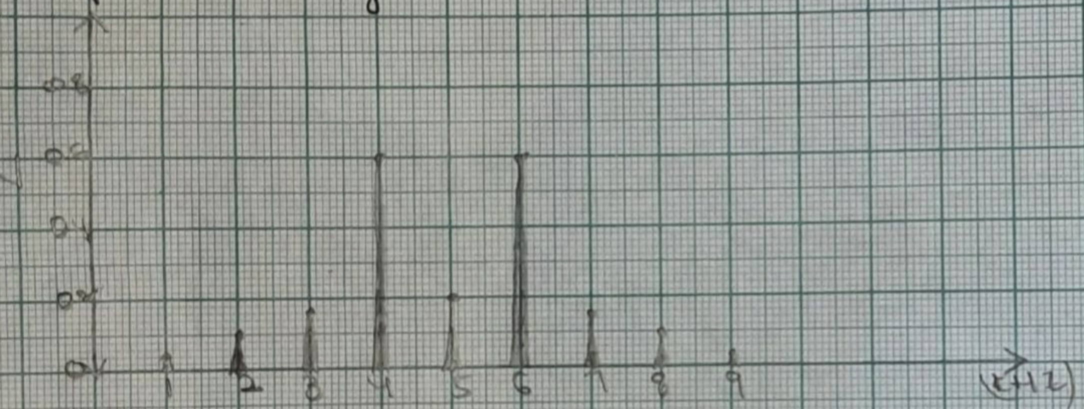
Spectrum analyzer

3) $f_c = 5kHz$ $A_c = 1V$ $f_m = 1kHz$ $A_m = 1.2V$ modulation constant = 8.8kHz/V

modulating signal

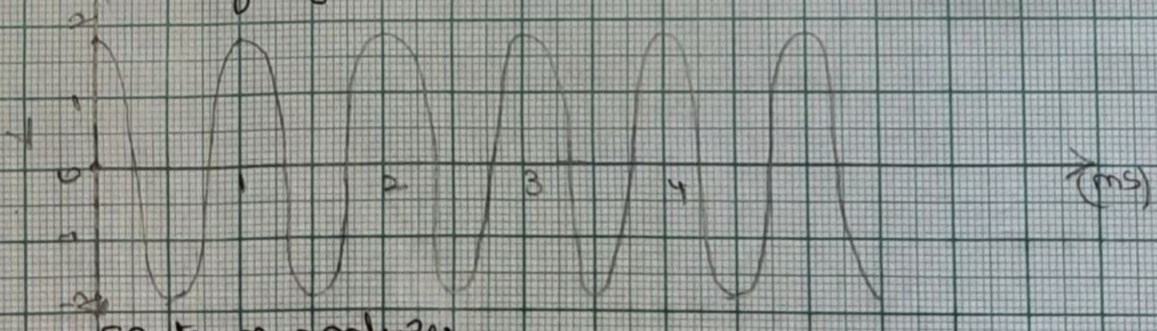


Spectrum analyzer

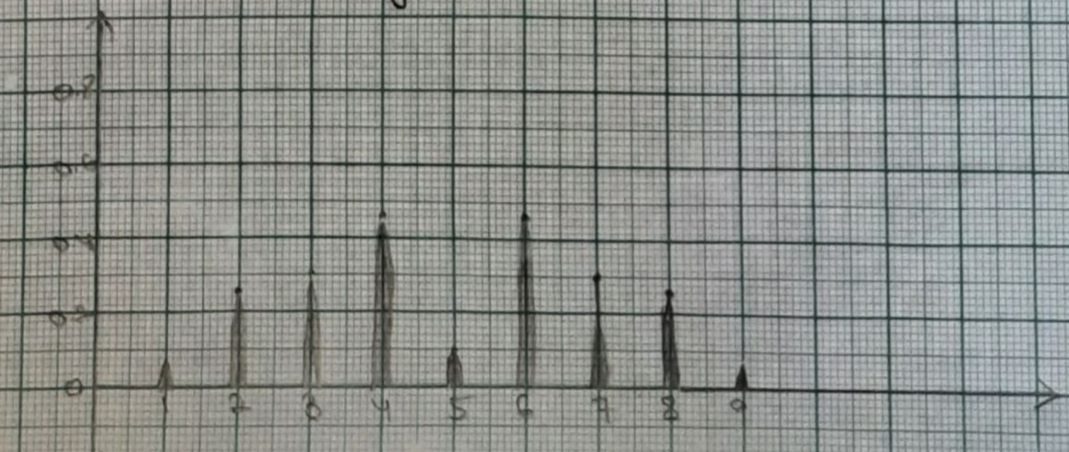


4) $f_c = 5kHz$ $A_c = 1V$ $f_m = 1kHz$ $A_m = 1.8V$ modulation constant = 9kHz/V

modulating signal



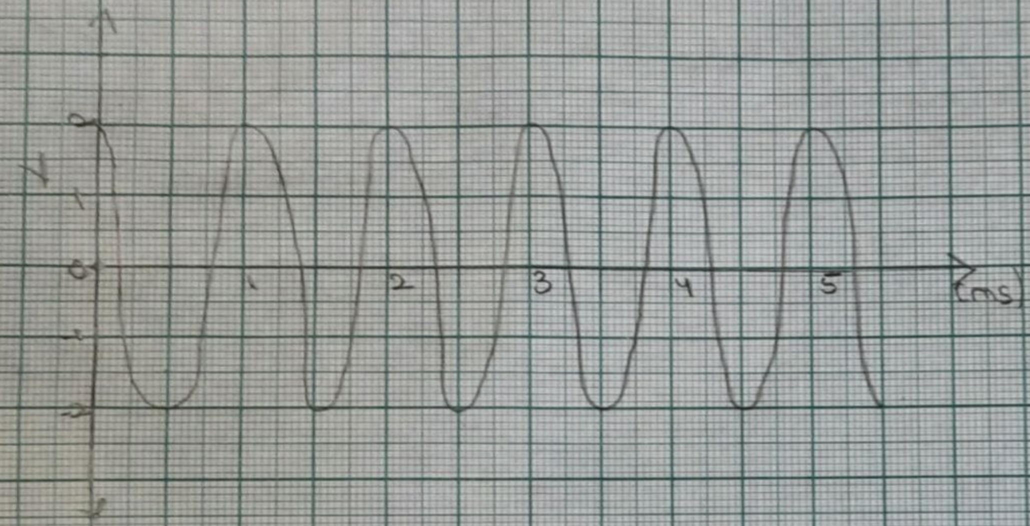
Spectrum analyzer



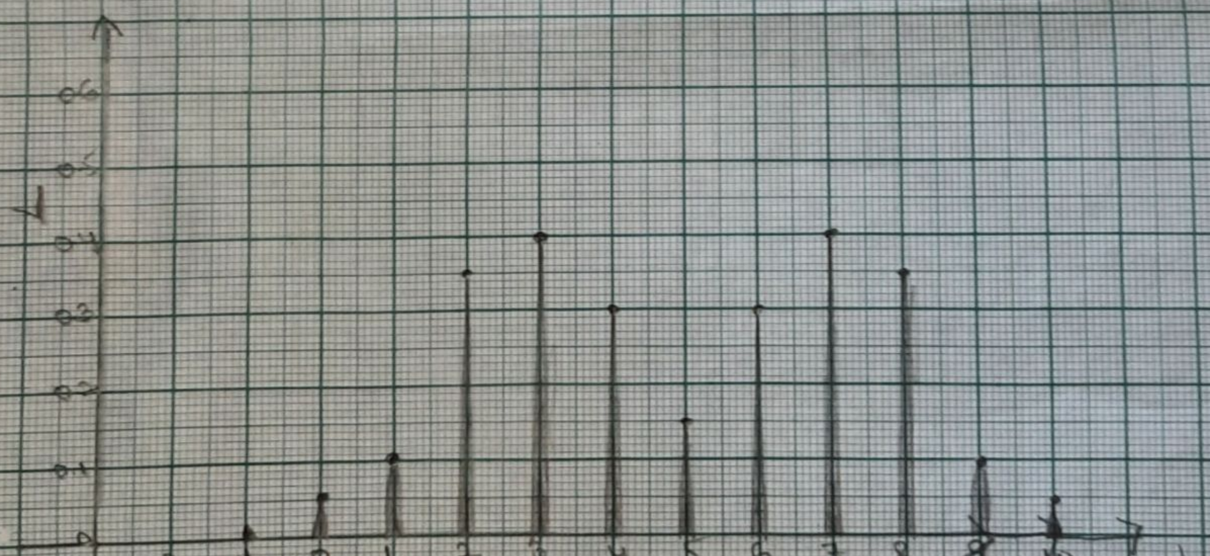
5) $f_c = 5\,kHz$ $A_c = 1V$ $f_m = 1\,kHz$ $A_m = 2V$

modulation Constant = 9.3 kHz/V

modulating signal



spectrum analyzer



Applications and advantages of Fm:-

1) Fm is resilient to noise and interference. Therefor it's used for high quality broadcast transmission
2) Fm is ideal for mobile radio communication application including more general two way radio communication or Portable applications where signal levels are likely to vary considerably.
3) It is used in magnetic force tape system record synthesis.
4) Radar, telemetry, observing infants for sezure through E.C.G, music.

Conclusion:- Successfully demonstrated frequency modulation and demodulation Process by observing waveforms in time domain and their spectra in frequency domain by varying Parameters of message signal.

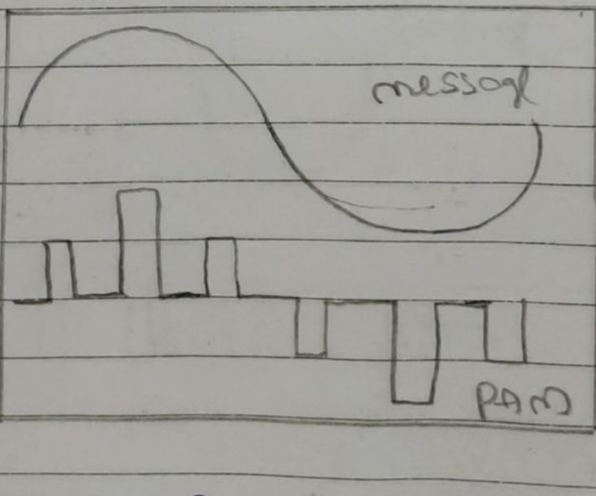
Aim:- To examine of Pulse amplitude modulation (PAM), Pulse Position modulation (PPM), Pulse width modulation (PWM) and verify and draw resutant waveforms. Illustrate circuit diagrams for PAM and PWM. show & draw outputs using matlab/Simulink using virtual mode.

Software/ Apparatus:- matlab

Theory:- To examine Pulse modulation is a type of modulation in which signal is transmitted in form of pulses. In Pulse modulation continuous signals are sampled at regular intervals.

Pulse modulation is further divided in to Analog and digital modulation and further analog in to PAM, PWM, PPM and Digital into PCM, DM.

PAM:- In PAM a pulse signal is used to sample an analog signal. The result is a train of constant-width pulses. The amplitude of each pulse is proportional to amplitude of message signal at the time of sampling. The PAM signal follows the amplitude of original signal as signal traces out the path of whole wave.



Properties:-

* Amplitude of Pulse is Proportional to amplitude of modulating signal.
* Bandwidth of transmission channel depends on the Pulse width.

Signature ....

* Instanteneous Power of transmitter Varies
* Noise interference is high.
* Similar to Amplitude modulation
* System is Complex to implement

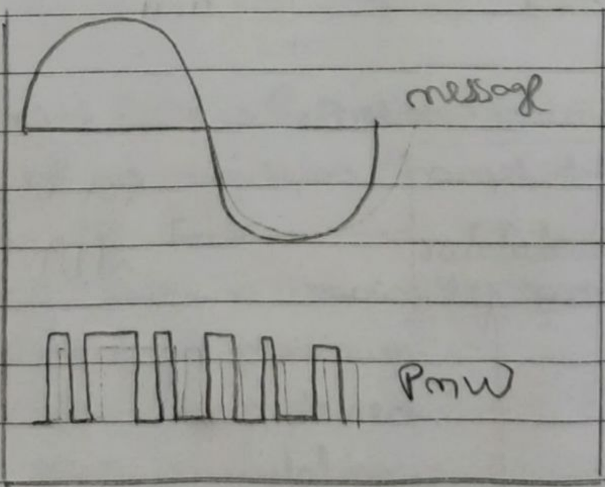
PAM Signal Generation:- We can generate PAM by two types:-

a) Natural Sampling b) Flat - top Sampling

Natural Sampling:- For a PAM signal Produced in the natural Sampling the sampled signal follows waveform of input signal during the time that each sample is taken.

Flat-top Sampling:- In this Sampled signal can be represented in Pulses for which amplitude of signal cannot be changed with respect to analog signal to be ~~seamed~~ Sampled.

PWM:- In this type, amplitude is maintained constant but duration or length or width of each is varied in accordance with instanteneous Values of analog signal.



