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BGS INSTITUTE OF TECHNOLOGY

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**ADVANCED COMMUNICATION
LABORATORY MANUAL
15ECL76**

For
VII Semester B.E.
2019-2020

**DEPARTMENT OF
ELECTRONICS AND COMMUNICATION ENGINEERING**

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Head of Department

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

VISION:

To develop high quality engineers with technical knowledge, skills and ethics in the area of Electronics and Communication Engineering to meet industrial and societal needs.

MISSION:

1. To provide high quality technical education with up-to-date infrastructure and trained human resources to deliver the curriculum effectively in order to impart technical knowledge and skills.
2. To train the students with entrepreneurship qualities, multidisciplinary knowledge and latest skill sets as required for industry, competitive examinations, higher studies and research activities.
3. To mould the students into professionally-ethical and socially- responsible engineers of high character, team spirit and leadership qualities.

PROGRAM EDUCATIONAL OBJECTIVES (PEO's):

After 3 to 5 years of graduation, the graduates of Electronics and Communication Engineering will;

1. Engage in industrial, teaching or any technical profession and pursue higher studies and research.
2. Apply the knowledge of Mathematics, Science as well as Electronics and Communication Engineering to solve social engineering problems.
3. Understand, Analyze, Design and Create novel products and solutions.
4. Display professional and leadership qualities, communication skills, team spirit, multidisciplinary traits and lifelong learning aptitude.

ADVANCED COMMUNICATION LAB SYLLABUS

Course Learning Objectives:

1. Design and demonstrate the digital modulation techniques
2. Demonstrate and measure the wave propagation in micro strip antennas
3. Characteristics of micro strip devices and measurement of its parameters.
4. Model an optical communication system and study its characteristics.
5. Simulate the digital communication concepts and compute and display various Parameters along with plots/figures.

Laboratory Experiments

1. Time Division Multiplexing and De-multiplexing of two band limited signals.
2. ASK generation and detection
3. FSK generation and detection
4. PSK generation and detection
5. Measurement of frequency, guide wavelength, power, VSWR and attenuation in microwave test bench.
6. Measurement of directivity and gain of micro strip dipole and Yagi antennas.
7. Determination of :
 - a. Coupling and isolation characteristics of micro strip directional coupler.
 - b. Resonance characteristics of micro strip ring resonator and computation of dielectric constant of the substrate.
 - c. Power division and isolation of micro strip power divider.
8. Measurement of propagation loss, bending loss and numerical aperture of an optical fiber.
9. Simulate NRZ, RZ, half-sinusoid and raised cosine pulses and generate eye diagram for binary polar signalling.
10. Simulate the Pulse code modulation and demodulation system and display the waveforms.
11. Simulate the QPSK transmitter and receiver. Plot the signals.

Beyond Syllabus:

1. QPSK MODULATION AND DEMODULATION

Course Outcomes:

1. Outline the digital modulation techniques-ASK, FSK and PSK.
2. Experiment with microwave test bench to determine frequency, guide wavelength, VSWR and attenuation.
3. Measure the parameters of micro strip devices-directional coupler, micro strip ring resonator and power divider.
4. Construct the experimental setup to find out propagation loss, bending loss and numerical aperture.
5. Demonstrate the simulation of types of digital signals - pulse code modulation, QPSK and DPSK.

Expt No-1. TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING OF TWO BANDLIMITED SIGNAL

Aim of the Experiment: To study the performance of the circuit that is used for multiplexing of two band limited signals.

Components required:

IC 7493

IC 4051

Theory:

Time-division multiplexing (TDM) is a type of digital or (rarely) analog multiplexing in which two or more signals or bit streams are transferred apparently simultaneously as sub-channels in one communication channel, but are physically taking turns on the channel. The time domain is divided into several recurrent timeslots of fixed length, one for each sub-channel. A sample byte or data block of sub-channel 1 is transmitted during timeslot 1, sub-channel 2 during timeslot 2, etc. One TDM frame consists of one timeslot per sub-channel. After the last sub-channel the cycle starts all over again with a new frame, starting with the second sample, byte or data block from sub-channel 1, etc.

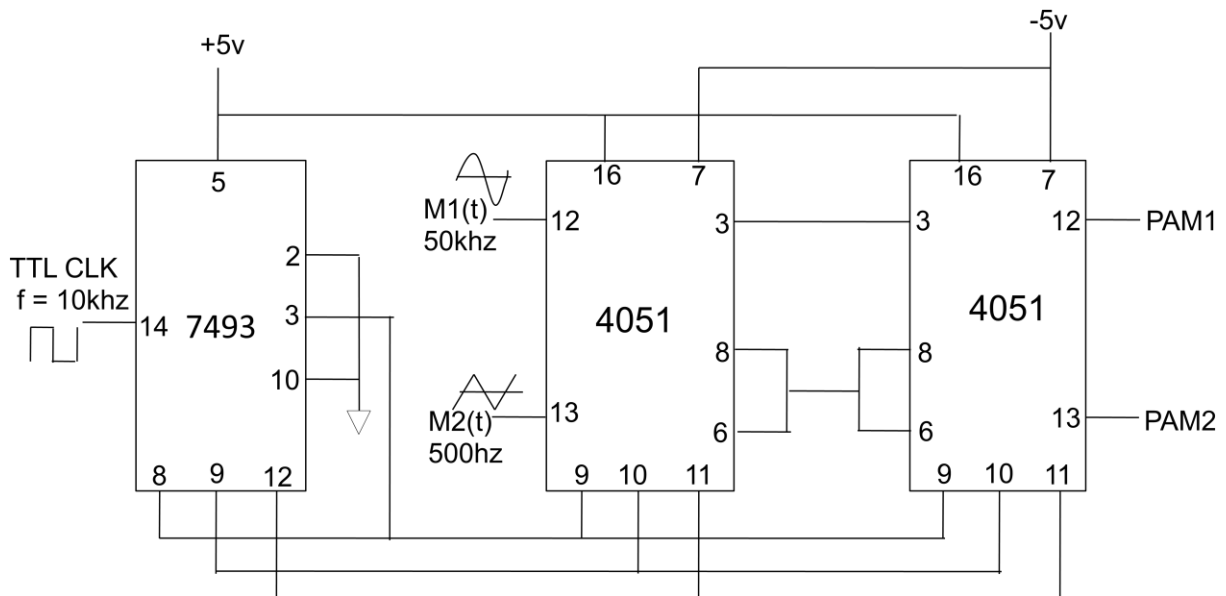
Applications:

- The plesiochronous digital hierarchy (PDH) system, also known as the PCM system, for digital transmission of several telephone calls over the same four-wire copper cable (T-carrier or E-carrier) or fiber cable in the circuit switched digital telephone network.
- The SDH and synchronous optical networking (SONET) network transmission standards, that have surpassed PDH.
- The RIFF (WAV) audio standard interleaves left and right stereo signals on a per-sample basis
- The left-right channel splitting in use for stereoscopic liquid crystal shutter glasses

TDM can be further extended into the time division multiple accesses (TDMA) scheme, where several stations connected to the same physical medium, for example sharing the same frequency channel, can communicate. Application examples include:

- The GSM telephone system

Circuit diagram:

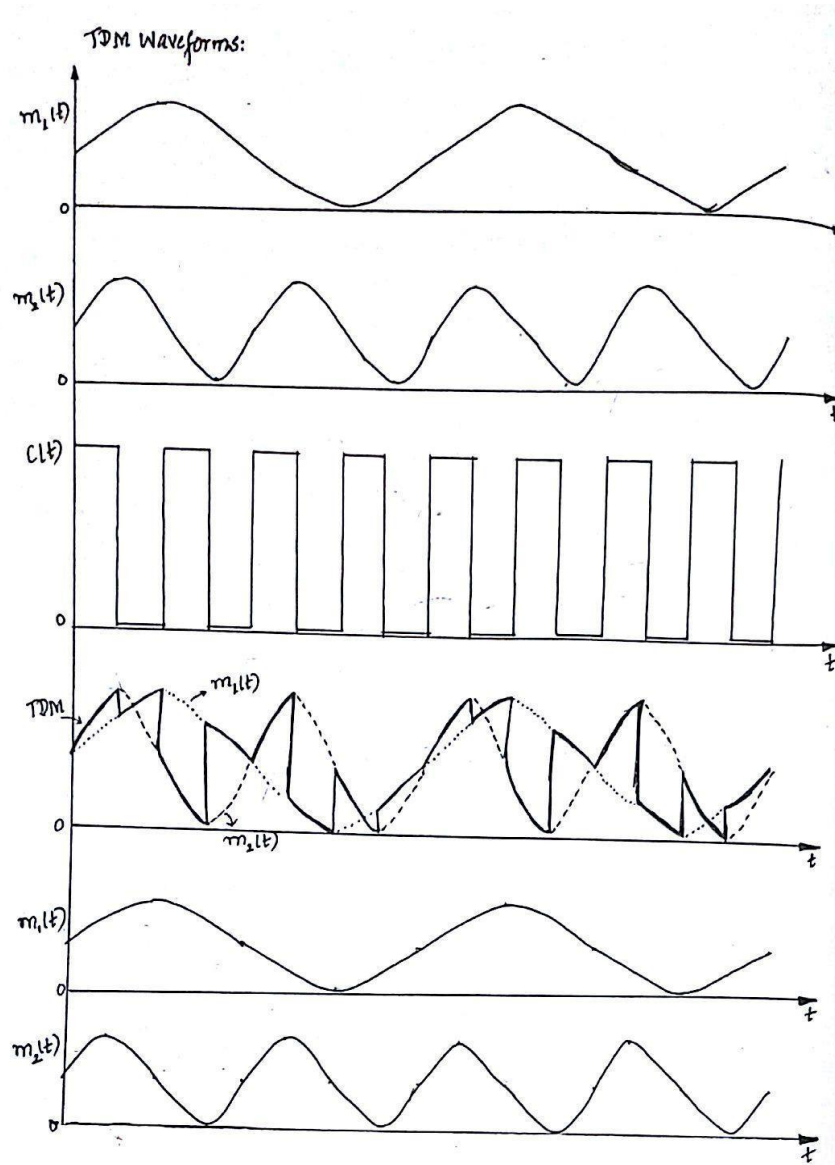


Procedure:

1. Rig up the circuit as shown in the figure.
2. Switch on the power supplies and adjust the V_{CC} to the required values.
3. Apply the TTL clock of 10 KHz at pin 14(IC 7493 counter).
4. Apply the input signals at pin 12(sine wave 3Vp-p 500hz) And pin 13(triangular wave 3Vp-p 50 Hz) of IC 4051.
5. Observe the TDM output signal at pin 3 of 1st IC 4051(multiplexed output).
6. Change the frequency and amplitude of the input signals and observe the output.
- 7 The 2nd IC 4051 acts as de-multiplexer. Observe the de-multiplexed output at pin 12 and pin 13.

8. Compare the input and de-multiplexed output.

Waveforms:



Results:

Observations/ Comments:

Expt No-2. AMPLITUDE SHIFT KEYING MODULATION AND DEMODULATION

Aim of the Experiment: To design and verify the operation of ASK generator and demodulator.

Components Required:

1. Transistor (BC107/108)
2. Diode(OA79),
3. Resistor – 22K Ω ,2.2K Ω , 100K Ω ,
4. Capacitor -0.01 μ F,
5. Op-Amp(μ A741)

Theory:

Amplitude-shift keying (ASK) is a form of amplitude modulation that represents digital data as variations in the amplitude of a carrier wave. In an ASK system, the binary symbol 1 is represented by transmitting a fixed-amplitude carrier wave and fixed frequency for a bit duration of T seconds. If the signal value is 1 then the carrier signal will be transmitted; otherwise, a signal value of 0 will be transmitted.

