SRI CHANDRASEKHARENDRA SARASWATHI VISWA MAHAVIDYALAYA

(University established under section 3of UGC Act 1956)
(Accredited with 'A' Grade by NAAC)
Enathur, Kanchipuram – 631 561

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

COURSE MATERIAL FOR

ANALOG AND DIGITAL COMMUNICATION LABORATORY

FULL TIME B.E II YEAR, IV- SEMESTER

2021-22

Prepared by
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Assistant Professor/ECE

ANALOG AND DIGITAL COMMUNICATION LABORATORY IV Semester

OBJECTIVES

- To construct basic circuits of Analog and Digital communication system. To Design and construct experiments for performing modulation and sampling.
- To analyze the Performance characteristics of both analog and Digital Communication Systems.

LIST OF EXPERIMENTS

1. Study of Multisim, VisSim and MATLAB
2. AM modulator and Demodulator.
3. DSB-SC modulator and Demodulator.
4. SSB modulator and Demodulator.
5. FM modulator and Demodulator.
6. PAM modulator and Demodulator.
7. PPM & PWM Modulator
8. Pre-emphasis and De-emphasis in FM.
9. Signal Sampling and Reconstruction (Sampling Theorem)
10. Pulse Code Modulation and Demodulation
11. Delta modulation and Adaptive Delta modulation.
12. Amplitude Shift Keying (ASK) and Frequency Shift Keying (FSK) modulator and Demodulator.
13. Phase Shift keying (PSK) and Binary Phase Shift Keying (BPSK) Modulator and Demodulator.

OUTCOMES

- Apply the practical knowledge to construct Analog and Digital communication.
- Evaluate Analog and Digital modulated waveform in time /frequency domain.
- Analyze and evaluate the performance of Analog and Digital communication systems

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Study of Multisim

The Multisim Documentation Set

Multisim documentation consists of a Getting Started and Tutorial manual, this User Guide, and on- line help. All Multisim users receive PDF versions of the Getting Started and Tutorial manual and the User Guide. Depending on your version of Multisim, you may also receive a printed version of the manuals.

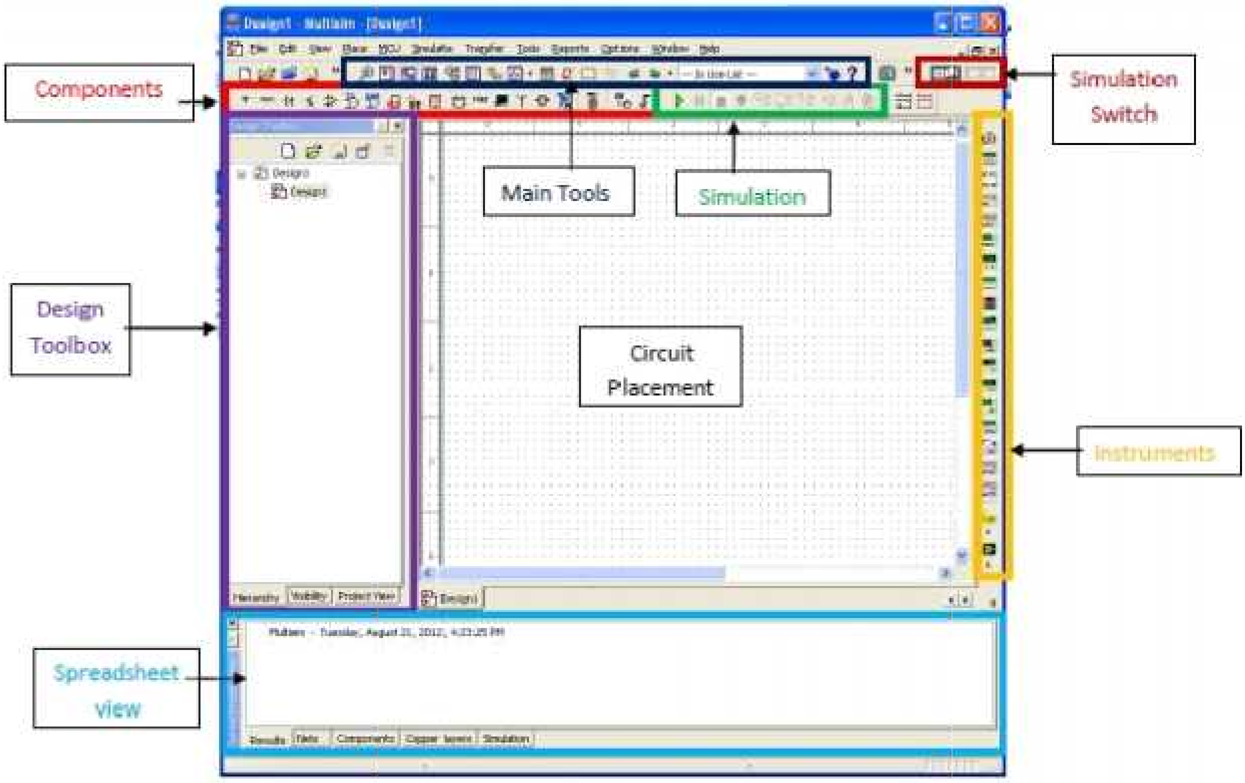
Getting Started and Tutorial

The Getting Started and Tutorial manual introduces you to the Multisim interface. It also offers an introductory tutorial that takes you through the stages of circuit design, simulation, analysis and reporting.

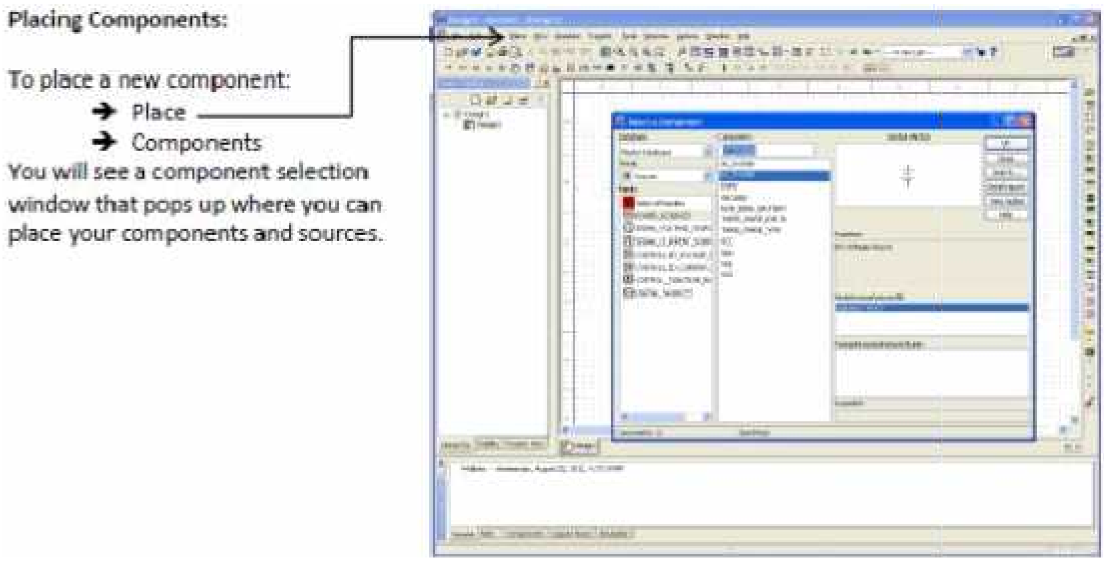
What is Multisim ?

Multisim is the latest generation of the world’s most popular electronic design and edu- cation software from Electronics Workbench. It is a complete system design tool that offers a large component database, schematic entry, full analog/digital SPICE simulation, VHDL/Ver- ilog design entry/simulation, FPGA/CPLD synthesis, RF capabilities, postprocessing features and seamless transfer to PCB layout packages such as Ultiboard, also from Electronics Work-bench. It offers a single, easy-to-use graphical interface for all your design and analysis needs.

Introduction to the Multisim Interface

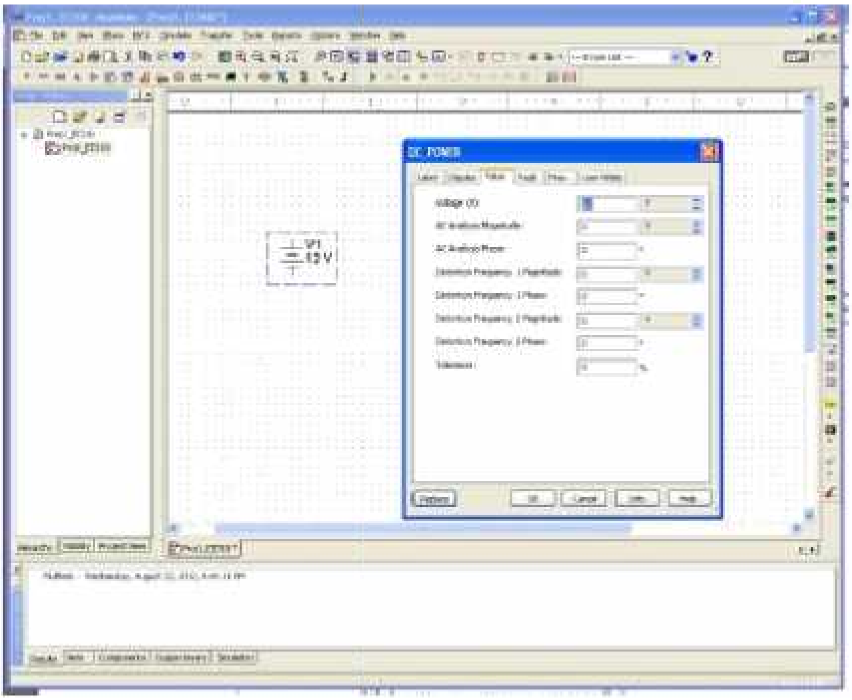


Placing of Component:

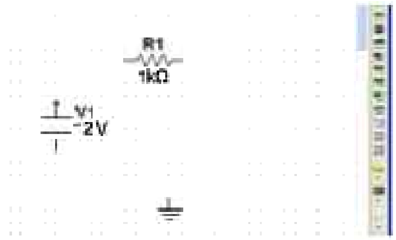


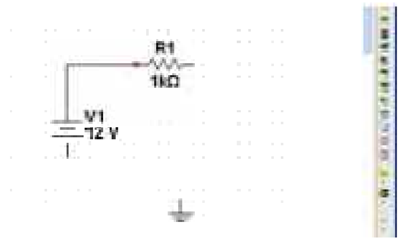
Changing Component Parameters:

To change the values or names of the component, you can either right click on the component and select properties, or double click on the component, and a screen will pop up allowing you to change the properties and values of the component.

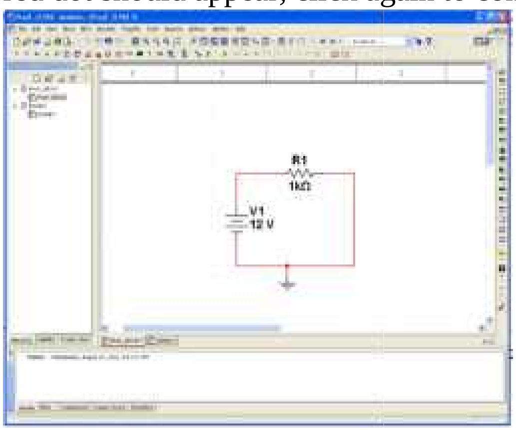


Wiring:



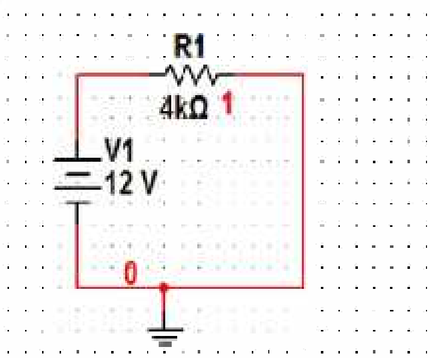


When wiring, place cursor at the corners of component and you should see a black dot appearing, click the mouse once and drag your wire to the edge of another component where a red dot should appear, click again to connect.



Displaying Node numbers/ Wire names:

To show the net names of the wires, select the desired wire and right click Properties, and a Net Settings window should pop up. Checking the show net name box will allow one to see the wire names/net on the circuit.

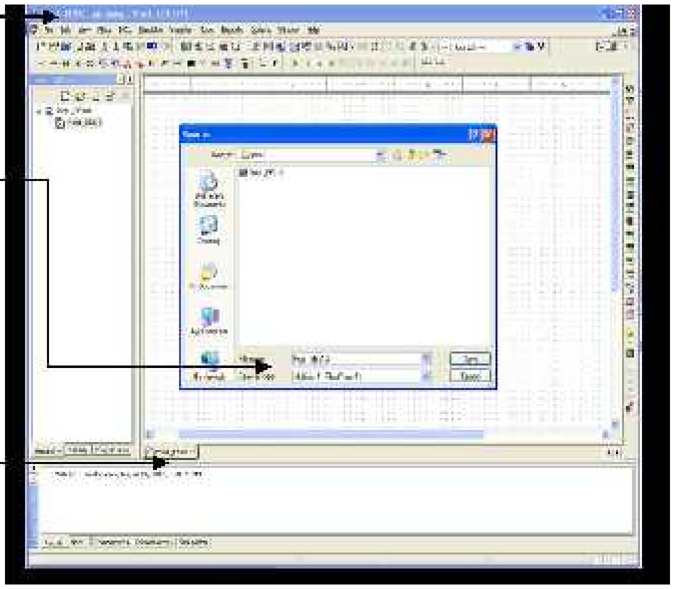


One can als [redacted]

Saving:

To save your project:

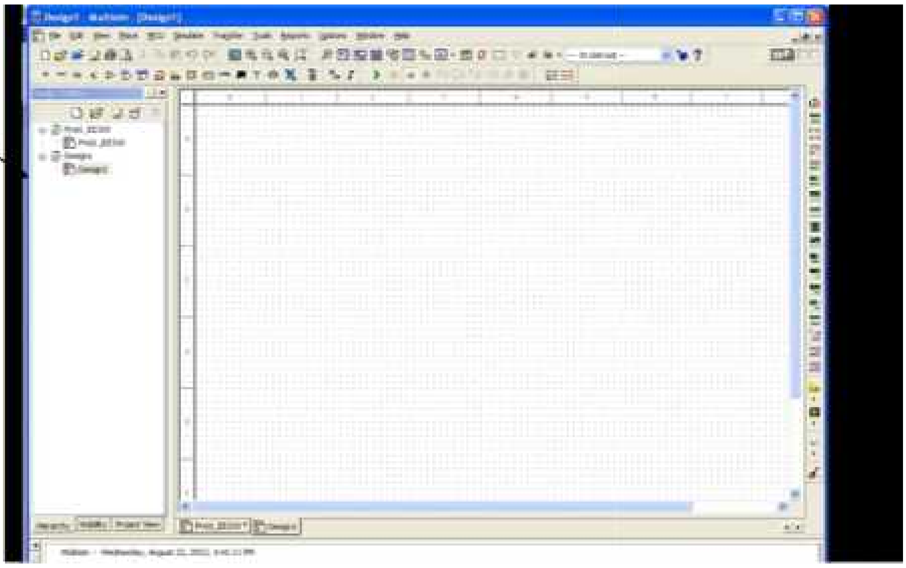
[redacted]



Once saved, you will see the name appearing on of the window and the original name Design1 change to your new project name.

You can also create more than one project in one setting by:

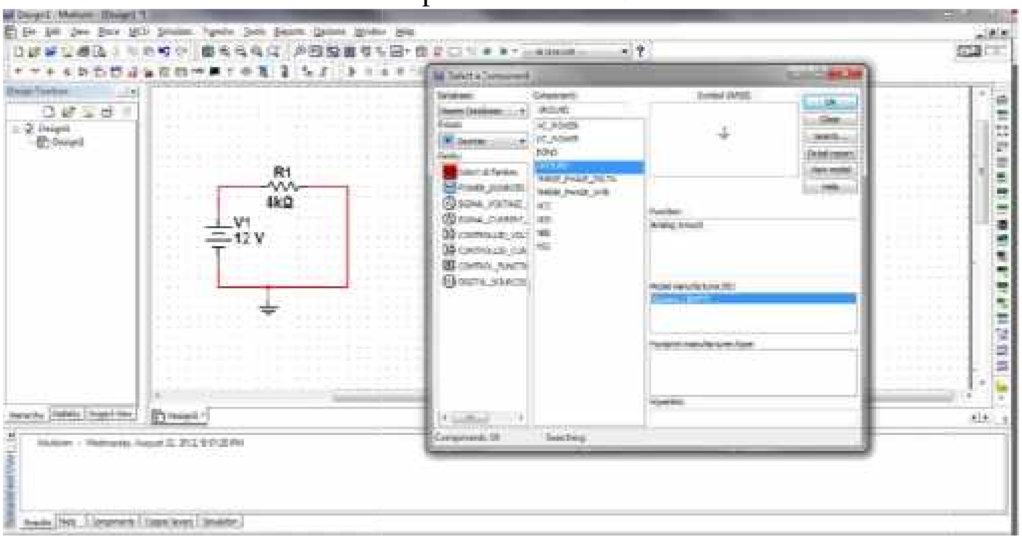
- File
- New
- Design



Simulation Analysis:

First, draw the desire circuit:

1. Place -> Components
2. In the Select Component windows, under Group, select Sources for DC_POWER and GROUND.
3. In Group, select Basic and chose RESISTOR.
4. Connect the wires to complete the circuit.

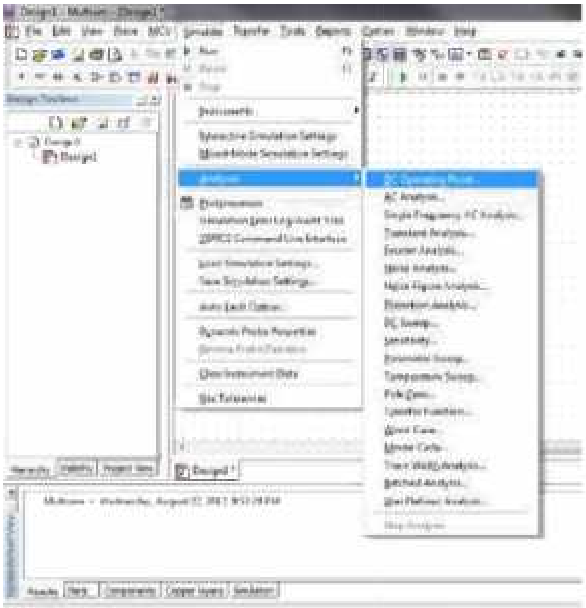


DC Operating Points:

To see the DC Bias points:

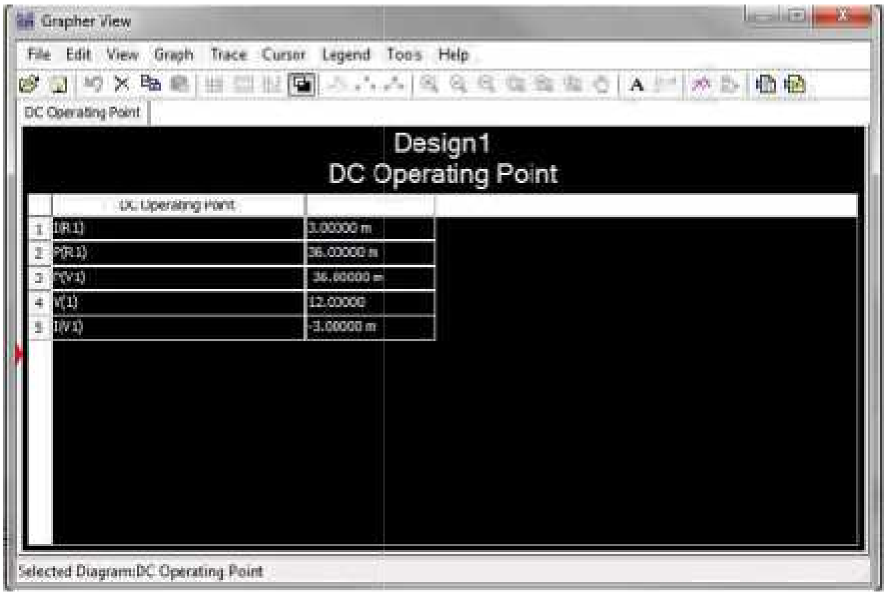
- Simulate
- Analyses
- DC Operating Point

The DC Operating Point Analysis window will pop up
with three tabs, Output, Analysis options and Summary.



For the Output tab, one can select the esired variables by Adding or Removing. click Simulate
when done. You can also change the simulation title name in the Analysis options tab.

A Grapher View window will pop up after clicking on the Simulate button, showing the DC operating points corresponds to the components in the circuit.



Copy Graphs and Circuits:

To copy circuit schematic: Select the entire circuit using the mouse and right click [redacted]opy

To copy the graph desired from Grapher View: Ed[redacted]copy graph

DC Sweep:

[redacted]Simulate
[redacted]Analyses
[redacted]DC Sweep

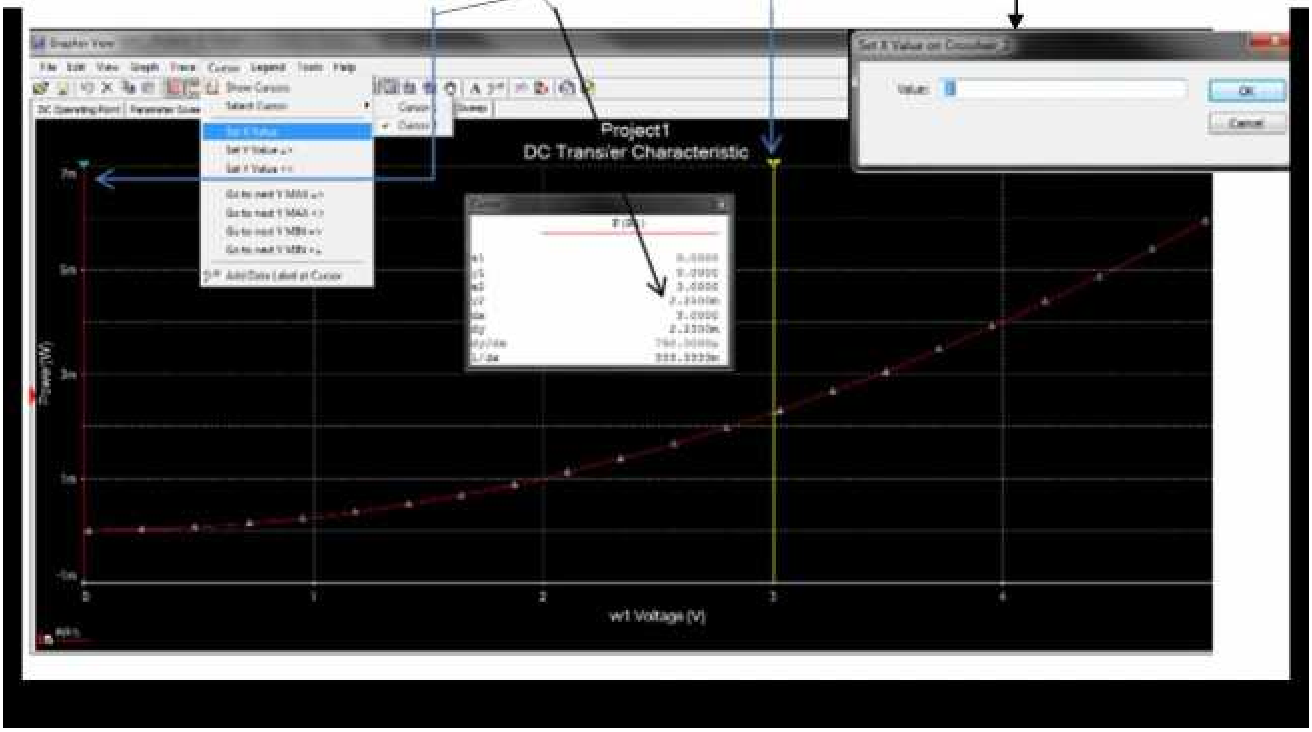
As shown above, we can see the graph of Power vs. Voltage for our Resistor 1.

To see different point values or the use of **Cursors** in the Grapher View

- window**:** Cursor
- Show Cursors (a Cursor window should pop up) and two cursors should show
- up Cursor
- Set X value (a Set X value window should pop

  up) o Set value to 3 and click OK

  o This sets the Voltage value to 3 and this allow us to see what is he y value, or Power value when the voltage is at 3 volts



Parametric Sweep:

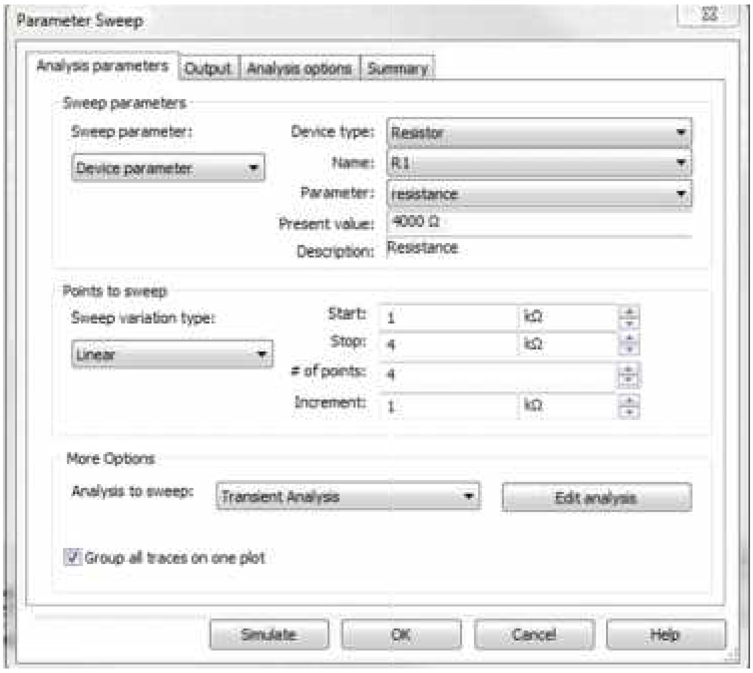
To see the different Current values when we change the Resistor values, we can do a parametric sweep.

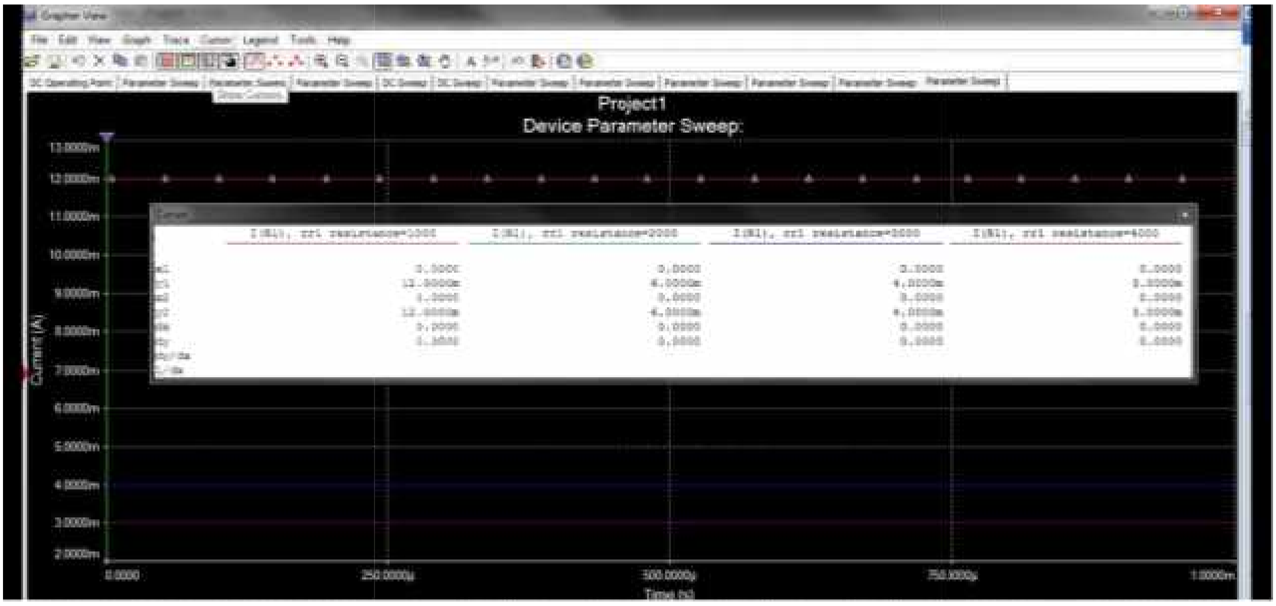
- Simulate
- Analyses
- Parameter Sweep

In the Parameter Sweep window, change the following parameters:

After changing the parameters in the Analysis parameters, go to the Output tab and select the desired output to I(R1).

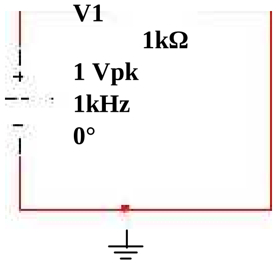
After clicking on Simulate, the Grapher View should pop up showing the Graph with 4 different curves. Click on the show Cursors will allow you to see the different current values for different resistance values.

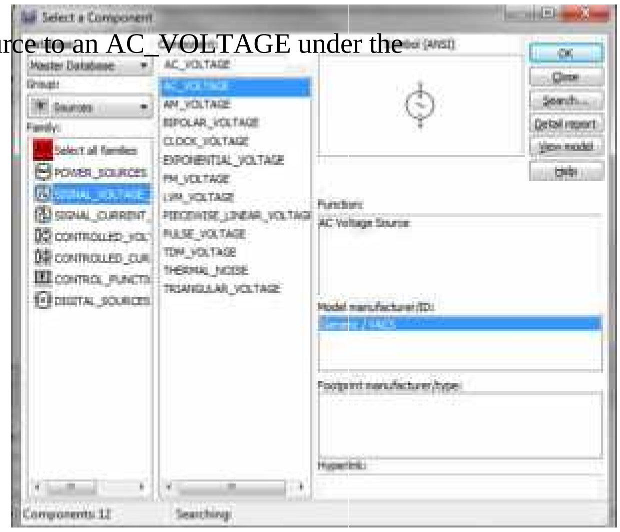




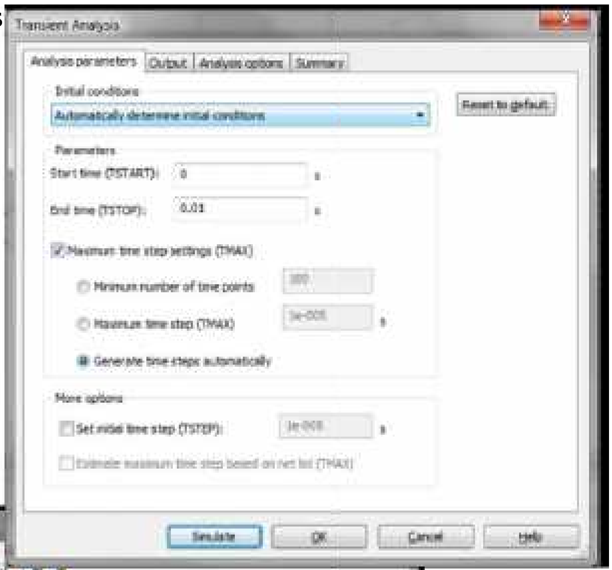
Transient Analysis:

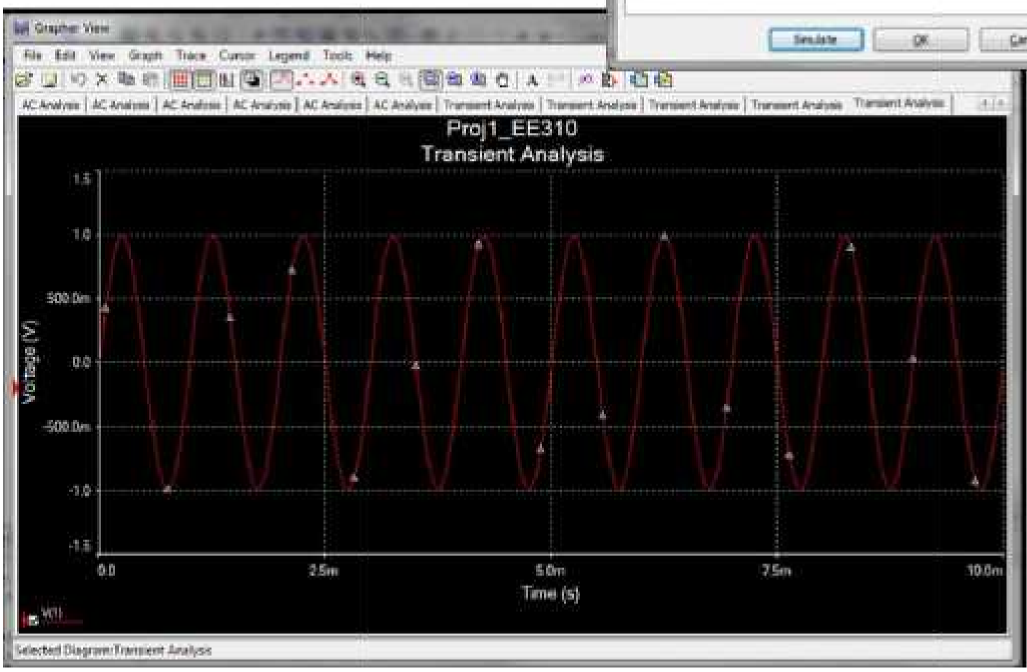
For transient analysis, we can change the voltage source to an AC_VOLTAGE under the **SIGNAL_VOLTAGE_SOURCES.**





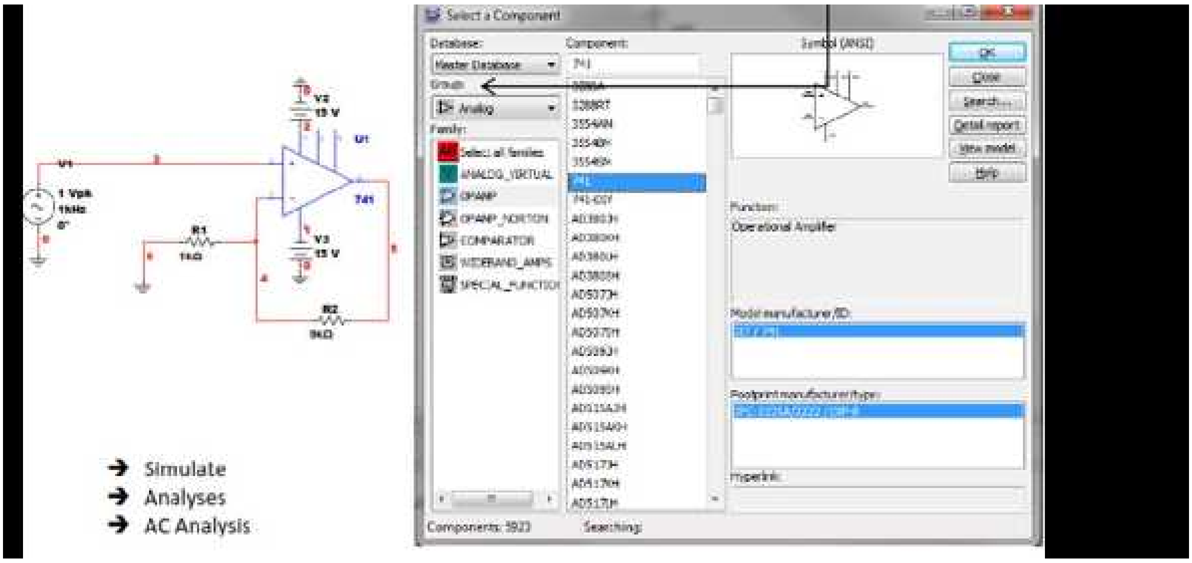
- Simulate
- Analyses
- Transient Analysis
  - Change the following parameters
  - For output, we can verify our voltage source by adding V(1) to our output





AC Analysis:

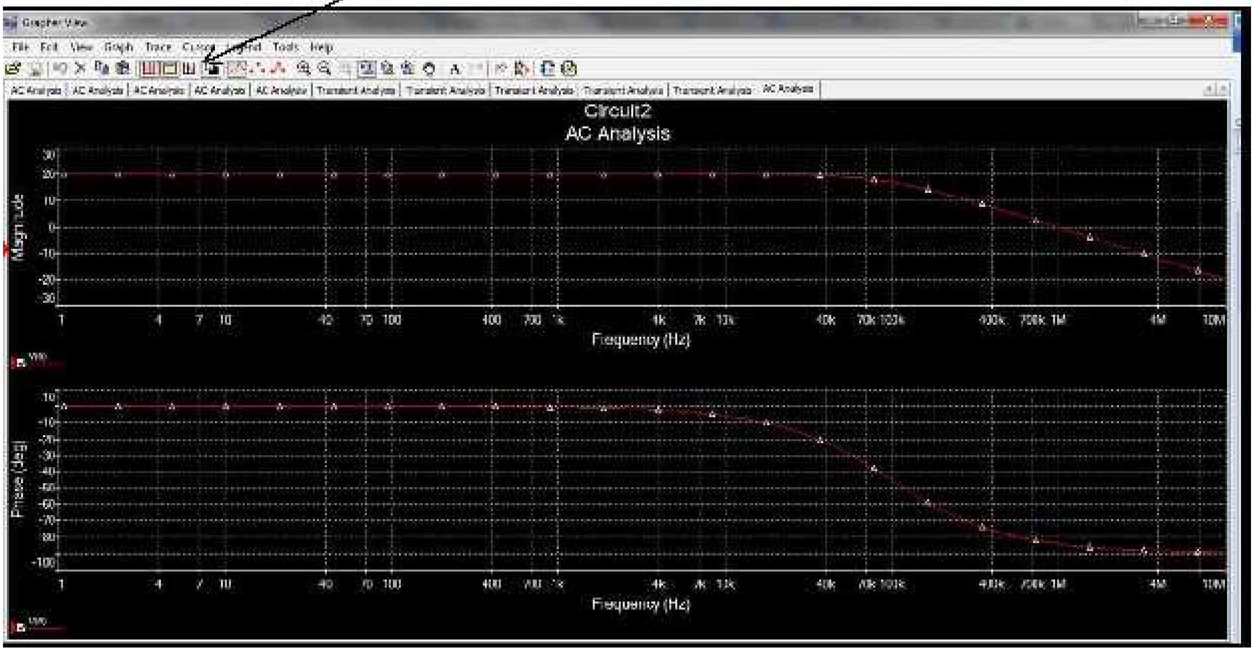
Draw the following circuit (closed loop inverting amplifier), for the amplifier, under Group, select Analog and select the OPAMP under Family and type in 741 for the amplifier part.



