Experiment No. 1

Study of data sheets of all IC's

Aim: To study the data sheets of all IC's which are required for hardware implementation of all the experiments.

- IC's and its specifications

(i) MC1496 : Used in amplitude modulation and demodulation

Pin Diagram:

Signal input	1	M	14	V_{EE}
Gain adjust	2	C	13	N/C
Gain adjust	3	1	12	Output
Signal input	4	4	11	N/C
Bias	5	9	10	Carrier Input
Output	6	6	9	N/C
N/C	7		8	Input Carrier

- Excellent carrier suppression – 65dB typical @0.5MHz
  50dB typical @10MHz
- High Common Mode Rejection – 85dB
- Maximum Bias current – 10mA
- Operating Ambient temperature range – 0 to 70°C
- Differential Input Signal – $-5 \pm (5 + 15R_e)$
- Input offset current – 0.7µA typically, 7µA maximum
- Output offset current – 14µA typically, 20µA max
- DC power dissipation – 33mW

ii ICL8038: This is a 14 pin IC used in frequency modulati-on.

Sine wave adjust	1	I	14	N/C
Sine wave out	2	C	13	N/C
Triangle out	3	L	12	Sine wave adjust
Duty cycle	4	8	11	V or GND
Frequency adjust	5	0	10	Timing Capacitor
V+	6	3	9	Square wave output
FM bias	7	8	8	FM Sweep input

1) Supply voltage operating range : Min → 10V
Max → 30V

2) Supply current : typical → 12mA , maximum → 20mA

3) Maximum frequency of oscillation → 100KHz
4) Source frequency of FM input → typically 10KHz
5) Leakage current → min 1A
6) Saturation voltage → typically 0.2V, maximum 0.4 V
7) Rise time (RL = 4.7KΩ) → typically 180nS
8) Fall time (RL = 4.7KΩ) → typically 40nS
9) Triangle / Sawtooth / Ramp Amplitude → typically 0.30 X Vsupply

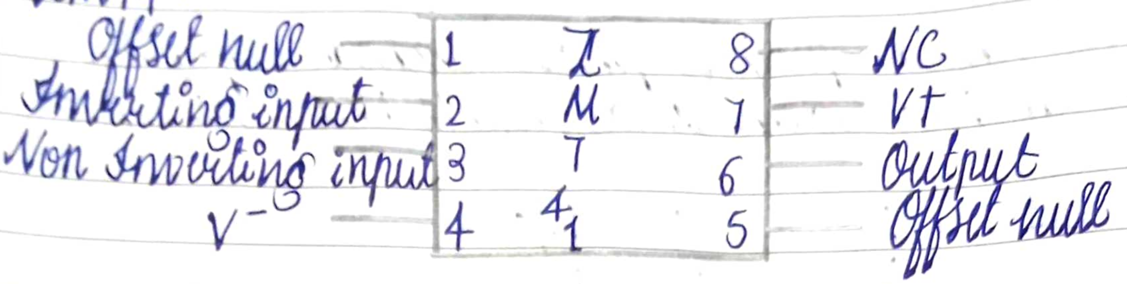
10) Output Impedance → 200Ω

3 LM565: This is general purpose phase locked loop, this IC is used in FSK demodulation, FM modulation etc

-Vcc	1	L	14	N/C
Input	2	M	13	N/C
Input	3	5	12	N/C
VCO Output	4	6	11	N/C
VCO Input	5	5	10	+Vcc
Reference Output	6		9	Timing Capacitor
VCO control voltage	7		8	Timing Capacitor

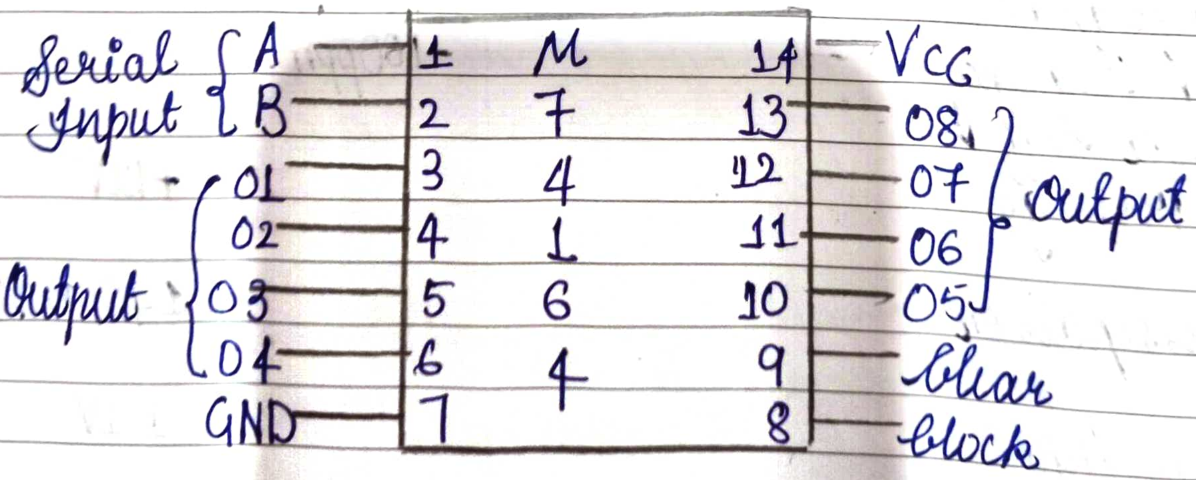
Parameter	Condition	LM565 Min	Max
Power supply current	–	–	12.5mA
Input impedance	–	7KΩ	
VCO maximum operating frequency	Co = 2.7pF	300KΩ	
Operating frequency temprature coeff	–	-100ppm	
Supply voltage			+12V
Power Dissipation			1400mW
Differential Input voltage			±1V

iv LM741



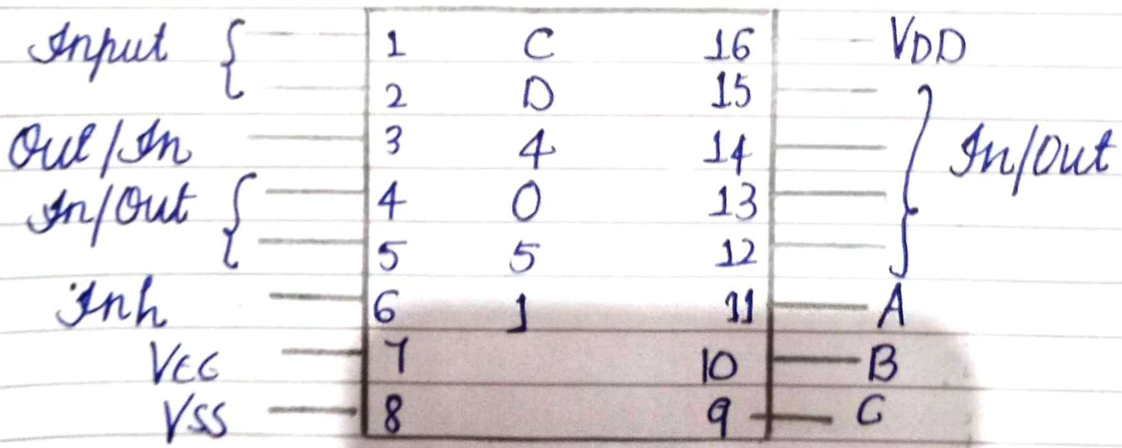
- Supply voltage → ±22V
- Power Dissipation → 500mW
- Differential input voltage → ±30V
- Input voltage → ±15V
- Operating temprature → -50 to 125 °C
- junction temprature → 150°C
- Input offset current → 200nA
- Output voltage swing → ±14V
- Common mode rejection ratio → 95dB
- Supply voltage rejection ratio → 96dB
- Slew Rate → 0.5V/μs
- Bandwidth at 25°C → 1.5MHz

v M74164 : Used in PN sequence generator



Symbol	Parameter	Value	Unit
Vcc	Supply Voltage		
VI	DC Input Voltage	-0.5 to +7	V
Vo	DC Output Voltage	-0.5 to Vcc	V
ISIG	DC input diode current	-0.5 to Vcc	V.
IOIG	DC output diode current	±20	mA
Io	DC output source sink	±20	mA
	current per output pin	±25	mA
Io.	DC Vcc or GND current	±50	mA
Pd	Power dissipation	500	mW
Tsig	Storage Temprature	-65 to +150	°C
TL	Lead Temprature	300	°C

vi CD4051IC: Used in PSK



Parameter	Value	Unit
(i) DC supply voltage (VDD)	+0.5 VDD to 18 VDC	V
(ii) Input Voltage (Vin)	-0.5 VDC to VDD	V
(iii) Storage Temprature Range (TS)	-65 to +150	°C
iv Power dissipation (PD) Dual in line	700	mW
(v) Small Outline	500	mW
(vi) Lead temprature (TL)	260	°C
(vii) Input current (Iin)	-0.1	µA
ix Quiesent device current (IDD) if VDD = 10V at +85°C	300	µA
x Low level input voltage	4	V

Experiment No 2

Amplitude Modulation

- Aim: Design and implementation of modulation and demodulation (AM) using the hardware components and multisim.

- Objectives: (i) Comparing the modulated signal and demodulated signal
  (ii) Observing input/output waveforms for different modulation index values
  (iii) Using different design we have to implement amplitude modulation and demodulation

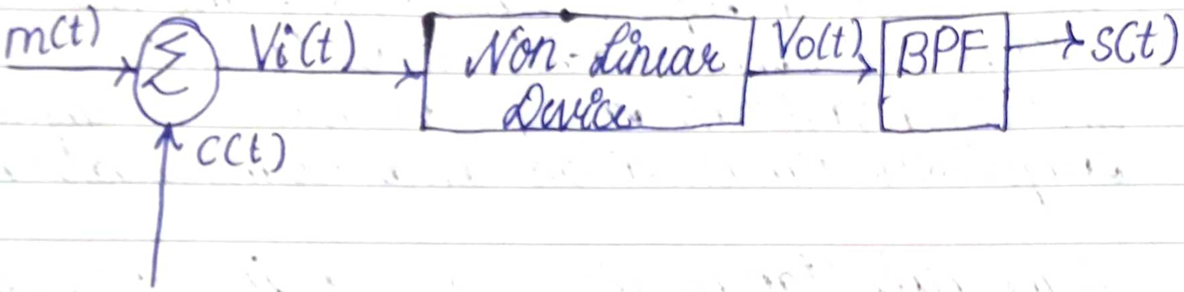
- List of Components/Equipments required to carry out the experiment

	Components/Equipments	Specifications	Quantity
1	Regulated DC	0-30V, 2A	1
2	CRO	80 Vp-p/20MHz	1
3	Function generator	0-1MHz	2
4	BJT BC107	BC107	1
5	Resistors	10KΩ, 6KΩ, 22KΩ, 1KΩ	1 Each
6	Inductor	130mH	1
7	Capacitor	0.01μF	1
8	BNC probes	-	2
9	Wires	-	Assorted

- Theoritical Background: Amplitude modulation is a type of modulation technique, where the amplitude of the carrier signal is varied in accordance with the instatenous value of message signal keeping the frequency and phase constant.