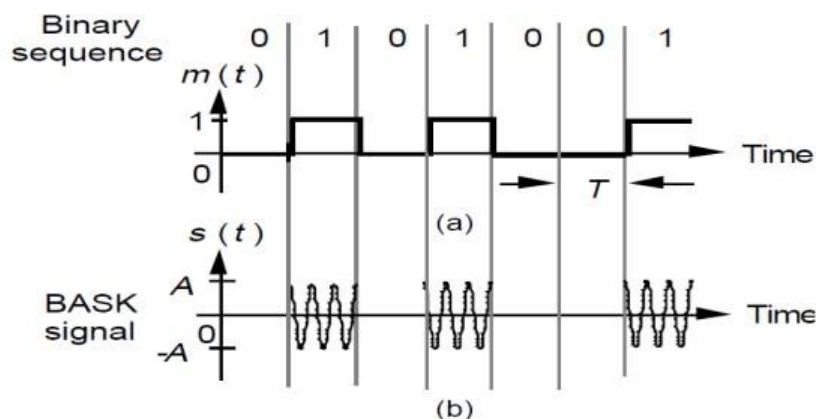


Figure.1 Binary modulating signal and (b) BASK signal.

ASK Generation: - A block diagram of a basic ASK modulator is shown in Fig.2.

The multiplier can be modeled by analog multiplier or by analog switch.

Demodulation methods

It is apparent from Figures 1 that the ASK signal has a well defined envelope. Thus it is amenable to demodulation by an envelope detector. A synchronous demodulator would also be appropriate.

Note that:

- Envelope detection circuitry is simple.
- Synchronous demodulation requires a phase-locked local carrier and therefore carrier acquisition circuitry .

With band limiting of the transmitted ASK neither of these demodulation methods would recover the original binary sequence; instead, their outputs would be a band limited version. Thus further processing - by some sort of decision-making circuitry for example - would be necessary.

Thus demodulation is a two-stage process:

- Recovery of the bandlimited bit stream
- Regeneration of the binary bit stream

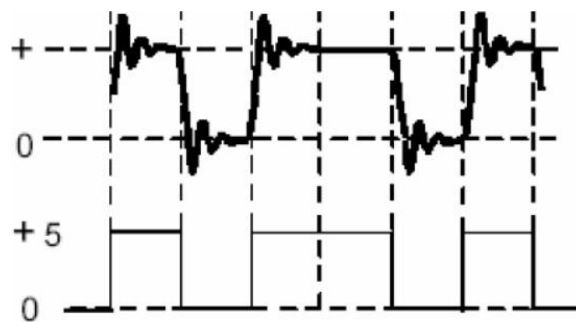


Figure.3 The two stages of the demodulation process

Envelope Detection

To demodulation by an envelope detector. Envelope detection is used if Signal to Noise Ratio

(SNR) is high. The envelope detector is a rectifier and low pass filter circuit. The circuit rejects the carrier frequency components and produces a dc output that corresponds to the original binary-data.

The output from the detector will be a distorted binary signal. A threshold device, such as comparator or decision maker, can be used to ‘clean up’ the recovered envelope detector output waveform. Figure.4 shows the block diagram for such system.

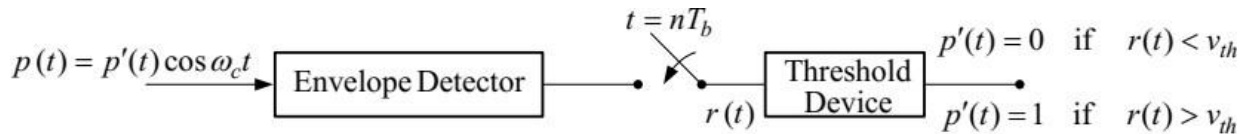
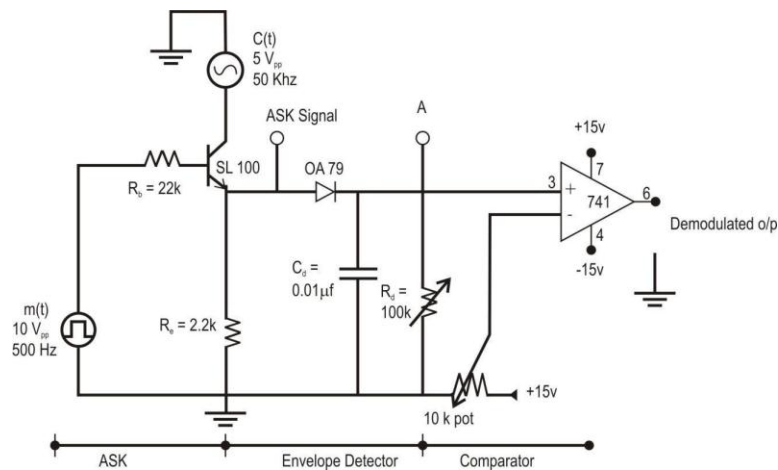


Figure.4 Amplitude Shift Keying Envelope Detection

Circuit Diagram:

ASK modulator and demodulator



Design:

Let $V_c = 5$ volts peak-to-peak, $V_m = 10$ volts peak-to-peak, $f_m = 500$ Hz, $f_c = 50$ kHz.

Assume $h_{fe} = 30$, $V_{BEsat} = 0.7$ volts, $V_{CEsat} = 0.3$ volts, $I_c = 1$ mA, $I_c = I_e$.

$$V_c \text{ peak} = V_{CEsat} + I_e R_e$$

$$2.5 = 0.3 + (1 \text{ m})R_e,$$

$$\Rightarrow R_e = 2.2 \text{ k}\Omega$$

$$V_{m,peak} = R_b I_b + V_{BE,sat} + I_e R_e$$

$$R_e 5 = R_b I_b + 0.7 + 2.2,$$

$$\text{where } I_b = I_c / h_{fe}$$

then $R_{bmax} = 63 \text{ k}\Omega$, Choose $R_b = 22 \text{ k}\Omega$

Envelope Detector:

$$1/f_m > R_d C_d > 1/f_c, \quad \text{hence} \quad 2\text{ms} > R_d C_d > 20\mu\text{s}$$

$$\text{Let } R_d C_d = 50/f_c = 1 \text{ ms}$$

Assume $C_d = 0.01 \mu\text{F}$, then $R_d = 100 \text{ k}\Omega$

Procedure:**Generation of ASK:**

1. Rig up the circuit as shown in circuit diagram.
2. Apply the message signal $m(t) > 8\text{vp-p}, 500\text{hz}$ square wave
3. Apply the carrier signal $c(t) 5\text{vp-p}, 50\text{khz}$ sine wave
4. Observe the ASK output at the emitter of transistor using a CRO.

Demodulation of ASK:

1. Rig up the circuit as shown in the figure.
2. Observe the output at the envelope detector (diode).
1. Observe the amplitude and waveform at the pin no 6 of op-amp which gives output

corresponding to envelope detector stage.

2. Vary the comparator voltage (pin 2) 0-5v to obtain the clear square wave.
3. The demodulated signal $m'(t)$ at the comparator (pin 6) matches with the message signal $m(t)$.
4. Calculate the bit error from $m'(t)$ and $m(t)$.

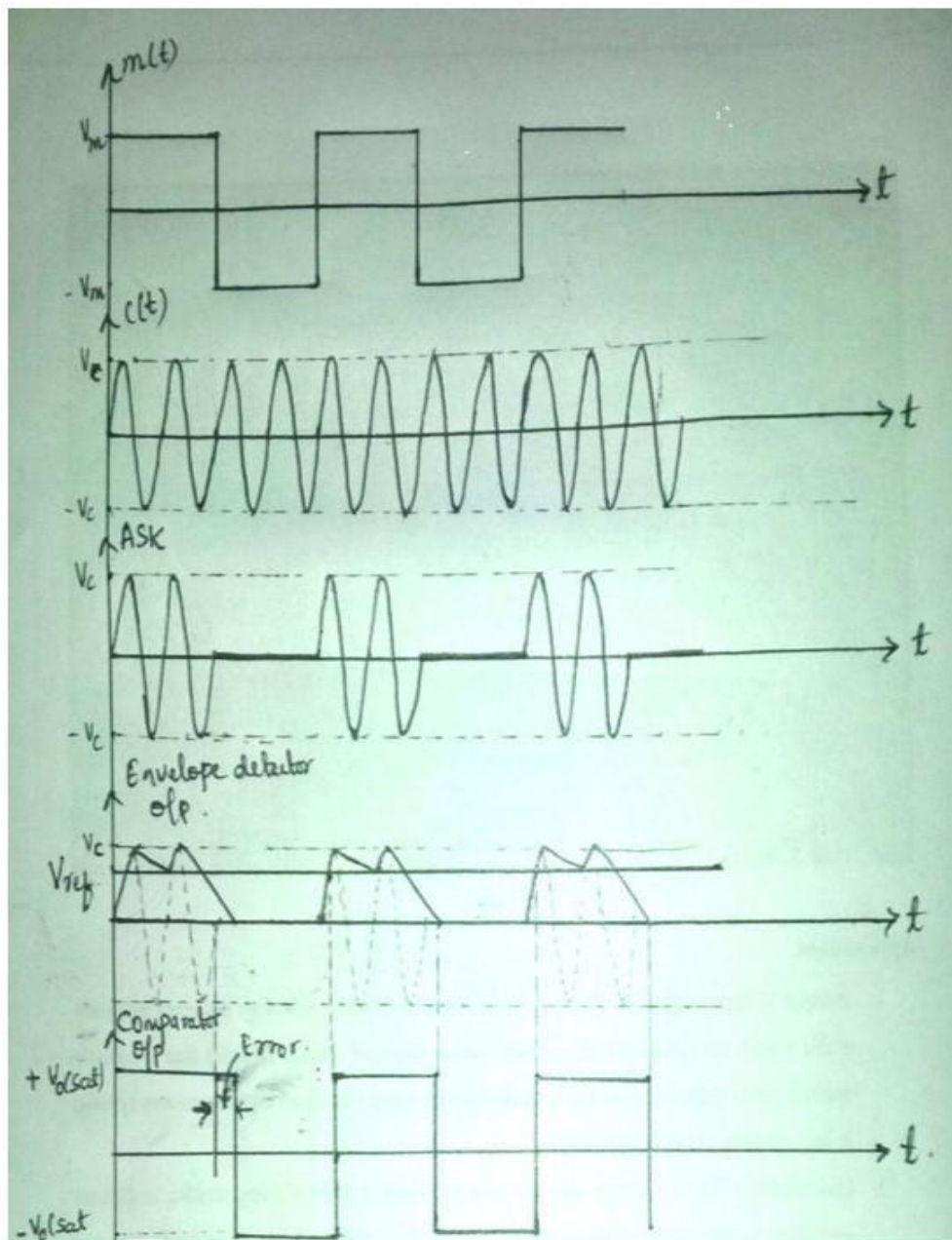
Working:

- Since we are using an NPN transistor (BC108 / 107) for proper working of a **transistor**, emitter base junction should be forward biased and CB junction should be reverse biased.
- When data '1' is applied to base then transistor works and it acts as a closed switch transmitting the carrier from collector to emitter.
- If data '0' is fed to base of the transistor then it acts as an open switch hence no output at the emitter terminal hence the ASK.
- Envelop detector is used to detect the digital data, when bit 1 is transmitted capacitor charges and discharges rapidly maintaining some finite voltage at the output. When bit 0 is transmitted capacitor completely discharges as shown in waveform.
- Finally comparator is used to compare the output of detector and the threshold, if output greater than threshold it generates a new bit 1 else 0.

Like AM, ASK is also linear and sensitive to atmospheric noise, distortions, propagation conditions on different routes in PSTN, etc. Both ASK modulation and demodulation processes are relatively inexpensive. **The ASK technique is also commonly used to transmit digital data over optical fiber.**

For LED transmitters, binary 1 is represented by a short pulse of light and binary 0 by the absence of light. Laser transmitters normally have a fixed "bias" current that causes the device to emit a low light level. This low level represents binary 0, while a higher-amplitude light wave represents binary 1.

ASK modulation and demodulation waveforms :



Results :

Observation :

Expt No-3. FREQUENCY SHIFT KEYING MODULATION AND DEMODULATION

Aim of the Experiment: To design and verify the operation of FSK generator and detector.

Components Required:

1. Transistor-BC107/108 & SK100
2. Diode(OA79)
3. Resistors 22K Ω , 10K Ω , 2.2K Ω , 100K Ω
4. Capacitors 0.01 μ F
5. Op-Amp(μ A741).

