G H Patel College of Engineering & Technology

Department of Electronics & Communication Engineering

LABORATORY MANUAL

Subject Code: 102060521

Subject Name: Principles of Electronics Communication

Bachelor of Engineering in Electronics & Communication

CVM University

Programme: Bachelor of Engineering in Electronics & Communication Engineering

Department of Electronics & Communication Engineering

Vision

Department of Electronics & Communication Engineering is committed to procreate the universally competing engineers, capable of undertaking and conquering professional challenges by nurturing them with analytical and practical knowledge.

Mission

- To create excellent teaching learning environment.
- To produce globally competent engineers by enhancing professional skills.
- To nurture engineers with self-confidence and self-reliance for handling the challenges and demands of industry in meaningful and sustainable manner.
- To generate outstanding research environment beneficial to industry and society.

Programme Educational Objectives (PEO)

The objectives of the Electronics and Communication Engineering department at **G H Patel College of Engineering & Technology** is to produce graduates with skills and abilities

1) To understand and apply the fundamental knowledge of core electronics and mathematical approaches for developing industrial applications.
2) To identify, formulate and solve problems related to electronics and communication engineering.
3) To excel in professional career with challenging attitude and effective communication skill.

Programme Outcomes (PO)

Graduates of the programme will be prepared …

Graduates of the Programme will be prepared for:

1. **Engineering knowledge**: Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.
2. **Problem analysis**: Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.
3. **Design/development of solutions**: Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.
4. **Conduct investigations of complex problems**: Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.
5. **Modern tool usage**: Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modeling to complex engineering activities with an understanding of the limitations.
6. **The engineer and society**: Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
7. **Environment and sustainability**: Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.
8. **Ethics**: Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
9. **Individual and teamwork**: Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.
10. **Communication**: Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.
11. **Project management and finance**: Demonstrate knowledge and understanding of the engineering and management principles and apply these to one’s own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.
12. **Life-long learning**: Recognize the need for and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

List of Experiments

No.	Name of the Experiment
1.	To Study Amplitude Modulation and Demodulation using Kit and measure Modulation Index. Simulate Amplitude Modulation in MATLAB and plot the AM wave and its frequency spectrum.
2.	To Study Frequency Modulation and Demodulation using Kit and MATLAB simulation.
3.	To Study Sampling and Reconstruction of signal using kit (ST-2102), and MATLAB simulation for sampling theorem.
4	To Study Pulse Code Modulation (PCM) using Time Division Multiplexing (TDM) of the Signals.
5.	To Study Pulse Modulation (PAM, PWM, PPM) using kit (ST-2110) and Pulse Code Modulation (PCM) using (ST- 2103) kit.
6.	To Study ASK, FSK and BPSK Modulation Schemes using MATLAB.
7.	To Study DTMF Telephone Trainer Kit (2654).
8.	Introduction to basic IP commands. To implement Network Topology on Cisco Packet Tracer.
9.	To study Fiber Optic Link: (a) To Setting up of Fiber optic Analog link using ST-2501. To study setting up of Fiber optic Digital link ST-2501.
10.	Study experimental set up for establishment of Analog satellite Link - with and without an Emulator
11.	To verify AT commands and introduction to GSM trainer kit ST-2133
12.	Open Ended Problem:

Experiment : 1

Aim : To Study Amplitude Modulation and Demodulation using Kit and measure Modulation Index. Simulate Amplitude Modulation in MATLAB and plot the AM wave and its frequency spectrum.

Apparatus : ST-2201 & 2202 kit, MATLAB/LABVIEW software

Theory :

Amplitude modulation is a type of modulation where the amplitude of the carrier signal is varied in accordance with the information bearing signal.

Amplitude modulation requires a high frequency constant carrier and a low frequency modulation signal.

A sine wave carrier is of the form $\mathbf{e_c = E_c \sin (w_c t)}$

A sine wave modulation signal is of the form $\mathbf{e_m = E_m \sin (w_m t)}$

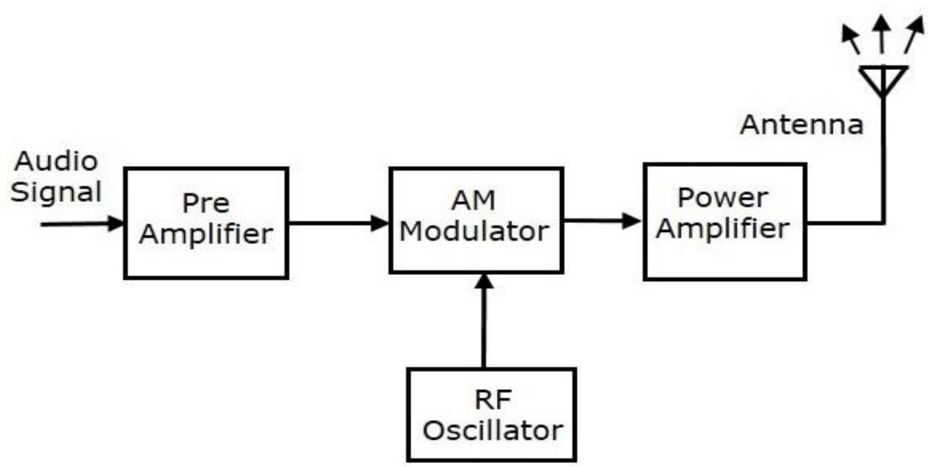
Notice that the amplitude of the high frequency carrier takes on the shape of the lower frequency modulation signal, forming what is called a modulation envelope.

The modulation index is defined as the ratio of the modulation signal amplitude to the carrier amplitude Where **m** is known as modulation index.

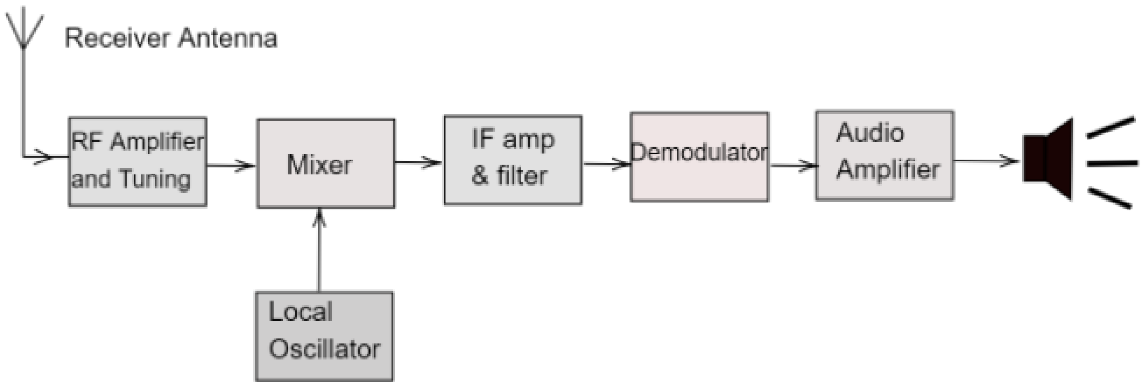
$$\therefore\ m = \frac{E_m}{E_c}$$

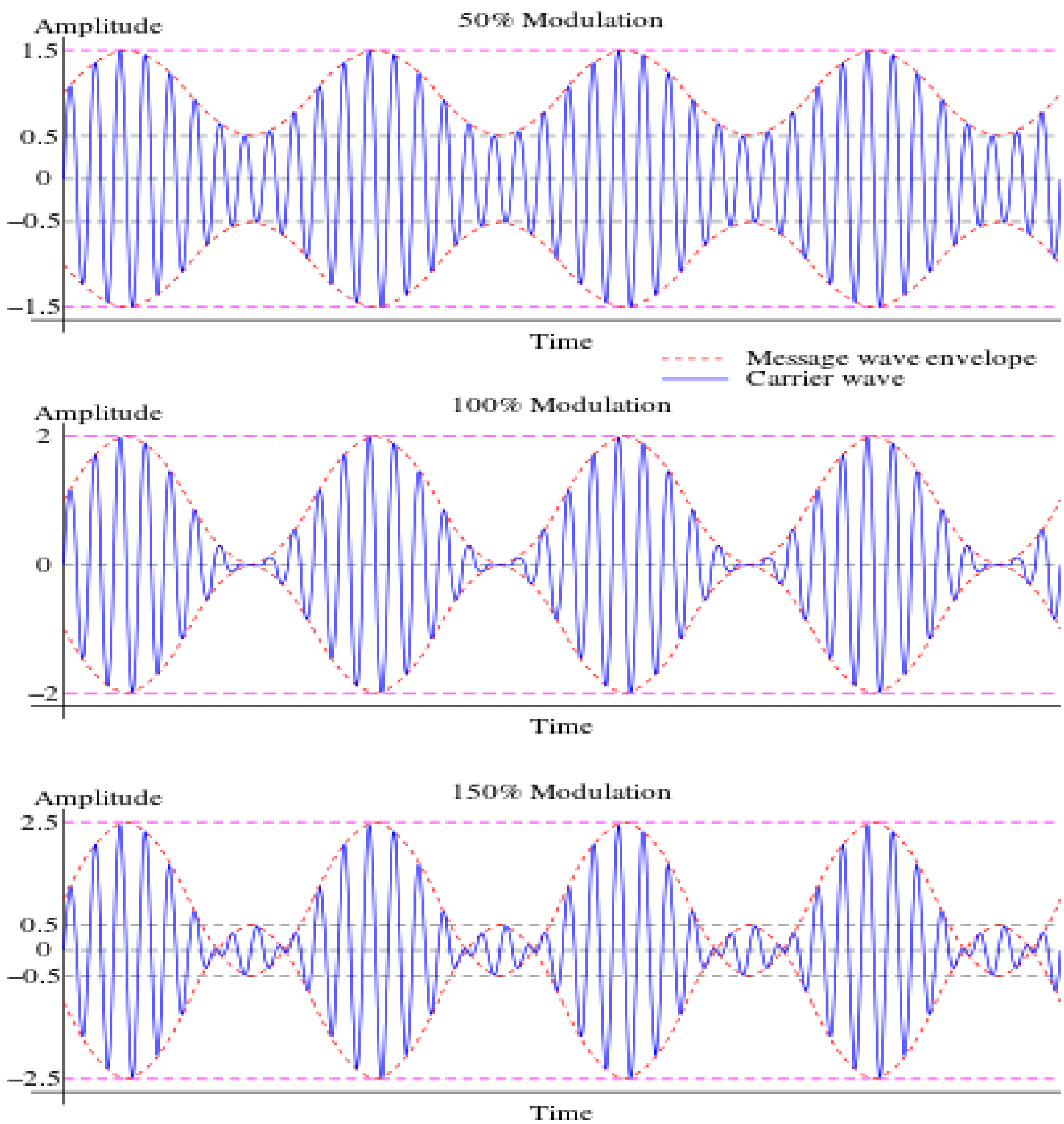
The overall signal can be described by : $\mathbf{Y_{AM} = [\, E_c + E_m \sin(w_m t)\,] \sin (w_c t)}$

AM Transmitter :



AM Receiver :





Procedure :

Task 1: Observation and analysis using ST-2201 & ST 2202 kit.