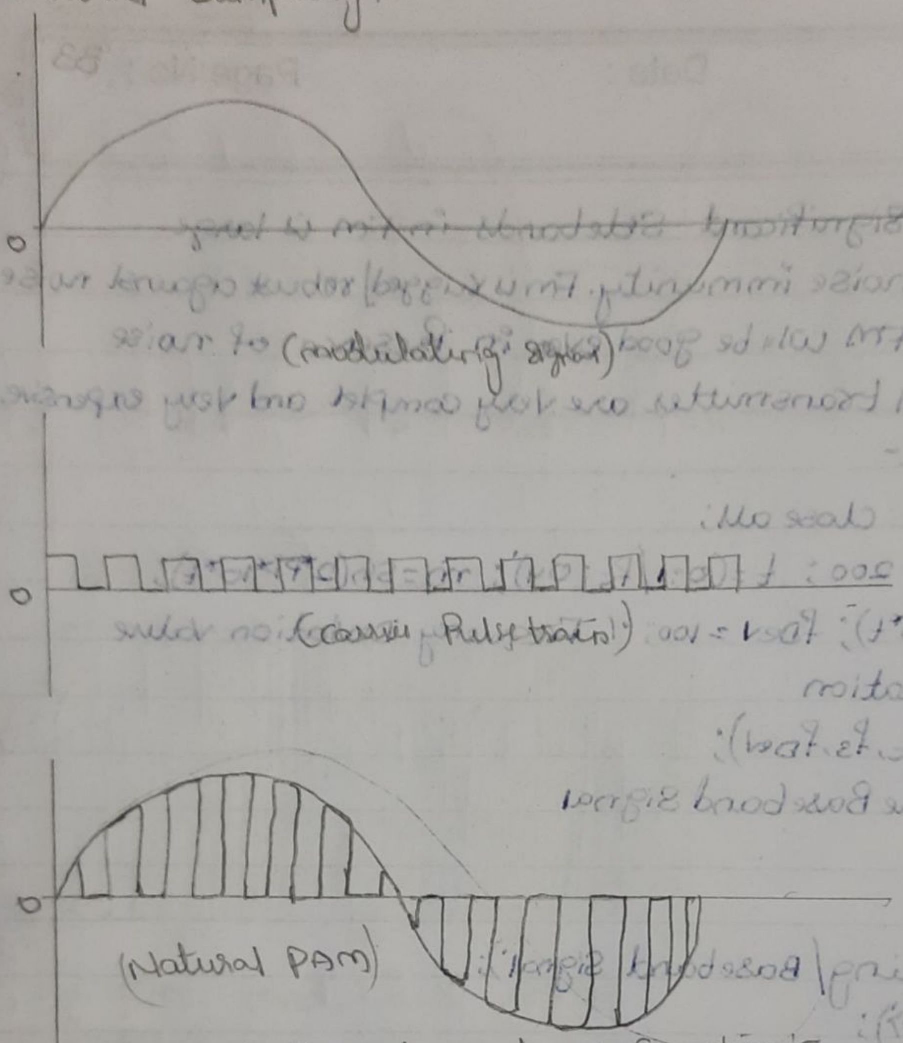
Properties:-

* width of Pulse is Proportional to amplitude of modulated signal
* Bandwidth of transmission channel depends on rise time of Pulse.

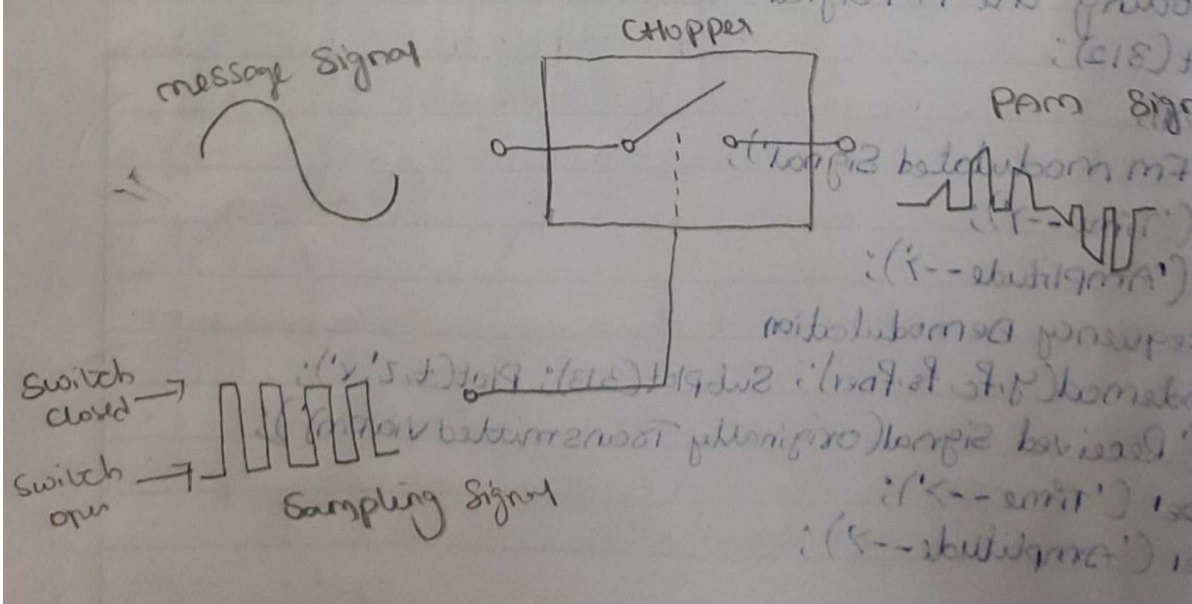
Signature ....

S.K.B.W

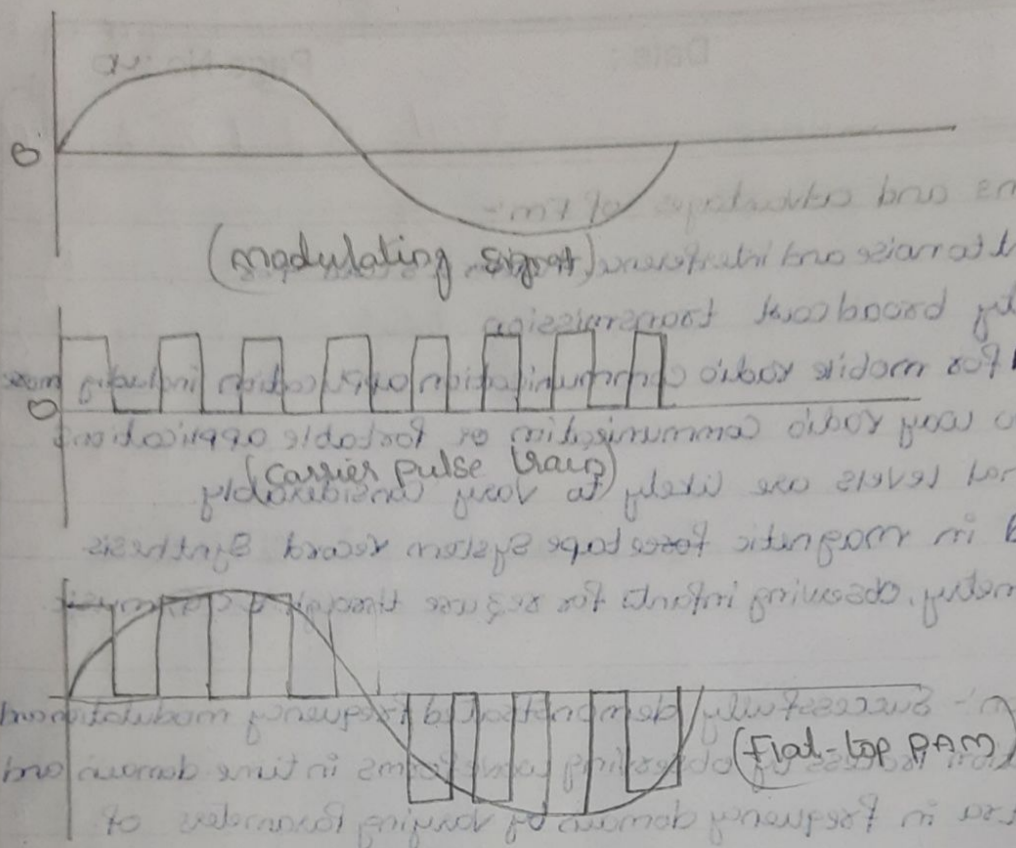
Natural Sampling :-



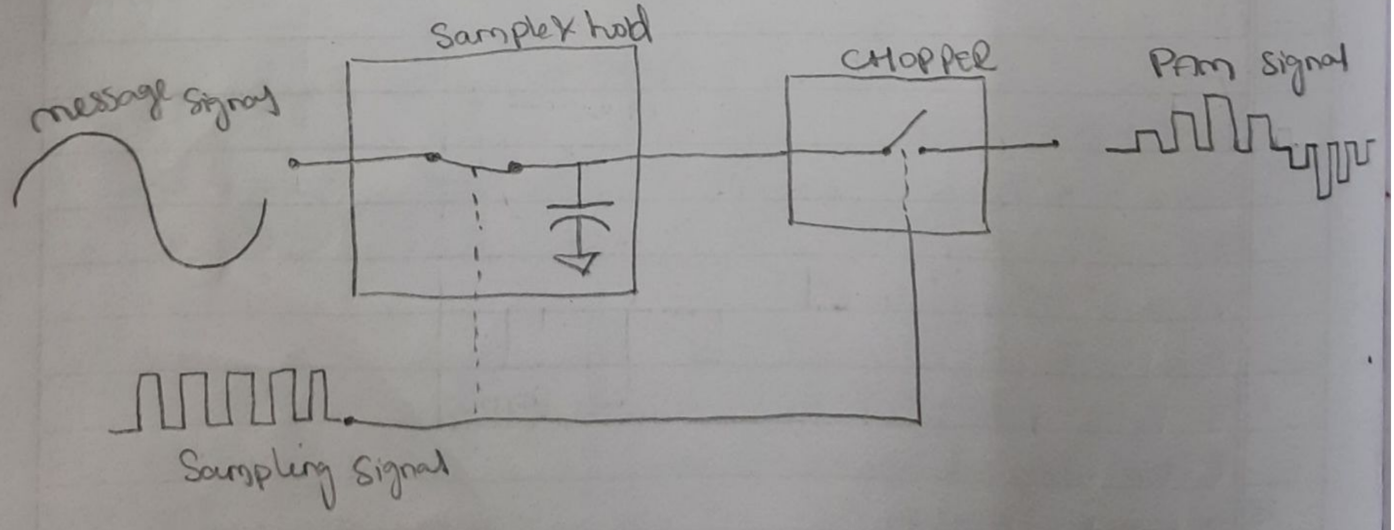
Generation of PAM by natural Sampling :-



Flat Top Sampling :-

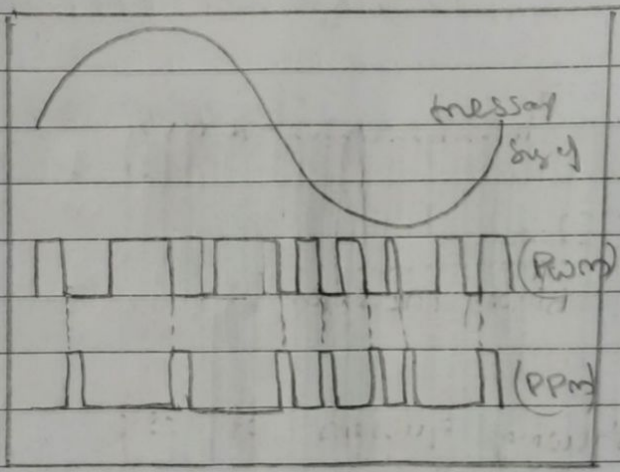


Generation of PAM by Flat-Top Sampling :-



* Instanteneous Power of transmitter varies
* Noise interference is minimum
* System is Simple to implement
* Similar to frequency modulation.