Theory:

FSK is one of the digital modulation technique. Here frequency of the carrier is switched between two values. A sinusoidal of amplitude 'A' and frequency $fc1$ is used to represent a binary '1' and frequency $fc2$ is used to represent binary '0'. FSK modulated waveform can be represented as,

Design

1. Modulator

$$R_C = \frac{V_{CC} - V_{CE}}{I_C}$$

$$R_C = 1K\Omega$$

$$R_B = \frac{V_{in} - V_{CE}}{I_B} = 10K\Omega$$

2. Demodulator

Low Pass filter $f = 2$ KHz, Let $c = 0.1$ μ F

$$f = \frac{1}{2\pi R c}$$

$$\text{So, } R = 7.9K\Omega$$

$$\frac{1}{f_m} \gg RC \gg \frac{1}{f_c}$$

$$f_m = 200\text{Hz}, f_c = 10\text{K Hz}$$

$$\frac{1}{200} \gg RC \gg \frac{1}{10 * 10^3}$$

Let C = 0.1 μf , and hence R = 10K Ω

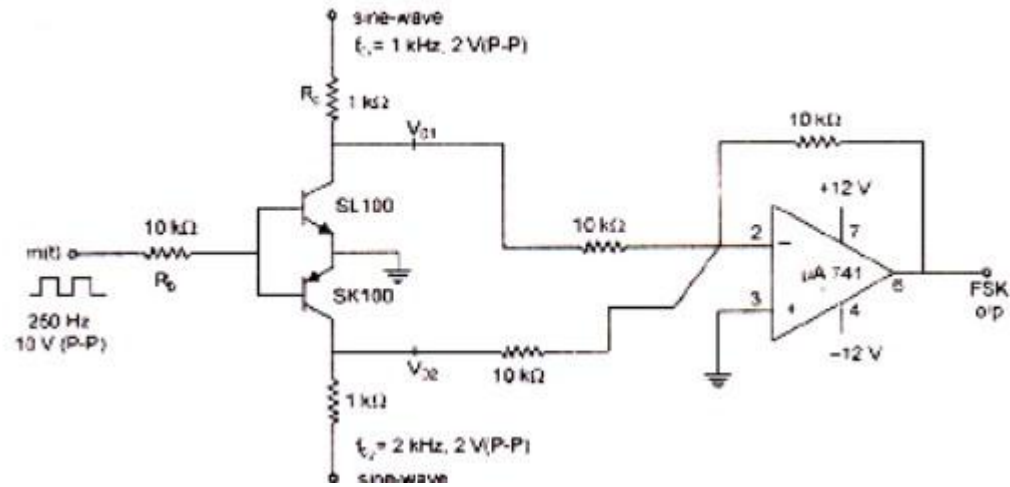
Application:

Some early CW transmitters employed an arc converter that could not be conveniently keyed. Instead of turning the arc on and off, the key slightly changed the transmitter frequency in a technique known as the *compensation-wave method*. The compensation-wave was not used at the receiver. Spark transmitters used for this method consumed a lot of bandwidth and caused interference, so it was discouraged by 1921.

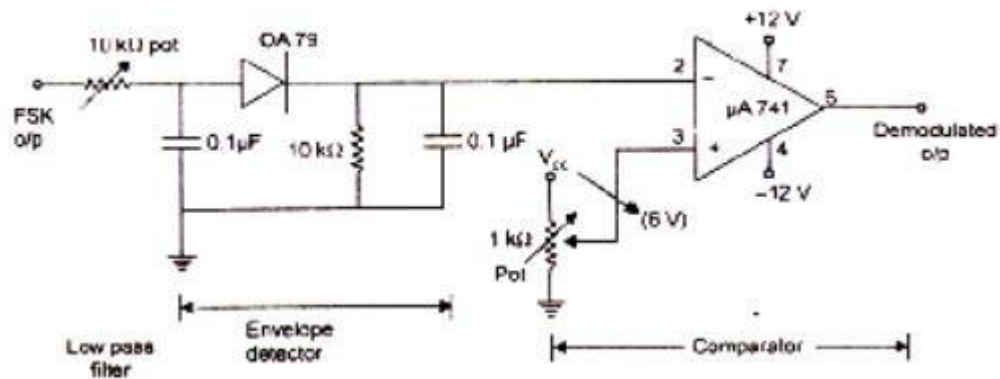
Most early telephone-line modems used audio frequency-shift keying (AFSK) to send and receive data at rates up to about 1200 bits per second. The Bell 103 and Bell 202 modems used this technique. Even today, North American caller ID uses 1200 baud AFSK in the form of the Bell 202 standard. Some early microcomputers used a specific form of AFSK modulation, the Kansas City standard, to store data on audio cassettes. AFSK is still widely used in amateur radio, as it allows data transmission through unmodified voice band equipment.

AFSK is also used in the United States' Emergency Alert System to transmit warning information. It is used at higher bitrates for Weather copy used on Weather radio by NOAA in the U.S.

Modulator

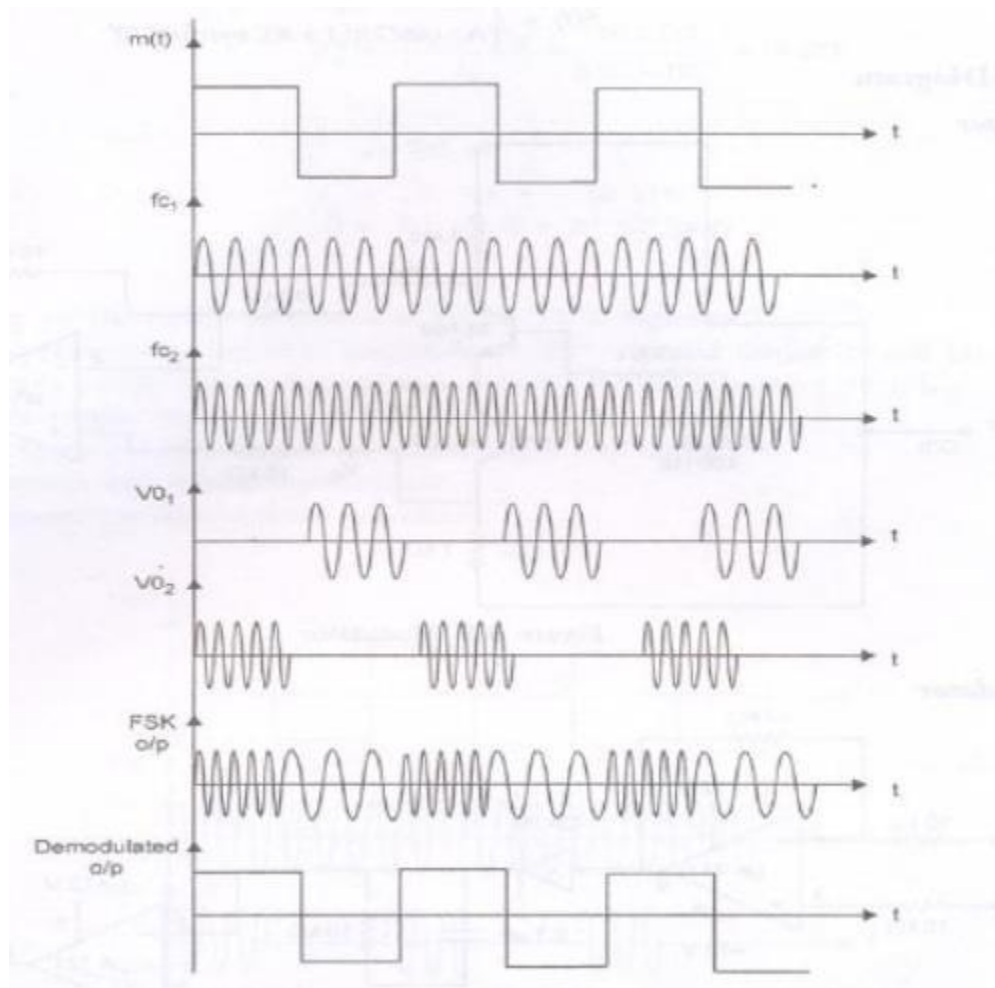


Demodulator



Procedure:

1. Rig up the modulator circuit as shown in the figure.
2. Apply carrier of amplitude 2 V(P- P) and frequency 1 kHz.
3. Apply carrier of amplitude 2 V(P- P) and frequency 2 kHz.
4. Apply message signal of amplitude 10 V(P - P) and frequency of 250 Hz. .
5. Observe ASK outputs at each collector of transistor, and also observe FSK output at pin 6 of op-amp.
6. Connect demodulator circuit.
7. Observe the demodulated output on CRO.



Expected Wave form:

Results :

Observation/Comments:

Expt No-4. PHASE SHIFT KEYING MODULATION AND DEMODULATION

Aim of the Experiment: To generate PSK signal and to demodulate the PSK signal.

Component Required:

1. Transistor-B108C & SK100
2. Diode(OA79)
3. Resistors 22K Ω , 10K Ω , 2.2K Ω , 100K Ω , 10K Ω pot, 45K Ω pot
4. Capacitors 0.01 μ F, 0.1 μ F
5. Op-Amp(μ A741).

Theory:

BPSK (also sometimes called PRK, phase reversal keying, or 2PSK) is the simplest form of phase shift keying (PSK). It uses two phases which are separated by 180° and so can also be termed 2-PSK. It does not particularly matter exactly where the constellation points are positioned, and in this figure they are shown on the real axis, at 0° and 180°. Therefore, it handles the highest noise level or distortion before the demodulator reaches an incorrect decision. That makes it the most robust of all the PSKs. It is, however, only able to modulate at 1 bit/symbol and so is unsuitable for high data-rate applications.

In the presence of an arbitrary phase-shift introduced by the communications channel, the demodulator is unable to tell which constellation point is which. As a result, the data is often differentially encoded prior to modulation. BPSK is functionally equivalent to 2- QAM modulation.

Applications:

Owing to PSK's simplicity, particularly when compared with its competitor quadrature amplitude modulation, it is widely used in existing technologies.

The wireless LAN standard, IEEE 802.11b-1999, uses a variety of different PSKs depending on the data-rate required. At the basic-rate of 1 M bit/s, it uses DBPSK (differential

BPSK). To provide the extended-rate of 2 M bit/s, DQPSK is used. In reaching 5.5 Mbit/s and the full-rate of 11 M bit/s, QPSK is employed, but has to be coupled with complementary code keying. The higher-speed wireless LAN standard, IEEE 802.11g-2003 has eight data rates: 6, 9, 12, 18, 24, 36, 48 and 54 M bit/s. The 6 and 9 M bit/s modes use OFDM modulation where each sub-carrier is BPSK modulated. The 12 and 18 M bit/s modes use OFDM with QPSK. Because of its simplicity BPSK is appropriate for low-cost passive transmitters, and is used in RFID standards such as ISO/IEC 14443 which has been adopted for biometric passports, credit cards such as American Express's Express Pay, and many other applications.

Bluetooth 2 will use $\pi / 4$ -DQPSK at its lower rate (2 M bit/s) and 8-DPSK at its higher rate (3 M bit/s) when the link between the two devices is sufficiently robust. Bluetooth 1 modulates with Gaussian minimum-shift keying, a binary scheme, so either modulation choice in version 2 will yield a higher data-rate. A similar technology, IEEE 802.15.4 (the wireless standard used by ZigBee) also relies on PSK. IEEE 802.15.4 allows the use of two frequency bands: 868–915 MHz using BPSK and at 2.4 GHz using OQPSK.

Notably absent from these various schemes is 8-PSK. This is because its error-rate performance is close to that of 16-QAM — it is only about 0.5 dB better — but its data rate is only three-quarters that of 16-QAM. Thus 8-PSK is often omitted from standards and, as seen above, schemes tend to 'jump' from QPSK to 16-QAM (8-QAM is possible but difficult to implement).

