RNS INSTITUTE OF TECHNOLOGY

**Dr. VISHNUVARDHAN ROAD, CHANNASANDRA,
BENGALURU - 560 098**

Department of Electronics and Communication Engineering

VI Semester

Communication Lab Manual

18ECL67

VISION of the College

Building RNSIT into a World - Class Institution

MISSION of the College

To impart high quality education in Engineering, Technology and Management with a difference, enabling students to excel in their career by

1. Attracting Quality Students and preparing them with a strong foundation in fundamentals so as to achieve distinctions in various walks of life leading to outstanding contributions.
2. Imparting value based, need based, and choice based and skill based professional education to the aspiring youth and carving them into disciplined, World class Professionals with social responsibility.
3. Promoting excellence in Teaching, Research and Consultancy that galvanizes academic consciousness among Faculty and Students.
4. Exposing Students to emerging frontiers of knowledge in various domains and make them suitable for Industry, Entrepreneurship, Higher studies, and Research & Development.
5. Providing freedom of action and choice for all the Stake holders with better visibility.

Vision of the department

Conquering technical frontiers in the field of Electronics and Communication

Mission of the department:

1. To achieve and foster excellence in core Electronics and Communication engineering with focus on the hardware, simulation and design.
2. To pursue Research, development and consultancy to achieve self-sustenance.
3. To create benchmark standards in electronics and communication engineering by active involvement of all stakeholders.

PROGRAM OUTCOMES (POs)

Engineering Graduates will be able to:

1. **PO1: Engineering knowledge:** Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization for the solution of complex engineering problems
2. **PO2: Problem analysis:** Identify, formulate, research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences and engineering sciences.
3. **PO3: Design/development of solutions:** Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for public health and safety, and cultural, societal, and environmental considerations.
4. **PO4: Conduct investigations of complex problems:** Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to

provide valid conclusions.

5. **PO5: Modern tool usage:** Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools, including prediction and modeling to complex engineering activities, with an understanding of the limitations.
6. **PO6: The engineer and society:** Apply reasoning informed by the contextual knowledge to assess Societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
7. **PO7: Environment and sustainability:** Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.
8. **PO8: Ethics:** Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
9. **PO9: Individual and team work:** Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.
10. **PO10: Communication:** Communicate effectively on complex engineering activities with the engineering community and with the society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.
11. **PO11: Project management and finance:** Demonstrate one's own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.
12. **PO12: Life-long learning:** Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

PROGRAM SPECIFIC OUTCOMES (PSOs)

The graduates of the department will be able to:

PSO1: Apply fundamental knowledge of Electronics, Communications, Signal processing, VLSI, Embedded and Control systems etc., in the analysis, design, and development of various types of real-time integrated electronic systems and to synthesize and interpret the experimental data leadin g to valid conclusions.

PSO2: Demonstrate competence in using Modern hardware languages and IT tools for the design and analysis of complex electronic systems as er industry standards along with analytical and managerial skills to arrive at appropriate solutions, either independently or in team.

Course objectives: This course will enable students to:

- Design and demonstrate the digital modulation techniques
- Demonstrate and measure the wave propagation in microstrip antennas
- Characteristics of microstrip devices and measurement of its parameters.
- Model an optical communication system and study its characteristics.
- Simulate the digital communication concepts and compute and display various parameters along with plots/figures.

Course Outcomes:

CO1	Determine the characteristics and response of microwave devices
CO2	Determine the characteristics of microstrip antennas and devices and compute the parameters associated with it
CO3	Understand the various optical fibre communication losses
CO4	Simulate the digital modulation schemes with the display of waveforms and computation of performance parameters
CO5	Design and test the digital modulation circuits/systems and display the waveforms

6 SEM	**SUB CODE: 18ECL67 SUB NAME: Communication Lab**														
		PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12	**PSO1**	**PSO2**
CO1		1		1						2	1		1	3	
CO2		1		1						2	1		1	3	
CO3		1		1						2	1		1	3	
CO4		1		1		3				2	1		1	3	3
CO5		1		1						2	1		1		
Final_CO_one_row		1		1		3				2	1		1	**3**	**3**

COMMUNICATION LAB
Subject Code: 18ECL67

I-CYCLE

1. FSK and PSK generation and detection
2. Time Division Multiplexing and Demultiplexing of two bandlimited signals.
3. Pulse sampling, Flat top sampling and reconstruction.

II-CYCLE

1. Amplitude Modulation and Demodulation:
   I. Standard AM,
   II. DSBSC (LM741 and LF398 ICs can be used).
2. Frequency modulation and demodulation (IC 8038/2206 can be used)
3. Obtain the Radiation Pattern and Measurement of directivity and gain of microstrip dipole and Yagi antennas.
4. Measurement of frequency, guide wavelength, power, VSWR and attenuation in microwave test bench
5. Determination of
   1. Coupling and isolation characteristics of microstrip directional coupler.
   2. Resonance characteristics of microstrip ring resonator and computation of dielectric constant of the substrate.
   3. Power division and isolation of microstrip power divider.

III-CYCLE

1. Simulate NRZ, RZ, half-sinusoid and raised cosine pulses and generate eye diagram for binary polar signaling.
2. Pulse code modulation and demodulation system.
3. Computations of the Probability of bit error for coherent binary ASK, FSK and PSK for an AWGN Channel and Compare them with their Performance curves.
4. 4. Digital Modulation Schemes i) DPSK Transmitter and receiver, ii) QPSK Transmitter and Receiver

Courses Related to Laboratory

- Digital Communication
- Antennas and Propagation
- Microwaves and Radar
- Optical Fiber Communication

Lab Equipment

1. Oscilloscopes :
    - CROs- Two Channel
    - DSOs- Two Channel and Four Channel
2. Signal Generators
3. Dual Regulated Variable Power Supplies
4. TDM Trainer Kit
5. Digital Communication Trainer Kits: ASK, FSK, PSK, DPSK, and QPSK Kits
6. VSWR Meter
7. Microwave signal generators
8. Simple Dipole and Folded Dipole Antenna trainer kit
9. Printed Dipole, Microstrip Patch antenna and Yagi antenna (printed) trainer kit
10. Microstrip Directional Coupler, Ring Resonator and Power Divider
11. OFC Trainer kits
12. Microwave Test Bench
13. Component Tester
14. Digital Multimeters
15. Analog and Digital ammeters and voltmeters

Experiment1: Generation and detection of DSBSC and standard Amplitude modulation (DSBFC) using IC398 and LM741

AIM: To generate DSBSC and AM signal using IC LF398 modulation for Carier Frequency = 10KHz and message frequency = 1KHz. Also plot the variations of modulating signal amplitude versus modulation index. Components required

SI.	Components	Range
1	IC 398 and 741	
3	Resistors	47KΩ (1), 10 KΩ(4)
4	Capacitors	0.01µF(1),0.001 µF(1)

Theory: Amplitude Modulation is a technique in which the amplitude of the carrier is varied in accordance with the instantaneous value of the message.

Transformer-less DSB-SC generation:

We now describe an alternate method of generating the DSB-SC. Consider the block diagram of figure 2, which has: (i) the analogue inverter, (ii) the two-channel switch and (iii) the band pass filter. The first block, the analogue inverter takes an input signal and provides a signal with 1800 phase shift at its output. The inverter can be realised using either a transformer or using an operational amplifier as an inverter. The second block; the two-channel analogue switch, has two input signals (‘A’ and ‘B’) and a ‘select input’. The output of the switch at any instant is either ‘A’ or ‘B’, depending on the amplitude of the ‘select input’ being ‘high’ or ‘low’. The two-channel switch can be realized using two transistors (n-p-n and p-n-p), or using standard two-channel switch available as linear integrated circuit. The third block a BPF, tuned to the carrier frequency, and can be either an active or a passive filter, and is the only block similar to that in figure 1.Since transformer_less circuit are preferred, this combination is an alternative method of generating the DSB-SC waveform

The demodulation circuit is a simple envelope detector. The output of this detector is the envelope of its input. As the envelope of the AM is the modulating signal m(t), the demodulation can be carried out by using this circuit.

Modulation is defined as the process by which some characteristics of a carrier signal is varied in accordance with a modulating signal. The base band signal is referred to as the modulating signal and the output of the modulation process is called as the modulation signal.

Amplitude modulation is defined as the process in which is the amplitude of the carrier wave is varied about a means values linearly with the base band signal. The envelope of the modulating wave has the same shape as the base band signal provided the following two requirements are satisfied The carrier frequency fc must be much greater than the highest frequency components fm of the message signal m (t) i.e. fc>>fm

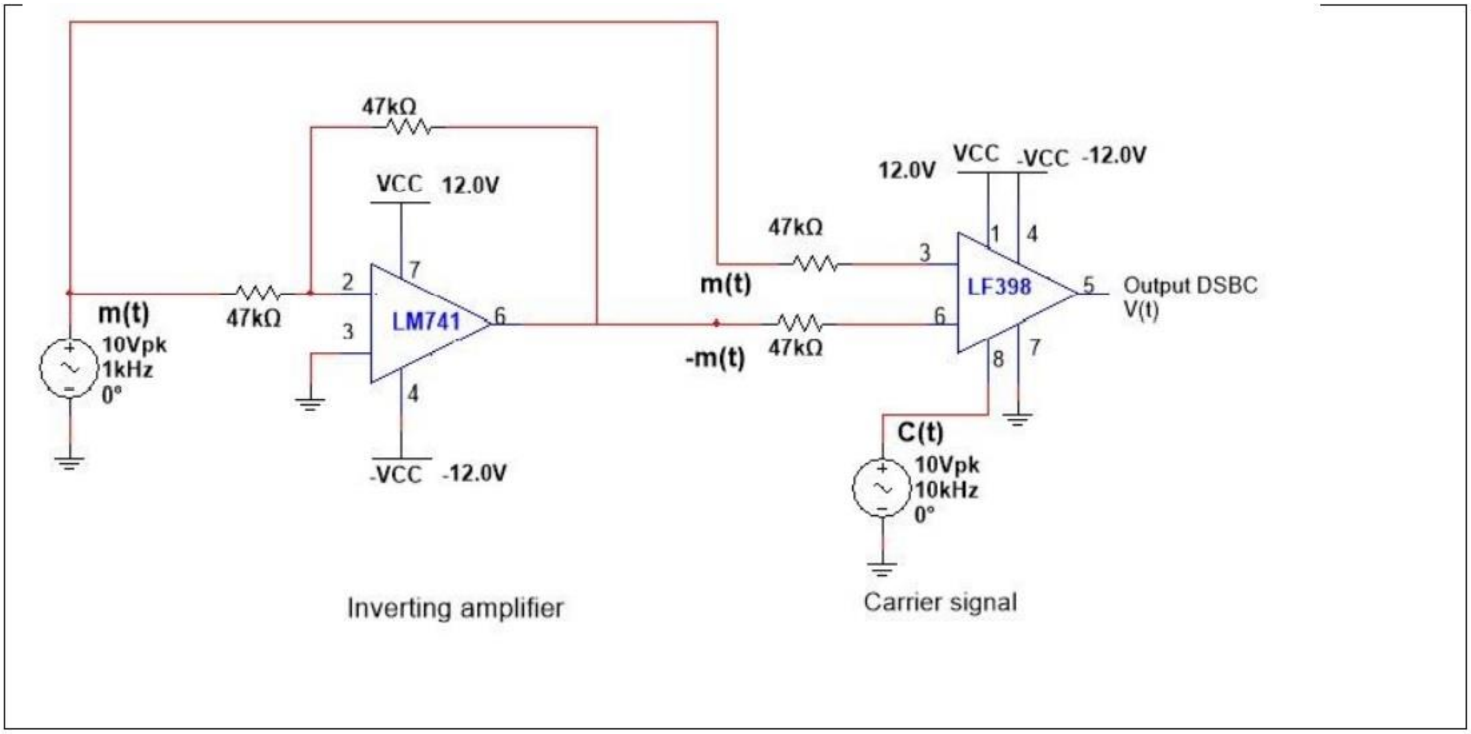
The modulation index must be less than unity. if the modulation index is greater than unity, the carrier wave becomes over modulated.

Amplitude Modulation

For the experimental verification of the alternate method of DSB-SC generation, we need an analogue inverter and an analogue switch. The analogue inverter and analogue switch can be realized using varied number of ICs, or even transistor_resistor combination. The individual circuits are given in figure. These blocks connected as in figure 2, produces the DSB_SC signal at the output of the BPF tuned to carrier frequency. In both cases, the square wave carrier is ≥ 10. KHz, while the sinusoidal message is of 1 KHz

Circuit diagram

1. Modulator



Design

Gain Av=Rf/R1

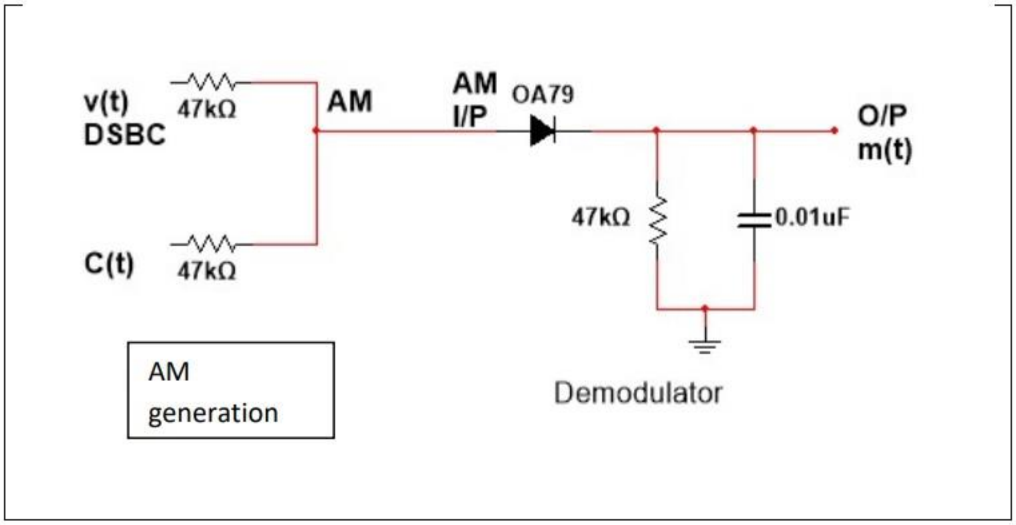
Gain=1

Assume

Rf =47KΩ

Therefore R1=47KΩ

2. Demodulator



Design

1/fc << RC<< 1/fm

1/10K << RC << 1/1K

100µS << RC<<1ms

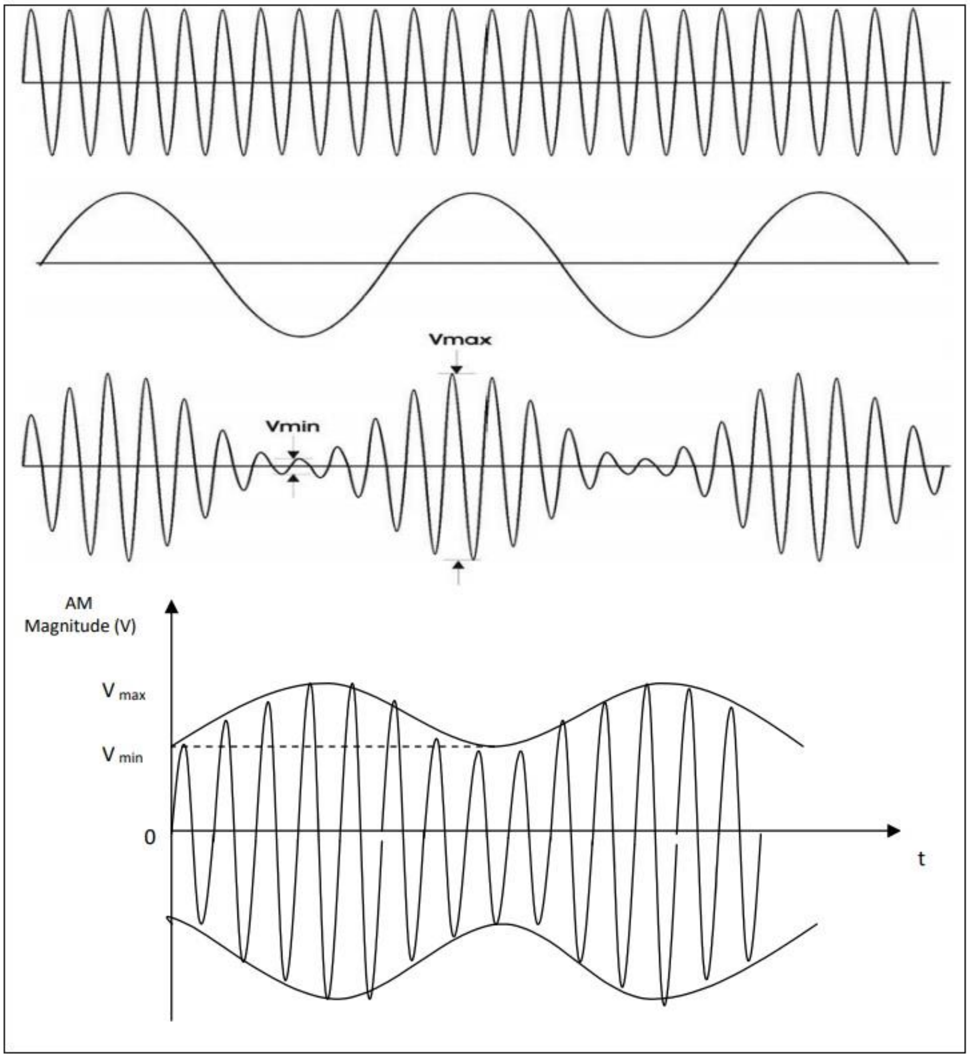
Assume RC = 500µs

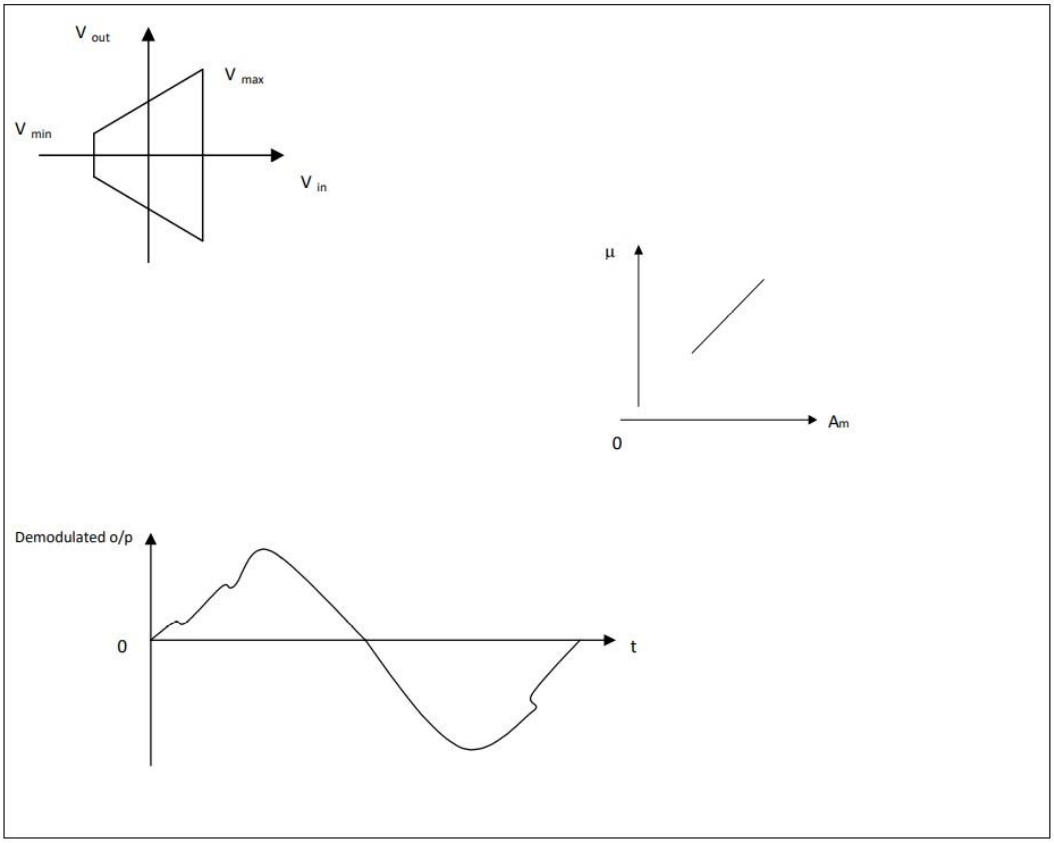
Let C=0.01µF

Then R =50kΩ

Assume R=47KΩ

Expected waveforms





Pre-Lab Work:

1. Basic theory of DSB-SC modulation using balanced modulator and synchronous detection techniques.
2. Time and Frequency analysis of DSB-SC waves.
3. Understanding the circuit diagrams of DSB-SC generation using balanced modulator and envelope detection.
4. Understanding the data sheets of components used in the experiment.
5. Computer simulations (Multisim / pSpice) are performed and the objectives are obtained prior to the hardware experiment.

Procedure

DSB-SC Modulation: Analog Inverter:

1. Initially wire the circuit for analog inverter circuit as shown in Fig..

2. Set audio signal generator (modulating signal) to 1 kHz sine wave with 1V peak.

3. Now observe the output wave form at the output of analog inverter (pin No6 of LM 741). This waveform

should be inversion of the input sine wave.

Analog Multiplier:

4. Now wire the analog multiplier circuit as shown in Fig.

5. A square wave of 10 kHz with 10 Vpp is connected to the Pin. No.8 of LF 398.

6. Observe the DSB-SC output at Pin No. 5 of LF 398.

7. Now slightly increase and decrease the modulating signal and note how the DSB-SC modulation changes.

Demodulation:

8. Now the demodulation circuit is connected across the output of DSB-SC modulator circuit.

9. By connecting and disconnecting the ‘C ‘ observe the demodulated signal.

10. By varying the modulating voltage in the DSB-SC modulation circuit, observe the demodulated signal.

11. Similarly, by varying the modulating signal frequency in the modulation circuit, observe the demodulated signal.

Tabular Column

Magnitude of m(t) Am (V)	Vmax	Vmin	η = (Vmax–Vmin) / (Vmax + Vmin)	% η

Viva Questions:

1. What are discrete frequencies in DSB-SC?
2. What is the advantage of DSB-SC over AM?
3. Mention the names of methods for DSB-SC generation?
4. What do you mean by coherence detection and non-coherent detection?
5. How a message signal recovered from DSBSC wave?
6. What is the disadvantage of DSB-SC?
7. What is the bandwidth of DSB-SC?
8. Why DSB-SC is not used for commercial broad casting?
9. Mention few applications for DSB-SC

Experiment 2: Frequency modulation using IC 8038 and demodulation using IC565

AIM: To design and conduct an experiment to generate an FM wave using IC 8038 and demodulation using IC 565.

THEORY: Frequency modulation: FM is that form of angle modulation in which the instantaneous frequency is varied linearly with the message signal. The IC 8038 waveform generator is a monolithic integrated circuit capable of producing high accuracy sine, square, triangular, saw tooth and pulse waveforms with a minimum number of external components. Figure 1 shows Pin diagram of IC 8038. It is a Precision waveform Generator/voltage contrived oscillator. It is a monolithic IC capable of producing high frequency sine, square, triangular, saw tooth &pulse wave (frequency range 0.001hz to 300khz).

APPLICATIONS:-

1. Sine wave output Buffer Amplifier.
2. Burst generator.
3. Variable Audio oscillator (20hz to 20khz)
4. Linear voltage controlled oscillator

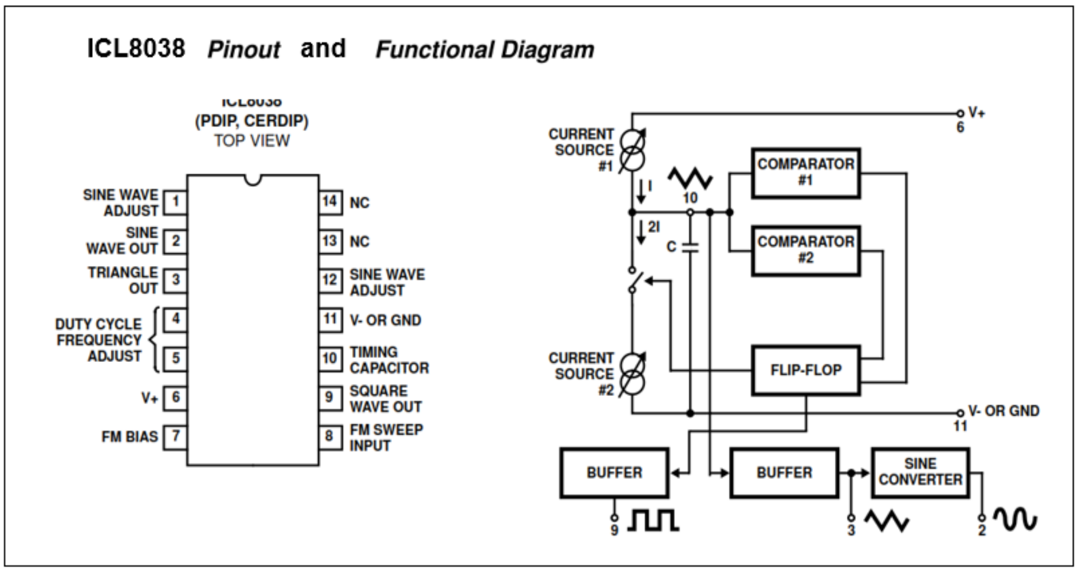


Figure 1: IC 8038 pinout and functional diagram

Basic principle of IC 8038: InFigure1of functional block diagram of IC 8038, an external capacitor C is charged and discharged by two current sources. Current source #2 is switched on and off by a flip-flop, while current source #1 is on continuously. Assuming that the flip-flop is in a state such that current source #2 is off, and the capacitor is charged with a current I, the voltage across the capacitor rises linearly with time. When this voltage reaches the level of comparator #1 (set at 2/3 of the supply voltage), the flip-flop is triggered, changes states, and releases current source #2.

This current source normally carries a current 2I, thus the capacitor is discharged with a net-current I and the voltage across it drops linearly with time. When it has reached the level of comparator #2 (set at 1/3 of the supply voltage), the flip-flop is triggered into its original state and the cycle starts again. Four waveforms are readily obtainable from this basic generator circuit. With the current sources set at I and 2I respectively, the charge and discharge times are equal. Thus, a triangle waveform is created across the capacitor and the flip-flop produces a square wave. Both waveforms are fed to buffer stages and are available at pins 3 and 9. The triangular wave is then passed through the on-chip wave shaper to generate sign wave.

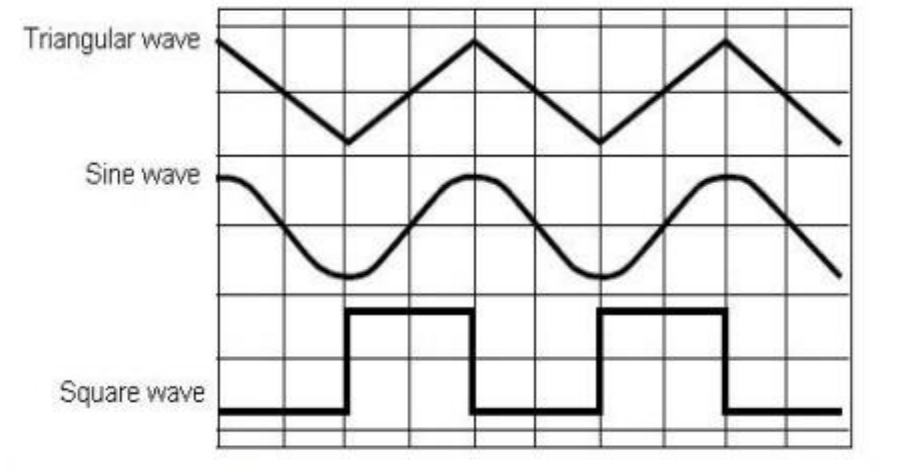


Figure 2: Output waveforms from function generator IC 8038

Part 2

The objective is to generate single- and multi-tone FM signals using a voltage-controlled oscillator and observe the waveforms.