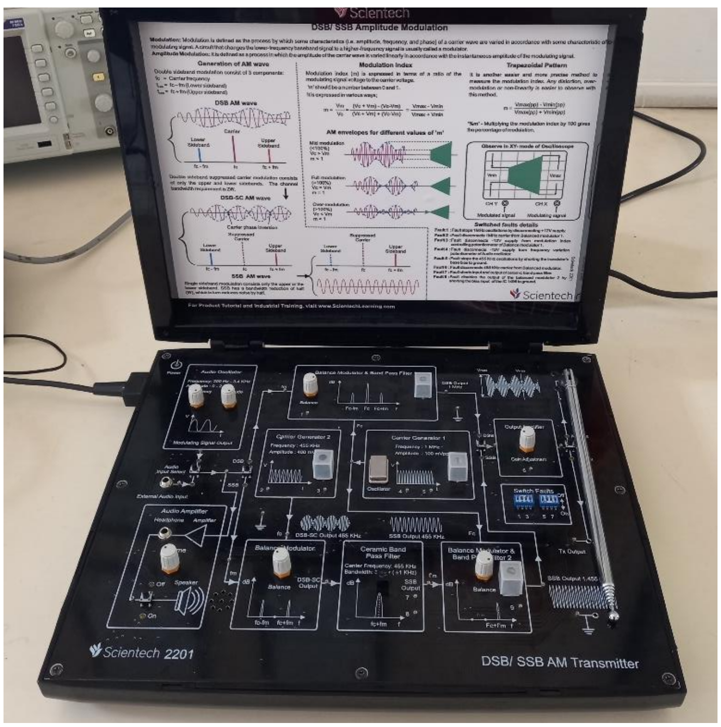
1) Connect the Modulating signal and carrier signal and observe the modulated output on MSO

2) Find the Modulation Index, and observe the vary the Amplitude of Modulating signal and observe the effect on Modulating Index.
3) Observe the Frequency Spectrum of AM signal and locate the carrier signal and sidebands.
4) Connect ST-2201 kit (AM Modulation Kit) output with ST- 2202(AM demodulation Kit), learn function of each components of Super Heterodyne receiver (ST-2202), Observe the recovered base band signal at the output.
5) Repeat the process of SSB-SC signal in the kit. Also repeat the whole process of Audio signal instead of sinusoidal Signal.

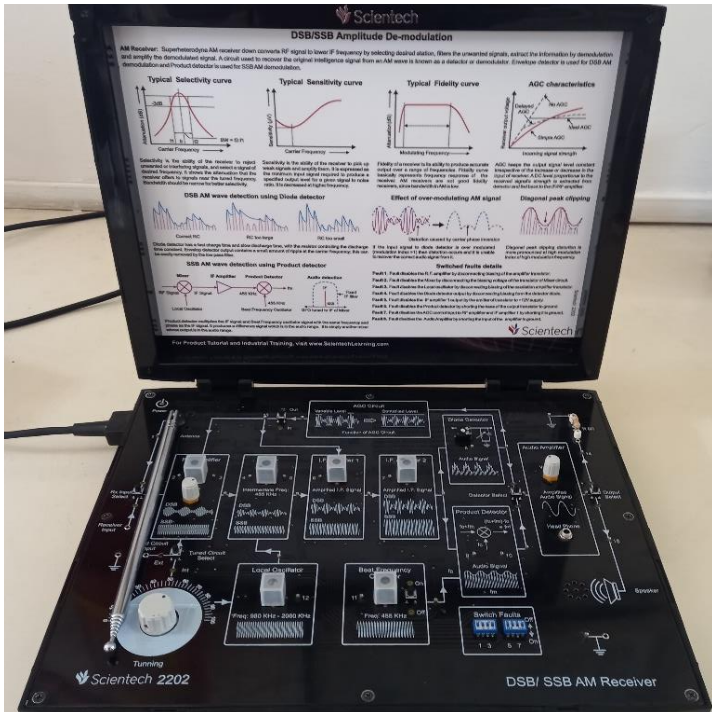
Task 2: Perform Experiment using MATLAB/LABVIEW

1) Generate the AM signal with carrier using based band sinusoidal signal and carrier signal.
2) Observe the frequency spectrum of AM signal.

ST-2201 kit for Amplitude Modulation :



ST-2202 kit for Amplitude Demodulation



Task-2: Write a Matlab Code to implement Amplitude modulation for different modulation index and plot all the waveforms.

Output:

Lab Tutorials :

1) What is the Modulation Index? How to find AM by observing signal on MSO without knowing actual value of Amplitude of base-band and carrier signal ?
2) How to define Under modulation and over modulation based on the value of modulation index (m) ?
3) How to find the sideband if carrier signal and modulating signal frequency is known?
4) State the difference between Coherence and Non- coherence demodulation?
5) What is the role of RF amplifier, IF amplifier and mixer in super heterodyne receiver?
6) How Local oscillator works in tracking with RF amplifier, Explain in detail.

Conclusion :

Experiment: 2

Aim : To Study Frequency Modulation and Demodulation using Kit and MATLAB simulation.

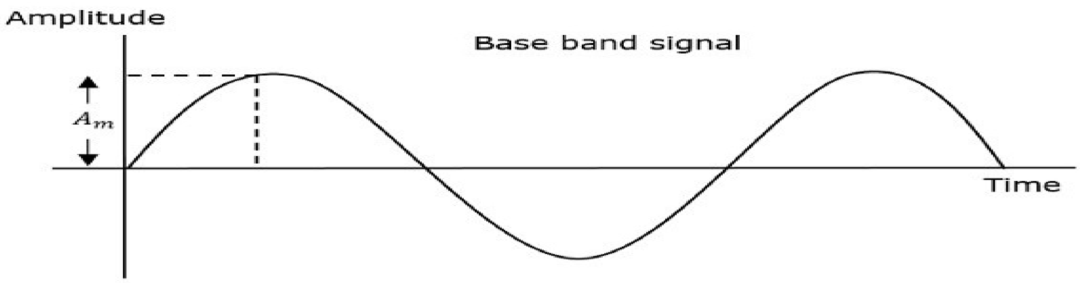
Apparatus : ST-2203 kit, MATLAB/LABVIEW software

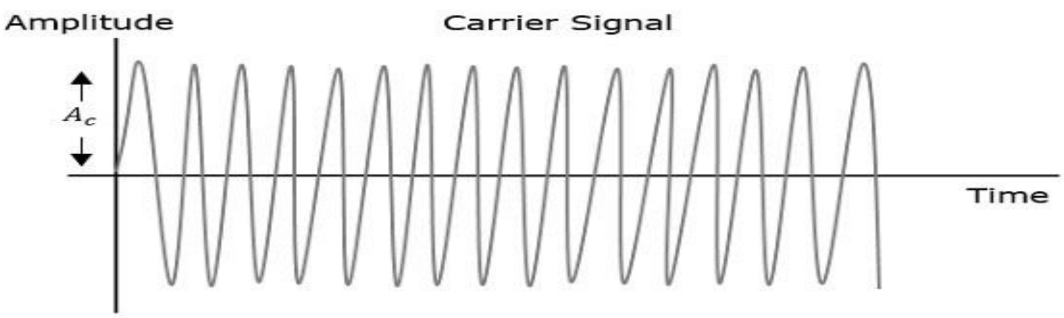
Theory :

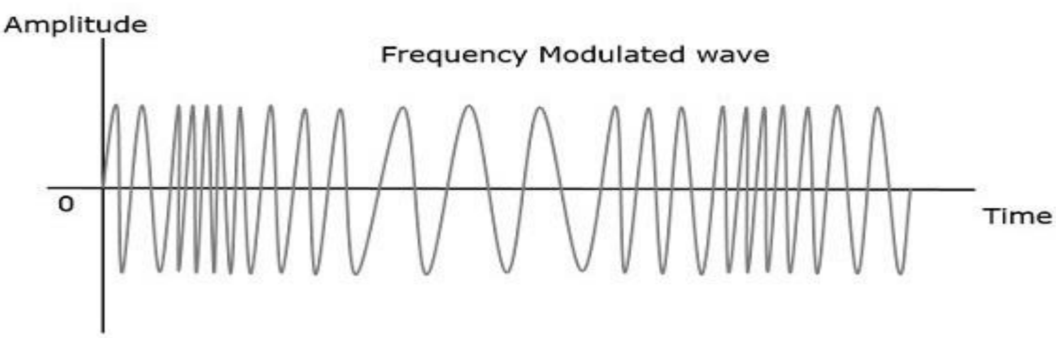
Frequency Modulation :

In amplitude modulation, the amplitude of the carrier signal varies. Whereas, in **Frequency Modulation (FM)**, the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.

Hence, in frequency modulation, the amplitude and the phase of the carrier signal remains constant.







The frequency of the modulated wave increases, when the amplitude of the modulating or message signal increases. Similarly, the frequency of the modulated wave decreases, when the amplitude of the modulating signal decreases. Note that, the frequency of the modulated wave remains constant and it is equal to the frequency of the carrier signal, when the amplitude of the modulating signal is zero.

The equation for instantaneous frequency f_i in FM modulation is,

$$fi = fc + k_f\, m(t)$$

Where,

fc is the carrier frequency

k_f is the frequency sensitivity

m(t) is the modulating signal

We know the relationship between angular frequency ωi and angle θi(t) as

$$wi = \frac{d\theta_i(t)}{dt}$$

$$2\pi fi = \frac{d\theta_i(t)}{dt}$$

$$\theta i(t) = 2\pi \int fi\, dt$$

$$\theta i(t) = 2\pi \int (fc + k_f\, m(t))\, dt$$

$$\theta i(t) = 2\pi fct + 2\pi k_f \int m(t)\, dt$$

Substitute, θi(t) value in the standard equation of angle modulated wave.

$$s(t)=E_c \cos(2\pi fct + 2\pi k_f \int m(t)\, dt)$$

This is the **equation of FM wave**.