The general form for BPSK follows the equation

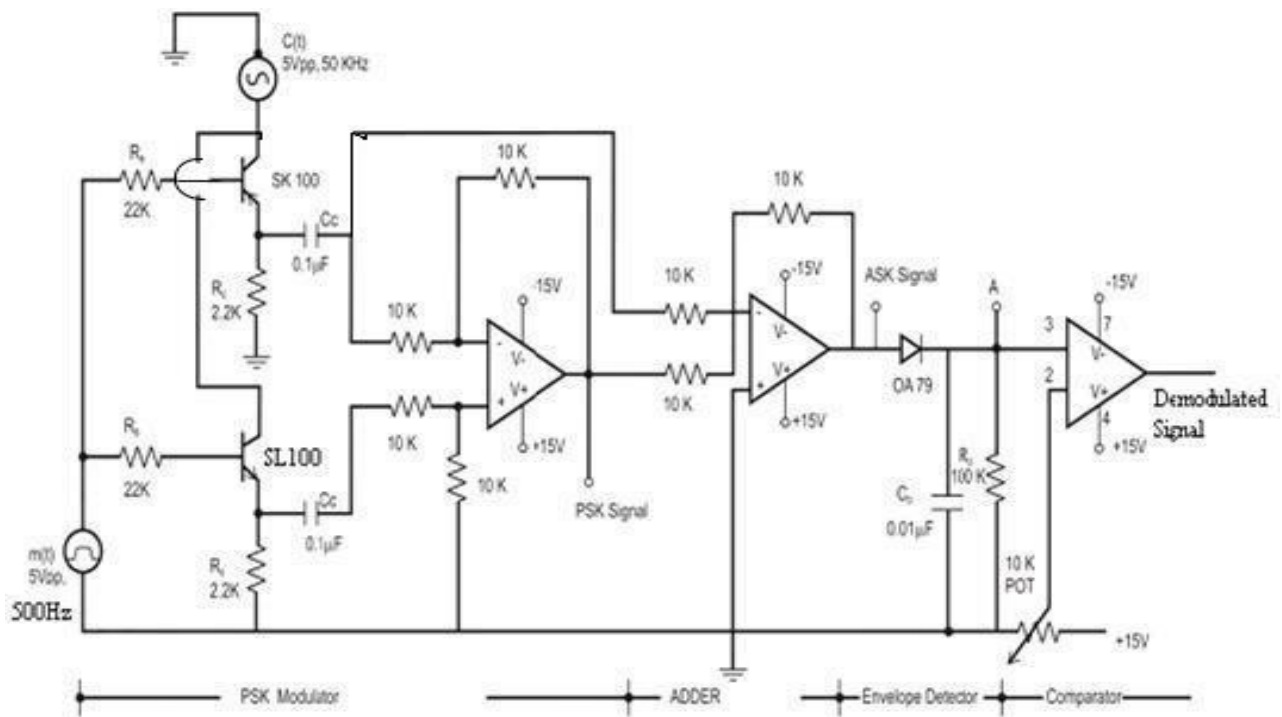
$$S_n(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi ft + \pi(1 - n)), n = 0,1$$

This yields two phases, 0 and π . In the specific form, binary data is often conveyed with the following signals:

$$S_0(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi ft + \pi), \text{ for } = 0$$

$$S_1(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi ft + \pi), \text{ for } = 1$$

where f is the frequency of the base band.



Design:

Let $V_c = 5$ volts peak-to-peak, $V_m = 10$ volts peak-to-peak, $f_m = 500$ Hz, $f_c = 50$ kHz.

Assume $h_{fe} = 30$, $V_{BEsat} = 0.7$ volts, $V_{CEsat} = 0.3$ volts, $I_B = 0.1$ mA, $I_C = 1$ mA Applying KVL to CE-Loop

From the circuit

Substituting the values in above equation we get,

Envelope Detector:

$$1/f_m > R_d C_d > 1/f_c, \quad \text{hence} \quad 2ms > R_d C_d > 20\mu s$$

$$\text{Let } R_d C_d = 50/f_c = 1 \text{ ms}$$

Assume $C_d = 0.01 \mu F$, then $R_d = 100 \text{ k}\Omega$

Procedure:**Generation of PSK:**

1. Rig up the circuit as shown in circuit diagram.
2. Apply the message signal $m(t) > 8\text{vp-p}, 500\text{hz}$ square wave
3. Apply the carrier signal $c(t) 5\text{vp-p}, 50\text{khz}$ sine wave
4. Observe ASK1 and ASK2 output at SL100 and SK100 emitter respectively.
5. Observe the PSK output at the end of adder stage using a CRO.

Demodulation of PSK:

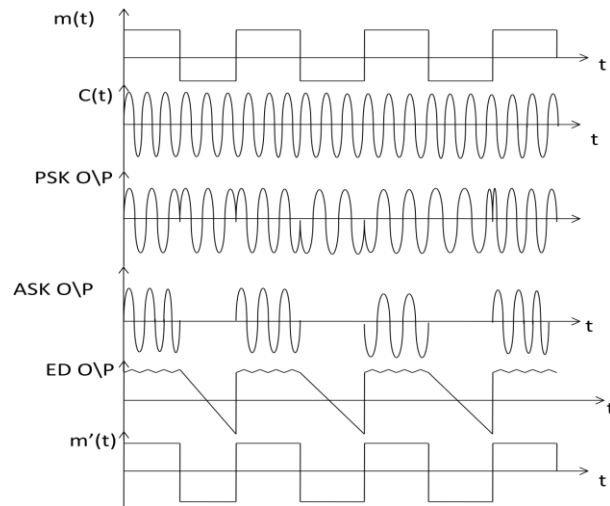
1. Rig up the circuit as shown in the figure.
2. Observe the output at the envelope detector (diode).
3. Observe the amplitude and waveform at the pin no 6 of op-amp which gives output corresponding to envelope detector stage.
4. Vary the comparator voltage (pin 2) 0-5v to obtain the clear square wave.
5. The demodulated signal $m^1(t)$ at the comparator (pin 6) matches with the message signal $m(t)$.
6. Calculate the bit error from $m^1(t)$ and $m(t)$.

Working: Here we are using two transistors one is of NPN and other PNP transistor.

During bit '1' NPN transistor is switched on and gives ask1 output. During bit '0' PNP transistor is switched on and NPN is turned off again yielding ask2. ASK1 and ASK2 are summed up by using summer circuit as shown in circuit producing PSK output.

In the demodulation, first carrier in one bit duration (during bit 0) is eliminated again by using adder circuit to obtain ASK. Finally an envelope Detector and comparator is used to obtain binary data as explained in ASK demodulator.

Expected BPSK modulation and de modulation waveforms:



Result:

Observations /Comments:

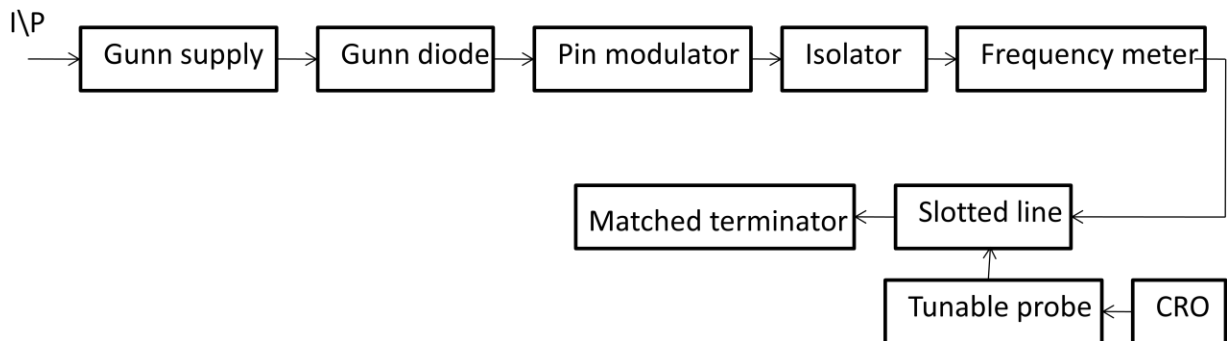
Expt No-5. MEASUREMENT OF FREQUENCY, WAVELENGTH, POWER USING MICROWAVE

Aim of the Experiment: Measurement of frequency, guide wavelength, power, VSWR and attenuation in microwave test bench.

Equipment's Required:

- Gunn oscillator,
- Gunn power supply,
- isolator,
- variable attenuator,
- CRO,
- signal generator,
- Connecting probes.

Block diagram:



Procedure:

1. Initially keep the GUNN voltage and pin bias in minimum position.
2. Select AM, switch on the GUNN supply.
3. Increase the GUNN supply voltage and keep it between 6 to 8V. so that GUNN diode is in the negative resistance region to work as an oscillator and produce microwave signal.

4. Apply pin bias maximum and observe the output on an oscilloscope. A square wave is expected.
5. Vary the micrometer provided at the GUNN oscillator unit to change the operating frequency and output power.

Observations and calculations:

Frequency (f_0) = _____ GHz.

Wavelength:

L1= _____ cm

L2= _____ cm

L3= _____ cm

L4= _____ cm

V_{max}= _____ mV

V_{min}= _____ mV

V_{max}= _____ mV

V_{min}= _____ mV

VSWR= _____

Result:

Expt No-6. MEASUREMENT OF DIRECTIVITY AND GAIN OF ANTENNAS

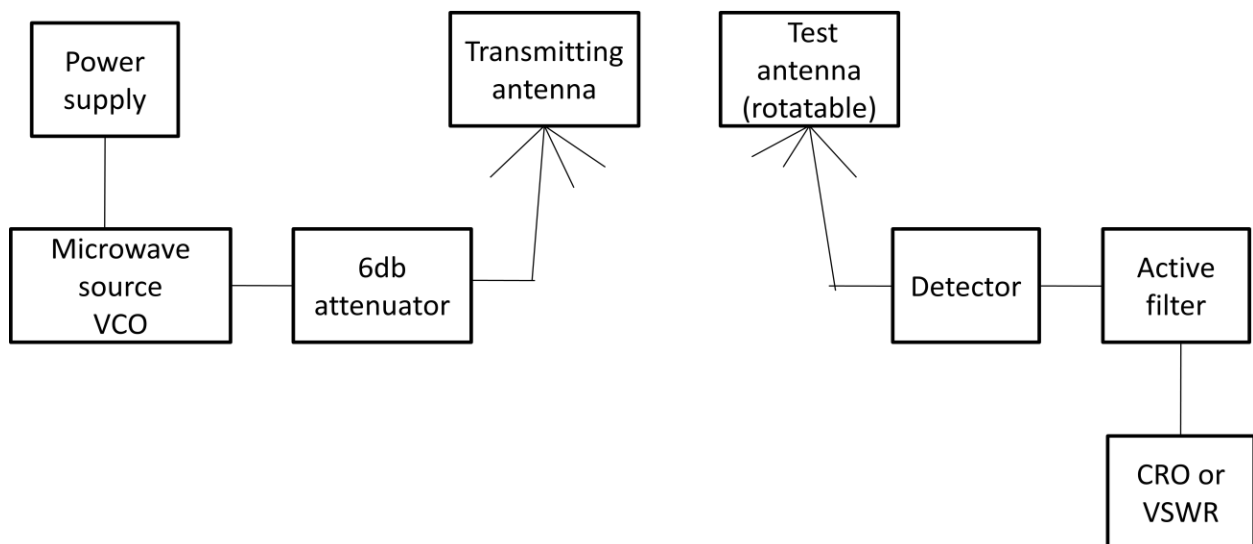
Aim of the Experiment: To measure the directivity and gain of antennas

- (a) Printed dipole (standard dipole)
- (b) Patch antenna
- (c) Yagi antenna

Equipment's required:

- Power supply,
- VCO,
- 50ohm transmission line
- directional coupler (branch line coupler and parallel line coupler),
- 50ohm terminations, and cables with SMA connectors,
- VSWR.

Block diagram:



Procedure:

Set up the system as shown in figure for standard dipole antenna.

- Keeping the voltage at minimum switch on the power supply
- Vary the power supply voltage and check the output for different VCO frequencies.
- Keeping at the resonant frequency, calculate and keep the minimum distance for field between the transmitting and receiving antenna using the formula

$$S \propto \frac{2d^2}{\lambda}$$

Where d is broader dimension of the antenna.

- Keeping the line of sight properly (0^0 at the turn table); tabulate the output (E_m).
- Rotate the turn table in clock wise and anticlockwise for different angle of deflection and tabulate the output for every angle (E_ϕ).
- Plot a graph: angle Vs output.
- Find the half power beam width from the points where the becomes half(3dB or 0.707V)
- Directivity of the antenna can be calculated using the formula

Where HPBW is the half power beam width in degrees. E_m and E_ϕ are the output signals measured at the receiving antenna for 0^0 and for different angle respectively.