A window for the AC analysis should pop up, change to the following parameters:

For the Output tab, choose V(6), the output of the amplifier.

Two graphs should be show in the Grapher View, where the first graph shows the magnitude of the gain of the amplifier, and the second graph shows the Phase vs. Frequency (we will usually look at only the first graph). Again, if we would like to see or plot points on the graph, we can choose Cursors, and move or type in our desired location.



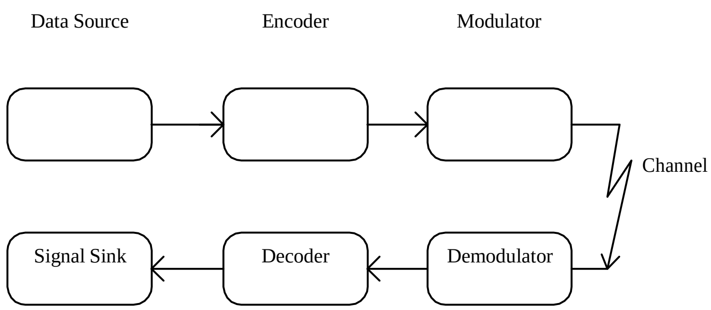
Use the link to have a full manual for the Multisim:
http://www.swarthmore.edu/NatSci/echeeve1/Class/e11/DL/Userguid.pdf

Study of VisSim

Introduction

A typical communication link includes, at a minimum, three key elements: a transmitter, a communication medium (or channel), and a receiver. The ability to simulate all three of these elements is required in order to successfully model any end-to-end communication system.

The transmitter and receiver elements can in turn be further subdivided into sub-systems, as shown in the figure below. These include a data source (analog or digital), an optional data encoder, a modulator, a demodulator, an optional data decoder, and a signal sink.



The data source generates the information signal that is intended to be sent to a particular receiver. This signal can be either an analog signal such as speech, or a digital signal such as a binary data sequence. This signal is typically a baseband signal represented by a voltage level.

For analog signals, it is often desirable to digitally encode the signal prior to transmission by undergoing a quantization process. This step converts the analog signal into a digital signal. While some information is lost in this process, the resulting digital signal is often far less susceptible to the effects of noise in the transmission channel.

An encoder can also be used to add redundancy to a digital data stream, in the form of additional data bits, in a way that provides an error correction capability at the receiver. This overall process is referred to as Forward Error Correction (FEC). Among the most popular FEC schemes are convolutional coding, block coding and trellis coding. It is important to note that usually the output bit rate of an encoder is not equal to the input bit rate. To properly distinguish between the two bit rates, the transmitter's input rate is referred to as the information data rate, while the transmitter output rate is referred to as the channel data rate.

Depending on the type of information signal and the particular transmission medium, different modulation techniques are employed. Modulation refers to the specific technique used to represent the information signal as it is physically transmitted to the receiver. For example, in Amplitude Modulation (AM), the information is represented by amplitude variations of the carrier signal.

Once the signal is modulated, it is sent through a transmission medium, also known as a *channel*, to reach the intended receiver. This may be a copper wire, coax cable, or the atmosphere in the case of a radio transmission. To some extent, all channels introduce some form of distortion to the original signal. Many different channel models have been developed to mathematically represent such distortions. A commonly used channel model is the Additive White Gaussian Noise

(AWGN) channel. In this channel, noise with uniform power spectral density (hence the term *white*) is assumed to be added to the information signal. Other types of channels include fading channels and multipath channels.

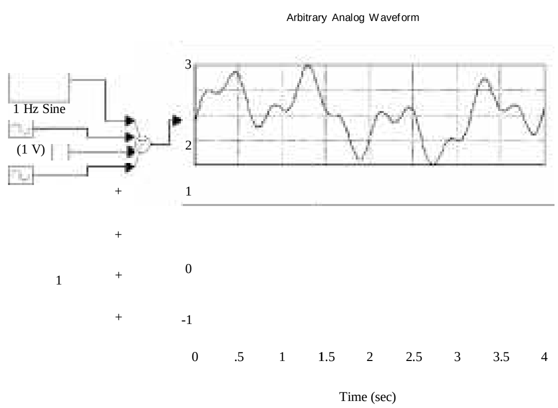
When the transmitted signal reaches the intended receiver, it undergoes a *demodulation* process. This step is the opposite of modulation and refers to the process required to extract the original information signal from the modulated signal. Demodulation also includes any steps associated with signal synchronization, such as the use of phase-locked loops in achieving phase coherence between the incoming signal and the receiver's local oscillator.

When data encoding is included at the transmitter, a data decoding step must be performed prior to recovering the original data signal. The signal decoding process is usually more complicated than the encoding process and can be very computationally intensive. Efficient decoding schemes, however, have been developed over the years—one example is the Viterbi decoding algorithm, which is used to decode convolutionally encoded data.

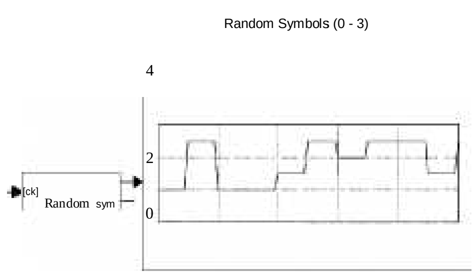
Finally, an estimate of the original signal is produced at the output of the receiver. The receiver’s output port is sometimes referred to as the signal sink. As communications engineers, we are usually interested in knowing how well the source information was recreated at the receiver’s output. Several metrics are used by engineers to evaluate the success of the data transmission. The most common metric, in the case of digital signals, is the received Bit Error Rate (BER). Other valuable performance indicators include the received signal to noise ratio, eye patterns and phase scatter plots to name a few.

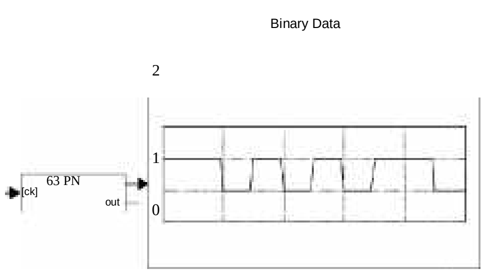
Signal Sources

VisSim/Comm provides both analog and digital signal sources. By combining basic signal producers blocks, the user can generate just about any analog time domain waveform that is desired. The basic building blocks provided include step functions and impulses to name a few. The user can also import a signal from an external file, including wave files (.WAV), as shown below.

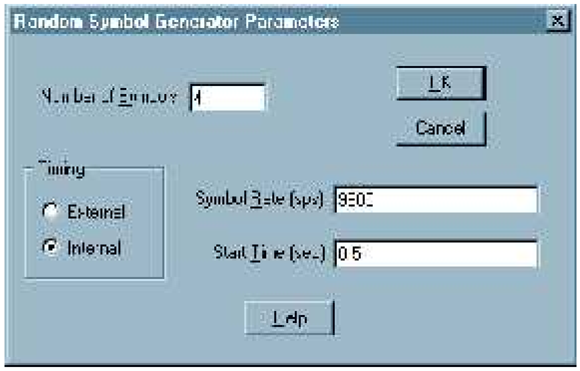


When working with digital signals, the VisSim/Comm environment provides sources for generating random bits or symbols, as well as pseudo-random bit patterns such as PN sequences. A bit can assume a value of either “1” or “0”, while a symbol is defined as a value which is a member of the set of M possible integer values {0,1,2,…,M-1}. Symbol values are typically used within VisSim/Comm when dealing with high order modulation constellations.





Data source parameters are easily specified by accessing each block's Setup dialog box. For example, the block allows the user to specify the symbol alphabet size (2 for binary, 4 for QPSK), the symbol rate in hertz (Hz), and a data start time. In addition, this block allows the use of a user-supplied clock by selecting external timing within the Setup dialog box.



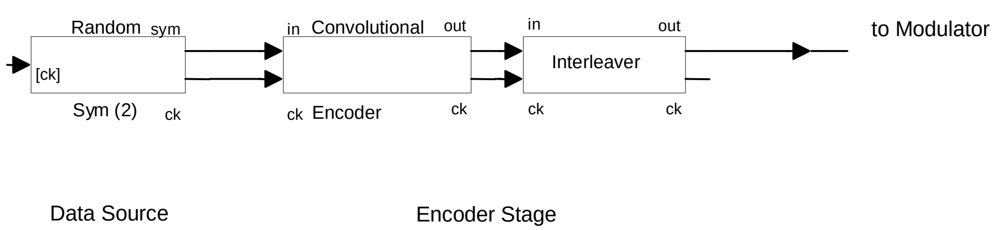
When working with digital sources, the simulation sampling frequency should be chosen so as to provide, at a minimum, on the order of 8 or 10 samples per symbol. This procedure will ensure that any filtering operations in the model will be accurately simulated. Of course, should there be another signal requiring a higher sampling frequency (e.g., a carrier), then the latter would dictate the necessary sampling frequency for the simulation.

Signal Encoding

As previously mentioned, signal encoding is performed to increase the reliability of information transfer. Signal encoding can assume several forms, such as the quantization of an analog signal, or the addition of smart redundancy into a digital data stream for FEC purposes.

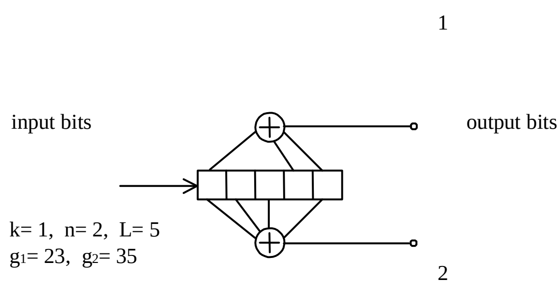
The quantization of an analog signal to obtain a digital data stream is usually performed using an Analog to Digital Converter (ADC). The VisSim/Comm environment provides several basic blocks to support this process, including an block and a data block. Companding is a nonlinear process used prior to quantization to provide higher resolution to the lower voltage range of a signal.

For digital signals, VisSim/Comm provides several methods for implementing forward error correction of digital data streams. Support is included for both convolutional codes and trellis coded modulation. Blocks to perform data interleaving and de-interleaving are also provided. The example below illustrates digital data encoding in VisSim/Comm.



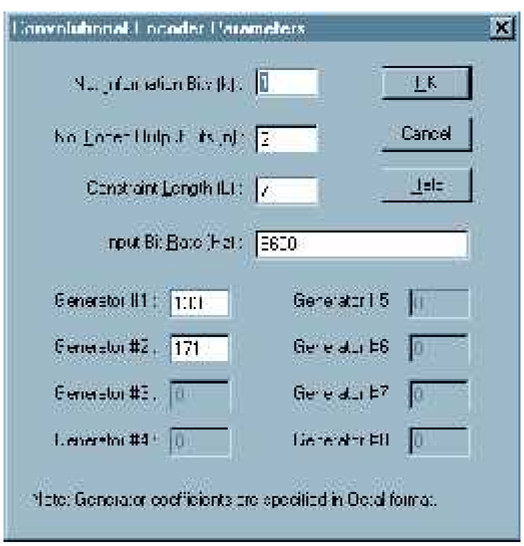
Convolutional Codes

Convolutional encoding is performed by passing a binary data sequence through a finite length shift register. A total of n outputs are then generated using different linear combinations (binary modulo addition) of the shift register contents. The specific tap connections are specified by the code generator coefficients, which are usually represented in octal format. Data bits are shifted k bits at a time into the encoder. The shift register size divided by k is referred to as the constraint length L of the encoder. The ratio k/n is referred to as the code rate, and specifies the amount of overhead added by the code. For example, for a code rate of ½, two channel bits are produced for each input information bit (100% overhead). This process is illustrated in the following figure.



Specifying a convolutional encoder in VisSim/Comm is very simple. The block will automatically generate the required encoder structure based on user-supplied generator coefficients and other block parameters. The example below is for a commonly used

R=½, k= 7 code.



Trellis Codes

Trellis coding differs from other coding methods because it combines into one operation the data encoding and constellation mapping steps. For this reason, the term trellis coded modulation is often used. The output of a trellis encoder is a baseband (I, Q) pair (i.e. a modulator constellation point). Unlike other codes, the transmitted channel symbol rate is often equal to the information symbol rate. This quality is highly desirable when the available channel bandwidth is limited. The coding overhead is hidden in an increase in the number of constellation points available to the transmitter. For example, the trellis coding approach used within the V.32 modem standard maps

a 16-ary input word (4 bits) to a 32 point signal constellation (5 bits). In this fashion, a 25% coding bit rate increase is achieved without a corresponding increase in the necessary channel bandwidth. As can be easily shown, signal bandwidth is related to the transmitted symbol rate, as opposed to the channel bit rate.

The trellis mapping is specified in VisSim/Comm via an external input file. This file specifies the modulator IQ outputs and the encoder new state value for each possible input value and current state combination.

The example below illustrates a few lines from the VisSim/Comm V.32 trellis map file.

V.32 Trellis Map File					(*1st header line*)
4	5	8			(*k, n, # states*)
0	0	0	-4	1	
0	1	0	0	-3	
…	…	…	…	…	
0	15	2	-2	1	
1	0	2	-4	1	
…	…	…	…	…	
7	15	7	-1	4	

In this example there are four input bits (*k*), indicating an input symbol range of 0 to 15. The number of coded output bits is five (*n*), which specifies one of 32 different constellation points. This particular trellis structure has eight states (0 - 7).

Data Interleaving

Most codes that have been developed perform well under uniform channel error conditions, as those produced by a Gaussian channel. When channel errors tend to occur in bursts, however, additional steps are required to maximize the performance of the coding process. One common technique is to scramble the channel bits prior to transmission by using an interleaver.

Data interleaving alters a coded bit stream so as to minimize the effects of burst type channel degradations. By using this technique, adjacent bit errors caused by signal fades or bursts of interference are spread out in time across a longer interval, giving the decoder a greater probability of recovering the original data stream. The following figure illustrates the operation of a very simple 4x4 data interleaver.

Modulation

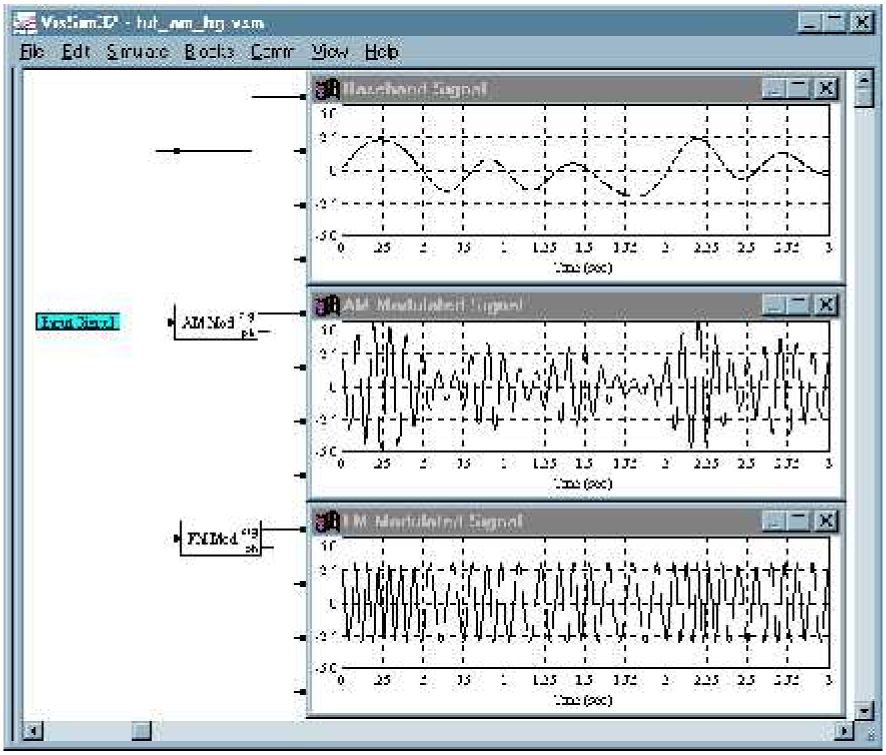
In order to send an information signal from one point to another, the signal must often be altered prior to transmission. This process is commonly known as *modulation*. There are many reasons why it is desirable for a new signal to represent the original waveform. The most common reasons relate to bandwidth allocation and signal recovery considerations. The modulation process, for example, allows the original baseband information signal to be translated in frequency so that many signals can coexist simultaneously without interfering with each other. This is achieved by allocating each modulated signal to a slightly different region of the available frequency spectrum.

Many types of modulation techniques have been devised for representing an information signal as it is being transmitted. In general, all modulation schemes rely somehow on varying the amplitude, phase or frequency of a carrier waveform. At a high level, modulation techniques can be subdivided into two basic groups: analog modulation and digital modulation.

Analog Modulation

In analog modulation, the transmitted signal can be varied continuously over a specified range as opposed to assuming a fixed number of predetermined states. Examples of analog modulation techniques include Amplitude Modulation (AM), Phase Modulation (PM), and Frequency Modulation (FM). As the name implies, an AM transmitter operates by varying the amplitude of the carrier according to the voltage of the input signal. In a PM transmitter, the input signal is used to control the instantaneous phase of the carrier phase. With FM, on the other hand, the input signal is used to vary the instantaneous frequency of the carrier. In all three cases the input signal is an analog signal, such as a voice signal.

The following VisSim example compares the AM and FM modulation techniques using an arbitrary input waveform. The same carrier frequency is used in generating both the AM and FM waveforms.

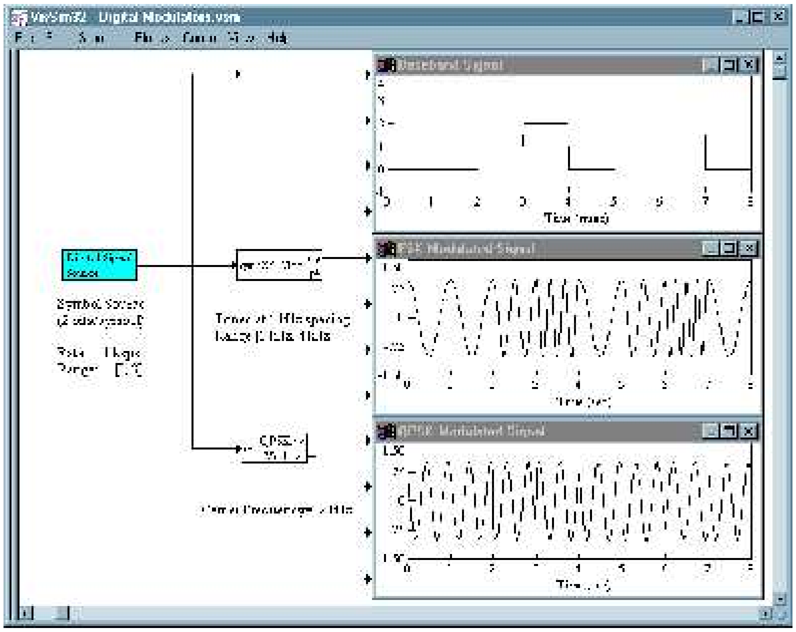


Digital Modulation

In digital modulation, the transmitted signal can assume only a fixed number of predetermined states, usually referred to as the *alphabet size* or *constellation size* of the modulated signal. These include discrete amplitude levels, discrete phases, discrete frequencies, or combinations of the above. Examples of digital modulation techniques include Phase Shift Keying (PSK), Quadrature Amplitude Modulation (QAM), Frequency Shift Keying (FSK) and Pulse Position Modulation (PPM). Each of the above techniques can be implemented at various levels of complexity depending primarily on the total number of distinct states (constellation points) that are allowed within the modulator.

Digital modulation has inherent benefits over analog modulation because its distinct transmission states can more easily be detected at a receiver in the presence of noise than an analog signal, which can assume an infinite number of values. When the transmitted signal originates as an analog waveform, a trade-off occurs at the encoding stage since some information is always lost in the quantization process.

The following VisSim example compares two digital modulation techniques using an arbitrary digital input signal. The top shows the baseband input signal which can assume one of four specified levels [0 – 3]. The additional two show the difference between the output of a PSK modulator and an FSK modulator given the same input signal. PSK and FSK are described in more detail later in this section.



FSK and MSK Modulation

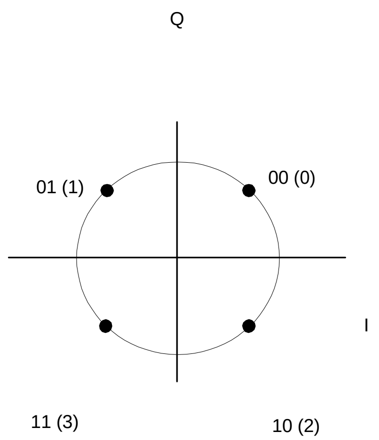
In Frequency Shift Keying (FSK) modulation, the digital information is transmitted by assigning discrete output frequencies to each of the possible input symbols. The carrier amplitude remains constant. Two classes of FSK modulators exist: those that maintain a continuous carrier phase between states, and those that do not. The first case applies when a frequency synthesizer is used

to generate the modulated output. The latter occurs when multiple independent oscillators are used to generate the various FSK tones. The bandwidth occupied by an FSK signal is directly proportional to the signaling rate. VisSim/Comm provides a basic block and also an (Minimum Shift Keying) block, which represents a special case of continuous phase FSK where the two output tones are spaced exactly by the symbol rate R. An example of FSK modulation is shown in the figure on the preceding page.

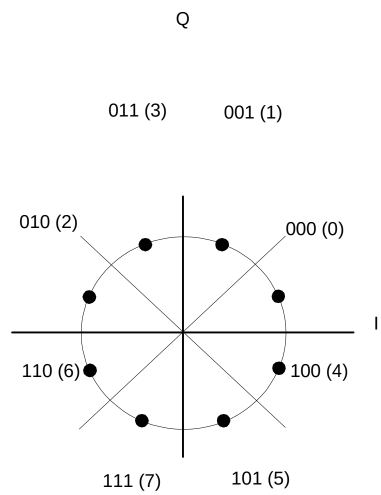
PSK Modulation

In PSK modulation, the digital information is transmitted by varying the carrier phase between known phase states. The carrier amplitude remains constant. Among its desirable properties are its constant envelope characteristic, as all constellation points have equal power. The bandwidth occupied by a PSK signal is directly proportional to the symbol rate. Thus as the constellation size increases, no additional bandwidth is required. Of course, as the constellation points move closer together, higher power is required to maintain a given BER performance level.

VisSim/Comm offers the following PSK formats: BPSK, QPSK, SQPSK, 8-PSK, and 16-PSK. The user is given complete control over the mapping from input value to output constellation point. The examples below illustrate the signal constellations in the (I, Q) plane for QPSK and 8- PSK using a Gray encoded constellation mapping. Such a mapping ensures that neighboring constellation points differ by only one bit.

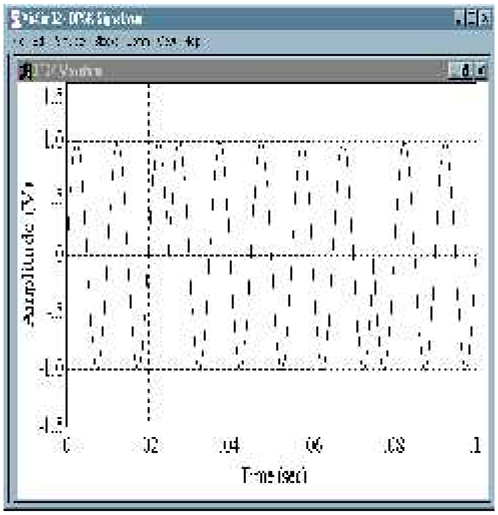


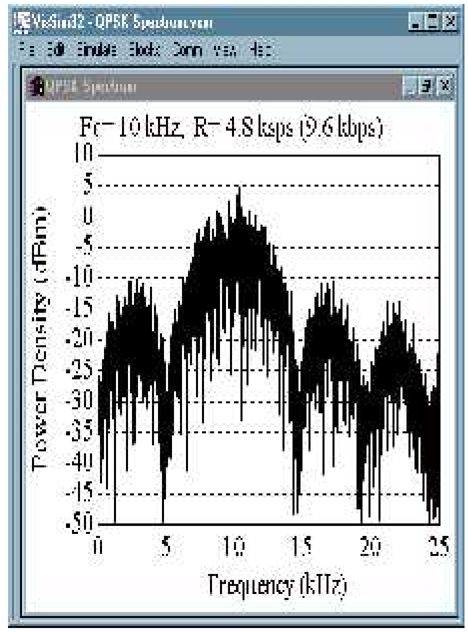
QPSK



8 PSK

An example of a BPSK modulated waveform and the spectrum of a QPSK signal are shown in the illustrations below. The null to null bandwidth of a PSK signal is $2R$, where R represents the symbol signaling rate. For QPSK, the symbol rate is ½ the bit rate, as each constellation point represents two bits. The output spectrum is centered about the modulator carrier frequency f_C.





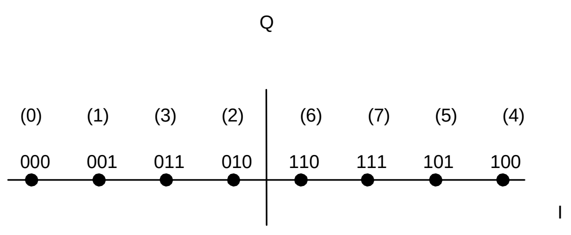
QAM Modulation

In QAM modulation, the digital information is transmitted by varying both the carrier phase and carrier amplitude between known constellation points in the (I, Q) plane. While QAM signals are not constant envelope, they make very efficient use of available bandwidth, with constellation sizes up to 256 points being common. As with PSK signals, the null to null bandwidth of a QAM modulated waveform is $2R$, where R represents the *symbol* signaling rate. For 16-QAM, the symbol rate is ¼ the bit rate as each constellation point represents four bits. The output spectrum is centered about the modulator carrier frequency f_C.

VisSim/Comm offers the following QAM formats: 16-QAM, 32-QAM, 64-QAM, and 256-QAM. The user is given complete control over the mapping from input value to output constellation point. The following example illustrates the signal constellation in the (I, Q) plane for 16-QAM using a Gray encoded mapping.

PAM Modulation

In pulse Amplitude Modulation (PAM) modulation, also known as Amplitude Shift Keying (ASK), the digital information is transmitted by varying the carrier amplitude between known discrete levels. The following PAM formats are offered in VisSim/Comm: 4-PAM and 8-PAM. The user is given complete control over the mapping from input value to output constellation point. The example below illustrates the signal constellation in the (I, Q) plane for 8-PAM using a Gray encoded constellation mapping.



8 PAM

Differential Modulation Formats

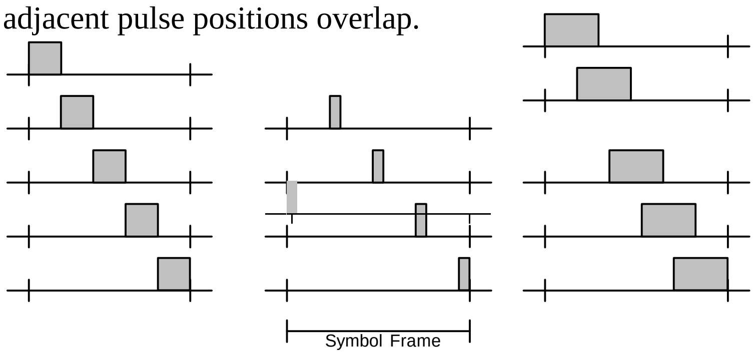
VisSim/Comm also offers several differential modulation formats, such as DQPSK and π/4-DQPSK. In addition the user can easily convert any digital modulator into a differential modulator by appropriately modifying the input bit stream(s) into the transmitter.

Differential modulation schemes are used when coherent phase detection cannot be readily implemented. Rather than mapping each information symbol to a particular carrier phase state, a differential modulator will map the input signal bits into carrier phase transitions, which from a detection point of view can be more easily identified. The table below illustrates the mapping from input symbol to output phase transition for both DQPSK and π/4-DQPSK.

Input Symbol	*DQPSK*	*π/4-DQPSK*
0	0°	45°
1	90°	135°
2	-90°	-45°
3	180°	-135°

Pulse Position Modulation

In Pulse Position Modulation (PPM) the information is transmitted by varying the occurrence of a rectangular or shaped pulse within a pre-defined symbol frame. The location of the pulse is proportional to the input signal level. This scheme requires a synchronized transmitter and receiver to maintain symbol frame timing. As shown in the following illustration, the pulse width can be varied from relatively narrow to wide enough that adjacent pulse positions overlap.



Study of MATLAB

Introduction

The tutorials are independent of the rest of the document. The primarily objective is to help you learn quickly the first steps. The emphasis here is \learning by doing". Therefore, the best way to learn is by trying it yourself. Working through the examples will give you a feel for the way that MATLAB operates. In this introduction we will describe how MATLAB handles simple numerical expressions and mathematical formulas.

The name MATLAB stands for MATrix LABoratory. MATLAB was written originally to provide easy access to matrix software developed by the LINPACK (linear system package) and
EISPACK (Eigen system package) projects. MATLAB [1] is a high-performance language for technical computing. It integrates computation, visualization, and programming environment. Furthermore, MATLAB is a modern programming language environment: it has sophisticated data structures, contains built-in editing and debugging tools, and supports object-oriented programming. These factors make MATLAB an excellent tool for teaching and research.

MATLAB has many advantages compared to conventional computer languages (e.g., C, FORTRAN) for solving technical problems. MATLAB is an interactive system whose basic data element is an array that does not require dimensioning. The software package has been commercially available since 1984 and is now considered as a standard tool at most universities and industries worldwide.

It has powerful built-in routines that enable a very wide variety of computations. It also has easy to use graphics commands that make the visualization of results immediately available. Specific applications are collected in packages referred to as toolbox. There are toolboxes for signal processing, symbolic computation, control theory, simulation, optimization, and several other fields of applied science and engineering.

Later and with the addition of several toolboxes the capabilities of Matlab were expanded and today it is a very powerful tool at the hands of an engineer.

- Typical uses include:
- Math and Computation
- Algorithm development
- Modelling, simulation and prototyping
- Data analysis, exploration and visualisation
- Scientific and engineering graphics
- Application development, including graphical user interface building.

Matlab is an interactive system whose basic data element is an ARRAY. Perhaps the easiest way to visualise Matlab is to think it as a full-featured calculator. Like a basic calculator, it does simple math like addition, subtraction, multiplication and division. Like a scientific calculator it handles square roots, complex numbers, logarithms and trigonometric operations such as sine, cosine and tangent. Like a programmable calculator, it can be used to store and retrieve data; you can create, execute and save sequence of commands, also you can make comparisons and control the order in which the commands are executed. And finally as a powerful calculator it allows you to perform matrix algebra, to manipulate polynomials and to plot data.

Basic features

As we mentioned earlier, the following tutorial lessons are designed to get you started quickly in MATLAB. The lessons are intended to make you familiar with the basics of MATLAB.

A Minimum MATLAB session

The goal of this minimum session (also called starting and exiting sessions) is to learn the first steps:

How to log on

- Invoke MATLAB
- Do a few simple calculations
- How to quit MATLAB

Starting MATLAB

After logging into your account, you can enter MATLAB by double-clicking on the MATLAB shortcut icon (MATLAB R2013a) on your Windows desktop. When you start MATLAB, a special window called the MATLAB desktop appears. The desktop is a window that contains other windows.

When MATLAB is started for the first time, the screen looks like the one that shown in the above figure. This illustration also shows the default configuration of the MATLAB desktop.

Now, we are interested in doing some simple calculations. We will assume that you have sufficient understanding of your computer under which MATLAB is being run.

Using MATLAB as a calculator As an example of a simple interactive calculation, just type the expression you want to evaluate. Let's start at the very beginning. For example, let's suppose you want to calculate the expression, 1 + 2 x 3. You type it at the prompt command (>>) as follows,

```
>> x=1+2*3
>> ans=7
```

will result in x being given the value 1 + 2 *3 = 7. This variable name can always be used to refer to the results of the previous computations. Therefore, computing 4x will result in

```
        >> 4*x
ans = 28.0000
```

Before we conclude this minimum session, Table 1.1 gives the partial list of arithmetic operators.

Quitting MATLAB

To end your MATLAB session, type quit in the Command Window, or select File Exit MATLAB in the desktop main menu.

Getting started

After learning the minimum MATLAB session, we will now learn to use some additional operations.

Creating MATLAB variables

MATLAB variables are created with an assignment statement. The syntax of variable assignments

For example,

variable name = a value (or an expression) >> x = expression

where expression is a combination of numerical values, mathematical operators, variables, and function calls. On other words, expression can involve:

- manual entry
- built-in functions
- user-defined functions

Overwriting variable

Once a variable has been created, it can be reassigned. In addition, if you do not wish to see the intermediate results, you can suppress the numerical output by putting a semicolon (;) at the end of the line. Then the sequence of commands looks like this:

```
>> t = 5;
>>      t = t+1 t = 6
```

Error messages

If we enter an expression incorrectly, MATLAB will return an error message. For example, in the following, we left out the multiplication sign, *, in the following expression

```
>> x = 10;
>> 5x
??? 5x
    |Error: Unexpected MATLAB expression.
```

Making corrections

To make corrections, we can, of course retype the expressions. But if the expression is lengthy, we make more mistakes by typing a second time. A previously typed command can be recalled with the up-arrow key ". When the command is displayed at the command prompt, it can be modified if needed and executed.

Controlling the hierarchy of operations or precedence

Let's consider the previous arithmetic operation, but now we will include parentheses. For example, 1 + 2 X 3 will become (1 + 2) X3

```
>>      (1+2)*3 ans = 9
```

and, from previous example

```
>> 1+2*3
ans=7
```

By adding parentheses, these two expressions give different results: 9 and 7.

The order in which MATLAB performs arithmetic operations is exactly that taught in high school algebra courses. Exponentiations are done first, followed by multiplications and divisions, and finally by additions and subtractions. However, the standard order of precedence of arithmetic operations can be changed by inserting parentheses.

For example, the result of 1+2£3 is quite different than the similar expression with parentheses (1+2)X3.

The results are 7 and 9 respectively. Parentheses can always be used to overrule priority, and their use is recommended in some complex expressions to avoid ambiguity.

Therefore, to make the evaluation of expressions unambiguous, MATLAB has established a series of rules. The order in which the arithmetic operations are evaluated is given in Table 1.2. MATLAB arithmetic operators obey the same precedence rules as those in most computer programs. For operators of equal precedence, evaluation is from left to right.

Table 1.2: Hierarchy of arithmetic operations

PRECEDENCE	MATHEMATICAL OPERATIONS
First	The contents of all parentheses are evaluated first, starting from the innermost parentheses and working outward.
Second	All exponentials are evaluated, working from left to right
Third	All multiplications and divisions are evaluated, working from left to right
Fourth	All additions and subtractions are evaluated, starting from left to right

Controlling the appearance of floating point number

MATLAB by default displays only 4 decimals in the result of the calculations, for example - 163:6667, as shown in above examples. However, MATLAB does numerical calculations in double precision, which is 15 digits. The command format controls how the results of computations are displayed. Here are some examples of the different formats together with the resulting outputs.

```
>> format short
>> x=-163.6667
```

If we want to see all 15 digits, we use the command format long

```
>> format long
>> x= -1.636666666666667e+002
```

To return to the standard format, enter format short, or simply format.