PPM:- In this type of modulation both amplitude and width of Pulse are kept constant. we vary the Position of each Pulse according to instanteneous Sampled values of message Signal. PPM modificat of PWM

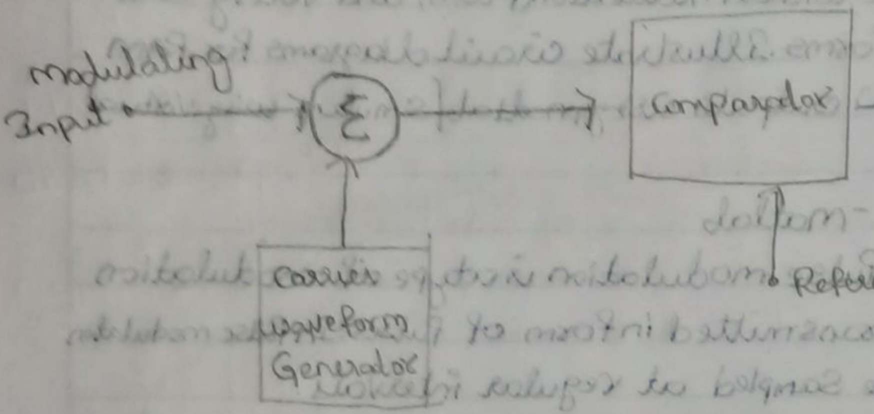


Properties:-

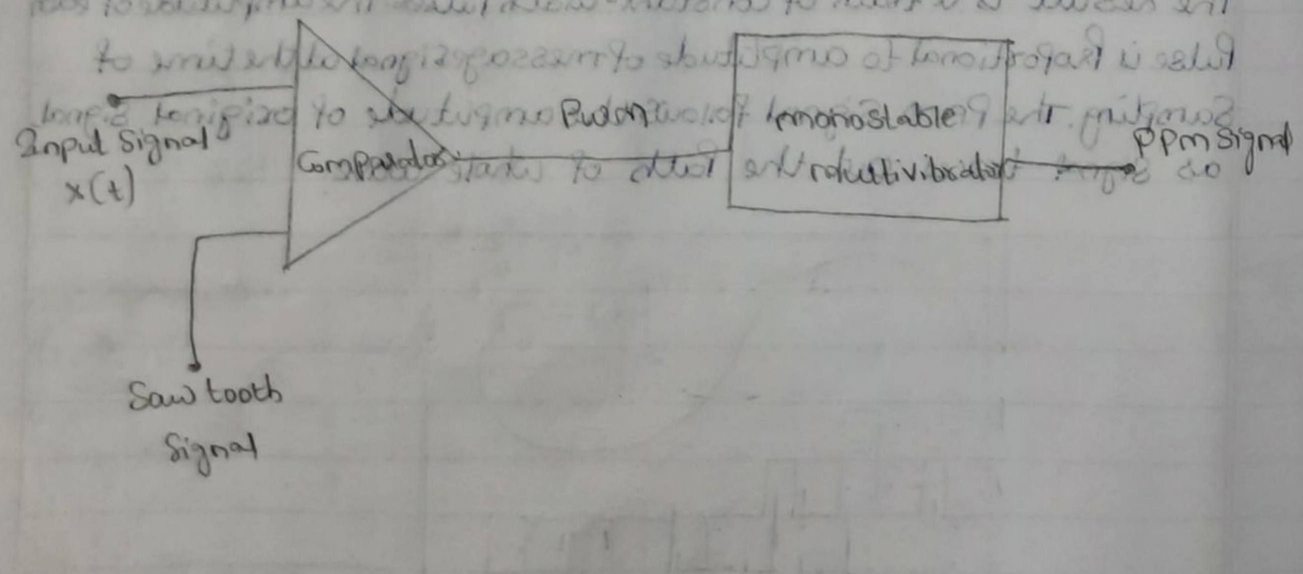
* Relative Position of Pulse is Proportional to amplitude of modulating Signal
* Band width of transmission channel depends on the rising time of Pulse.
* Instanteneous Power of transmitter remains Constant.
* Noise interference is minimum.
* System is Simple to implement
* Similar to phase modulation.

Signature ....

PMW Signal Generation :-

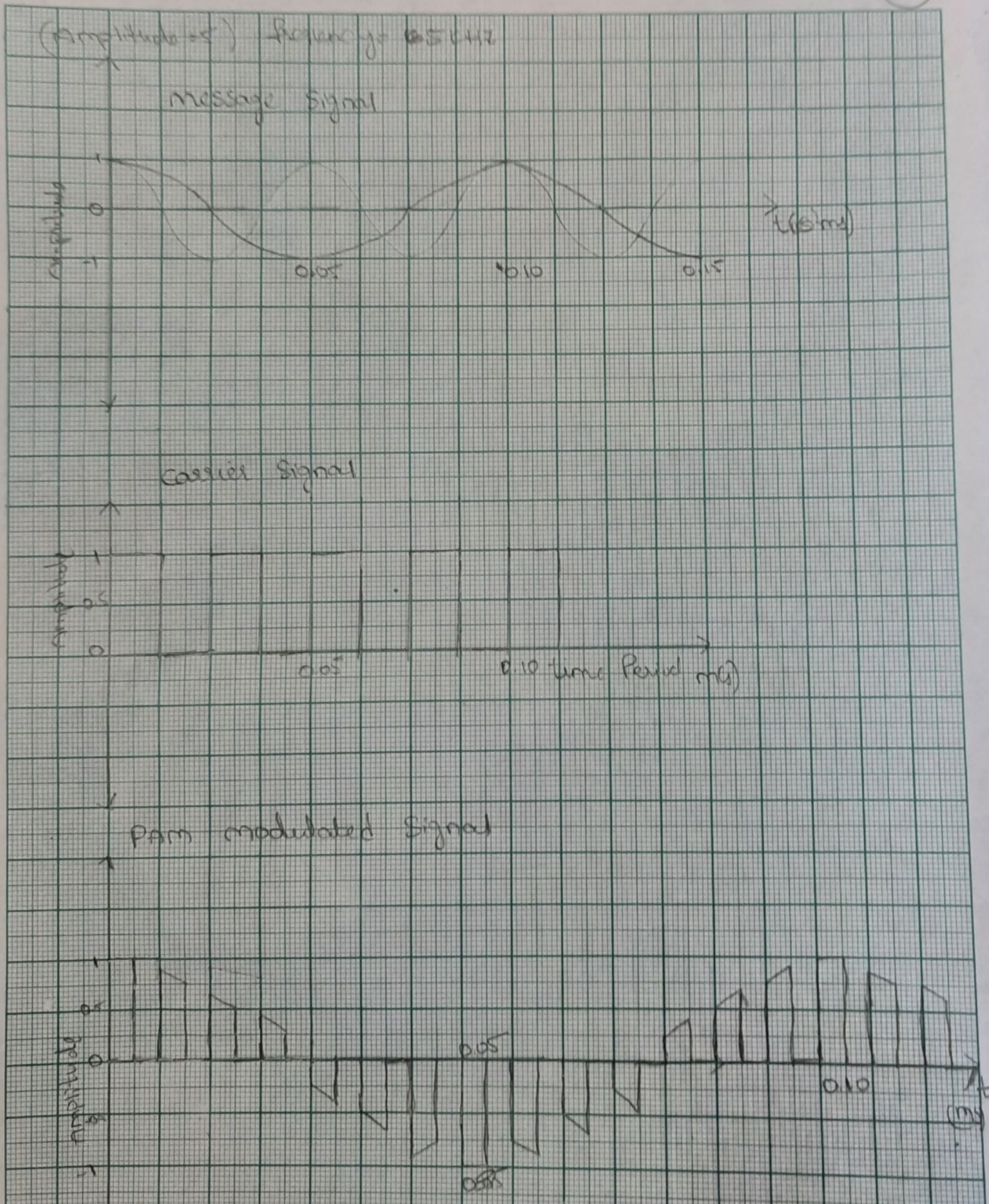


PPM Signal Generation



matlab code :-

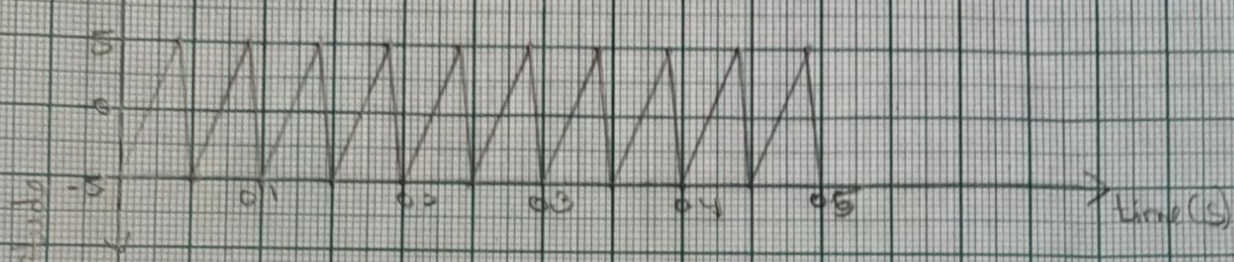
```
% PAM
clc;
clear all;
close all;
fc=100; fm=fc/10; fs=100*fc;
t=0:1/fs : 4/fm; msg_sg1 = cos(2*pi*fm*t)
carr_sg1 = msg_sg1.*carr_sg1;
tt=[]
for i=1: length(mod_sg1);
    if mod_sg1(i)=0;
        tt = [tt, mod_sg1(i)];
    else
        tt= [tt, mod_sg1(i)+2];
    end
end
figure(1)
subplot(4,1,1); plot(t,msg_sg1,'m'); title('message signal');
xlabel('Time Period'); ylabel('Amplitude');
subplot(4,1,2); plot(t,carr_sg1); title('~~Message Sig~~ Carrier signal')
xlabel('Time Period'); ylabel('Amplitude');
subplot(4,1,3);
plot(t,mod_sgl,'r');
title('PAM modulated Signal');
x label('Time Period');
y label('Amplitude');
```


message signal
Amplitude
1
0
-1
0.05
0.10
0.15
t(s ms)
carrier signal
Amplitude
1
0.5
0
0.05
0.10 time Period ms)
PAM modulated signal
Amplitude
1
0
-1
0.05
0.10
t
(ms)

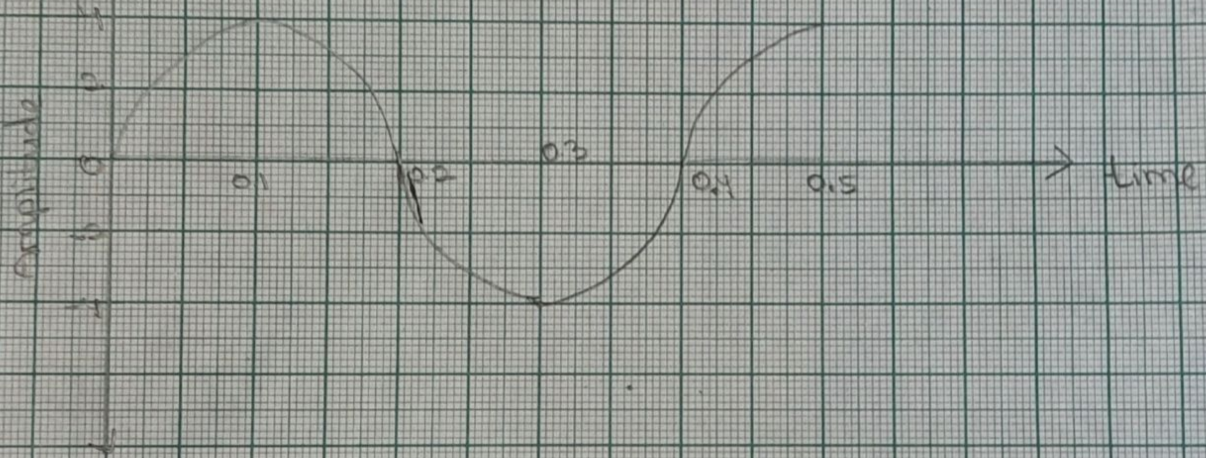
```
% PWM
clc;
close all; clear all;
t=0:0.0001:1;
s=sawtooth(2*pi*10*t+pi); m=0.75*sin(2*pi*1*t);
n=length(s);
for i=1:n
 if (m(i)>=s(i))
 pwm(i)=1;
 elseif (m(i)<=s(i))
 pwm(i)=0
 end
end
plot(t,pwm,'o',t,m,'r',t,s,'b');
ylabel('Amplitude'); axis([0 1 -1.5 1.5]);
xlabel('Time index'); title('Pmw wave');
grid on;
% PPM
clc; clear all; close all;
fc=1000; fs=10000; fm=200;
t=0:1/fs:((2/fm)-(1/fs));
x=0.5*cos(2*pi*fm*t)+0.5; y=modulate(x,fc,fs,'ppm');
subplot(2,1,1); plot(x);
title('msg signal');
subplot(2,1,2); plot(y);
axis([0 500 -0.2 1.2]);
title('PPM');
```

Signature ....