★ Methods of Modulation:

(1) Square law Modulator



Let the modulated signal s(t) be given by

$s(t) = Ac[1 + Ka\,m(t)] \cos 2\pi f_c t$,

Ka = Amplitude Sensitivity

$s(t) = Ac \cos 2\pi f_c t + Ka\,m(t) \cos 2\pi f_c t$

Let $m(t) = Am \cos 2\pi f_m t$

$s(t) = Ac \cos 2\pi f_c t + Ka\,Am\,Ac (\cos 2\pi f_c t)(\cos 2\pi f_m t)$

$s(t) = Ac \cos 2\pi f_c t + \mu Ac (\cos 2\pi f_c t)(\cos 2\pi f_m t)$

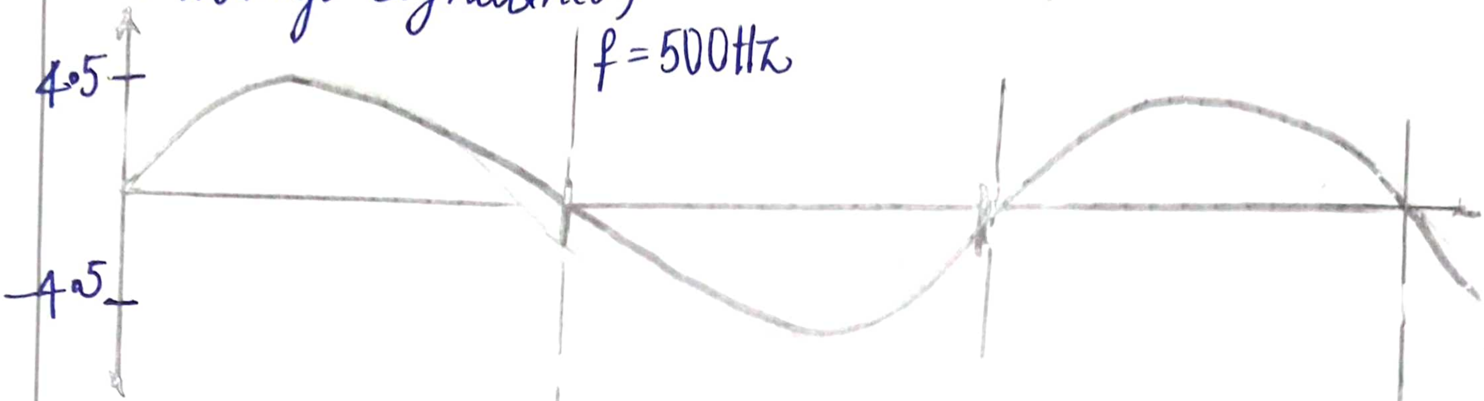
$\mu \rightarrow$ Modulation Index

$\mu < 1$ Overdamped

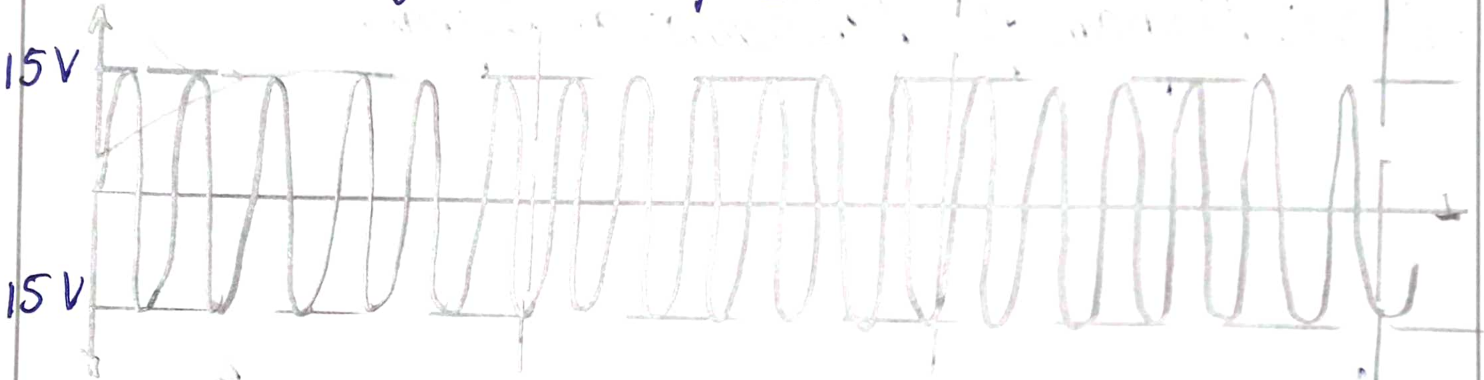
$\mu = 1$ Critical Modulation

$\mu < 1$ Under modulation

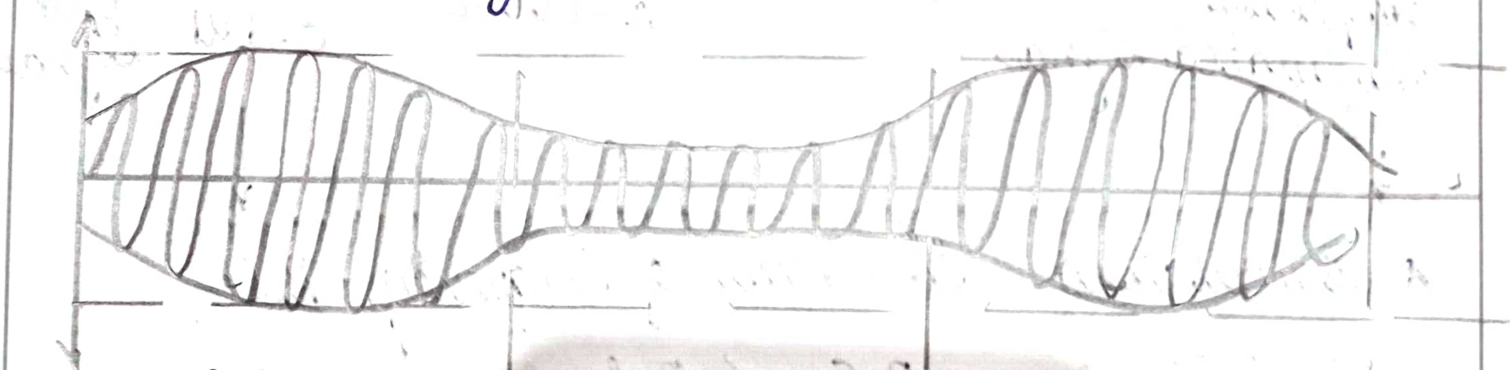
* Message Signal (m(t))



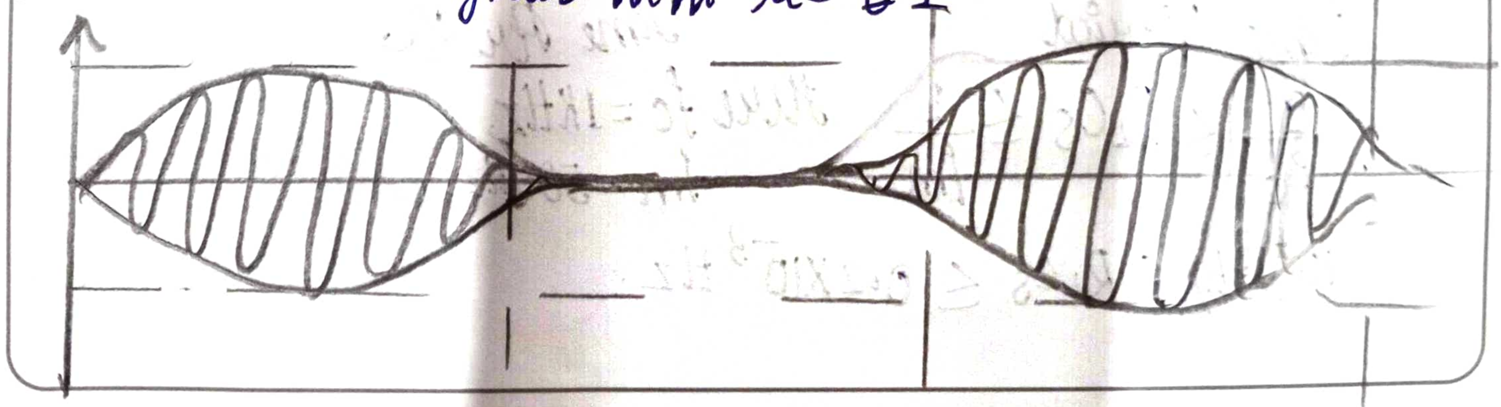
* Carrier Signal (c(t)) f = 1KHz



* Modulated Signal with $\mu = 0.3$

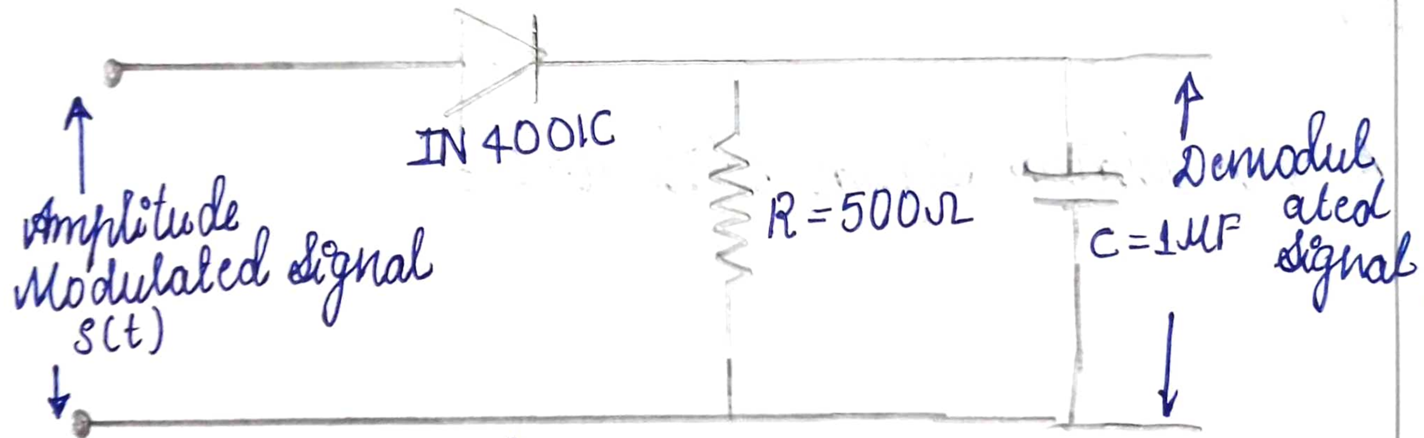


* Modulated Signal with $\mu = 1$



★ Modulated Signal with $\mu = 2.25$

Circuit Diagram for Demodulated Signal.