There are several other formats. For more details, see the MATLAB documentation, or type help format.

Note - Up to now, we have let MATLAB repeat everything that we enter at the prompt (>>). Sometimes this is not quite useful, in particular when the output is pages en length. To prevent MATLAB from echoing what we type, simply enter a semicolon (;) at the end of the command. For example,

```
>> x=-163.6667;
```

and then ask about the value of x by typing, >> x

x = -163.6667

Managing the workspace

The contents of the workspace persist between the executions of separate commands. Therefore, it is possible for the results of one problem to have an effect on the next one. To avoid this possibility, it is a good idea to issue a clear command at the start of each new independent calculation.

```
>> clear
```

The command clear or clear all removes all variables from the workspace. This frees up system memory. In order to display a list of the variables currently in the memory,

type >> who

while, whos will give more details which include size, space allocation, and class of the variables.

Keeping track of your work session

It is possible to keep track of everything done during a MATLAB session with the diary command.

```
>> diary
```

or give a name to a created file, >> diary FileName

where FileName could be any arbitrary name you choose.

The function diary is useful if you want to save a complete MATLAB session. They save all input and output as they appear in the MATLAB window. When you want to stop the recording, enter diary off. If you want to start recording again, enter diary on. The ¯le that is created is a simple text ¯le. It can be opened by an editor or a word processing program and edited to remove extraneous material, or to add your comments. You can use the function type to view the diary file or you can edit in a text editor or print. This command is useful, for example in the process of preparing a homework or lab submission.

Entering multiple statements per line

It is possible to enter multiple statements per line. Use commas (,) or semicolons (;) to enter more than one statement at once. Commas (,) allow multiple statements per line without suppressing output.

```
>> a=7; b=cos(a),
c=cosh(a) b = 0.6570
c = 548.3170
```

Miscellaneous Commands

Here are few additional useful commands:

- To clear the Command Window, type clc
- To abort a MATLAB computation, type ctrl-c
- To continue a line, type . . .

Getting help

To view the online documentation, select MATLAB Help from Help menu or MATLAB Help directly in the Command Window. The preferred method is to use the Help Browser. The Help Browser can be started by selecting the ? icon from the desktop toolbar. On the other hand, information about any command is available by typing

```
>> help Command
```

Another way to get help is to use the look for command. The look for command differs from the help command. The help command searches for an exact function name match, while the look for command searches the quick summary information in each function for a match. For example, suppose that we were looking for a function to take the inverse of a matrix. Since MATLAB does not have a function named inverse, the command help inverse will produce nothing. On the other hand, the command lookfor inverse will produce detailed information, which includes the function of interest, inv.

```
>> lookfor inverse
```

Note - At this particular time of our study, it is important to emphasize one main point. Because MATLAB is a huge program; it is impossible to cover all the details of each function one by one. However, we will give you information how to get help. Here are some examples:

Use on-line help to request info on a specific function >> help sqrt

In the current version (MATLAB version 7), the doc function opens the on-line version of the help manual. This is very helpful for more complex commands.

```
>> doc plot
```

AMPLITUDE MODULATOR AND DEMODULATOR

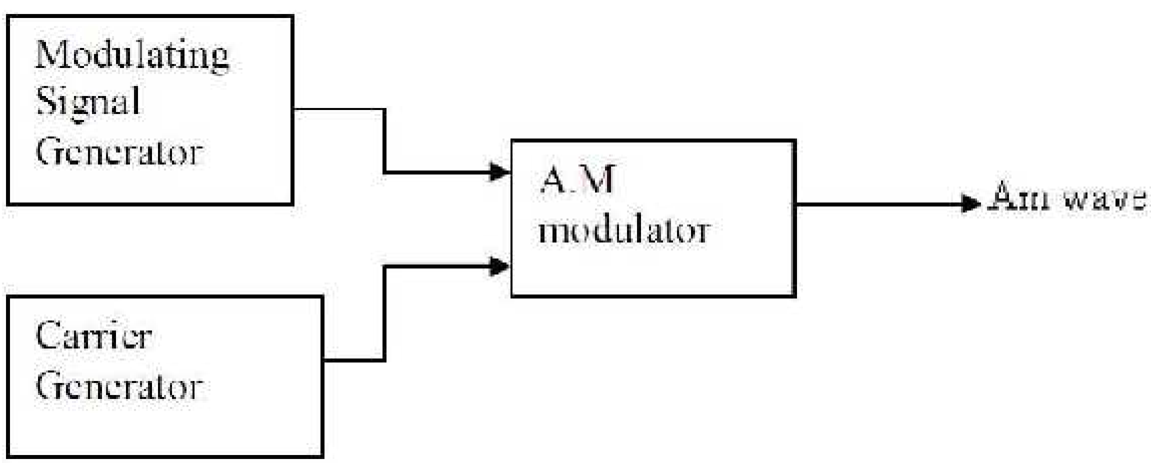
AIM:

To study the function of Amplitude Modulation & Demodulation (under modulation, perfect modulation & over modulation) and also to calculate the modulation index, efficiency, total power

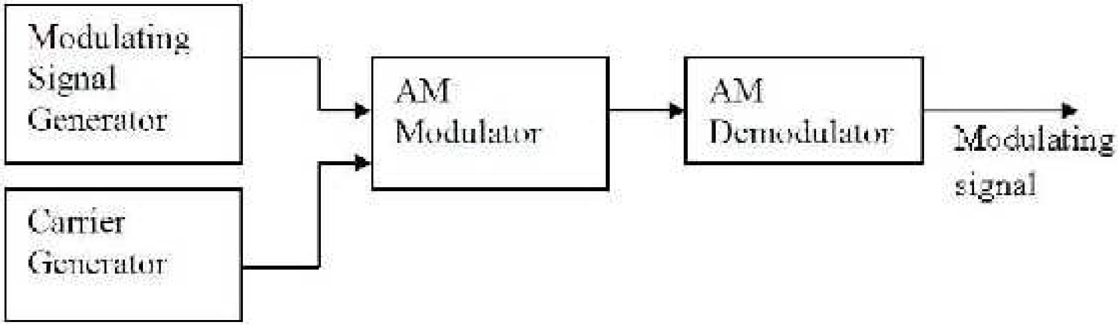
APPARATUS:

1. Amplitude Modulation & De modulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting cords & probes.
5. PC with windows(95/98/XP/NT/2000)
6. Amplitude Modulation & De modulation trainer kit.

BLOCK DIAGRAM:



AM MODULATOR



AM DEMODULATOR

THEORY :

Amplitude modulation (AM) is defined as a process in which the amplitude of the carrier wave c(t) is varied about a mean value, linearly with the base band signal m(t).

An AM wave may thus be described, in its most general form, as a function of time as follows.

$$S(t)=A\,[1+K_a m(t)]\,\text{Cos}\,(2\pi f_c t)$$

The amplitude of $K_a m(t)$ is always less than unity, that is $|K_a m(t)| <1$ for all t. It ensures that the function $1+K_a m(t)$ is always positive. When the amplitude sensitivity K_a of the modulator is large enough to make $|K_a m(t)| >1$ for any t, the carrier wave becomes over modulated, resulting in carrier phase reversals. whenever the factor $1+K_a m(t)$ crosses zero.

The absolute maximum value of $K_a m(t)$ multiplied by 100 is referred to as the percentage modulation.

PROCEDURE:

1. Connect the AC Adapter to the mains and the other side to the Experimental Trainer. Switch „ON" the power.

2. Observe the carrier and modulating waveforms and note their frequencies. (Carrier frequency is around 100 KHz and amplitude is variable from 0 -8Vp-p, modulating signal is 1KHz).

3. Connect the carrier and modulating signals to the modulator circuit.

4. Observe the amplitude modulated wave.

5. Connect Carrier I/P to ground and apply a 2V peak to peak AF Signal input to (modulating I/P)and adjust P1 in anti-clock wise position to get minimum A.C output.

6. Connect modulating I/P to ground and apply a 3V peak to peak carrier signal to carrier I/P andadjust P2 in clock wise direction to get minimum A.C output..

7. Connect modulating input &carrier input to ground and adjust P3 for zero D.C output.

8. Make modulating i/p 2 Vpp and carrier i/p 3 Vpp peak to peak and adjust potentiometer P4 formaximum output.

9. Calculate maximum and minimum points on the modulated envelope on a CRO and calculatethe depth of modulation.

10. Observe that by varying the modulating voltage, the depth of modulation varies.

11. During demodulation connect this AM output to the input of the demodulator.

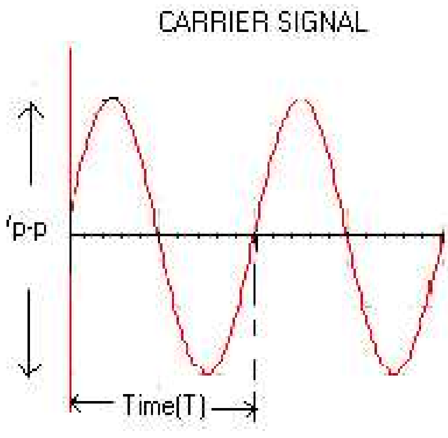
12. By adjusting the RC time constant (i.e., cut off frequency) of the filter circuit we getminimum distorted output.

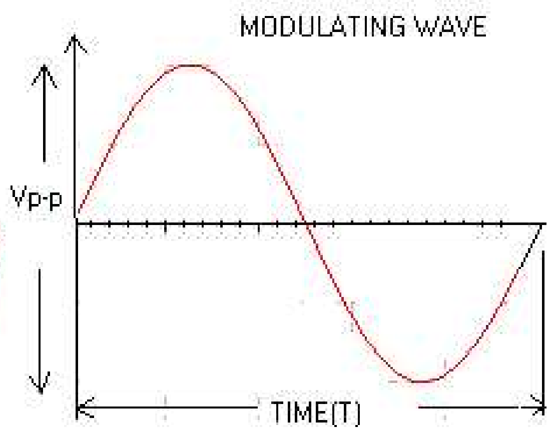
13. Observe that this demodulated output is amplified has some phase delay because of RCcomponents.

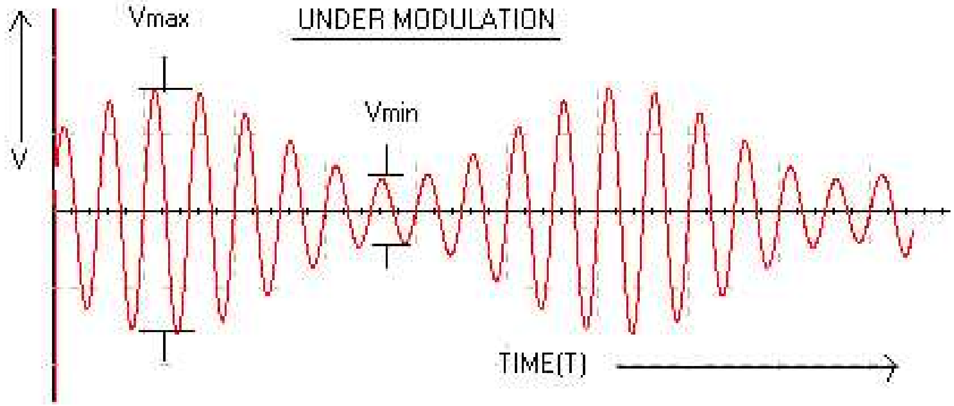
14. Also observe the effects by changing the carrier amplitudes.

15. In all cases, calculate the modulation index.

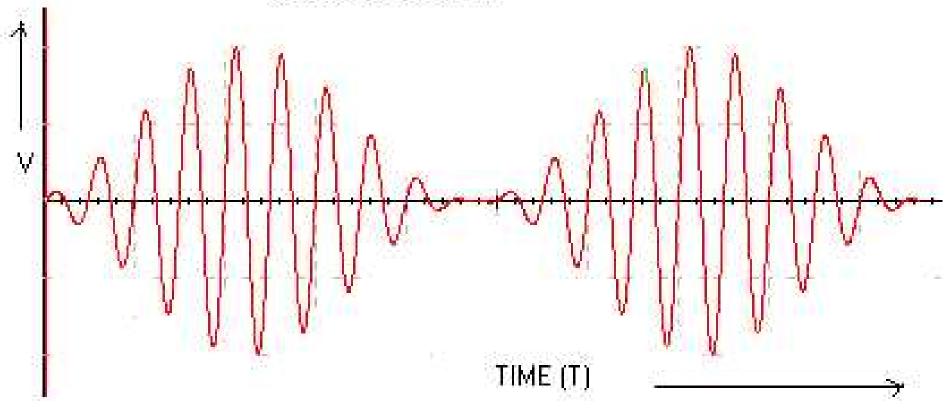
Expected Wave Forms



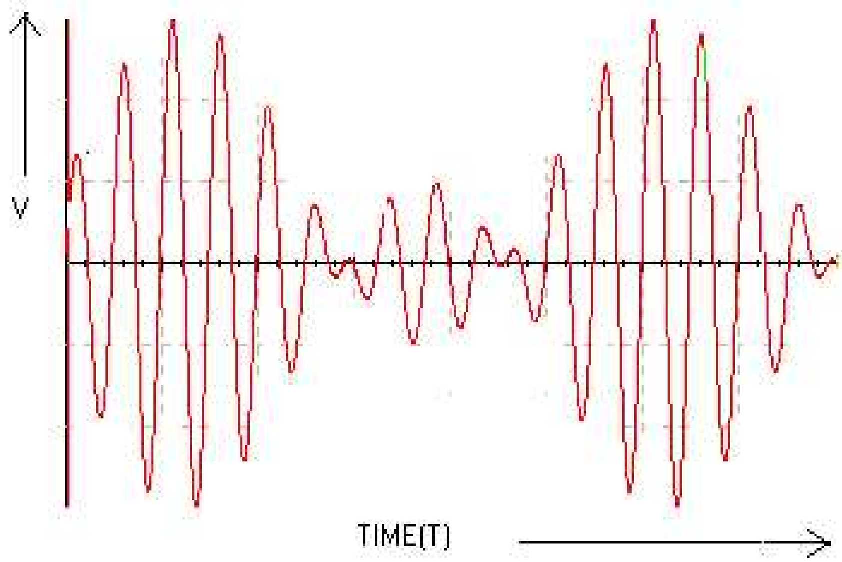




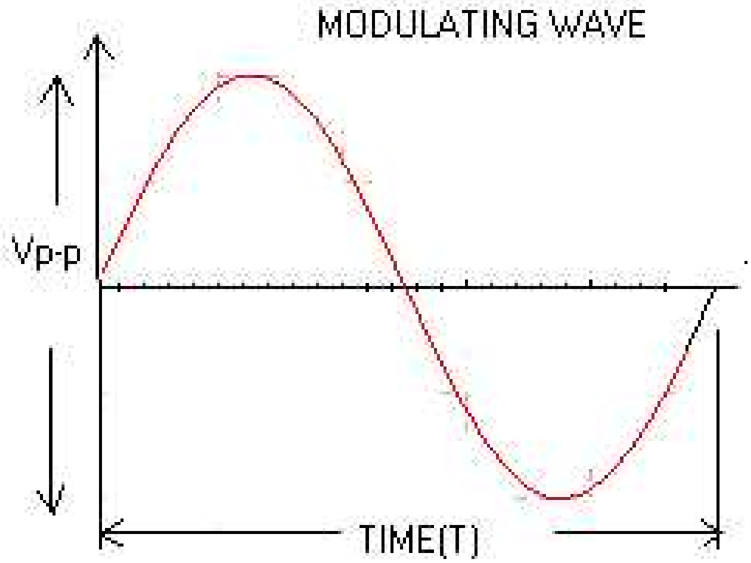
PERFECT MODULATION/
100% MODULATION



OVER MODULATION



Demodulated signal



OBSERVATIONS:

MODULATION:

MODULATIONS	V_c	V_m	V_{max}	V_{min}	$\mu = \frac{(V_{max} - V_{min})}{(V_{max} + V_{min})}$	$\mu = \frac{V_m}{V_c}$
Under modulation						
Perfect modulation						
Over modulation						

DEMODULATION:

Modulating signal frequency	Demodulated signal frequency

RESULT :

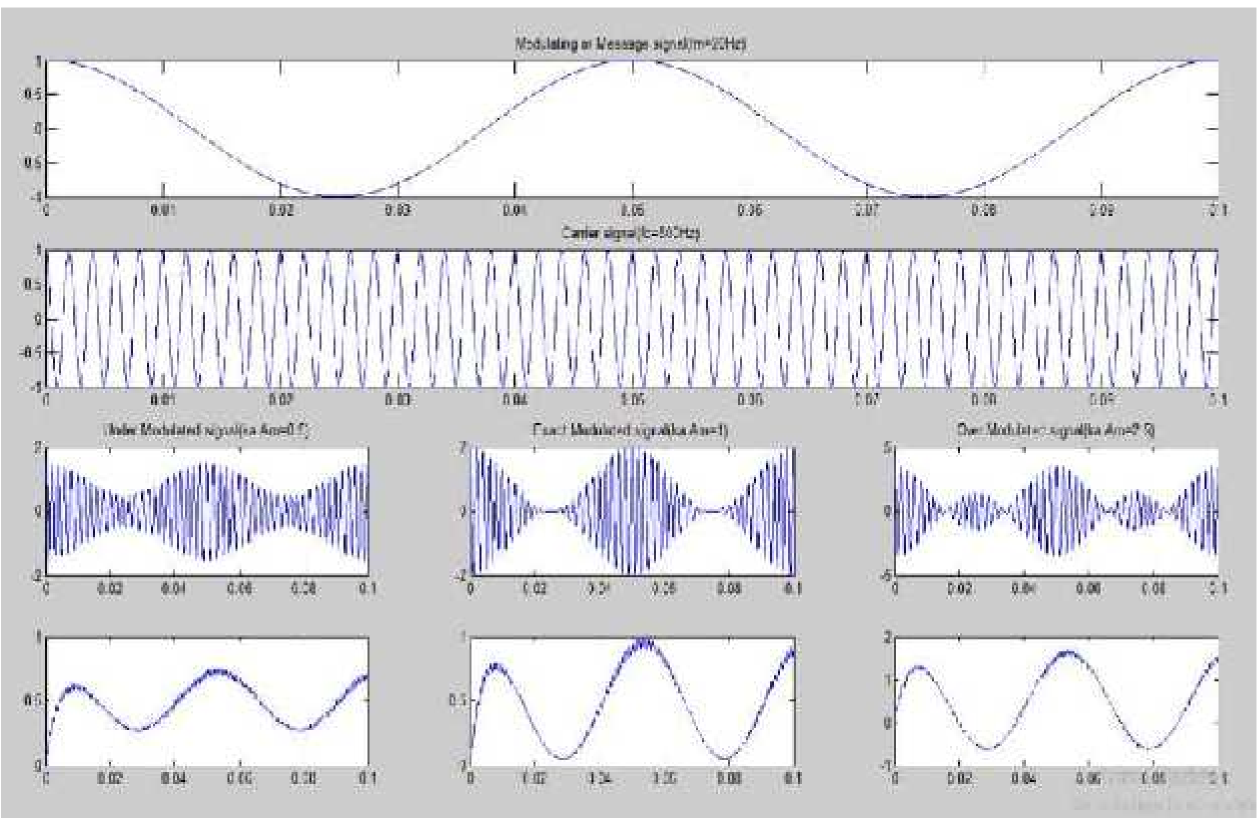
Thus the Amplitude modulated wave is observed for different modulation indexes.

CONCLUSION:

In amplitude modulation by increasing the message amplitude we observed different modulationindexes such as under modulation ($\mu<1$), over modulation($\mu>1$) and exact modulation($\mu=1$).

MATLAB CODE:

```
fs=8000; fm=20; fc=500; Am=1; Ac=1;
t=[0:.1*fs]/fs;
m=Am*cos(2*pi*fm*t); c=Ac*cos(2*pi*fc*t); ka=0.5;
u=ka*Am; s1=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t); subplot(4,3,1:3);
plot(t,m);
title('Modulating or Message signal(fm=20Hz)'); subplot(4,3,4:6);
plot(t,c);
title('Carrier signal(fc=500Hz)'); subplot(4,3,7);
plot(t,s1);
title( 'Under Modulated signal(ka.Am=0.5)'); Am=2;
ka=0.5;
u=ka*Am; s2=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t); subplot(4,3,8);
plot(t,s2);
title( 'Exact Modulated signal(ka.Am=1)'); Am=5;
ka=0.5;
u=ka*Am; s3=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t); subplot(4,3,9);
plot(t,s3);
title('Over Modulated signal(ka.Am=2.5)'); r1= s1.*c;
[b a] = butter(1,0.01);
mr1= filter(b,a,r1); subplot(4,3,10);
plot(t,mr1); r2= s2.*c;
[b a] = butter(1,0.01);
mr2= filter(b,a,r2); subplot(4,3,11);
plot(t,mr2);
r3= s3.*c;
[b a] = butter(1,0.01); mr3= filter(b,a,r3); subplot(4,3,12);
plot(t,mr3);
```



POST LAB QUESTIONS

1. Define AM and draw its spectrum?
2. Draw the phase's representation of an amplitude modulated wave?
3. Give the significance of modulation index?
4. What are the different degrees of modulation?
5. What are the limitations of square law modulator?

EVALUATION SHEET

Department of Electronics and Communication Engineering

SCSVMV UNIVERSITY, Enathur, Kanchipuram

Title of the Experiment	
Name of the Candidate	
Register Number	
Date of Submission	

S.No	Marks Split up	Maximum Marks	Marks Earned
1	Attendance	5	
2	Pre lab viva questions	5	
3	Execution of Experiments	20	
4	Calculation/Evaluation of Result	10	
5	Post lab viva questions	10	
6	Grand Total	50	

Signature of the Faculty Handling the Lab

DSB-SC MODULATION & DETECTION

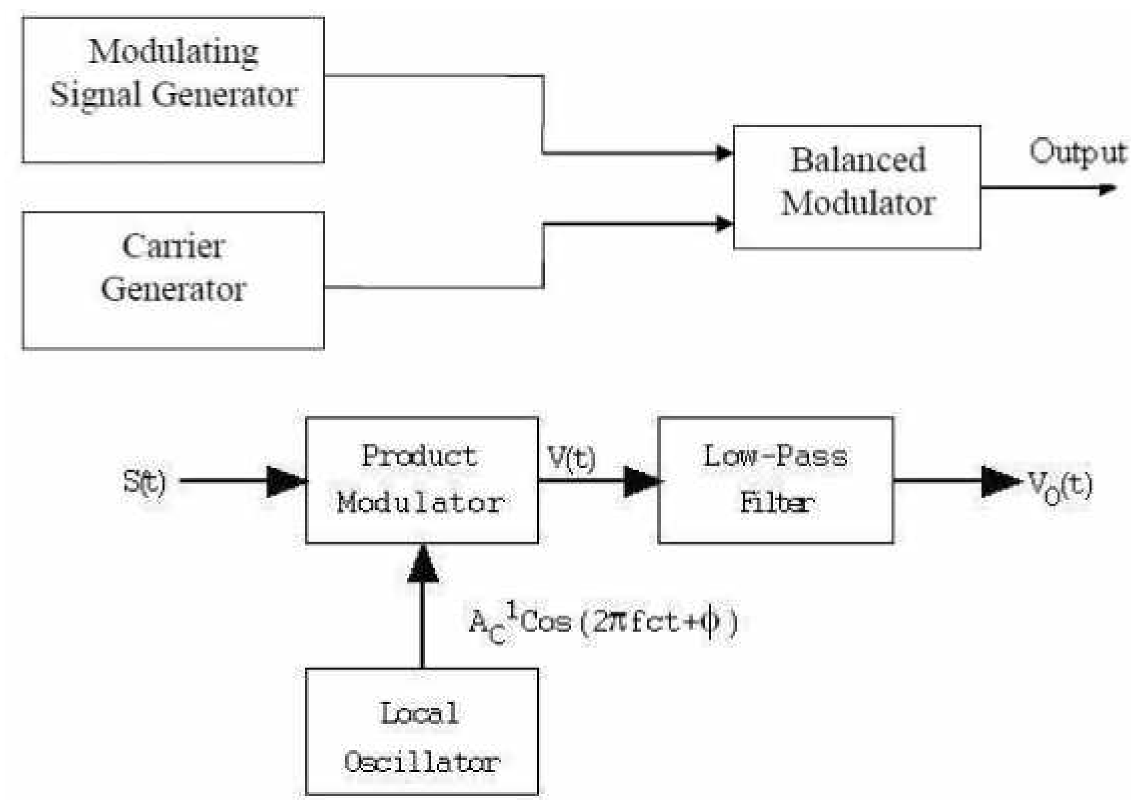
AIM:

To study and observe the function of Balanced Modulator and Demodulator, observe its efficiency

APPARATUS:

1. Amplitude Modulation & De modulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting cords & probes.
5. PC with windows(95/98/XP/NT/2000)
6. DSBSC Modulation & De modulation trainer kit.

BLOCK DIAGRAM: DSBSC MODULATOR &DEMODULATOR



THEORY:

The carrier of amplitude modulation wave does not convey any information. It is obvious from the fact that the carrier component remains constant in amplitude and frequency. No matter what the modulating signal does. It is thus, seen that no information is conveyed by the carrier. If the carrier is suppressed, only the side bands remains and a saving of two third powers can achieve at 100% modulation such suppression of carrier doesn't affect the massage signal in any way. This idea has resulted in the evolution of suppressed carrier modulation. Thus, the short coming of the conventional AM in regard of power wastage is overcome by suppressing the carrier from the modulated wave resulting in double side band suppressed carrier modulation. A balanced is used to generate DSBSC wave. A DSBSC signal is basically the product of the base band signal and the carrier wave.S(t) = m(t) *c(t)

Where m(t) is base band signal C(t) is carrier signal C(t) = A_c cost 2Π fct

The modulated wave under goes a phase reversal when ever base band signal m(t) crosses zero. Spectrum of base band signal

$S(f) = A_C/2\ [(M(f-f_c) + M(f+f_c)]$

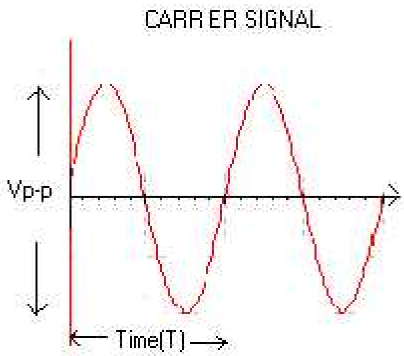
Where M (f) is the fourier trans form of m(t)A_c is carrier amplitude
And f_c is frequency of the carrier.

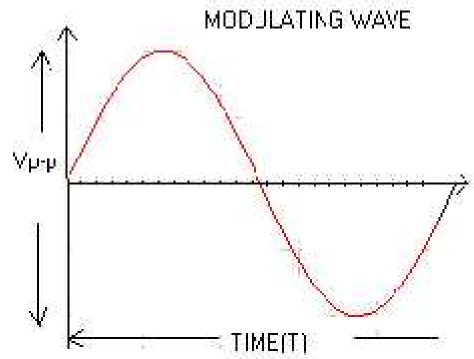
The band width of DSBSC signal is same as that of conventional AM i.e., 2W.The base band signal m(t) can be uniquely recovered from a DSB-SC wave S(t) by first multiplying s(t) with a locally generated sinusoidal wave and then low-pass filtering the product, as in fig. below. It is assumed that the local oscillator signal is exactly coherent or synchronized, in both frequency and phase, with the carrier wave C(t) used in the product modulator to generate S(t). This method of demodulation is known as Coherent or Synchronous demodulation.

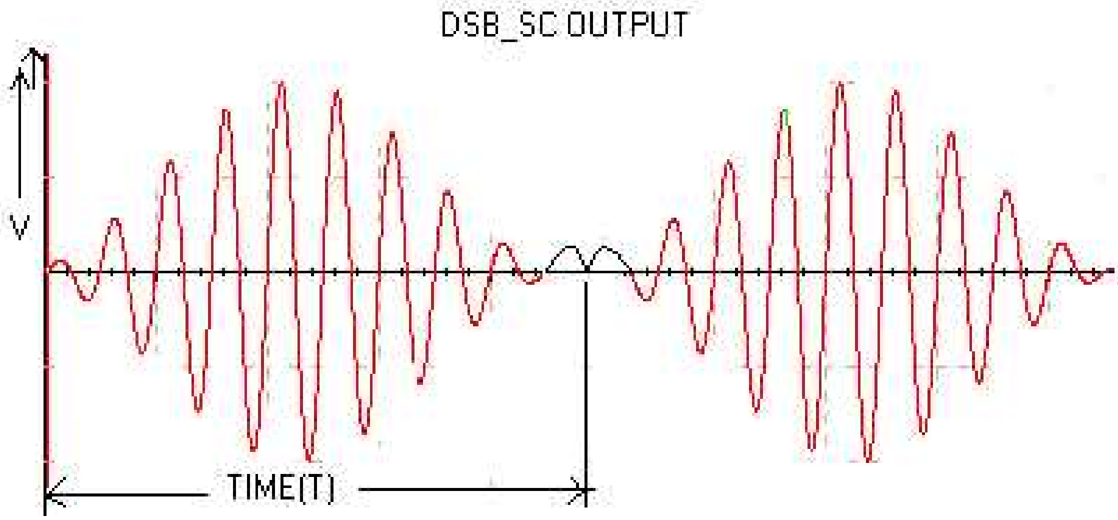
PROCEDURE:

1. Apply DSB-SC signal to DSB-SC signal input of the synchronous detector and RF generatoroutput to RF input of synchronous detector.
2. Observe the synchronous detector output on CRO and compare it with the original AF signal.

EXPECTED WAVEFORMS:







OBSERVATIONS:

Carrier Signal		Message Signal		Modulated SignalOutput		Demodulated signalOutput	
F_c(Hz)	V_c(volts)	F_m(Hz)	V_m(volts)	F_o(Hz)	V_o(volts)	F(Hz)	V(volts)

RESULT:

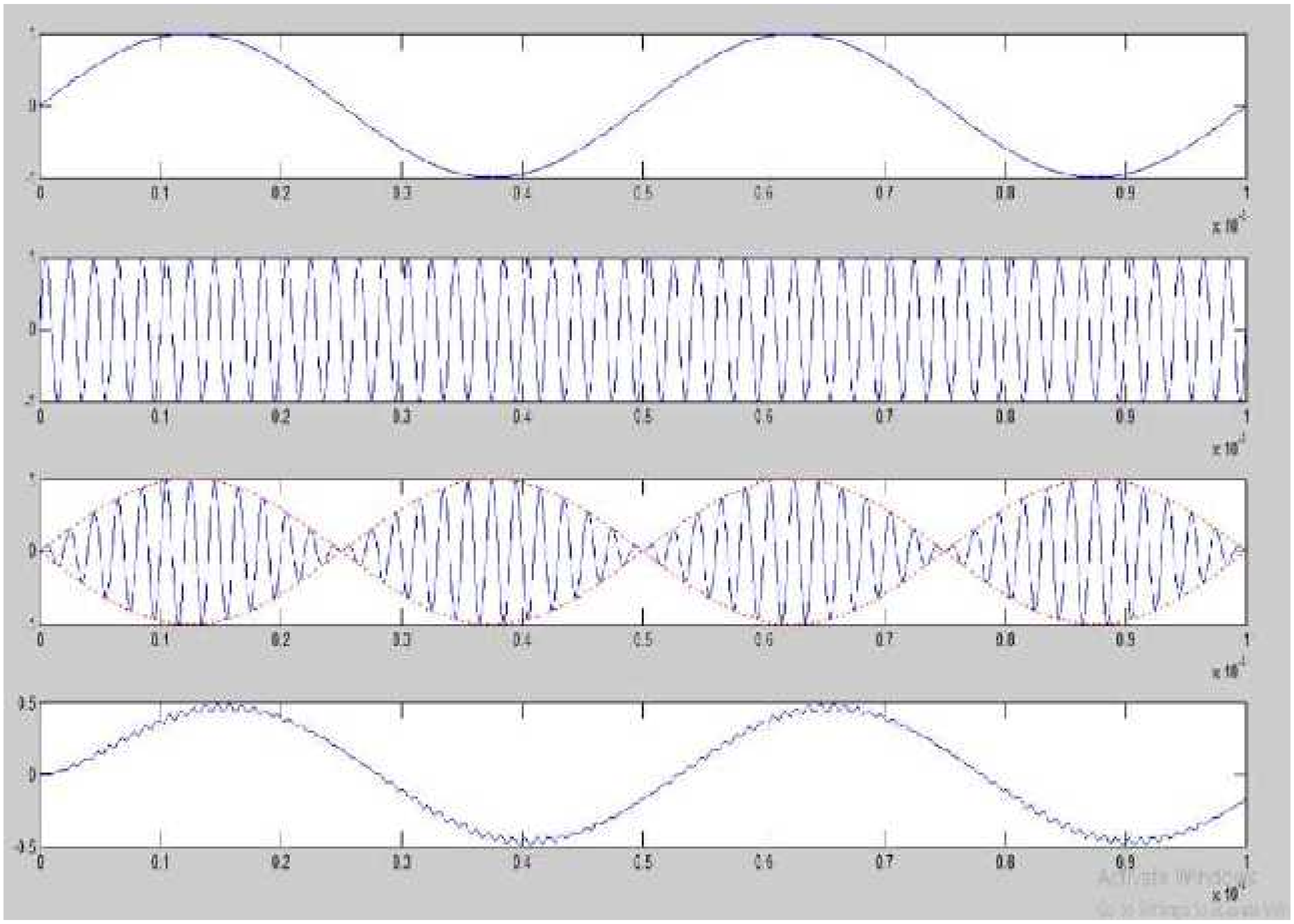
DSB-SC modulated wave is observed and demodulation is performed by using synchronousdetector

CONCLUSION:

Thus the balanced modulator and demodulator, modulation index is (μ>1),Efficiency is 100% was studied and observed.

MATLAB CODE:

```matlab
function amplitude = dsbsc

fm = input('Enter the value of message signal frequency: ');
fc = input('Enter the value of carrier signal frequency: ');
Am = input('Enter the value of message signal amplitude: ');
Ac = input('Enter the value of carrier signal amplitude: ');

Tm = 1/fm;
Tc = 1/fc;

t1 = 0:Tm/999:6*Tm;

message_signal = Am*sin(2*pi*fm*t1);
subplot(3,1,1)
plot(t1, message_signal, 'r');
grid();
title('Message signal');

carrier_signal = Ac*sin(2*pi*fc*t1);
subplot(3,1,2)
plot(t1, carrier_signal, 'b');
grid();
title('Carrier Signal');

amplitude = message_signal.*carrier_signal;
subplot(3,1,3)
plot(t1,amplitude, 'g');
grid();
title('DSBSC');

end
```



POST LAB QUESTIONS

1. What are the two ways of generating DSB-SC.
2. What are the applications of balanced modulator?
3. What are the advantages of suppressing the carrier?
4. What are the advantages of balanced modulator?
5. What are the advantages of Ring modulator?
6. Write the expression for the output voltage of a balanced modulator?
7. Give any two methods to avoid errors in synchronous demodulator?
8. What is quadrature null effect in synchronous demodulator?
9. What is beats in synchronous detector?
10. Give the block diagram of synchronous detector?
11. Give the working principle of coastas receiver.

EVALUATION SHEET

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SSB-SC MODULATOR & DEMODULATOR (PHASE SHIFT METHOD)

AIM:

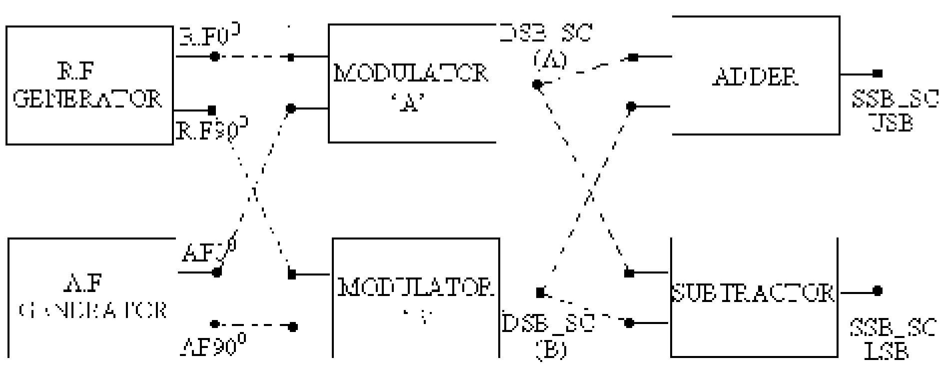
To generate SSB using phase Shift method and demodulation of SSB signal using Synchronous detector.