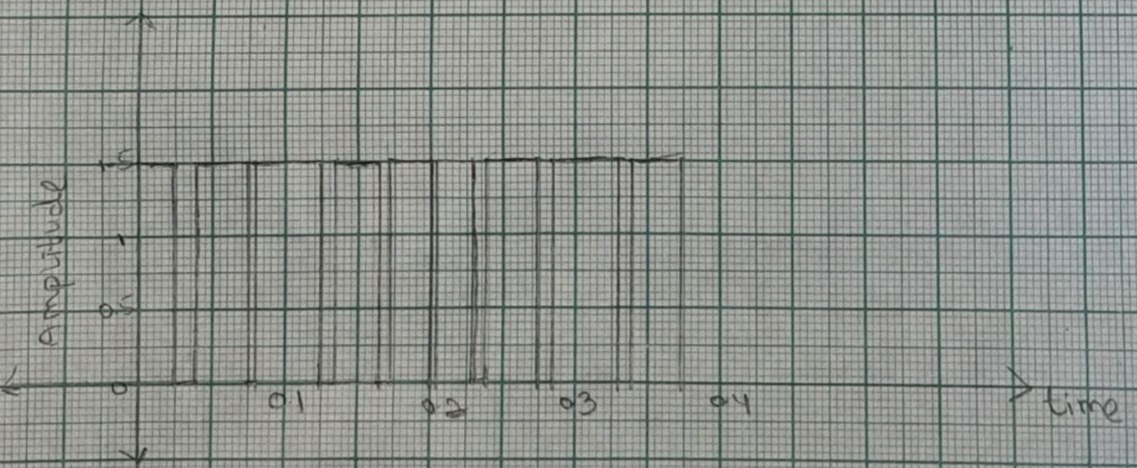
Carrier Sawtooth wave -

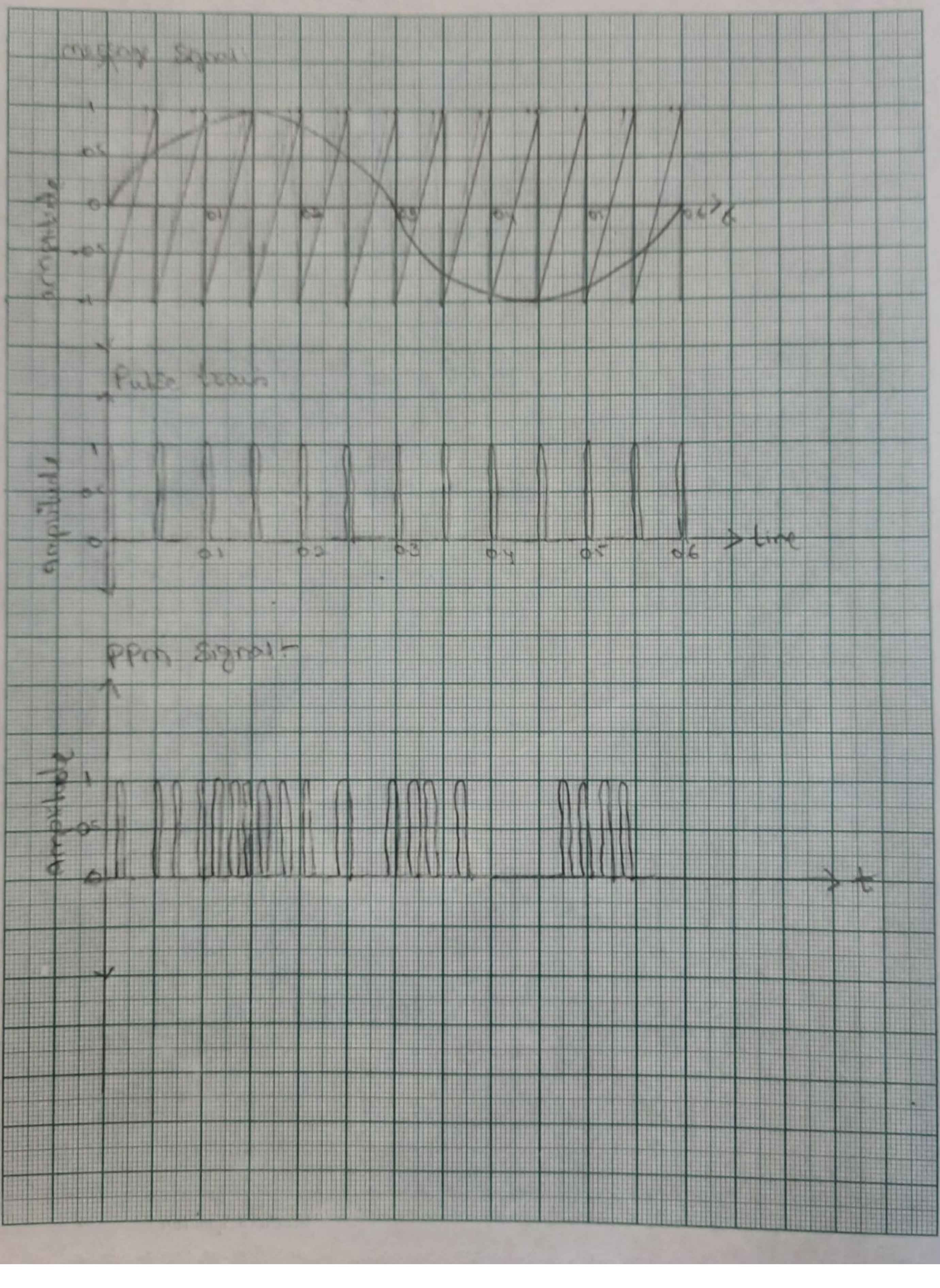


Message signal



Plot of PWM:-




message signal
amplitude
Pulse train
amplitude
time
PPM signal:-
amplitude
t

Advantages of Pulse modulation:-

* Pulses are quite short as compared to time in between, so a Pulse modulated wave remains off most of the time.
* Time interval between Pulses may be filled with sample values from other messages, so we can send many messages at a time on Pulse communication system.
* one of chief advantages of PM is that if we combine Pulse modulation with continuous modulation(AM, FM) we can obtain 'multichannel' communication system, a desirable feature for data transmission.

Conclusion:- Successfully Performed PAM, PWM, PPM and analysed and verified its input and output wave forms using matlab software.

Aim:- To study of amplitude shift keying (ASK), Frequency shift keying (FSK), Phase shift keying (PSK) modulation Technique and verify waveforms. Illustrate schematic diagrams for ASK, FSK and PSK. Show & draw the input/output waveforms using matlab code using virtual mode.

Software/ Apparatus:- matlab

Theory:-

ASK:- In case of amplitude shifting keying the amplitude of the resultant output (modulated) depend upon the input data. This is also a type of Amplitude modulation which represent the binary data in form of variation in amplitude of signal.

ASK is a digital modulation technique defined as process of shifting the amplitude of carrier signal between two levels, depending on whether 1 or 0 is to be transmitted.

Let the message signal be binary sequence of 1's and 0's. It can be represented as function of time as follow

$$V_m = \begin{cases} V_m, & \text{when Symbol is } 1 \\ 0, & \text{when Symbol is } 0 \end{cases}$$

Let the carrier be defined as $V_c = V_c \cos \omega t$

then corresponding ASK signal is given by product of V_m & V as

$V_{ASK} = V_m V_c \cos \omega t \rightarrow$ Symbol is 1

$= 0 \rightarrow$ Symbol is 0

FSK:- In case of frequency shifting keying, output signal will be either high or low, depending upon the input data applied.

FSK is digital modulation technique in which frequency of carrier signal varies accordingly to discrete digital changes. FSK is a scheme of frequency modulation.

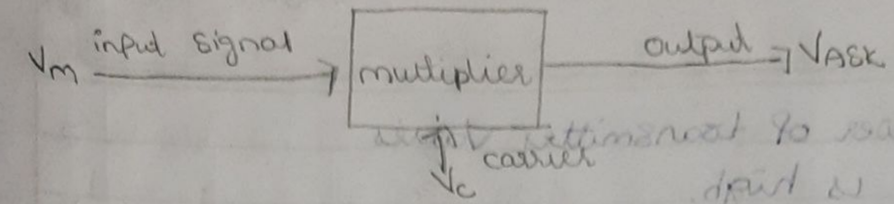
Let V_m be the message signal. $V_m(t) = V_m$

Signature ....

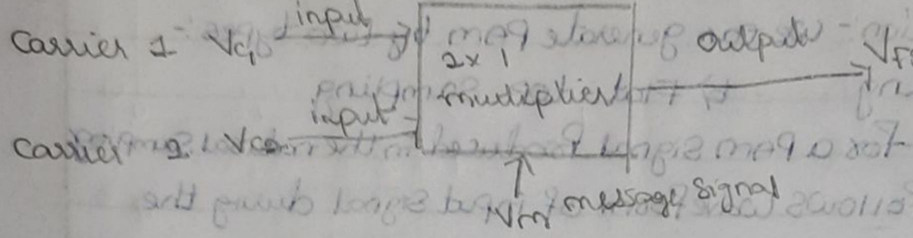
S.K.B.W

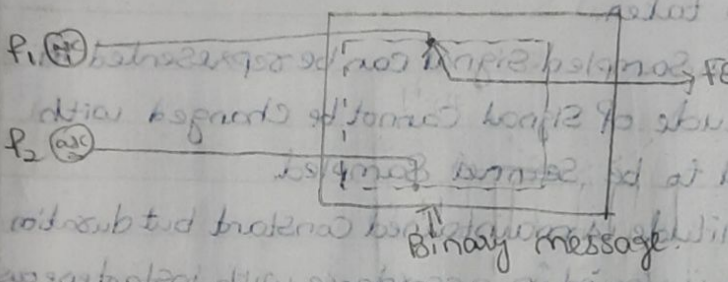
Block Diagram:

1) Block diagram of generation of ASK Signal →

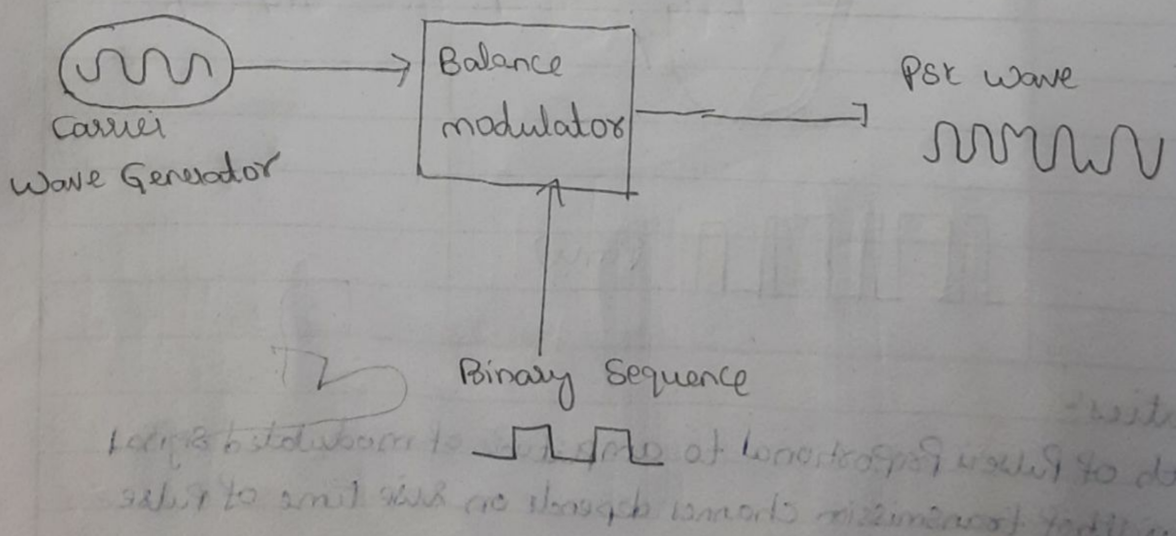


2) Block diagram of FSK Generator





3) Block diagram showing question of PSK



Let two carriers be defined as, $V_{c1} = V_c \cos\omega_1 t$; $V_{c2} = V_c \cos\omega_2 t$
then FSK corresponding signal is defined as
$V_{FSK} = V_m V_c \cos\omega_1 t$, Symbol is 1
$= V_m V_c \cos\omega_2 t$, Symbol is 0

PSK :- Phase shift keying (PSK) is digital modulation technique in which Phase of carrier signal is changed, by varying the sine and cosine input at a time. The phase of output signal gets shifted depending on input.

MATLAB CODE :-

```
% ASK
clc; close all; clear all;
fc = input('Enter the freq of Sine wave carrier:');
fp = input('Enter the freq of Periodic Binary Pulse (message):');
amp = input('Enter the amplitude (For carrier & Binary Pulse message):');
t=0:0.001:1;   c=amp.*sin(2*pi*fc*t);
Subplot(3,1,1); xlabel('Time'); Ylabel('Amplitude');
title('carrier wave');
m= amp/2.*square(2*pi*fp*t)+(amp/2);
Subplot(3,1,2);
Plot(t,m);
xlabel('Time');   Ylabel('Amplitude');
title('Binary message Pulses');
w=c.*m;
Subplot(3,1,3);
Plot(t,w);
xlabel('Time'); Ylabel('Amplitude');
title('Amplitude shift keyed signal');
```

Signature ....