★ Calculating the value of Frequency - R_LC_S

Input Signal Time Period $\leq R_LC_S \leq$ Output Signal Time Period

$$\frac{1}{f_c} \leq R_LC_S \leq \frac{1}{f_m}$$

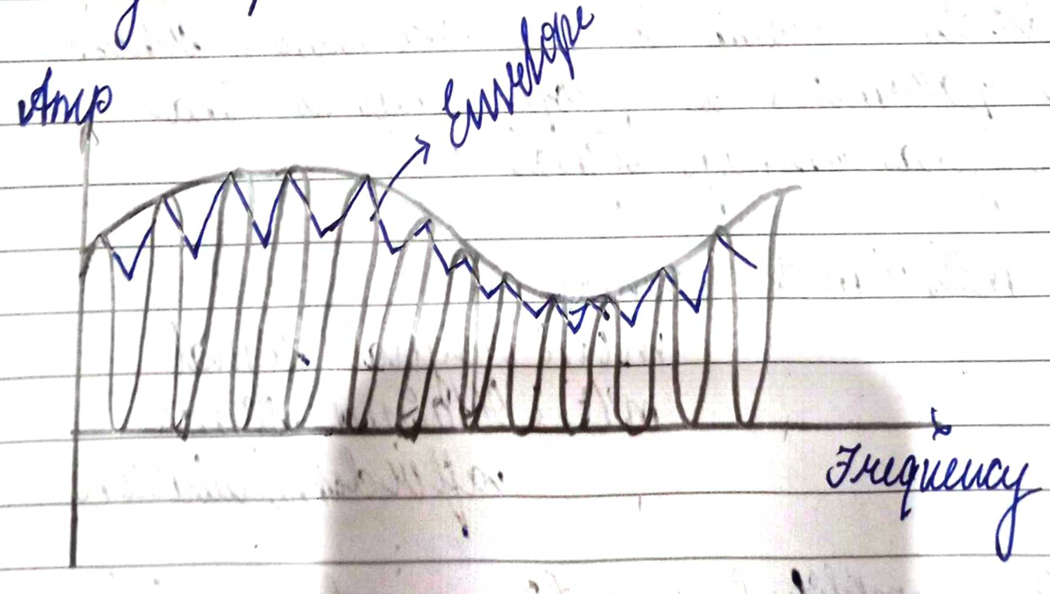
Here $f_c = 1kHz$
$f_m = 500Hz$

$$10^{-3} Hz \leq R_LC_S \leq 0.2 \times 10^{-3} Hz.$$

Demodulation is process of seprating the orignal information or 'signal' from the modulated carrier signal. It generally involves a device called 'demodulator' or 'detector' which produces a signal corresponding to the instantanious changes in amplitude. One of the common method of demodulation is 'Envelope Detector'.

- Envelope Detector: This charges a capacitor to a voltage to the peak voltage of incoming AM waveform
- When the input wave's amplitude increases, the capacitor voltage is increased via the rectifying diode
- When the input's amplitude falls, the capacitor voltage is reduced by being discharged by the resistor.

- Nature of Envelope Detected Wave, illustrated in a single peak.



★ Results and Discussion:

From the above ~~equati~~ experiment, we can clearly observed that the nature of the modulated wave depends upon the value of the 'Modulation Index μ'

(1) It determines the strength and the quality of the modulated signal

2) Greater the value of the modulation factor, the stronger and clearer the signal

3) However, if the modulation factor > 1, the carrier is said to be '<u>over modulated</u>' is distorted during the reception.

★ Conclusion:

The modulated wave [s(t)] when passed through the envelope detector results in the the wave, whose amplitude parameter is proportional to that of the original message signal. i.e

$m(t) \propto m'(t)$, where

$m(t)$: Orignal signal

$m'(t)$: Obtained signal after demodulation

Hence the amplitude modution and demodulation is verified.

Experiment No 3:
Frequency Modulation and Demodulation

Aim: Design and implement Frequency modulation and demodulation

Objectives: Implementation of FM modulation and demodulation circuit and observe the input/output waveforms for different modulation index values.

List of components/Equipments required to perform the experiments

	Components/Equipments	Specification	Quantity
1	Regulated DC	0-30V, 2A	1
2	CRO	80Vp-p/20MHz	1
3	Function Generator	0-1MHz	1
4	IC 8038	1	1
5	Resistors	39kΩ	2
6	Capacitors	0.1 μF	1
		100pF	1
7	BNC probes	-	2
8	Wires	-	Assorted

Theoretical Background:

Frequency Modulation is defined as change in the frequency of the carrier signal with respect to the instantaneous value of the the message signal keeping the amplitude and phase of the carrier constant.

In general, the modulated signal in FM is represented as:

$$s(t) = A_c \cos[2\pi f_c t + \beta \sin 2\pi f_m t]$$

β = Modulation Index
f_c = Frequency of carrier signal
f_m = Frequency of message signal.

If $\beta << 1$ → Narrow Band Frequency Modulation

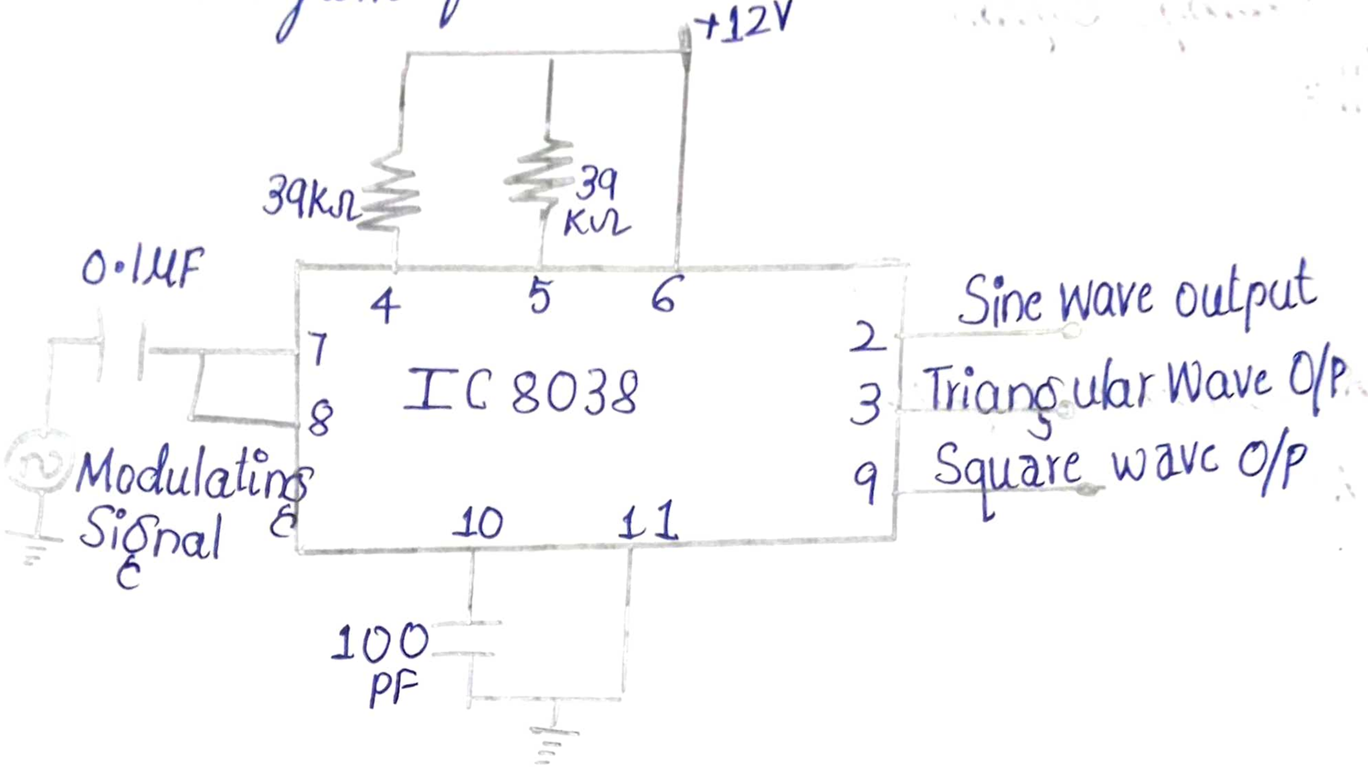
If $\beta >> 1$ → Wide Band Frequency Modulation.

★ Modulation Techniques
1. Direct Method
2. Indirect Method (Armstrong Method)