APPARATUS:

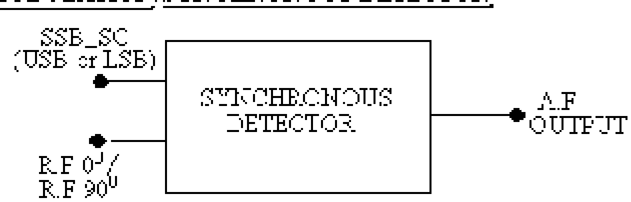
1. Amplitude Modulation & De modulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting cords & probes.
5. PC with windows(95/98/XP/NT/2000)
6. SSBSC Modulation & De modulation trainer kit.

BLOCK DIAGRAM:

SSB MODULATION



SSB DEMODULATION/SYNCHRONOUS DETECTOR



PROCEDURE:

SSB Modulation

1. Connect the circuit as per the given circuit diagram.

2 .Switch on the kit and measure the output of regulated power supplies positive and negativevoltages.

3.Observe the outputs of RF generators using CRO .Where one output is 0^0 phase the is 90^0phase shifted (or) is a sine wave and shifted w.r.t other (or) is a cosine wave.

4.Adjust the RF output frequency as 100 KHz and amplitude as ≥ 0.2 Vp-p (Potentiometers are provided to vary the output amplitude & frequency).

5. Observe the two outputs of AF generator using CRO.

6. Select the required frequency (2 kHz, 4 kHz and 6 kHz) form the switch positions for A.F.

7. Adjust the gain of the oscillator by varying the AGC potentiometer and keep the amplitudeof 10Vp-p.

8. Measure and record the above seen signals & their frequencies on CRO.

9. Set the amplitude of the R.F signal to 0. 2Vp-p and A.F signal amplitude to 8Vp-p and connect AF-0^0 and RF-90^0 to inputs of balanced modulator A and observe DSB-SC (A) output on CRO. Connect AF-90^0 and RF-0^0 to inputs of balanced modulator B and observe the DSB-SC (B) out put on CRO and plot the same on graph.

10. To get SSB lower side band signal connect balanced modulator outputs(DSB-SC) tosubtractor and observe the output wave form on CRO and plot the same on graph.

11. To get SSB upper side band signal, connect the output of balanced modulator outputs to summer circuit and observe the output waveform on CRO and plot the same on graph.

12. Calculate theoretical frequency of SSB (LSB & USB) and compare it with practical value.

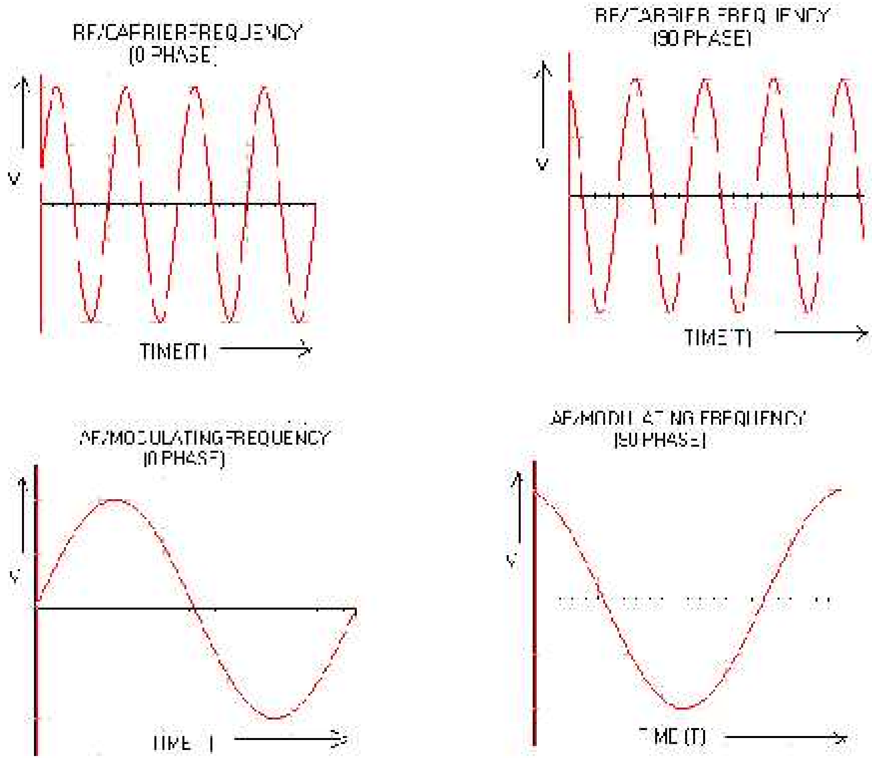
SSB Demodulation

1. Connect SSB signal from the summer or sub-tractor to the SSB signal input of Synchronousdetector and RF signal (0^o) to the RF input of the synchronous detector.
2. Observe the detector output on CRO and compare it with the modulating signal.

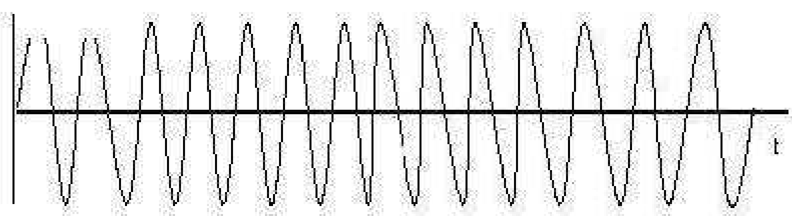
OBSERVATIONS:

Carrier signal		Modulatin gsignal		Balanced modulator -A		Balanced Modulator -B		Adder/Subtract orOutput		Synchron us detector	
F_c	V_c	F_m	V_m	V_{max}	V_{min}	V_{max}	V_{min}	V_{max}	V_{min}	F_d	V_d

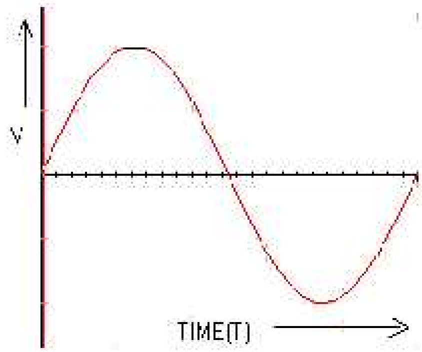
EXPECTED WAVEFORM:



SSB OUTPUT



SSB DEMODULATED OUTPUT



RESULT:

Thus the SSB modulation and demodulation is observed.

MATLAB CODE:

```
fs=8000;
fm=20;
fc=50;
Am=1;
Ac=1;
t=[0:0.1*fs]/fs;
subplot(4,1,1);
m1=Am*cos(2*pi*fm*t);
plot(t,m1);
title('Message Signa m1');
m2=Am*sin(2*pi*fm*t);
subplot(4,1,2);
plot(t,m2);
title('Message Signa m2');
c1=Ac*cos(2*pi*fc*t);
subplot(4,1,3);
plot(t,c1);
title('Carrier Signal c1');
c2=Ac*sin(2*pi*fc*t);
subplot(4,1,4);
plot(t,c2);
title('Carrier Signal c2');
Susb=0.5*m1.*c1-0.5*m2.*c2;
plot(t,Susb);
title('SSB-SC Signal with USB');
subplot(4,1,5);
plot(t,Susb);
Slsb=0.5*m1.*c1+0.5*m2.*c2;
subplot(4,1,6);
plot(t,Slsb);
title('SSB-SC Signal with LSB');
r = Susb.*c1;
[b a] = butter(1,0.001);
mr= filter(b,a,r);
subplot(4,2,5);
plot(t,mr);
```

POST LAB QUESTIONS:

1. What are the two ways of generating SSB?
2. What are the advantages of suppressing the sideband?
3. What are the advantages of phase discrimination method?
4. Write the expression for the output voltage of a SSB modulator?
5. What is the bandwidth required for SSB?
6. What is the power required for SSB?

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FREQUENCY MODULATION & DEMODULATION

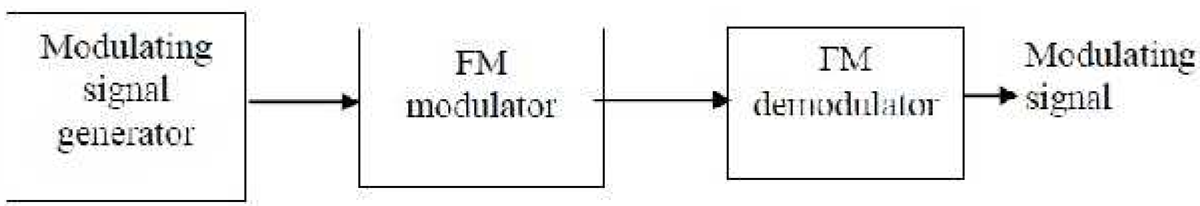
AIM:

To study the functioning of frequency modulation & demodulation and to calculate the modulation index.

APPARATUS:

1. Frequency modulation & demodulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting chords & probes.

BLOCK DIAGRAM: Frequency modulator & De modulator



THEORY:

FM is a system in which the amplitude of the modulated carrier is kept constant, while its frequency and rate of change are varied by the modulating signal. By the definition of FM, the amount by which the carrier frequency is varied from its unmodulated value, called the deviation, is made proportional to the instantaneous amplitude of the modulating voltage. The rate at which this frequency variation changes or takes place is equal to the modulating frequency. FM is that form of angle modulation in which the instantaneous frequency $f_i(t)$ is varied linearly with the message signal m(t), as

$$\mathbf{f_i(t) = f_C + k_f\, m(t)}$$

The term f_c represents the frequency of the unmodulated carrier, and the constant K_f represents the frequency sensitivity of the modulator expressed in Hertz per volt. Unlike AM, the spectrum of an FM signal is not related in a simple manner to that of modulating signal, rather its analysis is much more difficult than that of an AM signal.

PROCEDURE:

1. Switch on the experimental board.
2. Observe the FM modulator output without any modulator input which is the carrier signal and note down its frequency and amplitude.
3. Connect modulating signal to FM modulator input and observe modulating signal and FM output on two channels of the CRO simultaneously.
4. Adjust the amplitude of the modulating signal until we get less distorted FM output.
5. Apply the FM output to FM demodulator and adjust the potentiometer in demodulation until We get demodulated output

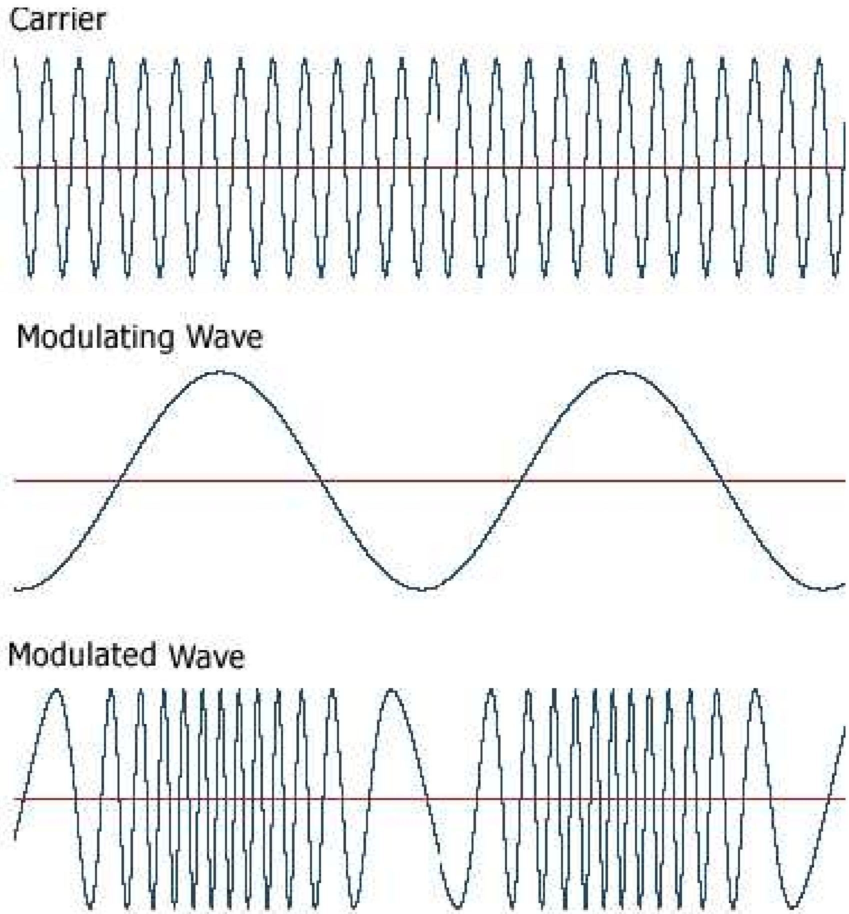
OBSERVATIONS: **MODULATION**

V_m	F_1	F_2	Frequency deviation $\Delta f = F_1 - F_2$	Modulating index $\beta = \frac{\Delta f}{F_m}$	Bandwidth $2(\beta+1)f_m$

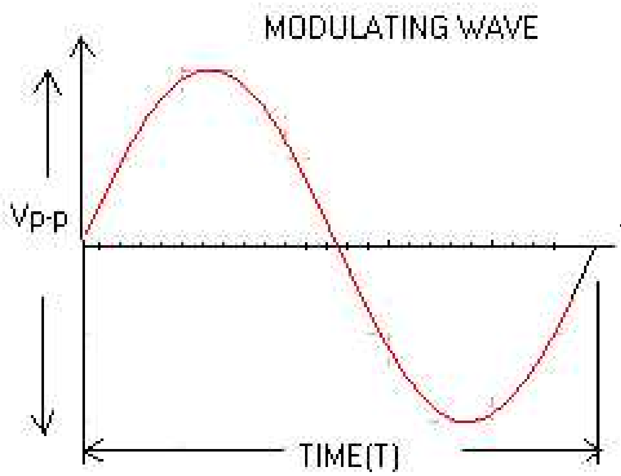
DEMODULATION

Modulating Signal frequency	Demodulating signal frequency

EXPECTED WAVE FORM:



Demodulated signal

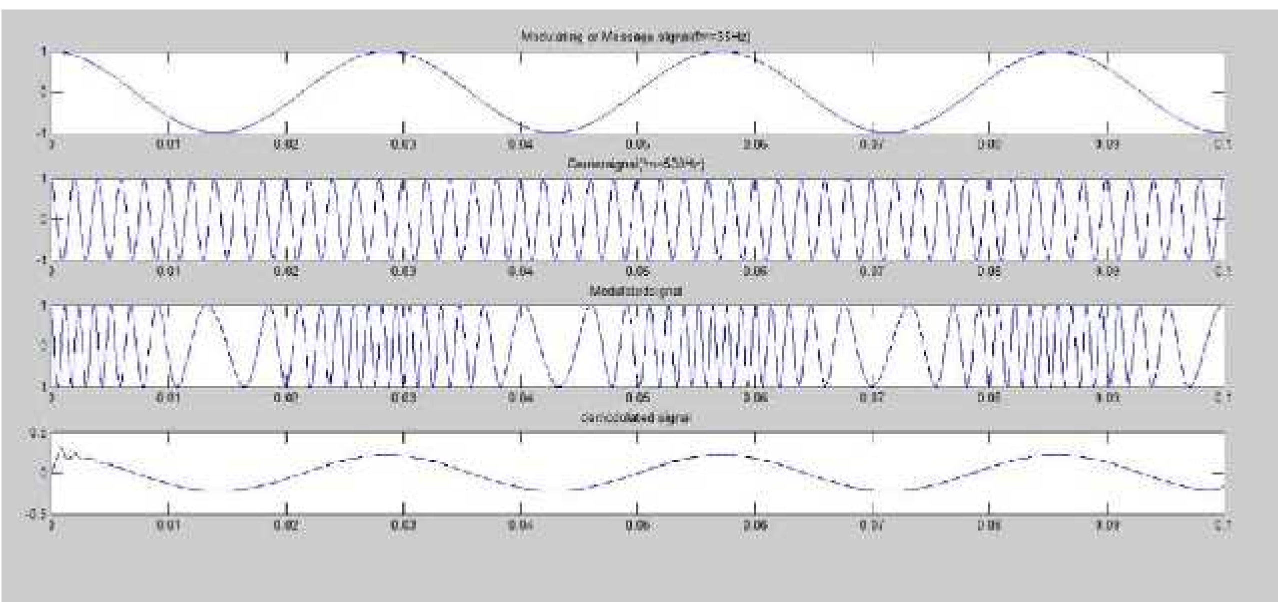


RESULT:

Thus the Frequency modulation and demodulation is observed.

MATLAB CODE:

```
%The frequency modulation (FM)waveform in time and frequency domain.
%fm=35HZ,fc=500HZ,Am=1V,Ac=1V,B=10
fs=10000;
Ac=1;
Am=1;
fm=35;
fc=500;
B=10;
t=(0:0.1*fs)/fs;
wc=2*pi*fc;
wm=2*pi*fm;
m_t=Am*cos(wm*t);
subplot(5,1,1);
plot(t,m_t);
title('Modulating or Message signal(fm=35Hz)');
c_t=Ac*cos(wc*t);
subplot(5,1,2);
plot(t,c_t);
title('Carriersignal(fm=500Hz)');
s_t=Ac*cos((wc*t)+B*sin(wm*t));
subplot(5,1,3);
plot(t,s_t);
title('Modulatedsignal');
d=demod(s_t,fc,fs,'fm');
subplot(5,1,4);
plot(t,d);
title('demodulated signal');
```



POST LAB QUESTIONS:

1. Define FM & PM.
2. What are the advantages of Angle modulation over amplitude modulation?
3. What is the relationship between PM and FM?
4. With a neat block diagram explain how PM is generated using FM.

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PAM modulator and Demodulator

AIM:

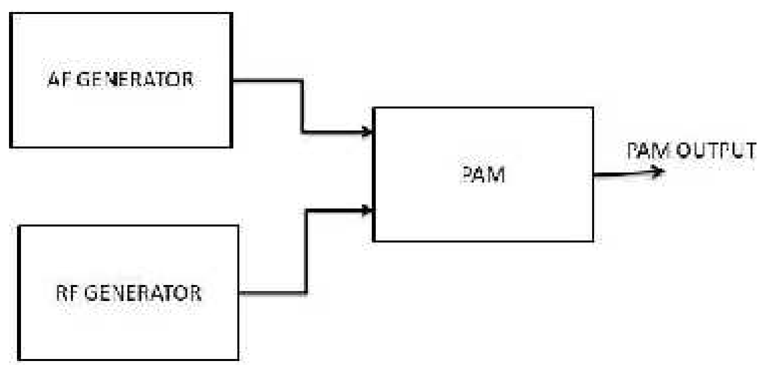
To study Pulse Amplitude modulation and demodulation process with relevant waveforms**.**

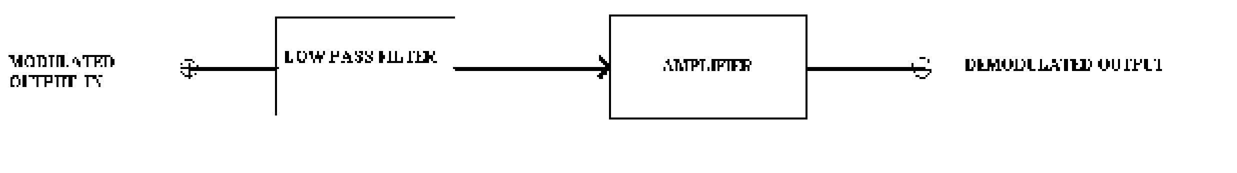
APPARATUS:

1. Pulse amplitude modulation & demodulation Trainer Kit.
2. Dual trace CRO.
3. Patch chords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

BLOCK DIAGRAM:

PULSE AMPLITUDE MODULATION AND DEMODULATION





THEORY:

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with syncing signals. At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse amplitude modulation is the simplest form of the pulse modulation. PAM is a pulse modulation system is which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cables are used to modulated carrier.

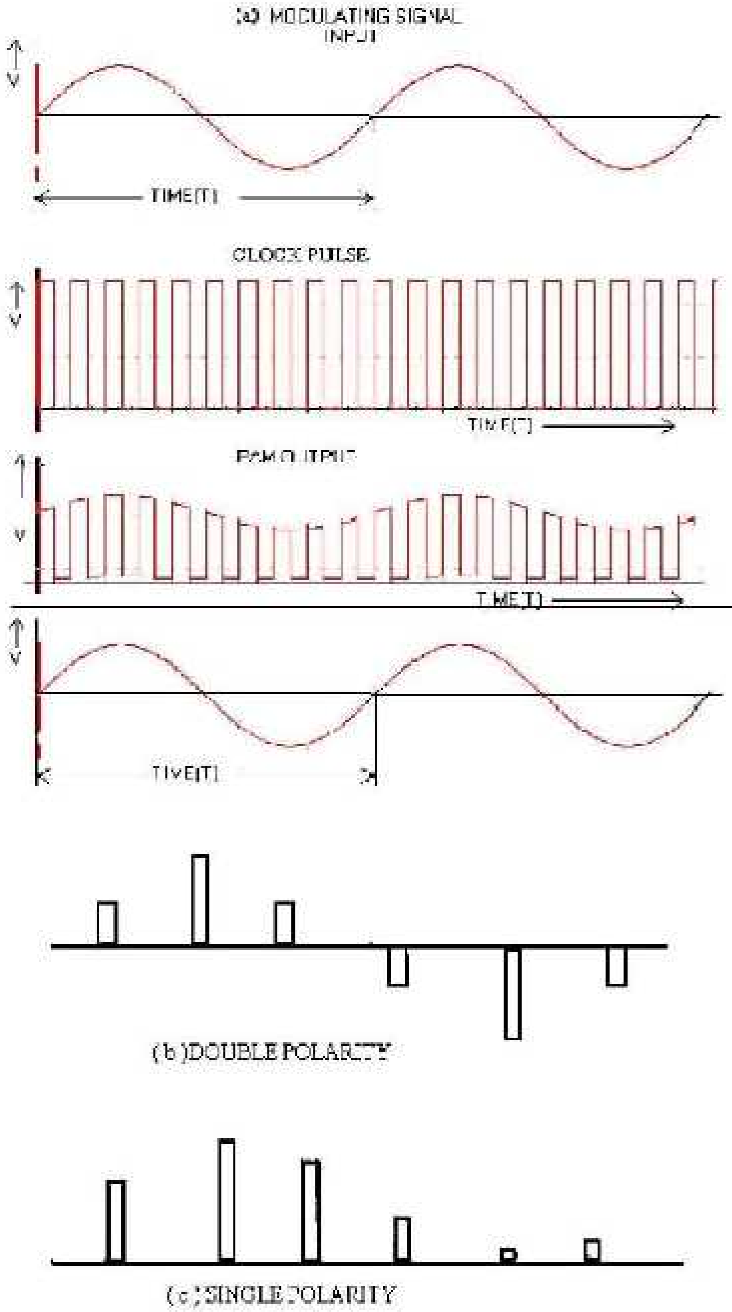
The two types of PAM are i) Double polarity PAM, and ii) the single polarity PAM, in which a fixed dc level is added to the signal to ensure that the pulses are always positive. Instantaneous PAM sampling occurs if the pulses used in the modulator are infinitely short. Natural PAM sampling occurs when finite-width pulses are used in the modulator, but the tops of the pulses are forced to follow the modulating waveform.

Flat-topped sampling is a system quite often used because of the ease of generating the modulated wave. PAM signals are very rarely used for transmission purposes directly. The reason for this lies in the fact that the modulating information is contained in the amplitude factor of the pulses, which can be easily distorted during transmission by noise, crosstalk, other forms of distortion. They are used frequently as an intermediate step in other pulse modulating methods, especially where time-division multiplexing is used.

PROCEDURE:

1. The 4016 integrated circuit is a CMOS bilateral switch which is used as a sampling switch. A positive voltage on pin 13 closes the CMOS transistor switch between pins 1&2. When pin13 is as zero volts, the switch is open.
2. Switch ON the trainer kit.
3. Connect a 10 KHz sine wave of 5V p-p from an audio generator at the point marked AF i/p.
4. Connect the oscilloscope to pin 2 of 4016 IC, adjust the 1K potentiometer (R1) to vary the amplitude of the modulating signal. Also adjust the frequency of the modulating signal to obtainstable display on the oscilloscope. The waveform obtained is a dual polarity PAM.
5. Vary the amplitude and frequency of the sine wave signal and observe the change in the output waveform.
6. Connect the modulated output to the input of the demodulator.
7. Connect channel 1 of the dual trace oscilloscope to the demodulator output and channel 2 to the input sine wave. Compare the two waveforms you will find that they are 180 out of phase.

EXPECTED WAVE FORMS:



RESULT:

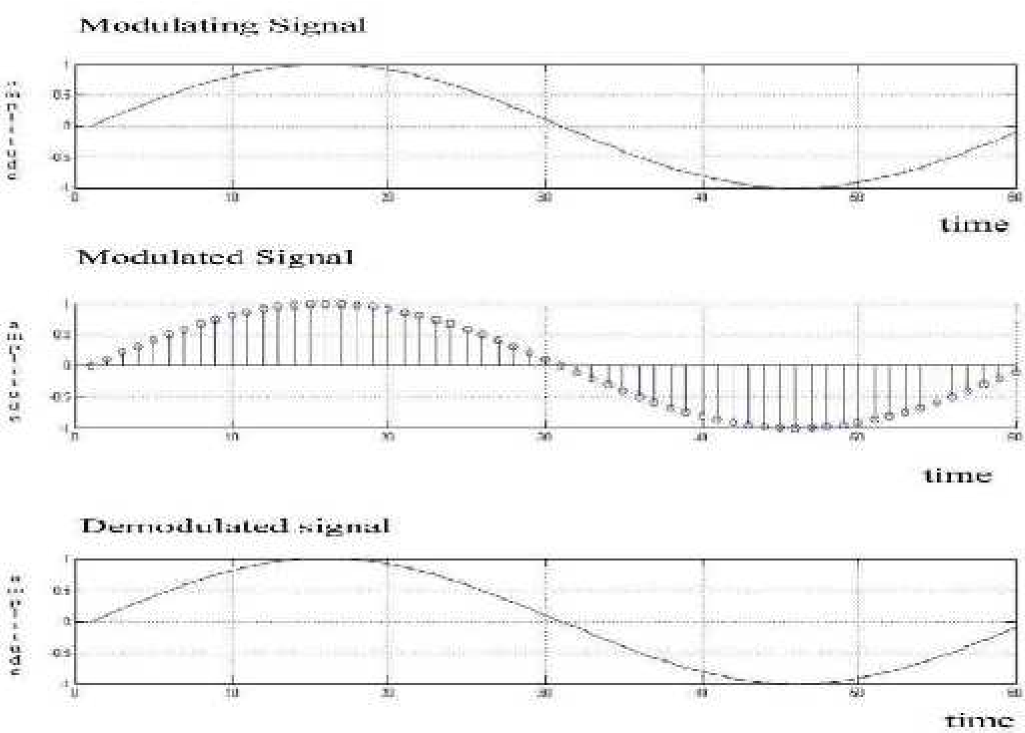
Thus the Pulse amplitude modulation and demodulation is observed and their respective wave formsare plotted.

CONCLUSION:

In Pulse Amplitude Modulation we observed that the characteristic parameter (amplitude) of carrier is varied according to the amplitude of the message signal, the samples are taken at regular intervals of time. Each sample is a pulse whose amplitude is determined by the amplitude of the variable at the instant of time at which the samples are taken, a reasonable approximating of the signal being sampled can be constructed at the receiving end

MATLAB CODE:

```
clc;
fc = 20;
fm =2;
fs = 1000;
t=1;
n = [0:1/fs:t];
n = n(1:end-1);
dutycycle = 50;
s = square(2*pi*fc*n,dutycycle);
s(find(s<0))=0;   %to make it unipolar
%plot(s);
m = sin(2*pi*fm*n);
period_sam = length(n)/fc;     %to find the number of samples in one period
ind = 1:period_sam:length(n);   %to find the starting sample index
on_samp = ceil(period_sam * dutycycle/100);   %no. of samples in on period of time
pam = zeros(1,length(n));
for i =1:length(ind)
    pam(ind(i):ind(i)+on_samp) = m(ind(i));
end
subplot(3,1,1);
plot(n,s);
ylim([-0.2 1.2]);
subplot(3,1,2);
plot(n,m);
ylim([-1.2 1.2]);
subplot(3,1,3);
plot(n,pam);
ylim([-1.2 1.2]);
```



POST LAB QUESTIONS:

1. TDM is possible for sampled signals. What kind of multiplexing can be used in continuous modulation systems?
2. What is the minimum rate at which a speech signal can be sampled for the purpose of PAM?
3. What is cross talk in the context of time division multiplexing?
4. Which is better, natural sampling or flat topped sampling and why?
5. Why a dc offset has been added to the modulating signal in this board? Was it essential for the working of the modulator? Explain.
7. Study about the frequency spectrum of PAM signal and derive mathematical expression for it?
8. Explain the modulation circuit operation?
9. Explain the demodulation circuit operation?

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PWM MODULATOR & DEMODULATOR

AIM:

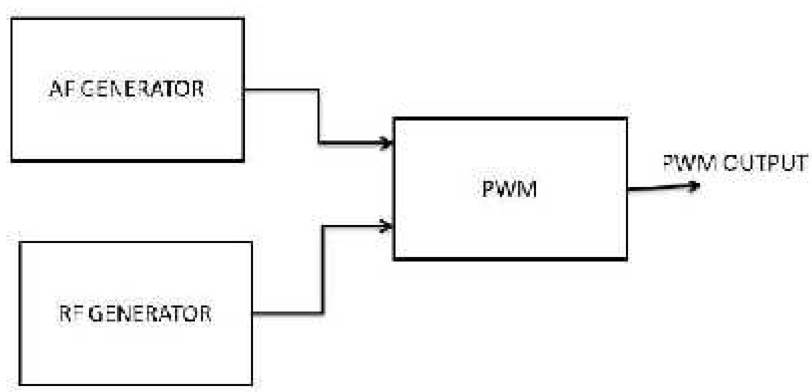
To generate the pulse width modulated and demodulated waves.

APPARATUS:

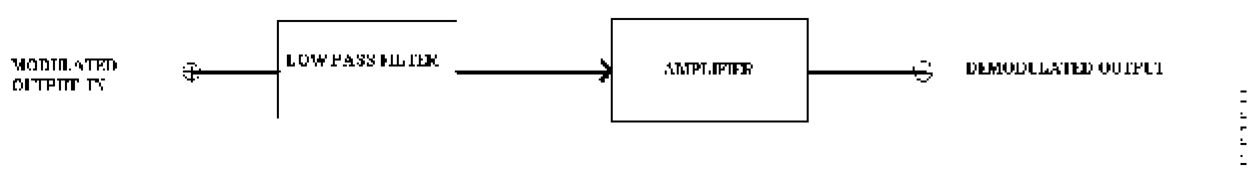
1. PWM trainer kit
2. C.R.O(30MHz)
3. Patch Chords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

BLOCK DIAGRAM:

PWM MODULATOR & DEMODULATOR



DEMODULATOR



PROCEDURE:

1. Switch “ON” the experimental kit.
2. Observe the clock generator output & modulation signal outputs.
3. Connect clock generator output to the clock input point of PWM modulator and observe the same clock on channel of a dual trace CRO.
4. Trigger the CRO with respect to CH 1.
5. Apply a variable DC voltage of 8 to 12 volts from any external regulated Power supply.
6. Observe the PWM output on CH 2.
7. If we observe the PWM output, it’s width varies according to the Modulating voltage.
8. A variable amplitude modulating signal is given to observe how the PWM are varying for AC modulating voltages.
9. In this case we have to trigger the CRO with respect to modulating voltage.

EXPECTED WAVE FORMS:

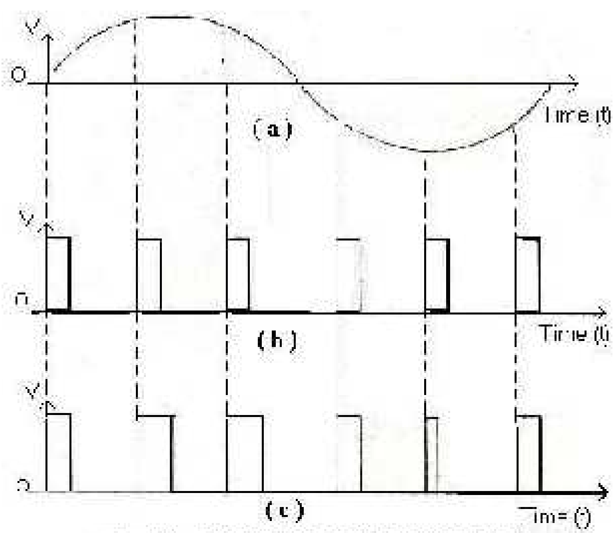


Fig (2) PULSE WIDTH MODULATION
(a) Signal
(b) Unmodulated pulses
(c) PWM

RESULT:

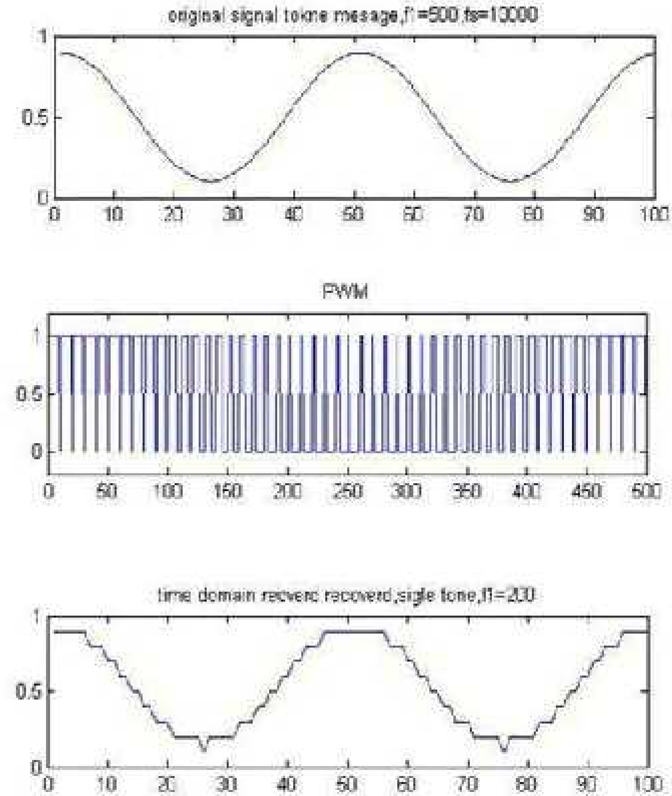
Thus the Pulse width modulation and demodulation is observed and their respective wave forms areplotted.

CONCLUSION:

In Pulse Width Modulation we observed that carrier signal width is varied according to the amplitude of the message signal, amplitude of the carrier signal is fixed.

MATLAB CODE:

```
fs=input('Comparator Sawtooth frequency:');
fm=input('Message frequency(Assuming it to be a sine wave):');
a=input('Enter Amplitude of Message:');

t=0:0.0001:1; %sampling rate of 10kHz

stooth=1.01*a.*sawtooth(2*pi*fs*t); %generating a sawtooth wave
%to make the two non zero lobes of pwm not to overlap the amplitude of
%sawtooth wave must be atleast more than a bit to the message amplitude

subplot(3,1,1);
plot(t,stooth); % plotting the sawtooth wave
title('Comparator Wave');

msg=a.*sin(2*pi*fm*t); %generating message wave

subplot(3,1,2);
plot(t,msg); %plotting the sine message wave
title('Message Signal');


for i=1:length(stooth)
if (msg(i)>=stooth(i))
    pwm(i)=1; %is message signal amplitude at i th sample is greater than
    %sawtooth wave amplitude at i th sample
else
    pwm(i)=0;
end
end

subplot(3,1,3);
plot(t,pwm,'r');
title('PWM');
axis([0 1 0 1.1]); %to keep the pwm visible during plotting.
```



original signal tokne mesage,f1=500,fs=10000
PWM
time domain reovere recoverd,sigle tone,f1=200

POST LAB QUESTIONS:

1. An audio signal consists of frequencies in the range of 100Hz to 5.5KHz.What is the minimum frequency at which it should be sampled in order to transmit it through pulse modulation?
2. Draw a TDM signal which is handling three different signals using PWM?
3. What do you infer from the frequency spectrum of a PWM signal?
4. Clock frequency in a PWM system is 2.5 kHz and modulating signal frequency is 500Hzhowmany pulses per cycle of signal occur in PWM output? Draw the PWM signal?
5. Why should the curve for pulse width Vs modulating voltage be linear?
6. What is the other name for PWM?
7. What is the disadvantage of PWM?
8. Will PWM work if the synchronization between Tx and Rx fails?
9. Why integrator is required in demodulation of PWM?
10. What kind of conversion is done in PWM generation?