S.K.B.W

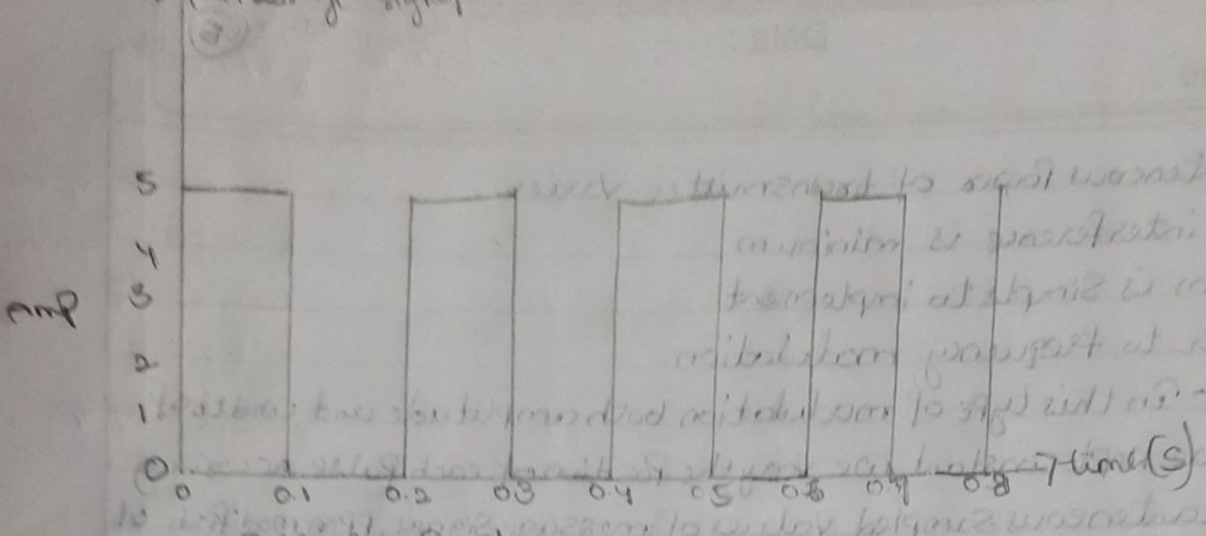
1)

ASK

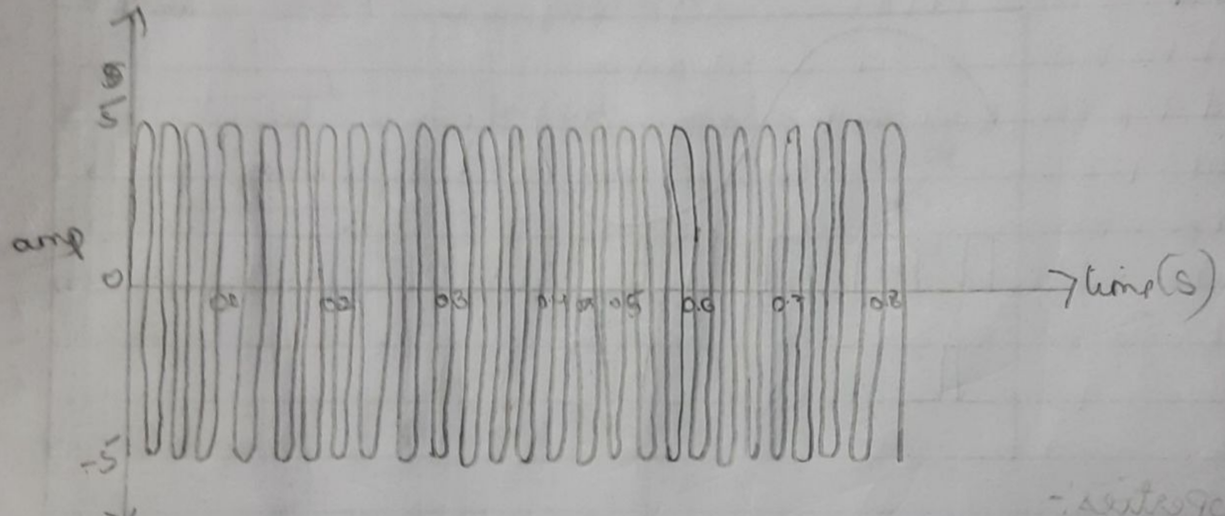
$f_c = 20Hz$ $f_p = 5Hz$
amp = 5

(56)

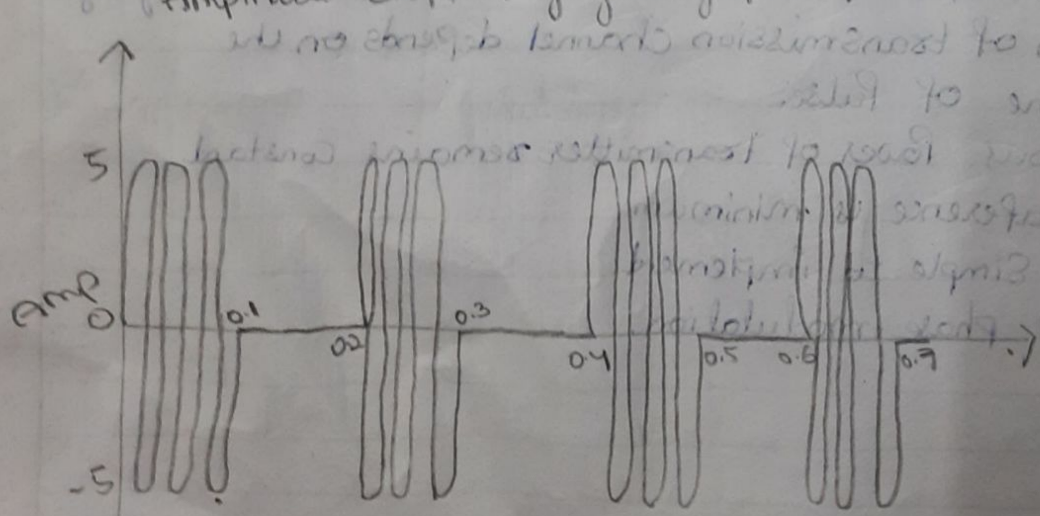
message signal



carrier signal



Amplitude shift keying signal :-

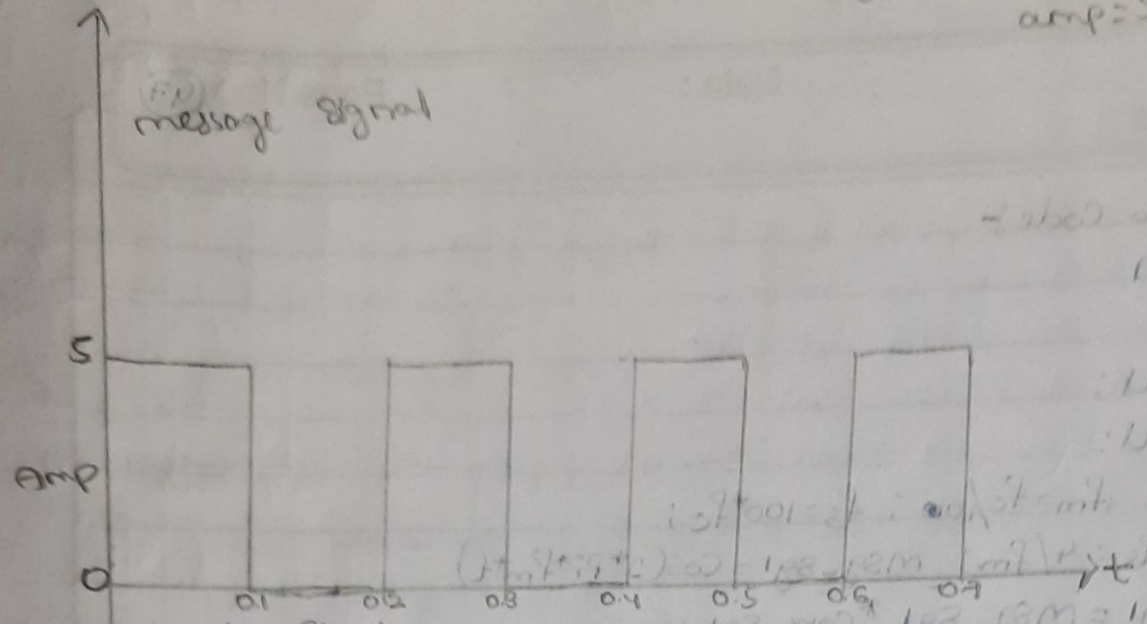


27 ASK

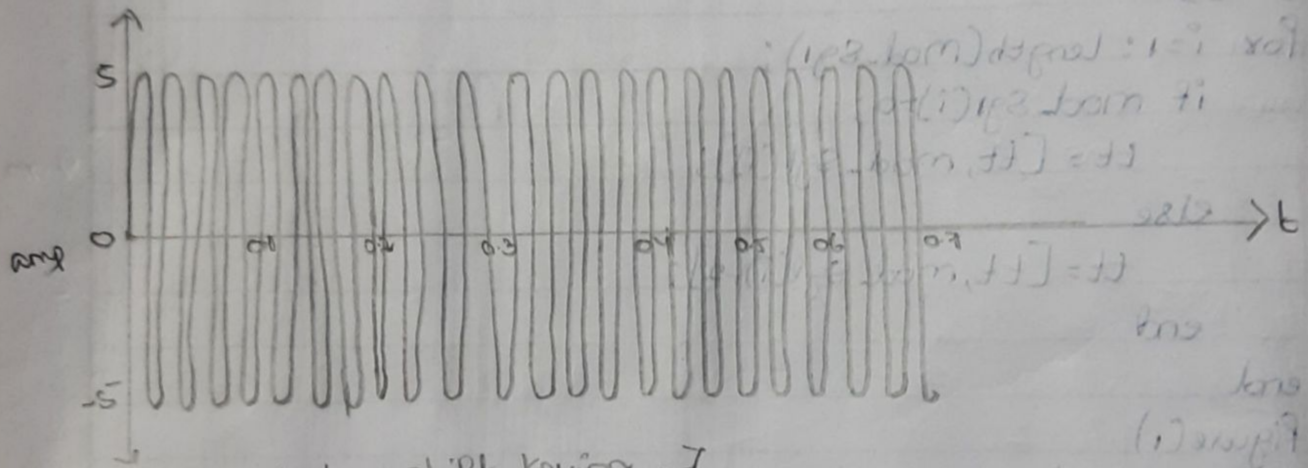
f = 45Hz, fp = 5Hz
amp = 5

(59)