Frequency Modulation &Sweeping: -

The frequency of the wave form generator is a direct function of the DC voltage at terminal ‘8’ (measured from v+) by altering this voltage, FM is performed. The frequency deviations can be archived by varying the voltage at pin No 8.

Implement the Frequency Modulator circuit shown in the data sheet (reproduced in Figure 4 below). The output of the VCO when no modulation is applied (i.e. the carrier frequency) is given by f = 0.33/RC, where R = RA = RB = RL Design the circuit for a carrier frequency of 33 kHz. Test the system response by feeding a 1-Hz audio-frequency tone to the FM sweep input pin. Observe the waveforms on the oscilloscope.

Design:-

Let R=Ra=Rb =10 KΩ

Let f=33 kHz

$$f = \frac{0.33}{RC}$$

Substituting for R and f in above equation

We get,   C =0.001μF

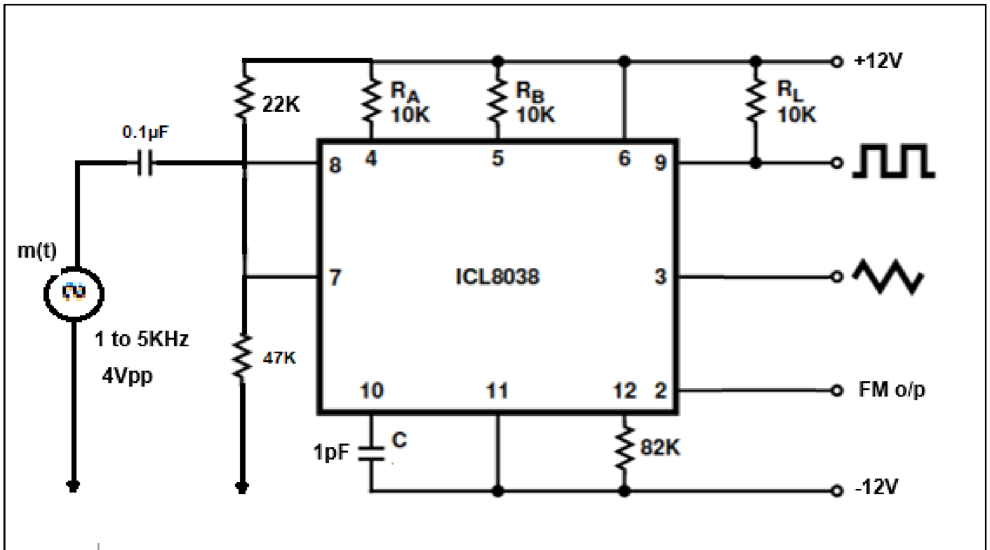


Figure3: Frequency modulation application circuit from ICL8038

PROCEDURE:

1. Rig up the circuit as in the Figure 3.
2. Observe the unmodulated sinusoidal carrier signal at Pin 2 of IC8038 and measure the amplitude and frequency without applying modulating signal.
3. Apply modulating signal with amplitude 4V Peak to peak and frequency of 1 KHz (approximate).
4. Observe the frequency modulated output at pin2.
5. Note down maximum and minimum frequencies corresponding to two peaks of the modulating signal.
6. Compute the parameters of the FM signal and tabulate them.

Waveforms:

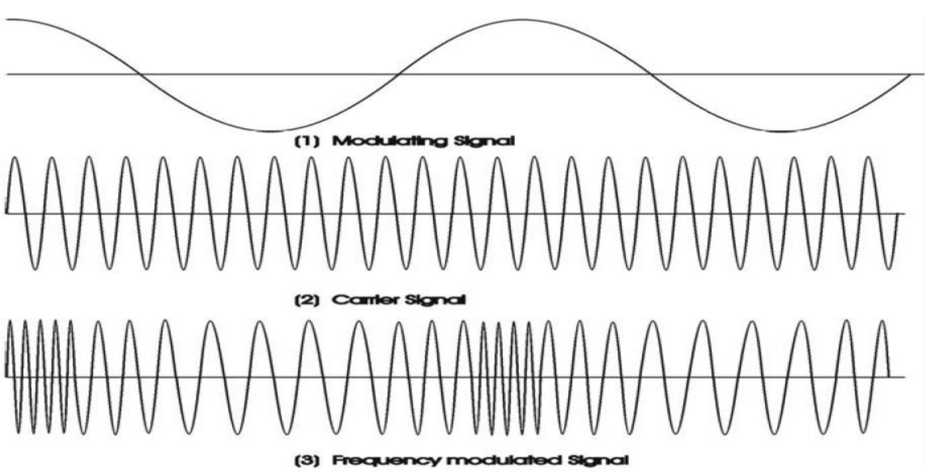


Figure 4: waveforms of FM

Result and calculations:

fmax=

fmin=

fm=

Am=

1. Frequency deviation: $\Delta f = \frac{f_{max} - f_{min}}{2} =$

2. Modulation index: $\beta = \frac{\Delta f}{f_m} =$

3. Frequency sensitivity = $K_f = \frac{\Delta f}{A_m}$ =

4. Bandwidth = $2(\Delta f + f_m)$ Hz =

Part 3
The objective is to generate an FM demodulated signal using IC 565 and observe the waveform.

Theory: **LM565** is a **general purpose PLL** (**Phase Locked Loop**) IC designed for demodulation, frequency multiplication and frequency division. The device mainly consists of two components, one is voltage controller oscillator (VCO) and other is phase detector (PD). In which VCO is designed for highly linear operation and PD with good carrier suppression. Pin diagram of IC 565 is shown in Figure 5.

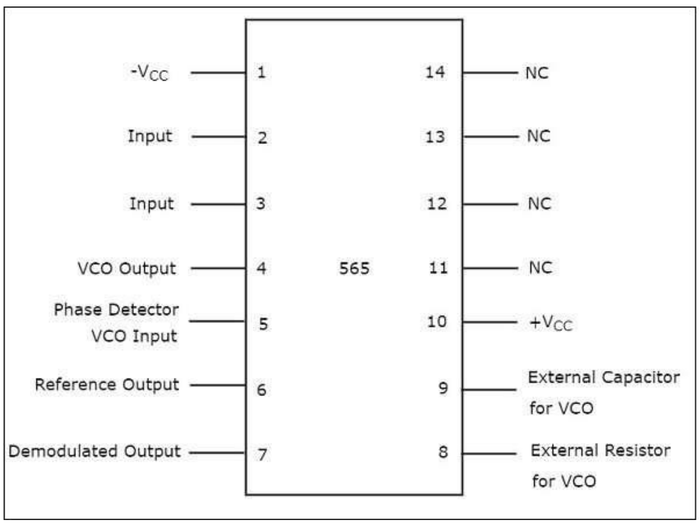


Figure 5: Pin diagram of IC 565

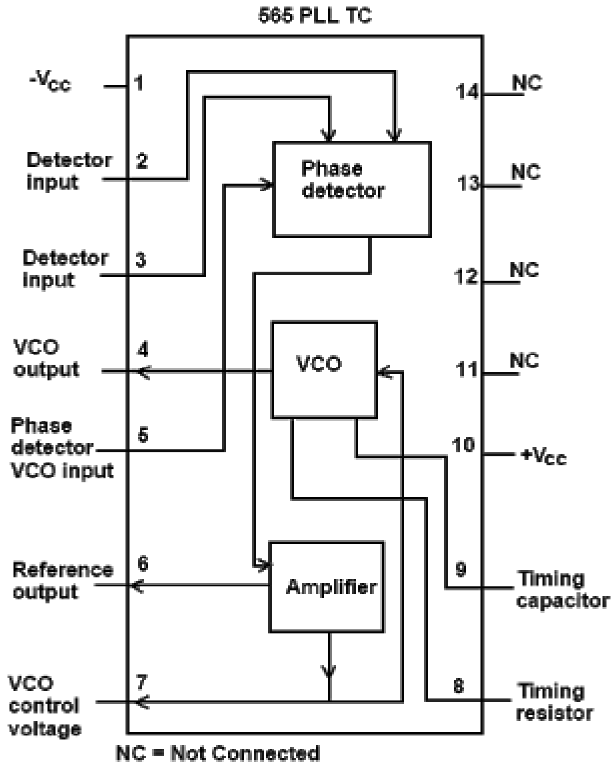


Figure 6: Internal block diagram of IC 565

Internal block diagram of IC 565 is shown in Figure 6. The output of a phase detector is applied as an input of active low pass filter with Amplifier. Similarly, the output of Amplifier is applied as an input to VCO.

The **working** of a PLL is as follows −

- **Phase detector** produces a DC voltage, which is proportional to the phase difference between the input signal having frequency of fin and feedback (output) signal having frequency of fout.
- A **Phase detector** is a multiplier and it produces two frequency components at its output − sum of the frequencies fin and fout and difference of frequencies fin & fout.
- An **active low pass filter (Amplifier)** produces a DC voltage at its output, after eliminating high frequency component present in the output of the phase detector. It also amplifies the signal.
- A **VCO** produces a signal having a certain frequency, when there is no input applied to it. This frequency can be shifted to either side by applying a DC voltage to it. Therefore, the frequency deviation is directly proportional to the DC voltage present at the output of a low pass filter.

The above operations take place until the VCO frequency equals to the input signal frequency. Based on the type of application, we can use either the output of active low pass filter or output of a VCO. PLLs are used in many **applications** such as FM demodulator, clock generator etc.

PLL operates in one of the **following three modes** −

- Free running mode
- Capture mode
- Lock mode

Initially, PLL operates in **free running mode** when no input is applied to it. When an input signal having some frequency is applied to PLL, then the output signal frequency of VCO will start change. At this stage, the PLL is said to be operating in the **capture mode**. The output signal frequency of VCO will change continuously until it is equal to the input signal frequency. Now, it is said to be PLL is operating in the **lock mode**.

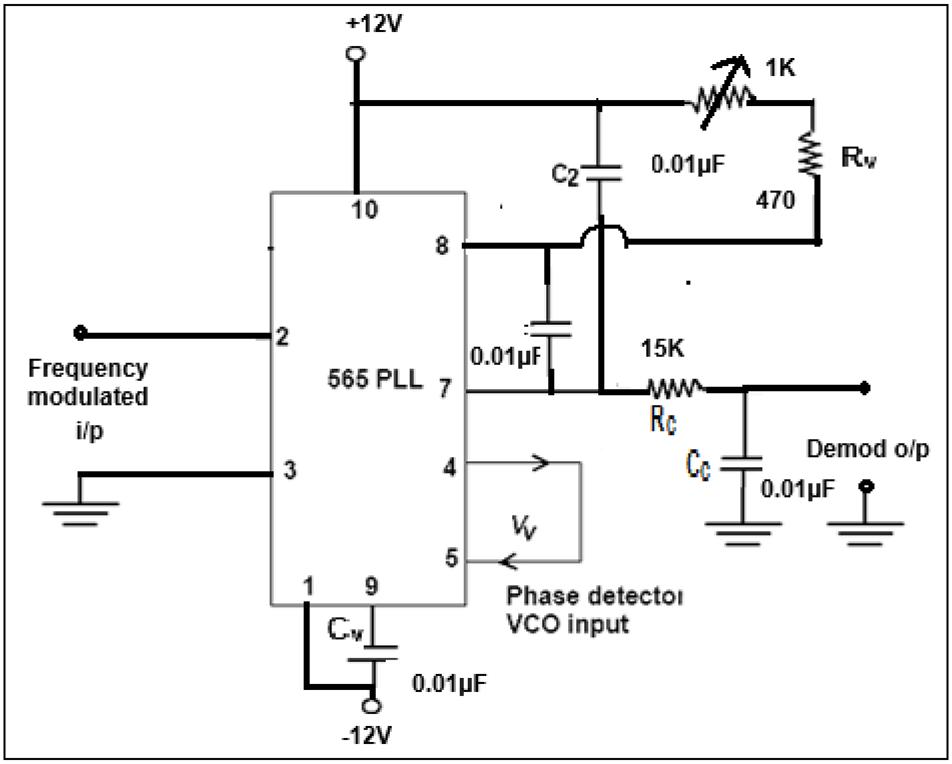


Figure 7: Frequency demodulation Circuit diagram using IC 565

Design:

The **VCO** produces an output at pin number 4 of IC 565, when the pin numbers 2 and 3 are grounded.

Free-running Frequency

Fig. 8 shows a PLL circuit with LM565. In the absence of the input signal, the output frequency of the VCO is called the free-running frequency fo. In the PLL circuit of Fig. 8, the free-running frequency of LM565 is determined by the timing components C_V and R_V, and can be found by

$$f_0 = \frac{0.25}{R_v C_v}$$

Let $C_v = 0.01\ \mu F$ and $f_0 = 1$ KHz

From above equation, $R_v = 833\ \Omega$

Low pass filter:

$$f_0 = \frac{1}{2\pi R_c C_c}$$

Let Cc=0.01 μF and f_0=1 KHz

From above equation R_c= 15 KΩ

- **Pin numbers 4 and 5**are to be shorted with an external wire so that the output of VCO can be applied as one of the inputs of phase detector.
- IC 565 has an internal resistance of 3.6KΩ. A capacitor, C has to be connected between pin numbers 7 and 10 in order to make a **low pass filter** with that internal resistance.

PROCEDURE:

1. Make connections as shown in circuit diagram of Figure 7.
2. Apply FM signal at pin 2.
3. Observe the demodulated signal. Measure and note down the output frequency.

RESULT:

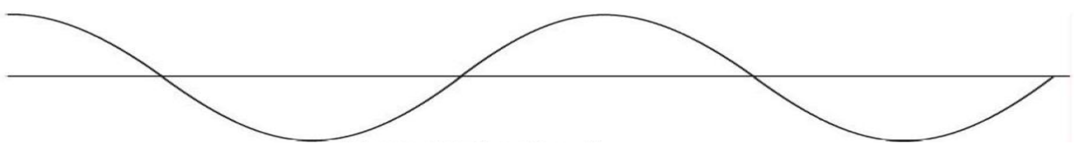


Figure 8: Baseband signal (message signal)

fm=

Experiment 3. Pulse Sampling , Flat Top Sampling and Reconstruction

Pulse Amplitude Modulation:

AIM: To conduct Pulse Sampling , Flat Top Sampling and Reconstruction and to observe the waveforms.

APPARATUS REQUIRED:

Transistor BC107/2N2222, Resistor 10kΩ 2nos, 22kΩ, Capacitor 1.7µF, Signal generator, CRO.

THEORY: Pulse Amplitude Modulation is defined as the variation of amplitude of the pulses (carrier) in accordance with the message signal. The output is a series of pulses, the amplitudes of which vary in proportion with the message signal. Sampling theorem states that "If the sampling rate exceeds twice the maximum signal frequency, then the original signal can be reconstructed in the receiver with minimal distortion". The most popular sampling method is the sample and hold method in which flat top samples is created. The top of the samples remains a constant and is equal to the instantaneous value of the baseband signal at the start of sampling. A flat top pulse has constant amplitude established by the sample value of the signal at the beginning of the pulse interval and each sample is extended to have duration of T seconds.

Circuit Diagram: Pulse sampling

The transistor conducts during the positive half cycle of c (t) and the transistor does not conduct during the –ve half cycle of c (t).For demodulating the sampled signal a simple RC low pass filter is used. The analog signal can be exactly recovered from the sampling provided the pulse repetition frequency of the pulses is greater than twice the analog frequency according to sampling theorem. The circuit and output wave form is as shown in fig 1 and 2 respectively.

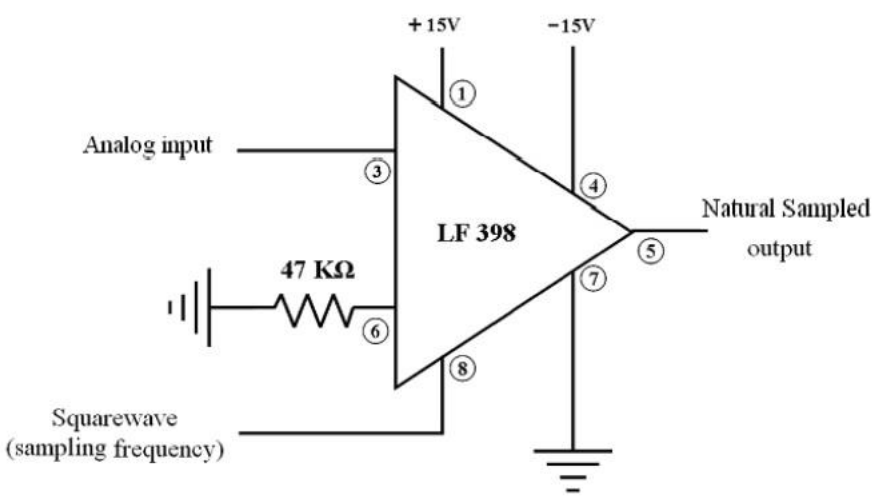
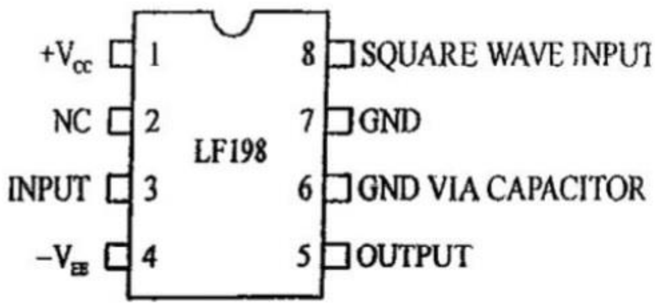
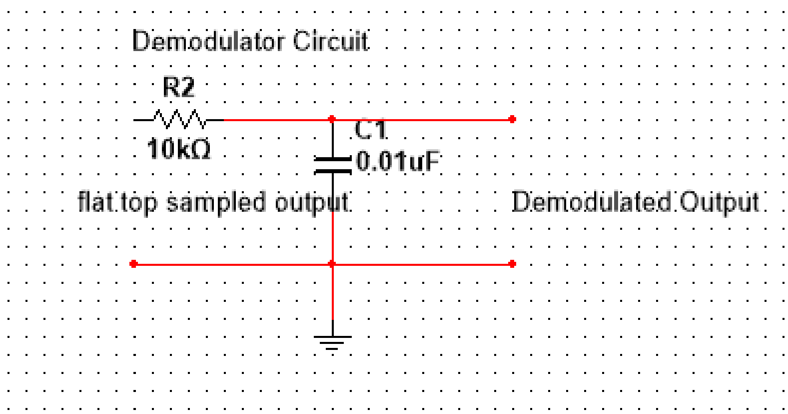


Fig 1 :PAM signal generation



Demodulation circuit:



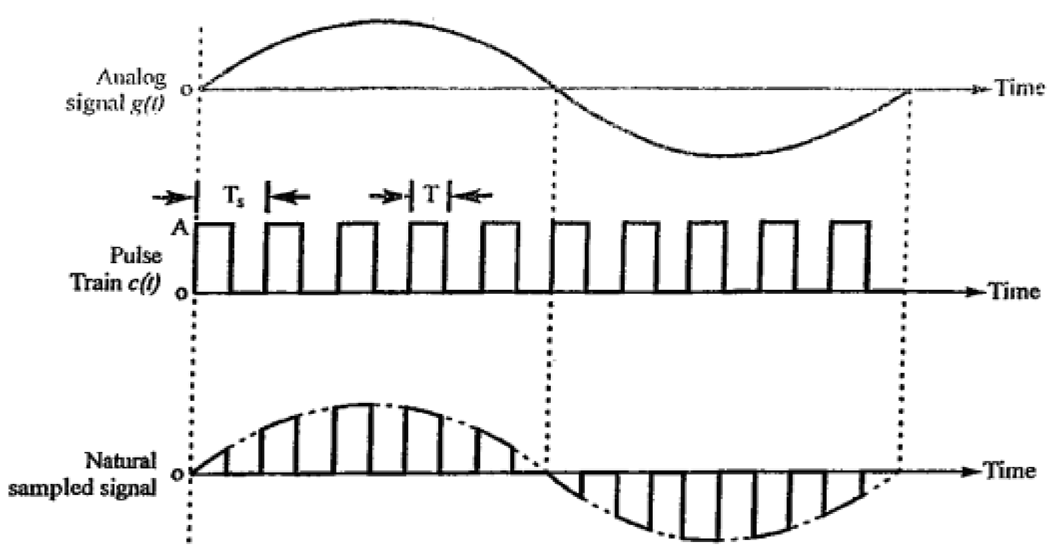


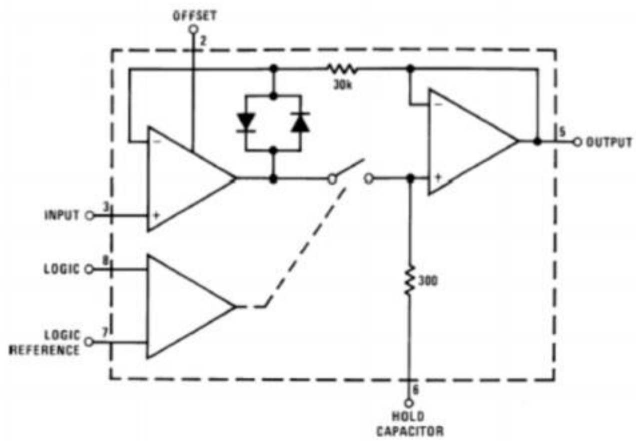
Fig : Natural Sampling Waveform

fig 2: Simulink PAM output

Circuit diagram: Flat top Sampling

The circuit diagram and output wavefrom for falt top sampling is as shown in fig 3 and 4 respectively

Functional Block Diagram of IC LF 198/398



Circuit Diagram:

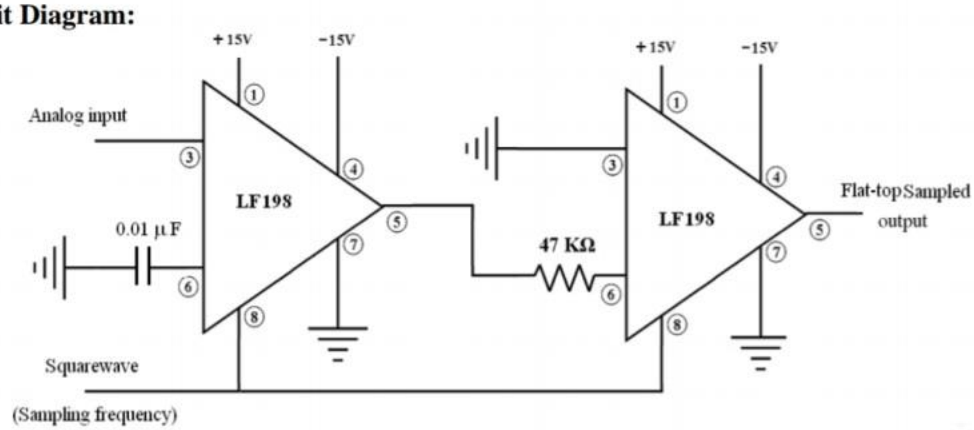


Fig: Circuit Diagram for Flat top Sampling

Demodulator Circuit:

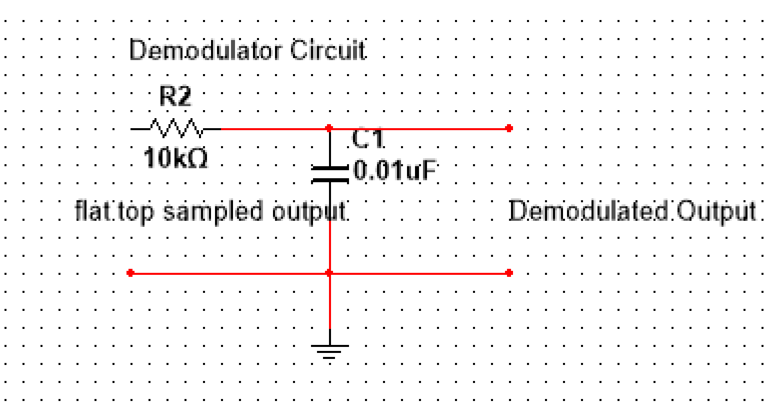


fig4: Flat Top Sampling Demodulation

Design: For Demodulation:

$1/fc << RC << 1 / f_m$

Let C = 0.1 µF

Then R = ……. kΩ

Expected Waveform:

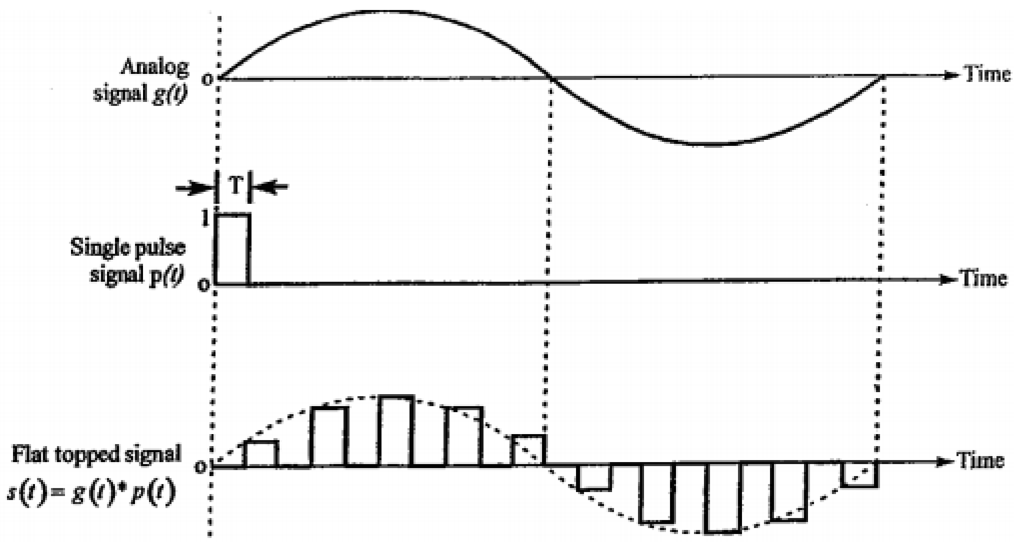


Fig: Flat top Sampling waveform

Procedure:

1. Connections are made as per the circuit diagram.
2. Sine wave input signal of 5V,1KHz and square wave(carrier signal) of 5V 10KHz is given as input to the circuit. Message signal is given to collector and carrier signal (pulse signal) of high frequency is given to base of the transistor.
3. Output is taken at emitter and observed across the CRO.
4. Repeat the procedure for different conditions of sampling i.e. $f_s= 2f_m$, $f_s> 2f_m$, $f_s<$
    a. $2f_m$.
5. Connect the demodulator circuit and the modulated output is given as input to the demodulator to obtain the message signal.obtain output for different conditions.
6. Plot the waveforms.

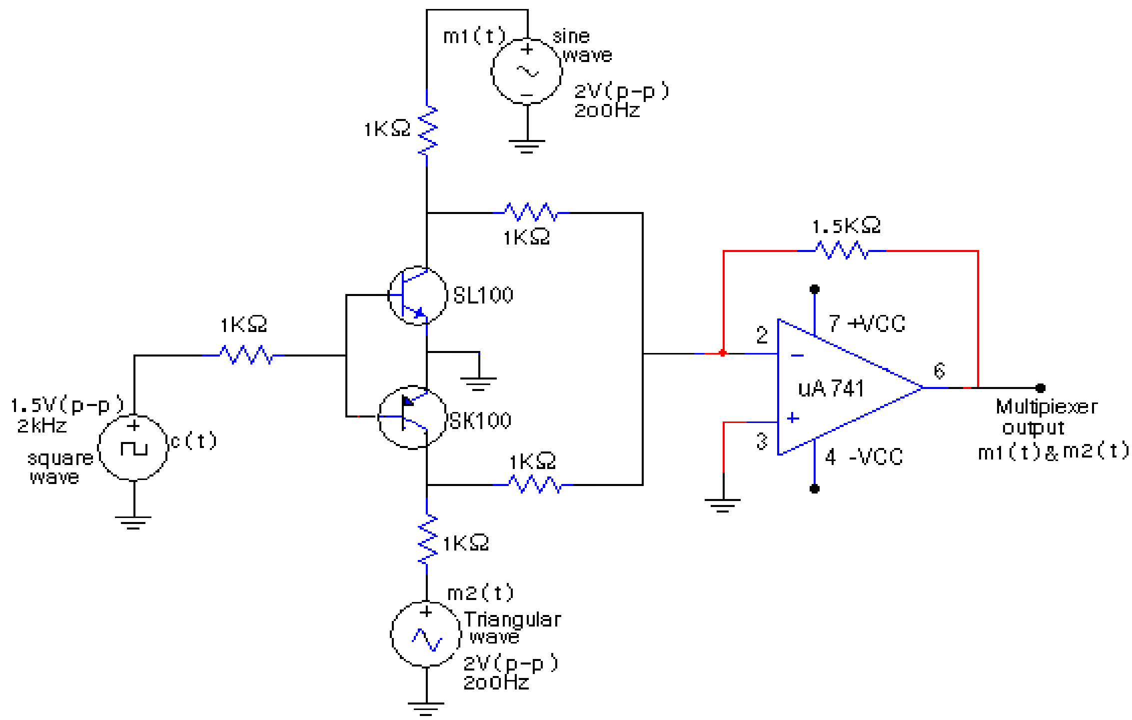
Result: Pulse sampling and flat top sampling was conducted and output waveforms were observed.