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1	Attendance	5	
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4	Calculation/Evaluation of Result	10	
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6	Grand Total	50	

Signature of the Faculty Handling the Lab

PULSE POSITION MODULATION AND DEMODULATION

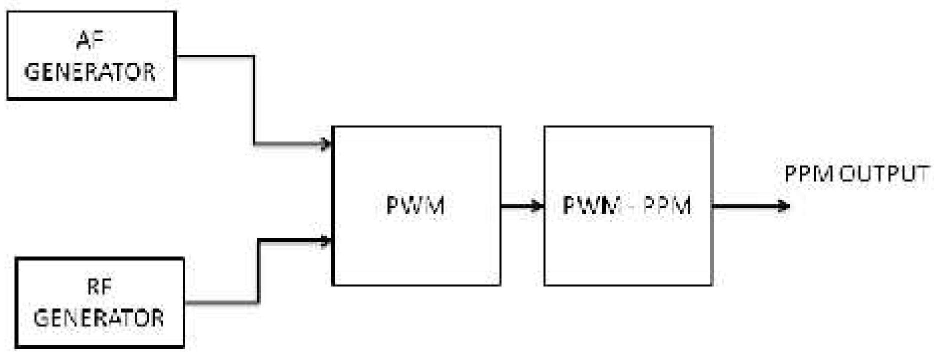
AIM:

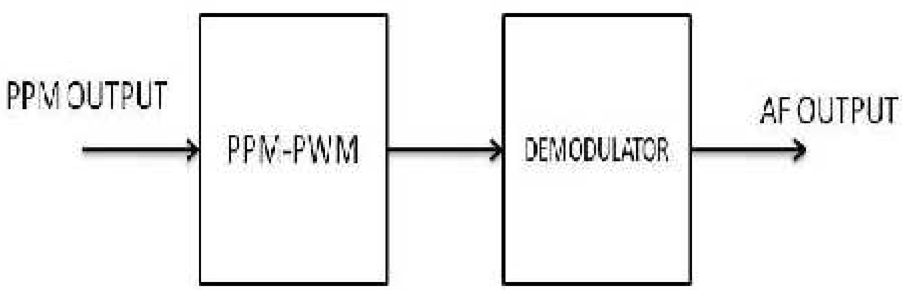
To study the generation Pulse Position Modulation (PPM) and Demodulation.

APPARATUS:

1. Pulse Position Modulation (PPM) and demodulation Trainer Kit.
2. C.R.O(30MHz)
3. Patch chords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

BLOCK DIAGRAM: PPM MODULATION AND DEMODULATION





THEORY:

Pulse modulation is used to transmit analog information, such as continuous speech or data. The data is sent at sampling times, with synchronizing pulses. The pulse position modulation is an analog modulation method, where in we have fixed amplitude of each pulse, but the position of each pulse is made proportional to the amplitude of the modulating signal at that instant.

PPM is derived from the pulse width modulated signal. To demodulate the PPM signal, it is fed to an integrating RC circuit (LPF) to obtain the modulating signal.

PROCEDURE:

MODULATOR:

1. Switch On the experimental kit.
2. Observe the clock generator output and modulating signal outputs.
3. Connect the clock generator output to the clock input point of PPM modulator and observe the same clock on CH1 of dual trace CRO.
4. Trigger the CRO w.r.t CH1.
5. Apply a variable D.C voltage of 8-12V from any external regulated power supply.
6. Observe the PPM output on CH2.
7. By varying the modulating voltage, PPM output changes position, but the width is maintained constant.

DEMODULATOR:

1. Apply PPM signal to the PPM demodulator and observe the output.
2. The output almost coincides with modulating signal.

RESULT:

Pulse position modulation and demodulation is observed and their respective waveforms are plotted.

MATLAB CODE:

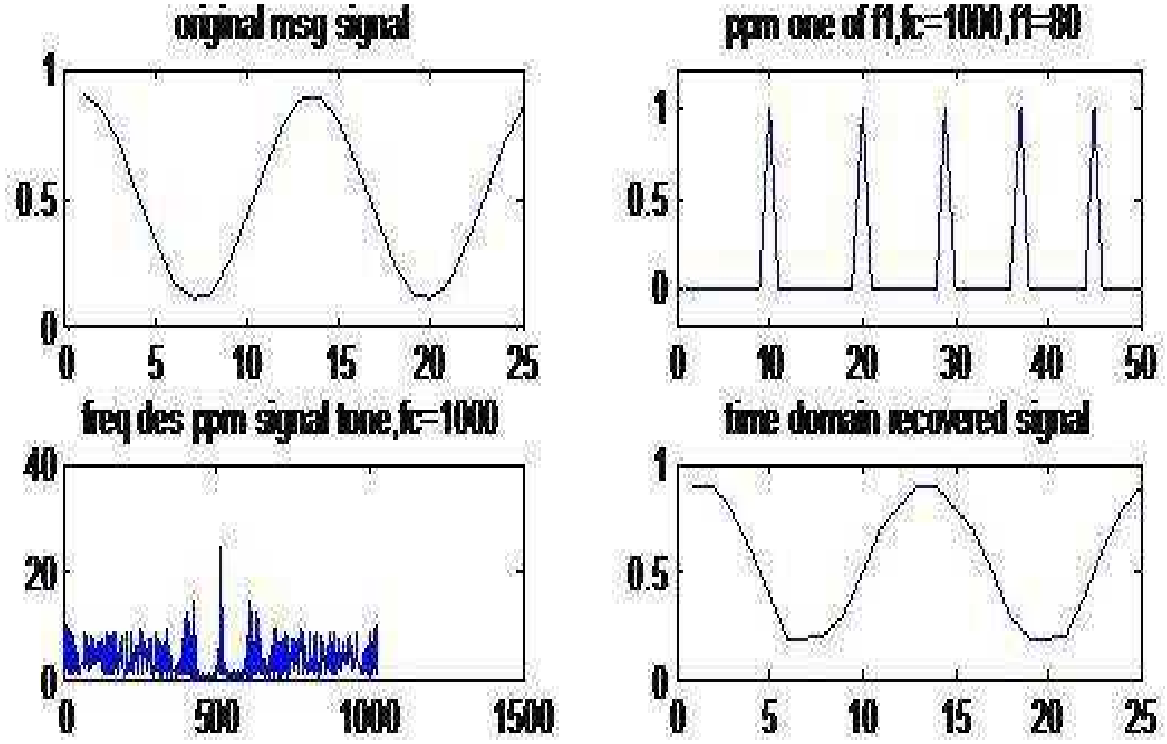
ALGORITHM

- Choose the sampling frequency f_s and modulating frequency f_1 such that Nyquist criteria are satisfied.
- Generate the message signal using f_1 and f_s .
- Modulate the message signal using the carrier frequency.
- FFT is applied to the modulated signal to get frequency spectrum.
- Demodulate the modulated signal using the same carrier frequency.
- Plot the graphs for the original message signal, modulated, frequency spectrum and demodulated signal.

PROGRAM

```
fc=100;
fs=1000;
f1=80;
f2=300 ;
t=0:1/fs:((2/f1)-(1/fs));
x1=0.4*cos(2*pi*f1*t)+0.5;
x2=0.2*(cos(2*pi*f1*t)+cos(2*pi*f2*t))+0.5 ;
subplot(4,2,1);
plot(x1);
title('original msg signal'); y1=modulate(x1,fc,fs,'ppm');
subplot(4,2,2);
plot(y1)
axis([0 50 -0.2 1.2]);
title('ppm one of f1,fc=1000,f1=80 ');
fx1=abs(fft(y1,1024));
fx1=[fx1(512:1024) fx1(1:513)];
f=[(511*fs/1024):(fs/1024):(512*fs/1024)];
subplot(4,2,3); plot(fx1);
title('freq des ppm signal tone,fc=1000');
x1_recov = demod(y1,fc,fs,'ppm');
subplot(4,2,4);
plot(x1_recov);
title('time domain recovered signal');
```

WAVE FORMS:


original msg signal
ppm one of f1,fc=1000,f1=80
freq des ppm signal tone,fc=1000
time domain recovered signal
1
0.5
0
5
10
15
20
25
30
40
50
500
1000
1500

POST LAB QUESTIONS:

1. Define and describe PPM?
2. Explain with waveforms how PPM is derived from PWM.
3. What is the fundamental difference between pulse modulation, on the one hand, andfrequency and amplitude modulation on the other?

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PRE-EMPHASIS & DE-EMPHASIS

AIM:

To study the frequency response of Pre-Emphasis and De-Emphasis circuits.

APPARATUS:

1. Pre-emphasis & De-emphasis trainer kits.
2. C.R.O (20 MHz)
3. Function generator (1MHz).
4. Patch chords and Probes.
5. PC with windows (95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

THEORY:

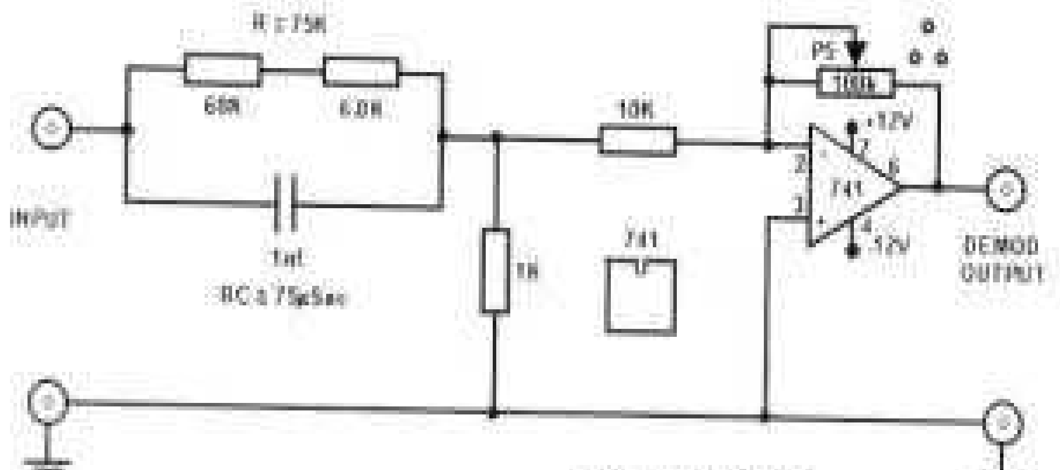
Frequency modulation is much immune to noise than amplitude modulation and significantly more immune than phase modulation. A single noise frequency will affect the output of the receiver only if it falls within its pass band.

The noise has a greater effect on the higher modulating frequencies than on lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, improvement in noise immunity could be expected. This booting of the higher frequencies, in accordance with a pre-arranged curve, is termed pre-emphasis, and the compensation at the receiver is called de-emphasis.

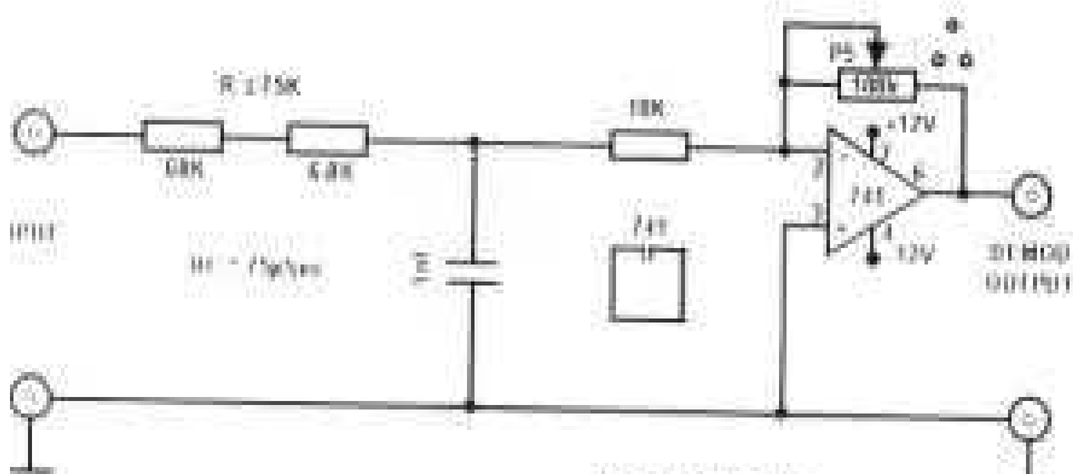
If the two modulating signals have the same initial amplitude, and one of them is pre-emphasized to (say) twice this amplitude, whereas the other is unaffected (being at a much lower frequency) then the receiver will naturally have to de-emphasize the first signal by a factor of 2, to ensure that both signals have the same amplitude in the output of the receiver. Before demodulation, I.e. while susceptible to noise interference the emphasized signal had twice the deviation it would have had without pre-emphasis, and was thus more immune to noise. Alternatively, it is seen that when this signal is de- emphasized any noise sideband voltages are de-emphasized with it, and therefore have a correspondingly lower amplitude than they would have had without emphasis again their effect on the output is reduced.

Apart from that, it would be difficult to introduce pre-emphasis and de-emphasis in existing AM services since extensive modifications would be needed, particularly in view of the huge numbers is receivers in use.

CIRCUIT DIAGRAM:



PRE EMPHASIS



DE EMPHASIS

PROCEDURE:

PRE-EMPHASIS

1. Connect the circuit as per the circuit diagram
2. Apply a sine wave to the input terminals of 2 $V_{P-P\ (Vi)}$
3. By varying the input frequency with fixed amplitude, note down the output amplitude (V_o) with respect to the input frequency.
4. Calculate the gain using the formula Gain = 20 log (V_O/ V_I) db

   Where V_O = output voltage in volts.

   V_I = Input voltage in volts. And plot the frequency response.

DE-EMPHASIS

5. Connect the circuit as per circuit diagram.
6. Repeat steps 2, 3 & 4 of Pre-Emphasis to de-emphasis also.

EXPECTED WAVEFORMS

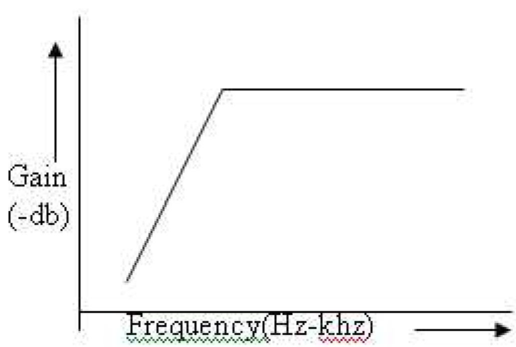


Fig: Pre-emphasis

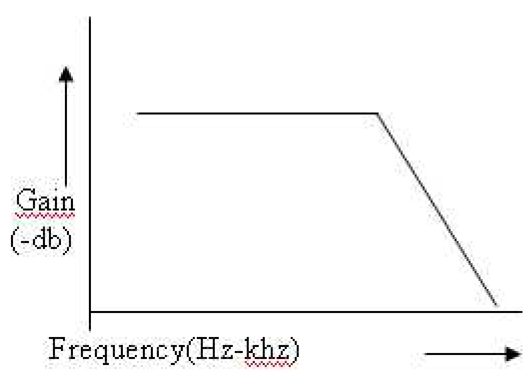


Fig: De-emphasis

TABLE:

Pre-emphasis

Frequency(f)	Vin	Vo	Vo/Vin	Gain in db (20logVo/Vin)

De-emphasis

Frequency (f)	Vin	Vo	Vo/Vin	Gain in db (20logVo/Vin)

RESULT:

Thus the frequency response of Pre-Emphasis and De-Emphasis circuits was done.

MATLABCODE:

```
% program for Pre-Emphasis and De-Emphasis
num_samples = 2^13;
fs=5000;
Ts=1/fs;
fm1=20;
fm2=30;
fc=200;
t=(0:num_samples-1)*Ts;
f=(-num_samples/2:num_samples/2-1)*fs/num_samples;
mt=sin(2*pi*fm1*t);
Mf=fftshift(abs(fft(mt)));
f_cutoff_pe=15;
Wn_pe=f_cutoff_pe/(fs/2);
[b_pe,a_pe]=butter(1,Wn_pe);
[H_pe,W]=freqz(a_pe,b_pe);
a_de=b_pe;
b_de=a_pe;
[H_de,W]=freqz(a_de,b_de);
mt_pe=filter(a_pe,b_pe,mt);
Mf_pe=fftshift(abs(fft(mt_pe)));
figure(1);
subplot(211);
plot(t,mt);
axis([0 .6 min(mt)-1 max(mt)+1]);
grid on
title('Modulating Signal (Time Domain)');
subplot(212);
plot(f,Mf);
grid on;
axis([-50 50 0 max(Mf)+100]);
title('Modulating Signal (Frequency Domain)');
figure(2);
subplot(211)
semilogx(W*pi*(fs/2),abs(H_pe),'m','linewidth',2);
axis([0 fs/2 0 50]);
grid on;
title('Pre-emphasis Filter Magnitude Response');
subplot(212);
semilogx(W*pi*(fs/2),abs(H_de),'m','linewidth',2);
axis([0 fs/2 0 1]);
grid on;
title('De-emphasis Filter Magnitude Response');
figure(3);
subplot(211);
plot(t,mt_pe);
axis([0 .6 min(mt_pe)-1 max(mt_pe)+1]);
title('preemphasised signal time domain');
subplot(212);
plot(f,Mf_pe);
title('pre-emphasised signal frequency domain');
grid on;
axis([-50 50 0 max(Mf_pe)+100]);
```

POST LAB QUESTIONS:

1. What is the need for pre-emphasis?

2. Explain the operation of pre-emphasis circuit?

3. Pre emphasis operation is similar to high pass filter explain how?

4. De emphasis operation is similar to low pass filter justify?

5. What is de-emphasis?

6. Draw the frequency response of a pre-emphasis circuit?

7. Draw the frequency response of a de-emphasis circuit?

8. Give the formula for the cutoff frequency of the pre-emphasis circuit?

9. What is the significance of the 3db down frequency?

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Signal Sampling and Reconstruction (Sampling Theorem)

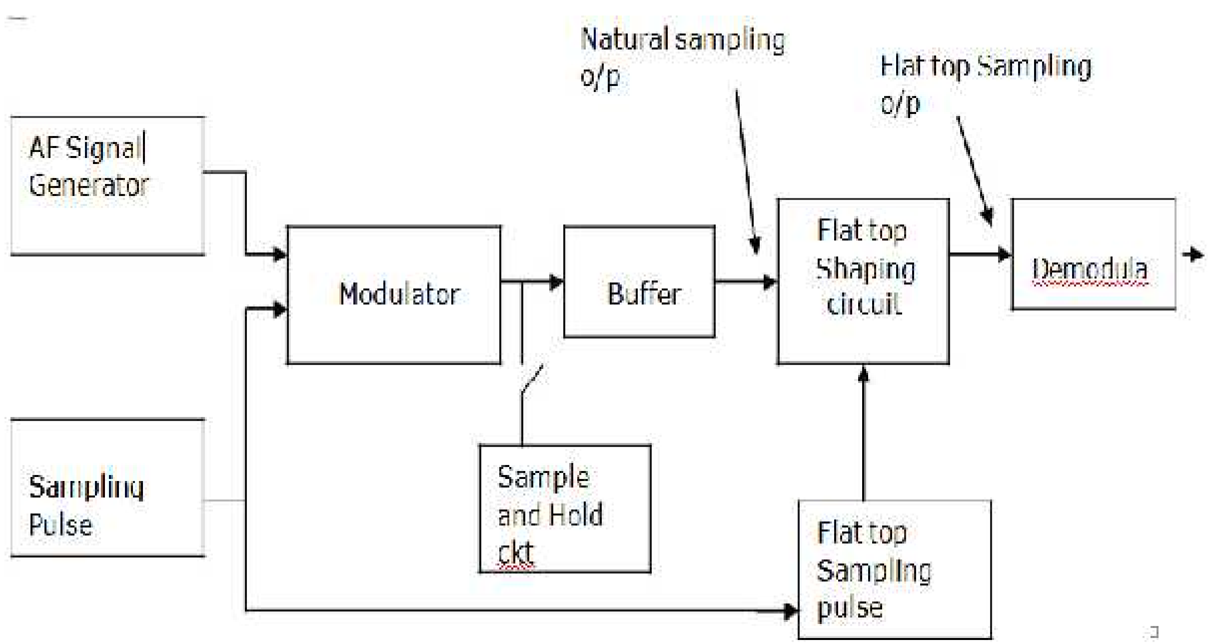
AIM:

1. To study the sampling theorem and its reconstruction.
2. To study the effect of amplitude and frequency variation of modulating signal on the output.
3. To study the effect of variation of sampling frequency on the demodulated output.

APPARATUS:

1. Sampling and reconstruction Trainer Kit.
2. C.R.O(30Mhz)
3. Patch cords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

BLOCK DIAGRAM:



THEORY:

‘ Pulse modulation is used to transmit analog information. In this system continuous waveforms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals. At the receiving end the original waveforms may be reconstituted from the informationregarding the samples.

Sampling theorem statement:
A band limited signal of finite energy which has no frequency components higher than fm Hz iscompletely, described by specifying the values of the signal at instants of time separated by +fm sec.

The sampling theorem states that if the sampling rate in any pulse modulation system exceeds twice the maximum signal frequency, the original signal can be reconstructed in the receiver with minimum distortion.

Fs> 2fm is called Nyquist rate Where Fs sampling frequency, Fm modulating signal frequency.

If we reduce the sampling frequency Fs less than 2fm the side bands and the information signal will overlap and we cannot recover the information signal by low pass filter. This phenomenon is called fold over distortion or aliasing.

There are two methods of sampling

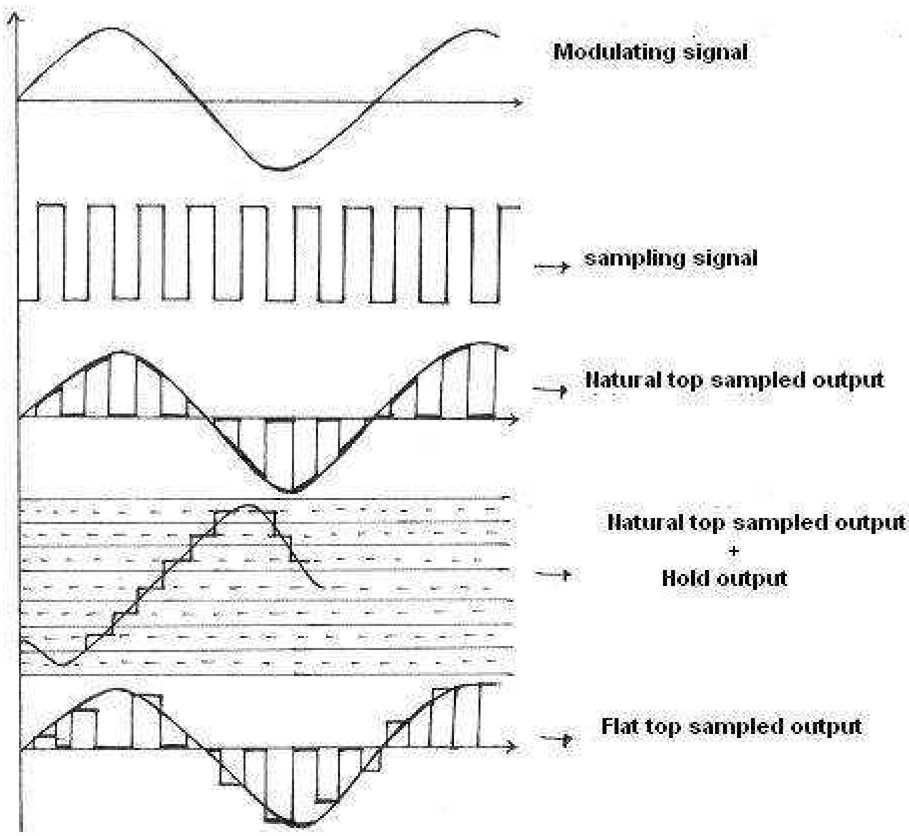
1. Natural sampling.
2. Flat top sampling.

Sample and hold circuit holds the sampled value until the next sample is taken. Sample and hold technique is used to maintain reasonable pulse energy. The duty cycle is defined as the ratio of pulse duration to the pulse repetition period.

PROCEDURE:

1. Switch on the ”power ON” switch.
2. Observe the AF signal generator output, it is a sine wave frequency varying from 2 KHz to 20 KHz and amplitude varying from 0-5 Vp-p.
3. Observe the sampling pulse generator output, it is a square wave of frequency varying from Date: 2 KHz to 32 KHz and pulse also adjustable.
4. Connect the sampling pulse generator output to the sampling pulse output of the modulator. Make sure that the frequency adjust pot is in its extreme clock wise direction.
5. Now adjust the output of the AF signal generator to 1 KHz, 5Vp-p and connect the same signal to analog input terminals of the modulator. If the pulse width is now made very narrow the SH signal will seen as if it is instantaneously sampled and held.
6. If the jumper J12 is removed-natural top sampling output is observed.

EXPECTED GRAPH:



RESULT:

Thus the Analog signal sampling and reconstruction with different sampling techniques are observed.

POST LAB QUESTIONS:

1. What is Nyquist rate?
2. What is aliasing?
3. What type of filter is used to recover the signal?
4. How many sampling techniques are there?
5. Give the differences between analog and discrete signals.
6. Give the difference between flat top sampling and natural sampling.

MATLABCODE:

```
% t=-10:.01:10;
T=4;
fm=1/T;
x=cos(2*pi*fm*t);
subplot(2,2,1);
plot(t,x);
xlabel('time');ylabel('x(t)')
title('continous time signal')
grid;
n1=-4:1:4
fs1=1.6*fm;
fs2=2*fm;
fs3=8*fm;
x1=cos(2*pi*fm/fs1*n1);
subplot(2,2,2);
stem(n1,x1);
xlabel('time');ylabel('x(n)')
title('discrete time signal with fs<2fm')
hold on
subplot(2,2,2);
plot(n1,x1)
grid;
n2=-5:1:5;
x2=cos(2*pi*fm/fs2*n2);
subplot(2,2,3);
stem(n2,x2);
xlabel('time');ylabel('x(n)')
title('discrete time signal with fs=2fm')
hold on
subplot(2,2,3);
plot(n2,x2)
grid;
n3=-20:1:20;
x3=cos(2*pi*fm/fs3*n3);
subplot(2,2,4);
stem(n3,x3);
xlabel('time');ylabel('x(n)')
title('discrete time signal with fs>2fm')
hold on
subplot(2,2,4);
plot(n3,x3)
grid
```

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Pulse Code Modulation and Demodulation

AIM:

To analyze a PCM system and interpret the modulated and demodulated waveforms for a sampling frequency of 4 KHz.

APPARATUS:

1. PCM modulator trainer
2. PCM Demodulator trainer
3. C.R.O(30MHz)
4. Patch chords.
5. PC with windows(95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

INTRODUCTION

In Pulse code modulation (PCM) only certain discrete values are allowed for the modulating signals. The modulating signal is sampled, as in other forms of pulse modulation. But any sample falling within a specified range of values is assigned a discrete value. Each value is assigned a pattern of pulses and the signal transmitted by means of this code. The electronic circuit that produces the coded pulse train from the modulating waveform is termed a coder or encoder. A suitable decoder must be used at the receiver in order to extract the original information from the transmitted pulse train.

This PCM system consists of

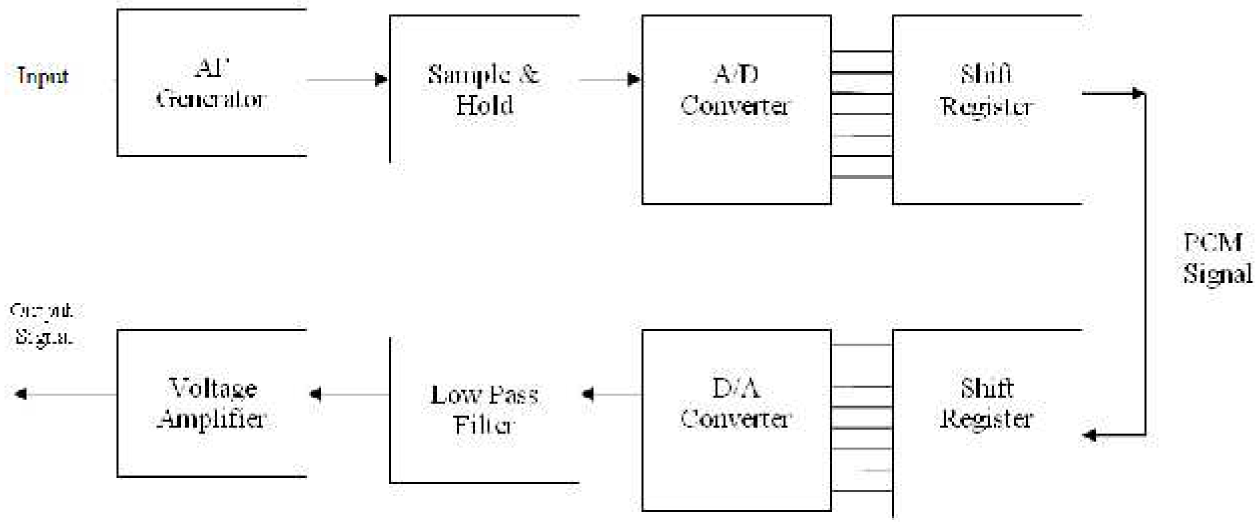
PCM Modulator

1. Regulated power supply
2. Audio Frequency signal generator
3. Sample & Hold circuit
4. 8 Bit A/D Converter
5. 8 Bit Parallel-Serial Shift register
6. Clock generator/Timing circuit
7. DC source

2.3.2. PCM Demodulator

1. Regulated power supply
2. 8 Bit Serial-Parallel to shift register
3. 8 Bit D/A converter
4. Clock generator
5. Timing circuit
6. Passive low pass filter
7. Audio amplifiers

BLOCK DIAGRAM: PCM MODULATOR & DEMODULATOR



Regulated power supply (68M & 68D):

This consists of a bridge rectifier followed by capacitor filters and three terminal regulators 7805 and 7905 to provide regulated DC voltages of +5V and +12V @ 300Ma each to the on board circuits. These supplies have been internally connected to the circuits, so no external connections are required for operation.

Audio Frequency (AF) Signal generator (68M):

Sine wave signal of 200Hz is generated to use as a modulating (message or information) signal to be transmitted. This is an Op-Amp based Wein bridge Oscillators using IC TL084. IC TL084 is a FET input general purpose Operational Amplifier. Amplitude control is provided in the circuit to vary the output amplitude of AF signal.

Clock generator/Timing circuit (68M & 68D):

A TTL compatible clock signal of 64 KHz and 4KHz frequency are provided on board to use as a clock to the various circuits in the system. This circuit is a astable multivibrator using 555 timer followed by a buffer and frequency dividers.

DC source (68M):

A 0 to +5V variable DC voltage is provided on board to use as a modulating signal instead of AF signal. This is useful to study step by operation of PCM modulation and demodulation. This is a simple circuit consisting of potentiometer and fixed power supply.

Low pass filters (68D):

This is a series of simple RC networks provided on board to smoothen the output of the D/A converter output (stair case signal). RC values are chosen such that the cutoff frequency would be at 200Hz

Amplifiers (68D):

This is an Op-amp (IC TL084) based non-inverting variable gain amplifiers provided on board to amplify the recovered message singles i.e. output of the Low pass filter to desired level. Amplitude control is provided in circuit to vary the gain of the amplifier between 0 and 3. AC/DC Switch facilitates to couple the input signal through capacitor or directly to the amplifier input.

Sample & Hold circuit (AET-68M):

This block (circuit) is a combination of buffer, level shifting network and sample & hold network. Op- amp IC TL084 is connected as buffer followed by non-inverting summer circuit. One of the inputs of summer is connected a voltage divider network and other being drawn as input. A dedicated sample& hold integrated circuit LF 398 is used as an active component followed by buffer. The LF198/LF298/LF398 is monolithic sample-and-hold circuits which utilize BI-FET technology to obtain ultra-high dc accuracy with fast acquisition of signal and low droop rate. Operating as a unity gain follower, dc gain accuracy is 0.002% typical and acquisition time is as low as 6μs to 0.01%. A bipolar input stage is used to achieve low offset voltage and wide bandwidth.

Input offset adjust is accomplished with a single pin, and does not degrade input offset drift. The wide bandwidth allows the LF198 to be included inside the feedback loop of 1 MHz op amps without having stability problems. Input impedance of 1010(Ohm) allows high source impedance to be used without degrading accuracy.

P-channel junction FET's are combined with bipolar devices in the output amplifier to give droop rates as low as 5mV/min with a 1μf hold capacitor. The JFET's have much lower noise than MOS devices used in previous designs and do not exhibit high temperature instabilities. The overall design guarantees no feed-through from input to output in the hold mode, even for input signals equal to the supply voltages.

Logic inputs on the LF198 are fully differential with low current, allowing direct connection to TTL, PMOS, and CMOS. Differential threshold is 1.4V. The LF198 willoperate from +5V to +18V supplies.

8 Bit A/D Converter (AET-68M):

This has been constructed with a popular 8 bit successive approximation A/D Converter IC ADC0808. The ADC0808, data acquisition component is a monolithic CMOS device with an 8-bit analog-to-digital converter, 8-channel multiplexer and microprocessor compatible control logic. The 8-bit A/D converter uses successive approximation as the conversion technique. The converter features a high impedance chopper stabilized comparator, a 256R voltage divider with analog switch tree and a successive approximation register. The 8-channel multiplexer can directly access any of

8-single-ended analog signals. A dedicated 1MHz clock generator is provided in side this block. For complete specifications and operating conditions please refer the data sheet of ADC0808.

8 Bit Parallel-Serial Shift register (AET-68M):

A dedicated parallel in serial out shift register integrated circuit is used followed by a latch The SN74LS166 is an 8-Bit Shift Register. Designed with all inputs buffered, the drive requirements are lowered to one 74LS standard load. By utilizing input clamping diodes, switching transients are minimized and system design simplified.

The LS166 is a parallel-in or serial-in, serial-out shift register and has a complexity of 77 equivalent gates with gated clock inputs and an overriding clear input.

The shift/load input establishes the parallel-in or serial-in mode. When high, this input enables the serial data input and couples the eight flip-flops for serial shifting with each clock pulse.

Synchronous loading occurs on the next clock pulse when this is low and the parallel data inputs are enabled. Serial data flow is inhibited during parallel loading. Clocking is done on the low-to-high level edge of the clock pulse via a two input positive NOR gate, which permits one input to be used as a clock enable or clock inhibit function.

Clocking is inhibited when either of the clock inputs are held high, holding either input low enables the other clock input. This will allow the system clock to be free running and the register stopped on command with the other clock input. A change from low-to-high on the clock inhibit input should only be done when the clock input is high. A buffered direct clear input overrides all other inputs, including the clock, and sets all flip-flops to zero. For complete specifications and operating conditions please refer the data sheet of SN74LS166.

8 Bit Serial-Parallel Shift register (AET-68D):

A dedicated serial in parallel out shift register integrated circuit is used followed by a latch. The SN74LS164 is a high speed 8-Bit Serial-In Parallel-Out Shift Register.

Serial data is entered through a 2-Input AND gate synchronous with the LOW to HIGH transition of the clock. The device features as asynchronous Master Reset which clears the register setting all outputs LOW independent of the clock. It utilizes the Schottky diode clamped process to achieve high speeds and is fully compatible with all TTL products. For complete specifications and operating conditions please refer the data sheet of SN74LS164.

8 Bit D/A Converter (AET-68D):

This has been constructed with a popular 8 bit D/A converter IC DAC 0808. The DAC0808 is an 8-bit monolithic digital-to-analog converter (DAC) featuring a full scale output current settling time of 150ns while dissipating only 33mW with +5V supplies No reference current (IREF) trimming is required for most applications since the full scale output current is typically +1 LSB of 255 IREF/256. Relative accuracies of better than +0.19% assure 8-bit monotonicity and linearity while zero level output current of less than 4μA provides 8-bit zero accuracy for IREF >= 2mA.

The power supply currents of the DAC0808 is independent of bit codes, and exhibits essentially constant device characteristics over the entire supply voltage range. For complete specifications and operating conditions please refer the data sheet of DAC0808.

PCM Operation:

The block diagram of the PCM system. The modulating signal is applied to sample & hold circuit. This applied signal will be super imposed by +2.5V DC so that the negative portion the modulating signal will clamped to positive, this process is needed, because input of the A/D Converter should be between 0 and +5V. After level shifting is done the signal will be passed to sample & hold circuit. Sample & hold circuit will sample the input signal during on period of the clock signal and will hold the sampled output till next pulse comes. Sampling rate is 4KHz in this system.

So input of the A/D Converter is a stable voltage of certain level in between 0 and +5V. A/D converter (encoder) will give a predetermined 8 bit code for the sampled input. This entire conversion process will be made at a fast rate as ADC0808 is operating at high frequency clock

i.e. 1MHz. Coded output of the A/D converter is applied to input of the parallel in serial out register through a latch (741s373). This shift register is operating at 64KHz (sampling frequency is 4KHz, so to shift 8 bits from parallel to serial we need 64KHz). This output (PCM) is transmitted through a co-axial cable which represents a communication channel.