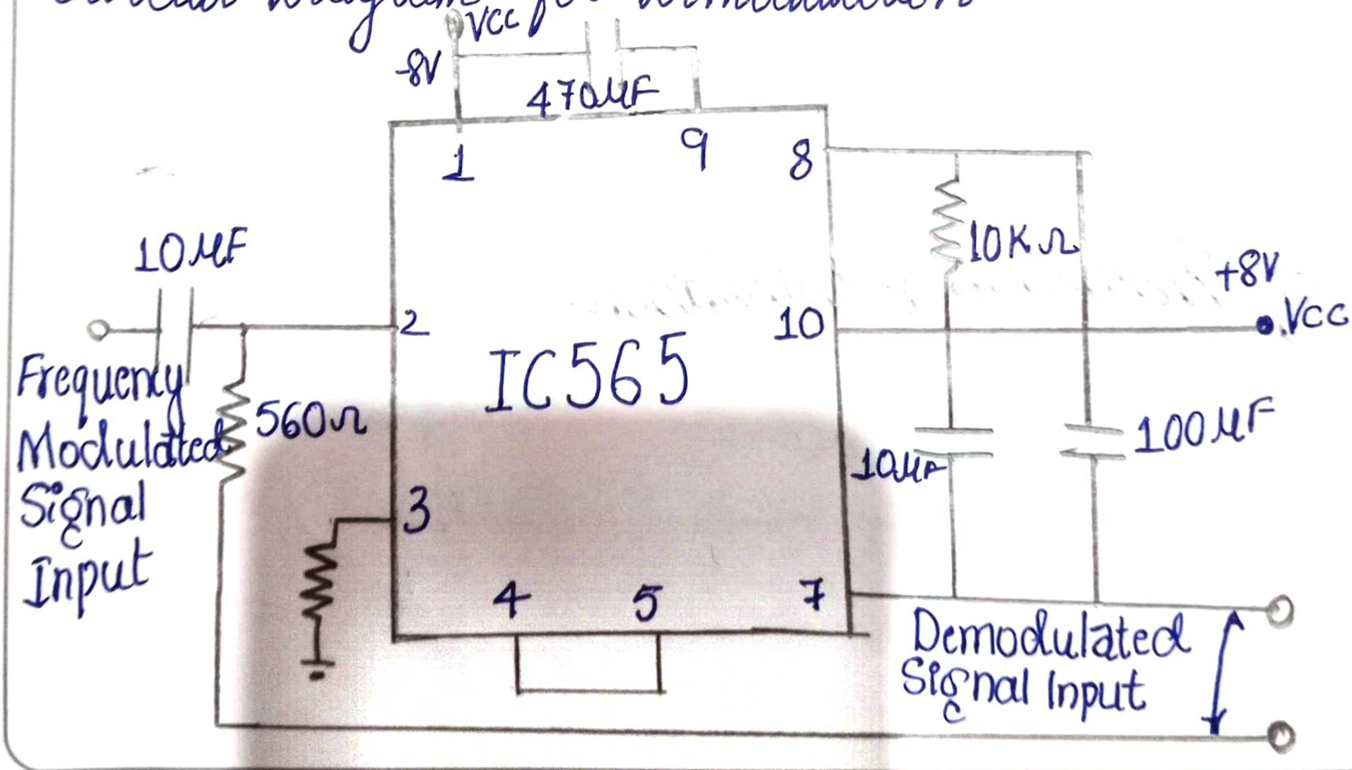
★ Demodulation Techniques

1. Balanced Slope Detector
2. Phase Locked Loop.

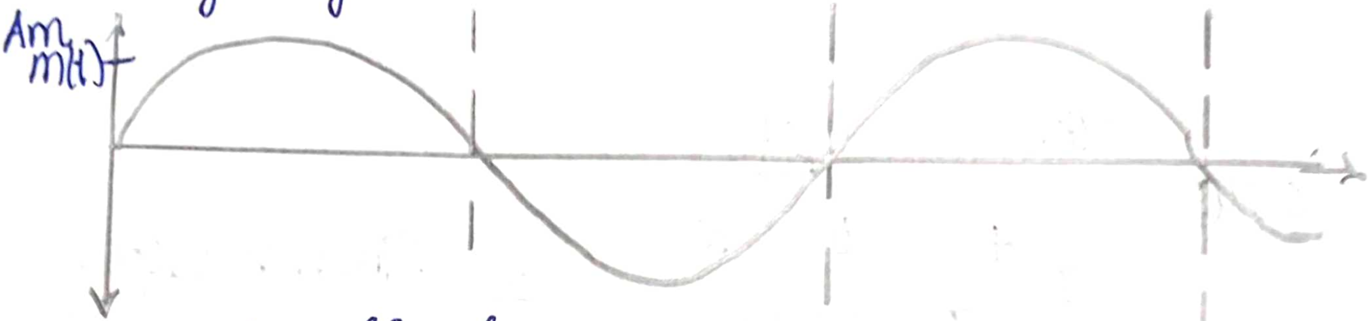
★ Circuit Diagram for Modulation



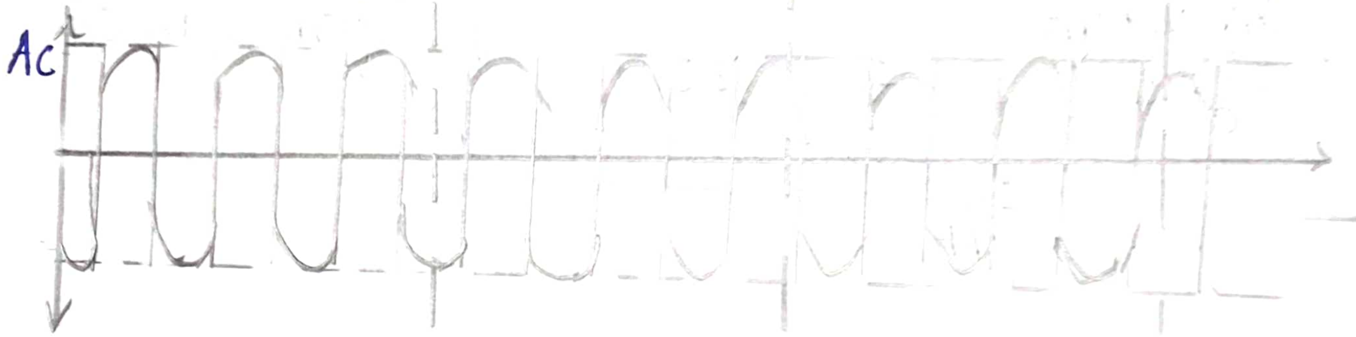
★ Circuit Diagram for Demodulation



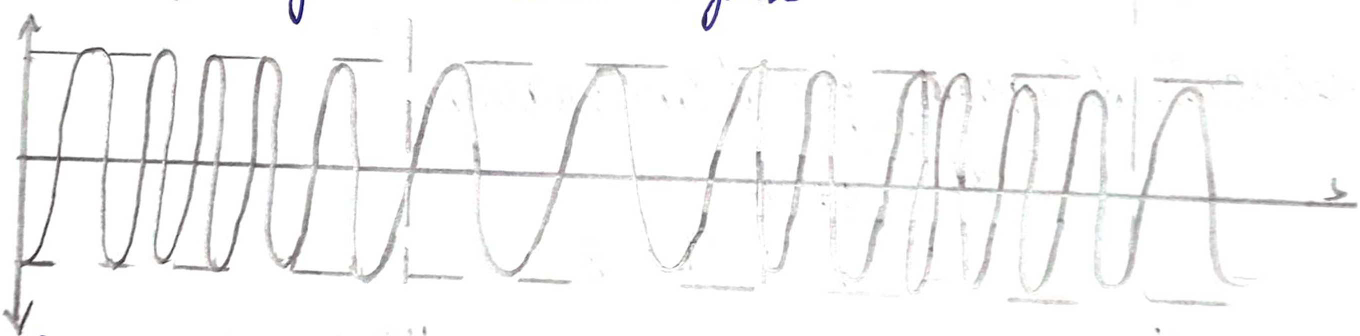
* Nature of Graphs
Message Signal



* Carrier Signal



* Frequency Modulated Signal



* Table of Observation & Calculation.

Experiment 4

Pulse Code Modulation and Demodulation

Aim: To implement Pulse code modulation and demodulation and DPCM using hardware components

Objectives: To implement PCM and DPCM circuit and observe the input/output waveforms.

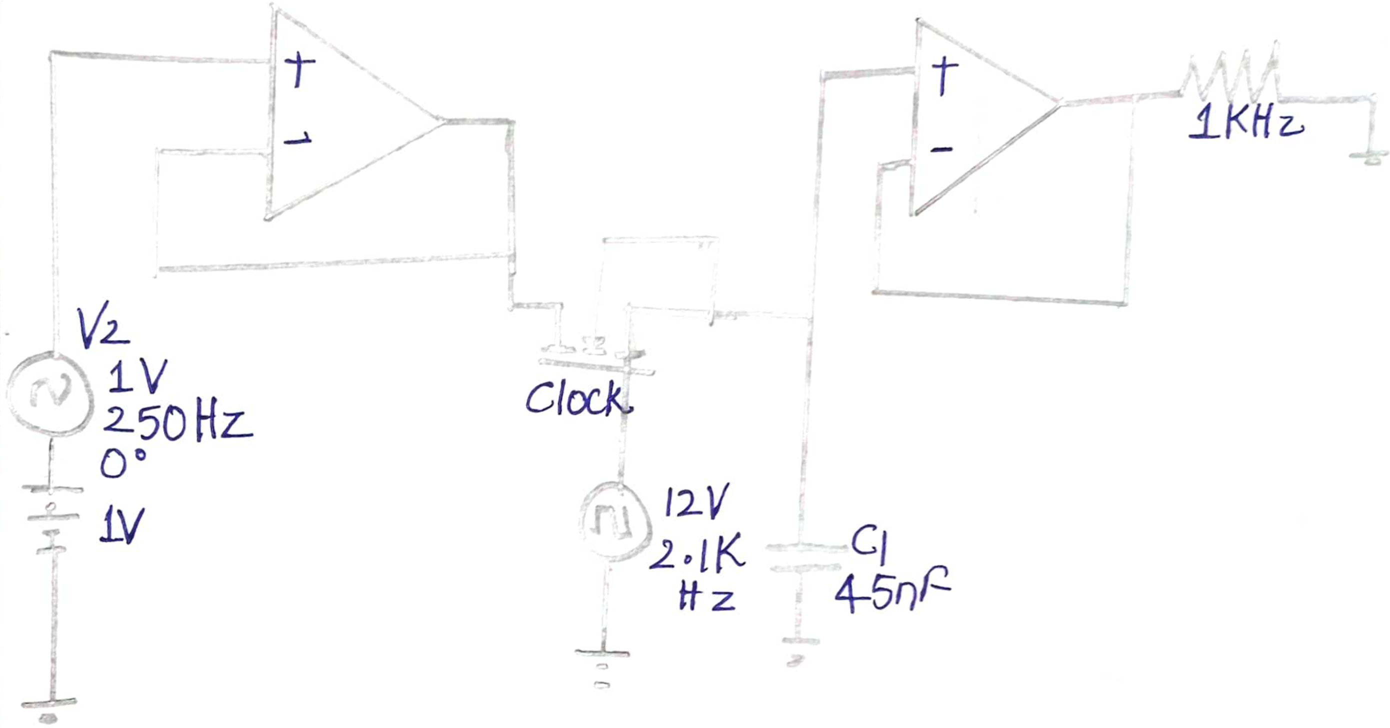
List of Components/Equipments required to perform the Experiments

	Components/Equipments	Specifications	Quantity
1	Regulated DC	0-30V, 2A	1
2	CRO	80Vp-p, 20MHz	1
3	Function generator	0-1MHz	1
4	Op-Amp μA741		2
5	Resistors	1KΩ	1
6	Capacitors	45nF	
7	BNC probes	–	2
8	Wires	–	Assorted.

Theoretical Background : The pulse-code modulation systems are complex in that the message signal is subjected to large number of operations.
The essential operations in the transmitter of the PCM system are sampling, quantising and encoding.
Regeneration of the impaired signals occur at the intermediate points along the transmission path.
At the receiver, the essential operations consist of one last stage of regeneration are usually performed in the same circuit.

In the use of PCM for digitisation of a voice or video signal, the signal is sampled at the rate which is slightly higher than the Nyquist rate. The resulting sampled signal is then found to exhibit high correla-tion between adjecent samples. The meaning of this high correlation is that, in an average sense, the signal doesnot change rapidly from one sample to the next with the result that the difference between the adjecent samples has the varience that is smaller than the varience of the signal itself.
When these highly correlated samples are encoded, the resulting signal contains redundant information. Therefore we use DPCM, in which input to a quantiser, is a signal which is difference between the unquantised input sample and the prediction of it. This predicted value is produced by a predictor the consists quantised version of input signal.

Circuit Diagram (PCM)



Experiment 5 Digital Modulators

Aim: To design and implement digital modulators and demodulators.

Objectives: Demonstrate the principle of ASK, FSK and PSK and generate the Eye Pattern.

Components/Equipments Required to Perform the Experiment

	Components/Equipments	Specifications	Quantity
1	CRO	80Vpp, 20MHz	1
2	Function generator	0-1MHz	1
3	Regulated DC	0-30V, 12A	01
4	Resistance	39KΩ 1.8KΩ 3.3KΩ 1.0KΩ 10KΩ	1 each
5	Variable Decade Resistance	10KΩ, 1KΩ	
6	Capacitors	10pF	
7	BNCs	–	3
8	Wires		Assorted

* Theoretical Background: In the ideal form of coherent detection, exact replicas of possible arriving signals are available at the receiver. This means that the receiver has exact knowledge of carrier waves phase

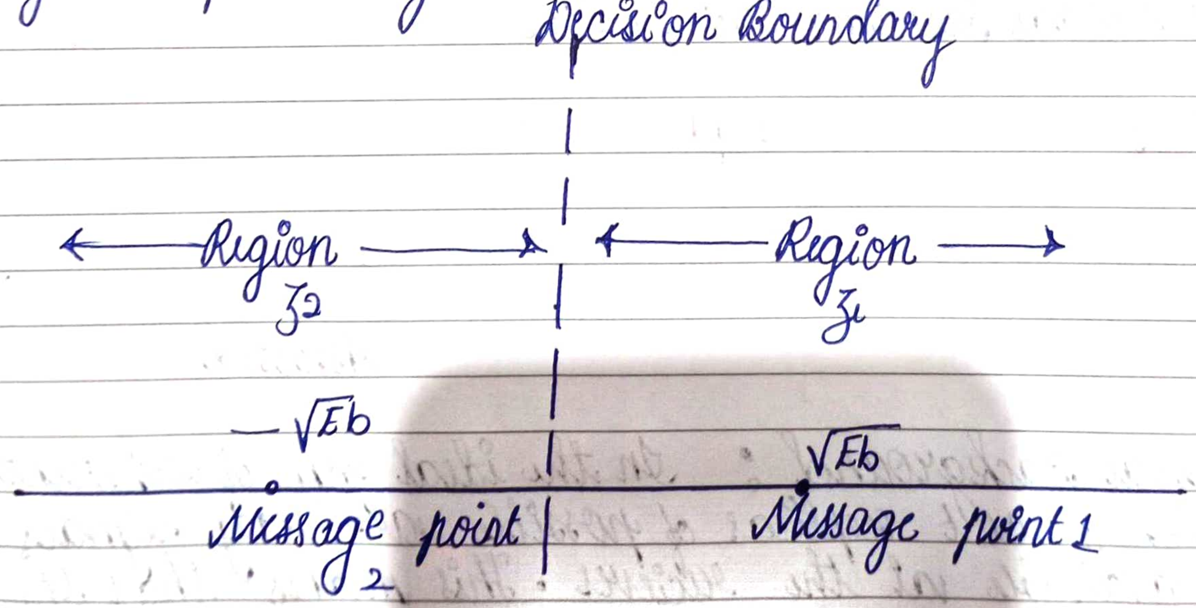
reference, in which case we say the receiver is phase locked to the transmitter. Coherent detection is performed by cross-correlating the received signal with each one of the replicas, and the making a decision based on the comparisions with preselected thresholds. Few of the coherent Binary Modulation Techniques include.