- Gain of the antenna can be calculated using the formula

$$\text{Gain in dB} = 10 \log G$$

Where E_T and E_r are the signal strength measured using an oscilloscope at the transmitting end at the receiving end respectively, When the line of sight is proper. S is the actual distance between the antennas and is the wavelength found using the formula

$$= c/f$$

- Repeat the experiment for a patch antenna and yagi antenna.

Note: for micro strip antenna $\lambda = \frac{\lambda_0}{\sqrt{\epsilon_r}}$

Tabulation:

Angle	Output on CRO or VSWR	
	Output(R)	Output(L)
0^0		
5^0		

10^0		
15^0		
20^0		
25^0		
30^0		
35^0		

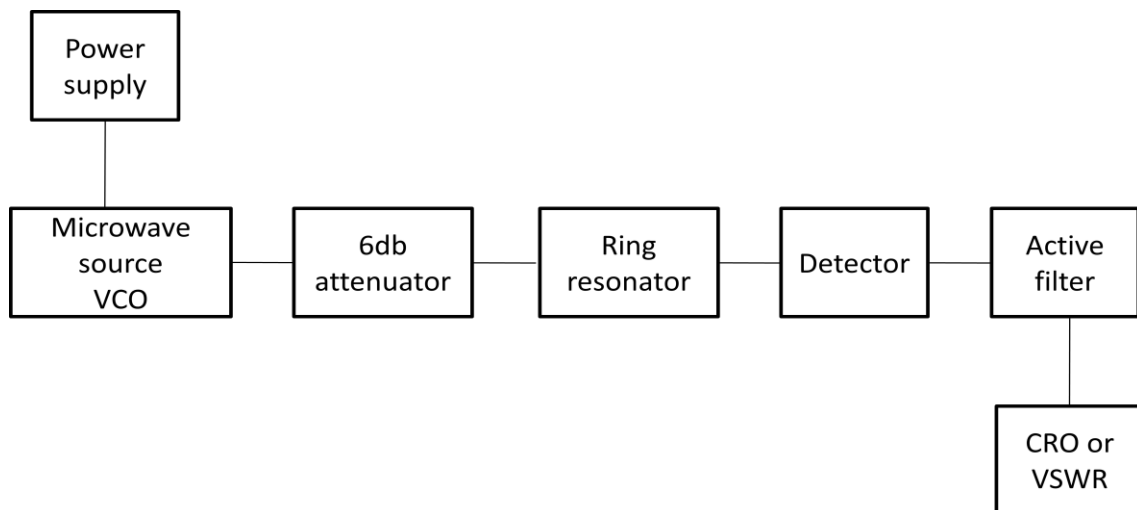
Results:

Expt No-7. DIRECTIONAL COUPLER & POWER DIVIDERS**Aim of the Experiment:**

- a) To measure the resonance characteristics of a micro strip ring resonator and determination of dielectric constant of the substrate.
- b) To measure the power division and isolation characteristics of a micro strip 3dB power Divider.
- c) To determine coupling and isolation characteristics of strip line directional coupler

Equipment's Required:

- Microwave source
- 6db attenuator
- 50Ω transmission line
- Ring resonator
- Detector
- Active filter
- VSWR or CRO

Block diagram:

Procedure:

a) I) Resonance Characteristics

- Set up the system as shown in the figure
- Keeping the voltage minimum switch on the power supply
- Insert a 50ohm transmission line and check for the output at the end of the system using CRO/VSWR/power meter.
- Vary the power supply voltage and check the output for different VCO frequency
- Replace 50ohm transmission line with ring resonator.
- Vary the supply voltage, tabulate VCO frequency Vs output.
- Plot a graph frequency Vs output and find the resonant frequency.

Tabulation:

SI No	Voltage	Frequency

II) Dielectric Constant

- Select a VCO frequency (say f_1) where there is a measurable output. Note down the magnitude/ power level of the output.
- Place the unknown dielectric material on top of the resonator ensure that there is no air gap between the dielectric and the resonator surface.
- Observe the change in the magnitude/ power level at the output.
- Now reduce supply voltage till maximum power level (before inserting the dielectric) is achieved. This is the new resonance condition due to the insertion of new dielectric material.

- Note down the VCO frequency(say f_2)
- Calculate the dielectric constant of the unknown material by using formula.

Calculation:

$$\lambda_1 = \frac{C}{f_1}$$

$$\lambda_2 = \frac{C}{f_2}$$

Where h = height of the unknown sample

W = width of the transmission line

Knowing the dielectric constant of the substrate used for the ring resonator the effective dielectric constant can be found by using Eq(3). Now the effective dielectric constant of the unknown material can be found using the relation

Where d_m = diameter of the ring resonator

ϵ_1 = effective dielectric constant of the known material

ϵ_2 = effective dielectric constant of unknown material.

a) Power Division & Isolation

- Set up the system as shown in the figure
- Keeping the voltage at minimum Switch on the power supply.
- Insert a 50ohm transmission line and check for the output at the end of the system using a CRO/ VSWR/ power meter.
- Vary the power supply voltage and the check the output for different VCO frequencies.
- Keep the VCO frequency constant note down the output. This value can be taken has the input to the power divider.
- Replace 50ohm transmission line with the power divider
- Tabulate the output at port2 and port3
- Calculate the insertion loss and coupling factor in each coupled arm.
- Calculate the isolation between port2 and port3 by feeding the input to port2 and measure output at port3 by terminating port1.
- Repeat the experiment for different VCO frequencies.

With VSWR meter

Isolation (dB) = $P_3 - P_2$

Power division at arm 3 (dB) = $P_1 - P_3$

Power division at arm 2 (dB) = $P_1 - P_2$

With CRO

Isolation between port 2 & 3 = $10 \log (V_2/V_3)$

Coupling factor at arm 3 = $10 \log (V_3/V_1)$

Coupling factor at arm 2 = $10 \log (V_2/V_1)$

b) Coupling & Isolation of directional coupler

- Set up the system as shown in figure
- Keeping the voltage at minimum, switch on the power supply.
- Insert 50ohm transmission line and check for the output at the end of the system using CRO/ VSWR/ power meter.
- Vary the power supply voltage and check the output for different VCO frequencies
- Not down the output for different VCO frequencies (this value can be taken as the input to branch line coupler)
- Replace 50ohm transmission line with branch line coupler.
- Check the output at port 2 (through output), 3 (coupled output), 4 (isolated output).
- Calculate the insertion loss, coupling factor and isolation using the formula given.
- Repeat the experiment for the parallel line coupler.

With VSWR meter:
 Insertion loss in (dB) = $P_1 - P_2$
 Coupling factor in (dB) = $P_1 - P_3$
 Isolation in (dB) = $P_3 - P_4$
 Directivity in (dB) = $P_3 - P_2$

With oscilloscope:
 Insertion loss in (dB) = $10 \log (V_1/V_2)$
 Coupling factor in (dB) = $10 \log (V_1/V_3)$
 Isolation in (dB) = $10 \log (V_1/V_4)$
 Directivity in (dB) = $10 \log (V_3/V_4)$

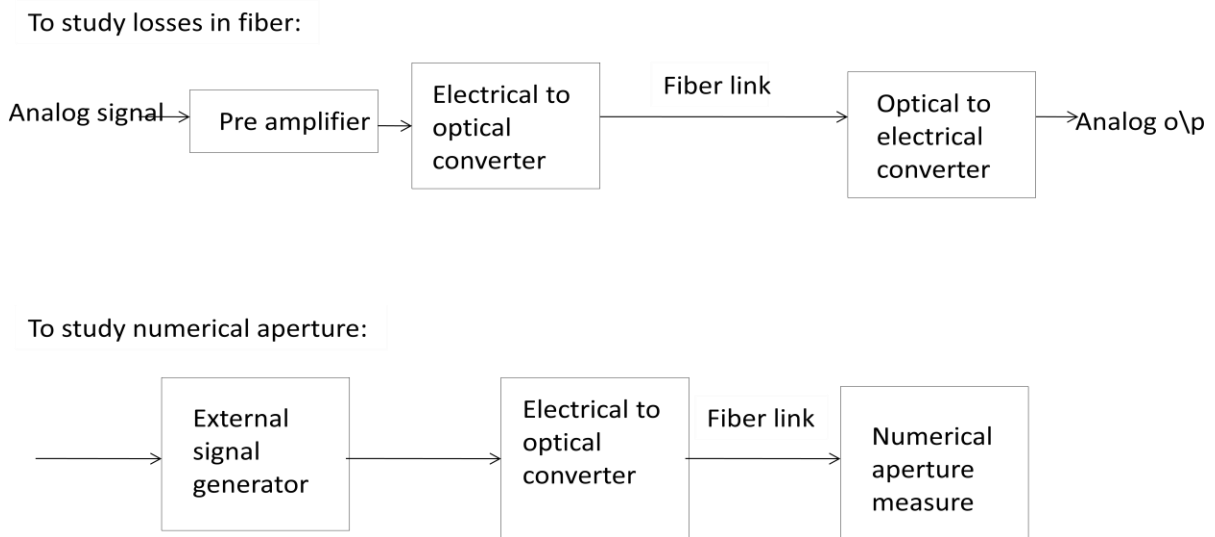
Expt No.-8 OPTICAL FIBER COMMUNICATION

Aim of the Experiment: To measure the losses (attenuation loss and bending loss) and the numerical aperture of an optical fiber cable.

Equipment’s required:

- Optical Fiber kit (OFT),
- optical fiber cable (1m, 0.5m),
- CRO.

Block Diagram:



Procedure:

1. Propagation loss\Attenuation loss:
2. Switch on the OFT kit.
3. Apply the sine wave (1V p-p, 1K Hz) at transmitting end.
4. Measure the output V_1 using CRO obtained by 0.5m optical cable at the receiving end.
5. Measure the output V_2 using CRO obtained by 1m optical cable at the receiving end.
6. Find the attenuation loss using the formula given by

$$V_1/V_2=e^{-\alpha (L_1 L_2)}$$

Where $L_1=0.5m$

$L_2=1m$

V_1 =output corresponding to 0.5m cable

V_2 =output corresponding to 1m cable

α =propagation loss/attenuation loss.

Bending loss:

1. Switch on the OFT kit.
2. Apply the sine wave (1V p-p, 1K Hz) at transmitting end.
3. Adjust the gain control knob to get the required output.
4. Make a loop (diameter more than 1cm) with the 1m cable as shown in the figure.
5. Increase the loop diameter and note down the corresponding value of diameter and output.
6. Observe the bending loss from the tabulated data.
7. Make sure that gain knob is not disturbed once set at starting.

Diameter(cm)	V_o (volts)

Numerical aperture:

1. The connections are made as per the user’s manual.
2. Switch on the OFT kit.
3. Connect the one end of the fiber to the transmitter and other end into the measurement jig provided. Adjust the fiber such that its tip is 10mm away from the paper.
4. Observe the shape of the light patch falling on the paper and measure the diameters twice in two perpendicular directions.
5. Calculate the average radius and determine the numerical aperture and critical angle

Width(W)	Length(L)	N.A	Critical angle(θ)

Result: Thus the losses are measured using optical and

Propagation loss=_____nepers/meter.

The numerical aperture is measured for different cases.

Observations/ Comments:

Expt No.-9 SIMULATE BINARAY POLAR SIGNAL

Aim of the Experiment: To Simulate NRZ, RZ, half-sinusoid and raised cosine pulses and generate eye diagram for binary polar signaling.

Equipment Required:

- a) Personal Computer
- b) MATLAB Software R2014a

Theory:

Line coding refers to the process of representing the bit stream (1's and 0's) in the form of voltage or current variations optimally tuned for the specific properties of the physical channel being used. The selection of a proper line code can help in so many ways: One possibility is to aid in clock recovery at the receiver. Some common types of line encoding in common-use nowadays are unipolar, polar, bipolar, Manchester and Duo-binary encoding.

These codes are explained here:

Unipolar (Unipolar NRZ and Unipolar RZ): Unipolar is the simplest line coding scheme possible. It has the advantage of being compatible with TTL logic. Unipolar coding uses a positive rectangular pulse $p(t)$ to represent binary 1, and the absence of a pulse (i.e., zero voltage) to represent a binary 0. Two possibilities for the pulse $p(t)$ exist: Non-Return-to-Zero (NRZ) rectangular pulse and Return-to-Zero (RZ) rectangular pulse. The difference between Unipolar NRZ and Unipolar RZ codes is that the rectangular pulse in NRZ stays at a positive value (e.g., +5V) for the full duration of the logic 1 bit, while the plus in RZ drops from +5V to 0V in the middle of the bit time.