Experiment 4.TDM OF TWO BAND LIMITED SIGNALS

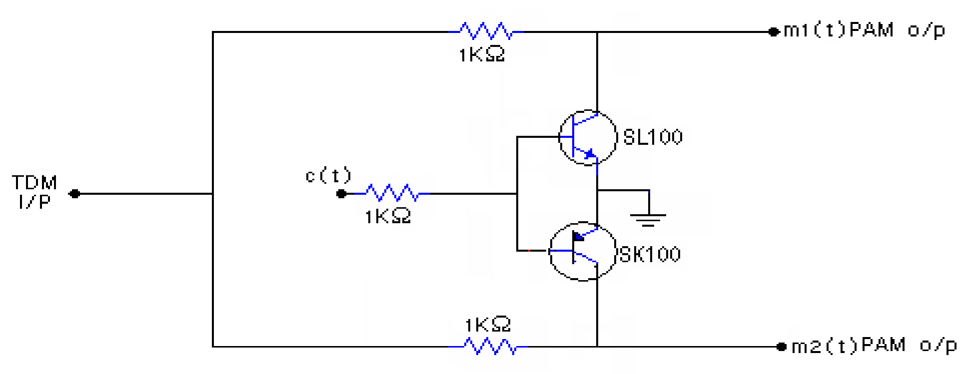
AIM: To design and demonstrate the working of TDM and recovery of two band-limited PAM signals.

COMPONENTS REQUIRED:
- Transistor SL100, SK100, Op-amp μA 741, Resistors 1KΩ8nos, 1.5KΩ 1no.
- Function Generator, Regulated DC Power supply, Spring Board, CRO.

CIRCUIT DIAGRAM:



DEMODULATION:



USING IC

CIRCUIT Diagram

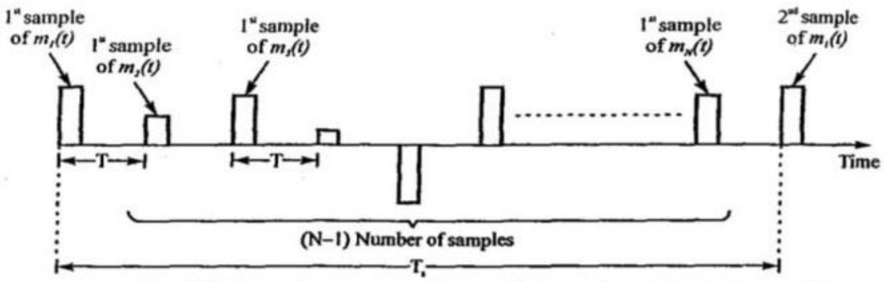


Fig 1.2 Illustrating accommodation of N samples within the interval T

The TDM process introduces a bandwidth expansion factor of N. at the receiver, the decommulated demultiplexer distributes the signals to appropriate low pass reconstruction filters. For satisfactory operation of the process, the synchronization of transmitter and receiver is essential.

Circuit diagram:

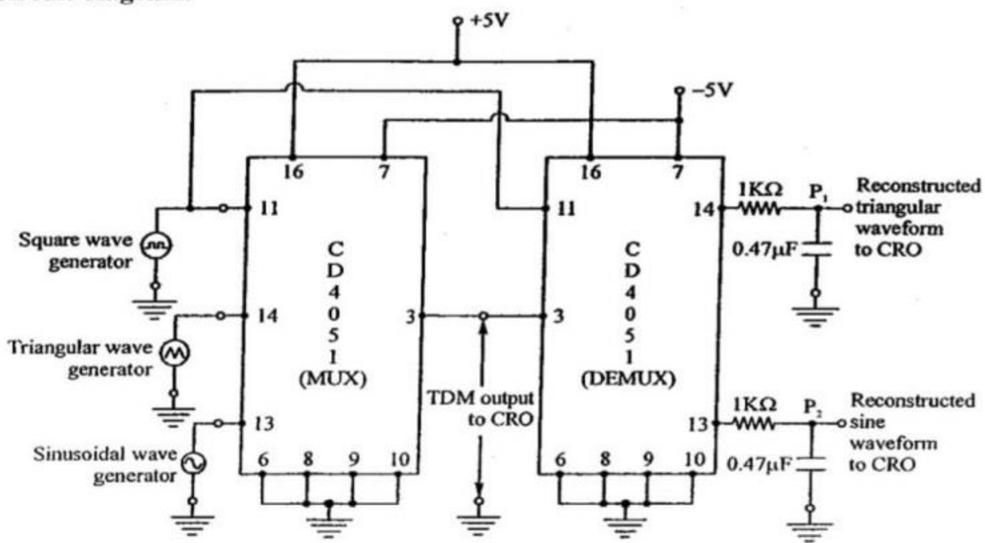


Fig 1.3 multiplexing and demultipexing of two different signals

THEORY:

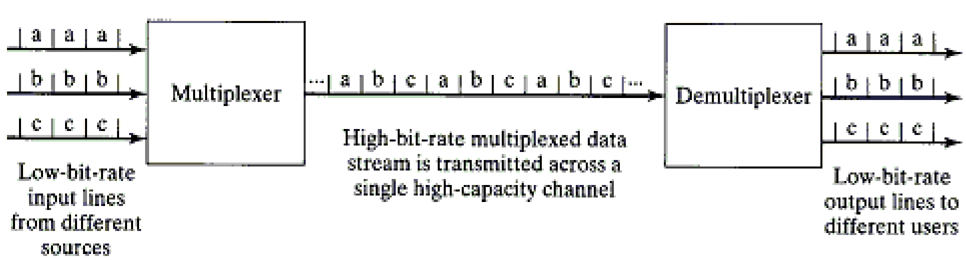
More efficient communication system can be obtained if a station transmits more than one “message” on the same carrier and on the same channel, or number of transmitters is transmitting simultaneously on the same channel. This process is known as “Multiplexing” and has been used for many years in long distance telephony. Multiplex transmissions have been used in commercial communications, not only for voice channels (wireless), but also for facsimile. Multiplexing has also been used since a long time for broadcasting. Remote data transmission (telemetry) would not be practicable were it not for multiplexing.

There are two types of multiplexing:

- Time division Multiplexing (TDM)
- Frequency division Multiplexing (FDM)

In practical it is to combine a set of low-bit-rate streams, each with a fixed and pre- defined bit rate, into a single high-speed bit stream that can be transmitted over a single channel. This technique is called *time division multiplexing* (TDM) and has many applications, including wireline telephone systems and some cellular telephone systems. The main reason to use TDM is to take advantage of existing transmission lines. It would be very expensive if each low-bit-rate stream were assigned a costly physical channel (say, an entire fiber optic line) that extended over a long distance.

Consider, for instance, a channel capable of transmitting 192 kbit/sec from Chicago to New York. Suppose that three sources, all located in Chicago, each have 64 kbit/sec of data that they want to transmit to individual users in New York. As shown in Figure given below, the high-bit-rate channel can be divided into a series of *time slots*, and the time slots can be alternately used by the three sources. The three sources are thus capable of transmitting all of their data across the single, shared channel. Clearly, at the other end of the channel (in this case, in New York), the process must be reversed (i.e., the system must divide the 192 kbit/sec multiplexed data stream back into the original three 64kbit/sec data streams, which are then provided to three different users). This reverse process is called *demultiplexing.*



Time division Multiplexing (TDM)

Choosing the proper size for the time slots involves a trade-off between efficiency and delay. If the time slots are too small (say, one bit long) then the multiplexer must be fast enough and powerful enough to be constantly switching between sources (and the demultiplexer must be fast enough and powerful enough to be constantly switching between users). If the time slots are larger than one bit, data from each source must be stored (buffered) while other sources are using the channel. This storage will produce delay. If the time slots are too large, then a significant delay will be introduced between each source and its user. Some applications, such as teleconferencing and videoconferencing, cannot tolerate long delays.

As shown in Example, the sources that are multiplexed may have different bit rates. When this occurs, each source is assigned a number of time slots in proportion to its transmission rate.

PROCEDURE:-

1. Rig up the circuit as shown in the circuit diagram for multiplexer.
2. Feed the input message signals m_1 and m_2 of 2V (P-P) at 200Hz.
3. Feed the high frequency carrier signal of 2V (P-P) at 2kHz.
4. Observe the multiplexed output.
5. Rig up the Demodulator circuit as shown in the circuit diagram for Demultiplexer.
6. Observe the Demultiplexer output in the CRO.

WAVEFORM:

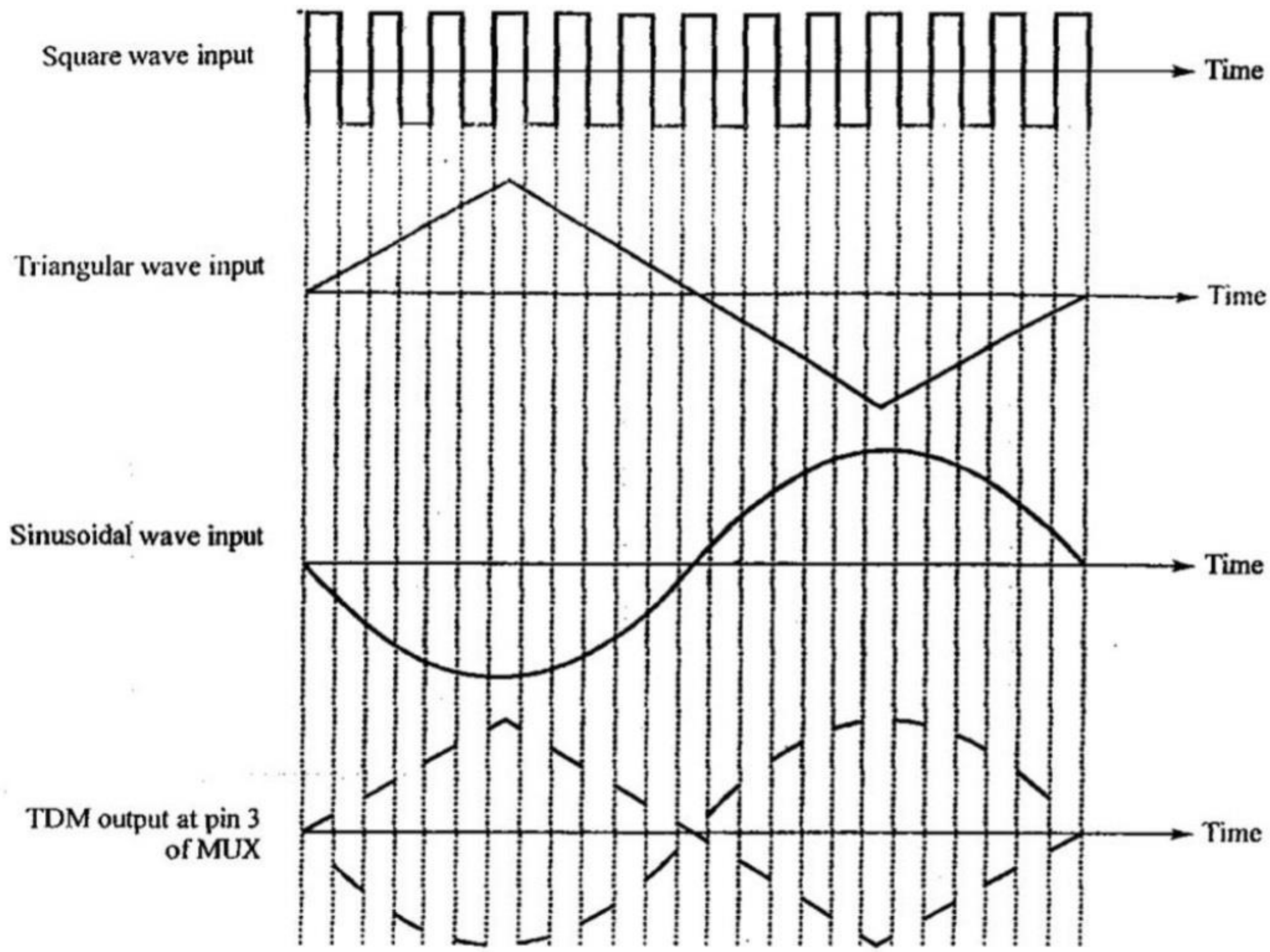
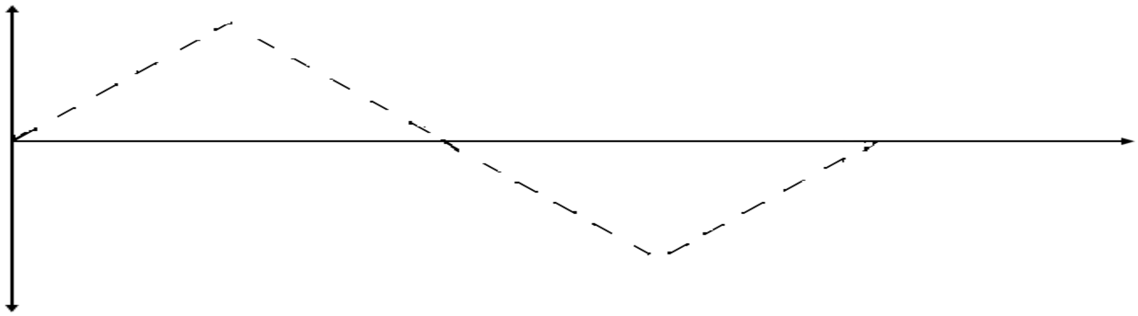
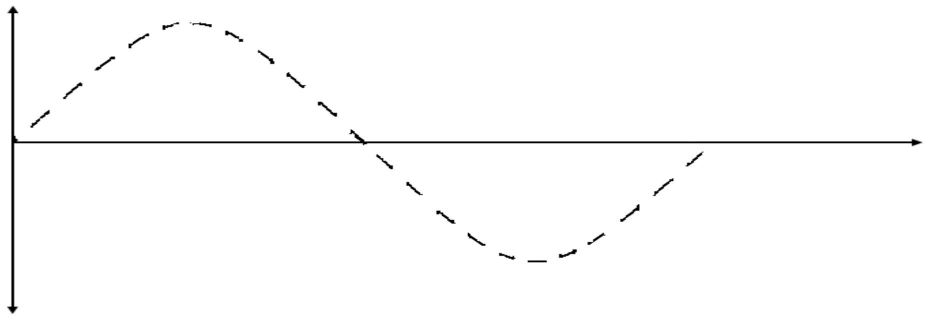


Fig 1.6 TDM output at high square wave frequency

DEMODULATION WAVEFORMS :





Experiment 5. FSK and PSK Generation and Detection (Binary)

FSK Generation and Detection

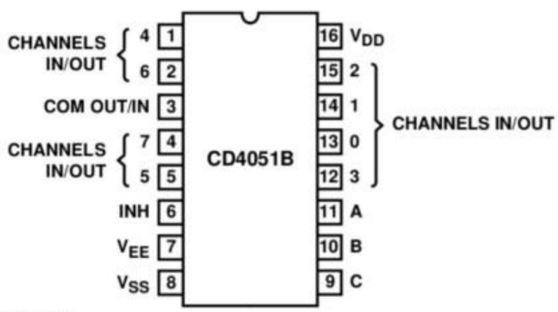
AIM: Design & Demonstrate the working of FSK with a suitable circuit for carrier signals of ___Hz & ___Hz, determine the frequency deviation & modulation index. Demodulate the above signal with the help of suitable circuit.

APPARATUS REQUIRED:

- Transistor SL100, SK100, Op-amp µA 741 2nos, Diode OA79, Resistors 680Ω2nos, 560Ω2nos, 10KΩ4nos, 1KΩ, 0KΩpot, Capacitors 0.1mF 3nos, 0.22mF. Function Generator, Regulated DC Power supply, Spring Board, CRO.

CIRCUIT DIAGRAM:

Modulation:



Circuit diagram:-
Modulator:-

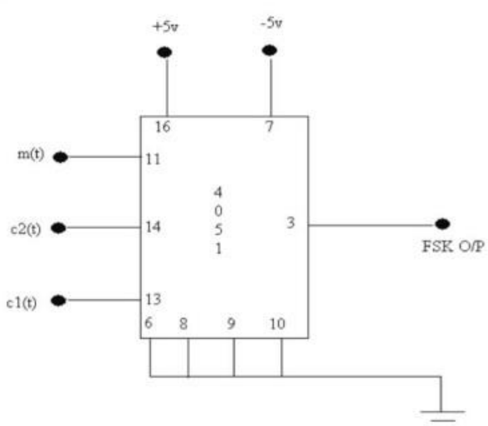
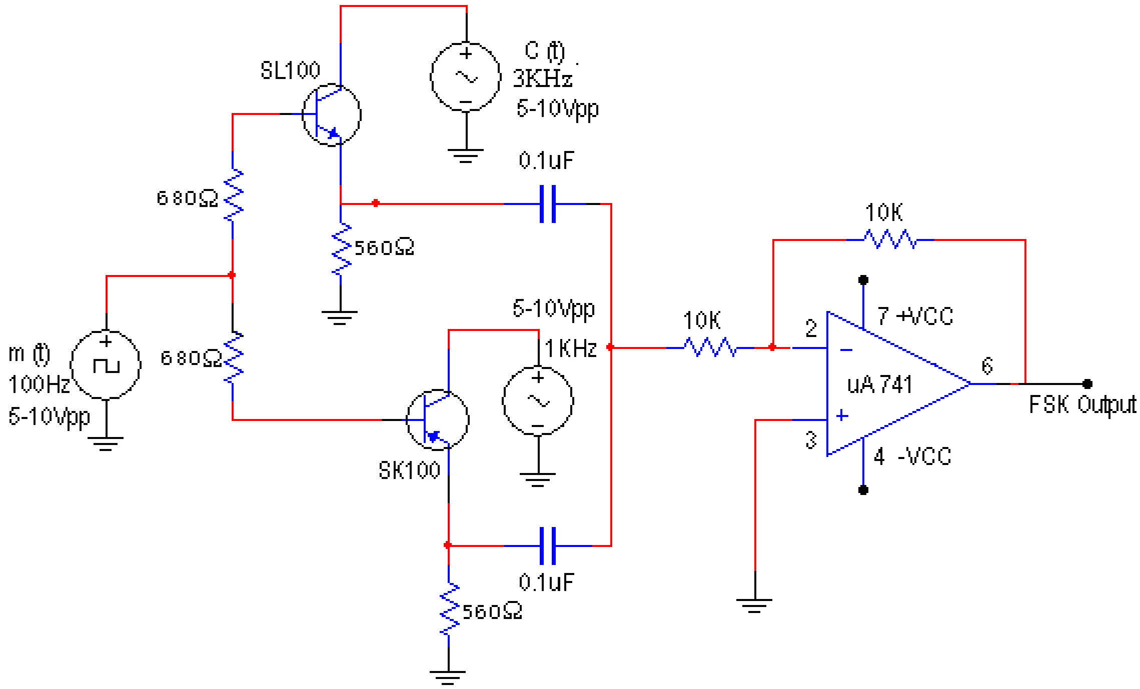
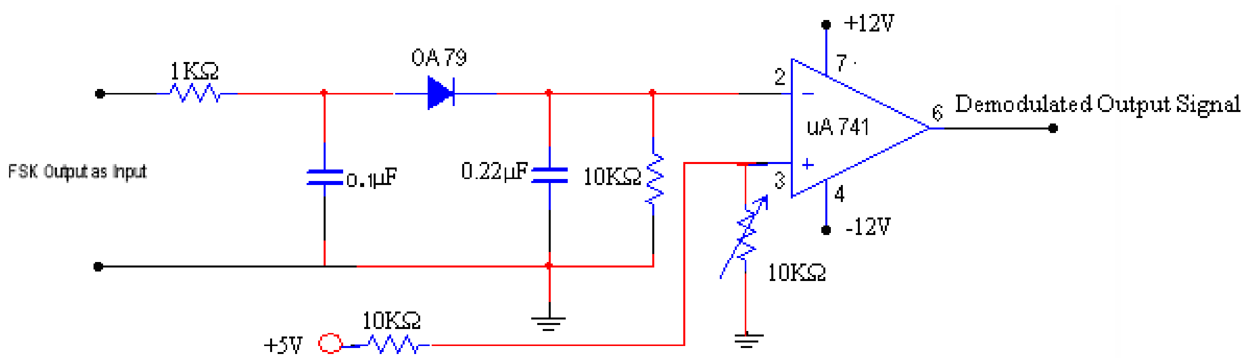


Fig 4.1 Circuit Diagram of FSK Modulator



Demodulation:



THEORY:

As its name suggests, a frequency shift keyed transmitter has its frequency shifted by the message. Although there could be more than two frequencies involved in an FSK signal, in this experiment the message will be a binary bit stream, and so only two frequencies will be involved. The word 'keyed' suggests that the message is of the 'on-off' (mark-space) variety, such as one (historically) generated by a Morse key or more likely in the present context, a binary sequence.

Modulation: Frequency Shift Keying is the process of generating a modulated signal from a digital data input. If the incoming bit is 1, a signal with frequency f1 is sent for the duration of the bit. If the bit is 0, a signal with frequency f2 is sent for the duration of this bit. This is the basic principle behind FSK modulation. Two transistors are used in switching configuration to obtain FSK signal. When the input bit is 1, NPN transistor is ON and the corresponding carrier signal is output. If the input bit is 0, PNP transistor is ON and the corresponding carrier is output.. In the demodulator circuit, the FSK modulated signal is applied to a low pass filter. This filter passes the frequency of either 0 or 1. This filter passes the selected frequency and rejects the other. Thus ASK is obtained and then demodulated using envelope detector and comparator.

Demodulation: There are different methods of demodulating FSK. A natural classification is into synchronous (coherent) or asynchronous (non-coherent).

Representative demodulators of these two types are the following:

Asynchronous Demodulator:
A close look at the waveform of Figure 1 reveals that it is the sum of two amplitude shift keyed (ASK) signals. The receiver of Figure 3 takes advantage of this. The FSK signal has been separated into two parts by band pass filters (BPF) tuned to the MARK and SPACE frequencies.

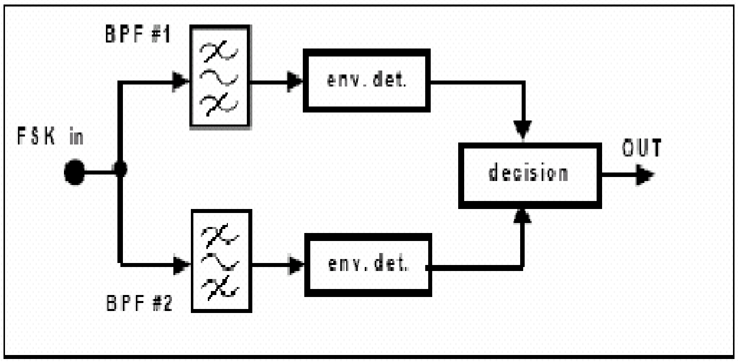


Figure: Demodulation by conversion to ASK

The output from each BPF looks like an amplitude shift keyed (ASK) signal. These can be demodulated asynchronously, using the envelope.

The decision circuit, to which the outputs of the envelope detectors are presented, selects the output which is the most likely one of the two inputs. It also re-shapes the waveform from a band limited to a rectangular form. This is, in effect, a two channel receiver. The bandwidth of each is dependent on the message bit rate. There will be a minimum frequency separation required of the two tones.

Synchronous Demodulator:

In the block diagram of Figure 4 two local carriers, on each of the two frequencies of the binary FSK signal, are used in two synchronous demodulators. A decision circuit examines the two outputs, and decides which is the most likely.

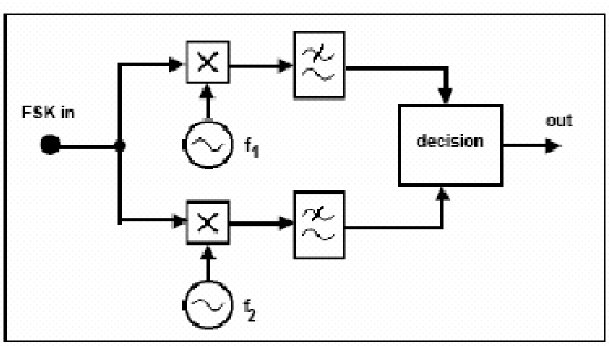


Figure: Synchronous Demodulation

This is, in effect, a two channel receiver. The bandwidth of each is dependent on the message bit rate. There will be a minimum frequency separation required of the two tones. This demodulator is more complex than most asynchronous demodulators.

DESIGN:

(a) MODULATION:

I_{Csat} = 2mA; h_{fe} = 30; V_{BEsat} = 0.7V; V_{CEsat} = 0.2v

We know that $I_B = I_{C/\ hfe}$

= 2mA/0.3

I_B = 0.0667mA

Choose I_B=6.67mA (To make Q point be very much in the saturation region) Also R_E =

V_E/I_E == $(Vc-V_{CE})/I_E$

= (5-0.2)/8.667mA

= 553.82Ω (=560 Ω)

Also $R_B = (V_m-V_{BE}-V_E)/I_B$

= (10-0.7-0-4.8)/ I_B

R_B=675 Ω (=680 Ω)

(b) DEMODULATION:

For LPF :

To convert FSK signal to ASK, the LPF must allow 1KHz signal but reject 3KHz signal or vice versa.

Here, the LPF is designed to pass 1KHz. o the design equation can be written as :

$$1KHz <= 1/2piRC \quad f_{c,LPF} < 3KHz .$$

Choose f_c=1.5KHz.