1) Coherent Binary PSK : In a coherent binary PSK system the pair of signals, $s_1(t)$ and $s_2(t)$ used to represent binary symbols 1 and 0 respectively and, are defined by

$$s_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t)$$

$$s_2(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t + \pi) = -\sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t)$$

Signal Space Diagram



PSK transmitter :

Binary wave (polar form) → Product Modulator → Binary PSK wave

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi f_c t)$$

• PSK Reciever

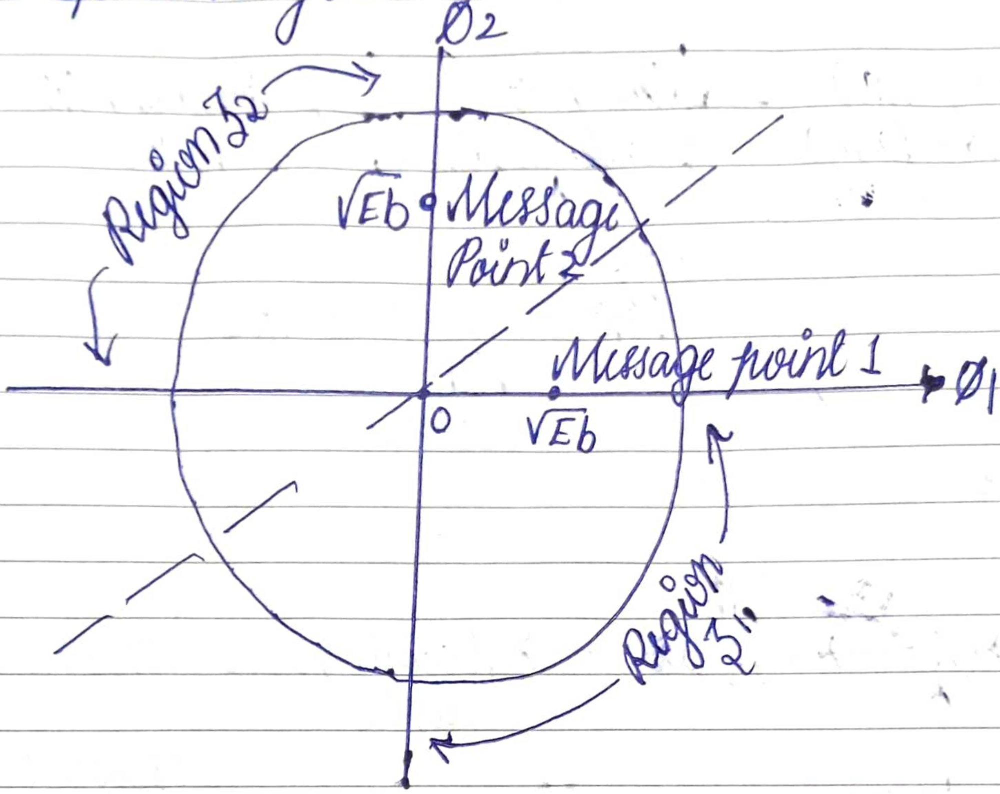
$x(t) \rightarrow \otimes \rightarrow \int_0^{T_b} dt \rightarrow$ Decision device $\rightarrow \begin{cases} \text{Choose 1 if } x_1 > 0 \\ \text{Choose 0 if } x_1 < 0 \end{cases}$

(input to $\otimes$: $\phi_1(t)$)

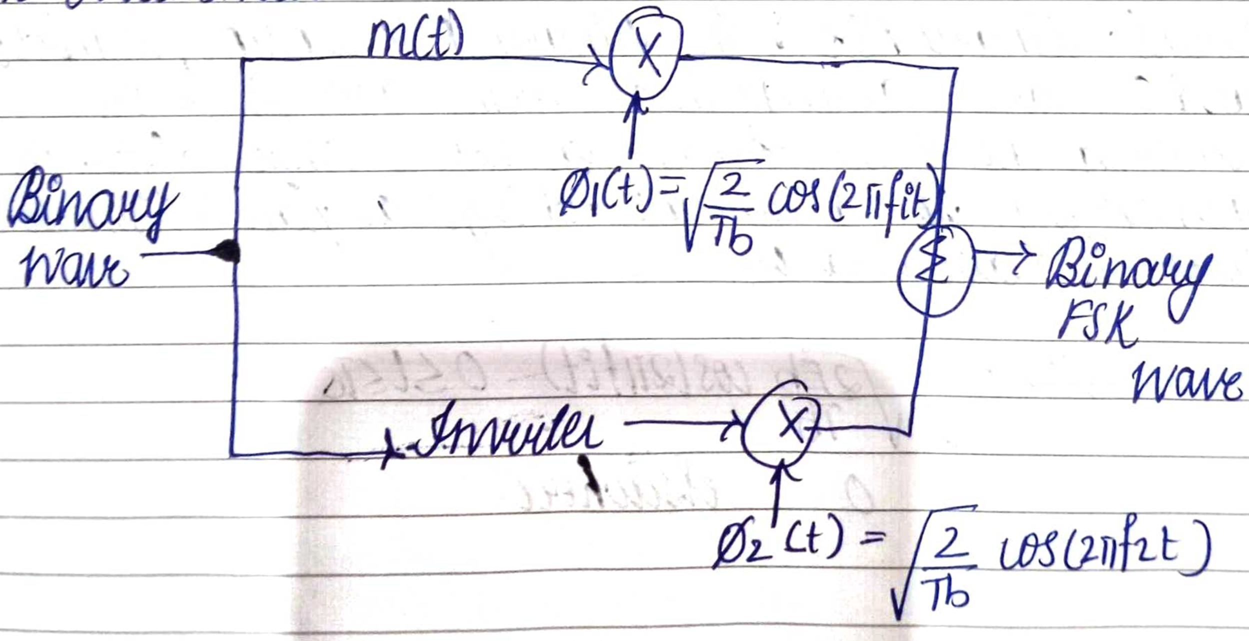
★ Coherent Binary FSK : In Binary FSK system, symbols 1 and 0 are distinguished from each other by transmitting one of two sinusoidal waves that differ in frequency by fixed amount. A typical pair of sinusoidal waves is described by

$$s_i(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_i t) & 0 \le t \le T_s \\ 0 & \text{elsewhere} \end{cases}$$