PCM signal from modulator (encoder) is applied to serial to parallel register. This shift register is also operating at 64KHz clock at which parallel to serial shift register is operating at PCM modulator (these both the clock signals should be in synchronized with each other in order to get proper decoded output). So the output of the serial to parallel register is a 8 bit code. This 8 bit code is applied to 8 bit D/A converter. Output of the D/A converter will be a staircase signaling between 0 and +5V. This stair case signal is applied a low pass filter. This low pass will smoothen the staircase signal so that we will get a recovered AF signal. We can use a voltage amplifier at the output of the low pass filter to amplify the recovered AF signal to desiredvoltage level.

PROCEDURE:

1. Connect the modulator trainer to the mains and switch on the power supply.
2. Observe the output of the AF generator using CRO, it should be a sine wave of 200Hzfrequency with 3Vpp amplitude.
3. Verify the output of the DC source with multimeter/scope, output should vary from 0 to +5V.
4. Observe the output of the clock generator using CRO, they should be 64KHz and 4KHzfrequency of square wave with 5Vpp amplitude.
5. The clock signals are internally connected the circuit so no external connections are required
6. Connect the demodulator trainer to the mains and switch on the power supply.
7. Observe the output of the clock generator using CRO, it should be 64KHz square wave with5Vpp amplitude.

PCM Operation (with DC input):
Modulation:

8. Set DC source to some value say 4.4V with the help of multimeter and connect it to the A/D converter input and observe the output LED's
9. Note down the digital code i.e. output of the A/D converter and compare with the theoreticalvalue. Keep CRO in dual mode. Connect one channel to 4KHz signal (one which is connected tothe Shift register) and another channel to the PCM output.
10. Observe the PCM output with respect to 4 Khz signal and sketch the waveforms.Compare them with the given waveforms

Demodulation

11. Connect PCM signal to the demodulators(S-P shift register) from the PCM modulator(AET-68M) with the help of coaxial cable.
12. Connect clock signal (64KHz) from the transmitter (AET-68M) to the receiver (AET-68D)using co axial cable.
13. Connect transmitter clock to the timing circuit.
14. Observe and note down the S-P shift register output data and compare it with transmitted data(i.e. output A/D converter at transmitter).You will notice that the output of the S-P shift register is following the A/D converter output in the modulator.
15. Observe D/A converter output (Demodulated output) using multimeter /scope and compare it with the original signal and you can observe that there is no loss in information in process of conversion and transmission.

Sample work sheet:

1. Modulating signal : 4.4 V
2. A/D Output (theoretical) : 1110 0001(2)
3. A/D Output (practical) : 1110 0001(2)
4. S-P Output : 1110 0001(2)
5. D/A Converter output : 4.4 V(Demodulated output)

PCM Operation (with AC input):
Modulation:

16. Connect AC signal of 2Vpp amplitude to Sample & Hold circuit.
17. Keep the CRO in dual mode. Connect one channel to the AF signal and another channel tothe Sample & Hold output. Observe and sketch the sample & hold output.
18. Connect the Sample and Hold output to the A/D converter and observe the PCM output usingStorage oscilloscope.
19. Observe PCM output by varying AF signal voltage.

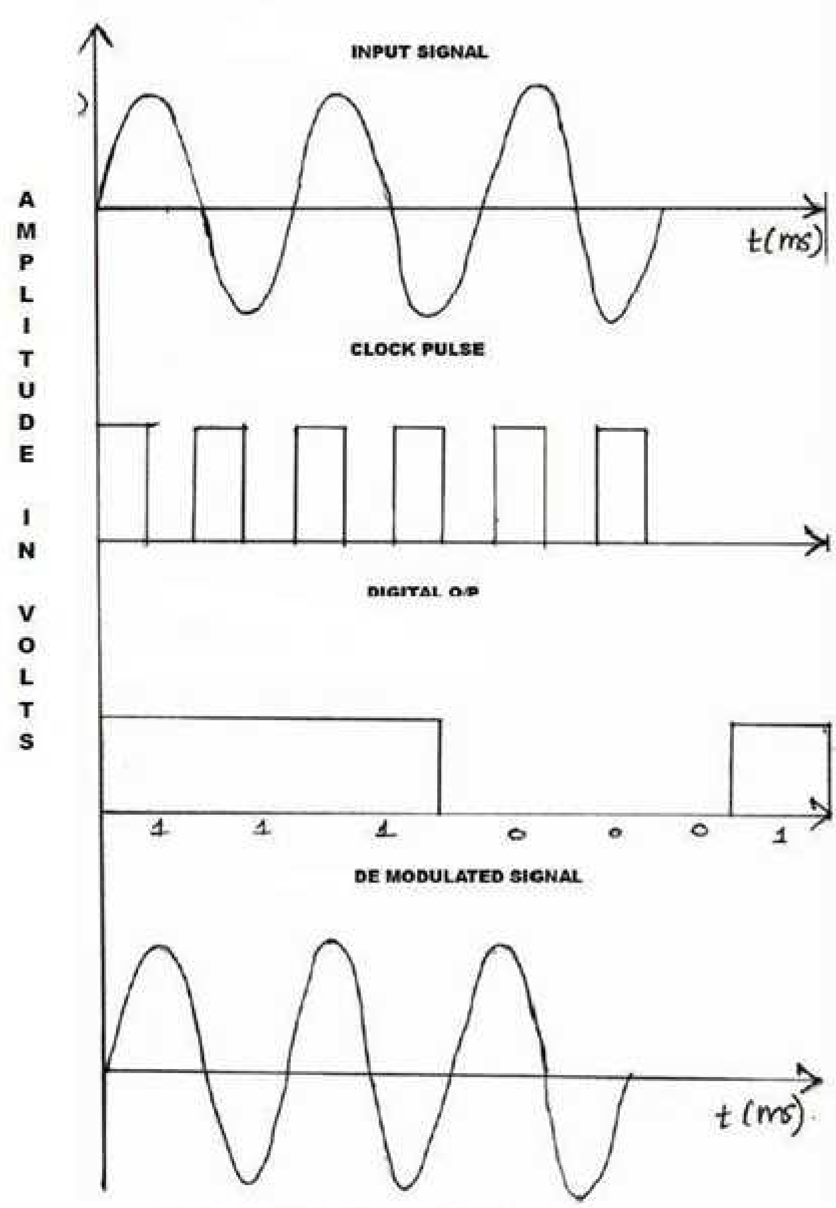
Demodulation:

20. Connect PCM signal to the demodulator input (AET-68D) (S-P shift register) from the PCMmodulator (AET- 68M) with the help of coaxial cable (supplied with the trainer).
21. Connect clock signal (64 KHz) from the transmitter (AET-68M) to the receiver (AET-68D)using coaxial cable.
22. Connect transmitter clock to the timing circuit.
23. Keep CRO in dual mode. Connect CH1 input to the sample and hold output (AET-68M) andCH2 input to the D/A converter output (AET-68D)
24. Observe and sketch the D/A output.
25. Connect D/A output to the LPF input.
26. Observe the output of the LPF/Amplifier and compare it with the original modulating signal(AET-68M).
27. From above observation you can verify that there is no loss in information (modulating signal) in conversion and transmission process.
28. Disconnect clock from transmitter (AET-68M) and connect to local oscillator (i.e.,Clock generator output from AET-68D) with remaining setup as it is.

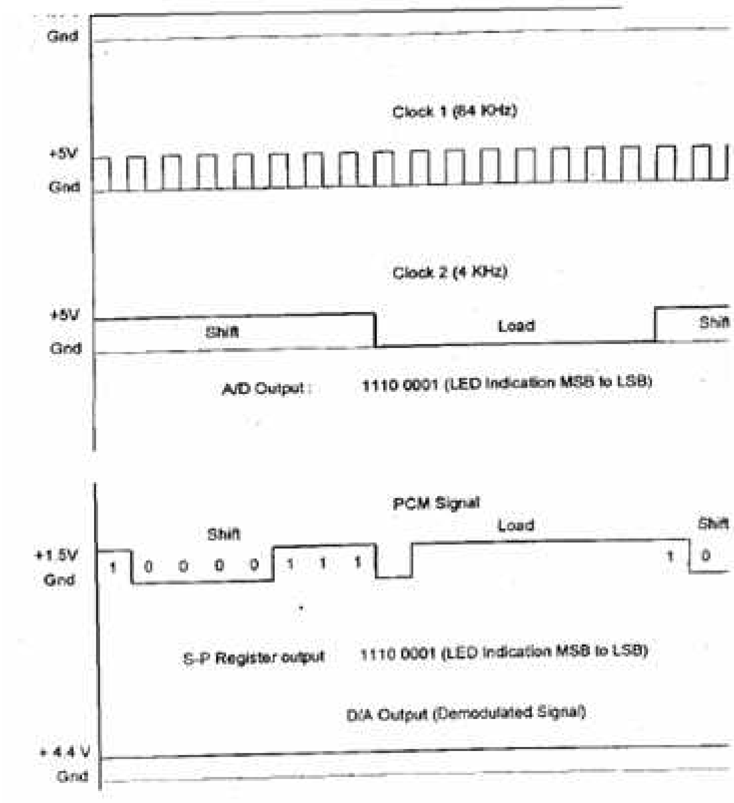
Observe D/A output and compare it with the previous result. This signal is little bit distorted in shape. This is because lack of synchronization between clock at transmitter and clock at receiver.

Note: You can take modulating signals from external sources. Maximum amplitude should not exceed 4V incase of DC and 3 Vpp incase AC (AF) signals.

EXPECTED WAVE FORMS:



WITH DC INPUT:



OBSERVATIONS: PCM Modulation with AC input

	Amplitude	Time period
AC input		
Sample and hold circuit		
Clock signal(4KHz)		
Clock signal(64KHz)		
PCM Output		
D/A converter output signal		
LPF output signal		
Demodulated output		

PCM Modulation with DC input

	Amplitude	Time period
DC input		
Clock signal(4KHz)		
Clock signal(64KHz)		
PCM Output		

PCM Demodulated (with DC input)

	Amplitude	Time period
D/A converter output signal		
LPF output signal		
Demodulated output		

RESULT:

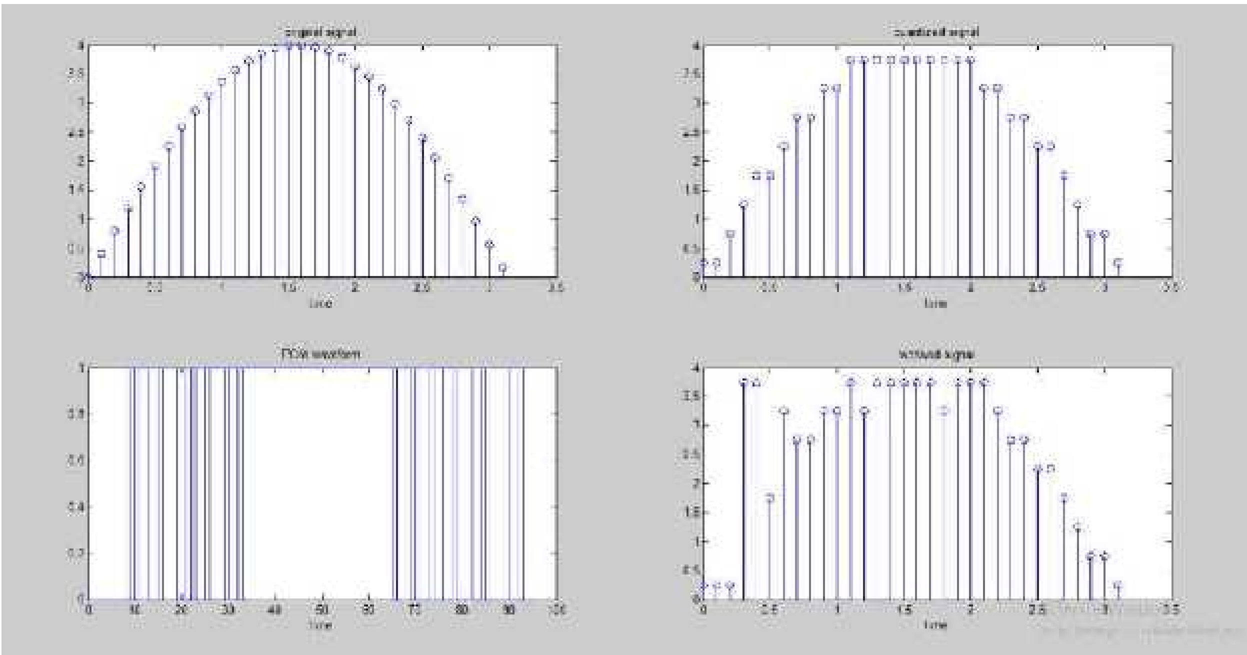
Thus the Pulse Code modulation and demodulation were performed and graphs were plotted.

MATLAB CODE:

```
clc;
close all;
clear all;
n=input('Enter n value for n-bit PCM system :  ');
n1=input('Enter number of samples in a period : ');
L=2^n;
% % Signal Generation
% x=0:1/100:4*pi;
% y=8*sin(x);                             % Amplitude Of signal is 8v
% subplot(2,2,1);
% plot(x,y);grid on;
% Sampling Operation
x=0:2*pi/n1:4*pi;               % n1 nuber of samples have tobe selected
s=8*sin(x);
subplot(3,1,1);
plot(s);
title('Analog Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(3,1,2);
stem(s);grid on;  title('Sampled Signal');  ylabel('Amplitude--->');  xlabel('Time---
>');
 %  Quantization Process
 vmax=8;
 vmin=-vmax;
 del=(vmax-vmin)/L;
 part=vmin:del:vmax;                                  % level are between vmin and vmax
with difference of del
 code=vmin-(del/2):del:vmax+(del/2);        % Contaion Quantized valuses
 [ind,q]=quantiz(s,part,code);                     % Quantization process
                                                                      % ind contain
index number and q contain quantized  values
 l1=length(ind);
 l2=length(q);

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

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

 %   Demodulation Of PCM signal

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

OUTPUT WAVE FORM



POST LAB QUESTIONS:

1. What do you mean by quantizing process?

2. What will happen when sampling rate is greater than Nyquist rate ?

3. What will happen when sampling rate is less than Nyquist rate ?

4. Find the A/D Converter output for input DC voltage of 3.6V.

5. Fig shown below shows a PCM wave in which the amplitude levels of +1 volt and -1 volt are used to represent binary symbols 1 and 0 respectively. The code word used consists of three bits. Find the sampled version of an analog signal from which this PCM wave is derived.

6. Mention some applications of PCM.

7. What is the function of Sample and Hold circuit?

EVALUATION SHEET

Department of Electronics and Communication Engineering

SCSVMV UNIVERSITY, Enathur, Kanchipuram

Title of the Experiment	
Name of the Candidate	
Register Number	
Date of Submission	

S.No	Marks Split up	Maximum Marks	Marks Earned
1	Attendance	5	
2	Pre lab viva questions	5	
3	Execution of Experiments	20	
4	Calculation/Evaluation of Result	10	
5	Post lab viva questions	10	
6	Grand Total	50	

Signature of the Faculty Handling the Lab

Delta modulation and Adaptive Delta modulation

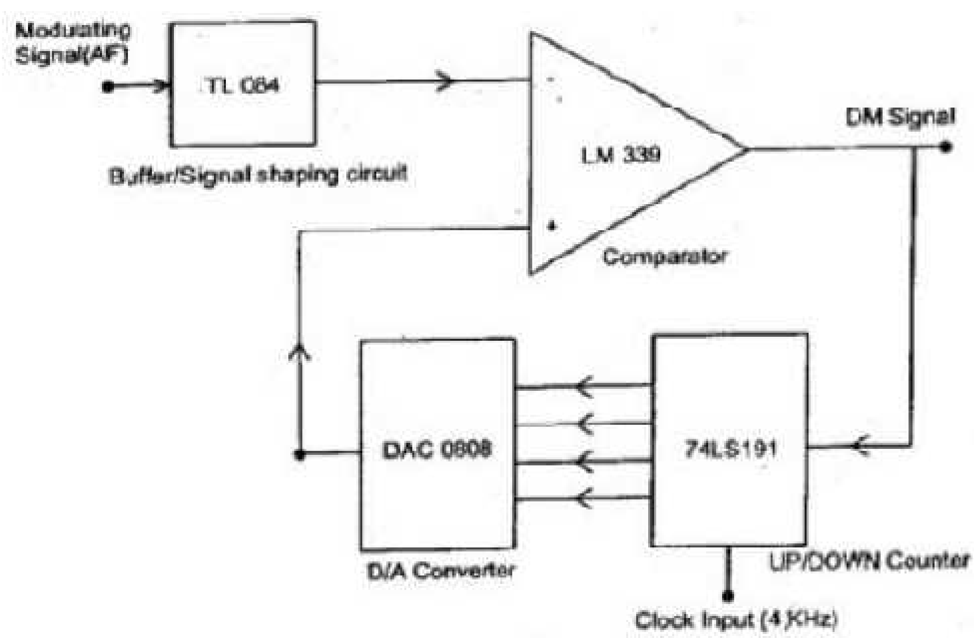
AIM:

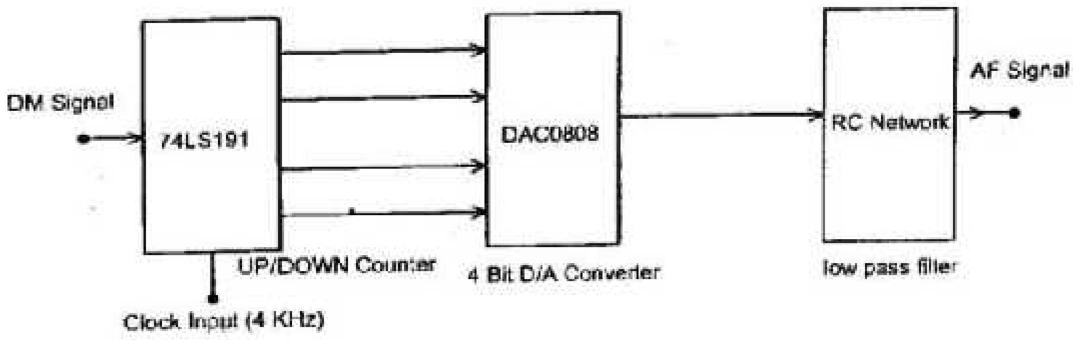
To analyze a Delta modulation system, Adaptive Delta modulation and interpret the modulated and demodulated waveforms

APPARATUS:

1. PCM Modulator trainer- AET-73M
2. PCM Demodulator trainer-AET-73D
3. C.R.O(30MHz)
4. Patch chords.
5. PC with windows(95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

BLOCK DIAGRAM: DELTA MODULATOR & DEMODULATOR





INTRODUCTION

Delta Modulation is a form of pulse modulation where a sample value is represented as a single bit. This is almost similar to differential PCM, as the transmitted bit is only one per sample just to indicate whether the present sample is larger or smaller than the previous one. The encoding, decoding and quantizing process become extremely simple but this system cannot handle rapidlyvarying samples. This increases the quantizing noise.

The trainer is a self sustained and well organized kit for the demonstration of delta modulation &demodulation .The system consist of :

DM Modulator (AET-73M) trainer kit

1. Regulated power supply
2. Audio Frequency signal generator
3. Buffer/signal shaping network
4. Voltage comparator
5. 4 Bit UP/DOWN counter
6. Clock generator/Timing circuit
7. 4 Bit D/A converter
8. DC source

DM Demodulator (AET-73D) trainer kit

1. Regulated power supply
2. 4 Bit UP/DOWN counter
3. 4 Bit D/A converter
4. Clock generator
5. Passive low pass filter
6. Audio amplifier

Regulated power supply (73M & 73D):

This consists of a bridge rectifier followed by Capacitor filters and three terminal regulators 7805 and 7905 to provide regulated DC voltages of +-5V and +12V@ 300Ma each to the on board circuits. These supplies have been internally connected to the circuits. so no external connectionsare required for operation.

Audio Frequency (AF) S signal generator (73M): Sine wave signal of 100 Hz is generated to use as a modulating (message or information) signal to be transmitted. This is an Op-Amp based Wein bridge Oscillators using IC TL084 is a FET.input general purpose Operational Amplifier .Amplitude control is provided in the circuit tovary the output amplitude of AF signal.

Clock generator/Timing circuit (73M & 73D):

A TTL compatible clock signal of 4 KHz frequency is provided on board to use as a clock to the various circuits in the system. This circuit is a astable multivibrator using 555 timer followed bya buffer.

DC Source (73M):

A 0 to +5V variable DC voltage is provided on board to use as a modulating signal instead of AF Signal. is useful to study step by step operation of Delta modulation and Demodulation. This is a simple circuit consists of potentiometer and fixed power supply.

Buffer/Signal shaping circuit (73M):

A non inverting buffer using IC TL 084 is provided at the input of the DM modulator followed by a level shifting network. Buffer provides the isolation between DM circuit and the signal source. Signal Shaping super imposes the 1.5V DC on incoming modulating signal so that the input of the comparator lies between 0 and +3V maximum.

Voltage comparator (73D):

This circuit is build with IC LM339 The LM339 series consists of four independent precision voltage comparators with an offset voltage specification as low as 2mV for all four comparators. These were designed specifically to operate from a single power supply over a wide range of voltages .Operation from split power supplies is also possible and the low power supply current drain is independent of the magnitude of the power supply voltage. These comparators also have a unique characteristic in that the common mode voltage range includes ground, even though operated from a single power
supply voltage. Application areas include limit comparators simple analog to digital converters: pulse, square and time delay generators. wide range VCO; MOS clock timers; multivibrators and high voltage digital logic gates .The LM139 series was designed to directly interface with TTL and CMOS. When operated from both plus and minus power supplies, they will directly interface with MOS logic where the low power drain of the LM339 is a distinct advantage over standard comparators .For circuit connections and other operating conditions.

Low pass filters (73D):

This is a series of simple RC networks provided on board to smoothen the output of the D/A converter output. RC values are chosen such that the cutoff frequency would be at 100 Hz.

Amplifiers (73D):

This is an Op-amp (IC TL084) based non-inverting variable gain amplifiers provided on board to amplify the recovered message singles i.e. output of low pass filter to desired level. Amplitude control is provided in circuit to vary the gain of the amplifier between 0 and 6.AC/DC Switch facilitates to couple the input signal through capacitor to directly to the amplifier input.

4 Bit UP/DOWN Counter (73M & 73 D):

This circuit is made using Synchronous 4-Bit Up/Down Counter with Mode Control IC 74LS191. The DM 74LS191 circuit is a synchronous, reversible, counter. Synchronous operation is provided by having all flip-flops clocked simultaneously. So that the outputs change simultaneously when so instructed by the steering logic. This mode of operation eliminates the output counting spikes normally associated with the asynchronous counters. The outputs of the four master slave flip flops are triggered on a LOW to HIGH level transition of the clock input. if the enable input is LOW a HIGH at the enable input inhibits counting .Level changes at either the enable input or the down/up input should be made only when the clock input is HIGH. The direction of the count is determined by the level of the down/up input. When LOW the counter counts up and when HIGH it counts down. The counter is fully programmable that is the outputs may be preset to either level by placing a LOW on the load input and entering the desired data at the data inputs. The output will change independent of the level of the clock input. This feature allows the counters to be used as modulo-N dividers by simply modifying the count length with the preset inputs. The clock, down/up and load inputs are buffered to lower the drive requirement which significantly reduces the number of clock drivers required for parallel words. The ripple clock input produces a low level output pulse equal in width to the low level portion of the clock input when an overflow or underflow condition exists. The counters can be easily cascaded by feeding the ripple clock output to the enable input of the succeeding counter if parallel clocking is used, or to the clock input if parallel enabling is used. The maximum/minimum count output can be used to accomplish look-ahead for high speed operation.

4 Bit D/A converter (AET-73M & 73D):

This has been constructed with a popular 8 bit D/A Converter IC DAC 0808.The DAC0808 is an 8-bit monolithic DAC featuring a full scale output current settling time of 150 Ns while dissipating only 33 maw with +-5V supplies. No reference current (I REF) trimming is required for most applications since the full scale output current is typically +- 1 LSB of 255 I REF/256

.Relative accuracies of better than +- 0.19 % assure 8 bit monotonic and linearity while zero level output current of less than 4 μA provides 8-bit zero accuracy for I REF [greater than or equal] 2 math power supply currents of the DAC0808 is independent of bit codes, and exhibits essentially constant device characteristics over the entire supply voltage range.4 LSB Bits are permanently grounded

to make 4 bit converter. For complete specifications and operating conditions please refer the data sheet of DAC0808.

DM Operation:

Figure 4.1 shows the basic block diagram of the PCM system. The modulating signal is applied to buffer /signal shaping network. This applied signal will be superimposed by +1.5V DC so that the negative portion the modulating signal will clamped to positive ,this process is needed ,because input of the comparator should be between 0 and +3V. After level shifting is done the signal will be passed to inverting input of the comparator. on inverting input of the comparator is connected to output of the 4 Bit D/A converter. Comparator is operating at +5V single supply .So output of the comparator will be high (i.e. +vet Vast) when modulating signal is less than the reference signal i.e. D/A output, otherwise it will be 0V. And this signal is transmitted as DM signal .same signal is also connected as UP/DOWN control to the UP/DOWN counter (74LS 191). UP/DOWN counter is programmed for 0000 starting count. So initially output of the counter is at 0000 and the D/A converter will be at 0V .Comparator compares the modulating signal is greater than the reference signal. For next clock pulse depending on the UP/DOWN input counter will count up or down. If the UP/DOWN input is low (nothing but comparator output). Counter will make up and output will be 0001. So the D/A converter will convert this 0001 digital input to equivalent analog signal(i.e. 0.3V 1 LSB Value).Now the reference signal is 0.3V.If still modulating signal is greater than the D/A output again comparator output(DM) will be low and UP count will occur. If not DOWN Count will take place. This process will continue till the reference signal and modulating signal voltages are equal. So DM signal is a series of 1 and 0. DM signal is applied to a UP/DOWN input of the UP/DOWN counter at the receiver. This UP/DOWN counter is programmed for 1001 initial value (i.e. power on reset) and mode control is activated. So depend on the UP/DOWN input for the next clock pulse counter will count UP or DOWN. This output is applied to 4 Bit D/A converter. A logic circuit is added to the counter which keeps the output of the counter in between 0000 to 1111 always. Output of the D/A converter will be a staircase signal lies between 0 and +4.7V.This staircase signal is applied a low pass filter .This low pass will smoothen the staircase signal so that original AF signal will be recovered. We can use a voltage amplifier at the output of the low pass filter to amplify the recovered AF signal to desired voltage level.

PROCEDURE:

DM Modulator:

1. Study the theory of operation
2. Connect the trainer (AET-73M) -
3. Observe the output of AF generator using CRO; it should be a Sine wave of 100 Hz frequencywith 3Vpp amplitude.
4. Verify the output of the DC source with multimeter/scope; output should vary 0 to +4V
5. Observe the output of the clock generator using Crotchety should be 4 KHz frequency ofsquare wave with 5 Up amplitude.

Note: This clock signal is internally connected to the up/down counter so no external connectionis required.

DM With DC Voltage as modulating signal:

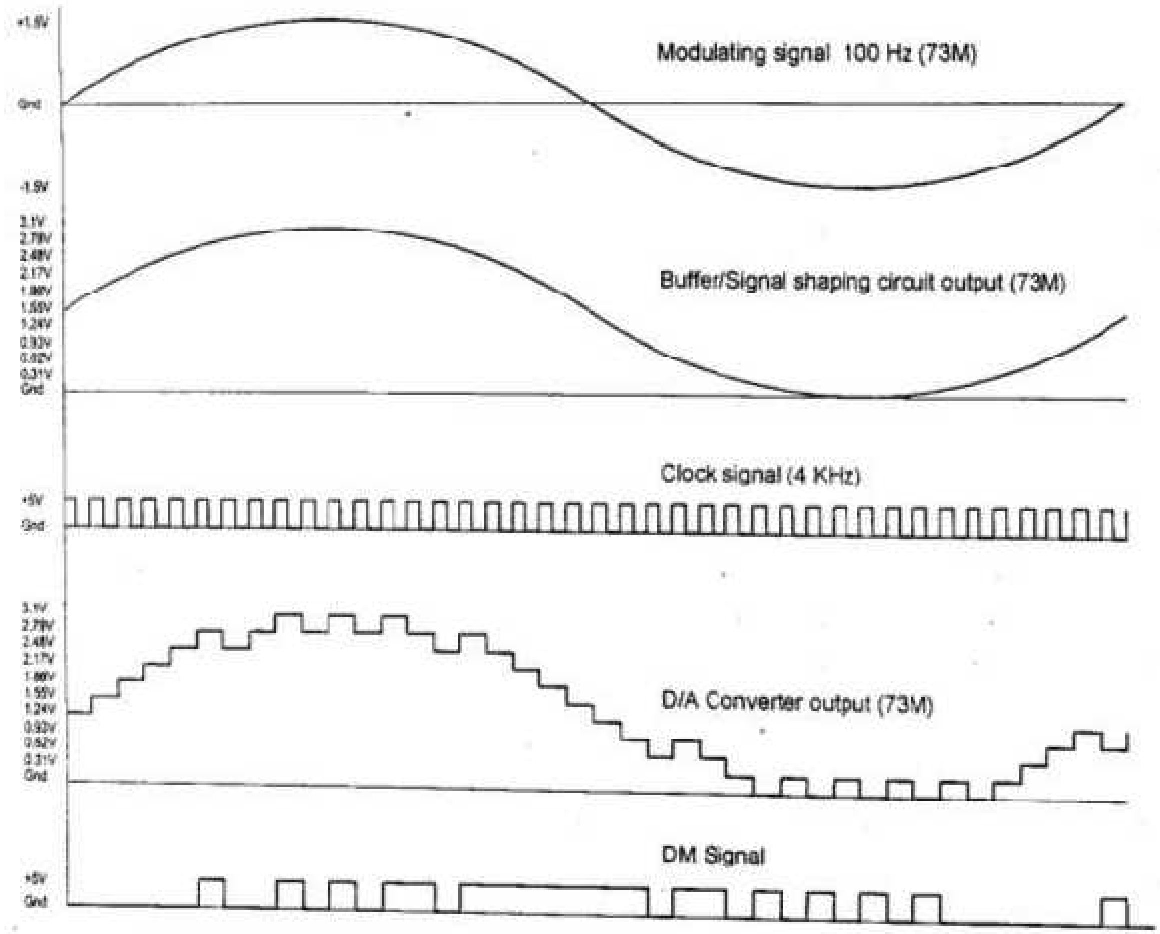
6. Connect DC signal from the DC source to the inverting input of the comparator and set somevoltage says 3V.
7. Observe and plot the signals at D/A converter output (i.e. non-inverting input of the comparator), DM signal using CRO and compare them with the waveforms given in figure.
8. Connect DM signal (from 73M) to the DM input of the demodulator.
9. Connect clock (4KHz) from modulator (73M) to the clock input of the demodulator (73D). Connect clock input of UP/DOWN counter (in 73D) to the clock from transmitter with the help of springs provided.
10. Observe digital output (LED indication) of the UP/DOWN counter (in 73 D) and compare it with the output of the UP/DOWN (in 73M) .By this you can notice that the both the outputs are same.
11. Observe and plot the output of the D/A converter and compare it with the waveforms given in figure.
12. Measure the demodulated signal (i.e. output of the D/A converter 73D with the help of multimeter and compare it with the original signal 73 M. From the above observation you can notice that both the voltages are equal and there is no loss in process of modulation, transmissionand demodulation.
13. Similarly you can verify the DM operation for different values of modulating signal.

DM With AF Voltage as modulating signal:

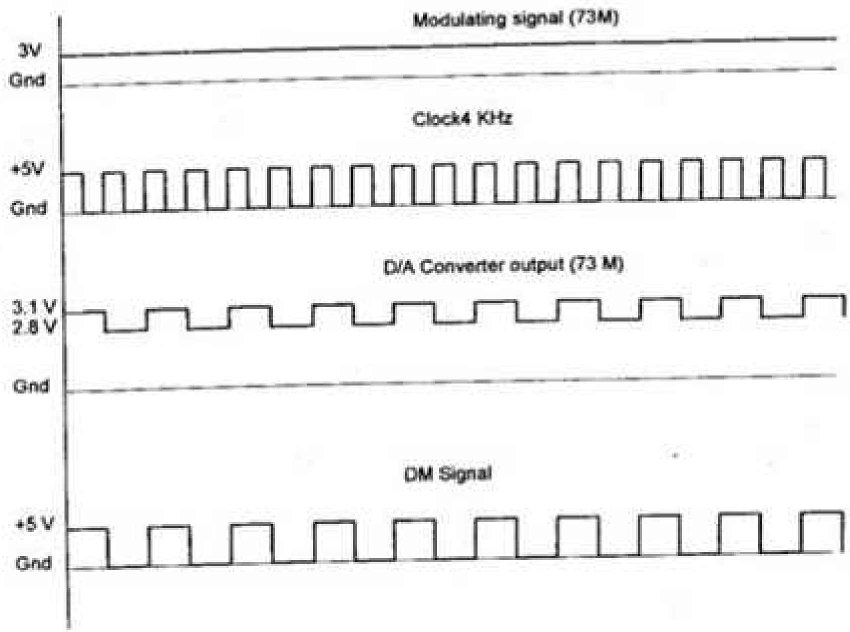
14. Connect AF signal from the AF source to the inverting input of the comparator and set omevoltage says 3V.
15. Observe and plot the signals at D/A converter output (i.e. non-inverting input of the comparator), DM signal using CRO and compare them with the waveforms given in figure.
16. Connect DM signal (from 73M) to the DM input of the demodulator.
17. Connect clock (4KHz) from modulator (73M) to the clock input of the demodulator (73D).
18. Connect clock input of UP/DOWN counter (in 73D) to the clock from transmitter with thehelp of springs provided.
19. Observe and plot the output of the D/A converter and compare it with the waveforms givenin figure.
20. Observe and sketch the D/A output.
21. Connect D/A output to the LPF input.
22. Observe the output of the LPF/Amplifier and compare it with the original modulating signal(AET-73M).
23. From the above observation you can verify that there is no loss in information in conversion and transmission process.
24. Disconnect clock from transmitter (AET-73M) and connect to local oscillator (i.e. clock generator output from AET-73D) with remaining setup as it is. Observe demodulated signal output and compare it with the previous result. This signal is little bit distorted in shape. This is because lack of synchronization between clock at transmitter and clock at receiver.

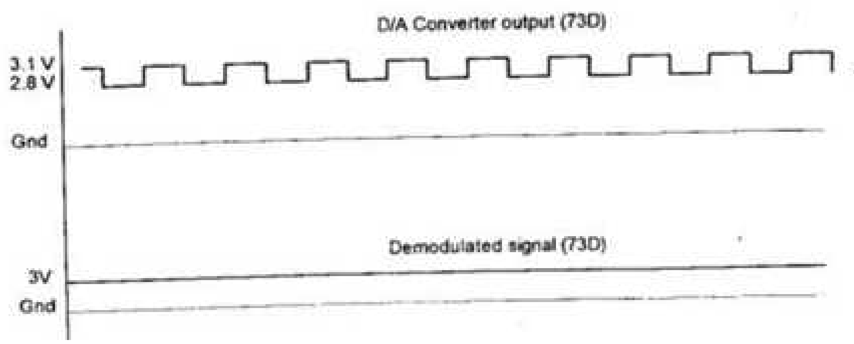
Note: you can take modulating signals from external sources. Maximum amplitude should not exceed 4 V in case of DC and 3 Vpp in case of AC (AF) signals.

EXPECTED WAVE FORMS: FOR AC INPUT



FOR DC INPUT


Modulating signal (73M)
3V
Gnd
Clock4 KHz
+5V
Gnd
D/A Converter output (73 M)
3.1 V
2.8 V
Gnd
DM Signal
+5 V
Gnd


D/A Converter output (73D)
3.1 V
2.8 V
Gnd
Demodulated signal (73D)
3V
Gnd

OBSERVATIONS: DM MODULATION WITH AC INPUT

	amplitude	Time period
AC input		
D/A converter output		
Clock signal (4 KHz)		
DM output		

DM DEMODULATION WITH AC INPUT

	amplitude	Time period
DM input		
D/A converter output signal		
Demodulated Output		
Clock signal (4 KHz)		

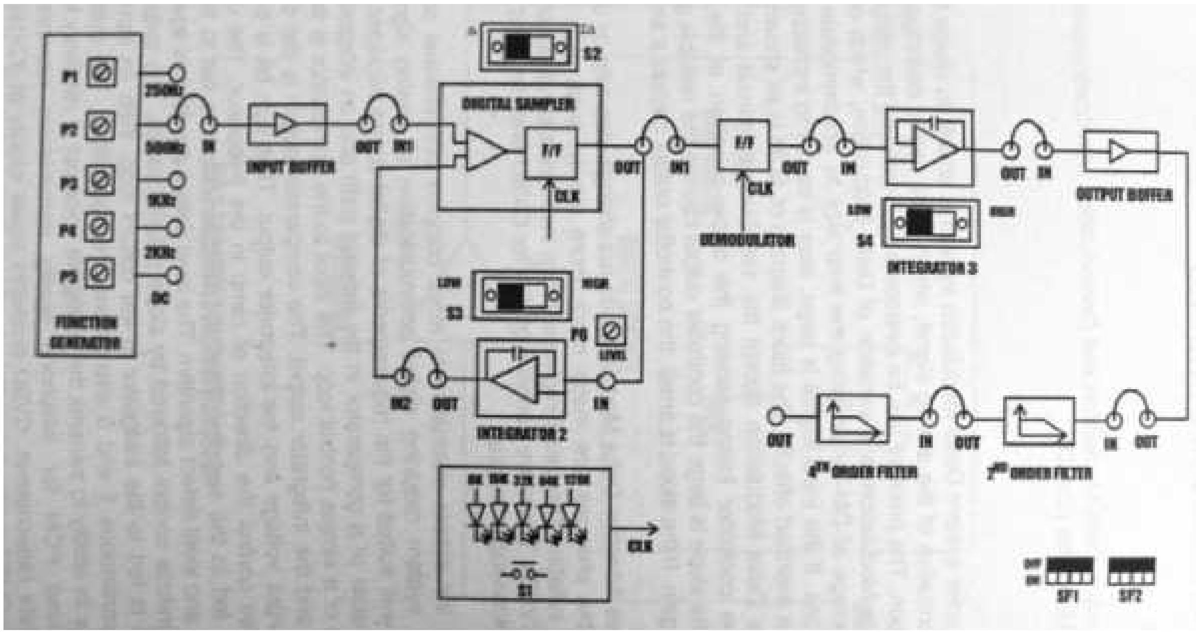
DM MODULATION WITH DC INPUT

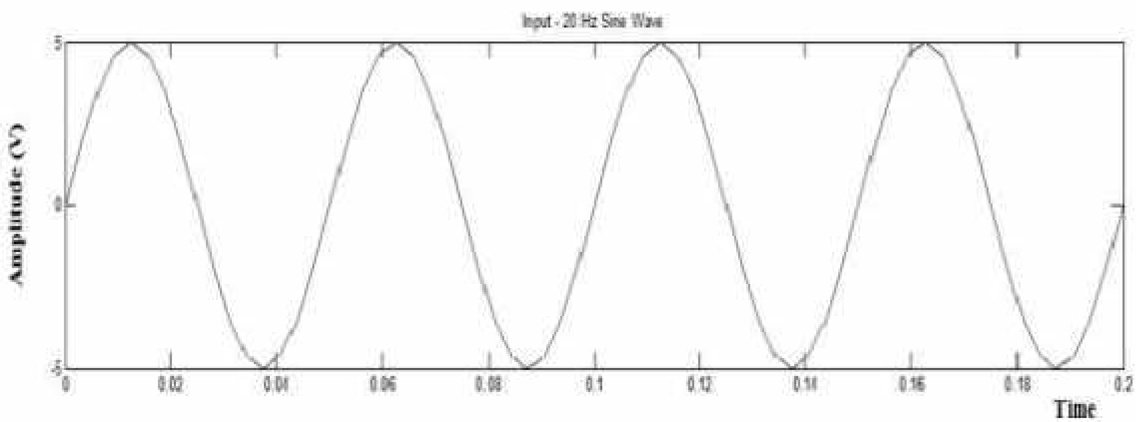
	amplitude	Time period
DC input		
D/A converter output		
Clock signal (4 KHz)		
DM output		

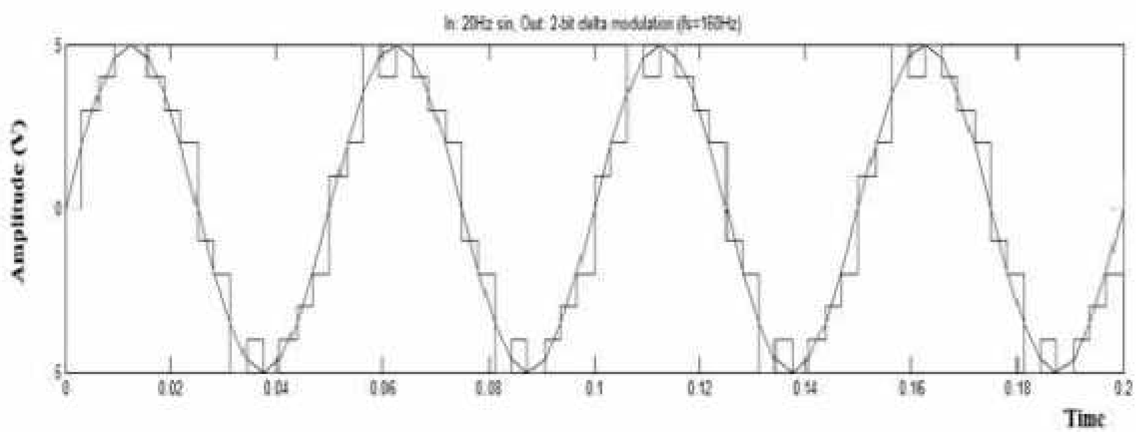
DM DEMODULATION WITH DC INPUT

	amplitude	Time period
DM input		
D/A converter output signal		
Demodulated Output		
Clock signal (4 KHz)		

Adaptive Delta modulation







	AMPLITUDE	TIME PERIOD	FREQUENCY
Message Signal			
Digital Sampler O/P			
Integrator -3 O/P			
Filter O/P			

RESULT:

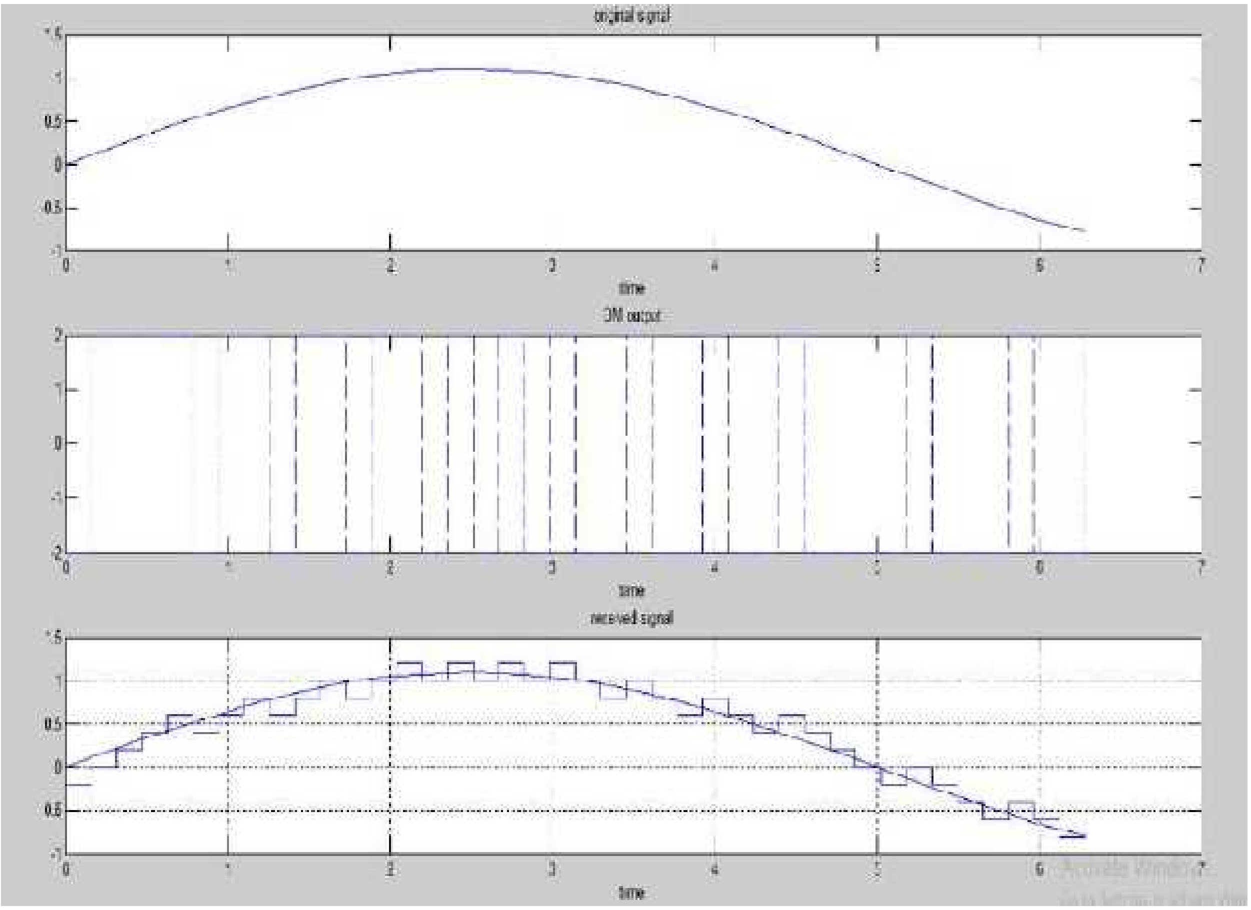
Thus the Delta modulation and demodulation, Adaptive Delta modulation & Demodulation were performed and graphs wereplotted.

MATLAB CODE:

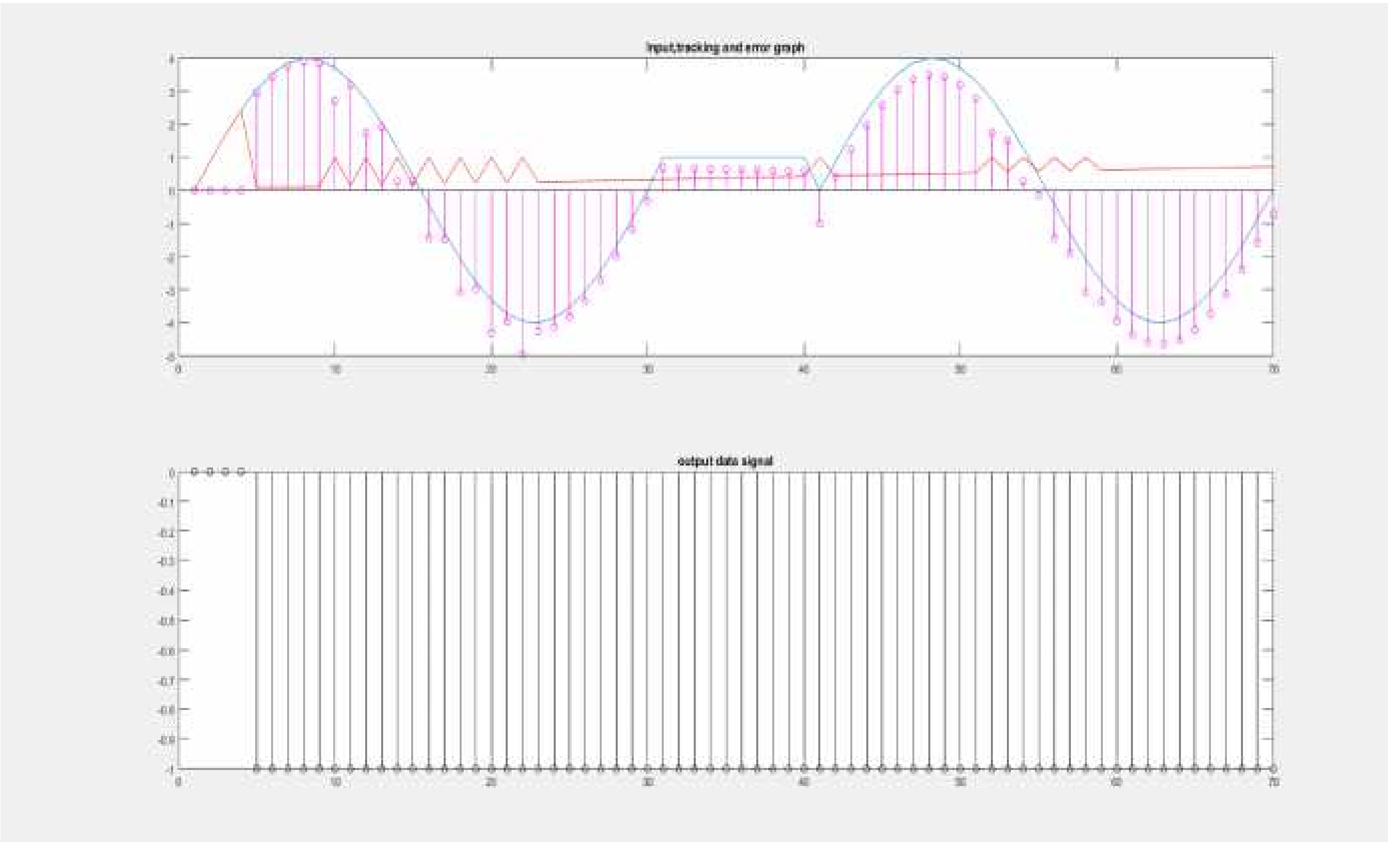
```
%% Delta Modulation (DM)