Let C=0.1 m F, then

$$R = \frac{1}{2\text{pi} f_{c,LPF} C} = 1.06\,K\text{W}$$

(Choose R=1KΩ

For envelope detector :

We know that $1/f_c < R \times C < 1/f_m$

Take f_{cmax} = 3KHz

Then, 1/3KHz < R x C <1/100

Take R x C = 2.2ms, and let C = 0.22mF Then

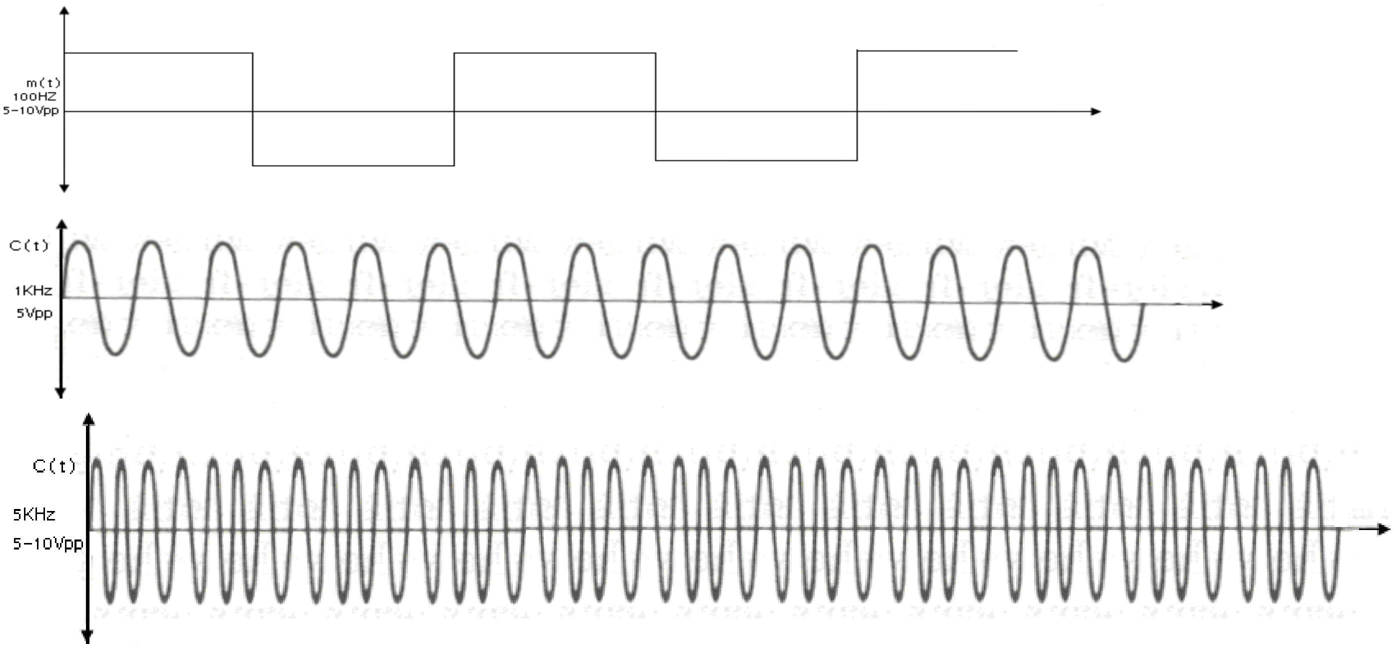
R=10KΩ

PROCEDURE:

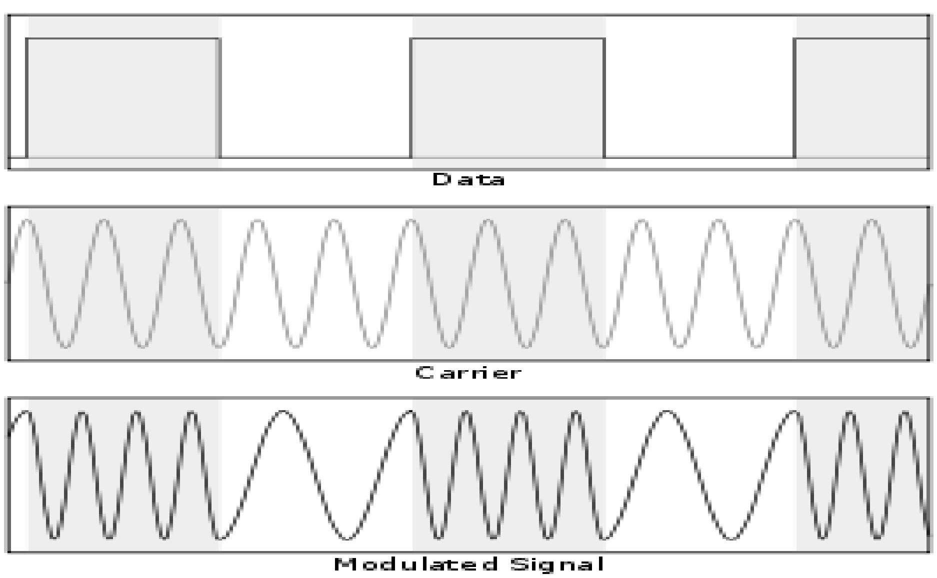
1. Rig up the circuit as shown in the circuit diagram.
2. Give two different frequencies for carrier and message to the base of NPN and PNP transistor.
3. FSK signals are obtained at the inverting amplifier terminal.
4. For demodulation, convert FSK to ASK then demodulate the message signal by envelope detector and comparator.

Implies, 0.3ms< R x C < 10ms.

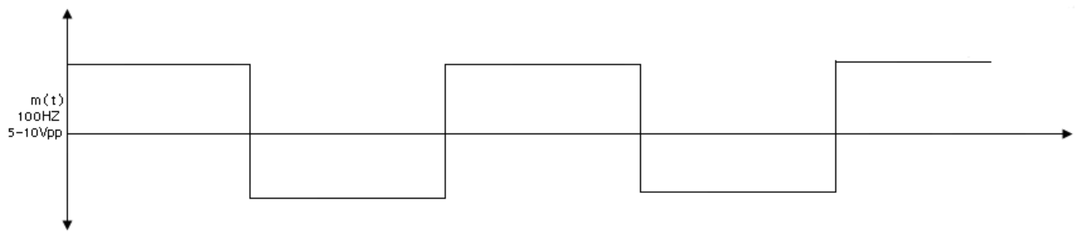
MODEL WAVEFORMS:



MODULATION WAVEFORMS:



DEMODULATION WAVEFORMS:



Results:

b.PSK Generation and Detection

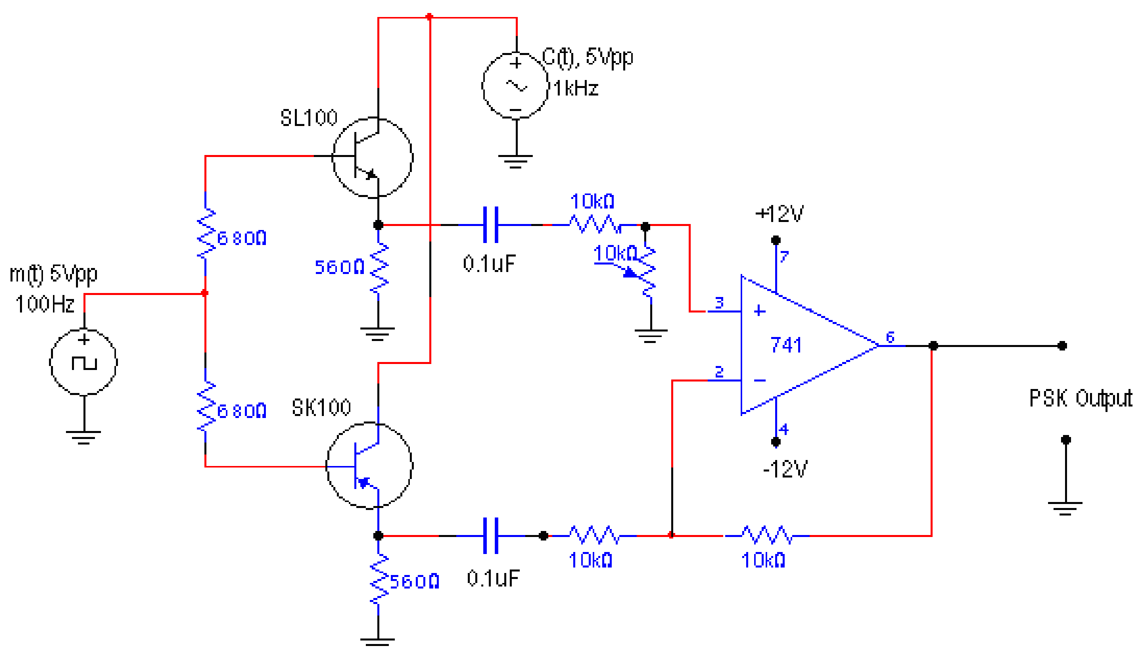
AIM: Design & Demonstrate the working of BPSK modulated signals for a given carrier signal of frequency 1 KHz to transmit a given digital data of frequency 100 Hz. Demodulate the BPSK Signal to recover the digital data.

APPARATUS REQUIRED:

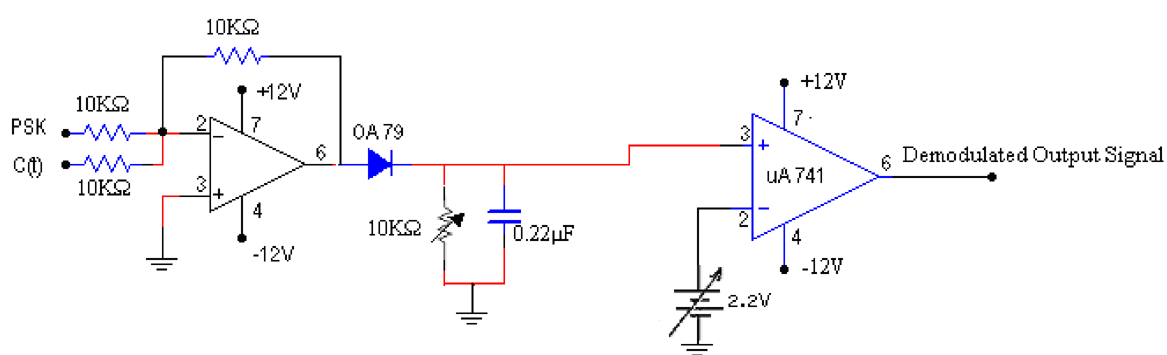
Transistor SL100, SK100, Op-amp µA 741 3nos, Diode OA 79, Resistors 560Ω 2nos, 680Ω 2nos, 10KΩ 6nos, Capacitors 0.1mF 2nos, 0.22mF, Potentiometer 10KΩ 2nos. Function Generator, Regulated DC Power supply, Spring Board, CRO.

CIRCUIT DIAGRAMS:

MODULATION:



DEMODULATION:



THEORY:

Phase Shift Keying is a digital modulation Technique. A cosinusoidal carrier of a fixed amplitude and frequency is taken. The digital data of 1's and 0's is converted to

$$s(t) = A_C \cos(2\pi f_c t)$$

$$s(t) = A_C \cos(2\pi f_c t + \pi)$$

respectively. Here, Phase shift keying is obtained using a switch followed by OP-AMP circuitry. If incoming bit is 0, the output is same as the carrier; if it is zero, the output is 180^0 phase shifted version of the carrier signal. For demodulation the PSK signal is converted to ASK signal using OPAMP circuit and then ASK signal is demodulated using an envelope detector and comparator.

Design:

(a) MODULATION:

I_{Csat} = 2mA; h_{fe} = 30; V_{BEsat} = 0.7V; V_{CEsat} = 0.2v

We know that $I_B > I_C / h_{fe}$

> 2mA/30

I_B >0.0667mA. So choose I_B=6.67mA Also

$R_E = V_E/I_E == (Vc-V_{CE})/I_E$

= (5-0.2)/8.667mA = 553.82Ω (=560 Ω)

Also $R_B= (V_m-V_{BE}-V_E)/I_B$

= (10-0.7-0-4.8)/ I_B =675 Ω (=680 Ω)

(b) DEMODULATION:

We know that $1/f_c$ < R x C < $1/f_m$. Here f_{cmax} = 1 kHz

Then, 1/1kHz < R x C <1/100

Implies, 1ms< R x C < 10ms.

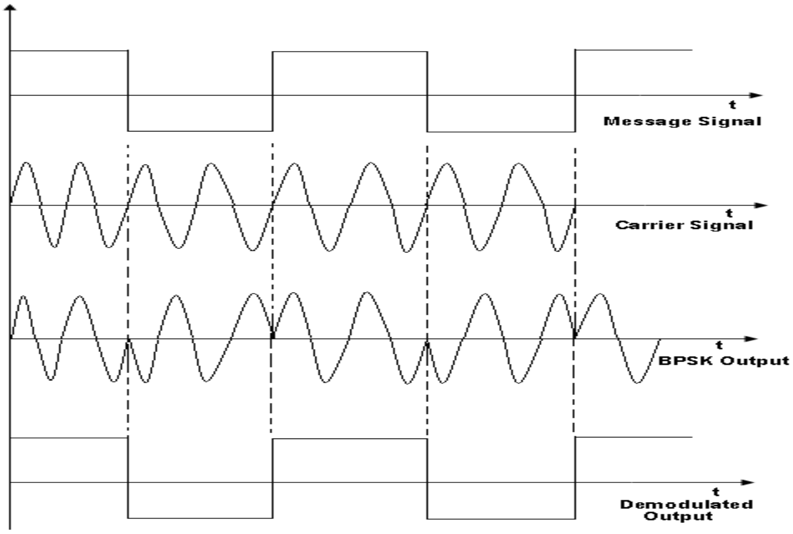
Take R x C = 2.2ms, and let C = 0.22mF

Then **R = 10KΩ (pot)**

Procedure:

1. Rig up the circuit as shown in the circuit diagram.
2. Carrier wave and message is given at the input of switching circuit.
3. Observe the PSK output on the CRO.
4. PSK out put is fed as input to the demodulation circuit.
5. Observe the demodulated output on the CRO.

MODEL WAVEFORMS:



t
Message Signal
t
Carrier Signal
t
BPSK Output
t
Demodulated Output

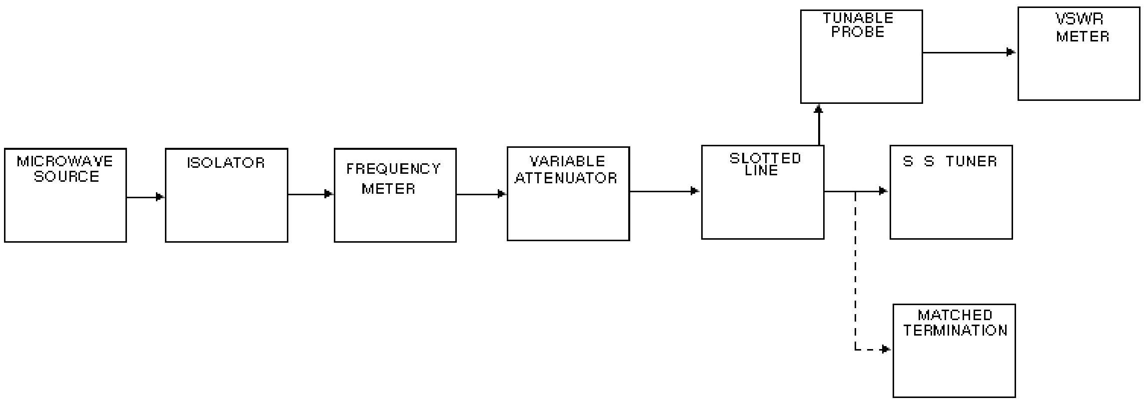
Experiment 6. Measurement of Frequency, Guide Wavelength, Power, VSWR & Attenuation in a Microwave Test Bench

AIM: To measure the frequency, guide wavelength, power, VSWR and attenuation in a microwave test bench.

Apparatus: Microwave test bench and accessories

A. Measurement of Frequency & Guide Wave Length

Block diagram



Theory:

A microwave test bench consists of a microwave source (modulation option also exists), isolator (to protect the source form reflected power), variable attenuator (to control the amount of power launched into the waveguide), a frequency meter (to measure the frequency), a slotted line (which is a waveguide cut in the longitudinal edge to provide for measurements), Slide Screw tuner (which is used to create various load impedances), a tunable probe along with a crystal detector and VSWR meter to measure power or VSWR values for a given load value.

The VSWR value can be used to further calculate the magnitude of the reflection coefficient of a load. The phase shift w.r.t. to a reference plane can be measured by moving the tunable probe along the slotted section of the waveguide and the phase of the reflection coefficient can also be calculated. In practice if unknown load impedance has to be measured, we can find out the load impedance value using the reflection coefficient measurements.

A VSWR meter is a sensitive high gain, high Q, low noise voltage amplifier tuned normally at a fixed frequency of 1 KHz at which the microwave signal is modulated. The output of the tunable probe is connected to a crystal detector. The output of this detector is amplified and given as input to the VSWR meter. The detector works in the square law region. The meter directly gives the VSWR reading V_{max}/V_{min} for an input of V_{min}, after the meter is adjusted to unity VSWR for an Input corresponding to V_{max}.

Procedure:-

1. Set up the components and equipments as shown in the block diagram above. First use a matched termination and disconnect the S. S. Tuner. (or retain the S. S. Tuner and disconnect the matched termination)
2. Set up variable attenuator at minimum attenuation position.
3. Keep the control knobs of VSWR Meter as below: -

Range - 50 dB

Input switch - Crystal low impedance
Meter switch - Normal position
Gain (Coarse & Fine) - Mid position

4. Keep the Control Knobs of Klystron power supply as below
   Beam Voltage - OFF

   Mod-switch - AM

   Beam voltage Knob Fully anticlockwise
   Reflector Voltage - Fully clockwise
   AM-Amplitude Knob - Around fully clockwise
   AM-Frequency Knob - Around mid position.
5. Switch 'ON' the Klystron power supply, VSWR Meter/CRO.
6. Switch 'ON' the beam voltage switch and set beam voltage at 250V with the help of beam voltage knob.
7. Adjust the reflector voltage to get some deflection in VSWR Meter.
8. Maximize the deflection with AM amplitude and frequency control knob of power supply. The following controls can also control the deflection in the VSWR meter.
   a. Plunger of Klystron Mount can also be tuned for maximum deflection.
   b. The reflector voltage knob can be tuned for minimum deflection.
   c. The probe in VSWR meter can be tuned for maximum deflection in VSWR Meter.
9. Tune the frequency meter knob to get a 'dip' on the VSWR Scale and note down the frequency directly from frequency meter. (Note: This dip can also be observed in a CRO instead of VSWR meter)
10. Now, replace the matched termination (if connected) with movable short, and detune the frequency meter.
11. Move the probe along the slotted line. The deflection in VSWR meter will vary.
    Move the probe to be at a minimum deflection position, to get accurate reading. If necessary increase the VSWR meter range dB switch to a higher position. Note and record the probe position. These positions can also be noted with the help of a CRO instead of VSWR meter.
12. Move the probe to next minimum position and record the probe position again.
13. Calculate the guide wavelength as twice the distance between two successive minima positions, obtained from above. These calculations can also be done using successive maxima positions. Tabulate the results of the positions as shown in the table below.
14. Measure the waveguide inner broad dimension 'a' which will be around 22.86 mm for X-band.
15. Calculate the frequency by using the equations given after the table.

Slotted –line carriage position (M)	Distance between successive minima	Distance between successive maxima
M1=	D1=M1-M3=	d1=m2-m4=
m2=		
M3=	D2=M3-M5=	d2=m4-m6=
m4=		
M5=	D3=M5-M7	d3=m6-m8=
m6=		
M7=	D4=M7-M9	
m8=		
M9=		

*m –maxima position M—minima position

Observations & Calculations:

Operating frequency f (using frequency meter) = D_{av} = (D1 + D2 + D3 + --------+ Dn) / n = ________________________

d_{av} = (d1 + d2 + d3 + --------+ dn) / n = _________________

Guide wavelength (λ_g) =2[(D_{av} + d_{av}) / 2]

Cut-**off wavelength** (λ_C) = (2 a) = __________________

where a = broader dimension of waveguide and m = 1 for TE_{10} mode

$$1/(\lambda_O)^2 = 1/(\lambda_C)^2 + 1/(\lambda_g)^2$$

λ_O = free-space wavelength = ___________________

Operating frequency f = c/ λ_O = __________

(Compare this value to the readin obtained from the frequency meter)

-

B. Measurement of VSWR and Power:

Using the set up shown in the block diagram previously (connect the S. S. Tuner), first move the probe along the slotted line to get maximum deflection in VSWR Meter.

1. Adjust VSWR meter gain control knob or variable attenuator until the meter indicates 1.0 on normal VSWR Scale.
2. Keep the entire control knob as it is, move the probe to the next minimum position. Avoid moving in both the directions on the slotted line. Continue in a single direction when the measurement is being performed. Read the VSWR on scale and power in dB from the VSWR meter. {Note: If the maxima, is set to 1 (using gain control on VSWR meter, the reading seen on the VSWR meter for the minima is directly the VSWR value.}
3. Repeat the above step for change of S.S. Tuner Probe depth and record the corresponding VSWR.
4. If the VSWR is between 3.2 and 10, change the range dB switch to next higher position and read the VSWR on second VSWR scale of 3 to 10.

C. Measurement of Attenuation:

1. Fix a particular attenuation value in the attenuator. Then Move the probe along the slotted line to get maximum deflection in VSWR Meter. Adjust the gain control knob of the VSWR meter if necessary. Note the power value (P_1) in dB as seen from the VSWR meter.
2. Vary the variable attenuator knob by x-dB. Without disturbing any settings on the VSWR meter, note the power value for this reading on the attenuator (P_2).
3. Compute P_2-P_1. This must be equal to x-dB, which is the attenuation of the attenuator.
4. The above experiment can be done to calculate the attenuation of a given unknown attenuator, AUT. (AUT: Attenuator under Test. This is different from the variable attenuator included in the set up) In this case find P_1 without the AUT and then find P_2 with AUT. Find the difference between both the power values to find the attenuation value.

D. Measurement of Relative and Absolute Power:

1. In the above measurement of attenuation (C), if P_1 is made as a reference power level, the power output P_2 is the relative power output level due to the device, AUT or due to the extra x-dB setting in the variable attenuator. If by some other method (ex: using power meter etc.), we know the exact power level for the case of P_1, then P_2 can be mapped to its absolute value and the absolute power measurement can be done.

Other information:

<u>VSWR (Voltage standing wave Ratio) Meter:-</u>

A gain control can be used to adjust the reading to the desired range. The over all gain is nearly 125dB which can be altered in step of 10dB.

There are three scales on the VSWR meter,

- When the VSWR is between 1 & 4, reading can be taken from the top SWR Normal scale.
- For VSWR between 3.2 & 10, bottom of SWR NORMAL scale is used.
- When the VSWR is less than 1.3, a more accurate reading can be taken by selecting the expanded scale, graduated from 1 to 1.2.
- The third scale at the bottom is graduated in dB.

Slotted line carriage

Slotted line carriage cantinas a co-axial E-field probe which penetrates inside a rectangular wave guide slotted section or a co-axial slotted line section from the outer wall & is able to traverse a longitudinal narrow slot. The longitudinal slot is cut along the centre of the waveguides broad wall or alone the outer conductor of the co-axial line over a length of 2-3 wave lengths were the electric current on the wall does not have any transverse component. The slot should be narrow enough to avoid any distortion in the original field inside the waveguide. the two ends of the slot is tapered to zero width for reducing the effect of discontinuity. The probe is made at move longitudinally at a constant small depth to achieve a uniform coupling Co-efficient between the electric field inside the line of the probe current at all positions. The probe samples the electric field which is proportional to the probe voltage. This unit is primarily used for the determination of locations of the voltage sending wave maximum & minimum along the line. The probe carriage contains a stub tunable Co-axial probe detector to obtain a low frequency modulating signal output to a slope or VSWR meter. The probe should be very thin compared to the wave length & the depth also should be small enough to avoid any field distortion. The slotted line with tunable probe detector is used to measure VSWR & Standing wave pattern, wavelength, impedance, reflection Co-efficient & Return loss measurements by the minima shifts method.

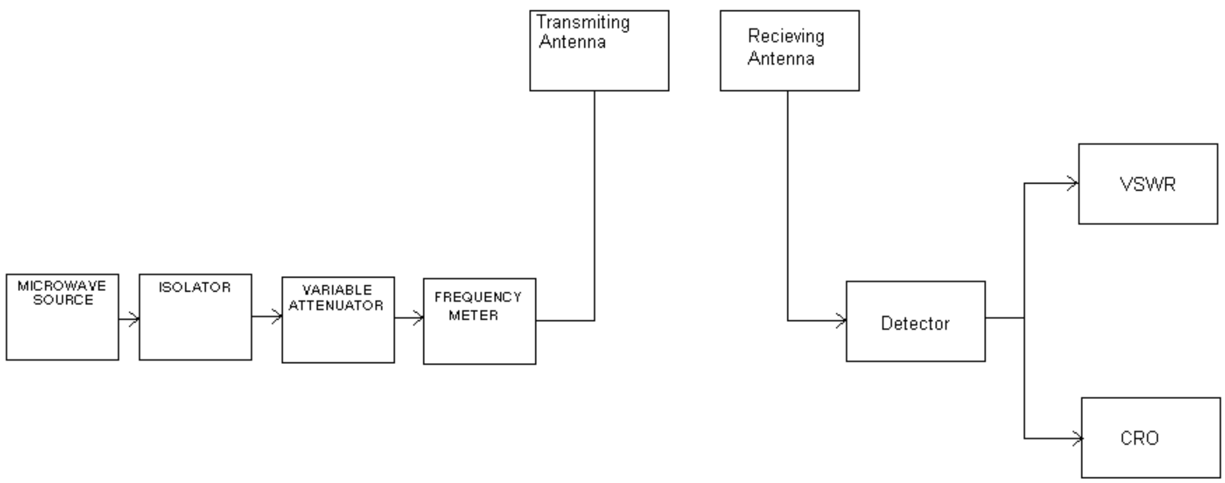
Experiment 7. Measurement of Directivity and Gain of Microstrip patch antenna,dipole & Yagi antenna.

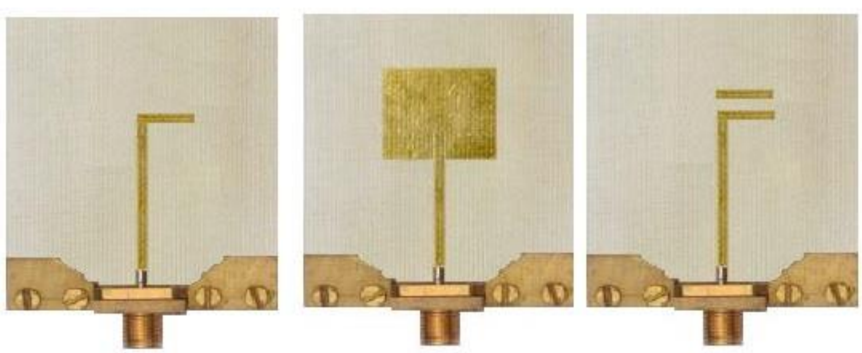
AIM:- To measure the Directivity and gain of antenna standard dipole, Microstrip patch antenna & Yagi antenna.

COMPONENTS:

Power supply
Microwave source Different type transmitting & receiving antennas
Frequency meter
Active filter
VSWR

CIRCUIT DIAGRAM:





THEORY:

If a transmission line propagating energy is left open at one end, there will be radiation from this end. The Radiation pattern of an antenna is a diagram of field strength or more often the power intensity as a function of the aspect angle at a constant distance from the radiating antenna. An antenna pattern is of course three dimensional but for practical reasons it is normally presented as a two dimensional pattern in one or several planes. An antenna pattern consists of several lobes, the main lobe, side lobes and the back lobe. The major power is concentrated in the main lobe and it is required to keep the power in the side lobes arid back lobe as low as possible. The power intensity at the maximum of the main lobe compared to the power intensity achieved from an imaginary omni-directional antenna (radiating equally in all directions) with the same power fed to the antenna is defined as gain of the antenna.

As we know that the 3dB beam width is the angle between the two points on a main lobe where the power intensity is half the maximum power intensity. When measuring an antenna pattern, it is normally most interesting to plot the pattern far from the antenna. It is also very important to avoid disturbing reflection. Antenna measurements are normally made at anechoic chambers made of absorbing materials. Antenna measurements are mostly made with unknown antenna as receiver. There are several methods to measure the gain of antenna. One method is to compare the unknown antenna with a standard gain antenna with known gain. Another method is to use two identical antennas, as transmitter and other as receiver. From following formula the gain can be calculated.

$$P_{r} = \frac{P_t \lambda^2}{}$$

Where

P_t is transmitted power
P_r is received Power,
G_1, G_2 is gain of transmitting and receiving antenna
S is the radial distance between two antennas
lo is free space wave length.