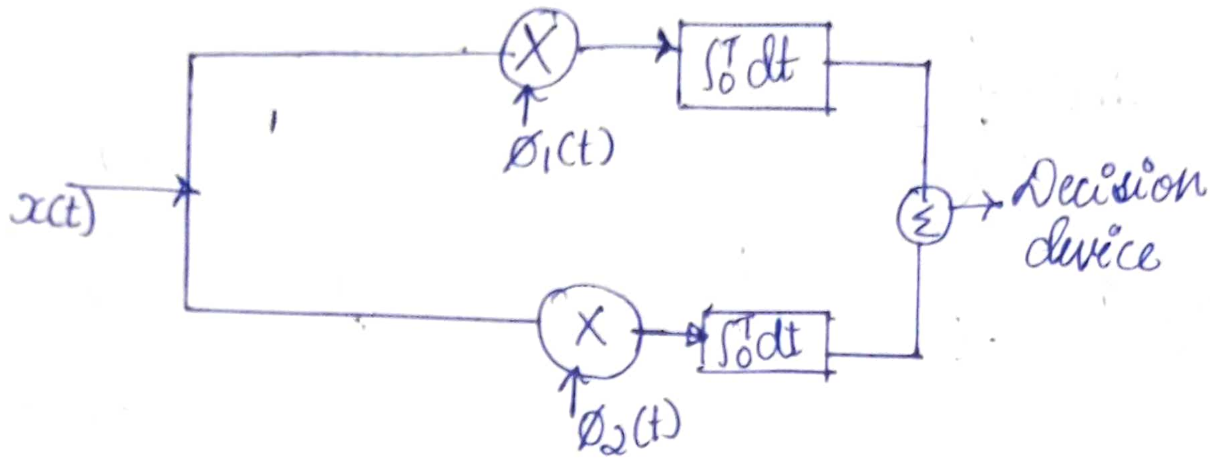
Signal Space Diagram



- FSK Transmitter

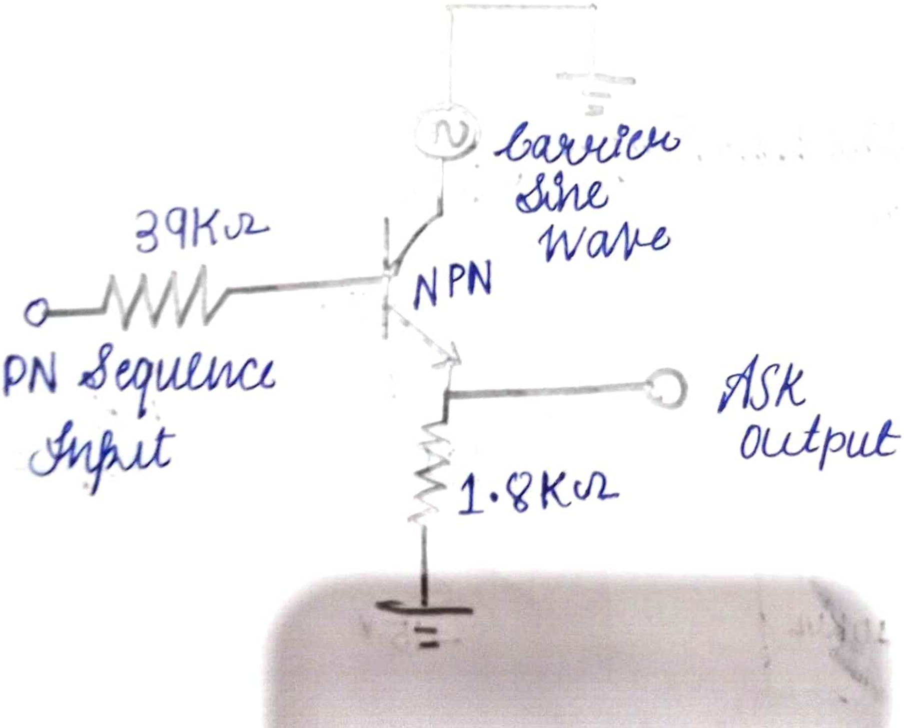


FSK Reciever

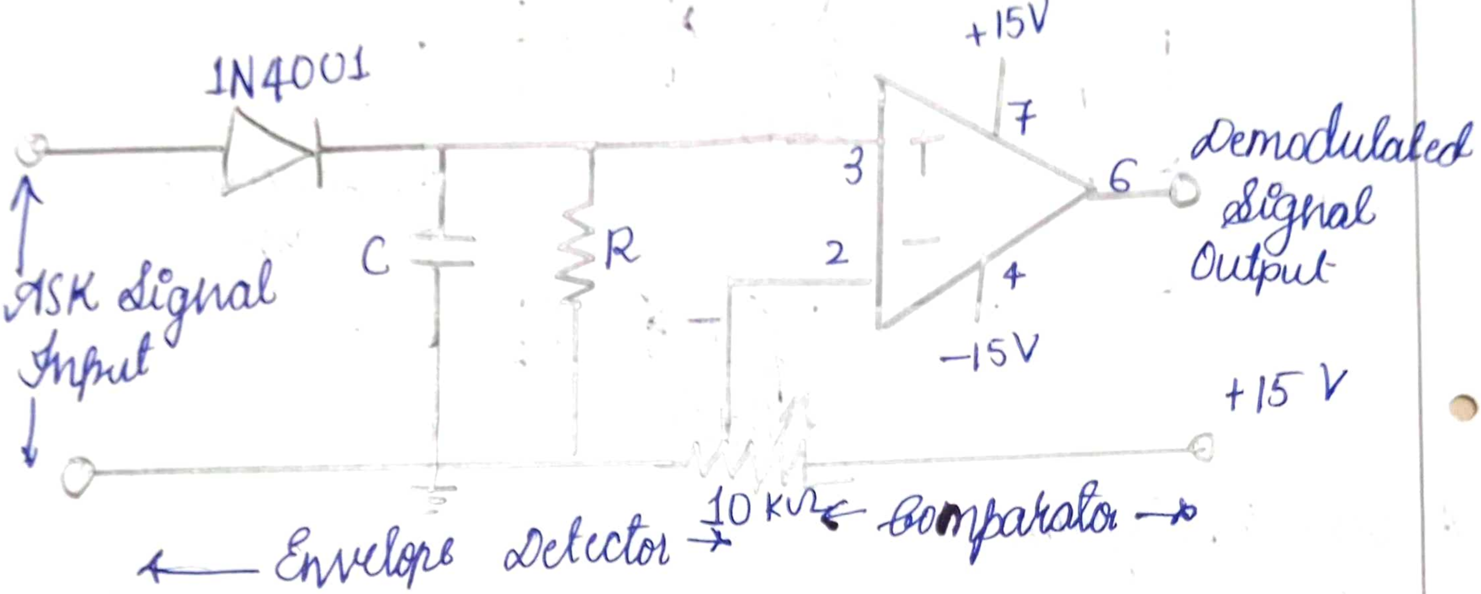


★ Circuit Diagram

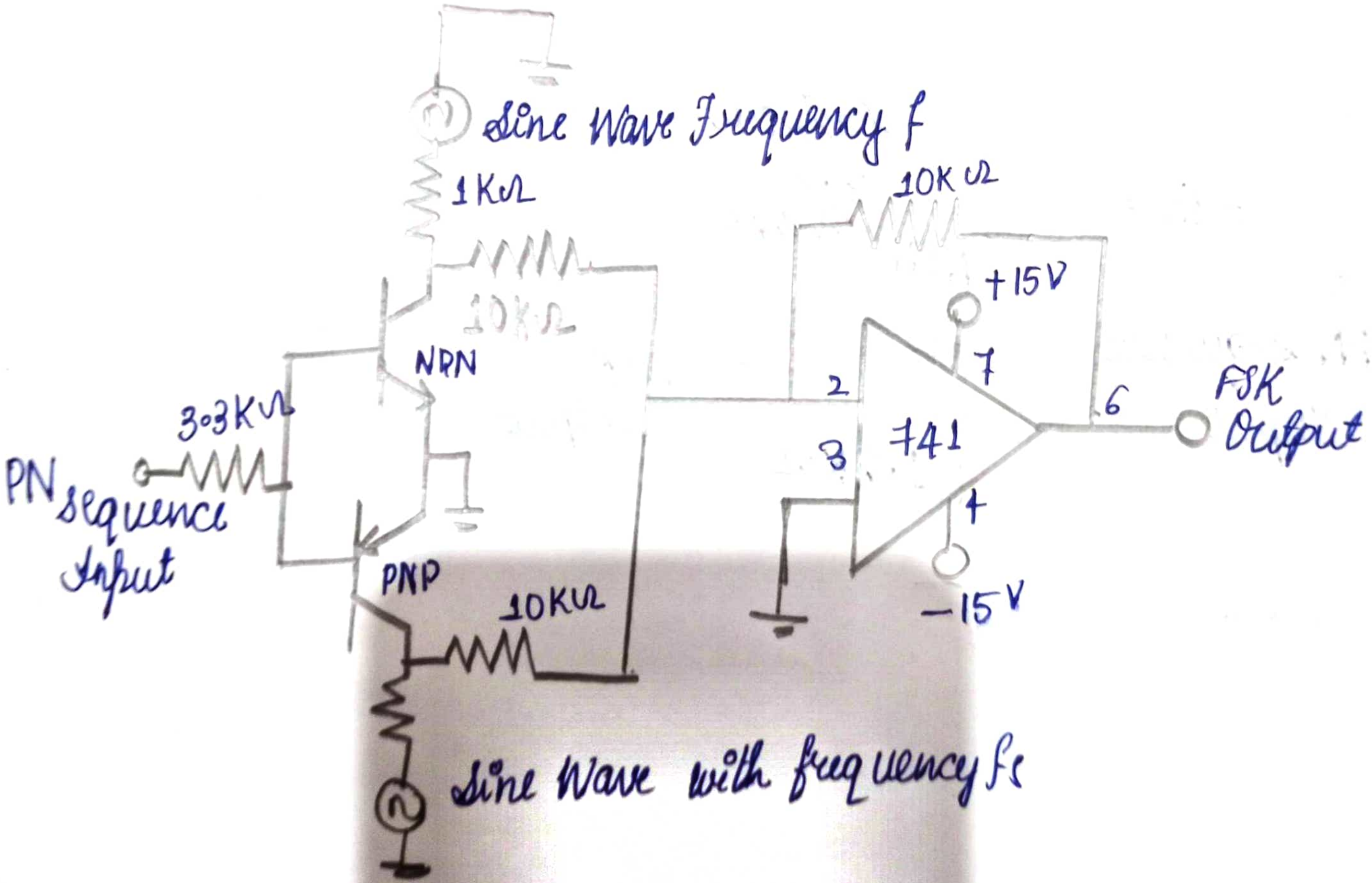
- ASK Modulation



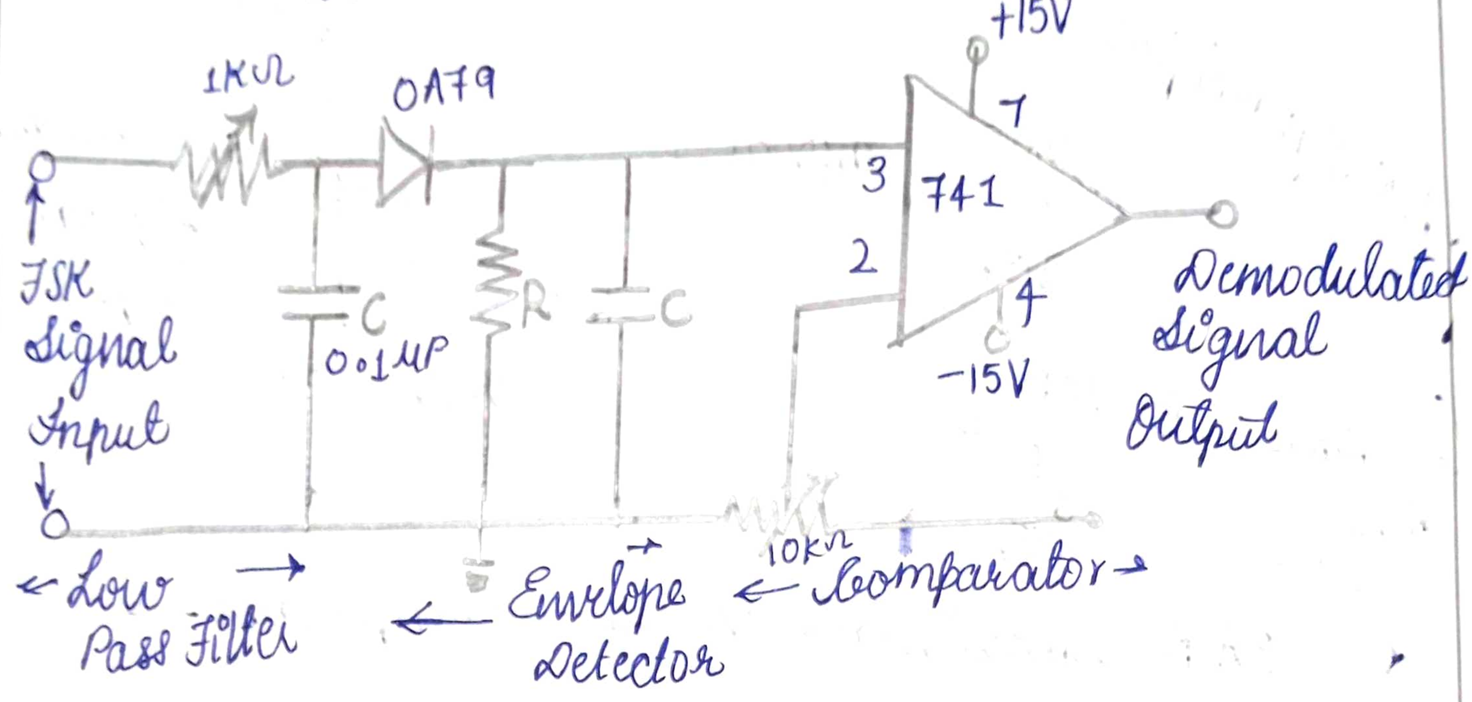
• ASK Demodulation



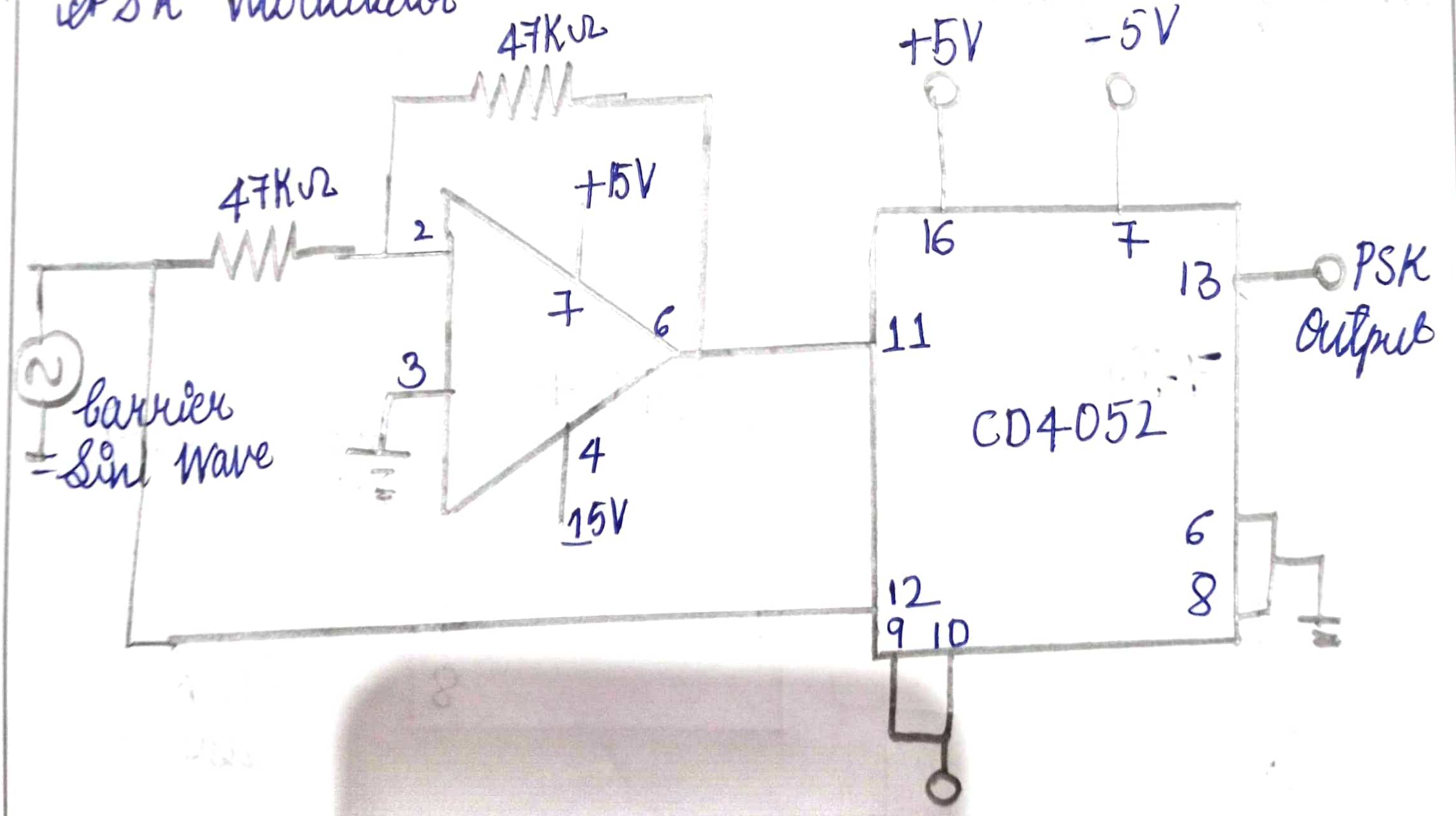
• FSK Modulation



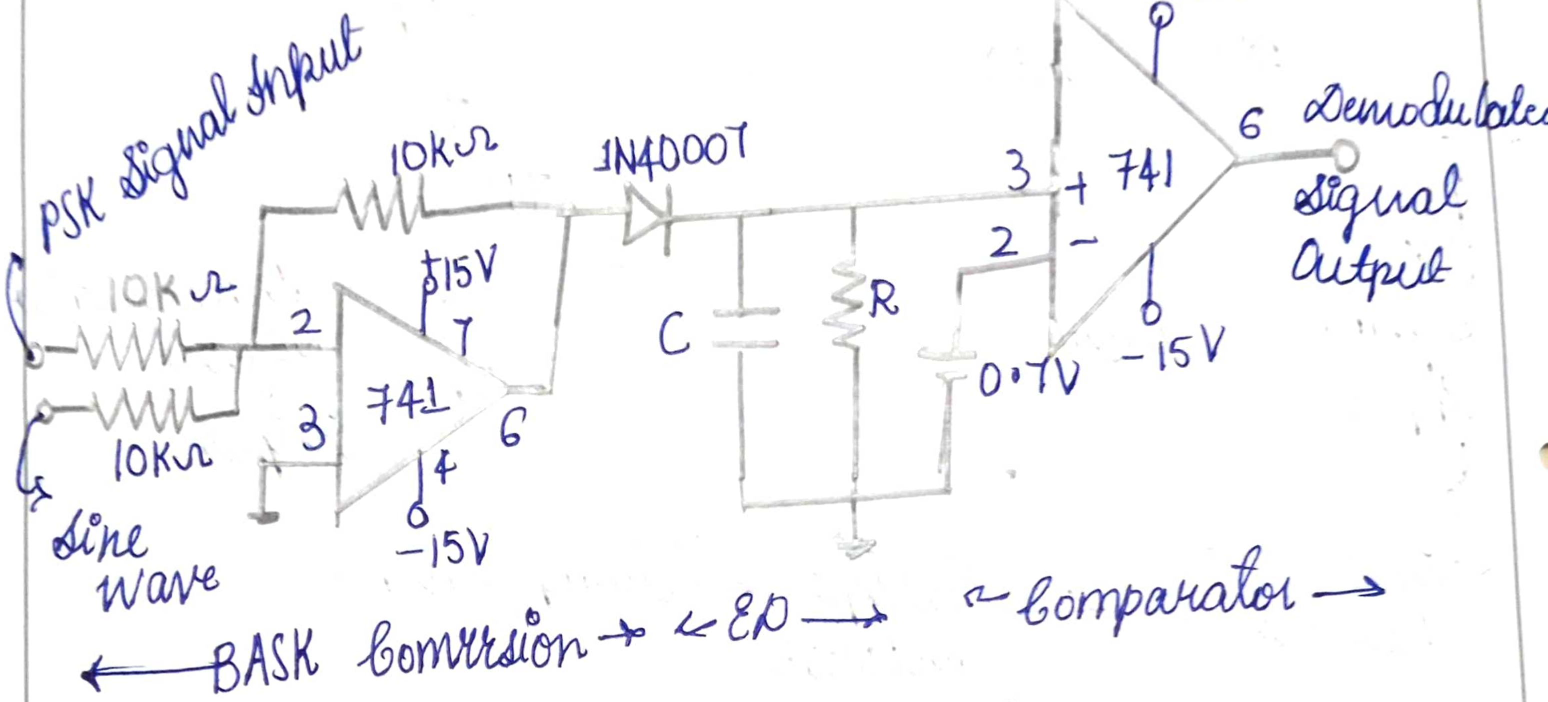
FSK Demodulator



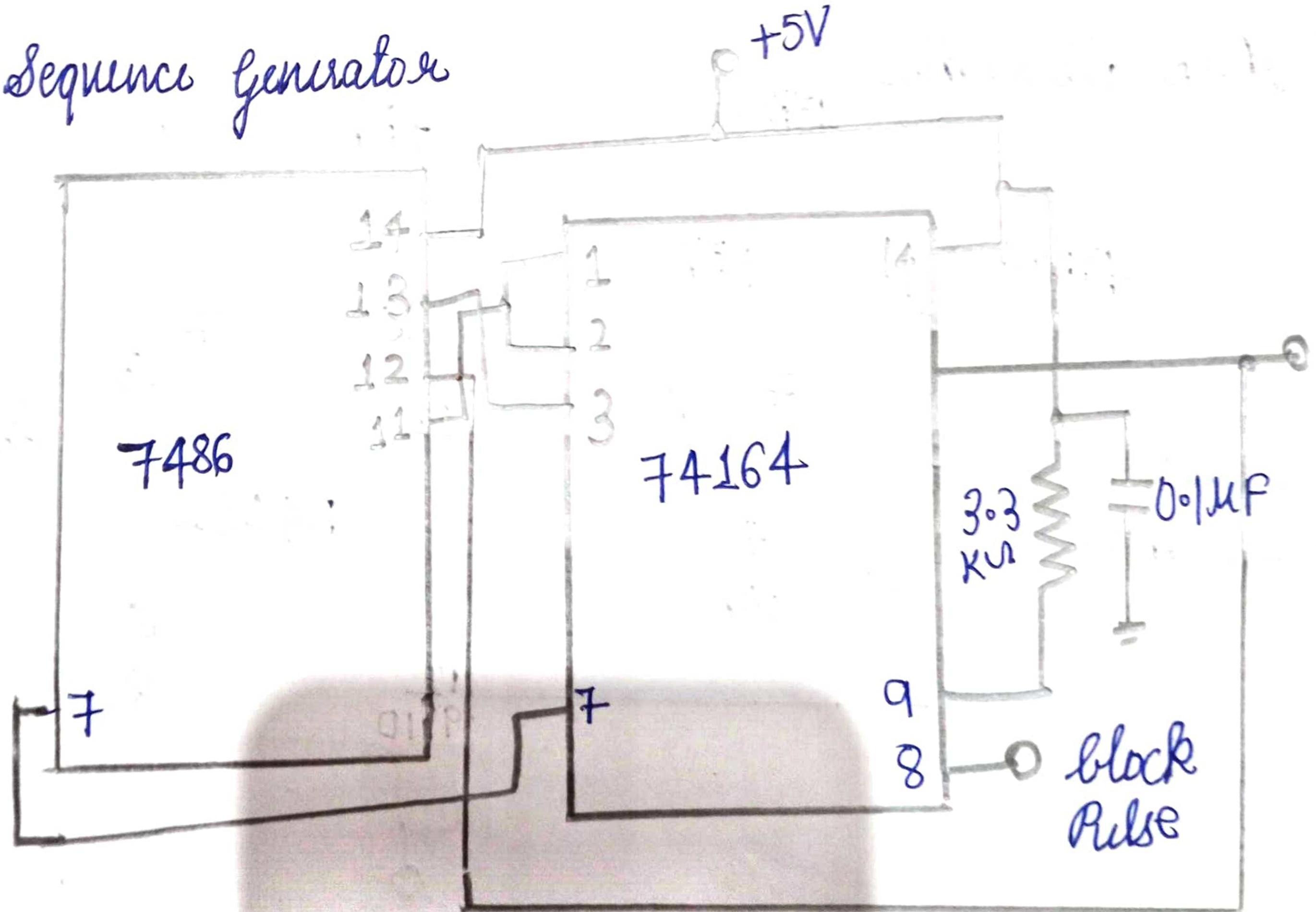
PSK Modulator



PSK Demodulator



PN Sequence Generator

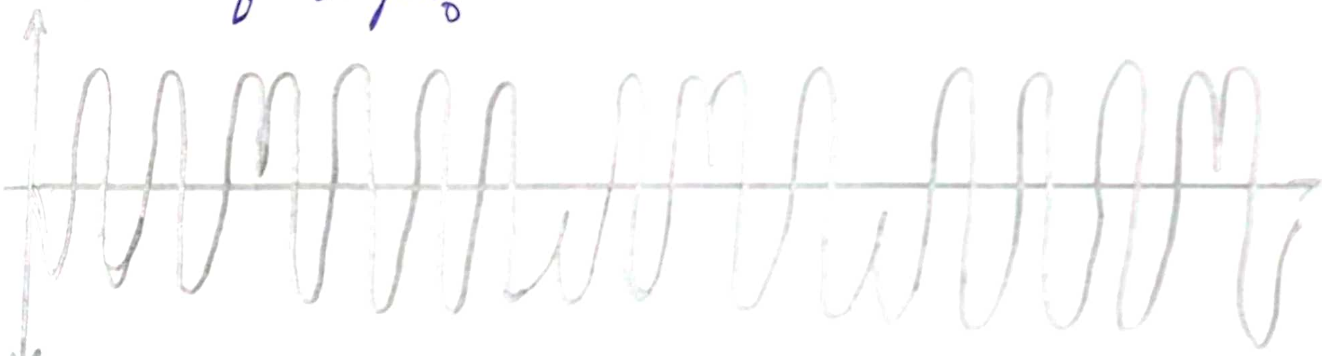


Nature of Graphs

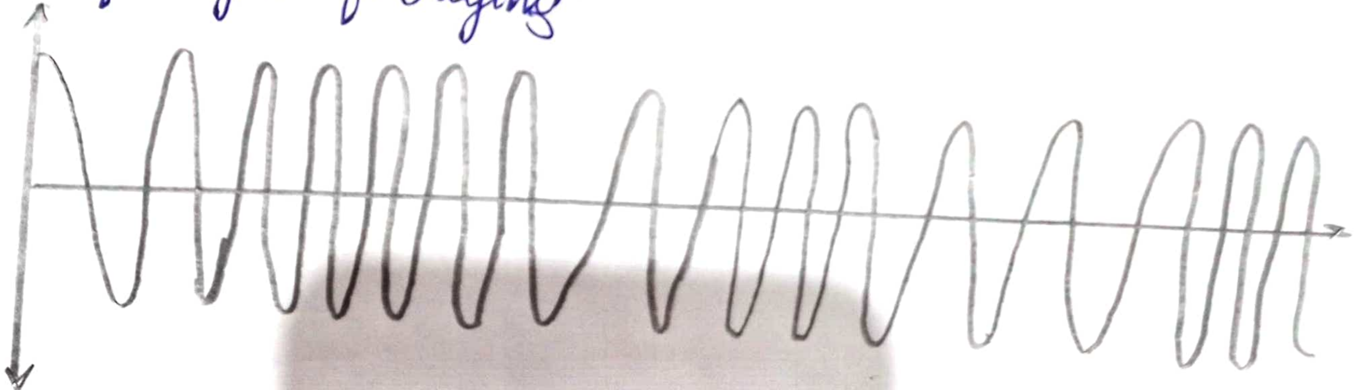
Binary Data 0 1 1 0 1 0 0 1

Amplitude Shift Keying

Phase Shift Keying



Frequency Shift Keying



Results and discussion:

The above experiment confirms the waveforms obtained by the modulation and demodulation of different digital modulation techniques namely ASK, PSK and FSK and are verified with the ideal output

Experiment (7)

Probability of Error Analysis

Aim: To analyse the probability of error using communication toolbox and MATLAB

Objectives: To analyse the probability of error by considering the n-bit sequence, Modulate it with digital modulation schemes
Transmit the signal through AWGN channel
Vary the SNR, compare the theoretical and simulated probability of error

Software Used: Matlab.

Theoretical Background: Suppose that the observation space Z is partitioned in accordance with the maximum-likelihood decision rule, into M regions $\{Z_i, i=1,2,3\cdots M\}$. Suppose also that symbol m_i is transmitted and the observation vector x is recieved. Then an error occurs whenever the recieved signal point represented by x does not fall inside Z_i associated with the message point represented by s_i

Averaging over all the possibly transmitted symbols, we readily see that P_e equals

$$P_e = \sum_{i=1}^{M} P(m_i \text{ sent})\, P(x \text{ does not lie inside } Z_i / m_i)$$

$$= \frac{1}{M}\sum_{i=1}^{M} P(x \text{ does not lie inside } Z_i / m_i)$$

$$= 1 - \frac{1}{M}\sum_{i=1}^{M} P(x \text{ lies inside } Z_i / m_i \text{ sent})$$

$$\therefore \quad P_e = 1 - \frac{1}{M} \sum_{i=1}^{M} \int_{z_i} f_X(x|m_i)\, dx$$

* Probability of Error's for Coherent Binary Modulation Techniques

Modulation Technique.	Probability of error
1) ASK	$\frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)$
2) PSK	$\frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)$