If the modulating signal is m(t) = Em cos(2πfmt), then the equation of FM wave will be

$$s(t) = Ec \cos(2\pi fct + \beta \sin(2\pi fmt))$$

Where,

β = **modulation index** = $\frac{\Delta f}{f_m} = \frac{k_f\, E_m}{f_m}$

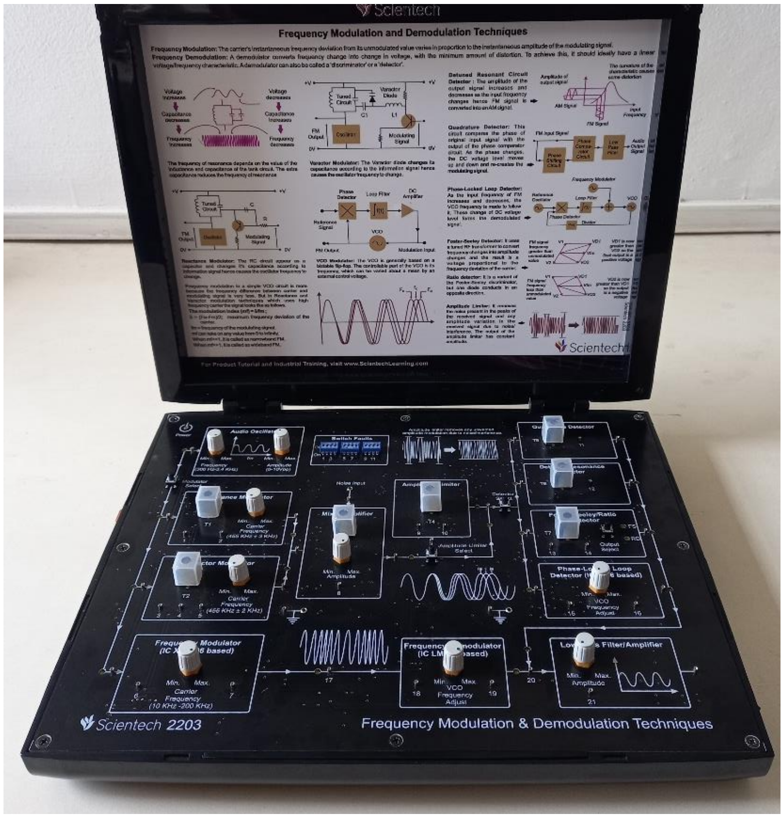
Procedure : (Task 1)

1) Produce Frequency Modulated signal in time domain and fine it’s Frequency Spectrum.
2) Add Noise in the signal and remove it using Amplitude Modulator.
3) Produce Original signal using FM demodulator.

Task 2: Perform Experiment on Matlab

1) Generate and observe the Frequency Modulated signal in time and frequency domain.
2) Perform Demodulation of FM signal and restore original signal.

ST-2203 kit for Frequency Modulation and Demodulation Techniques.



Observation :

Matlab Code :

Output :

Lab Tutorials:

1) State the Merits of FM over AM.

2) What is the role of Amplitude Limiter in FM receiver.

3) Discuss and make comparative analysis of Various types of Frequency Modulation.

4) Discuss and make comparative analysis of Various types of Frequency Demodulatoion.

Conclusion :

EXPERIMENT:3

Aim : To Study Sampling and Reconstruction of signal using ST-2101 kit and study Sampling theorem using MATLAB simulation.

Apparatus : ST-2101 kit, Matlab/LABVIEW software

Theory :

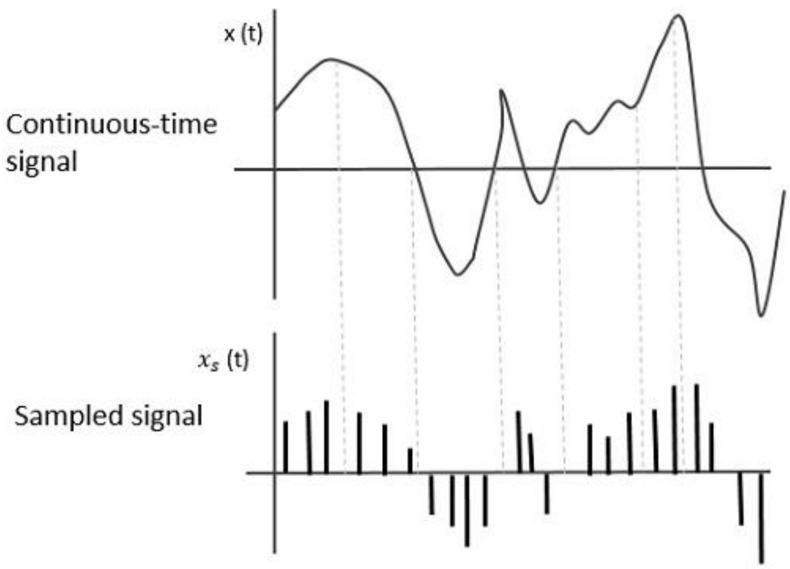
Sampling :

It is defined as, "The process of measuring the instantaneous values of continuous-time signal in a discrete form."

Sample is a piece of data taken from the whole data which is continuous in the time domain.

When a source generates an analog signal and if that has to be digitized, having **1s** and **0s** i.e., High or Low, the signal has to be discretized in time. This discretization of analog signal is called as Sampling.

The following figure indicates a continuous-time signal **x (t)** and a sampled signal $\mathbf{x_s}$ (t). When **x (t)** is multiplied by a periodic impulse train, the sampled signal $\mathbf{x_s}$ **(t)** is obtained.



Sampling Rate

To discretize the signals, the gap between the samples should be fixed. That gap can be termed as a sampling period Ts.

Sampling Frequency = $\frac{1}{T_s}$ **= fs**

Where, Ts is the sampling time

fs is the sampling frequency or the sampling rate

Sampling frequency is the reciprocal of the sampling period. This sampling frequency can be simply called as **Sampling rate**. The sampling rate denotes the number of samples taken per second, or for a finite set of values.

Reconstruction :

For an analog signal to be reconstructed from the digitized signal, the sampling rate should be highly considered. The rate of sampling should be such that the data in the message signal should neither be lost nor it should get over-lapped. Hence, a rate was fixed for this, called as Nyquist rate.

Nyquist Rate

Suppose that a signal is band-limited with no frequency components higher than Bandwidth. That means, BW is the highest frequency. For such a signal, for effective reproduction of the original signal, the sampling rate should be twice the highest frequency.

Which means,

$$\mathbf{f_S = 2\ BW}$$

Where,

$\mathbf{f_S}$ is the sampling rate

BW is the bandwidth of signal

This rate of sampling is called as Nyquist rate.

A theorem called, Sampling Theorem, was stated on the theory of this Nyquist rate.

Sampling Theorem :

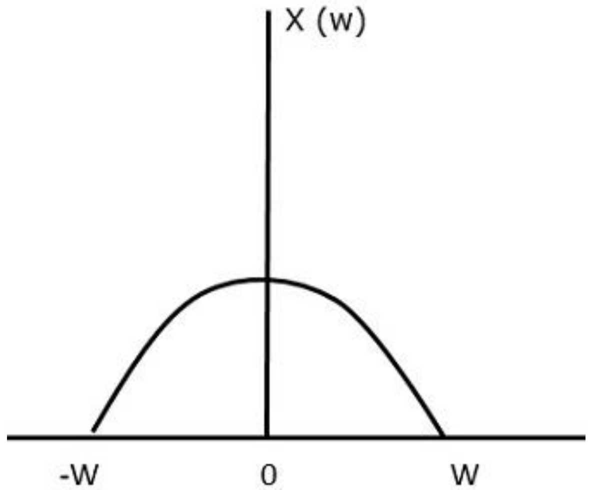
The sampling theorem, which is also called as Nyquist theorem, delivers the theory of sufficient sample rate in terms of bandwidth for the class of functions that are bandlimited.

The sampling theorem states that, “a signal can be exactly reproduced if it is sampled at the rate fs which is greater than twice the maximum frequency BW.”

To understand this sampling theorem, let us consider a band-limited signal, i.e., a signal whose value is non-zero between some –BW and BW.

Such a signal is represented as $x(f) = 0$ for $|f| > BW$

For the continuous-time signal x (t), the band-limited signal in frequency domain, can be represented as shown in the following figure.

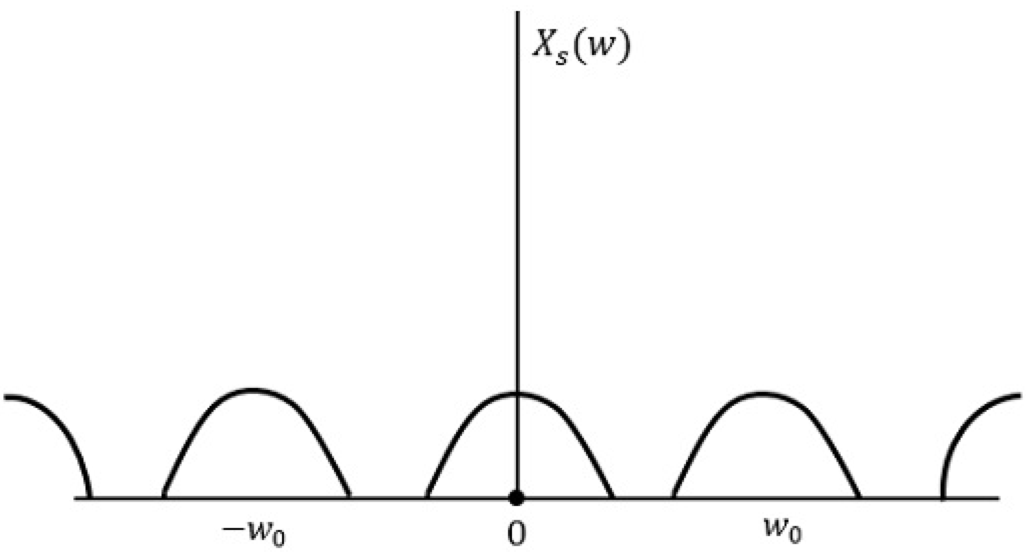


Band limited signal

We need a sampling frequency, a frequency at which there should be no loss of information, even after sampling. For this, we have the Nyquist rate that the sampling frequency should be two times the maximum frequency. It is the critical rate of sampling.

If the signal x(t) is sampled above the Nyquist rate, the original signal can be recovered, and if it is sampled below the Nyquist rate, the signal cannot be recovered.

The following figure explains a signal, if sampled at a higher rate than 2BW in the frequency domain.



The above figure shows the Fourier transform of a signal $x_s(t)$. Here, the information is reproduced without any loss. There is no mixing up and hence recovery is possible.