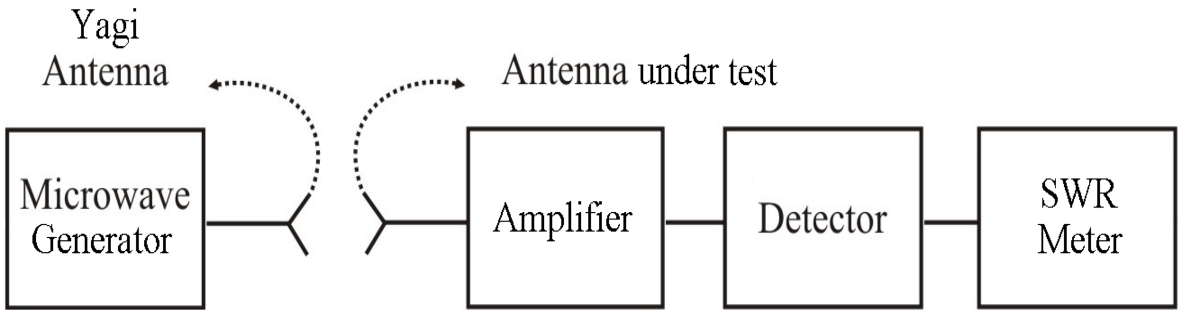
If both, transmitting and receiving antenna are identical having gain G then above equation becomes.

$$P_r = \frac{P_t \lambda_0^2 G^2}{(4\pi S)^2},$$

$$G = \frac{4\pi S}{\lambda_0} \sqrt{\frac{P_r}{P_t}}$$

In the above equation P_t, P_r and S and lo can be measured and gain can be computed. As is evident from the above equation, it is not necessary to know the absolute value of P_t and P_r only ratio is required which can be measured by SWR meter.

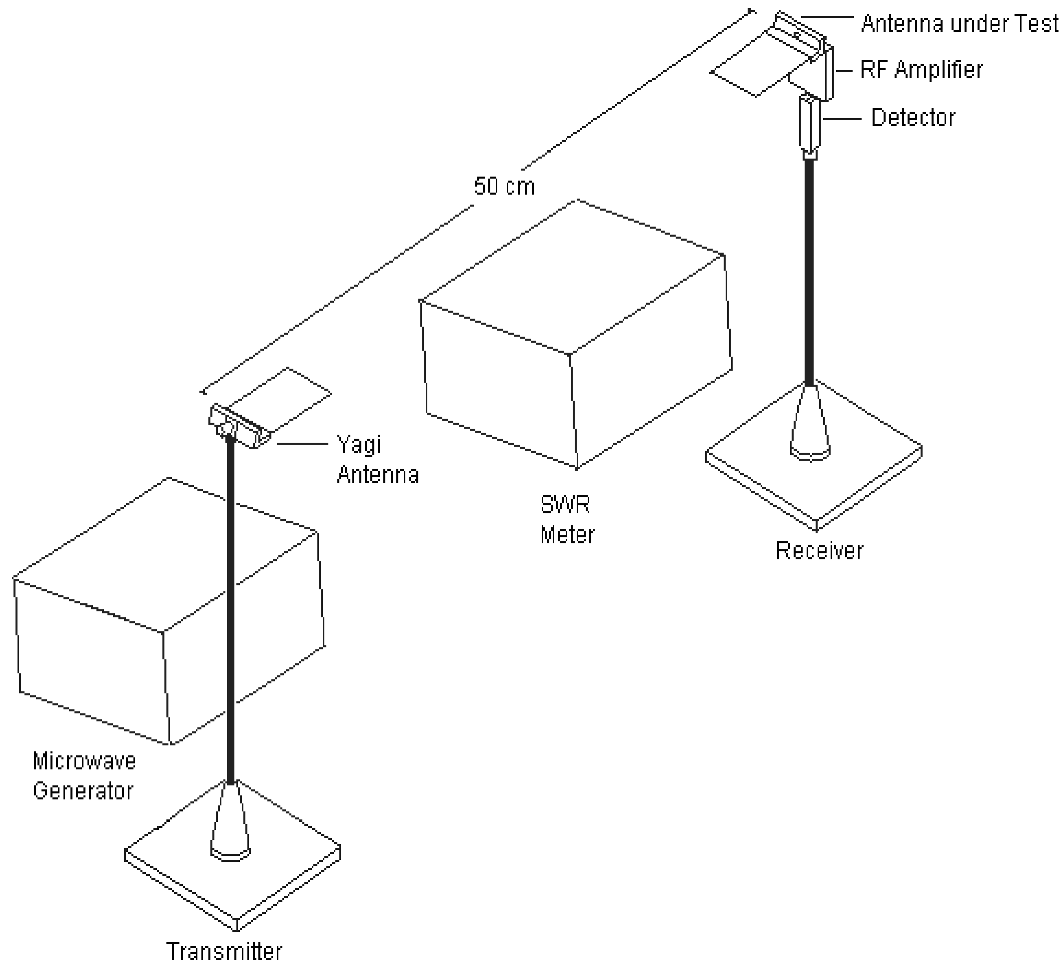
SETUP FOR DIRECTIVITY MEASUREMENT



PROCEDURE:

Directivity Measurement:

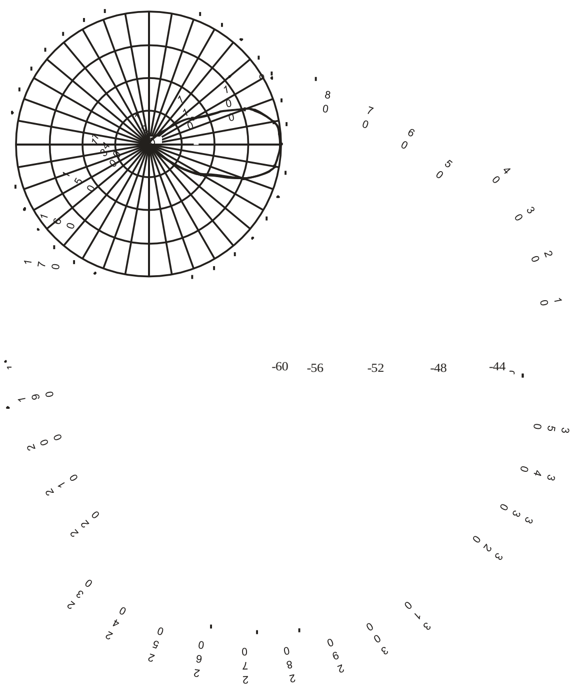
1. Connect a mains cord to the Microwave Generator and SWR Meter.
2. Now connect a Yagi antenna in horizontal plane to the transmitter mast and connect it to the RF Output of microwave generator using a cable (SMA to SMA).
3. Set both the potentiometer (Mod Freq & RF Level) at fully clockwise position.
4. Now take another Yagi antenna and RF Amplifier from the given suitcase.
5. Connect the input terminal of the Amplifier to the antenna in horizontal plane using an SMA (male) to SMA (female) L Connector.
6. Now connect the output of the Amplifier to the input of Detector and mount the detector at the Receiving mast.
7. Connect one end of the cable (BNC to BNC) to the bottom side of receiving mast, and another end to the input of SWR meter.
8. Now set the distance between Transmitter (feed point) and the receiver (receiving point) at half meter.



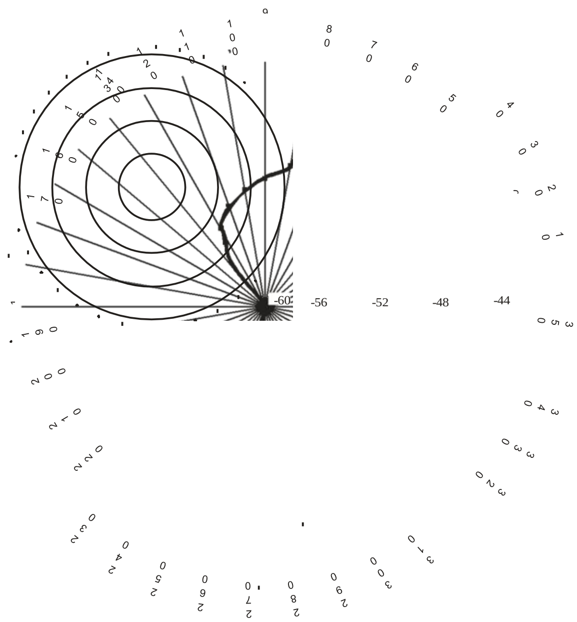
9. Now set the receiving antenna at zero degree (in line of Transmitter) and Switch on the power supply for Microwave Generator, SWR Meter. Also connect DC Adapter of RF Amplifier to the mains.
10. Select the transmitter for internal AM mode and press the switch “RF On”.
11. Select the range switch at SWR meter at – 40dB position with normal mode.
12. Set both the gain potentiometers (Coarse & Fine) at fully clockwise position and input select switch should be at 200 Ohm position. In case if reading is not available at – 40dB range then press 200 kOhm (Input Select) to get high gains reading.
13. Now set any value of received gain at – 40dB position with the help of -
    - Frequency of the Microwave Generator.
    - Modulation frequency adjustment.
    - Adjusting the distance between Transmitter and Receiver.
14. With these adjustments you can increase or decrease the gain.
15. Mark the obtained reading on the radiation pattern plot at zero degree position.
16. Now slowly move the receiver antenna in the steps of 10 degree and plot the corresponding readings.
17. This will give the radiation pattern of the antenna under test.
18. Directivity of the antenna is the measures of power density an actual antenna radiates in the direction of its strongest emission, so if the maximum power of rectivity will be ....dB at ....Degree.
19. In the same way you can measure the directivity of the Dipole antenna.

20. For directivity measurement of the transformer fed Patch antenna connect transmitter Yagi antenna in the vertical plane (Patch Antenna is vertically polarized). Since it is comparatively low gain antenna distance can be reduced between transmitter and receiver.

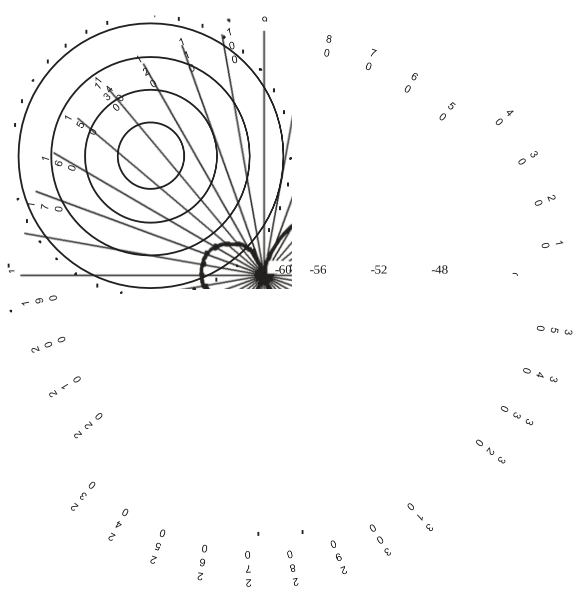
Radiation Patterns of Different Antennas:



Yagi Antenna



Patch Antenna



Dipole Antenna

Gain Measurement:

1. Connect a power cable to the Microwave Generator and SWR Meter
2. Now connect a Yagi antenna in horizontal plane to the transmitter mast and connect it to the RF Output of microwave generator using a cable (SMA to SMA).
3. Set both the potentiometer (Mod Freq & RF Level) at fully clockwise position.
4. Now take another Yagi antenna from the given suitcase.
5. Connect this antenna to the detector with the help of SMA (male) to SMA (female) L Connector.
6. Connect detector to the receiving mast.
7. Connect one end of the cable (BNC to BNC) to the bottom side of receiving mast, and another end to the input of SWR meter.
8. Now set the distance between Transmitter (feed point) and the receiver (receiving point) at half meter.
9. Now set the receiving antenna at zero degree (in line of Transmitter) and Switch on the power from both Generator & SWR Meter.
10. Select the transmitter for internal AM mode and press the switch "RF On".
11. Select the range switch at SWR meter at – 40dB position with normal mode.
12. Set both the gain potentiometers (Coarse & Fine) at fully clockwise position and input select switch should be at 200 Ohm position. In case if reading is not available at – 40dB range then press 200 kOhm (Input Select) to get high gain reading.
13. Now set the maximum gain in the meter with the help of following
    - Frequency of the Microwave Generator
    - Modulation frequency adjustment.
    - Adjusting the distance between Transmitter and Receiver.
14. Measure and record the received power in dB. Pr = ...dB
15. Now remove the detector from the receiving end and also remove the transmitting Yagi antenna from RF output.
16. Now connect the RF output directly to detector without disturbing any setting of the transmitter (SMA-F to SMA-F connector can be used for this).
17. Observe the output of detector on SWR meter that will be the transmitting power Pt.

    Pt = ...dB
18. Calculate the difference in dB between the power measured in step 14 and 17 which will be the power ratio Pt/Pr.

    Pt/Pr =........................

    Pr/Pt =........................
19. Now we know that the formula for Gain of the antenna is:

$$G = \frac{4\pi S}{\lambda_0}\sqrt{\frac{P_r}{P_t}}$$

Where:

P_t is transmitted power
P_r is received Power,
G is gain of transmitting/receiving antenna (since we have used two identical antennas)
S is the radial distance between two antennas
lo is free space wave length (approximately 12.5cm).

20. Now put the measured values in the above formula and measure the gain of the antenna which will be same for both the antennas. Now after this step you can connect one known gain antenna at transmitter end and the antenna under test at receiver end, to measure the gain of the antennas.
21. Gain can be measured with the help of absolute power meter also (Recommended Model NV105). For this, detector will not be used and directly the power sensor can be connected to both the ends as described earlier.

RESULT:- Patch Antenna:
Gain
Directivity
Dipole Antenna:
Gain
Directivity
Yagi Antenna:
Gain
Directivity

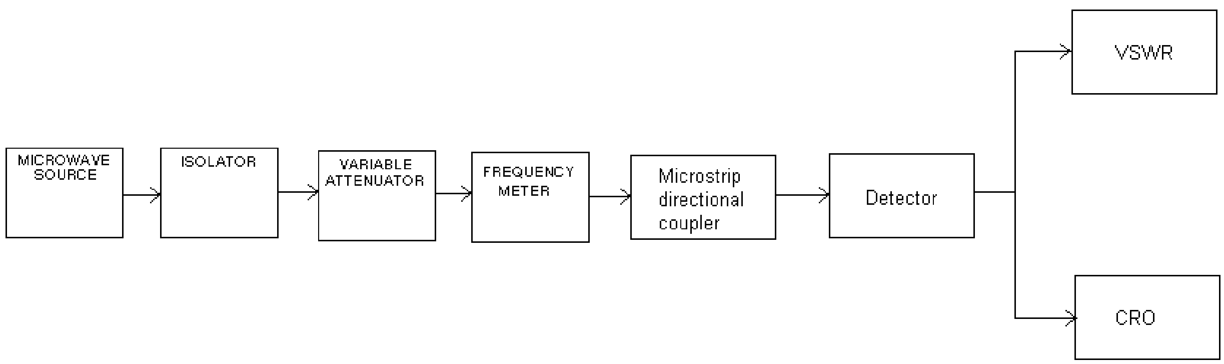
8a Determination of Coupling and Isolation Characteristics of Micro strip Directional Coupler

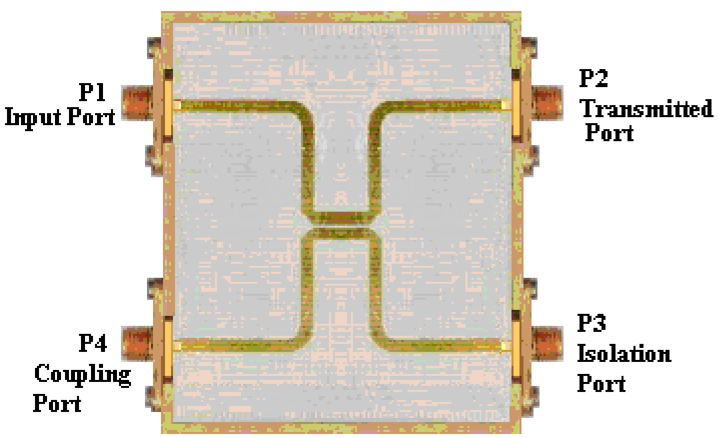
AIM:- To determine of coupling and isolation characteristics of Micro strip directional coupler.

COMPONENTS:

Power supply, Microwave source, 50 ohm transmission line, Branch line coupler, 11 line in coupler, Frequency meter , Active filter, VSWR

CIRCUIT DIAGRAM:





THEORY:

A directional coupler is a four port device, where in port 1 called input port is given input power. The output port is in the main line, which gets most of the input port. The coupled takes very small amount of input power and serves as a reference measurement in many application. The isolated power should not receive any power ideally. The port is isolated from the input port. Directional coupler is used after the antenna in a receiver circuit. This helps in monitoring reflected power. In a transmit circuit, antenna is connected after the directional coupler. This helps in monitoring incident and reflected power to the antenna and from the antenna.

PROCEDURE:-

1. Set up the system as shown in figure.
2. Keeping the voltage at minimum, switch on the power supply.
3. Insert a 50Ω transmission line and check for the output ate the end of the system.
4. Vary the power supply voltage at minimum, switch on the power supply.
5. Note down the output for different VCO frequency.
6. Replace the 50Ω transmission line with branch line coupler.
7. Check the output at ports 2,3 & 4.
8. Calculate insertion loss, coupling factor & isolation using the formulae given.
9. Repeat the experiment for a parallel line in coupler.

RESULT:-

<u>Using CRO</u>

Coupling factory = $10 \log \frac{p_1}{p_4}$

Directivity = $10 \log \frac{p_4}{p_3}$

Isolation loss = $10 \log \frac{p_1}{p_3}$

Insertion loss = $10 \log \frac{p_1}{p_2}$

<u>Using VSWR</u>

Coupling factory = $P_1 - P_4$

Directivity = $P_4 - P_3$

Isolation loss = $P_1 - P_3$

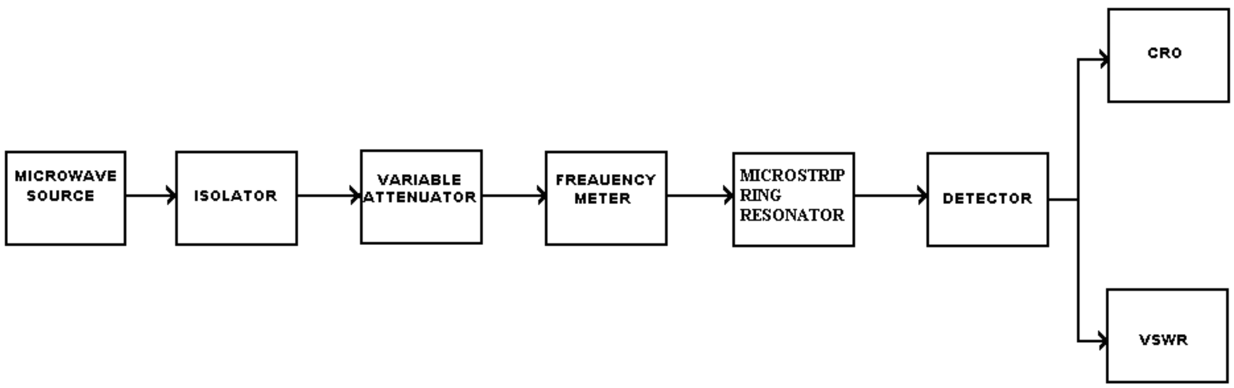
Insertion loss = $P_1 - P_2$

8b Measurement Of Resonance Characteristics Of A Microstrip Ring Resonator

AIM:- To measure the resonance characteristics of a Microstrip ring resonator & output.

COMPONENTS: Power supply, Microwave source, Frequency meter, Variable Attenuator, Microstrip ring resonator, Coaxial Wave guide Adapter, VSWR, CRO, Coaxial Cable, Cooling Fan.

CIRCUIT DIAGRAM:



THEORY:

This is a simplest printed planar resonator at microwave frequencies using microstrip technology. It is used in making filters (Band pass or Band stop) to some extent as an antenna element. It is a two port that it resonates the transmission characteristics of this two-port network show that is resonates at a center frequency where it has zero loss. But for other frequencies this loss increases.

The open-end effect encountered in a rectangular resonator at the feeding gaps can be minimized by forming the resonator as a closed loop. Such a resonator is called a ring resonator. Figure 7.4 shows the layout of a ring resonator along with the input and output feed lines. As explained in the case of the rectangular resonator, the coupling can be loose or tight depending on the gap width. Resonance is established when the mean circumference of the ring is equal to integral multiples of guide wavelength.

$$2pr_0 = nl = \frac{nn_0}{f_0\sqrt{e_{ef}}}, \qquad \text{for n = 1, 2,3…..} \qquad (7.6)$$

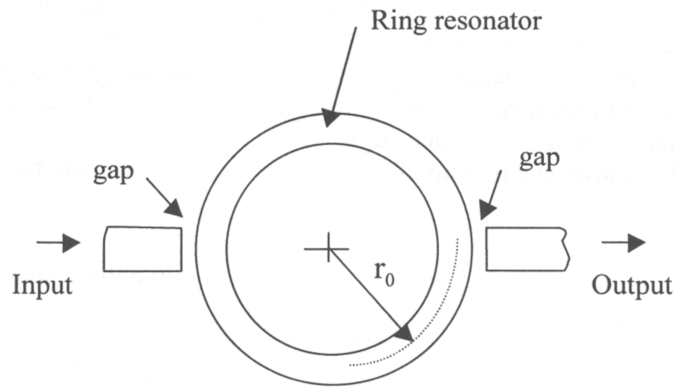


Fig. : Layout of a microstrip ring resonator with input and output lines

In Fig 1, r_o is the mean radius of the ring and *n* is the mode number. The other symbols are defined in section 7.1. The ring has the lowest order resonance for *n* = 1. For this mode, the field maxima occur at the two coupling gaps and nulls occur at 90° locations from the coupling gaps.

Equation (7.6) does not take into account the coupling effects at the gaps. As explained in section 7.1 for the rectangular resonator, the resonance frequency of the ring is also affected by the coupling gap. The deviation from the intrinsic resonant frequency, however, is much smaller than in the case of a rectangular resonator.

Coupling can be increased upto some extent by making the feeding lines as curved shown in fig.7.5.

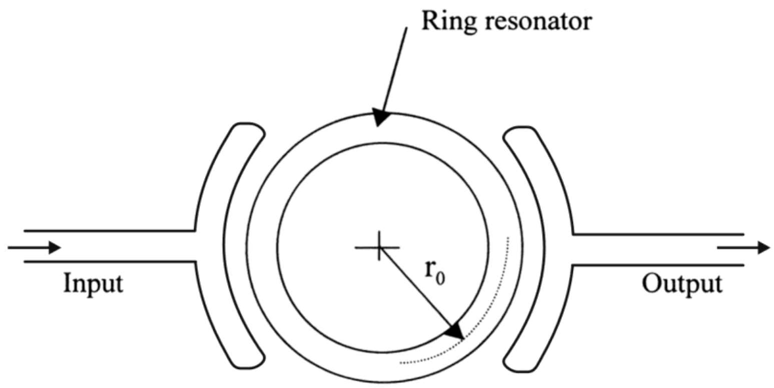


Fig. Layout with curved input and output feed line.

TABULAR COLUMN:

VOLTAGE (V)	FRE

PROCEDURE:-

1. Set up the system as shown in figure.
2. Keeping the voltage at minimum, switch on the power supply.
3. Insert a 50ohm transmission line and check for the output at the end of the system using a VSWR.
4. Vary the power supply voltage and check the output for different VCO frequency.
5. Replace the 50ohm transmission line with ring resonator.
6. Vary the supply voltage, tabulate VCO frequency vs output.
7. Plot a graph frequency vs output & find the resonant frequency.

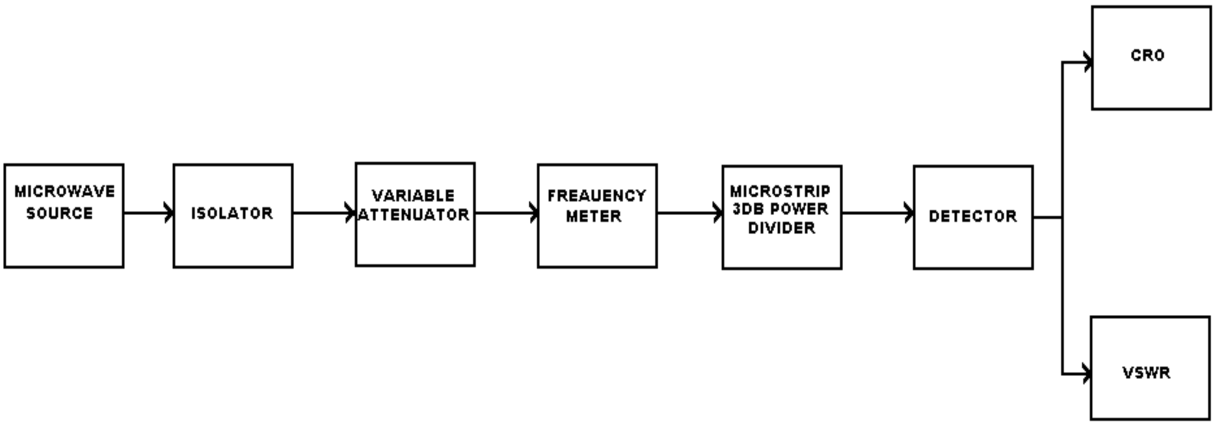
RESULT:- The resonant frequency fr =

8c. Measurement Of Power Division And Isolation Characteristics Of Microstrip 3db Power Divider

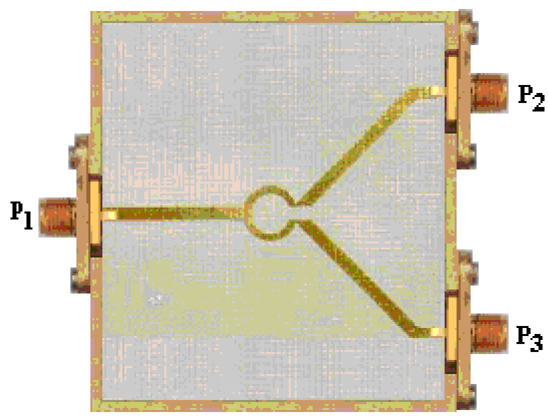
AIM: - To measure the power divider and isolation characteristics of Microstrip 3dB power divider.

COMPONENTS: Power supply, Microwave source, Frequency meter, Variable Attenuator, Micro strip 3dB power divider, Coaxial Wave guide Adapter, VSWR, CRO, Coaxial Cable, Cooling Fan.

BLOCK DIAGRAM:



MICROSTRIP 3DB POWER DIVIDER



THEORY:

A power divider is 3 port network. Input port is given power. This power is then divided into 2 parts and each of these powers is output to the output ports. So output power in any output port is less then the input port. If the power divider is ideal, two output powers sum is equal to the input power. A power divider can also act as a power combiner. If 2 power inputs are given to the output ports of power divider, the sum power is emitted from the input port of power divider. In communications often, we need power division, either in a receiver circuit or transmitter circuit. This device is the used. In optical domain, we have power dividers/couplers similar to the power dividers in the microwave region. A 3dB power divider divides input into two equal values.

PROCEDURE:-

1. Set up the system as shown in figure.

2. Keeping the voltage at minimum, switch on the power supply.
3. Insert a 50ohm transmission line and check for the output at the end of the system using a VSWR.
4. Vary the power supply voltage and check the output for different VCO frequency.
5. Keep the VCO frequency constant note down the output. This value can be taken as the input to the power divider.
6. Replace the 50ohm transmission line with the Wilkinson power divider.
7. Tabulate the output at parts 2 and 3.
8. Calculate insertion loss and coupling factor in each coupled arm.