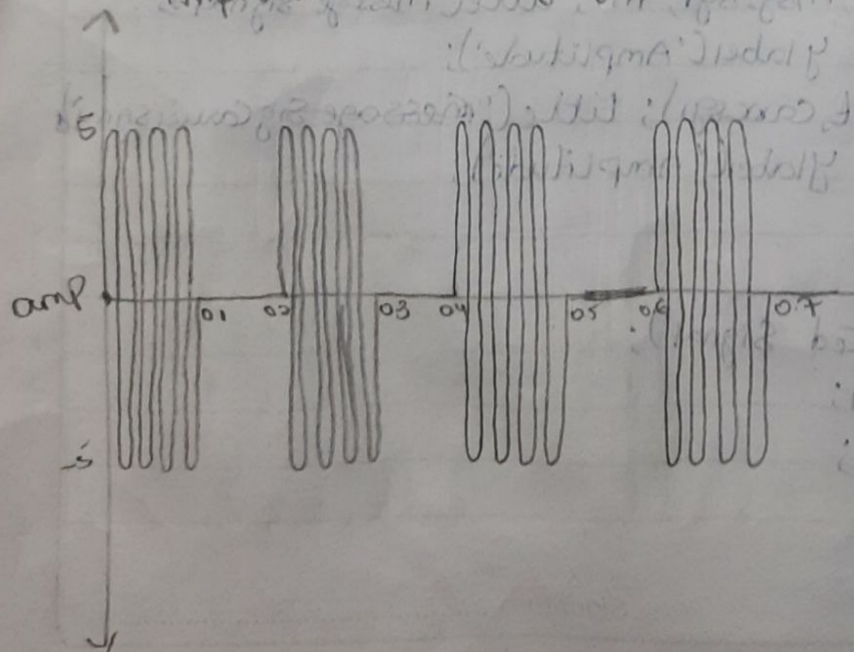
message signal



Carrier Signal

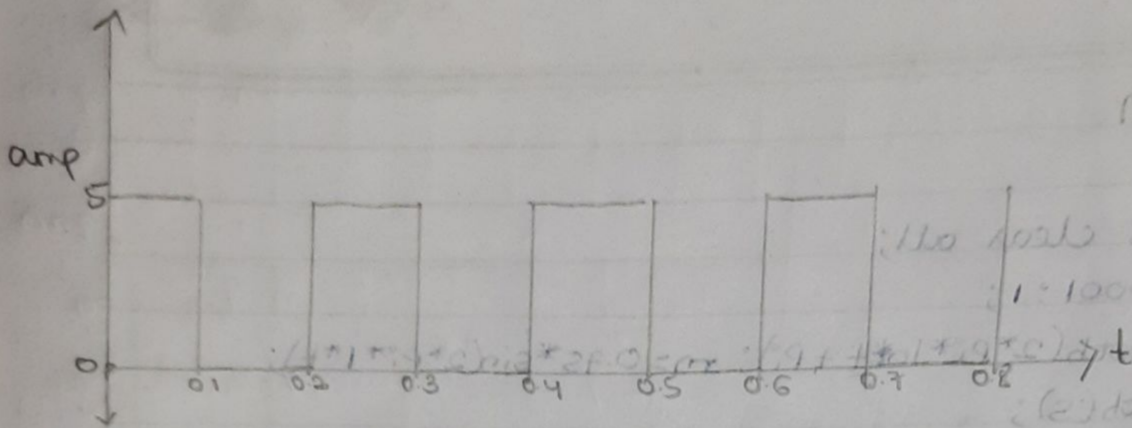


Amplitude Shift Keying →

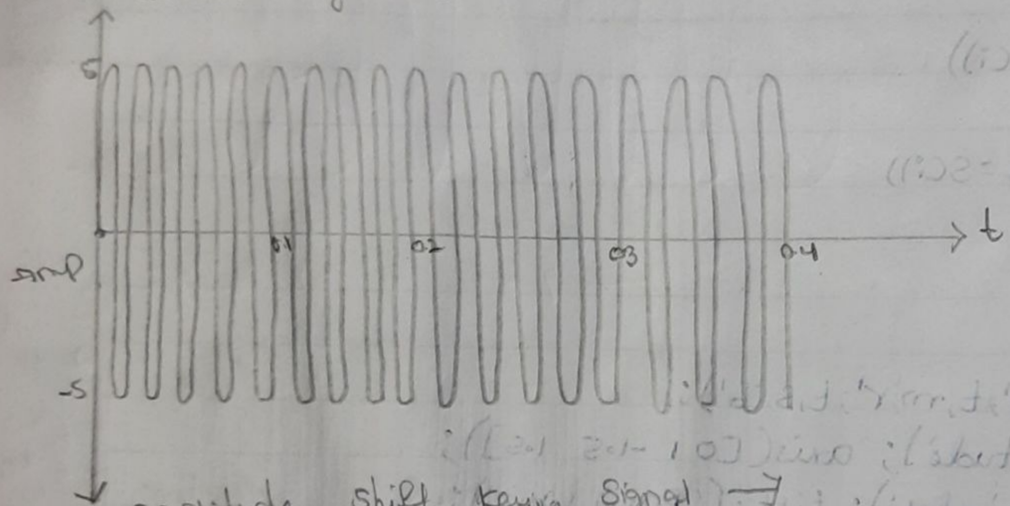


3] ASK $f_c = 50Hz$ $f_p = 10Hz$ amp = 5

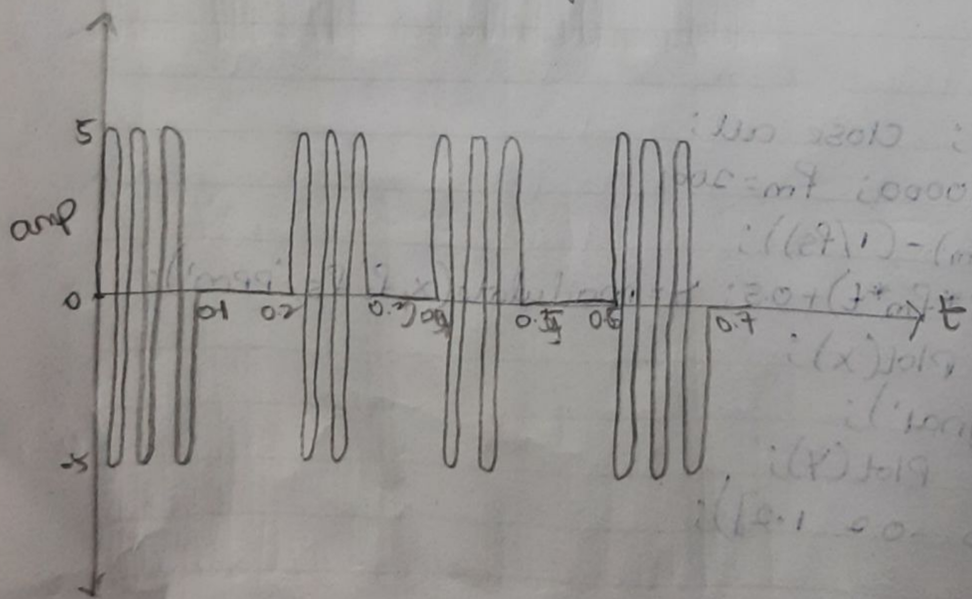
message signal →



carrier signal →



Amplitude shift keying signal →



```
%% FSK
clc; , close all;, clear all;,
fc1= input('Enter the freq of 1st sine wave carrier');
fc2= input('Enter the freq of 2nd sine wave carrier');
fp= input('Enter the freq of Period Binary Pulse(message):');
amp=input('Enter the amplitude(For Both carrier&message)?');
amp= amp/2 ; t=0:0.001:1;
c1 = amp.*sin(2*pi*fc1*t);  c2= amp.*sin(2*pi*fc2*t);
subplot(4,1,1); plot(t,c1);
xlabel('Time'); ylabel('Amplitude'); title('Carrier 1 wave');
subplot(4,1,2); plot(t,c2);
xlabel('Time'); ylabel('Amplitude'); title('Carrier 2 wave');
m=amp.*square(2*pi*fp*t)+amp; subplot(4,1,3); plot(t,m);
xlabel('Time'); ylabel('Amplitude'); title('Binary message Pulses');
for i=0:1000
    if m(i+1)==0
        mm(i+1)=c2(i+1);
        else
        mm(i+1)=c1(i+1);
    end
end
subplot(4,1,4);
plot(t,mm);
xlabel('Time');
ylabel('amplitude');
title('modulated wave');
```

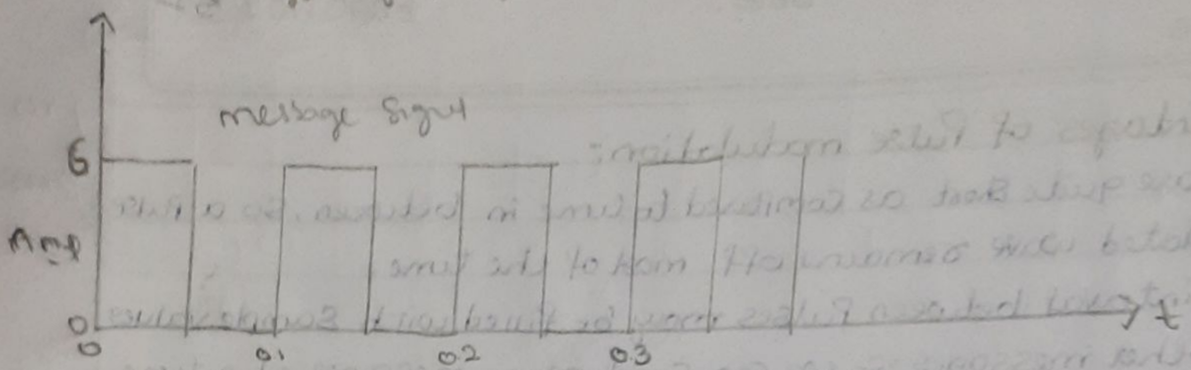
Signature ....

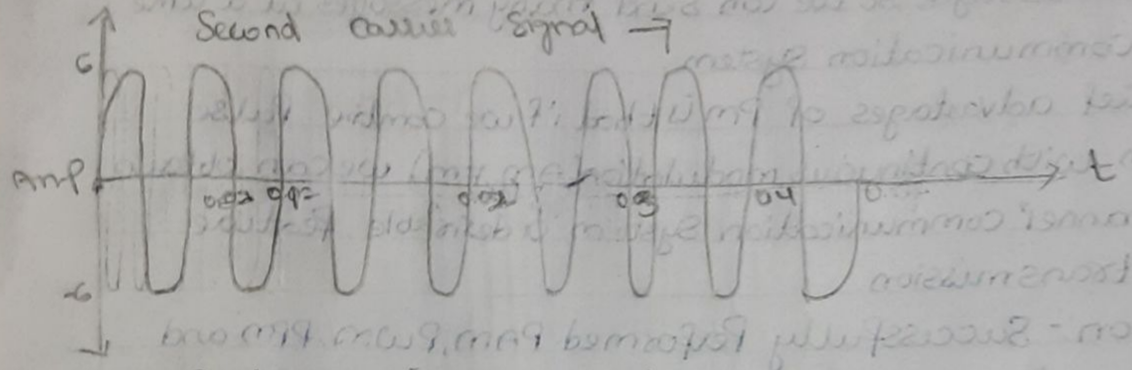
S.K.B.W

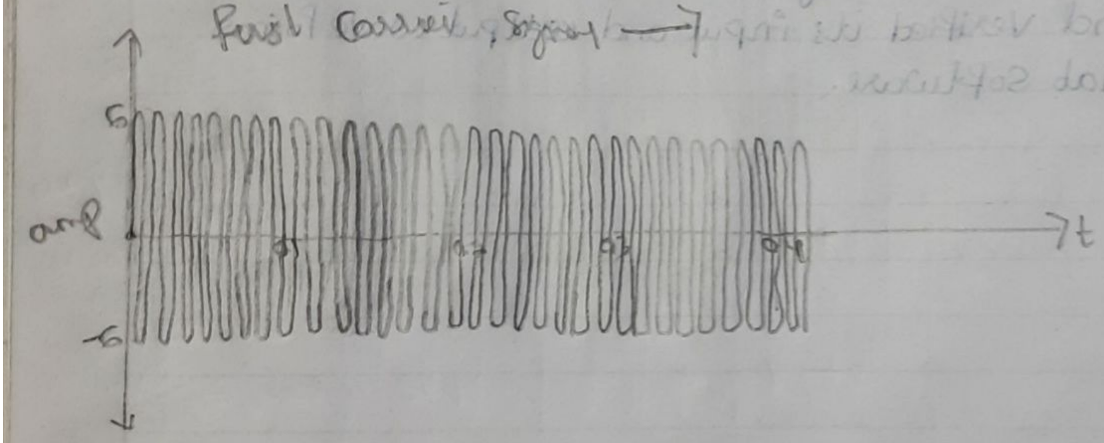
1) FSK $f_{c1} = 50Hz$ $f_{c2} = 20Hz$ $f_p = 10Hz$ amp = 6