The Fourier Transform of the signal $x_s(t)$ is

$$X_s(w) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(w - nw_0)$$

Where Ts = Sampling Period and w0 = $\frac{2\pi}{T_s}$

Let us see what happens if the sampling rate is equal to twice the highest frequency (2 BW)

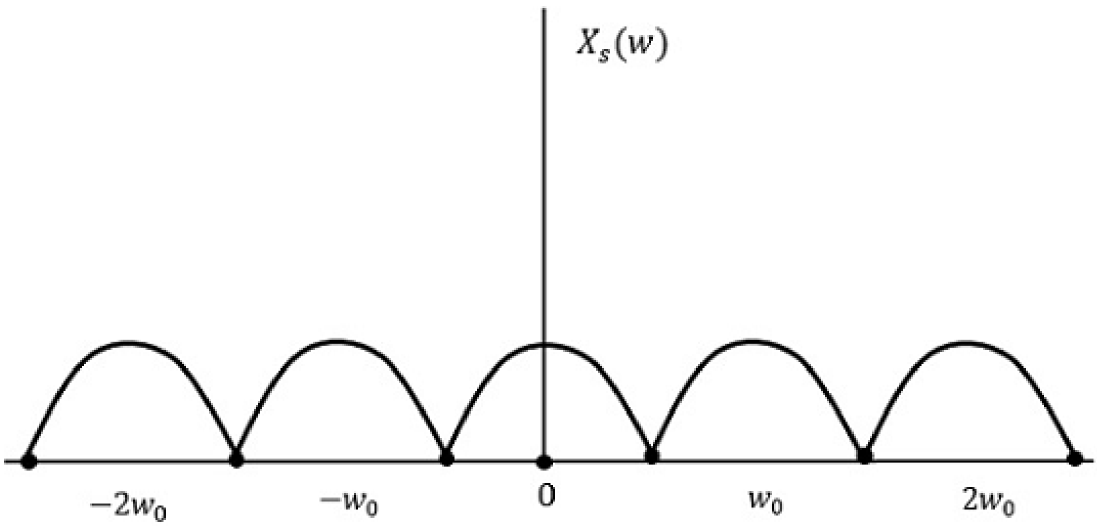
That means,

$$fs = 2\ BW$$

Where,

fs is the sampling frequency

BW is the highest frequency

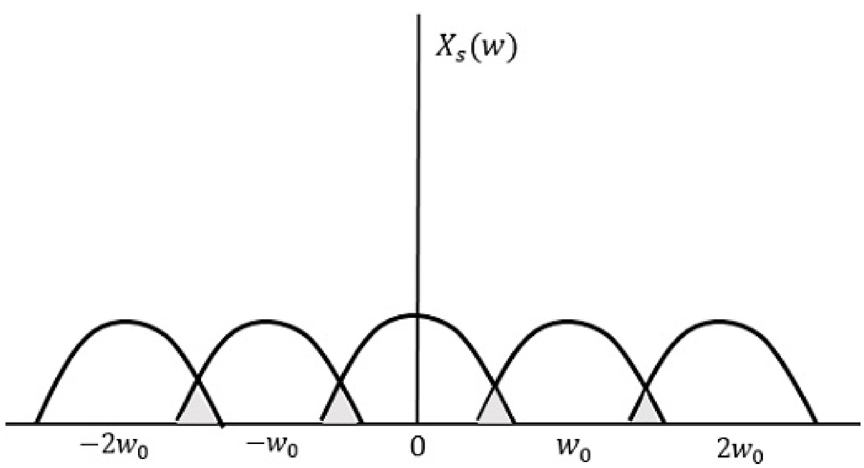


The result will be as shown in the above figure. The information is replaced without any loss. Hence, this is also a good sampling rate.

Now, let us look at the condition,

$$fs < 2W$$

The resultant pattern will look like the following figure.



We can observe from the above pattern that the over-lapping of information is done, which leads to mixing up and loss of information. This unwanted phenomenon of over-lapping is called as Aliasing.

Aliasing

Aliasing can be referred to as "the phenomenon of a high-frequency component in the spectrum of a signal, taking on the identity of a low-frequency component in the spectrum of its sampled version."

The corrective measures taken to reduce the effect of Aliasing are −

- In the transmitter section of PCM, a low pass anti-aliasing filter is employed, before the sampler, to eliminate the high frequency components, which are unwanted.
- The signal which is sampled after filtering, is sampled at a rate slightly higher than the Nyquist rate.

This choice of having the sampling rate higher than Nyquist rate, also helps in the easier design of the reconstruction filter at the receiver.

Procedure :

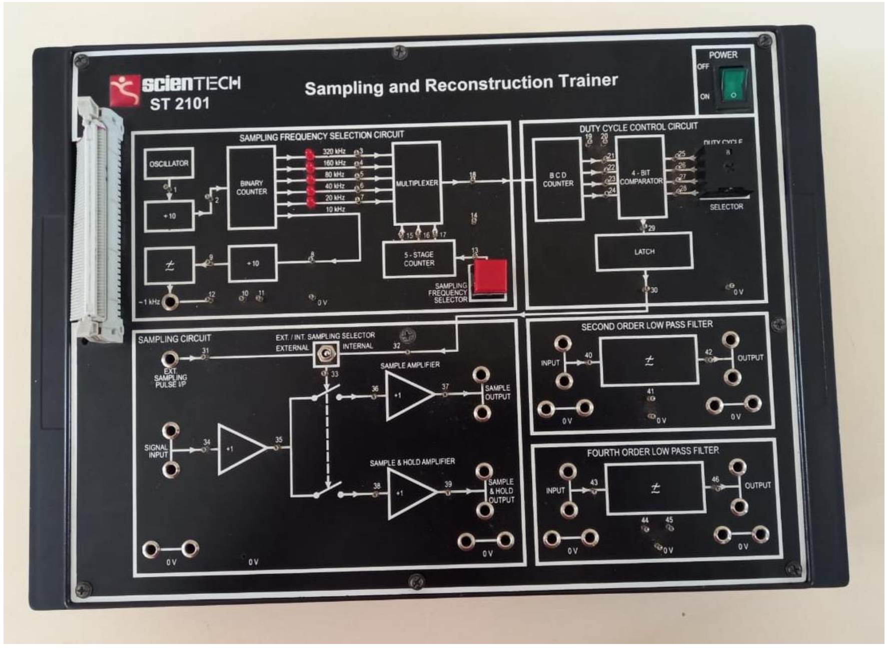
Task 1: Observation and analysis using ST-2101 kit.

1) Observe the sampled output of the signal at various sampling rate in time and frequency domain using MSO and store the waveform in USB drive. In frequency domain, observe the effect of frequency spectrum at various sampling rate.
2) Repeat the same experiment for sample and hold circuit.
3) Reconstruct the signal using 2nd and 4th order Low-pass filter. Find the difference in outcome at various sampling rate, store the waveform.
4) Repeat the same experiment with sample and hold circuit and note down the difference in output.
5) Change the duty cycle of signal and observe the output at Low-pass filter, find the significance of duty cycle in output.

Task 2: Perform Experiment using MATLAB.

1) Take a signal, sample it, observe the frequency spectrum of signal in frequency domain.
2) Apply Low-Pass Filter and restore the original signal.
3) See the effect of different sampling rate on the restored signal.

ST-2101 kit for Sampling and Reconstruction.



Observation :

Matlab Code :

Output :

Lab Tutorials :

1) What is Nyquist criteria ?

2) What is the significance of Nyquist criteria in Sampling and Reconstruction of signal ?

3) What is the basic difference between Natural Sampling and Sample and Hold circuit ?

4) How Low Pass Filter (LPF) is helpful to restore the original signal ?

5) What is the order of filter, what happened if order of filter is increased/ decreased?

Conclusion :

EXPERIMENT: 4

Aim : To Study Pulse Code Modulation (PCM) using Time Division Multiplexing (TDM) of the Signals.

Apparatus : ST-2103 & ST-2104 kit, MATLAB/LABVIEW software.

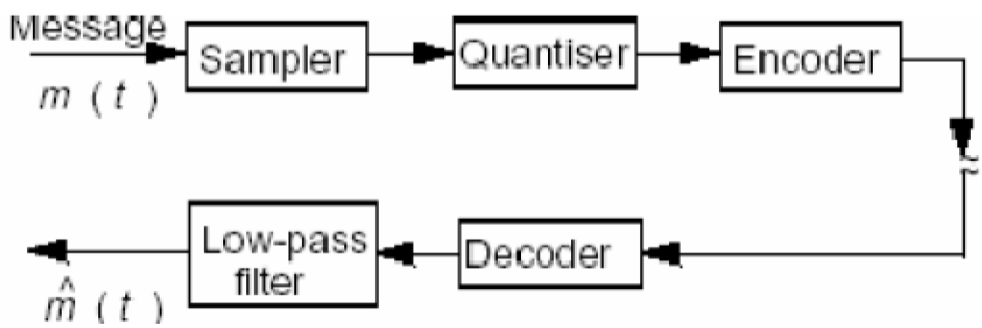
Theory :

PCM (Pulse code modulation) :

PCM is a method used to digitally represent sampled analog signals. It is the standard form of digital audio in computers, Compact Discs, digital telephony and other digital audio applications.

In PCM System the amplitude of the sampled waveform at definite time intervals is represented as a binary code. In PCM, first step is sampling, The analog signal is sampled according to the Nyquist criteria. The Nyquist criteria states that for faithful reproduction of the band limited signal, the sampling rate must be at least twice the highest frequency component present in the signal.

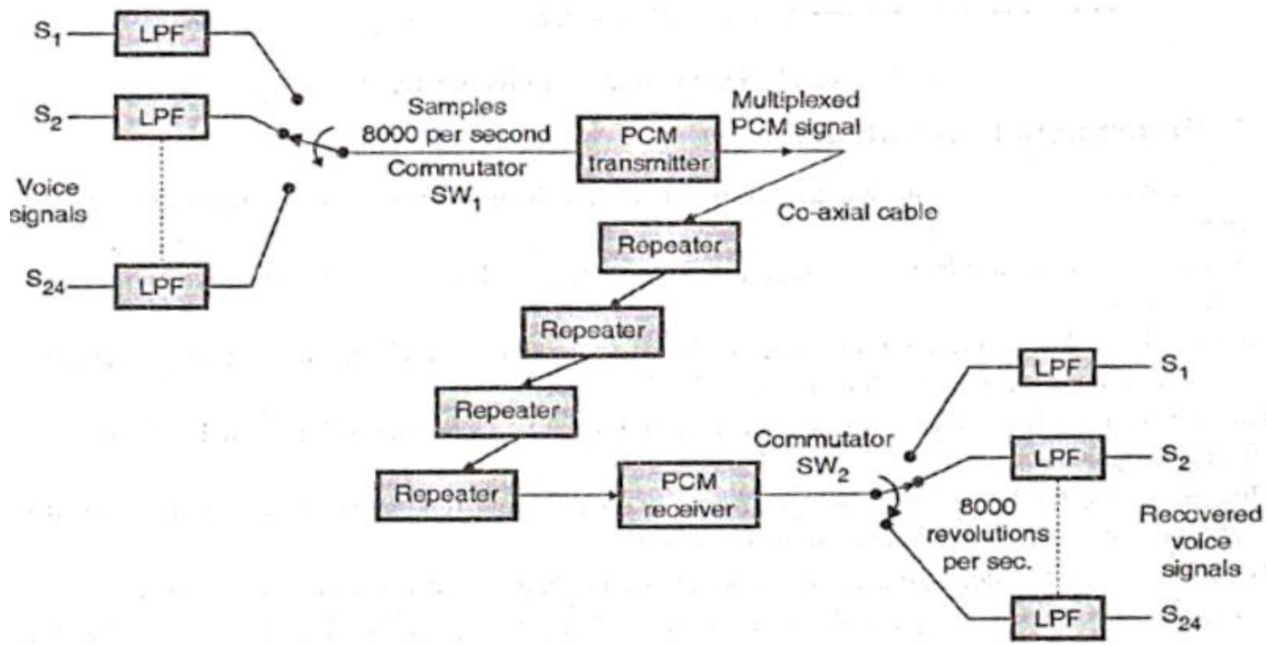
Each binary word defines a particular narrow range of amplitude level. The sampled value is then approximated to the nearest amplitude level. The sample is then assigned a code corresponding to the amplitude level, which is then transmitted. This process is called as *Quantization* & it is generally carried out by the A/D converter. Quantization levels are encoded using uniform and non uniform encoding method. Some error correcting codes & synchronization can also be transmitted along with the information signal.
At receiver, the data is decoded by the D/A converter; the recovered samples are filtered & reconstructed to provide the original waveform



Single channel PCM transmission system

TDM (Time-division multiplexing) :

TDM is a method of transmitting and receiving independent signals over a common signal path by means of synchronized switches at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern. It is used when the data rate of the transmission medium exceeds that of signal to be transmitted.