RESULT:- with VSWR meter: Isolation

(dB) $=P_3-P_2$

Power division arm 3(dB)= P_1-P_3

Power

DIGITAL COMMUNICATION LAB VIVA Questions

1. What is multiplexing?
2. What are different types of multiplexing?
3. What is TDM?
4. What do you mean by FDM?
5. What is Amplitude shift Keying (ASK) ?
6. What are different types of digital modulation?
7. What is Phase shift keying (PSK)?
8. What is Frequency shift keying (FSK) ?
9. What is Binary Phase shift Keying (BPSK) ?
10. What is bandwidth of BPSK signal?
11. What do you mean by Differential Phase Shift Keying (DPSK)?
12. What is Bandwidth of DPSK?
13. Show the phase change in DPSK at the digital input 10111101.
14. What is PSK?
15. Why JK flip-flop is being used in this type if Keying?
16. What are the advantages over other types of keying?
17. What is a GUNN DIODE?
18. How would you find the propagation loss for two different known length and voltage of optical fiber.
19. What is the used formula to find out the bending loss for an optical fiber?
20. Differentiate ASK and FSK.?
21. What is meant by PLL?
22. Applications of PLL.?
23. Define pre-emphasis & de-emphasis?
24. Why is ASK called as ON-OFF keying?
25. What are the advantages of digital transmission?
26. What is meant by frequency synthesizer?
27. Applications of frequency synthesizer?
28. What is meant by line coding?
29. Define polar encoding?
30. Define bipolar encoding?
31. Define unipolar encoding?
32. Define Manchester encoding?
33. What are the two function of fast frequency hopping?
34. What are the features of code Division multiple Accesses?
35. What is called multipath Interference?
36. Explain ARQ,FEC.HARQ?
37. What is sampling?
38. State sampling theorem ?
39. What is aliasing?
40. How to avoid aliasing effect ?
41. Define Nyquist Criteria.
42. Purpose of sampling?
43. what is DTV?
44. How many digital television terrestrial broadcasting standards (DTTB) are used?
45. Difference in HDTV and SDTV?
46. What is aperture effect?
47. How aperture effect can be corrected?
48. what is pulse stuffing?

49. Explain nysquist criterion and interval?
50. Which kind of multiplexing used in digital communication?
51. Define Aperture effect . How it can be avoided
52. Why is Sample & Hold circuit used?
53. An Analog signal is given as
X(t) = 3cos 50pt + 10 sin 300 pt + cos 100pt
Calculate Nyquist rate
54. The spectrum of a signal g(t) has bandwidth 0.8KHZ and centered around 10 KHZ. Find Nyquist Rate
55. why pm is better than fm ?
56. Difference in digital and discrete signals?
57. why we need quantization?
58. A PCM system uses a uniform quantizer followed by a 7 bit encoder,with bit rate 50Mbps.What is the message bandwidth for system to operate satisfactorially?
59. Properties of Matched filter?
60. Difference between CDMA & TDMA
61. What is Pulse code modulation(PCM)?
62. How the performance of a PCM is corrupted?
63. What are the two types of Spread spectrum technique?
64. We need to send 3 bits at a time at a bit rate of 3Mbps.The carrier frequency is 10MHz.Calculate No of levels and Bandwidth
65. What is a Constellation Diagram?
66. Which modulation is a combination of other 2 schemes?
67. Name two principles by which spread spectrum achieves its goal
68. Which of the 3 multiplexing techniques is common for fibre optic links?
69. What is the minimum number of bits in a PN sequence if we use FHSS with a channel bandwidth of 5kHz and Bss=120kHz?
70. Which theorem gives Maximum Data Rate of a channel?

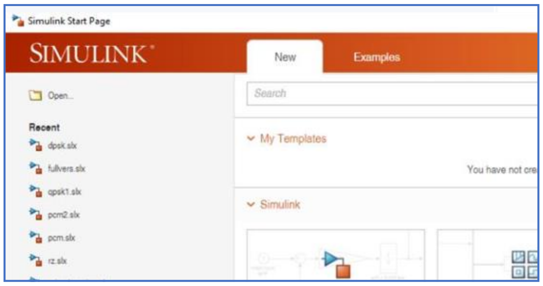
PART – B : SIMULATION EXPERIMENTS

Simulation Software : Simulink CONTENTS :

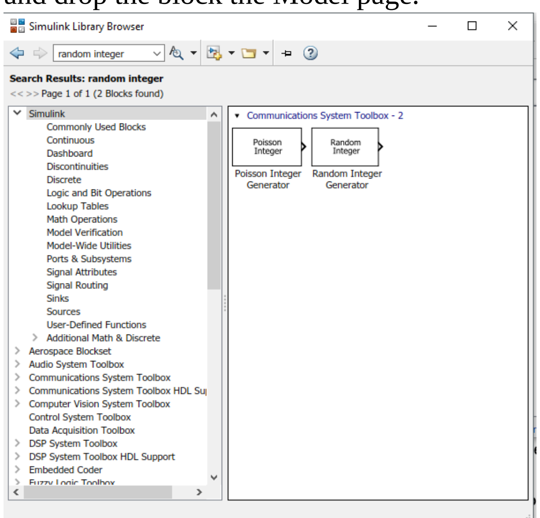
1. Simulate NRZ, RZ, Half-sinusoidal and raised cosine pulses and generate eye diagram for binary polar signalling.
2. Simulate the pulse code modulation and demodulation system and display the waveforms.
3. Computations of the Probability of bit error for coherent binary ASK, FSK and PSK for an AWGN Channel and Compare them with their Performance curves.
4. Digital Modulation Schemes i) DPSK Transmitter and receiver, ii) QPSK Transmitter and Receiver

GENERAL PROCEDURE:

i.Open Simulink. Select Blank Model.



ii. In the Model Page, Go to Library Browser . Type in the name of the block required in the search field. Drag and drop the block the Model page.



iii. Once all the blocks have been placed , rig up the connections by drawing wires. To do so just click on the open ends of each block and drag the wire to the required block.

iv. To change the properties of any block, double click on the block to get the parameters window. v. Save the design by going to File → Save.

vi. Go to Simulation → Run or Click on the ‘Play’ button. (Shortcut: CTRL + T).

vii. Double click on the CROs or Constellation Diagrams to view the waveforms.

viii. The properties of CROs or Constellation diagrams can be altered in the ‘Settings’ option after opening them.

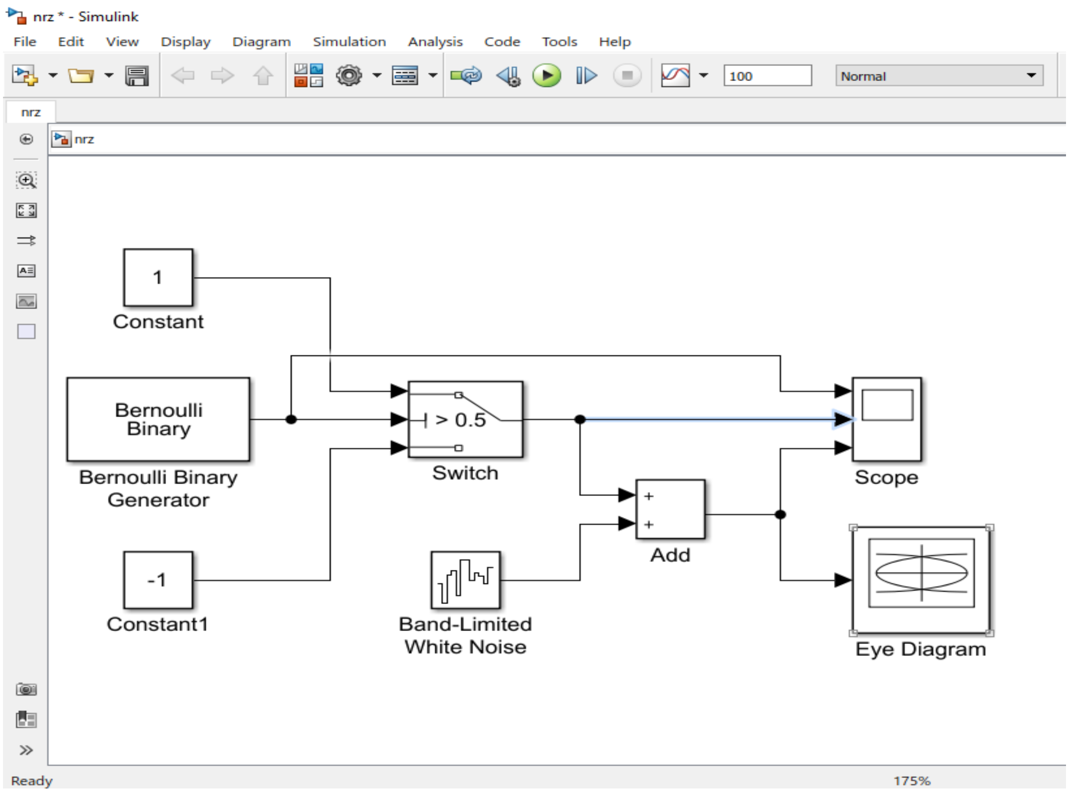
ix. To view the proper number of waveforms on the CRO, go to View → Layout and adjust accordingly.

x. Note that some waveforms in this manual can be different from the obtained, due to random number sequences used.

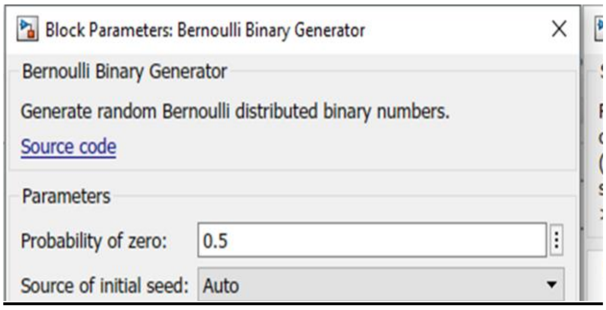
EXPERIMENT 1: Simulate NRZ, RZ, Half-sinusoidal and raised cosine pulses and generate eye diagram for binary polar signalling.

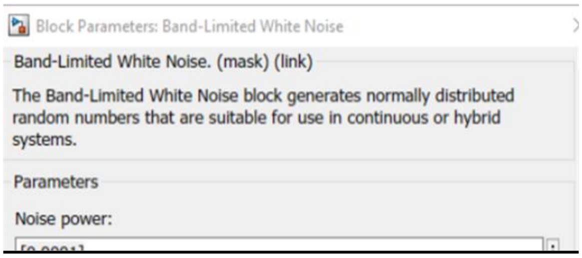
a. NRZ signalling (with Eye diagram for bipolar signalling):

CIRCUIT DIAGRAM:

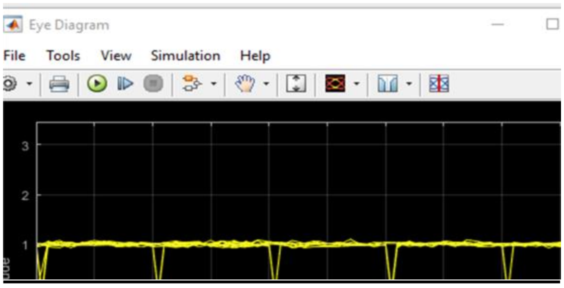


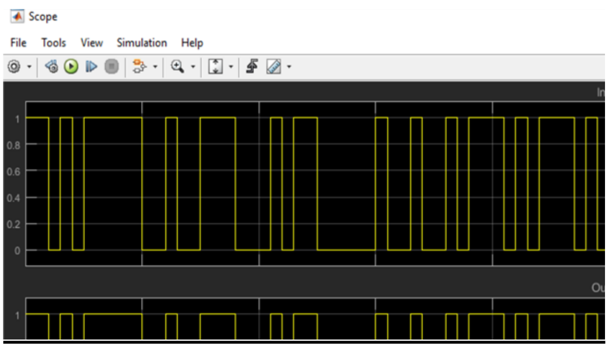
PARAMETERS:




Block Parameters: Band-Limited White Noise
Band-Limited White Noise. (mask) (link)
The Band-Limited White Noise block generates normally distributed random numbers that are suitable for use in continuous or hybrid systems.
Parameters
Noise power:

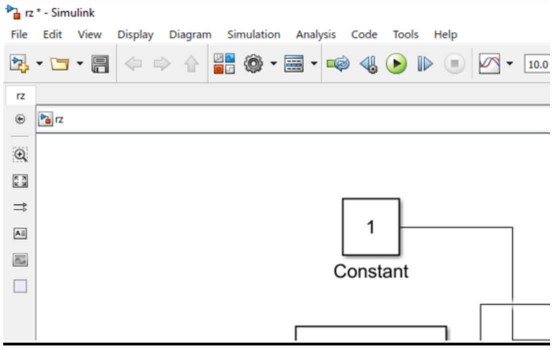
OUTPUTS:


Eye Diagram
File Tools View Simulation Help

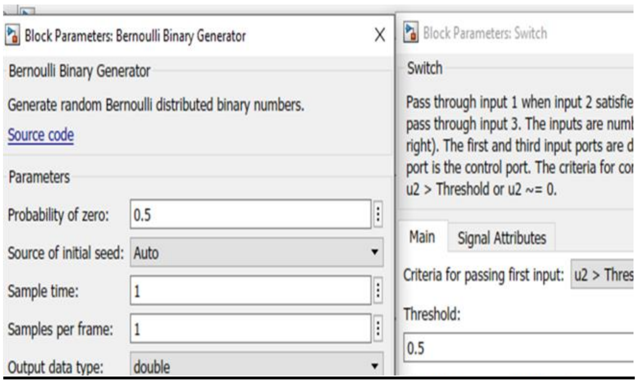

Scope
File Tools View Simulation Help

b. RZ Signalling:

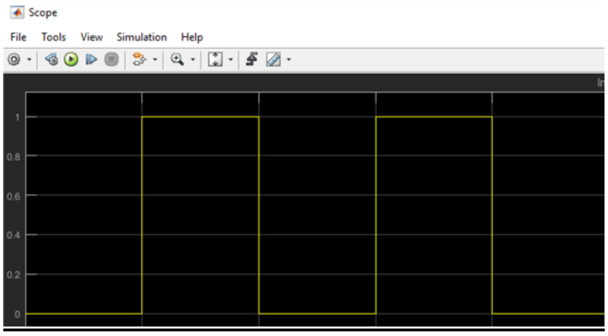
CIRCUIT DIAGRAM:



PARAMETERS:

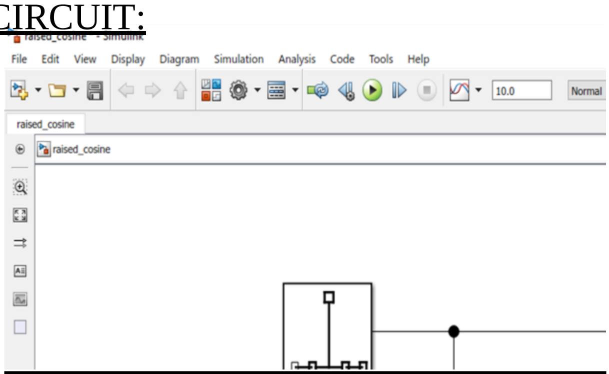


OUTPUT:

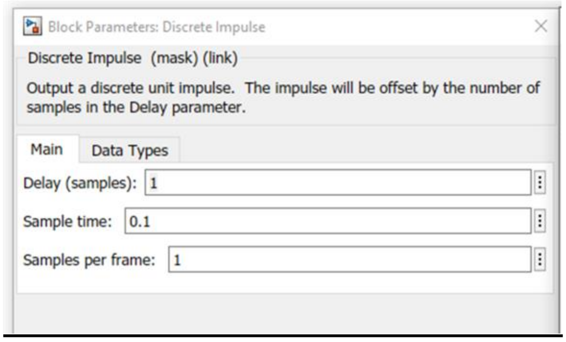


c. Raised Cosine Pulse:

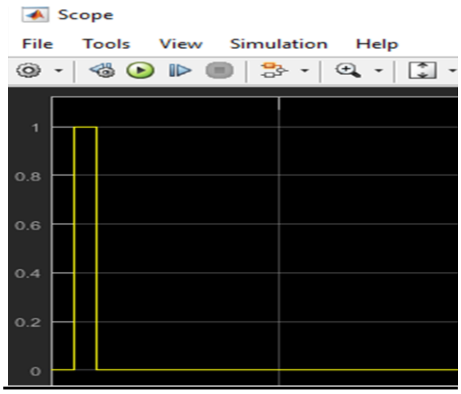
CIRCUIT:



PARAMETERS:

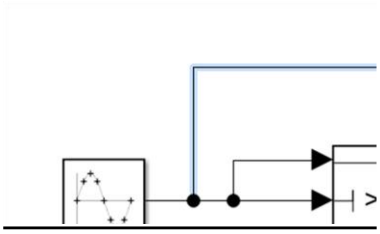


OUTPUT:

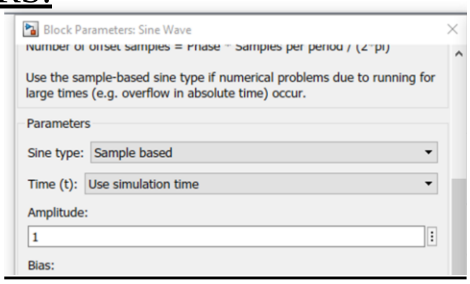


d. **Half Sinusoid:**

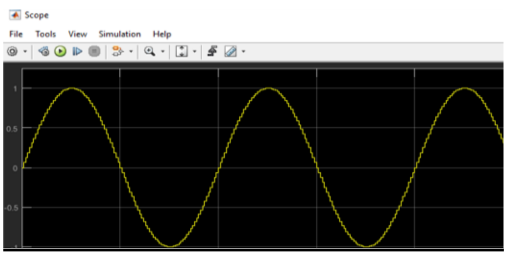
CIRCUIT:



PARAMETERS:

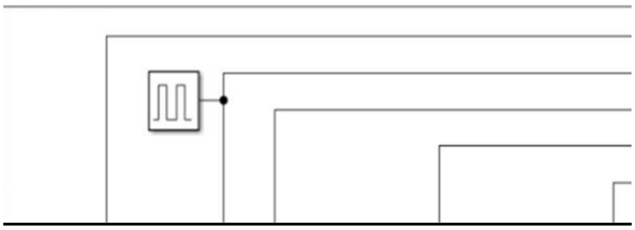


OUTPUT:

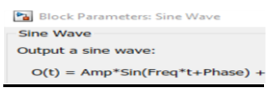


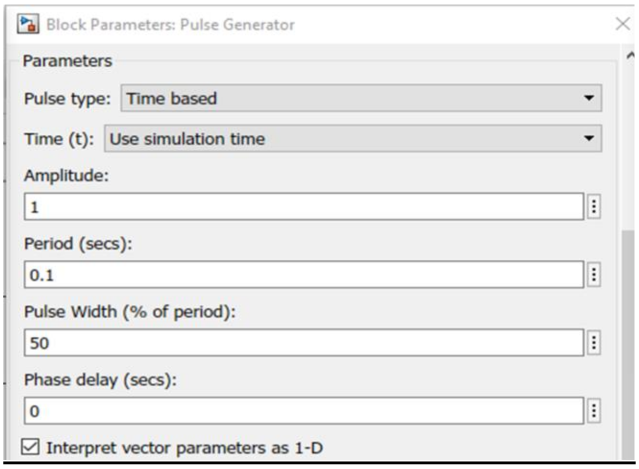
Experiment 2: Simulate the pulse code modulation an demodulation system and display the waveforms.

CIRCUIT:

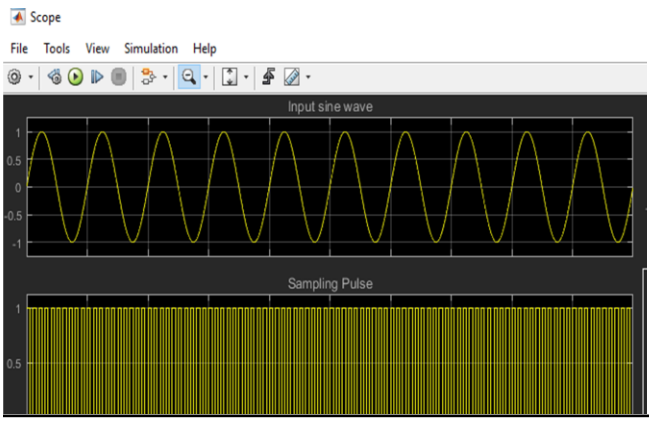


PARAMETERS:




Block Parameters: Pulse Generator
Parameters
Pulse type: Time based
Time (t): Use simulation time
Amplitude:
1
Period (secs):
0.1
Pulse Width (% of period):
50
Phase delay (secs):
0
Interpret vector parameters as 1-D

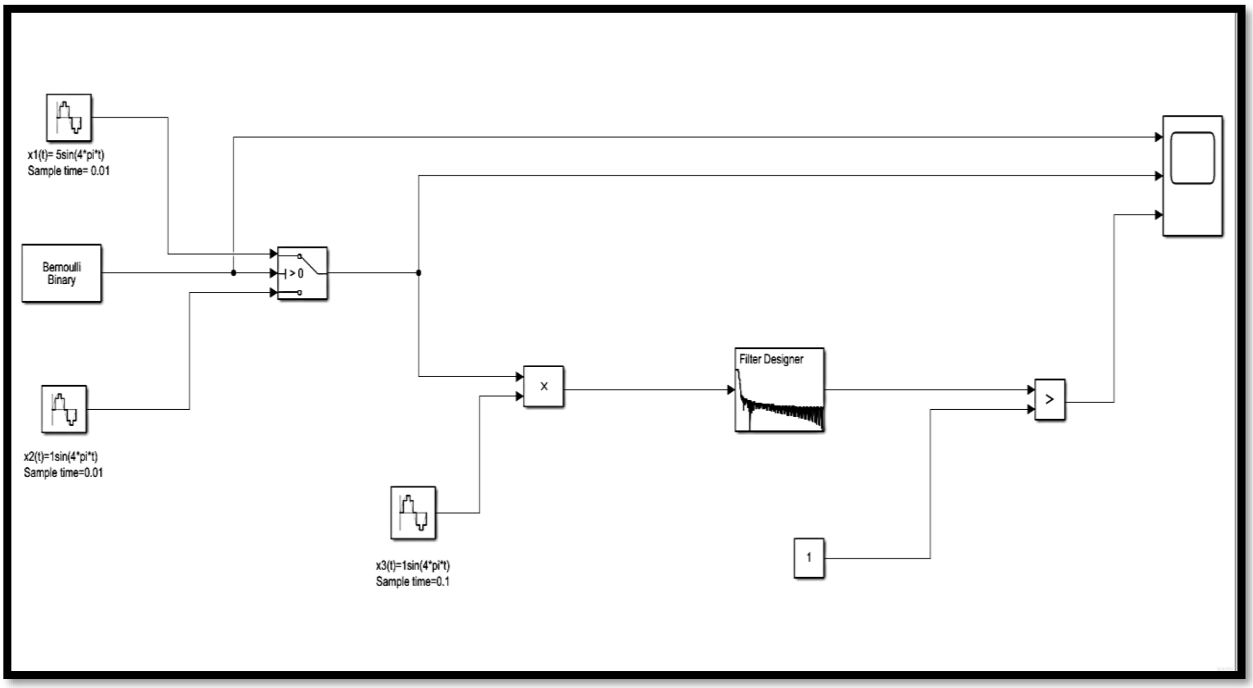
OUTPUT:


Scope
File Tools View Simulation Help
Input sine wave
1
0.5
0
-0.5
-1
Sampling Pulse
1
0.5

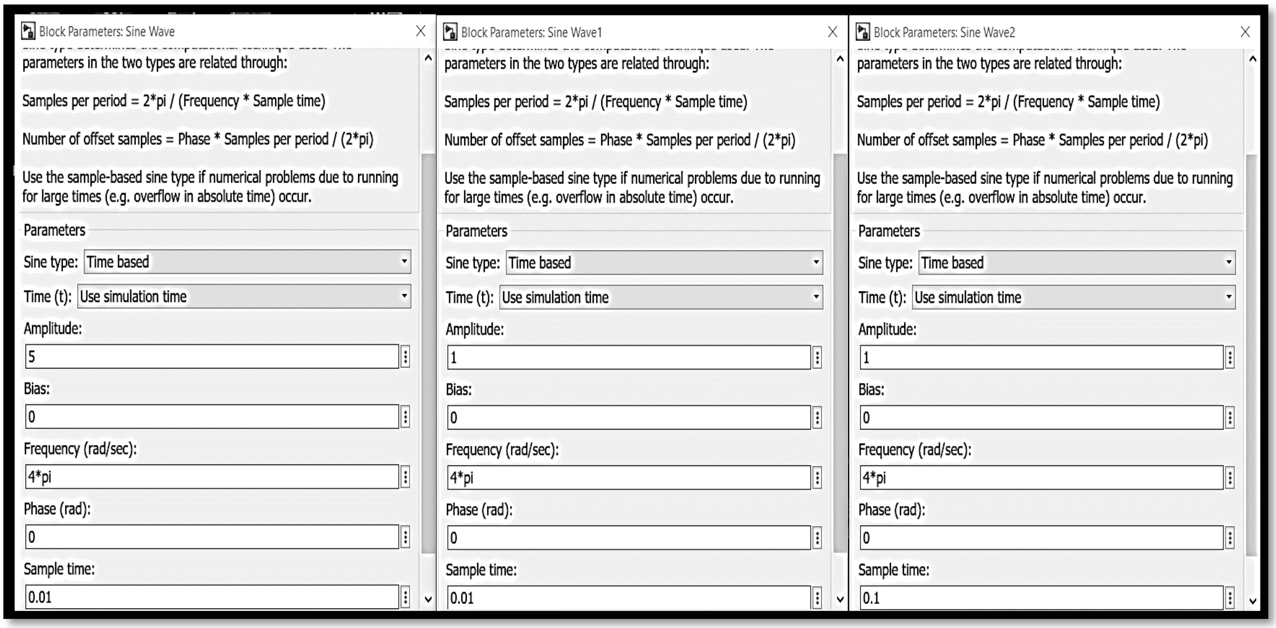
Experiment 3: Computations of the Probability of bit error for coherent binary ASK, FSK and PSK for an AWGN Channel and Compare them with their Performance curves.