3) FSK	$\frac{1}{2}\,\mathrm{erfc}\left(\sqrt{\frac{E_b}{2N_0}}\right)$

* Results and Discussion:

Upon simulating the probability of Error for various coherent binary modulation Techniques we conclude that the values of obtained from the simulation and the theoretical values are almost equal. Therefore P_e is verified.

Experiment No. 8

Pulse Shaping and Matched Filter.

Aim: To demonstrate matched filter concept using MATLAB

Software Tool: Matlab

Theoretical Background: Consider, for example, a linear filter with impulse response $h_j(t)$. With received signal $x(t)$ used as the filter input, the resulting filter output, $y_j(t)$, is defined by convolution integral

$$y_j(t) = \int_{-\infty}^{\infty} x(\tau)\, h_j(t-\tau)\, d\tau$$

Suppose we now set the impulse response

$$h_j(t) = \phi_j(T-t)$$

The Resulting signal output is

$$y_j(t) = \int_{-\infty}^{\infty} x(\tau)\, \phi_j(T-t+\tau)\, d\tau$$

Sampling this output at time $t=T$, we get

$$y_j(T) = \int_{-\infty}^{\infty} x(\tau)\, \phi_j(\tau)\, d\tau$$

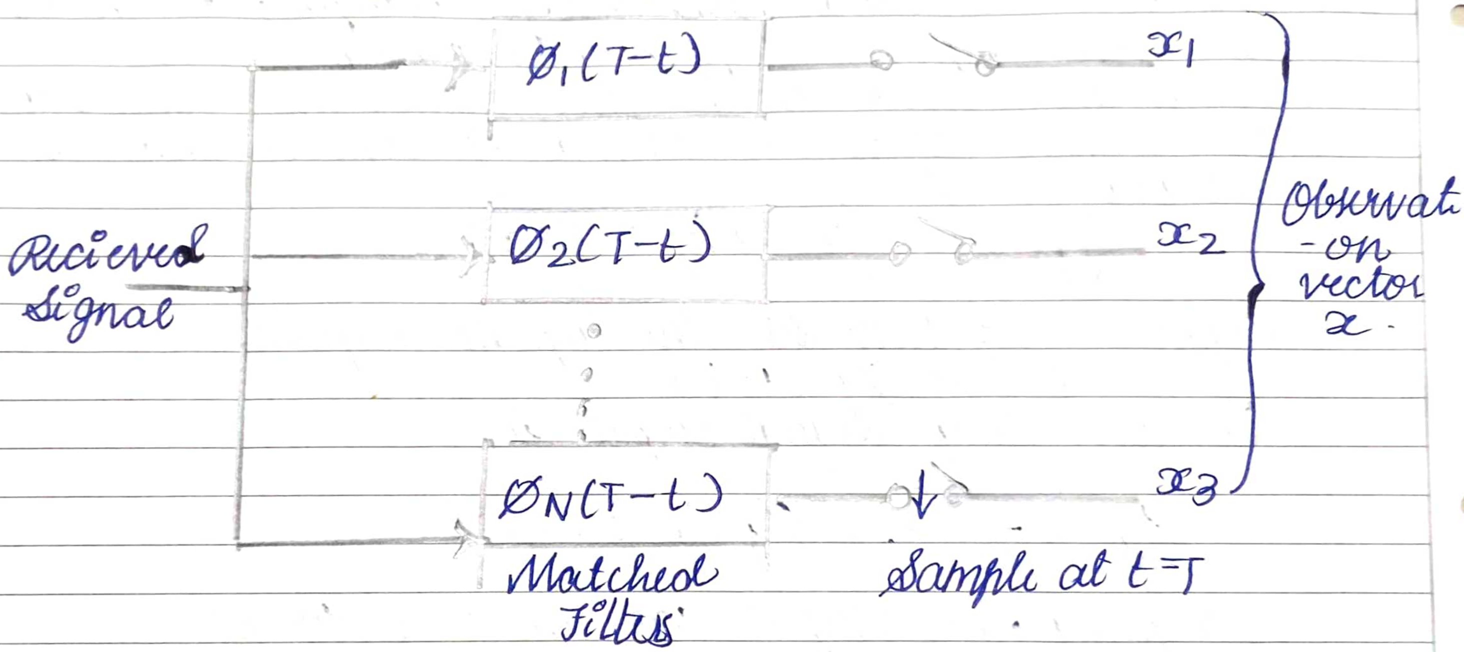
Since $\phi_j(T)$ is zero outside the interval $0 \le t \le T$, we finally get

$$y_j(T) = \int_0^T x(\tau)\, \phi_j(\tau)\, d\tau$$

A filter whose impulse response is a time-reversed and delayed version of some signal $\phi_j(t)$ is said to be matched to $\phi_j(t)$. It is used for

(1) Maximisation of Output signal to noise Ratio
(2) RF Pulse

* Detector Part of matched filter Reciever



→ Results and Dissussion

The filt functionality of the matched filter are analysed using matlab and the obtained results are verified.

Experiment 8

Adaptive Delta Modulation and Companding laws Using Matlab

Aim : To demonstrate the adaptive delta modulation and companding concepts using MATLAB

Software Tool : Matlab

Theoretical Background :

⇒ Adaptive Delta Modulation : The performance of a delta modulator can be significantly increased by making the step size of the modulator assume time varying form. In particular a step segment of input signal the step size is increased. Conversely, when the input signal is varing slowly, the step size is reduced. In this way, the step size is adapted to the level of input. In practical implementation of the system, the step size $\Delta(nT_s)$ or $2\delta(nT_s)$ is constrained to lie between the minimum and the maximum values. In particular we write

$$\delta_{min} \leq \delta(nT_s) \leq \delta_{max}$$