When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required. Figure shows the basic time division multiplexing scheme, called as the T1 digital system. This system is used to convey multiple signals over telephone lines using Wideband coaxial cable.

The operation of the PCM-TDM system is as follows: This system has been designed to accommodate 24 voice channels marked S1 to S24 . Each signal is band limited to 3.3 kHz, and the sampling is done at a standard rate of 8 kHz. This is higher than the Nyqvist rate.The sampling is done by the commutator switch SW1 . These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW1. Each sampled signal is then applied to the PCM transmitter which converts it into a digital signal by the process of A to D conversion and comparing, as explained earlier. The resulting digital waveform is transmitted over a co-axial cable.

Periodically, after every 6000 ft, the PCM-TDM signal is regenerated by amplifiers called “repeaters”. They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean PCM-TDM signal at their output. This ensures that the received signal is free from the distortions and noise. At the destination the signal is companded, decoded and demultiplexed, using a PCM receiver. The PCM receiver output is connected to different low pass filters via commutator switch SW2 . Synchronization between the transmitter and receiver commutators SW1 and SW2 is essential in order to ensure proper communication.

Procedure:

Task 1: Data Transmission and Reception using TDM-PCM kits (ST-2103 & ST-2104) and learn various synchronization modes.

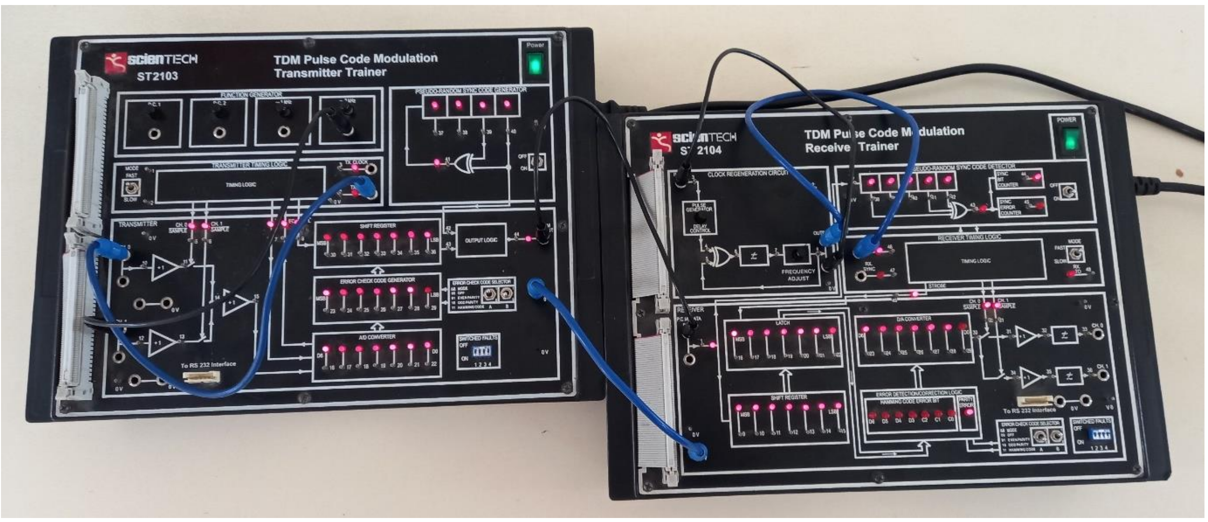
1) Connect the input to DC input to channel and see the A/D conversion (Refer Manual of ST- 2103)
2) Observe the PAM multiplexed output with 1Khz & 2Khz signal and it’s frequency spectra.
3) Observe the PCM output with different pattern of error correcting codes on ST-2103 kit.
4) Connect the ST-2103 & ST-2104 kits, operate them in all the modes of synchronisation and observe the restored signal at ST-2104.

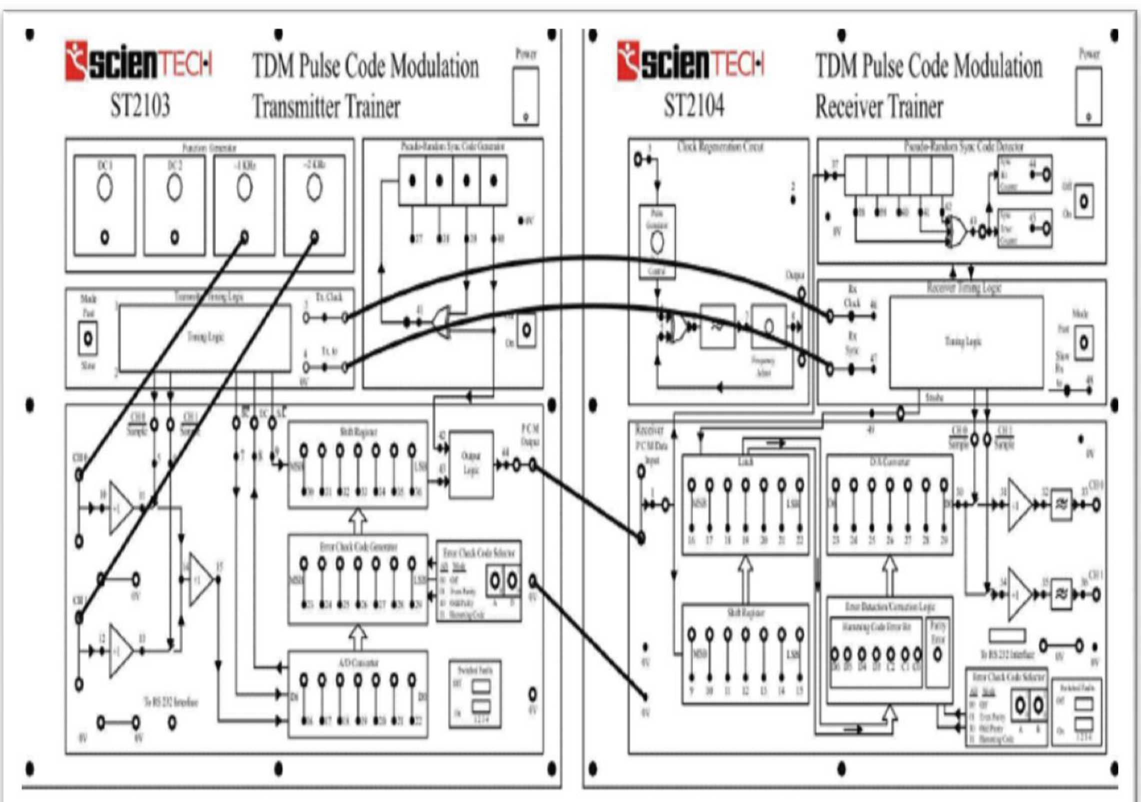
5) Learn the importance of synchronisation in signal reconstruction in TDM case.

Task 2: Perform Experiment using MATLAB/LABVIEW.

1) Take a signal, sample it, observe the frequency spectrum of signal in frequency domain.
2) Quantize the signal with 16 levels, and reconstruct the signal. Plot the original signal with quantized and reconstructed signal. Repeat the same with 256 levels and observe the difference.
3) See the effect of different sampling rate on the restored signal.

ST-2103 & ST-2104 kit for TDM Pulse Code Modulation Transmitter & Receiver :





Observation :

Signal	Amplitude	Frequency
m(t)		
CLK		
PCM o/p		
m(t)’		

Matlab Code :

Output :

Lab Tutorials:

1) What is Multiplexing? Discuss the types of Multiplexing.
2) State the Merits and De-merits of Time Division Multiplexing?
3) Explain the importance to synchronisation in TDM?
4) What is the role of error correcting codes, state the difference between various error correcting codes, you have studied in this experiment?
5) What is the importance of number of levels for quantization in reconstruction of signal in PCM?

Conclusion:

PRACTICAL-5

AIM: To study PAM, PWM and PPM

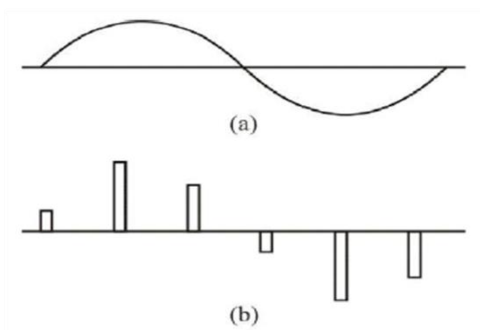
APPRATUS: ST-2110, Oscilloscope

THEORY:

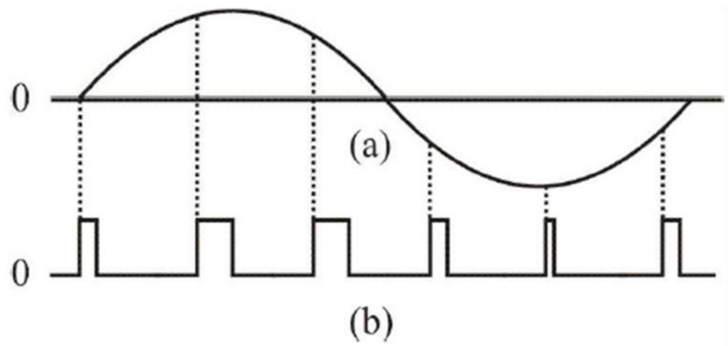
Pulse amplitude modulation is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling.