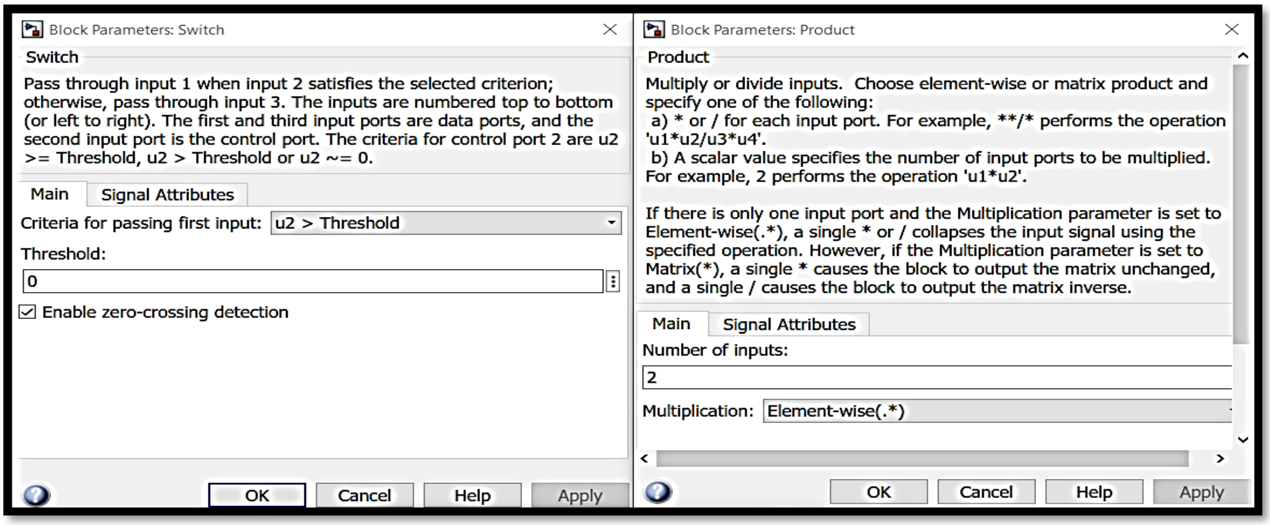
AMPLITUDE SHIFT KEYING

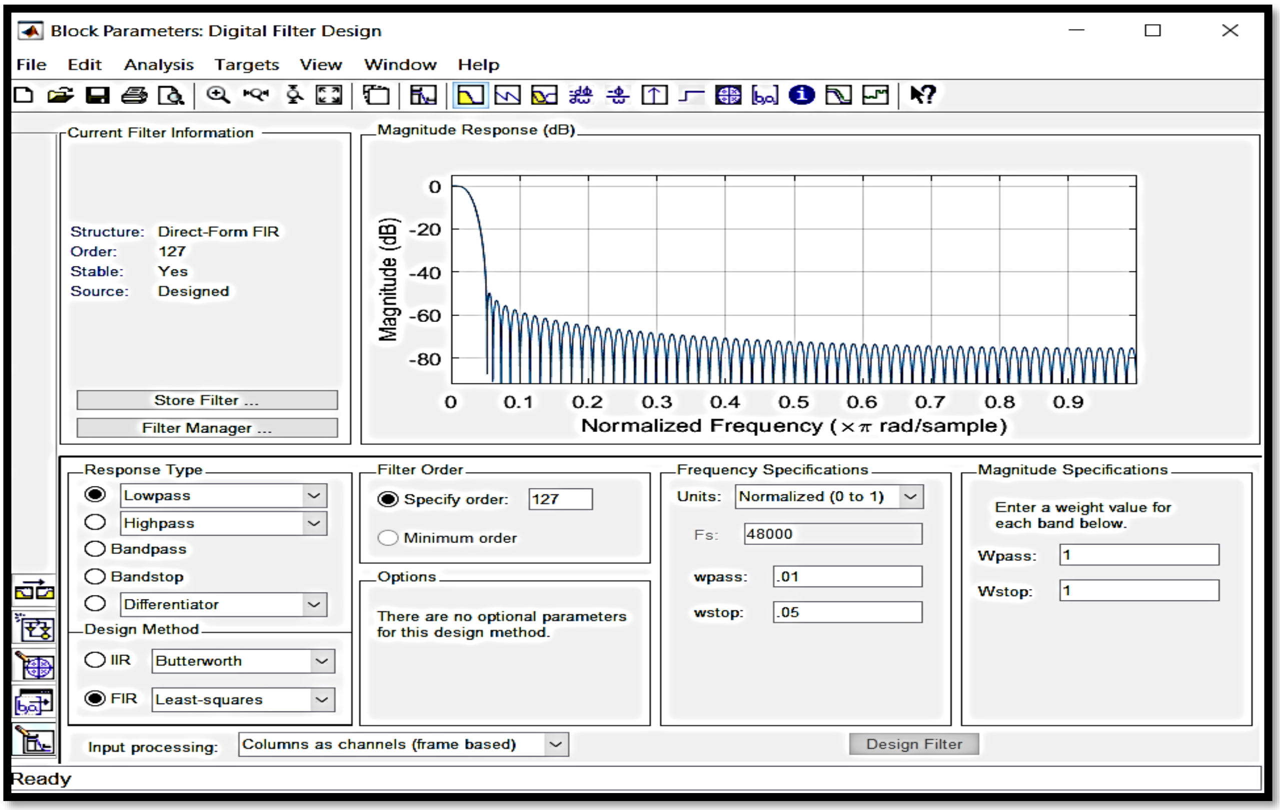
BLOCK DIAGRAM:



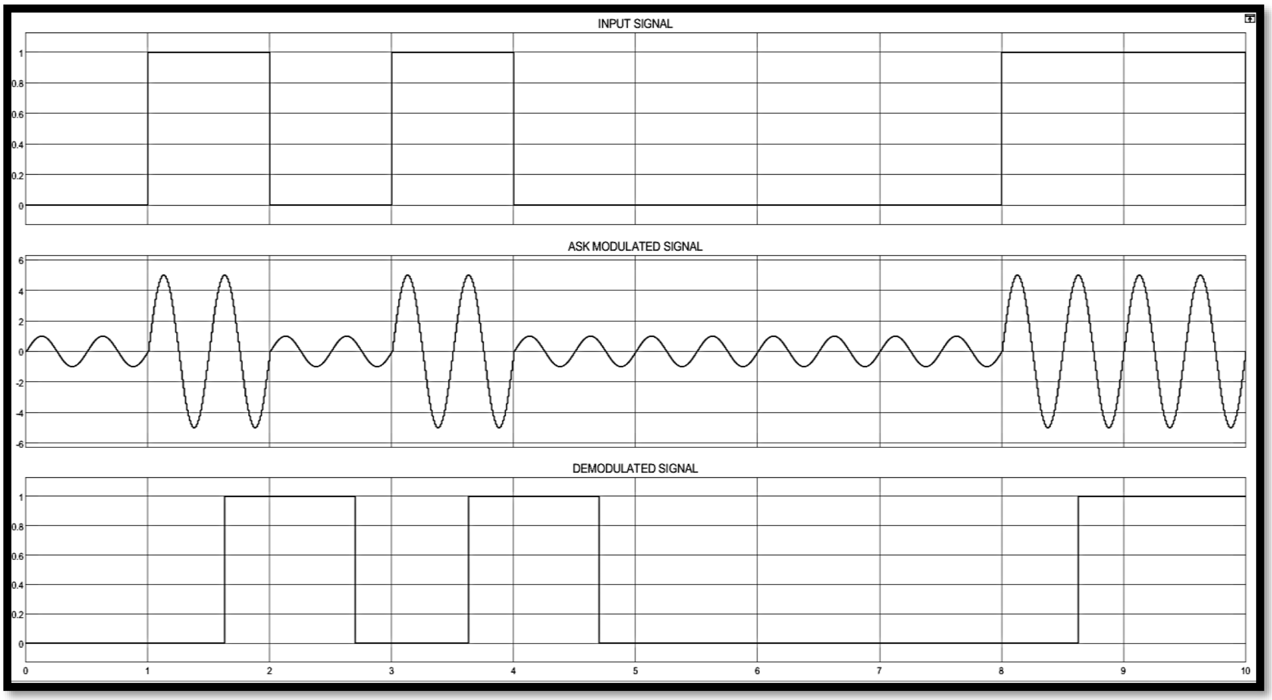
PARAMETERS:




Block Parameters: Switch
Switch
Pass through input 1 when input 2 satisfies the selected criterion; otherwise, pass through input 3. The inputs are numbered top to bottom (or left to right). The first and third input ports are data ports, and the second input port is the control port. The criteria for control port 2 are u2 >= Threshold, u2 > Threshold or u2 ~= 0.
Main
Signal Attributes
Criteria for passing first input: u2 > Threshold
Threshold:
0
Enable zero-crossing detection
OK
Cancel
Help
Apply
Block Parameters: Product
Product
Multiply or divide inputs. Choose element-wise or matrix product and specify one of the following:
a) * or / for each input port. For example, **/* performs the operation 'u1*u2/u3*u4'.
b) A scalar value specifies the number of input ports to be multiplied. For example, 2 performs the operation 'u1*u2'.
If there is only one input port and the Multiplication parameter is set to Element-wise(.*), a single * or / collapses the input signal using the specified operation. However, if the Multiplication parameter is set to Matrix(*), a single * causes the block to output the matrix unchanged, and a single / causes the block to output the matrix inverse.
Main
Signal Attributes
Number of inputs:
2
Multiplication: Element-wise(.*)
OK
Cancel
Help
Apply

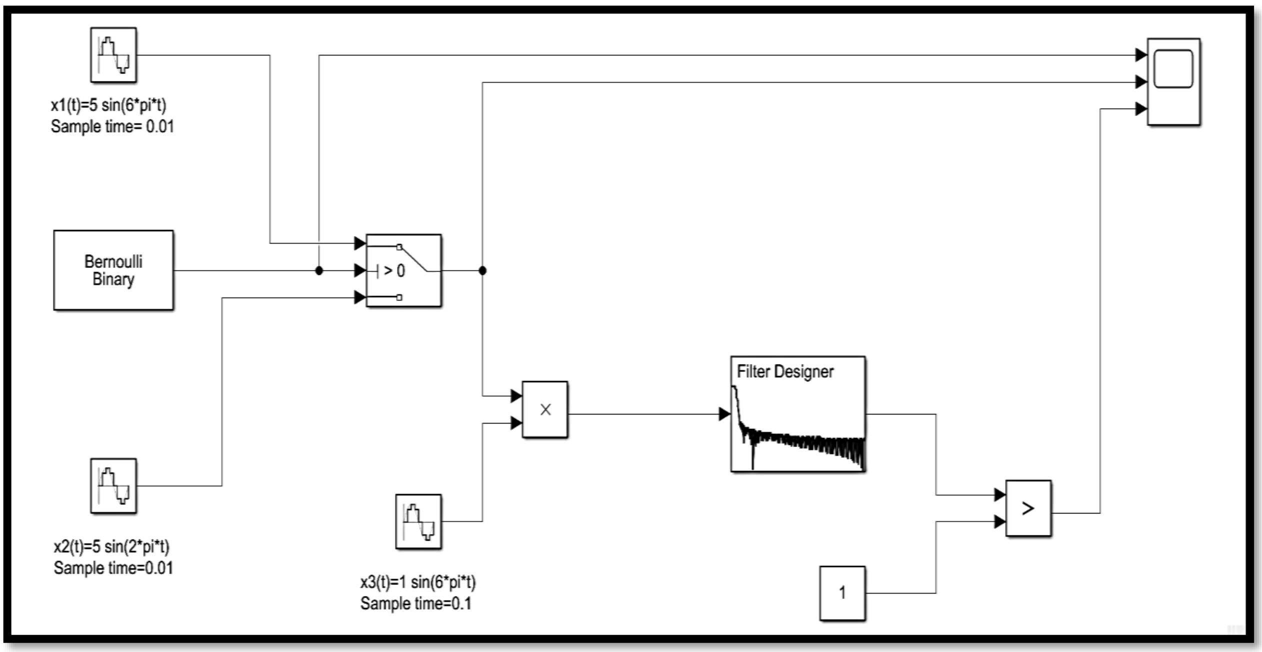

Block Parameters: Digital Filter Design
File Edit Analysis Targets View Window Help
Current Filter Information
Structure: Direct-Form FIR
Order: 127
Stable: Yes
Source: Designed
Store Filter ...
Filter Manager ...
Magnitude Response (dB)
Magnitude (dB)
0
-20
-40
-60
-80
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9
Normalized Frequency (×π rad/sample)
Response Type
Lowpass
Highpass
Bandpass
Bandstop
Differentiator
Design Method
IIR Butterworth
FIR Least-squares
Filter Order
Specify order: 127
Minimum order
Options
There are no optional parameters for this design method.
Frequency Specifications
Units: Normalized (0 to 1)
Fs: 48000
wpass: .01
wstop: .05
Magnitude Specifications
Enter a weight value for each band below.
Wpass: 1
Wstop: 1
Input processing: Columns as channels (frame based)
Design Filter
Ready

WAVEFORMS:

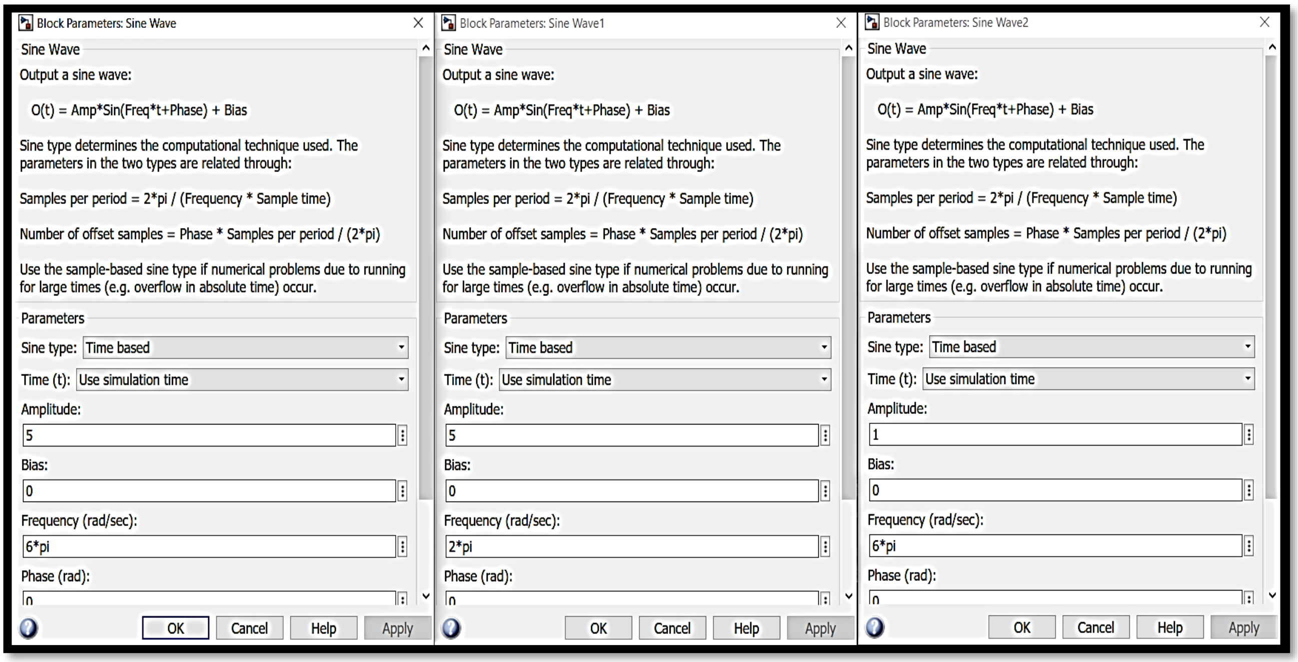


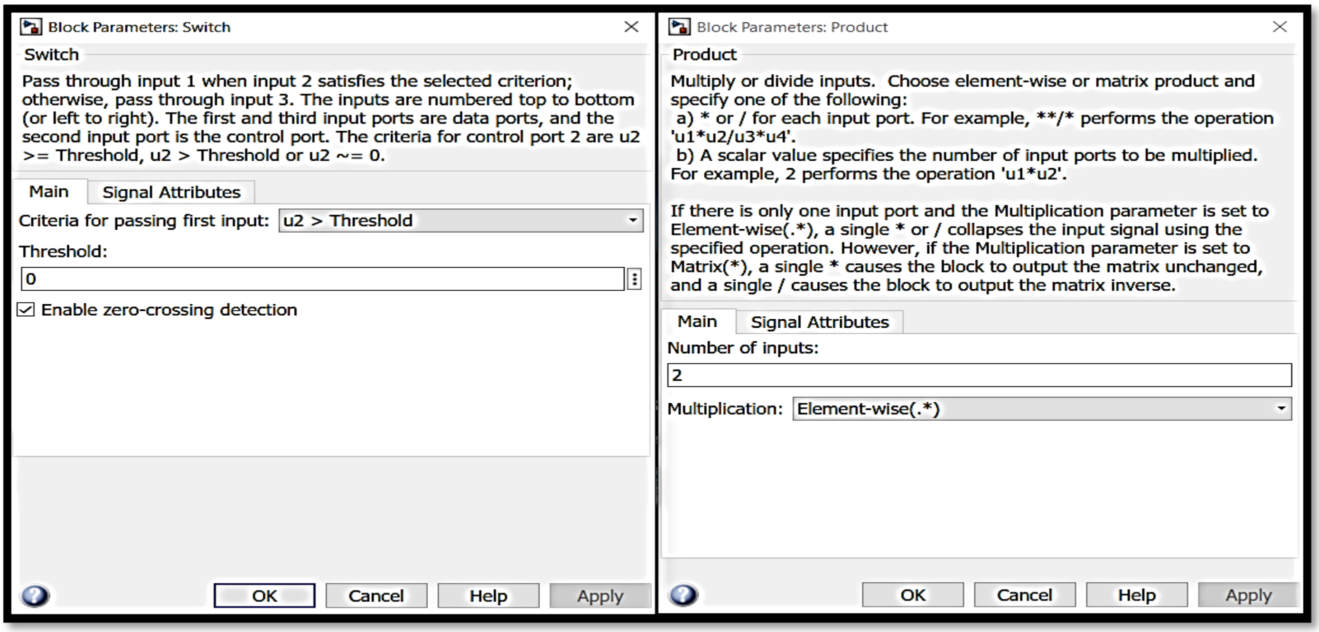
FREQUENCY SHIFT KEYING

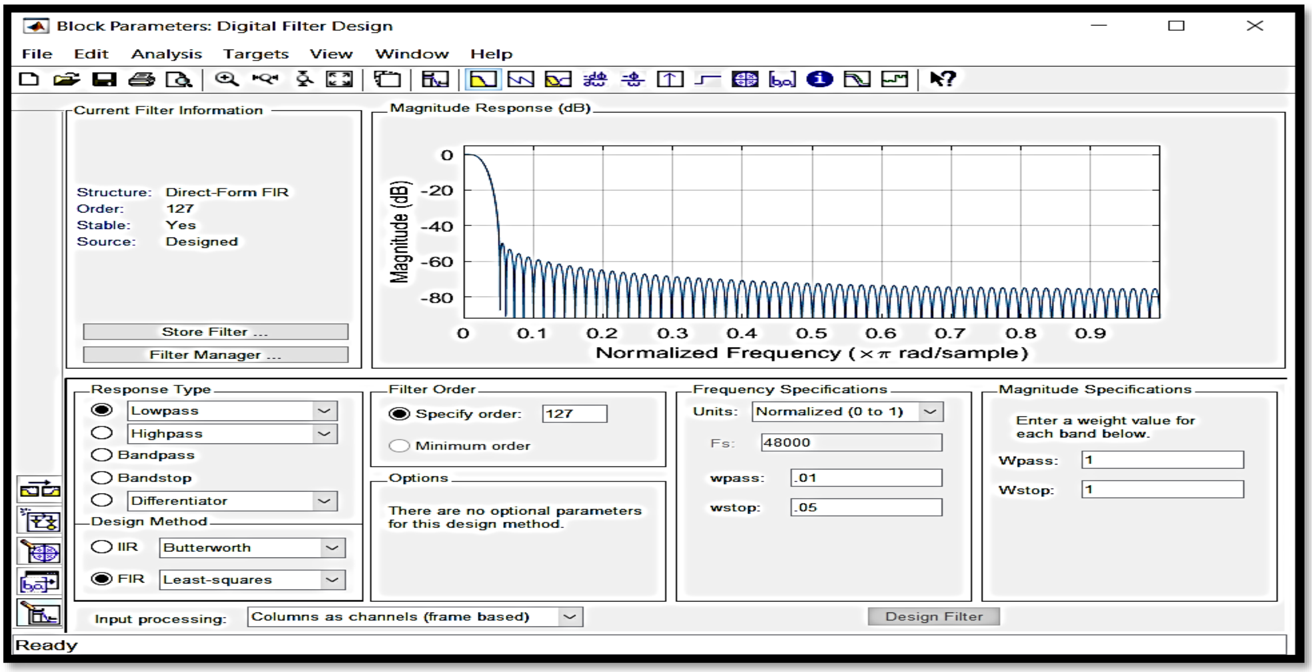
BLOCK DIAGRAM:



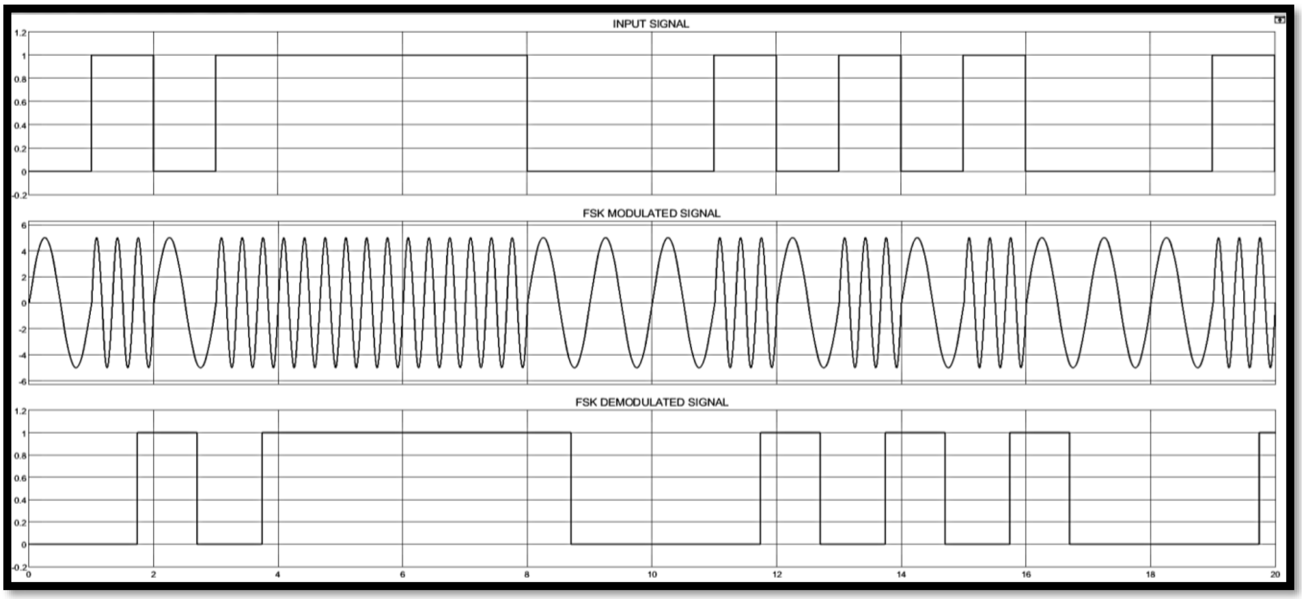
PARAMETERS:


Block Parameters: Sine Wave
Sine Wave
Output a sine wave:
O(t) = Amp*Sin(Freq*t+Phase) + Bias
Sine type determines the computational technique used. The parameters in the two types are related through:
Samples per period = 2*pi / (Frequency * Sample time)
Number of offset samples = Phase * Samples per period / (2*pi)
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
Parameters
Sine type: Time based
Time (t): Use simulation time
Amplitude:
5
Bias:
0
Frequency (rad/sec):
6*pi
Phase (rad):
OK
Cancel
Help
Apply
Block Parameters: Sine Wave1
Sine Wave
Output a sine wave:
O(t) = Amp*Sin(Freq*t+Phase) + Bias
Sine type determines the computational technique used. The parameters in the two types are related through:
Samples per period = 2*pi / (Frequency * Sample time)
Number of offset samples = Phase * Samples per period / (2*pi)
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
Parameters
Sine type: Time based
Time (t): Use simulation time
Amplitude:
5
Bias:
0
Frequency (rad/sec):
2*pi
Phase (rad):
OK
Cancel
Help
Apply
Block Parameters: Sine Wave2
Sine Wave
Output a sine wave:
O(t) = Amp*Sin(Freq*t+Phase) + Bias
Sine type determines the computational technique used. The parameters in the two types are related through:
Samples per period = 2*pi / (Frequency * Sample time)
Number of offset samples = Phase * Samples per period / (2*pi)
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
Parameters
Sine type: Time based
Time (t): Use simulation time
Amplitude:
1
Bias:
0
Frequency (rad/sec):
6*pi
Phase (rad):
OK
Cancel
Help
Apply


Block Parameters: Switch
Switch
Pass through input 1 when input 2 satisfies the selected criterion; otherwise, pass through input 3. The inputs are numbered top to bottom (or left to right). The first and third input ports are data ports, and the second input port is the control port. The criteria for control port 2 are u2 >= Threshold, u2 > Threshold or u2 ~= 0.
Main
Signal Attributes
Criteria for passing first input: u2 > Threshold
Threshold:
0
Enable zero-crossing detection
OK
Cancel
Help
Apply
Block Parameters: Product
Product
Multiply or divide inputs. Choose element-wise or matrix product and specify one of the following:
a) * or / for each input port. For example, **/* performs the operation 'u1*u2/u3*u4'.
b) A scalar value specifies the number of input ports to be multiplied. For example, 2 performs the operation 'u1*u2'.
If there is only one input port and the Multiplication parameter is set to Element-wise(.*), a single * or / collapses the input signal using the specified operation. However, if the Multiplication parameter is set to Matrix(*), a single * causes the block to output the matrix unchanged, and a single / causes the block to output the matrix inverse.
Main
Signal Attributes
Number of inputs:
2
Multiplication: Element-wise(.*)
OK
Cancel
Help
Apply

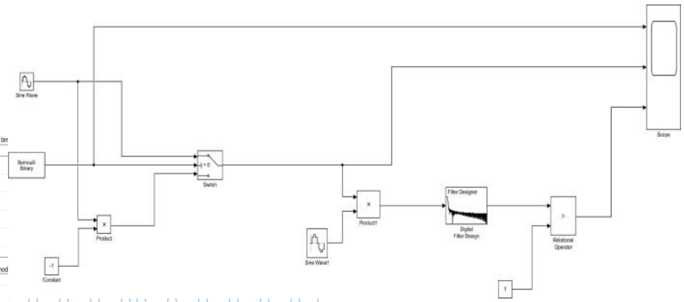

Block Parameters: Digital Filter Design
File Edit Analysis Targets View Window Help
Current Filter Information
Structure: Direct-Form FIR
Order: 127
Stable: Yes
Source: Designed
Store Filter ...
Filter Manager ...
Magnitude Response (dB)
Magnitude (dB)
Normalized Frequency (×π rad/sample)
Response Type
Lowpass
Highpass
Bandpass
Bandstop
Differentiator
Design Method
IIR Butterworth
FIR Least-squares
Filter Order
Specify order: 127
Minimum order
Options
There are no optional parameters for this design method.
Frequency Specifications
Units: Normalized (0 to 1)
Fs: 48000
wpass: .01
wstop: .05
Magnitude Specifications
Enter a weight value for each band below.
Wpass: 1
Wstop: 1
Input processing: Columns as channels (frame based)
Design Filter
Ready

WAVEFORMS:

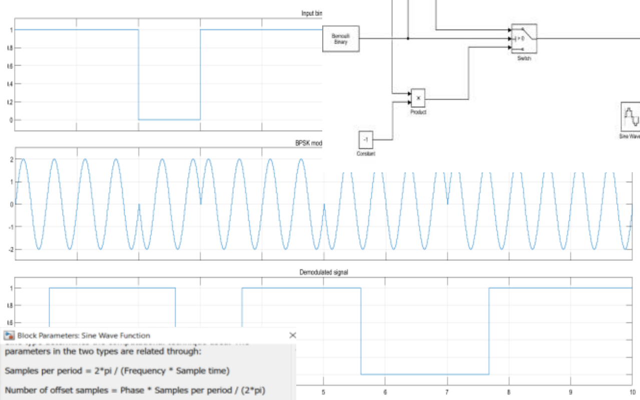

INPUT SIGNAL
FSK MODULATED SIGNAL
FSK DEMODULATED SIGNAL

BINARY PHASE SHIFT KEYING

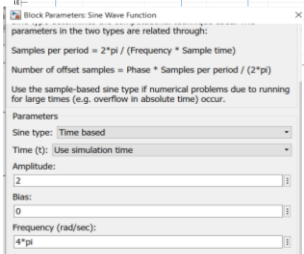
BLOCK DIAGRAM

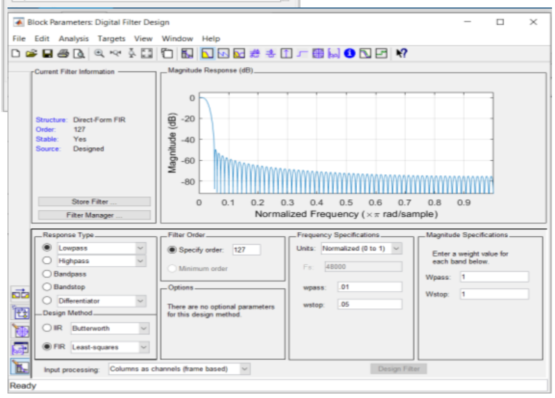


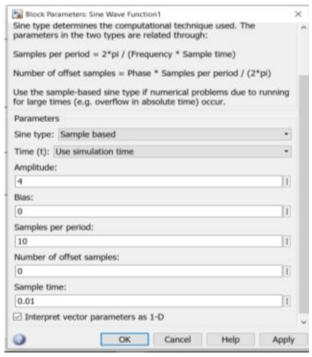
MODULATED AND DEMODULATED WAVEFORMS

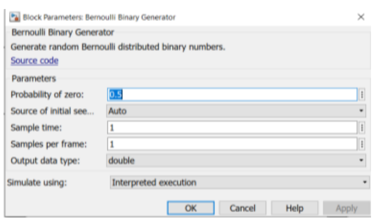


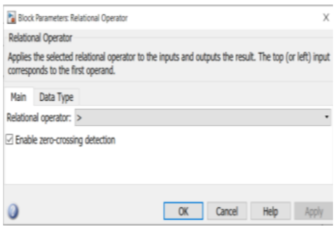
BLOCK PARAMETERS:



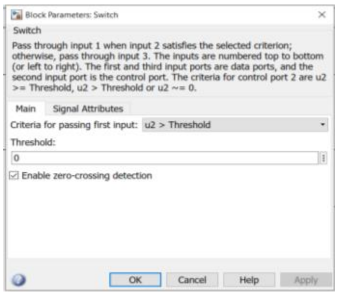


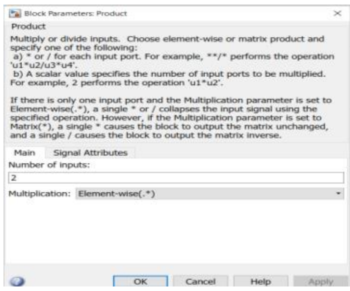






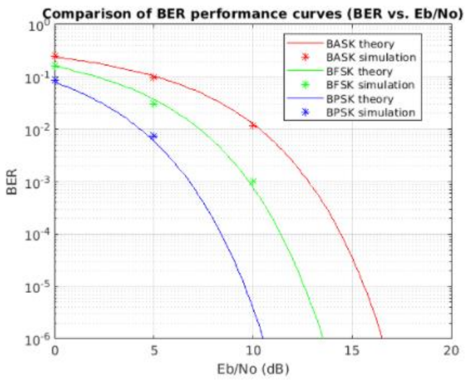
Computations of the Probability of bit error for coherent binary ASK, FSK and PSK for an AWGN Channel and Compare them with their Performance curves

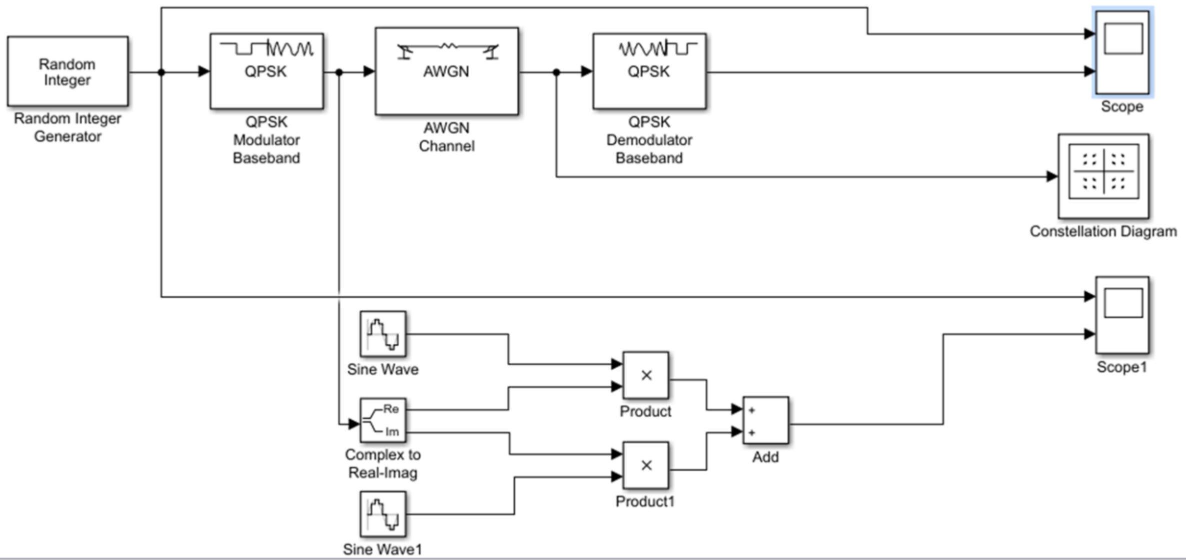


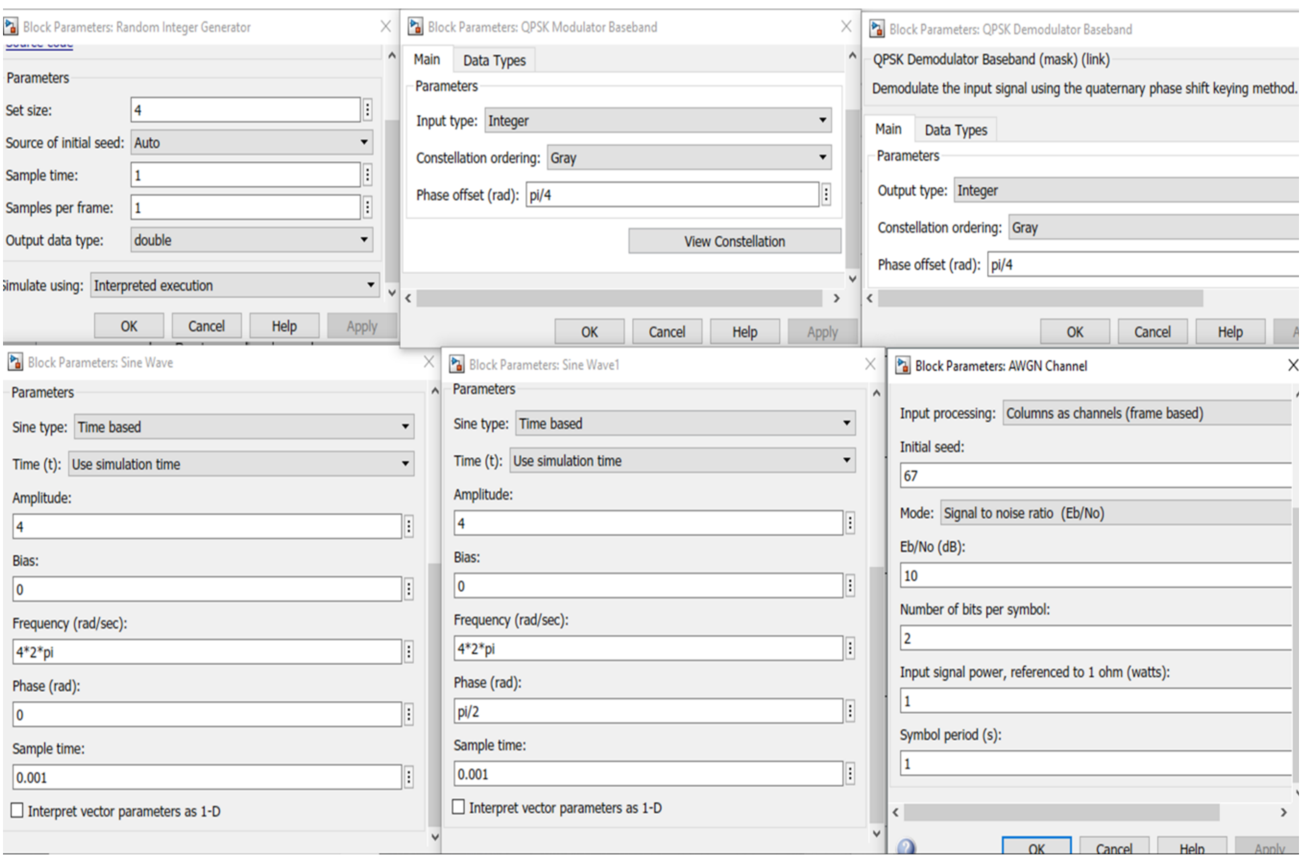



```
%A script to compare the BER of BASK, BFSK, BPSK from simulation and theory
%Variables:
%Sig_Length - bit length of the simulated signal
%Eb - bit energy of the simulated signal
Sig_Length = 2000;
Eb = 1;
%Eb/No for theoretical calculations (high resolution)
EtoN_dB = linspace(0,20,100);
EtoN = 10.^(EtoN_dB/10);
%Eb/No for simulation (five data points)
EtoN_dB_sim = linspace(0,20,5);
EtoN_sim = 10.^(EtoN_dB_sim/10);
%No for simulation purposes
No = Eb./EtoN_sim;
%Theoretical BER calculations
BER_BASK_te = (1/2)*erfc(sqrt(EtoN/4));
BER_BFSK_te = (1/2)*erfc(sqrt(EtoN/2));
BER_BPSK_te = (1/2)*erfc(sqrt(EtoN));


for iter = 1:length(EtoN_sim)
%Initialize error as 0
E_BASK = 0;
E_BFSK = 0;
E_BPSK = 0;


%Generate random string of bits with the length specified above
bit = randi([0 1],1,Sig_Length);
%Modulation of BFSK signal
x_BFSK = bit+1j*(~bit);
%Modulation of BASK signal
x_BASK = bit;
%Modulation of BPSK signal
x_BPSK = 2*bit-1;

% Generating noise
N_ril = sqrt(No(iter)/2)*randn(1,Sig_Length);
N_imj = sqrt(No(iter)/2)*randn(1,Sig_Length);
N = N_ril + 1j*(N_imj);


%Adding AWGN noise to the modulated signal
Y_BASK = x_BASK + N;
Y_BFSK = x_BFSK + N;
Y_BPSK = x_BPSK + N;
```

```
for iter2 = 1:Sig_Length
%BASK detector
Z_BASK(iter2) = (Y_BASK(iter2));
%Decision circuit BASK
if (Z_BASK(iter2) > 0.5 && bit(iter2) == 0) || (Z_BASK(iter2) < 0.5 && bit(iter2) == 1);
E_BASK = E_BASK + 1;
end
%BFSK detector
Z_BFSK(iter2) = real(Y_BFSK(iter2))-imag(Y_BFSK(iter2));
%Decision circuit BFSK
if (Z_BFSK(iter2) > 0 && bit(iter2) == 0) || (Z_BFSK(iter2) < 0 && bit(iter2) == 1);
E_BFSK = E_BFSK + 1;
end
%BPSK detector
Z_BPSK(iter2) = Y_BPSK(iter2);
%Decision circuit BPSK
if (Z_BPSK(iter2) > 0 && bit(iter2) == 0) || (Z_BPSK(iter2) < 0 && bit(iter2) == 1);
E_BPSK = E_BPSK + 1;
end


end

%Simulated BER calculations
BER_BASK_sim(iter) = E_BASK/Sig_Length;
BER_BFSK_sim(iter) = E_BFSK/Sig_Length;
BER_BPSK_sim(iter) = E_BPSK/Sig_Length;


End

%Making the graph
semilogy(EtoN_dB,BER_BASK_te,'k','color','red');
hold on
semilogy(EtoN_dB_sim,BER_BASK_sim,'k*','color','red');
semilogy(EtoN_dB,BER_BFSK_te,'k','color','green');
semilogy(EtoN_dB_sim,BER_BFSK_sim,'k*','color','green');
semilogy(EtoN_dB,BER_BPSK_te,'k','color','blue');
semilogy(EtoN_dB_sim,BER_BPSK_sim,'k*','color','blue');


legend('BASK theory','BASK simulation','BFSK theory','BFSK simulation','BPSK theory','BPSK
simulation','location','best');
axis([min(EtoN_dB) max(EtoN_dB) 10^(-6) 1]);
xlabel('Eb/No (dB)');
ylabel('BER');
title('Comparison of BER performance curves (BER vs. Eb/No)');
grid on;
hold off
```


Comparison of BER performance curves (BER vs. Eb/No)
BASK theory
BASK simulation
BFSK theory
BFSK simulation
BPSK theory
BPSK simulation
BER
Eb/No (dB)


Random Integer
Random Integer Generator
QPSK
QPSK Modulator Baseband
AWGN
AWGN Channel
QPSK
QPSK Demodulator Baseband
Scope
Constellation Diagram
Scope1
Sine Wave
Complex to Real-Imag
Sine Wave1
Product
Product1
Add


Block Parameters: Random Integer Generator
Parameters
Set size: 4
Source of initial seed: Auto
Sample time: 1
Samples per frame: 1
Output data type: double
Simulate using: Interpreted execution
OK Cancel Help Apply
Block Parameters: QPSK Modulator Baseband
Main Data Types
Parameters
Input type: Integer
Constellation ordering: Gray
Phase offset (rad): pi/4
View Constellation
OK Cancel Help Apply
Block Parameters: QPSK Demodulator Baseband
QPSK Demodulator Baseband (mask) (link)
Demodulate the input signal using the quaternary phase shift keying method.
Main Data Types
Parameters
Output type: Integer
Constellation ordering: Gray
Phase offset (rad): pi/4
OK Cancel Help
Block Parameters: Sine Wave
Parameters
Sine type: Time based
Time (t): Use simulation time
Amplitude: 4
Bias: 0
Frequency (rad/sec): 4*2*pi
Phase (rad): 0
Sample time: 0.001
Interpret vector parameters as 1-D
Block Parameters: Sine Wave1
Parameters
Sine type: Time based
Time (t): Use simulation time
Amplitude: 4
Bias: 0
Frequency (rad/sec): 4*2*pi
Phase (rad): pi/2
Sample time: 0.001
Interpret vector parameters as 1-D
Block Parameters: AWGN Channel
Input processing: Columns as channels (frame based)
Initial seed: 67
Mode: Signal to noise ratio (Eb/No)
Eb/No (dB): 10
Number of bits per symbol: 2
Input signal power, referenced to 1 ohm (watts): 1
Symbol period (s): 1
OK Cancel Help Apply

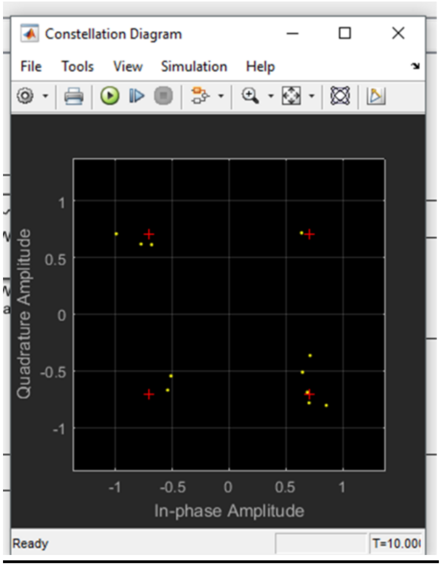
Experiment 4: Digital Modulation Schemes i) DPSK Transmitter and receiver, ii) QPSK Transmitter and Receiver

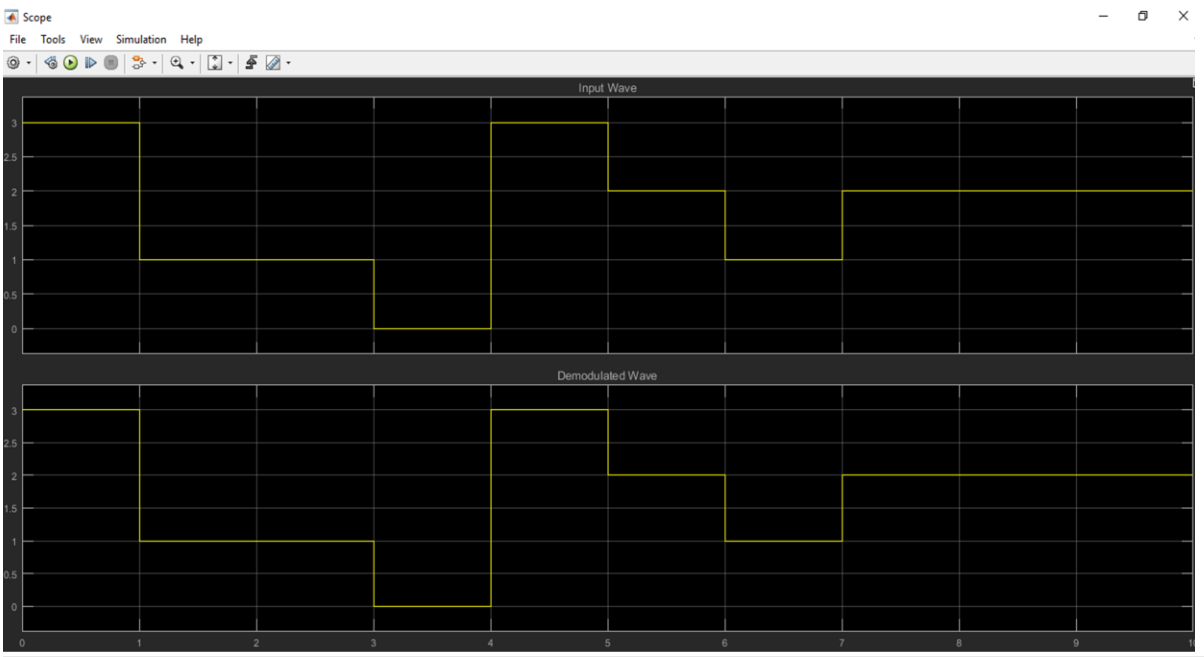
i) DPSK Transmitter and receiver

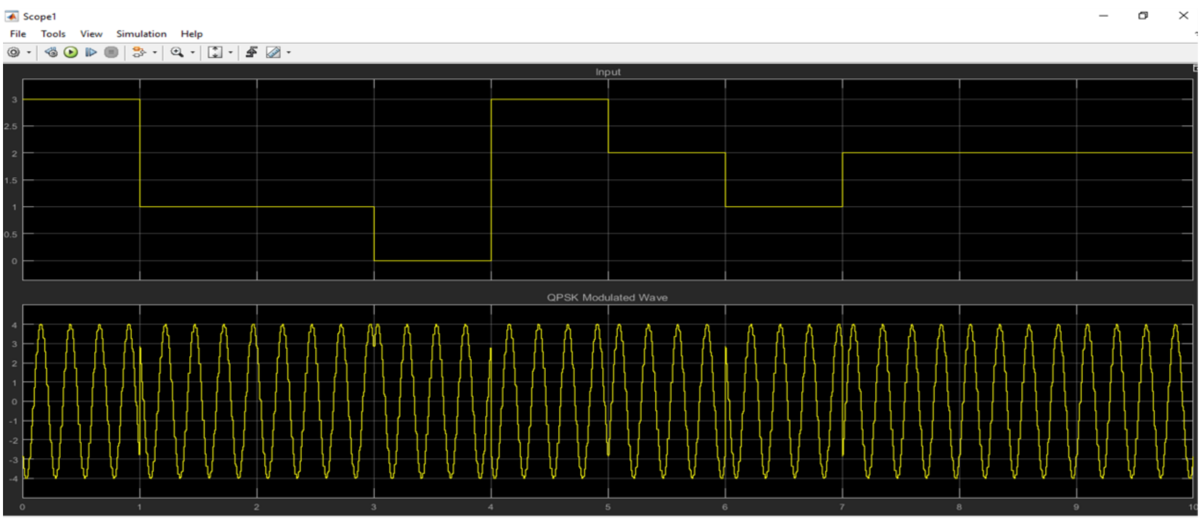
CIRCUIT:

PARAMETERS:

OUTPUTS:

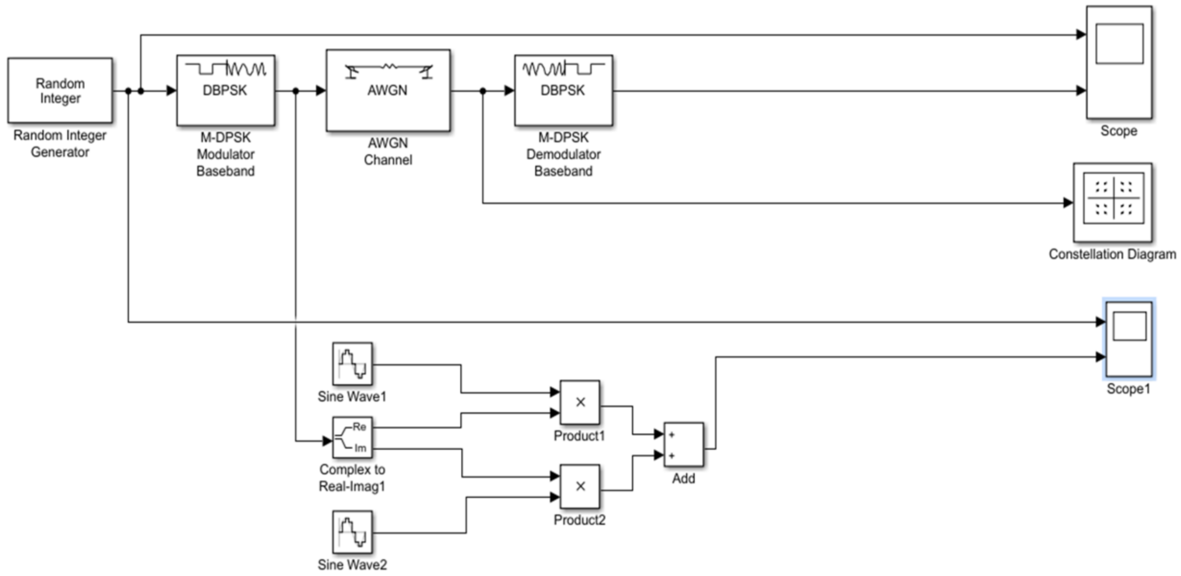

Constellation Diagram
File Tools View Simulation Help
Quadrature Amplitude
In-phase Amplitude
Ready
T=10.00


Scope
File Tools View Simulation Help
Input Wave
Demodulated Wave

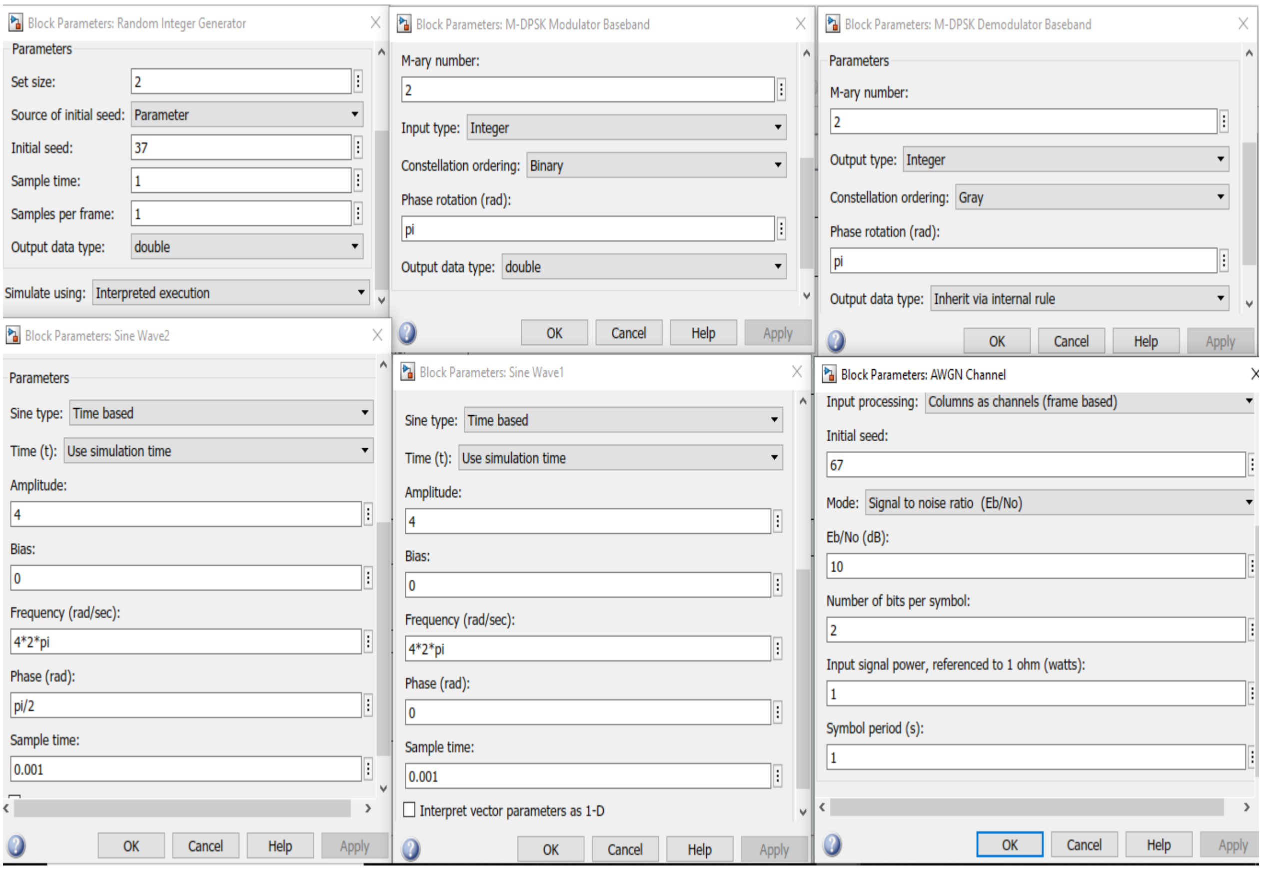

Scope1
File Tools View Simulation Help
Input
QPSK Modulated Wave

ii) QPSK Transmitter and Receiver

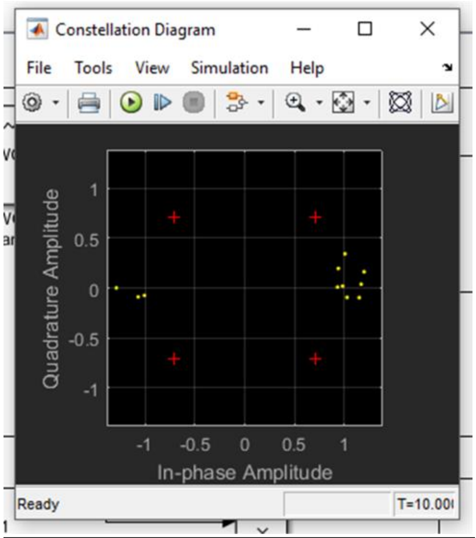
CIRCUIT:



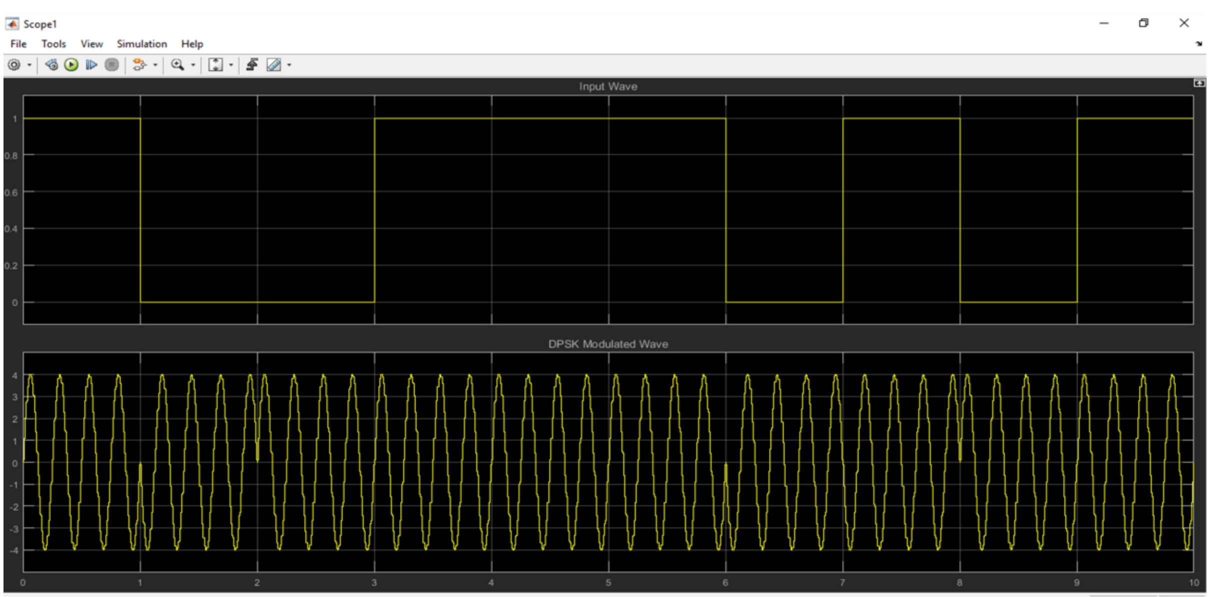
PARAMETERS:



OUTPUTS:


Constellation Diagram
File Tools View Simulation Help
Quadrature Amplitude
In-phase Amplitude
Ready
T=10.000


Scope
File Tools View Simulation Help
Input Wave
Demodulated Wave


Scope1
File Tools View Simulation Help
Input Wave
DPSK Modulated Wave