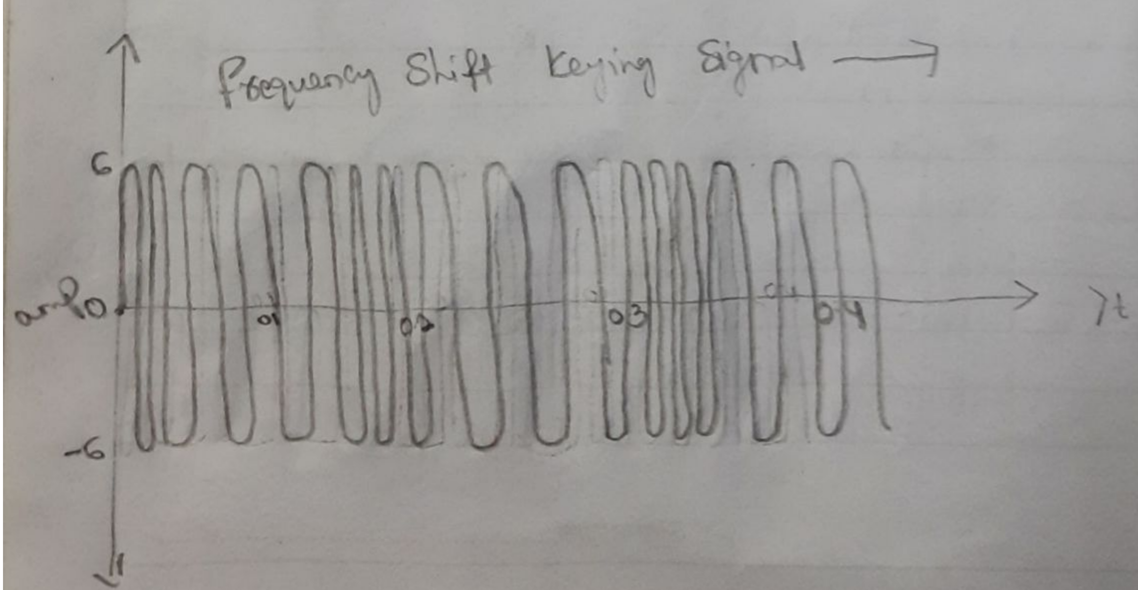
60

message signal →

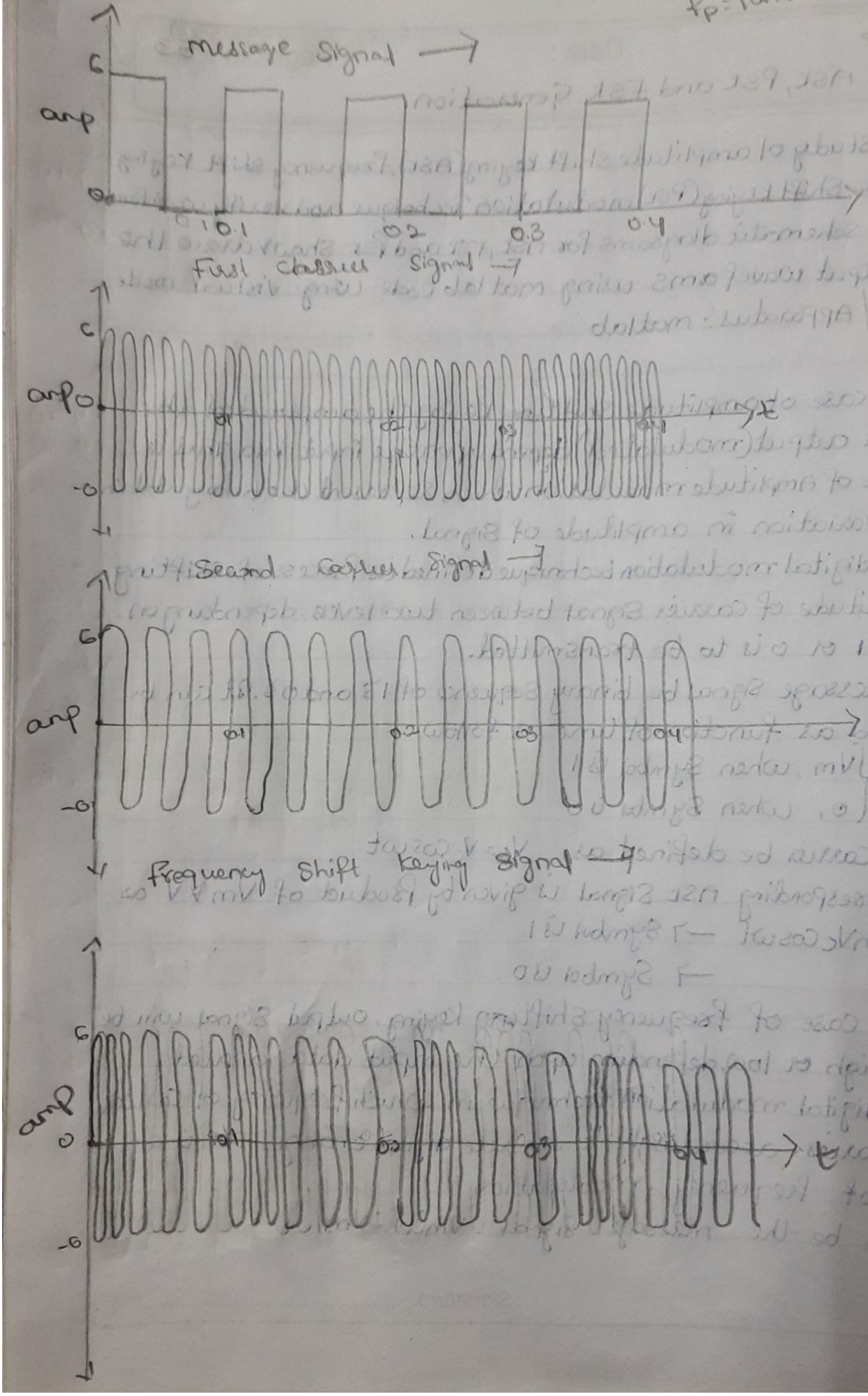






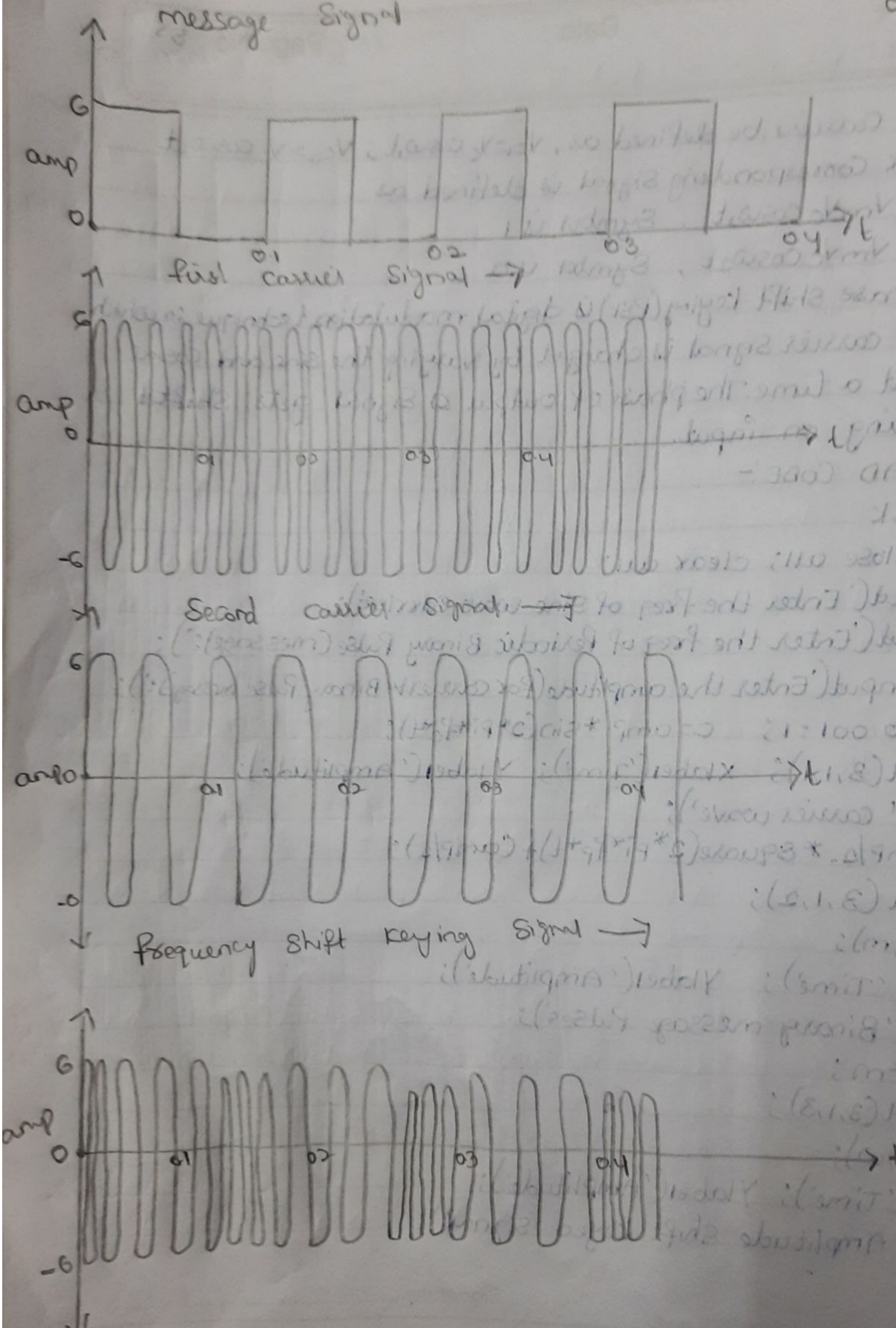


2) FSK $f_{c_1} = 50Hz$ $f_{c_2} = 30Hz$ $f_p = 10Hz$ amp = 6



3) FSK

$f_{c_1} = 30Hz$, $f_{c_2} = 10Hz$
$f_p = 5Hz$,
amp = 6



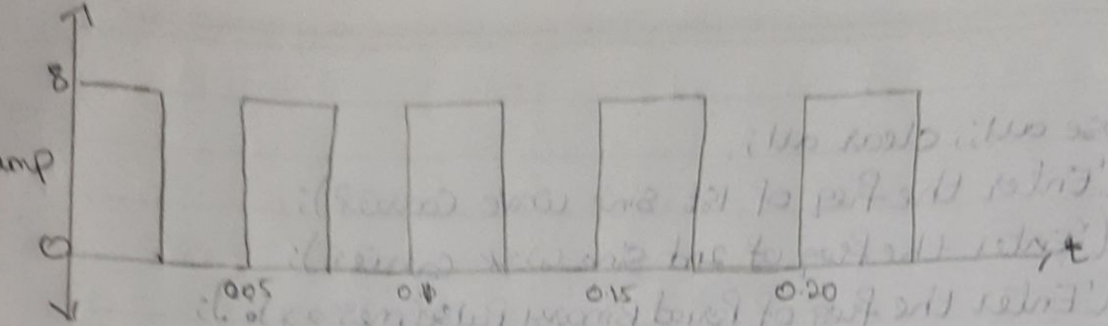
```
%% PSK
clc;
close all;
clear all;
fc=( input('Enter the freq of sine wave carrier');
fp= input('Enter the freq of Periodic binary Pulse(message)');
amp=input('Enter amplitude(For carrier binaryPulse message):');
t=0:0.0001:1
c=amp.*sin(2*pi*fc*t);
subplot(3,1,1);
plot(t,c);
xlabel('Time');
ylabel('Amplitude');
title('Carrier wave');
grid on;
m=square(2*pi*fp*t);
subplot(3,1,2);
plot(t,m);
xlabel('Time');
ylabel('Amplitude');
title('Binary message Pulses');
w=c.*m
subplot(3,1,3);
plot(t,w);
xlabel('Time');
ylabe('Amplitude');
grid on;
```

Signature ....