In Pulse position modulation The Amplitude and width of the pulses is kept constant in this system, while the position of each pulse, in relation to the position of a recurrent reference. Pulse is varied by each instantaneous sampled value of the modulating wave.

In Pulse width modulation, we have fixed amplitude and starting time of each pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant. As mentioned in connection with pulse width modulation, pulse-position modulations has the advantage of requiring constant transmitter power output, but the disadvantages of depending on transmitter receiver is synchronization



PAM



PWM

PROCEDURE:

Part 1: Voice Link Using Pulse Amplitude Modulation

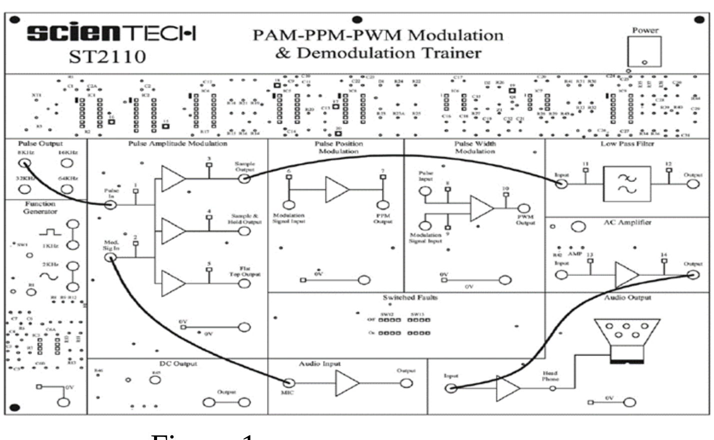


Figure 1

1. Connect the circuit as shown in Figure 1 and also described below for clarity.

a) Connect a microphone in the MIC socket in audio input block.
b) Connect the output of audio input block to MOD input of PAM block.
c) Connect the 8 KHz pulse output to pulse input of PAM block.
d) The sample output of PAM block to input of low pass filter.
e) Output of low pass filter to AC amplifier.
f) Gain pot of AC amplifier in mid position.
g) Output of AC amplifier to input of audio output block.

2. Switch 'On' the power supply.

3. You can observe the pulse being modulated by audio signal at output of sample output, sample & hold & flat top outputs.

4. Also, you can observe its demodulation and hear the same voice in speaker /headphone which was fed in the microphone in the input.

5. Switch 'Off' the power supply.

Part 2: Voice Link Using Pulse Position Modulation

1. Connect the circuit as shown in Figure 2 and also described below for clarity.

a) Connect a microphone in MIC socket of audio input block.
b) Output of audio input block to input of PPM block.
c) Output of PPM block to input of low pass filter block.
d) Output of low pass filter block to input of AC amplifier block.
e) Keep the frequency selector switch in 1 KHz position.
f) Keep the gain preset of AC amplifier in mid position.
g) Connect the output of AC amplifier block to input of audio output block.

2. Switch 'On' the power supply.

3. You can study the PPM using voice, by observing the waveforms at different stages.

4. The input is heard by means of speaker or headphone.

5. Switch 'Off' the power supply

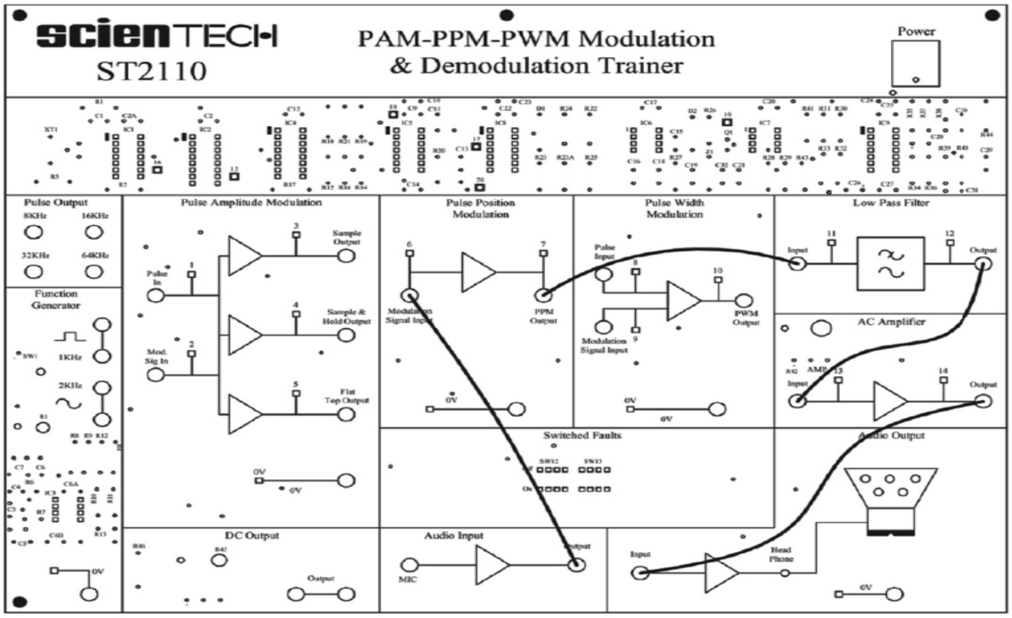


Figure 2

Part 3: Voice Link Using Pulse Width Modulation

1. Connect the circuit as shown in Figure 3 and also described below for clarity.

a) Connect a microphone in MIC socket of audio input block.
b) Output of audio input block to modulation signal in of PWM block.
c) 64 KHz Square Wave output to pulse input.
d) Output of PWM to input of low pass filter.
e) Output of low pass filter to input of AC Amplifier.
f) Keep the gain preset of AC amplifier in mid position.
g) Connect the output of AC amplifier block to input of audio output block.

2. Perform the pulse width modulation & demodulation experiment.
3. Switch ‘On’ the power supply.
4. You can study the pulse width modulation of a pulse signal by voice signal, by observing the outputs at different stages.
5. The voice output can be heard in speaker or headphone.
6. Switch ‘Off’ the power supply.

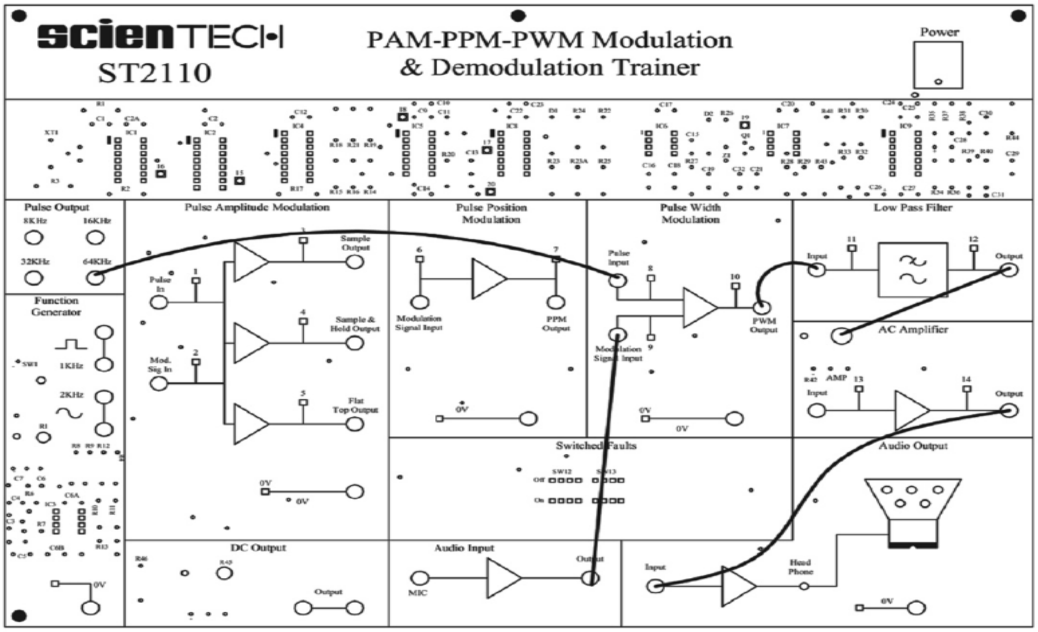


Figure 3

CONCLUSION:

EXPERIMENT: 6

AIM : To Study ASK, FSK and BPSK Scheme and it's function using MATLAB

Apparatus : MATLAB

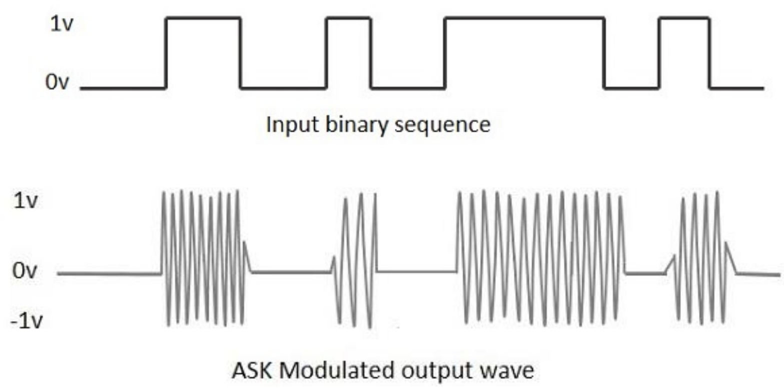
Theory :

Amplitude Shift Keying (ASK) :

Amplitude Shift Keying ASK is a type of Amplitude Modulation which represents the binary data in the form of variations in the amplitude of a signal.

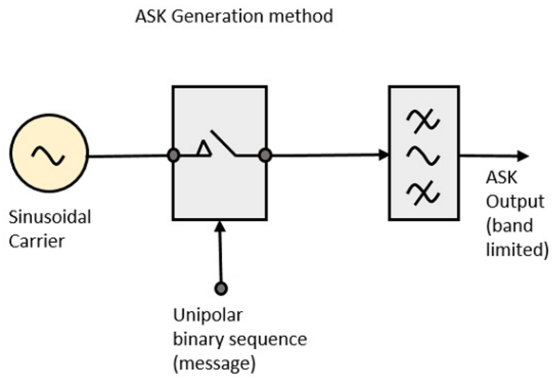
Any modulated signal has a high frequency carrier. The binary signal when ASK modulated, gives a zero value for Low input while it gives the carrier output for High input.

The following figure represents ASK modulated waveform along with its input.



ASK Modulator :

The ASK modulator block diagram comprises of the carrier signal generator, the binary sequence from the message signal and the band-limited filter. Following is the block diagram of the ASK Modulator.



The carrier generator sends a continuous high-frequency carrier. The binary sequence from the message signal makes the unipolar input to be either High or Low. The high signal closes the switch, allowing a carrier wave. Hence, the output will be the carrier signal at high input. When there is low input, the switch opens, allowing no voltage to appear. Hence, the output will be low.