%delta modulation = 1-bit differential pulse code modulation (DPCM)
predictor = [0 1]; % y(k)=x(k-1)
%partition = [-1:.1:.9];codebook = [-1:.1:1];
step=0.2; %SFs>=2pifA
partition = [0];codebook = [-1*step step]; %DM quantizer

t = [0:pi/20:2*pi];
x = 1.1*sin(2*pi*0.1*t); % Original signal, a sine wave
%t = [0:0.1:2*pi];x = 4*sin(t);
%x=exp(-1/3*t);
%x = sawtooth(3*t); % Original signal

% Quantize x(t) using DPCM.
encodedx = dpcmenco(x,codebook,partition,predictor);

% Try to recover x from the modulated signal.
decodedx = dpcmdeco(encodedx,codebook,predictor);
distor = sum((x-decodedx).^2)/length(x) % Mean square error

% plots
figure,subplot(2,2,1);plot(t,x);xlabel('time');title('original signal');
subplot(2,2,2);stairs(t,10*codebook(encodedx+1),'--');xlabel('time');title('DM
output');
subplot(2,2,3);plot(t,x);hold;stairs(t,decodedx);grid;xlabel('time');title('received
signal');
```

```matlab
%% Adaptive Delta Modulation (ADM)

clc;
clear all;
t =0:1/29:1;
f=1;
x=4*sin(2*pi*f*t);
x=[x ones(1,10) x];
y = zeros(1,length(x));
d = zeros(1,length(x));
e= zeros(1,length(x));
s=0.1;
for i=5:length(x)
if(x(i)-y(i-1))>=0
      y(i) = x(i)-s;
        d(i)=-1;
elseif(x(i)-y(i-1))<0
y(i)=x(i)-1;
d(i)=-1;
end
 if(sum(d(i-4:i)))>3
s=s+0.01;
 elseif(sum(d(i-4:i)))<-3
  s=s+0.01;
 elseif(sum(d(i-4:i)))==0
s=s-0.01;
else
s=s;
end
     end
subplot (2,1,1);
plot(x);
hold on;
stem(y,'m');
e=x-y;
plot(e,'r');
title('Input,tracking and error graph');
subplot (2,1,2);
stem (d,'k');
title('output data signal');
```

Delta Modulation



Adaptive Delta Modulation



POST LAB QUESTIONS

1. Compare DPCM ,PCM & Delta modulation.
2. How to reduce the quantization noise that occurs in DM?
3. A band pass signal has a spectral range that extends from 20 to 82 KHz.Find the acceptablesampling frequency.
4. Find the fourier series expansion of an Impulse train.
5. Mention the applications of DM.
6. What is the difference between DM and ADM?

EVALUATION SHEET

Department of Electronics and Communication Engineering

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5	Post lab viva questions	10	
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Signature of the Faculty Handling the Lab

Amplitude Shift Keying (ASK) modulator and Demodulator

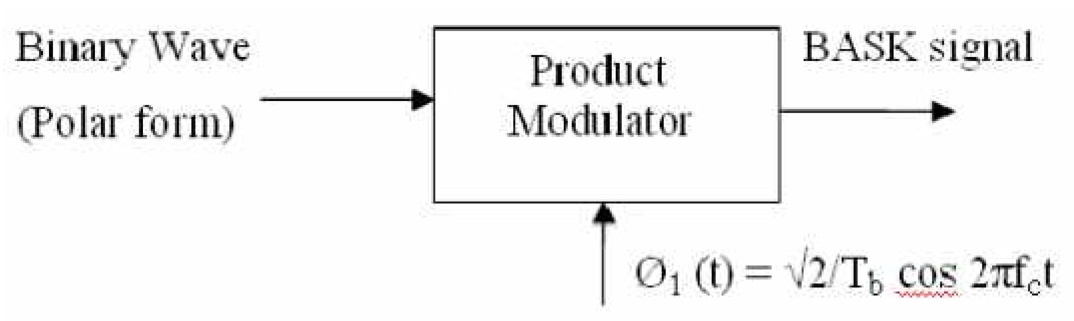
AIM:

To simulate Binary Amplitude shift keying technique using MATLAB software

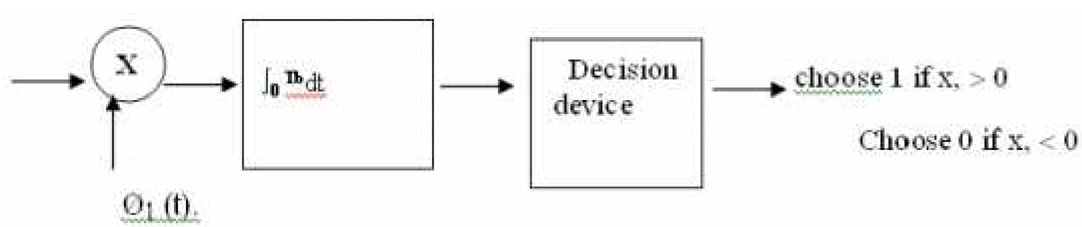
APPARATUS:

1. ASK Trainer Kit - AET-48
2. Dual Trace oscilloscope
3. Digital Multimeter
4. C.R.O(30MHz)
5. Patch chords.
6. PC with windows(95/98/XP/NT/2000)
7. MATLAB Software with communication toolbox

BLOCK DIAGRAM:



Transmitter



Receiver

THEORY:

Amplitude Shift Keying

ASK is a form of modulation that represents digital data as variations in the amplitude of a carrier wave.The amplitude of an analog carrier signal varies in accordance with the bit stream (modulating signal), keeping frequency and phase constant.
On-off keying (**OOK**) the simplest form of amplitude-shift keying (ASK) modulation that represents digital data as the presence or absence of a carrier wave. In its simplest form, the presence of a carrier for a specific duration represents a binary one, while its absence for the same duration represents a binary zero. In a ASK system, the pair of signal S1(t) used to represent binary symbols 1 & 0 are defined by

S1 (t) = $\sqrt{2Eb/\tau b}$ Cos $2\pi fct$
0 where $0 \leq t < Tb$ and

Eb = Transmitted signed energy for bit
The carrier frequency fc **=**n/Tb for some fixed integer n.

The input binary symbols are represented in polar form with symbols 1 & 0 represented by constant amplitude levels $\sqrt{Eb}$ & $-\sqrt{Eb}$. This binary wave is multiplied by a sinusoidal carrier in a product modulator. The result is a ASK signal.

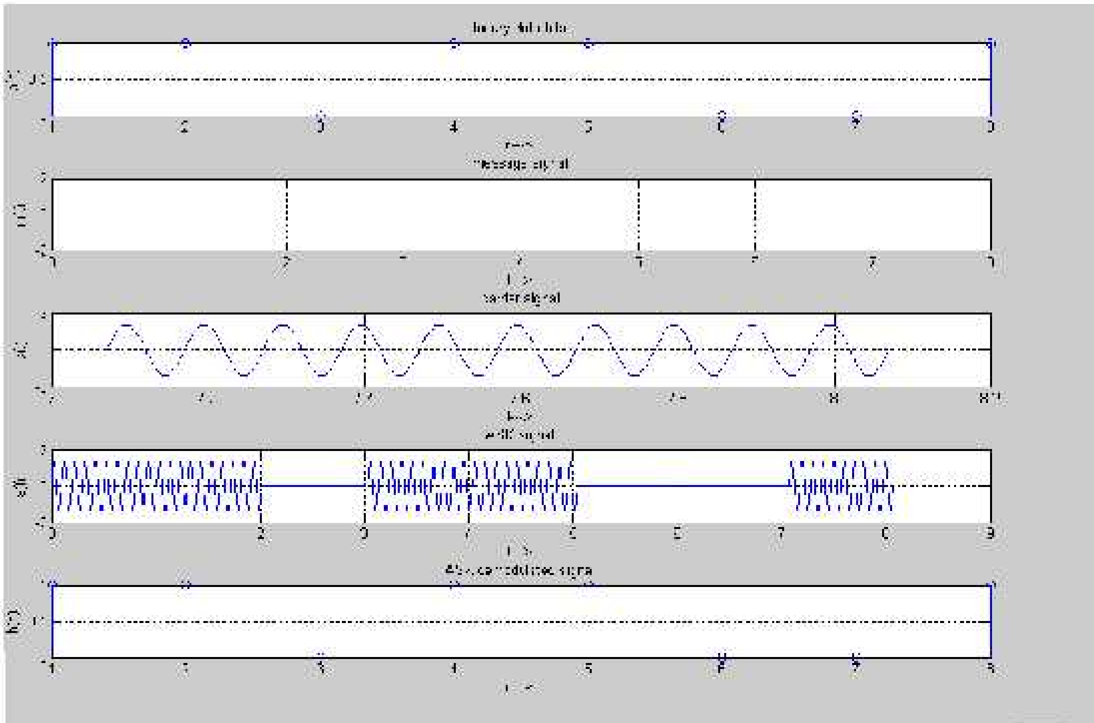
The received ASK signal is applied to a correlator which is also supplied with a locally generated reference signal Ø1 (t). The correlated o/p is compared with a threshold of zero volts. If x1> 0, the receiver decides in favour of symbol 1. If x1< 0, it decides in favour of symbol 0.

ALGORITHM:
1. Given a bandwidth of 5000 Hz for an ASK signal, what are the baud rate and bit rate?
2. Find the minimum bandwidth for an ASK signal transmitting at 2000bps.

ASK Modulation
3. Generate the carrier signal (Ø1 (t) = $\sqrt{2/Tb}$ cos $2\pi fct$)
4. Generate the base band data signal .
5. Multiply the polar form data signal and carrier signal. The resultant signal is a PSK signal.
6. Plot the carrier, data and PSK signal.

ASK Demodulation
7. Multiply the received PSK signal with the carrier signal (Ø1 (t) = $\sqrt{2/Tb}$ cos $2\pi fct$)
8. Integrate the resultant signal(x1) .
9. If x1 is greater than zero then choose 1 and if it is less than 0 choose 0.
10. Plot the demodulated signal.

EXPECTED WAVE FORMS:



MATLAB CODE:

```
% <<<<<<<<<<<<<<<<<<<< ASK Modulation and Demodulation >>>>>>>>>>>>>>>>>>>>
clc, clear all, close all;
% ******************* Digital/Binary input information *******************
x = input('Enter Digital Input Information = ');   % Binary information as stream of
bits (binary signal 0 or 1)
N = length(x);
Tb = 0.0001;   %Data rate = 1MHz i.e., bit period (second)
disp('Binary Input Information at Transmitter: ');
disp(x);
% ************* Represent input information as digital signal *************
nb = 100;   % Digital signal per bit
digit = [];
for n = 1:1:N
    if x(n) == 1;
       sig = ones(1,nb);
    else x(n) == 0;
        sig = zeros(1,nb);
    end
     digit = [digit sig];
end
t1 = Tb/nb:Tb/nb:nb*N*(Tb/nb);   % Time period
figure('Name','ASK Modulation and Demodulation','NumberTitle','off');
subplot(3,1,1);
plot(t1,digit,'LineWidth',2.5);
grid on;
axis([0 Tb*N -0.5 1.5]);
xlabel('Time(Sec)');
ylabel('Amplitude(Volts)');
title('Digital Input Signal');
% *************************** ASK Modulation *****************************
Ac1 = 12;     % Carrier amplitude for binary input '1'
Ac2 = 5;      % Carrier amplitude for binary input '0'
br = 1/Tb;    % Bit rate
Fc = br*10;   % Carrier frequency
t2 = Tb/nb:Tb/nb:Tb;   % Signal time
mod = [];
for (i = 1:1:N)
    if (x(i) == 1)
        y = Ac1*cos(2*pi*Fc*t2);   % Modulation signal with carrier signal 1
    else
        y = Ac2*cos(2*pi*Fc*t2);   % Modulation signal with carrier signal 2
    end

    mod = [mod y];
end
t3 = Tb/nb:Tb/nb:Tb*N;   % Time period
subplot(3,1,2);
plot(t3,mod);
xlabel('Time(Sec)');
ylabel('Amplitude(Volts)');
title('ASK Modulated Signal');
```

```
% ******************** Transmitted signal x *****************************
x = mod;
% ******************** Channel model h and w ****************************
h = 1;   % Signal fading
w = 0;   % Noise
% ******************** Received signal y ********************************
y = h.*x + w;   % Convolution
% ************************** ASK Demodulation ****************************
s = length(t2);
demod = [];
for n = s:s:length(y)
  t4 = Tb/nb:Tb/nb:Tb;    % Time period
  c = cos(2*pi*Fc*t4);    % Carrier signal
  mm = c.*y((n-(s-1)):n); % Convolution
  t5 = Tb/nb:Tb/nb:Tb;
  z = trapz(t5,mm);       % Intregation
  rz = round((2*z/Tb));
  Ac = ((Ac1 + Ac2)/2);   % Average of carrier amplitudes
  if(rz > Ac)             % Logical condition
    a = 1;
  else
    a = 0;
  end
  demod = [demod a];
end
disp('Demodulated Binary Information at Receiver: ');
disp(demod);
% ********** Represent demodulated information as digital signal **********
digit = [];
for n = 1:length(demod);
    if demod(n) == 1;
       sig = ones(1,nb);
    else demod(n) == 0;
        sig = zeros(1,nb);
    end

     digit = [digit sig];
end
t5 = Tb/nb:Tb/nb:nb*length(demod)*(Tb/nb);   % Time period
subplot(3,1,3)
plot(t5,digit,'LineWidth',2.5);grid on;
axis([0 Tb*length(demod) -0.5 1.5]);
xlabel('Time(Sec)');
ylabel('Amplitude(Volts)');
title('ASK Demodulated Signal');
```

RESULT:

The program for ASK modulation and demodulation has been simulated in MATLAB and necessary graphs are plotted.

POST LAB QUESTIONS:

1. Write a matlab program to generate sync function

2. Differentiate plot and subplot commands in MATLAB.

EVALUATION SHEET

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Signature of the Faculty Handling the Lab

Frequency Shift Keying (FSK) modulator and Demodulator

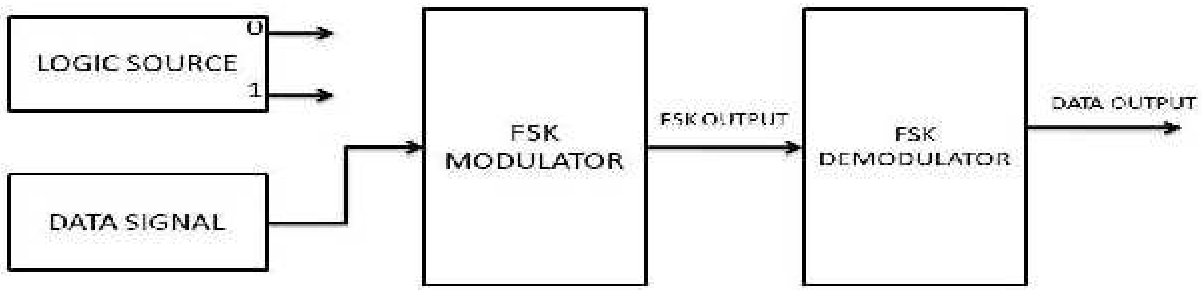
AIM:

To analyze a FSK modulation system and interpret the modulated and demodulated waveforms

APPARATUS:

1. FSK Trainer Kit - AET-48
2. Dual Trace oscilloscope
3. Digital Multimeter
4. C.R.O(30MHz)
5. Patch chords.
6. PC with windows(95/98/XP/NT/2000)
7. MATLAB Software with communication toolbox

BLOCK DIAGRAM:



THEORY:

In Frequency shift keying, the carrier frequency is shifted (i.e. from one frequency to another) corresponding to the digital modulating signal. If the higher frequency is used to represent a data ‘1’ & lower frequency a data ‘0’, the resulting FSK waveform appears.
Thus
Data =1 High FrequencyData =0 Low Frequency
It is also represented as a sum of two ASK signals. The two carriers have different frequencies & the digital data is inverted. The demodulation of FSK can be carried out by a PLL. As known, the PLL tries to ‘lock’ the input frequency. It achieves this by generating corresponding O/P voltage to be fed to the VCO, if any frequency deviation at its I/P is encountered. Thus the PLL detector follows the frequency changes and generates proportional O/P voltage. The O/P voltage from PLL contains the carrier components. Therefore to remove this, the signal is passed through Low Pass Filter. The resulting wave is too rounded to be used for digital data processing. Also, the amplitude level may be very low due to channel attenuation.

FSK Modulator
The FSK modulator using IC XR 2206. IC XR 2206 is a VCO based monolithic function generator capable of producing Sine, Square, Triangle signals with AM and FM facility. In this trainer XR2206 is used generate FSK signal. Mark (Logic 1) and space (logic 0) frequencies can be independently adjusted by the choice of timing potentiometers FO & Fl. The output is phase continuous during transitions. The keying signals i.e. data signal is applied to pin 9.

FSK Demodulator:

FSK demodulator in a combination of PLL (LM565) and comparator (Op-amp). The frequency-changing signal at the input to the PLL drives the phase detector to result in rapid change in the error voltage, which is applied to the input of the comparator. At the space frequency, the error voltage out of the phase detector is below the comparison voltage of the comparator. The comparator is a non-inverting circuit, so its output level is also low. As the phase detector input frequency shifts low (to the mark frequency), the error voltage steps to a high level, passing through the comparison level, causing the comparator output voltage to go high. This error voltage change will snap the comparator output voltage between its two output levels in manner that duplicates the data signal input to the XR22OS modulator. The free running frequency of the PLL (no input signal) is set midway between the mark and space frequencies. A space at 2025 Hz and mark at 2225 Hz will have a free running VCO frequency of 2125 Hz.

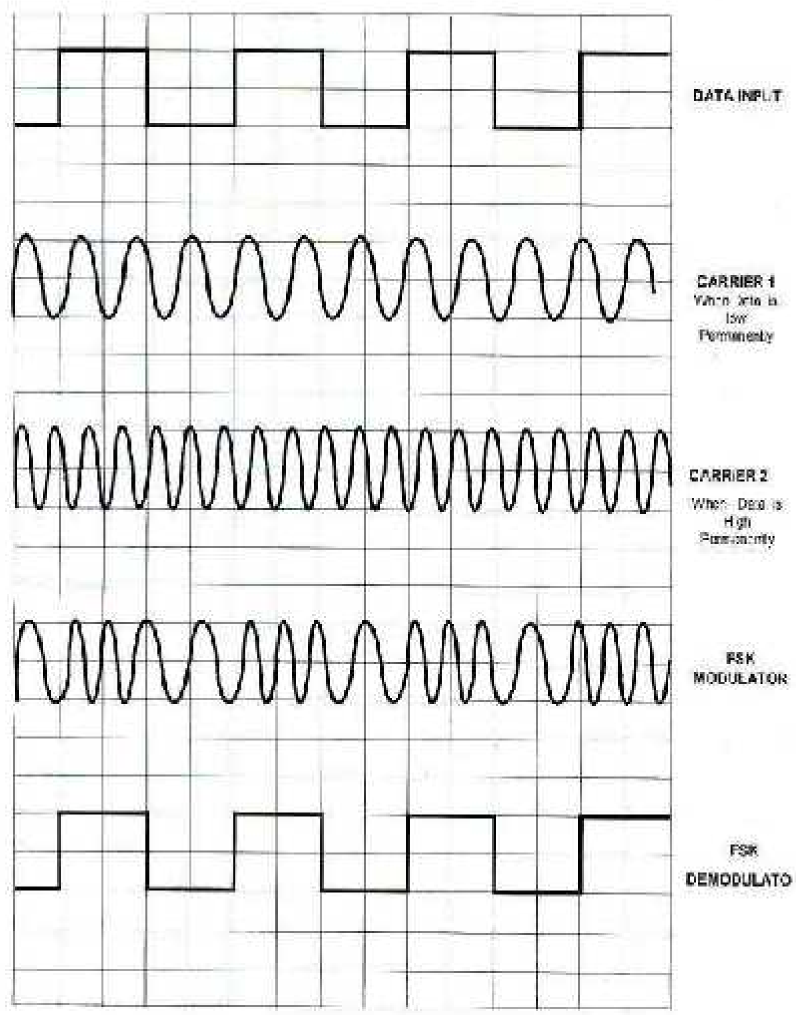
TEST PROCEDURE

1. Connect the trainer kit to the mains and switch on the power supply
2. Check internal RPS voltage (it should be 12V) and logic source voltage for logic one (it should be 12V)
3. Observe the data signal using oscilloscope. Note down the value. (Amplitude and Time Period)
4. Connect the output of the logic source to data input of the FSK modulator
5. Set the output frequency of the FSK modulator as 1.2 KHz using control F0 (this represents logic 0). Then set another frequency as 2.4 KHz using control F1 (this represents logic 1) using multimeter.
6. Connect the data input of the FSK modulator to the output of the data signal generator. Observe the signal that comes out of FSK modulator and note down the readings.
7. Connect the FSK modulator output to the input of the FSK demodulator. Observe the waveform of FSK demodulator output using CRO and note down the readings.

OBSERVATIONS:

Data source			Carrier signal		
Signal Type	Time Period	Amplitude	Signal name	Frequency	Amplitude
Square Wave			F1	2.4KHz	
			F0	1.2KHz	
Modulated Output			**Demodulated Output**		
Signal name	Frequency	Amplitude	Signal Type	Time Period	Amplitude
FSK	1.2KHz an d2.4Khz alternately appering		Square Wave		

EXPECTED WAVE FORMS:

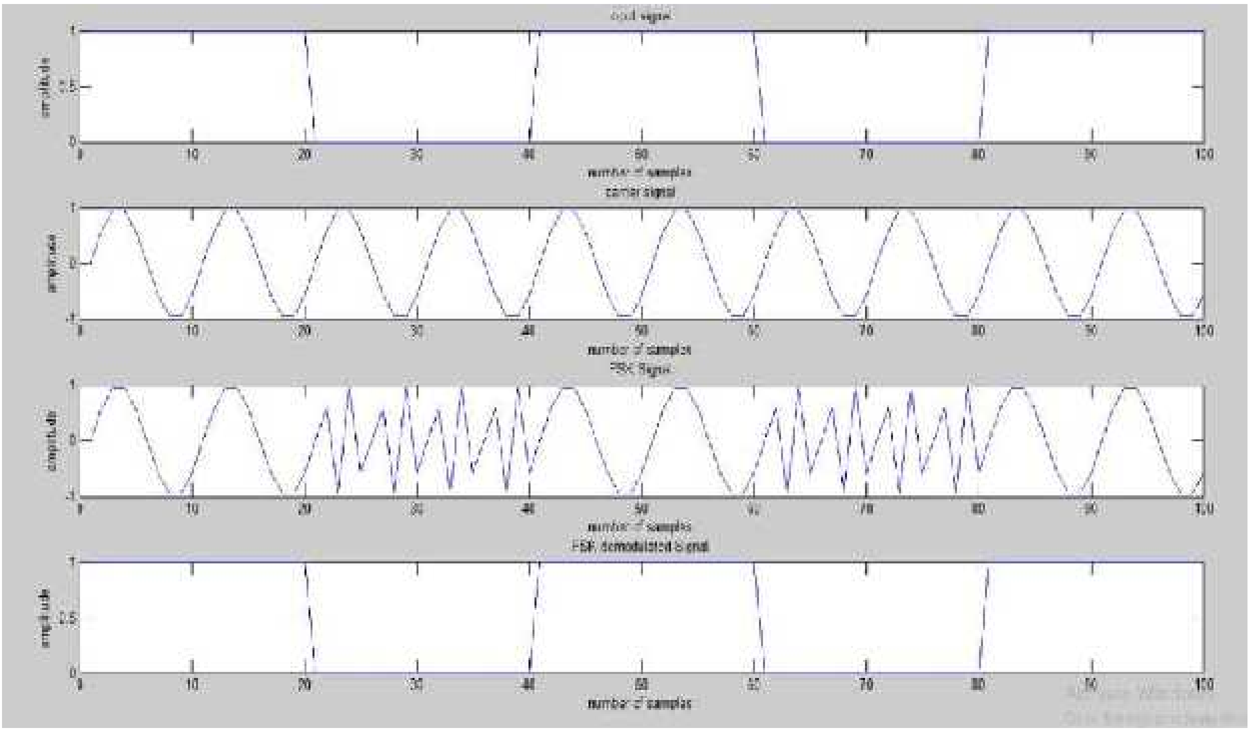


RESULT:

Thus the FSK modulation and demodulation were performed and required graphs were plotted.

MATLAB CODE:

```
clc;
clear all;
close all;
n=100;
x=[ones(1,20) zeros(1,20) ones(1,20) zeros(1,20) ones(1,20)];
subplot(4,1,1);
plot(x);
title('input signal');
xlabel('number of samples');
ylabel('amplitude');
f=1*10^6;
fs=10*10^6;
for i=0:n-1
d(i+1)=sin(2*pi*(f/fs)*i);
end
subplot(4,1,2);
plot(d);
title('carrier signal');
xlabel('number of samples');
ylabel('amplitude');
for i=0:n-1
    if(x(i+1)==1)
x(i+1)=sin((2*pi*(f/fs)*i));
    else
x(i+1)=sin((2*pi*((4*(f/fs)))*i));
    end
end
subplot(4,1,3);
plot(x);
title('FSK Signal');
xlabel('number of samples');
ylabel('amplitude');
for i=0:n-1 if(x(i+1)==sin((2*pi*(f/fs)*i))) x(i+1)=1;
else x(i+1)=0;
    end
end
subplot(4,1,4);
plot(x);
title('FSK demodulated Signal');
xlabel('number of samples');
ylabel('amplitude');
```


carrier signal
number of samples
FSK demodulated Signal

POST LAB QUESTIONS:

1. What is MSK?
2. For the given 8 bit data 10111010 draw the FSK output waveform.
3. Draw the constellation diagram of FSK.
4. What will happen if the same frequency is used for both the carriers?

EVALUATION SHEET

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BINARY PHASE SHIFT KEYING GENERATION & DETECTION

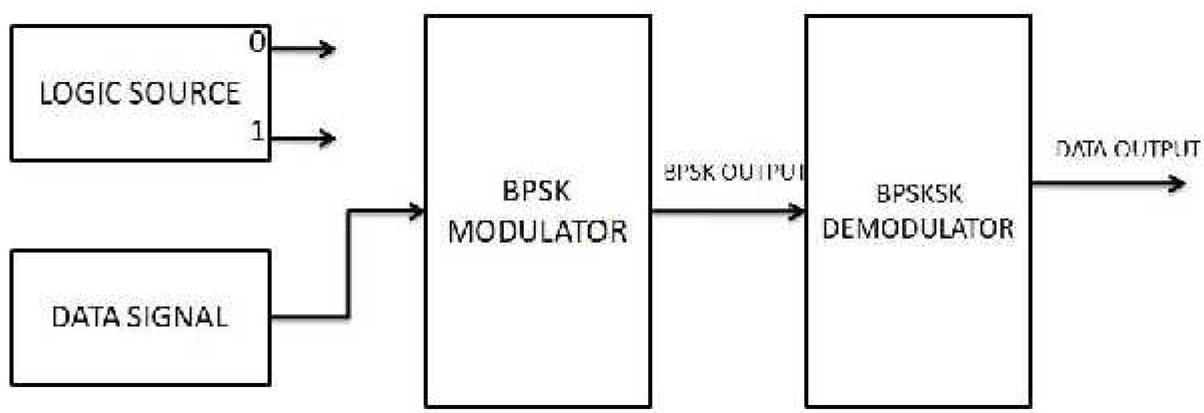
AIM:

To analyze a BPSK modulation system. And interpret the modulated and demodulated waveforms.

APPARATUS:

1. BPSK Trainer Kit - AET-48
2. Dual Trace oscilloscope
3. Digital Multimeter
4. C.R.O(30MHz)
5. Patch chords.
6. PC with windows(95/98/XP/NT/2000)
7. MATLAB Software with communication toolbox

BLOCK DIAGRAM:



INTRODUCTION

Phase shift keying is a modulation/data transmitting technique in which phase of the carrier signal is shifted between two distinct levels. In a simple PSK(ie binary PSK) unshifted carrier Vcosω0t is transmitted to indicate a 1 condition, and the carrier shifted by 1800 ie – Vcosω0t is transmitted to indicate as 0 condition.

PSK Modulator

Figure shows the PSK modulator. IC CD 4052 is a 4 channel analog multiplexer and is used as an active component in this circuit. One of the control signals of 4052 is grounded so that 4052 will act as a two channel multiplexer and other control is being connected to the binary signal ie data to be transmitted. Unshifted carrier signal is connected directly to CH1 and carrier shifted by 1800 is connected to CH2.phase shift network is a unity gain inverting amplifier using OP-amp (TL084).

When input data signal is 1 ie control signal is at high voltage, output of the 4052 is connected to CH1 and unshifted (or 0 phase)carrier is passed on to output. Similarly When data signal is 0 ie control signal is at zero voltage output of 4052 is connected to CH2 and carrier shifted by 1800 ispassed on to output.

PSK Demodulator:

Demodulation of PSK is achieved by subtracting the received carrier from a derived synchronous reference carrier of constant phase. Figure shows the simple coherent(synchronous) PSK modulator. Received PSK signal is converted to square wave using an op-amp(TL084) based zero crossing detector and connected to EX-OR circuit. The derived reference carrier is connected to other input of the EX-OR Gate through an op-amp based zero crossing detector. For the simplicity same carrier is used at receiver as reference carrier (In practical communication system reference carrier is generated at receiver).We can observe the exact operation of demodulator with the help of waveforms at various nodes in the circuit. Received PSK signal is converted to square wave using an op-amp(TL084) based zero crossing detector and connected to EX-OR circuit. The derived reference carrier is connected to other input of the EX-OR Gatethrough an op-amp based zero crossing detector. For the simplicity same carrier is used at receiver as reference carrier (In practical communication system reference carrier is generated at receiver).We can observe the exact operation of demodulator with the help of waveforms at various nodes in the circuit.

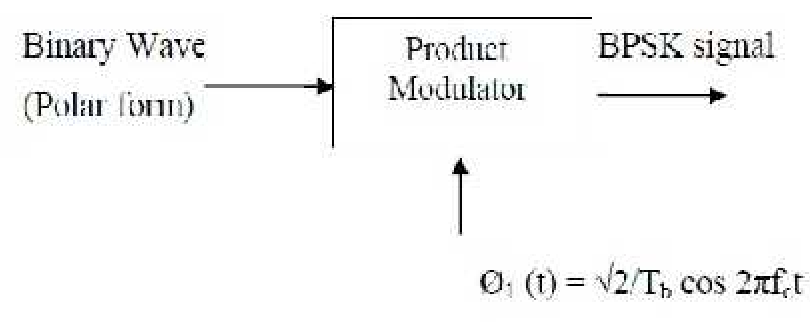
THEORY:

Binary Phase Shift Keying

In a coherent binary PSK system, the pair of signal S1(t) and S2 (t) used to represent binarysymbols 1 & 0 are defined by
S1 (t) = √2Eb/τb Cos 2πfct
S2 (t) =√2Eb/Tb (2πfct+π) = - √ 2Eb/Tb Cos 2πfct where 0 ≤ t< Tb and
Eb = Transmitted signed energy for bit
The carrier frequency fc **=**n/Tb for some fixed
integer n. In BPSK, there is only one basis function
of unit energy.Øb (t) = √2/Tb cos 2fπct 0≤ t< Tb
S1 (t) = √Eb Ø1 (t) 0≤ t
≤Tb S2 (t) = √Eb Ø1 (t) 0≤
t< Tb
The signal space is 1dimensional (N=1) having two message points (M = 2)

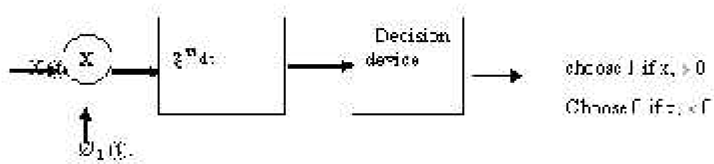
Block Diagram of BPSK Transmitter

The input binary symbols are represented in polar form with symbols 1 & 0 represented by constant amplitude levels √Eb & -√Eb. This binary wave is multiplied by a sinusoidal carrier in aproduct modulator. This results in a BSPK signal.



BSPK Receiver:

The received BPSK signal is applied to a correlator which is also supplied with a locally generated reference signal Ø1 (t). The correlated o/p is compared with a threshold of zero volts. If x1> 0, the receiver decides in favour of symbol 1. If x1< 0, it decides in favour of symbol 0



The received BPSK signal is applied to a correlator which is also supplied with a locally generated reference signal Ø1 (t). The correlated o/p is compared with a threshold of zero volts. If x1> 0, the receiver decides in favour of symbol 1. If x1< 0, it decides in favour of symbol 0

PROCEDURE

1. Connect the trainer to mains and switch on the power supply.
2. Measure the output of the regulated power supply ie +5V and -5V with the help of digital multimeter.
3. Observe the output of the carrier generator using CRO, it should be an 8KHZ sine with 5Vppamplitude.
4. Observe the various data signals(1KHZ,2KHZ and 4KHZ) using CRO

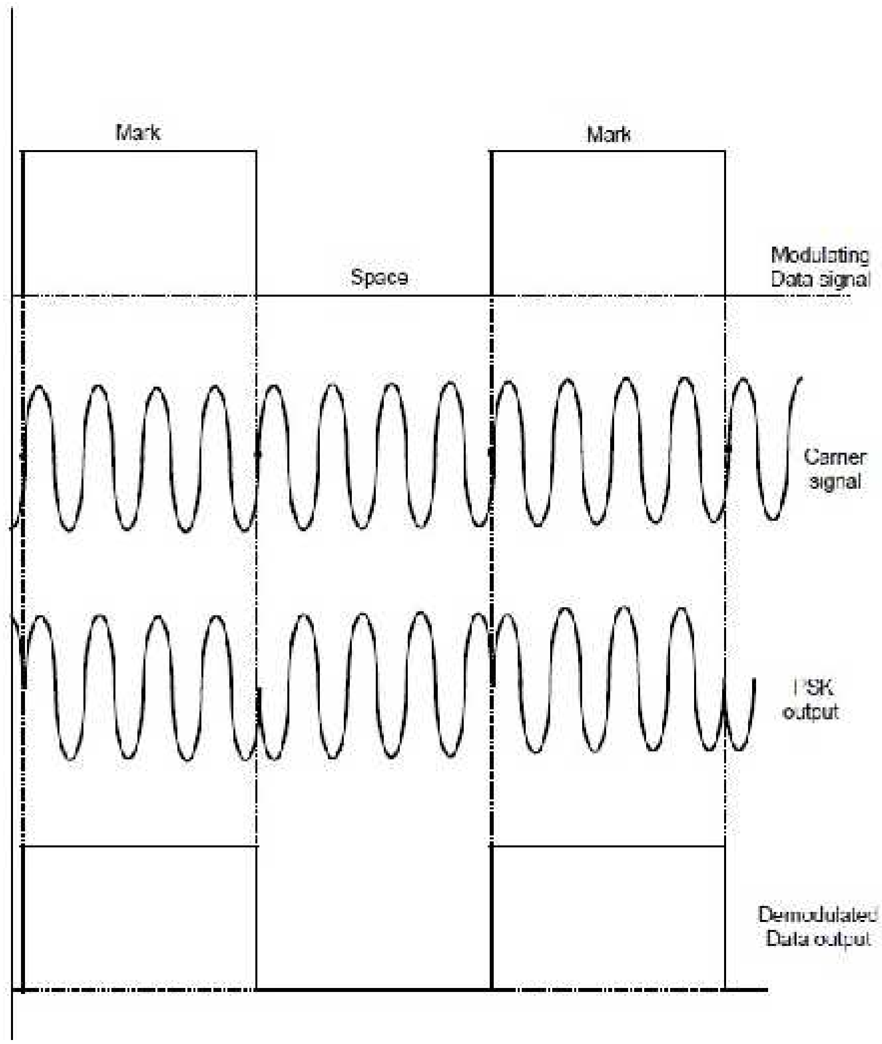
Modulation:

5. Connect carrier signal to carrier input of the PSK modulator.
6. Connect data signal say 4KHZ from data source to data input of the modulator.
7. Keep CRO in dual mode and connect CH1 input of the CRO to data signal and CH2 to theoutput of the PSK modulator.
8. Observe the PSK output signal with respect to data signal and plot the waveforms.

Demodulation:

9. Connect the PSK output to the PSK input of the demodulator.
10. Connect carrier to the carrier input of the PSK demodulator.
11. Keep CRO in dual mode and connect CH1 to data signal(at modulator) and CH2 to theoutput of the demodulator.
12. Compare the demodulated signal with the original signal. By this we can notice that there isno loss in modulation and demodulation process
13. Repeat the steps 6 to 12 with different data signals ie 2KHZ and 1KHZ

EXPECTED GRAPHS:



OBSERVATIONS: **BPSK MODULATION**

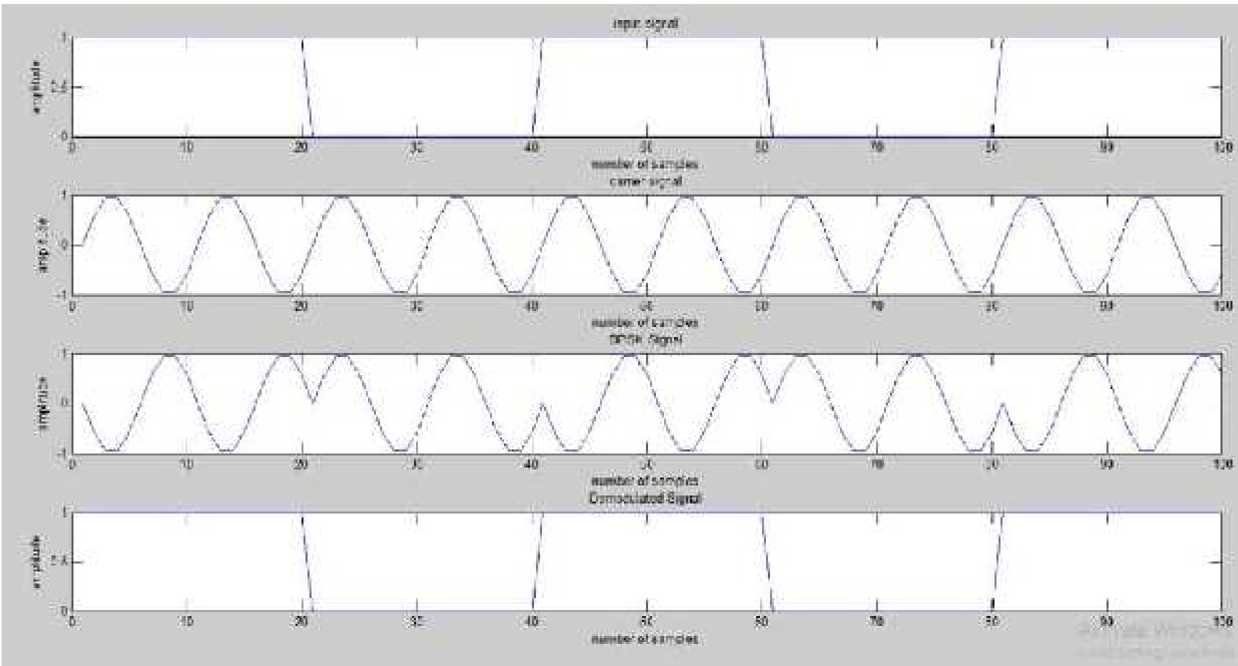
Carrier signal	**Amplitude**	**Time period**
Data Source For 4KHz For 2KHz		
For 1KHz		
Modulated Output For 4KHz For 2KHz For 1KHz		

BPSK DEMODULATION

	Amplitude	**Time period**
Demodulated Output For 4KHz For 2KHz For 1KHz		

MATLAB CODE:

```
clc;
clear all;
close all;
n=100;
x=[ones(1,20) zeros(1,20) ones(1,20) zeros(1,20) ones(1,20)];
subplot(4,1,1);
plot(x);
title('input signal');
xlabel('number of samples');
ylabel('amplitude');
f=1*10^6;
fs=10*10^6;
for i=0:n-1
d(i+1)=sin(2*pi*(f/fs)*i);
end
subplot(4,1,2);
plot(d);
title('carrier signal');
xlabel('number of samples');
ylabel('amplitude');
for i=0:n-1 if(x(i+1)==0)
x(i+1)=sin(2*pi*(f/fs)*i);
    else x(i+1)=sin(2*pi*(f/fs)*i+pi);
    end
end
subplot(4,1,3);
plot(x);
title('BPSK Signal');
xlabel('number of samples');
ylabel('amplitude');
for i=0:n-1 if(x(i+1)==sin(2*pi*(f/fs)*i)) x(i+1)=0;
else x(i+1)=1;
    end
end
subplot(4,1,4);
plot(x);
title('Demodulated Signal');
xlabel('number of samples');
ylabel('amplitude');
```


input signal
amplitude
number of samples
carrier signal
amplitude
number of samples
BPSK Signal
amplitude
number of samples
Demodulated Signal
amplitude
number of samples

RESULT:

Thus the PSK modulation and demodulation were performed and graphs were plotted.

POST LAB QUESTIONS:

1. Compare FSK and PSK.

2. List the Characteristics of TL084 op-amp.

3. Compare TL084 op amp with IC 741 op amp.

4. What do we infer from constellation diagrams of various modulation schemes?

EVALUATION SHEET

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Signature of the Faculty Handling the Lab

QPSK GENERATION & DETECTION

AIM:

To study modulation and demodulation of QPSK and sketch the relevant waveforms.

APPARATUS:

1. QPSK Trainer Kit
2. Dual Trace oscilloscope
3. Digital Multimeter
4. C.R.O(30MHz)
5. Patch chords.
6. PC with windows(95/98/XP/NT/2000)
7. MATLAB Software with communication toolbox

BLOCK DIAGRAM: QPSK MODULATOR & DEMODULATOR

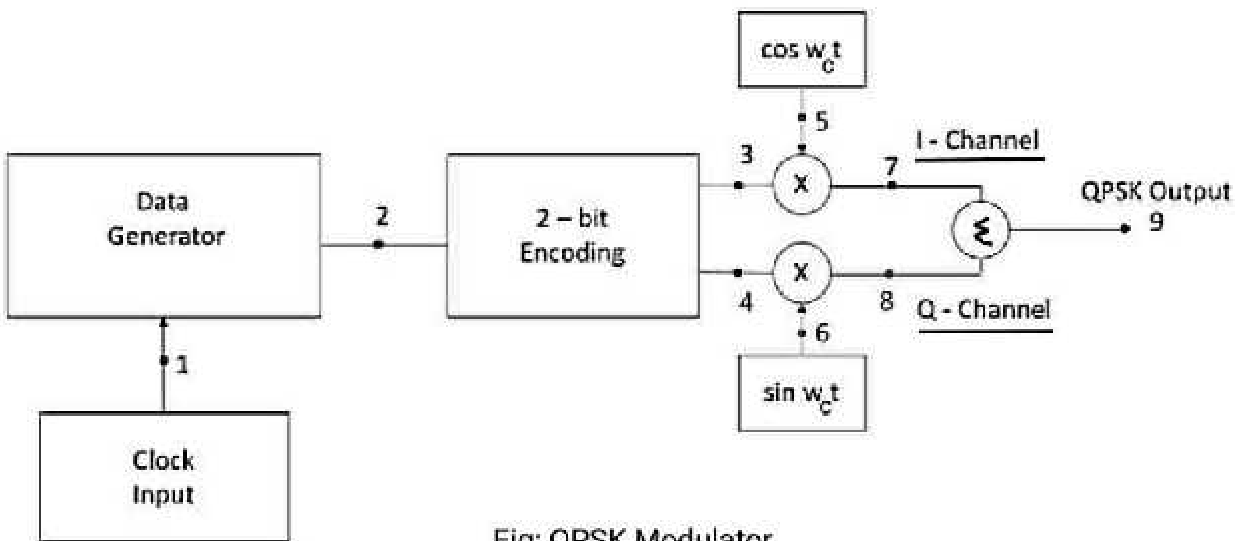


Fig: QPSK Modulator

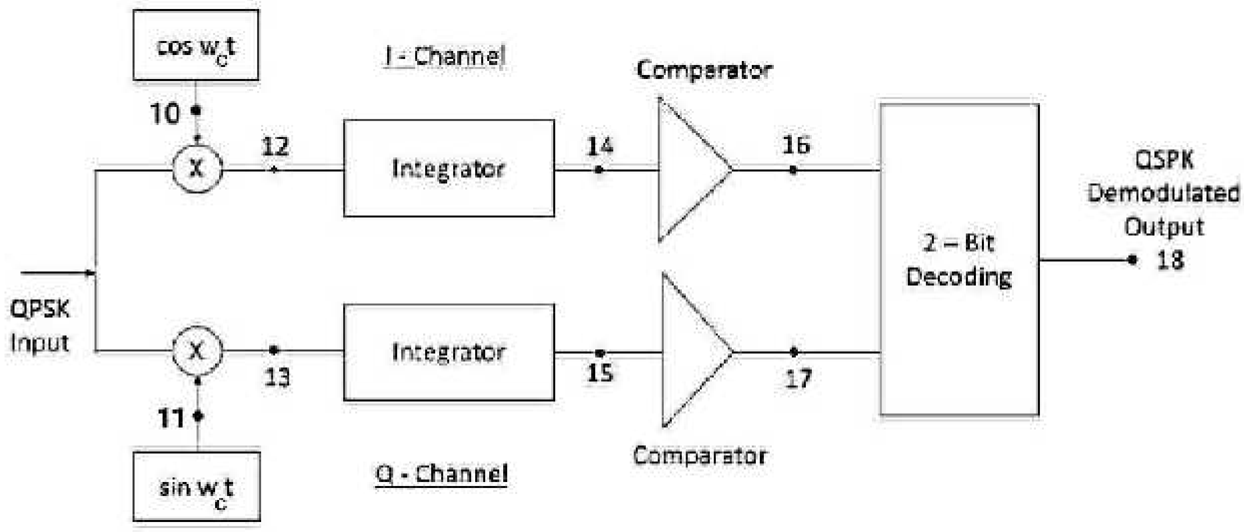


Fig: QPSK Demodulator

THEORY:

Quadrature Phase Shift Keying

Phase of the carrier takes on one of four equally spaced values such as π/4, 3π/4, 5π/4,7π/4. Si(t) = √2E/ Tb cos {2 π*f*ct + (2i – 1) π/4} , 0≤ t ≤ Tb

0, ,elsewhere

Where i = 1,2,3,4, & E= Tx signal energy per symbolTb = symbol duration

Each of the possible value of phase corresponds to a pair of bits called dibits.Thus the gray encoded set of dibits: 10,00,01,11

Si (t) = √2E/ Tb cos [(2i – 1)π/4] cos (2πfct) - √2E/ Tb sin [(2i –1) π/4)] sin (2πfct) ,0≤ t ≤Tb 0 , else where

There are two orthononormal basis functions

Ø1 (t) = √2/ Tb cos 2π*f*ct,

0≤ t ≤Tb Ø2 (t) = √2/ Tb sin

2π*f*ct, 0≤ t ≤Tb

The i/p binary sequence b(t) is represented in polar from with symbols 1 & 0 represented as +√E/2 and -√E/2 .This binary wave is demultiplexed into two separate binary waves consisting of odd & even numbered I/P bits denoted by b1 (t) & b2 (t) b1 (t) & b2(t) are used to modulate a pair of quadrature carrier or orthogonal Basis function Ø1 (t) & Ø2 (t). The result is two PSK waves' .These two binary PSK waves are added to produce the desired QPSK signal .

PROCEDURE:

1. Connect and switch on the power supply.
2. QPSK is selected by default and LEDs of corresponding technique will glow.
3. Select the bit pattern using push button i.e. 8 bit or 16 bit or 32 bit or 64 bit. Observe bitpattern on TP-2.
4. Select data rate using push button i.e. 2 KHz or 4 KHz or 8 KHz 16 KHz.

Modulation:

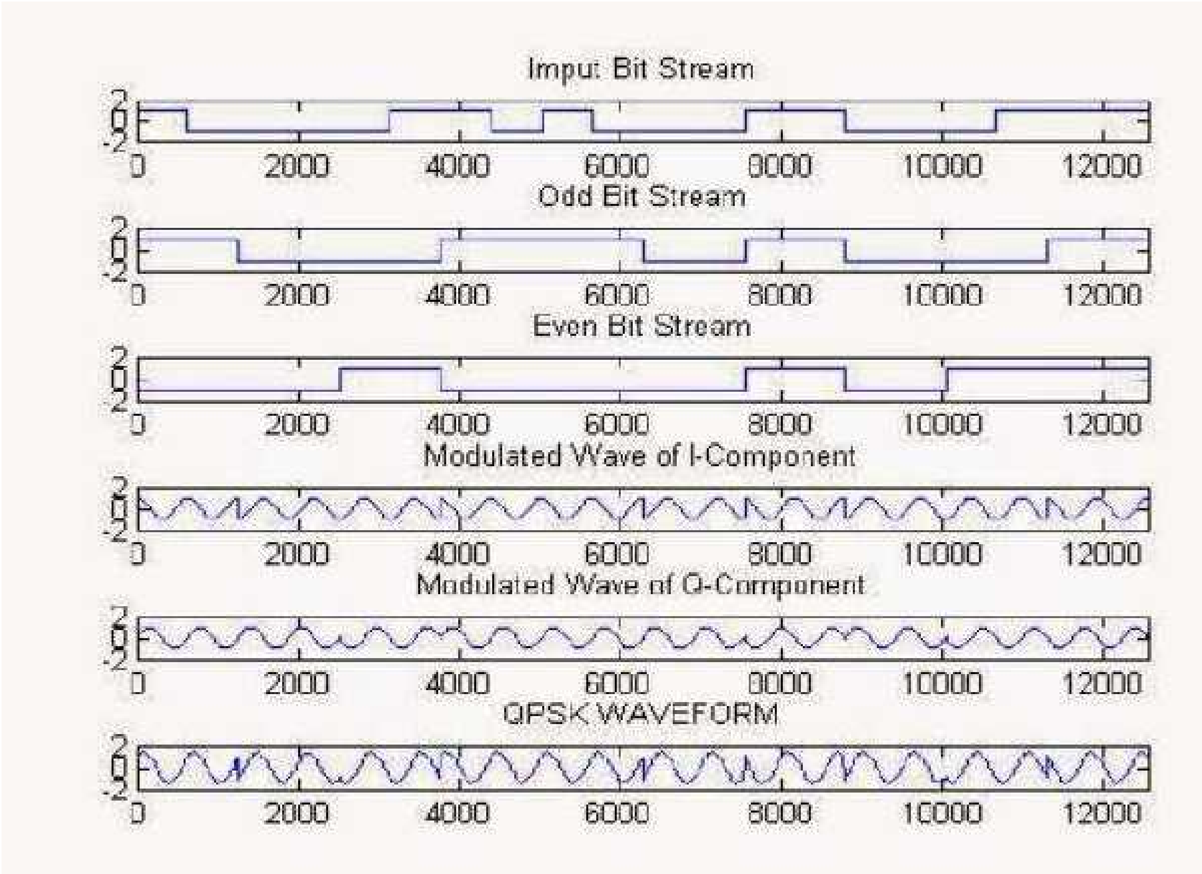
5. Observe the input bit pattern at TP-2 by varying bit pattern using respective push button.
6. Observe the data rate at TP-1 by varying data rate using respective push button.
7. Observe the Two- bit encoding i.e. I-Channel (TP-3) and Q-Channel (TP-4).
8. Observe carrier signal i.e. cosine wave (TP-5) and sine wave (TP-6). Frequency of carriersignal will change with respect to data rate.
9. Observe I-Channel (TP-7) and Q-Channel (TP-8) modulated signal.
10. Observe QPSK modulated signal at TP-9.

Demodulation:

11. Apply the QPSK modulated output to the demodulator input.
12. Observe the multiplied signal of QPSK and carrier signal, cosine at TP-12 and also observethe multiplied signal of QPSK and carrier signal, sine at TP-13.
13. Observe the integrated output at I-channel (TP-14) and Q-channel (TP-15).

Input Bits	**Phase of QPSK signal**	**Co –ordinates of message signal**	
		S1	**S2**
10	**π/4**	$\sqrt{E/2}$	$-\sqrt{E/2}$
00	**3π/4**	$-\sqrt{E/2}$	$-\sqrt{E/2}$
01	**5π/4**	$-\sqrt{E/2}$	$+\sqrt{E/2}$
11	**7π/4**	$+\sqrt{E/2}$	$+\sqrt{E/2}$

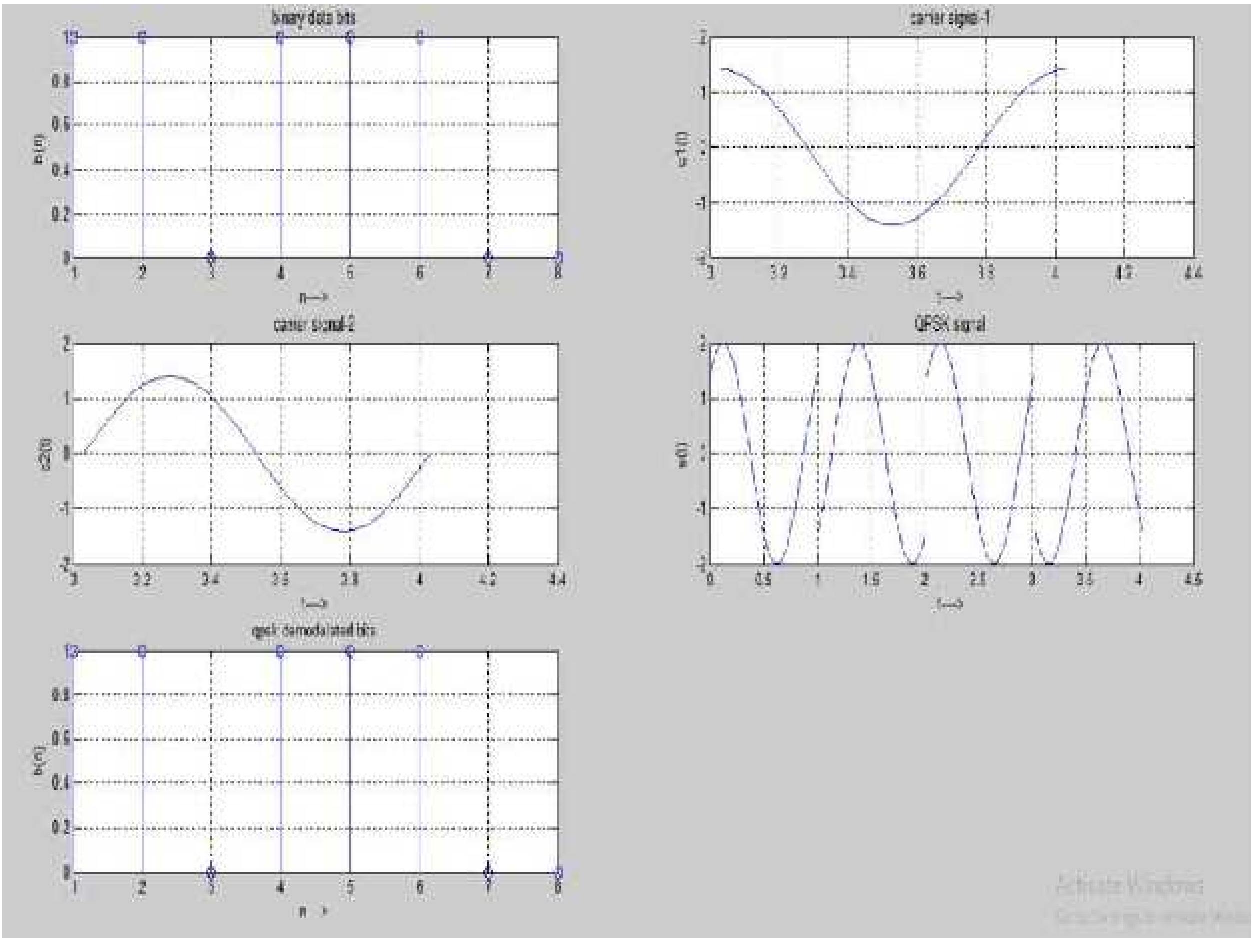
EXPECTED WAVE FORMS:



MATLAB CODE:

```
clc;
clear all;
close all;
data=[0  1 0 1 1 1 0 0 1 1]; % information
%Number_of_bit=1024;
%data=randint(Number_of_bit,1);
figure(1)
stem(data, 'linewidth',3), grid on;
title('  Information before Transmiting ');
axis([ 0 11 0 1.5]);
data_NZR=2*data-1; % Data Represented at NZR form for QPSK modulation
s_p_data=reshape(data_NZR,2,length(data)/2);  % S/P convertion of data
br=10.^6; %Let us transmission bit rate  1000000
f=br; % minimum carrier frequency
T=1/br; % bit duration
t=T/99:T/99:T; % Time vector for one bit information
% XXXXXXXXXXXXXXXXXXXXXXX QPSK modulation  XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
y=[];
y_in=[];
y_qd=[];
for(i=1:length(data)/2)
    y1=s_p_data(1,i)*cos(2*pi*f*t); % inphase component
    y2=s_p_data(2,i)*sin(2*pi*f*t) ;% Quadrature component
    y_in=[y_in y1]; % inphase signal vector
    y_qd=[y_qd y2]; %quadrature signal vector
    y=[y y1+y2]; % modulated signal vector
end
Tx_sig=y; % transmitting signal after modulation
tt=T/99:T/99:(T*length(data))/2;
figure(2)
subplot(3,1,1);
plot(tt,y_in,'linewidth',3), grid on;
title(' wave form for inphase component in QPSK modulation ');
xlabel('time(sec)');
ylabel(' amplitude(volt0');
subplot(3,1,2);
plot(tt,y_qd,'linewidth',3), grid on;
title(' wave form for Quadrature component in QPSK modulation ');
xlabel('time(sec)');
ylabel(' amplitude(volt0');
subplot(3,1,3);
plot(tt,Tx_sig,'r','linewidth',3), grid on;
title('QPSK modulated signal (sum of inphase and Quadrature phase signal)');
xlabel('time(sec)');
ylabel(' amplitude(volt0');
% XXXXXXXXXXXXXXXXXXXXXXXXXXXX QPSK demodulation XXXXXXXXXXXXXXXXXXXXXXXXXX
Rx_data=[];
Rx_sig=Tx_sig; % Received signal
for(i=1:1:length(data)/2)
    %%XXXXXX inphase coherent dector XXXXXXX
    Z_in=Rx_sig((i-1)*length(t)+1:i*length(t)).*cos(2*pi*f*t);
    % above line indicat multiplication of received & inphase carred signal

    Z_in_intg=(trapz(t,Z_in))*(2/T);% integration using trapizodial rull
    if(Z_in_intg>0) % Decession Maker
        Rx_in_data=1;
    else
```

```
        Rx_in_data=0;
    end



    %%XXXXXX Quadrature coherent dector XXXXXX
    Z_qd=Rx_sig((i-1)*length(t)+1:i*length(t)).*sin(2*pi*f*t);
    %above line indicat multiplication ofreceived & Quadphase carred signal

    Z_qd_intg=(trapz(t,Z_qd))*(2/T);%integration using trapizodial rull
        if (Z_qd_intg>0)% Decession Maker
        Rx_qd_data=1;
        else
       Rx_qd_data=0;
        end



        Rx_data=[Rx_data  Rx_in_data  Rx_qd_data]; % Received Data vector
end
figure(3)
stem(Rx_data,'linewidth',3)
title('Information after Receiveing ');
axis([ 0 11 0 1.5]), grid on;
```



binary data bits
carrier signal-1
carrier signal-2
QPSK signal
qpsk demodulated bits

RESULT:

Thus the QPSK modulation and demodulation wave forms are observed.

POST LAB QUESTIONS:

1. Write a matlab program to sample a message signal m(t) and reconstruct it.
2. Identify the error in the mat lab command Sin 3.
3. Draw the constellation diagram of QPSK.
4. Give some applications of QPSK modulation scheme
5. Find the output of the following command.5^ (2/3) – 25/(2*3)
6. What is the relationship between 4 QAM and QPSK?
7. Design a SIMULINK model for QPSK.

EVALUATION SHEET

Department of Electronics and Communication Engineering

SCSVMV UNIVERSITY, Enathur, Kanchipuram

Title of the Experiment	
Name of the Candidate	
Register Number	
Date of Submission	

S.No	Marks Split up	Maximum Marks	Marks Earned
1	Attendance	5	
2	Pre lab viva questions	5	
3	Execution of Experiments	20	
4	Calculation/Evaluation of Result	10	
5	Post lab viva questions	10	
6	Grand Total	50	

Signature of the Faculty Handling the Lab