The upper limit δ_{max} controlls the amount of slope overload distortion. The lower limit, δ_{min} controls the idle channel noise

* Companding laws

1) μ-law Companding : In μ law companding, the compressor characteristic $c(x)$ is continous, approximating the linear dependance on x for low levels and a logarithmic one for high level described by

$$\frac{c|x|}{x_{max}} = \frac{\ln\left(1+\mu\frac{|x|}{x_{max}}\right)}{\ln(1+\mu)} \qquad 0 \le \frac{|x|}{x_{max}} \le 1$$

2) A-law Companding : The compressor characteristic $c(x)$ is piecewise, made up of linear segments for low level inputs and logarithmic segment for high level inputs

$$\frac{c|x|}{x_{max}} = \begin{cases} \dfrac{A\frac{|x|}{x_{max}}}{1+\ln A}, & 0 \le \dfrac{|x|}{x_{max}} \le \dfrac{1}{A} \\[2ex] \dfrac{1+\ln\left(\frac{A|x|}{x_{max}}\right)}{1+\ln A} & \dfrac{1}{A} \le \dfrac{|x|}{x_{max}} \le 1 \end{cases}$$

- Results and Disscusion : The Adaptive Delta modulation along with the companding laws is analysed and verified upon simulation

Experiment 9

Huffman Coding

Aim: To demonstrate the concept of Huffman coding using MATLAB

Software Tool : Matlab

Theoretical Background: The Huffman code is a source code whose average word length approches the the fundamental limit set by the entropy of discrete memoryless source. The Huffman code is optimum, in the sense that no other uniquely decodable set of code words has a smaller average codeword length for given memoryless source. The essence of the algorithm used to synthesize the code is to replace the ~~A~~ prescribed set of source statistics of a discrete memoryless source with the simpler one. This reduction process is continued in step by step manner until we are left with the final set of source from ~~[illegible]~~ statistics of only Two, for which (0, 1) is a optimal code. Specifically the Huffmans encoding algorithm preceeds as follows.

1 The source symbols are listed in order of decreasing probability. The two source symbols of lowest probability are assigned a 0 and a 1

2 These two source symbols are regarded as being

combined into a new source symbol with probability equal to the sum of two original probabilities. The probability of the new symbol is placed in list in accordance with its value

3) The procedure is repeated until we are left with final list of source statistic of two which a 0 & 1 is assigned.

- Results and Discussion : The methodology of the Huffman coding is simulated and verified with the theoretical values.