The band-limiting filter shapes the pulse depending upon the amplitude and phase characteristics of the band-limiting filter or the pulse-shaping filter.

ASK Demodulator :

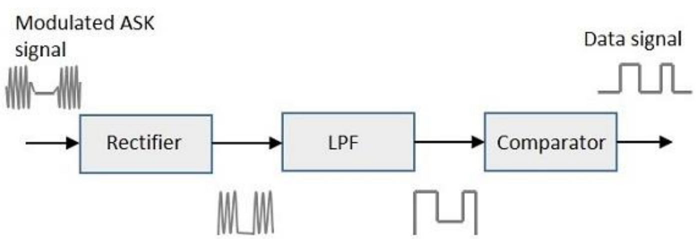
There are two types of ASK Demodulation techniques. They are

- Asynchronous ASK Demodulation/detection
- Synchronous ASK Demodulation/detection

The clock frequency at the transmitter when matches with the clock frequency at the receiver, it is known as a Synchronous method, as the frequency gets synchronized. Otherwise, it is known as Asynchronous.

Asynchronous ASK Demodulator :

The Asynchronous ASK detector consists of a half-wave rectifier, a low pass filter, and a comparator. Following is the block diagram for the same.

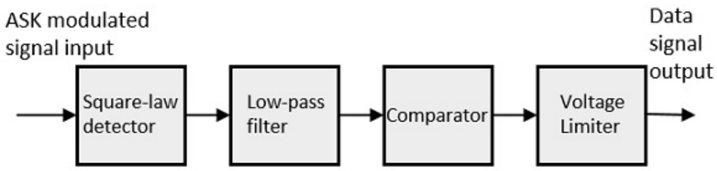


Asynchronous ASK detector

The modulated ASK signal is given to the half-wave rectifier, which delivers a positive half output. The low pass filter suppresses the higher frequencies and gives an envelope detected output from which the comparator delivers a digital output.

Synchronous ASK Demodulator :

Synchronous ASK detector consists of a square law detector, low pass filter, a comparator, and a voltage limiter. Following is the block diagram for the same.



Synchronous ASK detector

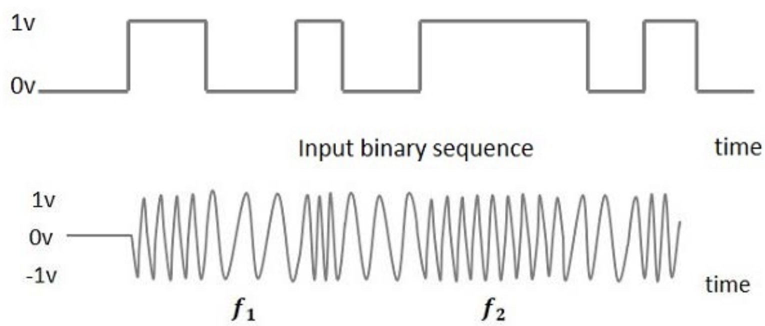
The ASK modulated input signal is given to the square law detector. A square law detector is one whose output voltage is proportional to the square of the amplitude modulated input voltage. The low pass filter minimizes the higher frequencies. The comparator and the voltage limiter help to get a clean digital output.

Frequency Shift Keying (FSK) :

Frequency Shift Keying FSK is the digital modulation technique in which the frequency of the carrier signal varies according to the digital signal changes. FSK is a scheme of frequency modulation.

The output of a FSK modulated wave is high in frequency for a binary High input and is low in frequency for a binary Low input. The binary 1s and 0s are called Mark and Space frequencies.

The following figure is the representation of FSK modulated waveform along with its input.



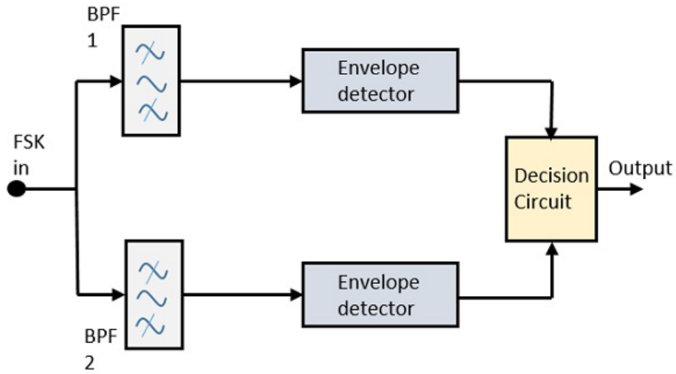
FSK Modulated output wave

FSK Demodulator :

There are different methods for demodulating a FSK wave. The main methods of FSK detection are asynchronous detector and synchronous detector. The synchronous detector is a coherent one, while asynchronous detector is a non-coherent one.

Asynchronous FSK Detector :

The block diagram of Asynchronous FSK detector consists of two band pass filters, two envelope detectors, and a decision circuit. Following is the diagrammatic representation.

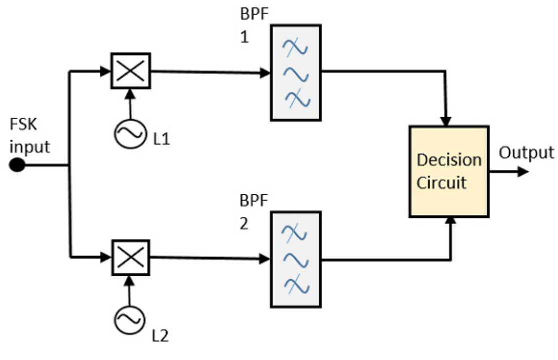


The FSK signal is passed through the two Band Pass Filters BPFs, tuned to Space and Mark frequencies. The output from these two BPFs look like ASK signal, which is given to the envelope detector. The signal in each envelope detector is modulated asynchronously.

The decision circuit chooses which output is more likely and selects it from any one of the envelope detectors. It also re-shapes the waveform to a rectangular one.

Synchronous FSK Detector :

The block diagram of Synchronous FSK detector consists of two mixers with local oscillator circuits, two band pass filters and a decision circuit. Following is the diagrammatic representation.



The FSK signal input is given to the two mixers with local oscillator circuits. These two are connected to two band pass filters. These combinations act as demodulators and the decision circuit chooses which output is more likely and selects it from any one of the detectors. The two signals have a minimum frequency separation.

Phase Shift Keying (PSK) is the digital modulation technique in which the phase of the carrier signal is changed by varying the sine and cosine inputs at a particular time. PSK technique is widely used for wireless LANs, bio-metric, contactless operations, along with RFID and Bluetooth communications.

PSK is of two types, depending upon the phases the signal gets shifted. They are −

Binary Phase Shift Keying (BPSK) :

Binary Phase Shift Keying (BPSK) is a type of digital modulation technique in which we are sending one bit per symbol i.e., ‘0’ or a ‘1’. Hence, the bit rate and symbol rate are the same. Depending upon the message bit, we can have a phase shift of 0^o or 180^o with respect to a reference carrier as shown in the figure above.

For example, we can have the following transmitted band-pass symbols:

$$S_1 = \sqrt{\frac{2E}{T}} \cos(2\pi f t) \rightarrow represents\ '1'$$

$$S_2 = \sqrt{\frac{2E}{T}} \cos(2\pi f t + \pi) \rightarrow represents\ '0'$$

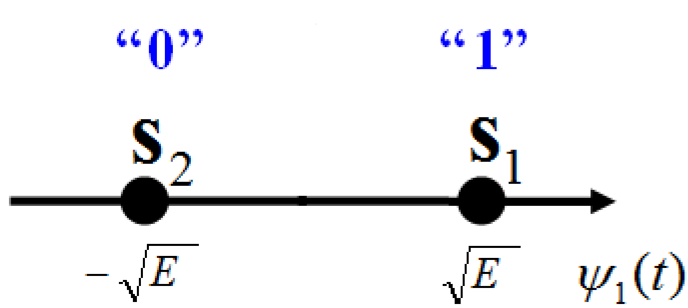
or

$$S_2 = -\sqrt{\frac{2E}{T}} \cos(2\pi f t) \rightarrow represents\ '0'$$

Where ‘E’ is the symbol energy, ‘T’ is the symbol time period, f is the frequency of the carrier. Using Gram-schmidt orthogonalization, we get a single orthonormal basis function, given as:

$$\psi_1 = \sqrt{\frac{2}{T}} \cos(2\pi f t)$$

Hence, the resulting constellation diagram can be given as:



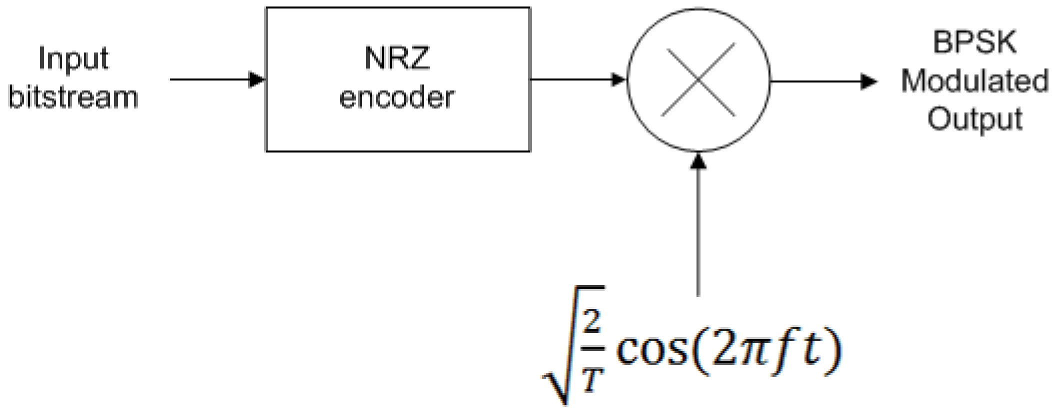
Constellation Diagram Of BPSK Signal

There are only two in-phase components and no quadrature component.