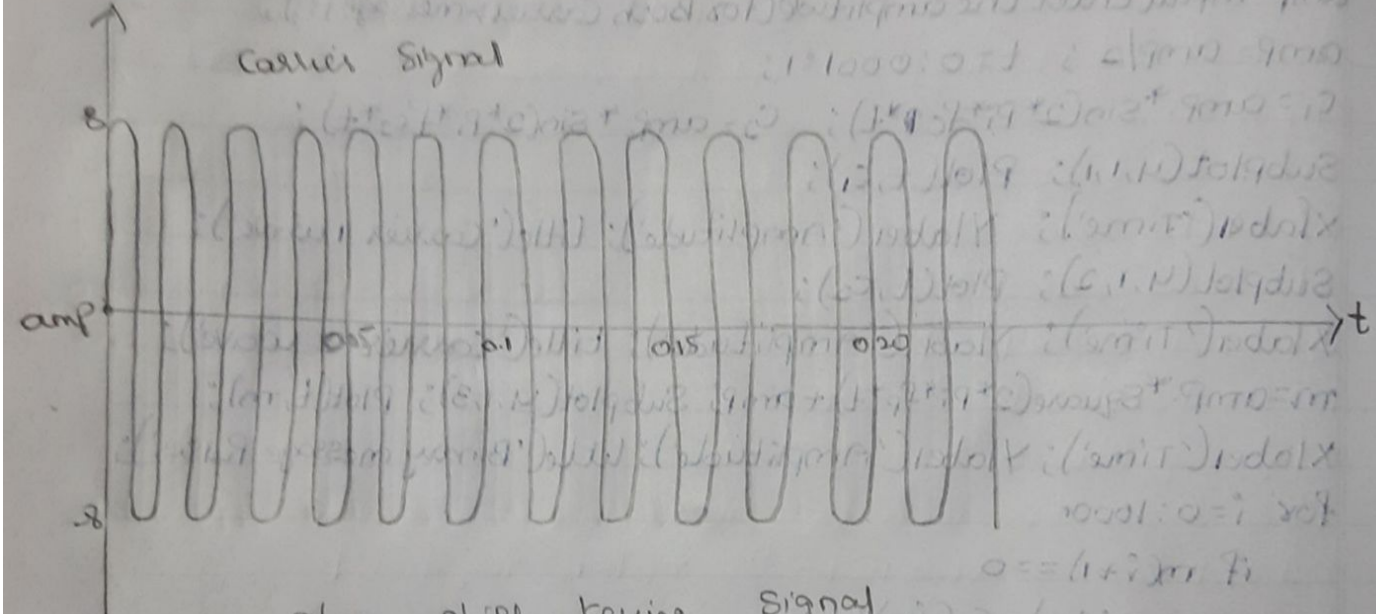
17) PSK

$f_c = 15Hz$ $f_p = 5Hz$
amp=8

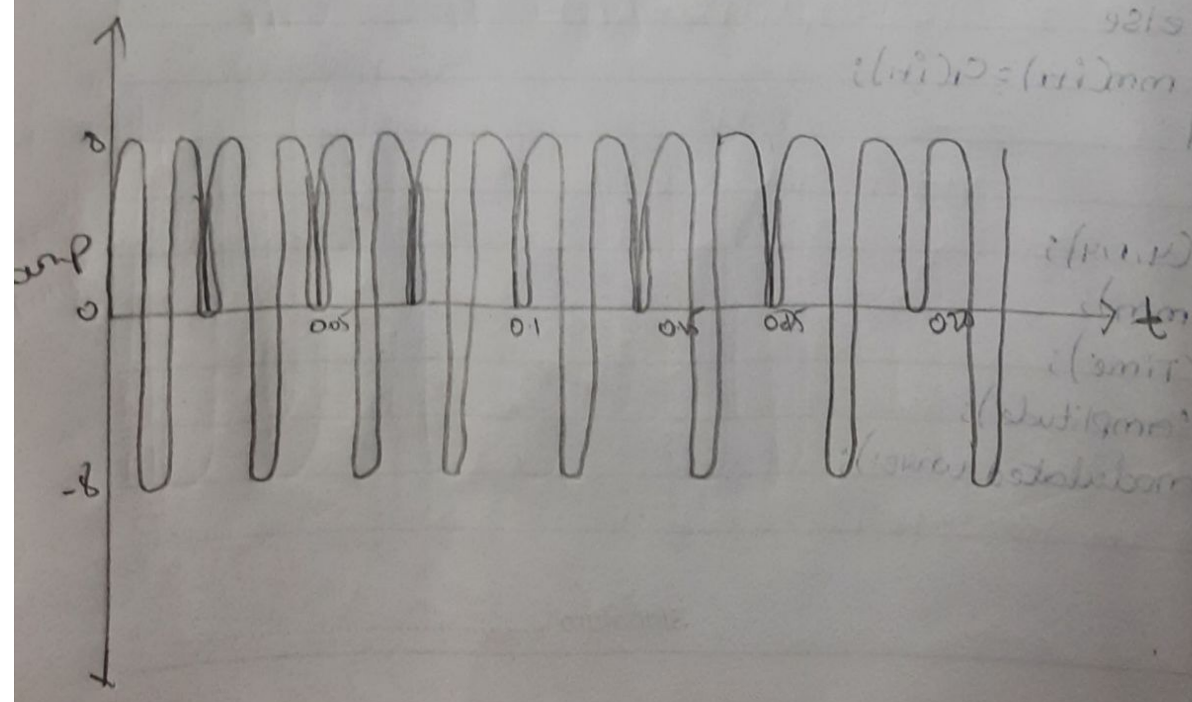
message signal



Carrier Signal



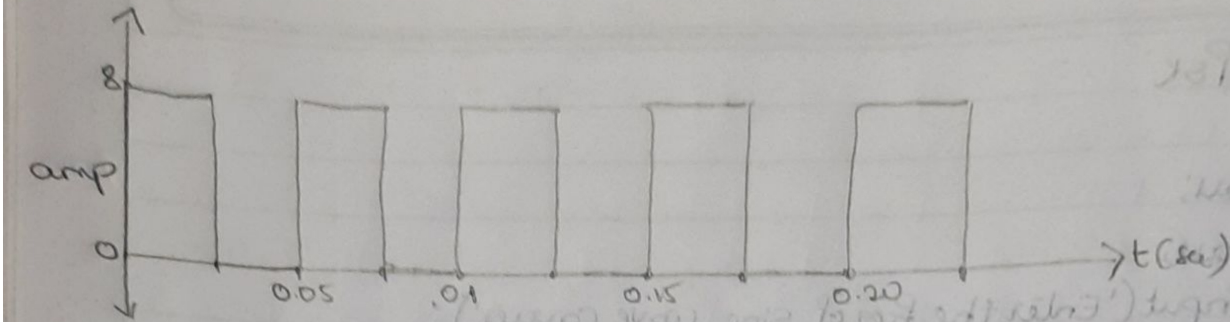
Phase Shift keying Signal



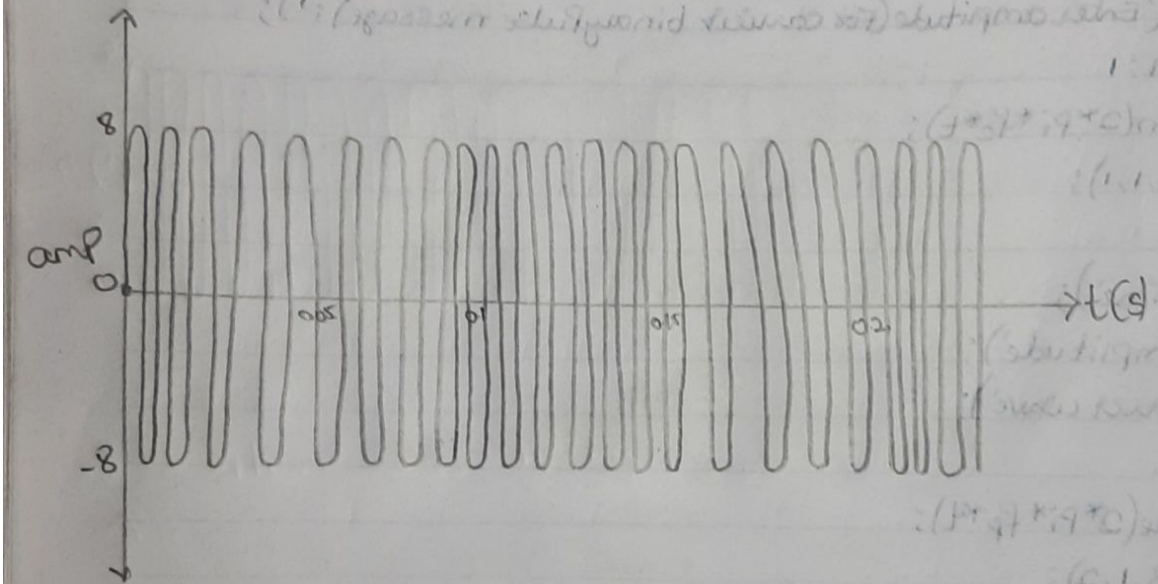
2) PSK

$f_c = 25Hz$, $f_p = 5Hz$
amp=8

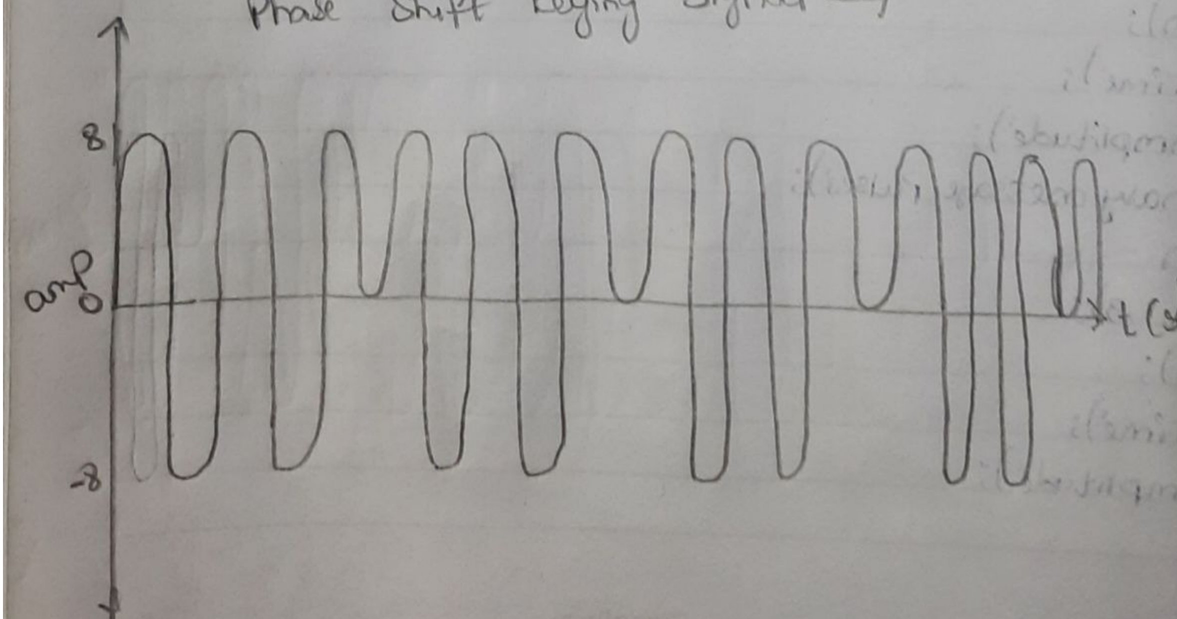
message signal →



carrier signal →

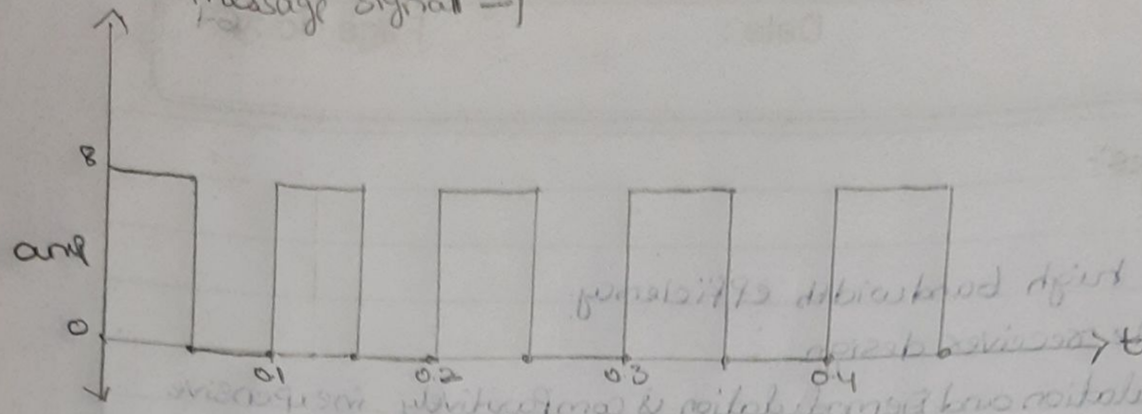


Phase shift keying signal →

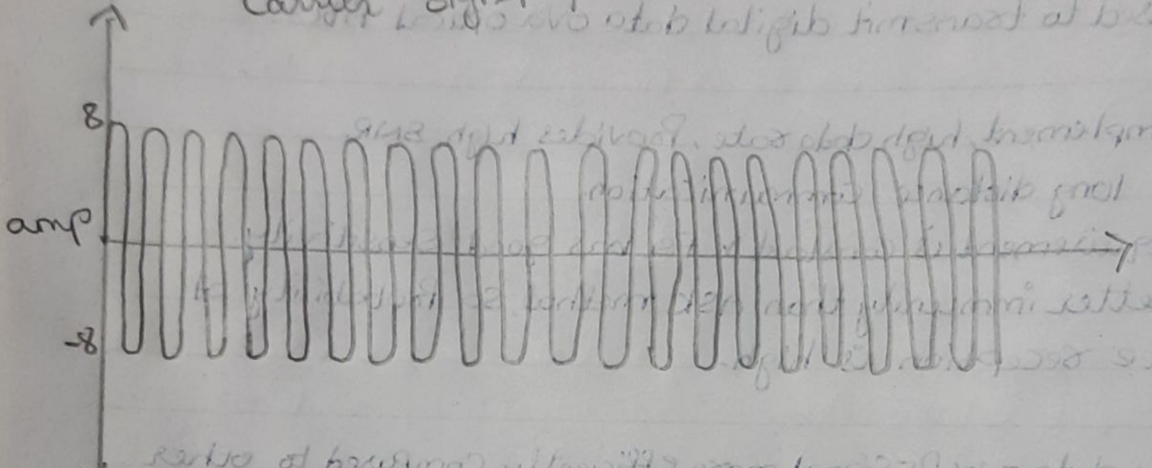


3/ PSK $f_c = 40Hz$ $f_p = 10Hz$ amp=8

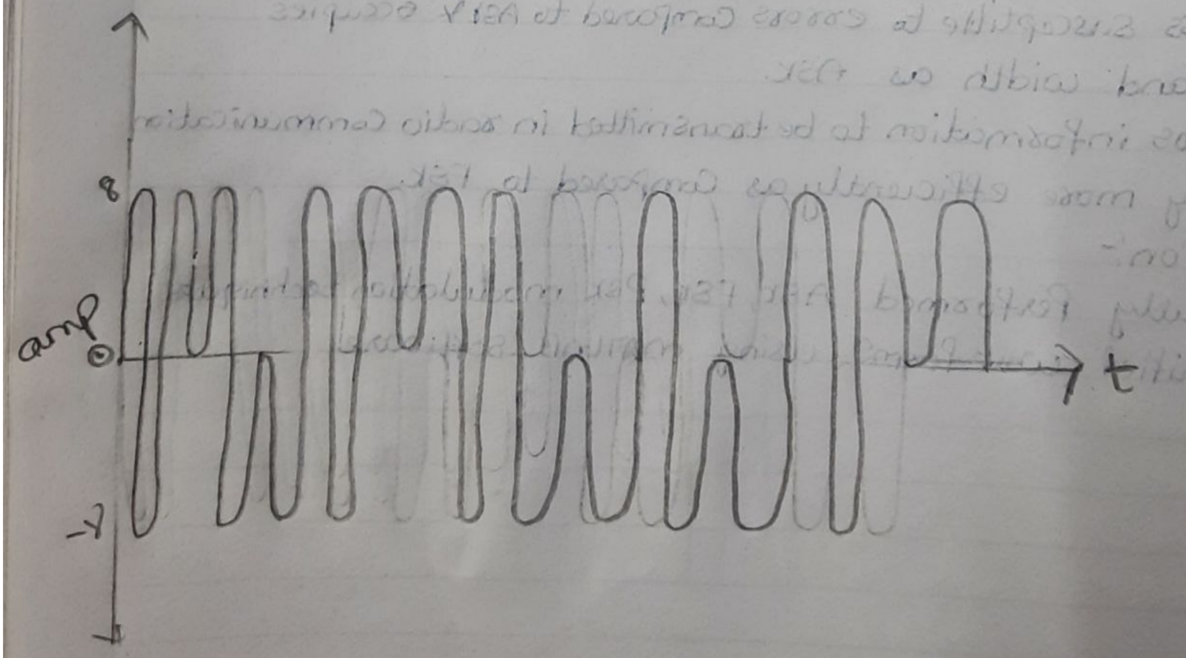
Message Signal →



Carrier Signal →



Phase Shift keying Signal →



Advantages:-

ASK:-

* It offers high bandwidth efficiency.
* Has simple receiver design.
* ASK modulation and Demodulation is comparitively inexpensive
* Can be used to transmit digital data over optical fibers.

FSK:-

* easy to implement, high data rate, Provides high SNR.
* used for long distance communication.
* Power requirement is constant & FSK has good sensitivity.
* It has better immunity than ASK method, so Probability of error-free reception is high.

PSK:-

* It carries data over RF signal more efficiently compared to other modulation types. Hence it is more efficient that ASK & FSK.
* It is less susceptible to errors compared to ASK & occupies same band width as ASK.
* It allows information to be transmitted in radio communication in a way more efficiently as compared to FSK.

Conclusion:-

Successfully Performed ASK, FSK, PSK modulation techniques and Verified waveforms using MATLAB Software.

Signature ....