A drawback of unipolar (RZ and NRZ) is that its average value is not zero, which means it creates a significant DC-component at the receiver (see the impulse at zero frequency in the corresponding power spectral density (PSD) of this line code.

Procedure:

1. Open Mat lab version R2014a
2. Open new file and enter the program and save it.
3. Add the path to the location of the file in the system.
4. Compile the program and check for any error and debug it.
5. Note down the output

Mat lab Code

```

clear all;
close all;
clc;
data=sign(randn(1,400)); % generate 400 random bits
Tau=64; % define the symbol period
for i=1:length(data)
    dataup((i-1)*64+1:i*64)=[data(i),zeros(1,63)]; end
dataup=upsample(data,Tau); % Generate impluse train
yrz=conv(dataup,prz(Tau)); % Return to zero polar signal
yrz=yrz(1:end-Tau+1);
ynrz=conv(dataup,pnrz(Tau)); % Non-return to zero polar
ynrz=ynrz(1:end-Tau+1);
ysine=conv(dataup,psine(Tau)); % half sinusoid polar
ysine=ysine(1:end-Tau+1);
Td=4; % truncating raised cosine to 4 periods
yrcos=conv(dataup,prcos(0.5,Td,Tau)); % rolloff factor=0.5
yrcos=yrcos(2*Td*Tau:end-2*Td*Tau+1); % generating RC pulse train
eye1=eyediagram(yrz,2*Tau,Tau,Tau/2);
title('RZ eye-diagram');
eye2=eyediagram(ynrz,2*Tau,Tau,Tau/2);
title('NRZ eye-diagram');
eye3=eyediagram(ysine,2*Tau,Tau,Tau/2);
title('Half-sine eye-diagram');
eye4=eyediagram(yrcos,2*Tau,Tau);
title('Raised-cosine eye-diagram');

```

Expt No.-10 SIMULATE PULSE CODE MODULATION & DEMODULATION

Aim of the Experiment: To Simulate the Pulse code modulation and demodulation system and display the waveforms.

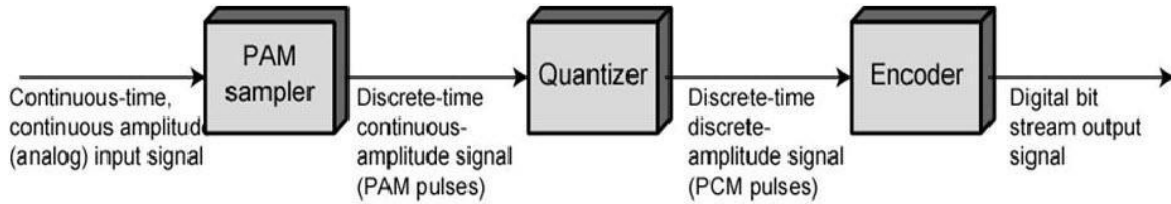
Equipment Required:

- a) Personal Computer
- b) MATLAB Software R2014a

Theory:

Pulse Code Modulation : Pulse code modulation is a method that is used to convert an analog signal into a digital signal, so that modified analog signal can be transmitted through the digital communication network. PCM is in binary form ,so there will be only two possible states high and low(0 and 1). We can also get back our analog signal by demodulation. The Pulse Code Modulation process is done in three steps Sampling, Quantization, and Coding. There are two specific types of pulse code modulations such as differential pulse code modulation(DPCM) and adaptive differential pulse code modulation(ADPCM)

Pulse Code Demodulation: Pulse Code Demodulation will be doing the same modulation process in reverse. Demodulation starts with decoding process, during transmission the PCM signal will effected by the noise interference. So, before the PCM signal sends into the PCM demodulator, we have to recover the signal into the original level for that we are using a comparator. The PCM signal is a series pulse wave signal, but for demodulation we need wave to be parallel. By using a serial to parallel converter the series pulse wave signal will be converted into a parallel digital signal. After that the signal will pass through n-bits decoder, it should be a Digital to Analog converter. Decoder recovers the original quantization values of the digital signal. This quantization value also includes a lot of high frequency harmonics with original audio signals. For avoiding unnecessary signals we utilize a low-pass filter at the final part.



Block Diagram of PCM

Procedure:

1. Open Matlab version R2014a
2. Open new file and enter the program and save it.
3. Add the path to the location of the file in the system.
4. Compile the program and check for any error and debug it.
5. Note down the output

Mat lab Code

```

clc;
close all; clear
all;
n=input('Enter n value for n-bit PCM system : ');
n1=input('Enter number of samples in a period : ');
L=2^n;

% % Signal Generation
% x=0:1/100:4*pi;
% y=8*sin(x); % Amplitude Of signal is 8v
% subplot(2,2,1);
% plot(x,y);grid on;
% Sampling Operation
x=0:2*pi/n1:4*pi; % n1 number of samples have to be selected s=8*sin(x);
subplot(3,1,1);
plot(s);
title('Analog Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(3,1,2);
stem(s);grid on; title('Sampled Sinal'); ylabel('Amplitude--->'); xlabel('Time--->');

```

```

% Quantization Process vmax=8;
vmin=-vmax;
del=(vmax-vmin)/L;
part=vmin:del:vmax;% level are between vmin and vmax with difference of del code=vmin-
(del/2):del:vmax+(del/2);          % Contain Quantized values
[ind,q]=quantiz(s,part,code);      % Quantization process
                                     % ind contain index number and q contain quantized values

l1=length(ind);
l2=length(q);

for i=1:l1
    if(ind(i)~=0)    % To make index as binary decimal so started from 0 to N ind(i)=ind(i)-1;
        end i=i+1;
    end
    for i=1:l2
        if(q(i)==vmin-(del/2))    % To make quantize value in between the levels
            q(i)=vmin+(del/2);
        end
    end
end
subplot(3,1,3);
stem(q);grid on;    % Display the Quantize values title('Quantized Signal');
ylabel('Amplitude--->');
xlabel('Time--->');

% Encoding Process
figure
code=de2bi(ind,'left-msb');        % Convert the decimal to binary
k=1;
for i=1:l1 for
    j=1:n
        coded(k)=code(i,j);        % convert code matrix to a coded row vector
        j=j+1;
        k=k+1;
    end i=i+1;
end
subplot(2,1,1); grid on;
stairs(coded);    % Display the encoded signal
axis([0 100-2 3]); title('Encoded Signal');
ylabel('Amplitude--->');
xlabel('Time--->');

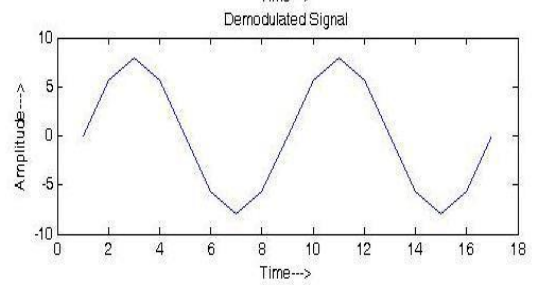
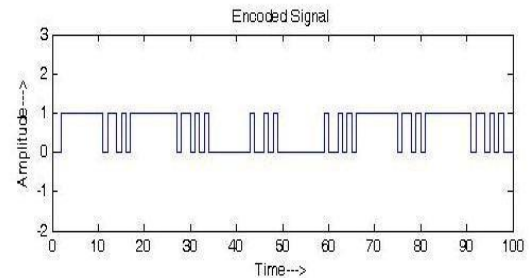
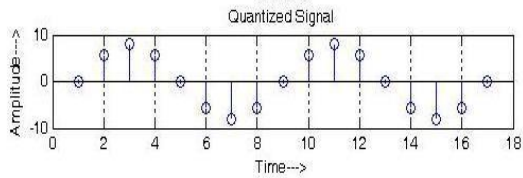
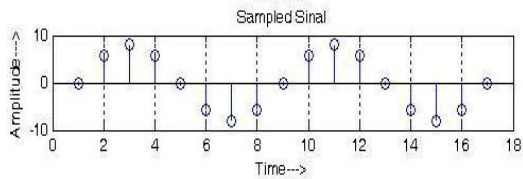
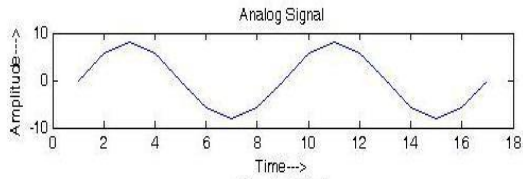
% Demodulation Of PCM signal

qunt=reshape(coded,n,length(coded)/n);
index=bi2de(qunt,'left-msb');      % Getback the index in decimal form
q=del*index+vmin+(del/2);          % getback Quantized values
subplot(2,1,2); grid on;
plot(q);    % Plot Demodulated signal
title('Demodulated Signal');
ylabel('Amplitude--->');
xlabel('Time--->');

```

Expected Output Waveform:

```
Enter n value for n-bit PCM system : 8
Enter number of samples in a period : 8
```



//

Expt No.-11 SIMULATE QPSK TRANSMITTER & RECIEVER

Aim of the Experiment: To Generate the Quadrature phase Shift Keying Modulation and Demodulation signals using MATLAB

Equipment Required:

1. Software required: MATLAB
2. Personal Computer

Algorithm:

Step1: The binary bit sequence is first taken into a variable.

Step2: The suitable carrier frequency is selected and two carriers one with zero phase and other with 90° phase are generated.

Step3: The input bit stream is given to the de-multiplexer and is decoded.

Step4: The de-multiplexer output is divided into two and one is multiplied with in phase component and the other with the out of phase component.

Step5: Both the multiplexer outputs are taken and are added together which gives us the QPSK output wave.

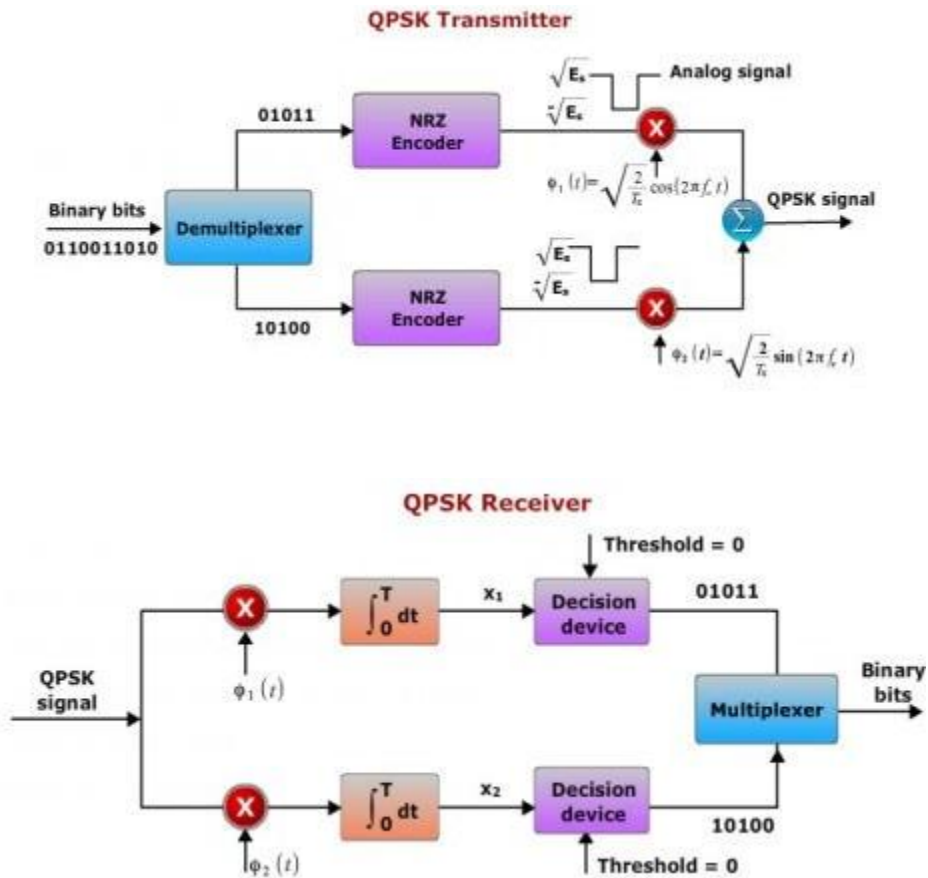
Step6: In the de-modulation process the input QPSK wave is multiplied with both the in phase and out phase components by an multiplier.

Step7: Both the outputs are passed through an low pass filter which acts as an integrator that adds up all the values of the signal.

Step8: The low pass filter outputs are given to the decision device which assigns the values to the wave.

Step9: The decision device output is given to the multiplexer circuit which gives the output as the input bit stream.

Block Diagrams:



Mat lab Code:

%Matlab program for QPSK wave

```

clc; clf;
clear all;
closeall;
b=input('enter binary data:');
fc=4000;
t=linspace(0,1/4000,50);
ec2=cos(2*pi*fc*t);
ec1=sin(2*pi*fc*t);
qpskout=[];bin=[];car1=[];car2=[];be=[];bo=[];bal1=[];bal2=[]; for
i=1:length(b);

```

```

bin=[bin,b(i)*ones(1,50)];
car1=[car1,ec1];
car2=[car2,ec2]; if
mod(i,2)==0 if
b(i)==0
be=[be,-ones(1,100)]; else
be=[be,ones(1,100)]; end;
bal1=[bal1,be(i*50-1)*ec1,be(i*50-1)*ec1]; else
if b(i)==1
bo=[bo,ones(1,100)];
else bo=[bo,-ones(1,100)]; end;
bal2=[bal2,bo(i*50-1)*ec2,bo(i*50-1)*ec2];
end;
end;
for i=1:2:length(b)
if b(i)==0 && b(i+1)==0
qpskout=[qpskout,-ec1-ec2,-ec1-ec2]; elseif
b(i)==0 && b(i+1)==1
qpskout=[qpskout,-ec1+ec2,-ec1+ec2]; elseif
b(i)==1 && b(i+1)==0
qpskout=[qpskout,ec1-ec2,ec1-ec2]; else
qpskout=[qpskout,ec1+ec2,ec1+ec2]; end;
end;
%ploting the graph
subplot(4,1,1);
plot(bin,'linewidth',3); title('binary
data'); xlabel('time');
ylabel('amplitude');
subplot(4,1,2);
plot(be,'r','linewidth',3);
title('even data'); xlabel('time');
ylabel('amplitude');
subplot(4,1,3);
plot(bo,'g','linewidth',3); title('odd
data');

```

```

xlabel('time');
ylabel('amplitude');
subplot(4,1,4);
plot(car1,'linewidth',3);
title('carrier 1');
xlabel('time');
ylabel('amplitude');
figure;
subplot(4,1,1);
plot(car2,'linewidth',3);
title('carrier2');
xlabel('time');
ylabel('amplitude');
subplot(4,1,2);
plot(bal1,'r','linewidth',3);
title('bal mod 1 data');
xlabel('time');
ylabel('amplitude');
subplot(4,1,3);
plot(bal2,'g','linewidth',3);
title('bal mod 2 data');
xlabel('time');
ylabel('amplitude');
subplot(4,1,4);
plot(qpskout,'linewidth',3);
title('modulated');
xlabel('time');
ylabel('amplitude');

                                %demodulation of qpsk
demod1=[ ];demod2=[ ];demod=[ ];synd1=[ ];synd2=[ ]; for
i=1:length(qpskout);

synd1=[synd1,car1(i)*qpskout(i)];
synd2=[synd2,car2(i)*qpskout(i)];
end;

for i=1:100:length(qpskout);

```

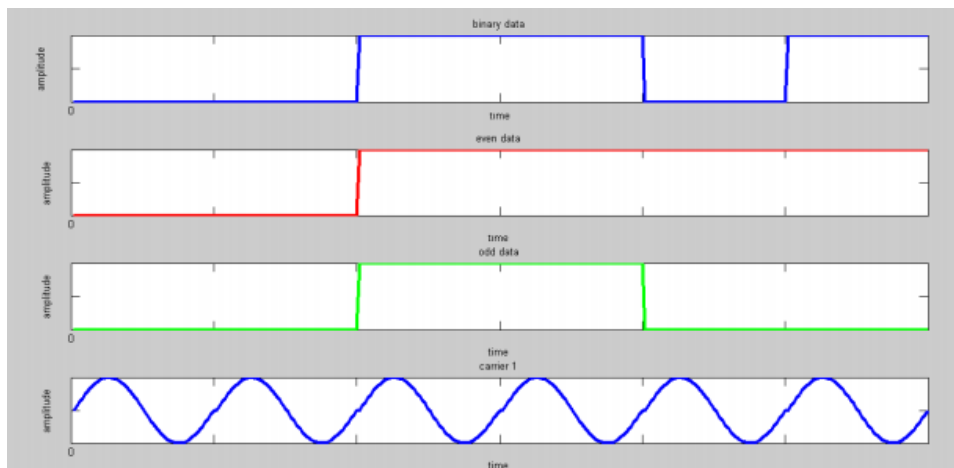
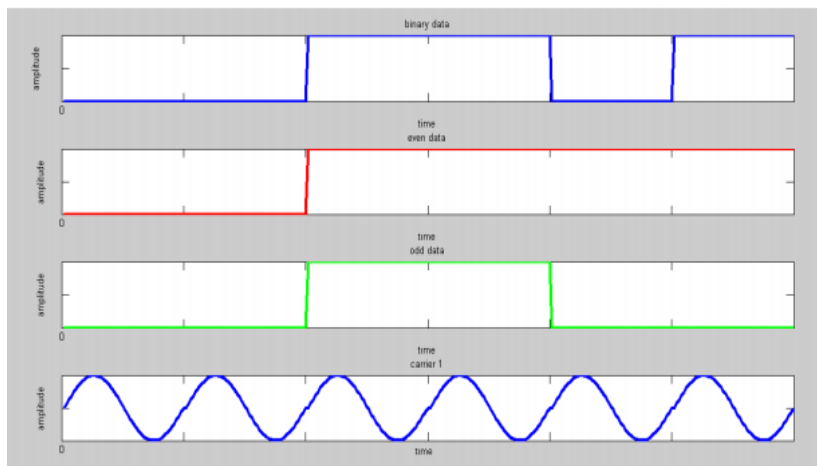
```
sum1=0;sum2=0;
for i=i:i+99
sum1=sum1+synd1(i);
sum2=sum2+synd2(i); end;
if sum1 > 0
demod1=[demod1,ones(1,50)]; else
demod1=[demod1,zeros(1,50)]; end;
if sum2 > 0
demod2=[demod2,ones(1,50)]; else
demod2=[demod2,zeros(1,50)]; end;
end; demod1
demod2
for i=1:50:length(demod1);
demod=[demod,demod1(i:i+49),demod2(i:i+49)]; end;

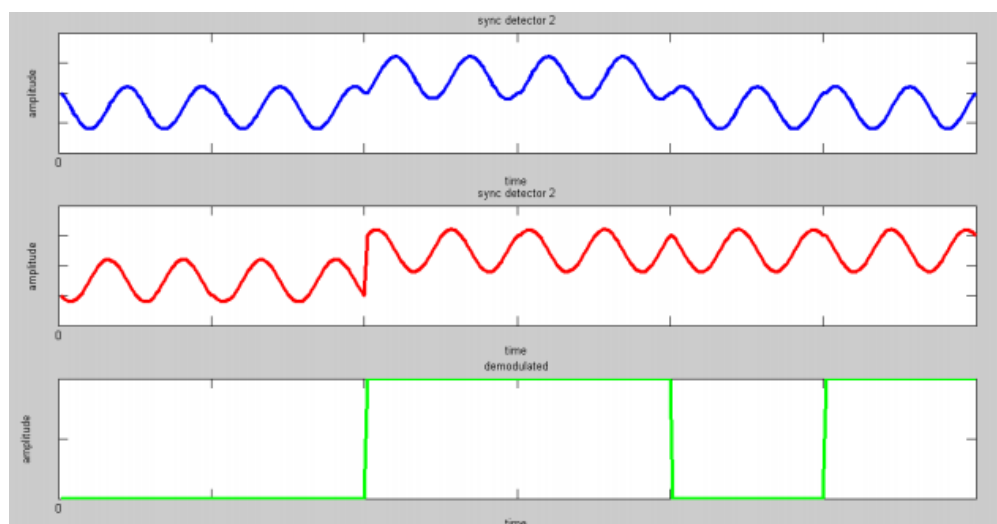
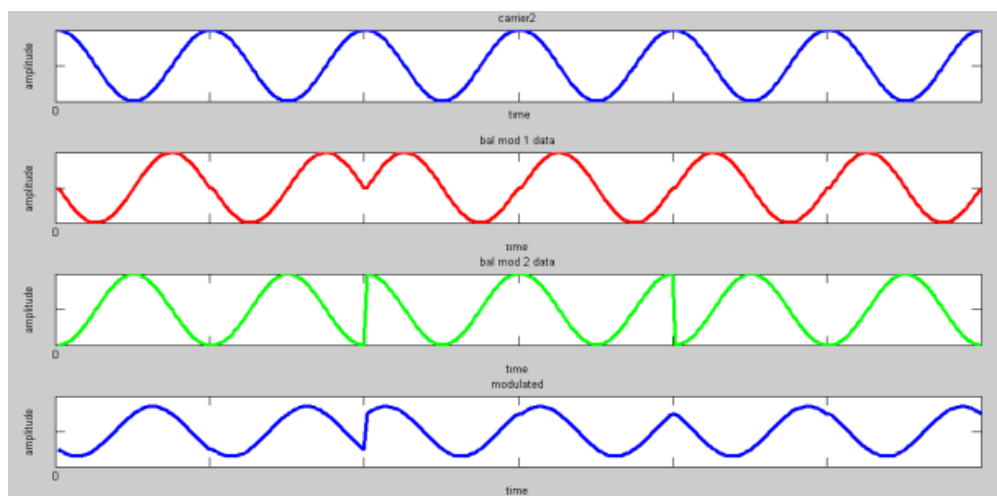
                                %ploting the graph

figure;
subplot(3,1,1)
plot(synd1,'linewidth',3);
title('sync detector 2');
xlabel('time');
ylabel('amplitude');
subplot(3,1,2)
plot(synd2,'r','linewidth',3);
title('sync detector 2');
xlabel('time');
ylabel('amplitude');

subplot(3,1,3)
plot(demod,'g','linewidth',3);
title('demodulated');
xlabel('time');
ylabel('amplitude');
```

Expected Output Wave Form





Beyond Syllabus:

QPSK MODULATION AND DEMODULATION

Introduction

The purpose of analog modulation is to impress an information waveform onto a carrier for transmission. The purpose of convert an information-bearing discrete waveform (perhaps impressed on a carrier). *Key concerns* - bandwidth efficiency and implementation These are affected by: baseband Phase transition characteristics

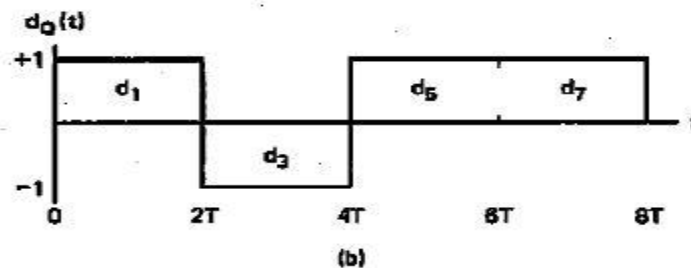
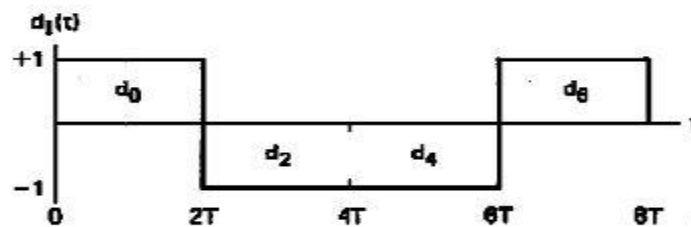
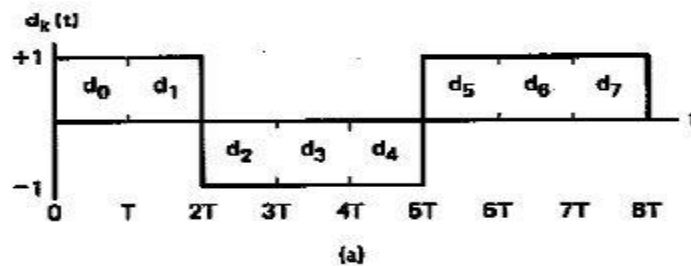
Digital modulation is the process by which digital symbols are transformed into waveforms that are compatible with the characteristics of the channel. Modulation can be used to minimize the effects of interference. Modulation can also be used to place a signal in a frequency band where design requ filtering and amplification, can be easily met. This is the case when radio (RF) signals are converted to an intermediate frequency (IF) in a receiver.

Theory

If we define four signals, each with a phase shift differing by 90 quadrature phase shift keying (QPSK). The input binary bit stream {d₁/T bits/sec and is separated into two data streams d even bits respectively.

$d_I(t) = d_0, d_2, d_4, \dots$

$d_Q(t) = d_1, d_3, d_5, \dots$



A convenient orthogonal realisation of a QPSK waveform, $s(t)$ is achieved by amplitude modulating the inphase and quadrature data streams onto the square wave

The pulse stream $d_I(t)$ modulates the equivalent to shifting the phase of the function by 0 or produces a BPSK waveform. Similarly the pulse stream $d_Q(t)$ shifted carrier function, yielding a BPSK waveform orthogonal to the summation of these two orthogonal waveforms is the QPSK waveform.

The values of $q(t) = 0, -(\pi/2),$ and π . Each of the four possible phases of carriers represents two bits of data. Thus there are two bits per symbol. Since the symbol rate for QPSK is half the bit rate, twice as much data can be carried in the same amount of channel bandwidth as compared to BPSK. This is possible because the two signals I and Q are orthogonal to each other and can be transmitted without interfering with each other.

In QPSK the carrier phase can change only once every $2T$ secs. If from one T interval to the next one, neither bit stream changes sign, the carrier phase remains unchanged. If one component occurs. However if both components change sign then a phase shift of π occurs. Figure below shows modulated wave

Constellation Diagram

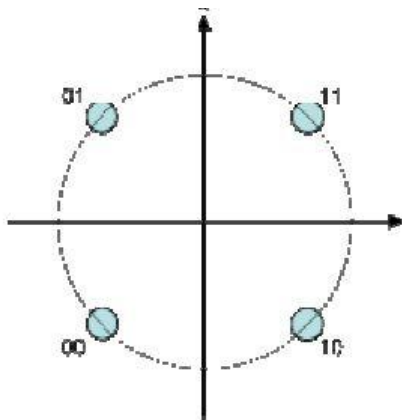


Figure shows constellation diagram for QPSK with symbol only differs by one bit. Sometimes known as quaternary or quadriphase PSK or 4 four points on the constellation diagram, equispaced at phases, QPSK can encode two bits per symbol, shown in the diagram with coding to minimize the BER be used either to double the data rate compared to a BPSK system while maintaining the bandwidth of the signal or to maintain the data

bandwidth needed. Although QPSK can be viewed as a quaternary modulation, it is easier to see it as two independently modulated quadrature carriers. With this interpretation, the even (or odd) bits are used to

modulate the in the odd (or even) bits are used to modulate the quadrature carrier. BPSK is used on both carriers and they can be independently demodulated.

As a result, the probability of bit-error for QPSK is the same as for BPSK:

$$P_b = Q \left(\sqrt{\frac{2E_b}{N_0}} \right).$$

However, with two bits per symbol, the symbol error rate is increased:

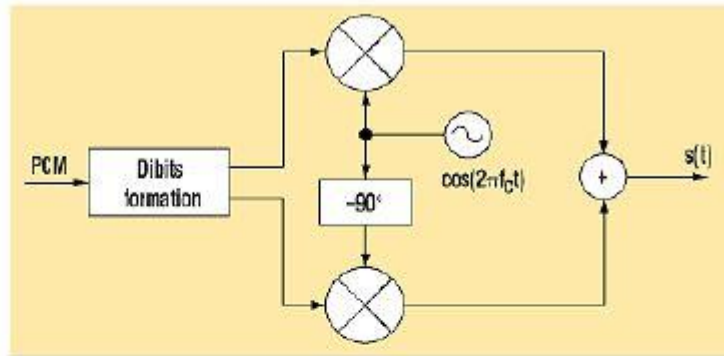
$$\begin{aligned} P_s &= 1 - (1 - P_b)^2 \\ &= 2Q \left(\sqrt{\frac{E_s}{N_0}} \right) - Q^2 \left(\sqrt{\frac{E_s}{N_0}} \right). \end{aligned}$$

If the *signal-to-noise ratio* is high (as is necessary for practical QPSK systems) the probability of symbol error may be approximated:

$$P_s \approx 2Q \left(\sqrt{\frac{E_s}{N_0}} \right)$$

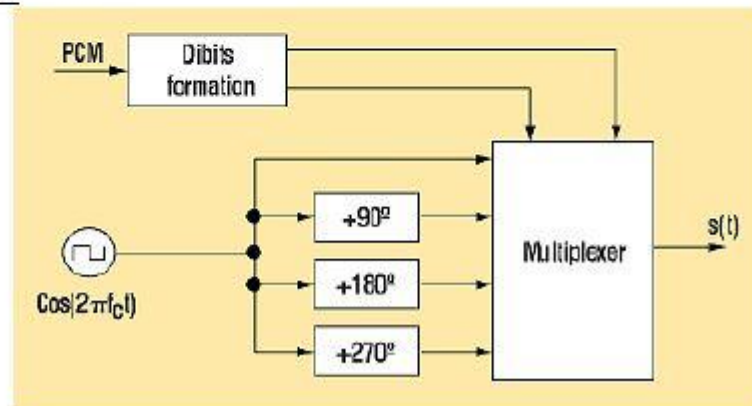
As with BPSK, there are phase ambiguity problems at the receiver and differentially encoded QPSK is more normally used in practice.

Block Diagram of QPSK Systems.



1. The classical method of implementing $s(t)$ for a QPSK modulator is shown by this block diagram.

Implementation:

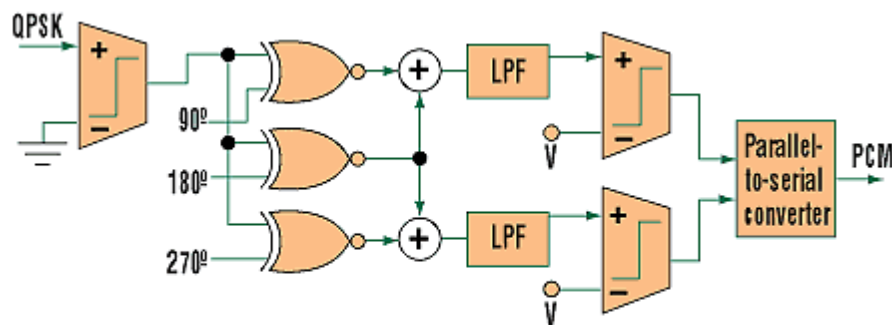


2. This technique for generating $s(t)$ eliminates the need for mixers and requires no adjustments.

QPSK Modulation

In QPSK modulation, the bit stream of the signal to be transmitted (PCM signal) is previously sampled and decomposed into two binary waves (dibits). Each pair of dibits is associated with one of four possible phases of the QPSK modulation. The modulated signal is: $s(t) = I_k c(t) + Q_k c(t-p/2)$ Where $c(t)$ is Sine wave carrier where I and Q are the in-phase and in-quadrature components of the modulated signal. The signal's values (1 or -1) determine four points (or symbols) in the QPSK signal space (or constellation). I and Q values are assigned according to associated dibits combinations. The classical circuitry to implement $s(t)$ is based on the block diagram shown in Figure If I and Q take their values from the set $[1,-1]$, the constellation shows four points at phases 45° , 135° , 225° , and 315° . In this case, the QPSK also is called 4-QAM. The realization of this circuit consists of rotating the angles of the QPSK constellation 135° , by increasing the set of possible values of I and Q components to $[-1,0,1]$. In this case, the table shows the modified phase distribution.

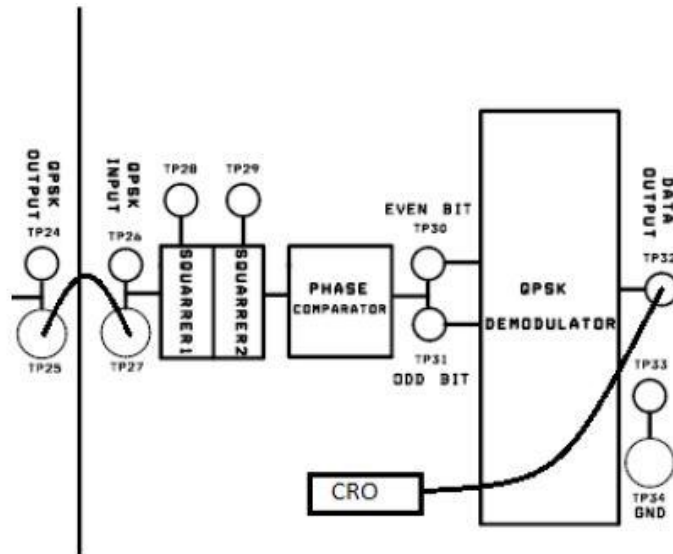
A demodulator that's suitable for the aforementioned QPSK modulator generates four modified carrier phases (0° , 90° , 180° , and 270°) obtained in the modulator by rotating the angles of the QPSK constellation. Then, a simple XNOR operation between each phase and the modulated signal will produce a logic "1" only if both coincide. If the received phase is 0° (related to dibit 00), all XNORs produce a logic "0". If the phase is 90° (dibit 10), only the upper branch gives a "1". This situation is inverted for 270° (dibit 01). In this case, the logic "1" is produced at the lower branch. But if the phase is 180° (dibit 11), a logic "1" is added to both branches. The result is the demodulated PCM signal.



2. An alternative technique to Figure 1 uses phase comparison to perform QPSK demodulation.

Connection Procedure:

1. Switch ON QPSK Trainer kit (DCT-006).
2. Observe the 8 bit Serial data out at (TP1 and TP2).
3. Connect the Serial Output Data (TP2) to Serial Data input (TP5).
4. Observe the carrier SIN0, SIN90, SIN180 and SIN270 in TP15, TP13, TP17 and TP11 respectively.
5. Connect the SIN0 (TP15) to SIN0 of QPSK Modulator (TP18).
6. Connect the SIN90 (TP15) to SIN90 of QPSK Modulator (TP19).
7. Connect the SIN180 (TP15) to SIN180 of QPSK Modulator (TP20).
8. Connect the SIN270 (TP15) to SIN270 of QPSK Modulator (TP21).
9. Observe the Even and odd bits at the TP7 and TP9 respectively.
10. Connect the Even bit (TP7) to the EVEN (TP22) of QPSK Modulator.
11. Connect the Odd bit (TP9) to the Odd (TP23) of QPSK Modulator.
12. Observe and Sketch the QPSK Modulated output at test point (TP25) using CRO.

Connection Diagram of QPSK Modulation:**Connection Procedure:**

1. Connect the QPSK Modulated output (TP25) to the QPSK input (TP27).
2. Observe and sketch the SQUARRER-1 output at TP28.
3. Observe and sketch the SQUARRER-2 output at TP29.
4. Phase comparison using CD4033 phase comparator to get EVEN and Odd Bit in TP30 and TP31 respectively.
5. To Combining the EVEN and Odd bits to get original Data bits at the TP32

VIVA QUESTIONS**VIVA QUESTIONS FOR ADVANCED COMMUNICATION LAB**

1. State different types of Digital modulation techniques?
2. What is shift keying?
3. What is a binary modulation technique?
4. Define ASK?
5. Define FSK?
6. Define PSK?
7. Define QPSK and DPSK?
8. Why QPSK is called quadrature shift keying?
9. Define TDM?

10. Define FDM?
11. State coherent detection?
12. What is standing wave?
13. Define Modes?
14. Define VSWR?
15. State application of Klystron?