Now, we can easily see that the two waveform of S_o and S_1 are inverted with respect to one another and we can use following scheme to design a BPSK modulator:

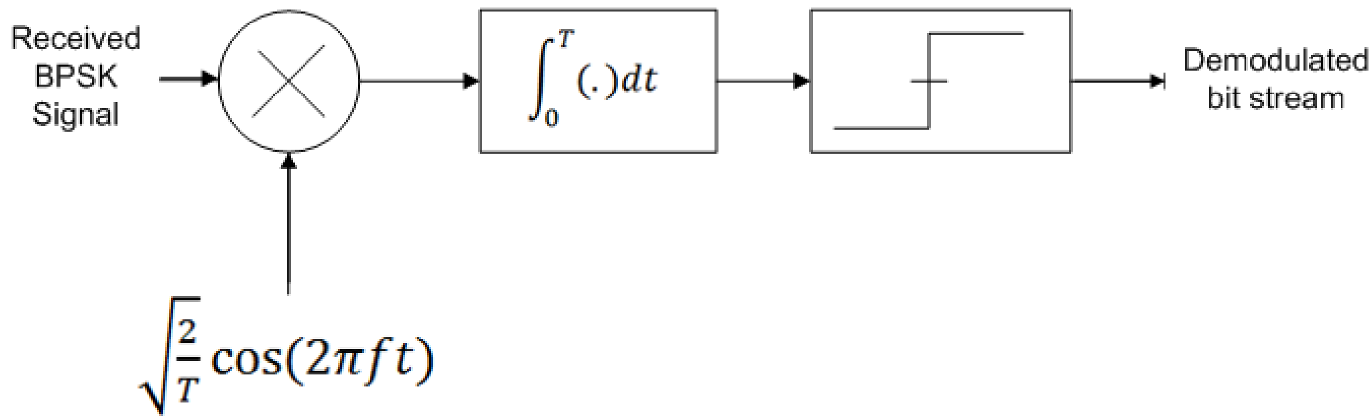
BPSK modulator

First the NRZ encoder converts these digital bits into impulses to add a notion of time into them. Then NRZ waveform is generated by up-sampling these impulses. Afterwards, multiplication with the carrier (orthonormal basis function) is carried out to generate the modulated BPSK waveform.



Demodulator Design:

We do coherent demodulation of the BPSK signal at the receiver. Coherent demodulation requires the received signal to be multiplied with the carrier having the same frequency and phase as at the transmitter. The phase synchronization is normally achieved using Phase Locked Loop (PLL) at the receiver. PLL implementation is not done here, rather we assume perfect phase synchronization. Block diagram of BPSK modulator is shown in the figure below. After the multiplication with the carrier (orthonormal basis function), the signal is integrated over the symbol duration ‘T’ and sampled. Then thresholding is applied to determine if a ‘1’ was sent (+ve voltage) or a ‘0’ was sent (-ve voltage).

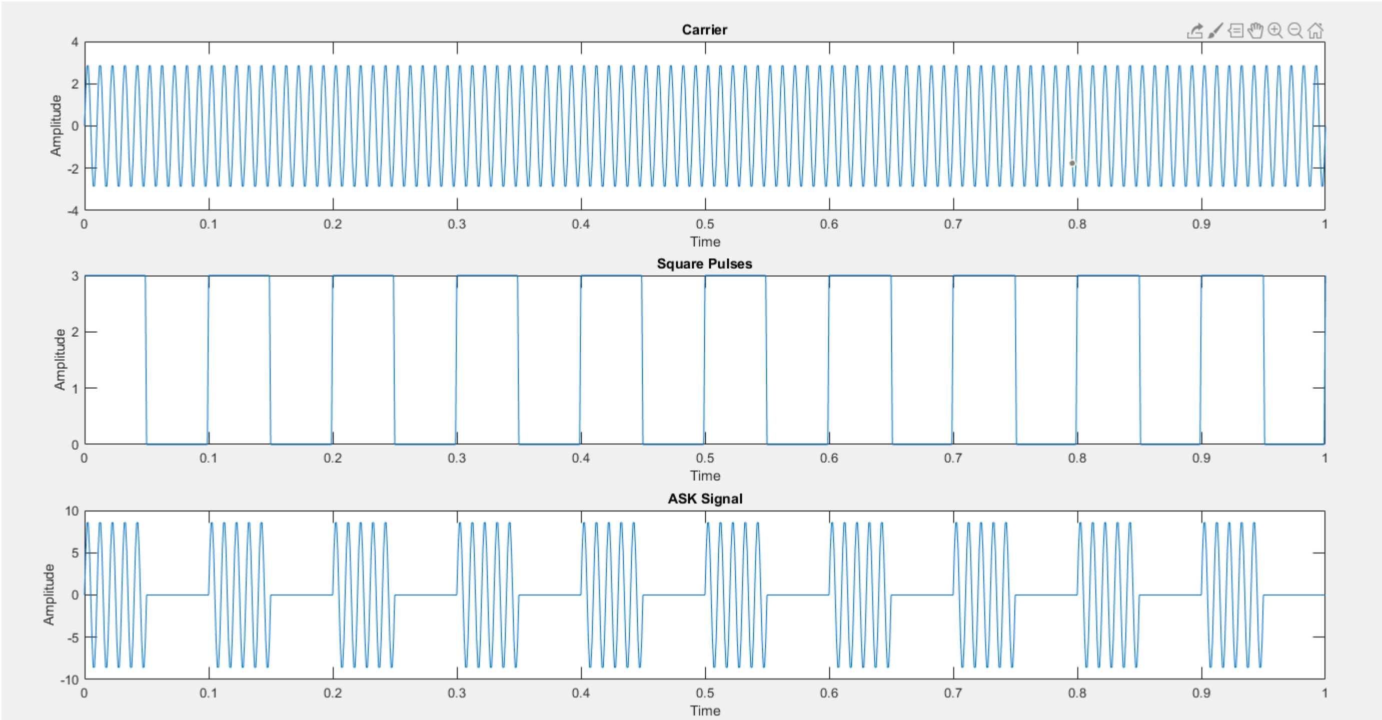


BPSK Receiver Design

The Matlab simulation code is given below. Here for the sake of simplicity, the bit rate is fixed to 1 bit/s (i.e., T=1 second). It is also assumed that Phased Locked Loop (PLL) has already achieved exact phase synchronization.

MATLAB Code: ASK

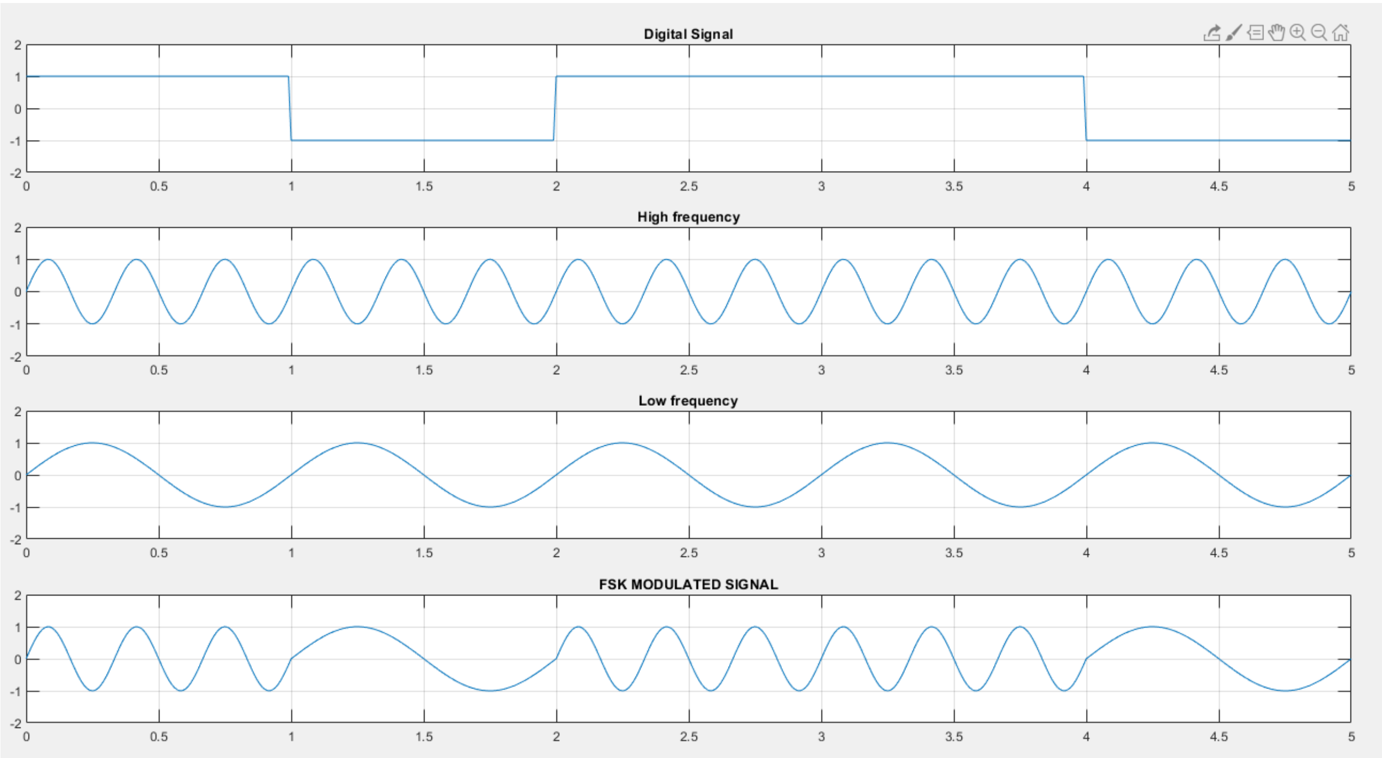
```
close all;
F1=input('Enter the frequency of carrier=');
F2=input('Enter the frequency of pulse=');
A=3;        %Amplitude
t=0:0.001:1;
x=A.*sin(2*pi*F1*t);     %Carrier Sine wave
u=A/2.*square(2*pi*F2*t)+(A/2);       %Square wave
v =x.*u;
subplot(3,1,1);
plot(t,x);
xlabel('Time');
ylabel('Amplitude');
title('Carrier');
subplot(3,1,2);
plot(t,u);
xlabel('Time');
ylabel('Amplitude');
title('Square Pulses');
subplot(3,1,3);
plot(t,v);
xlabel('Time');
ylabel('Amplitude');
title('ASK Signal');
```



MATLAB Code: FSK

```
clear;
clc;
b = input('Enter the Bit stream \n '); %b = [0 1 0 1 1 1 0];
n = length(b);
t = 0:.01:n;
x = 1:1:(n+1)*100;
for i = 1:n
if (b(i) == 0)
b_p(i) = -1;
else
b_p(i) = 1;
end
for j = i:.1:i+1
bw(x(i*100:(i+1)*100)) = b_p(i);
end
end
bw = bw(100:end);
wo = 2*(2*pi*t);
W = 1*(2*pi*t);
sinHt = sin(wo+W);
sinLt = sin(wo-W);
st = sin(wo+(bw).*W);
subplot(4,1,1)
plot(t,bw)
grid on ; axis([0 n -2 +2])
title('Digital Signal');
subplot(4,1,2)
plot(t,sinHt)
```

```
grid on ; axis([0 n -2 +2])
title('High frequency');
subplot(4,1,3)
plot(t,sinLt)
title('Low frequency');
grid on ; axis([0 n -2 +2])
subplot(4,1,4)
plot(t,st)
```



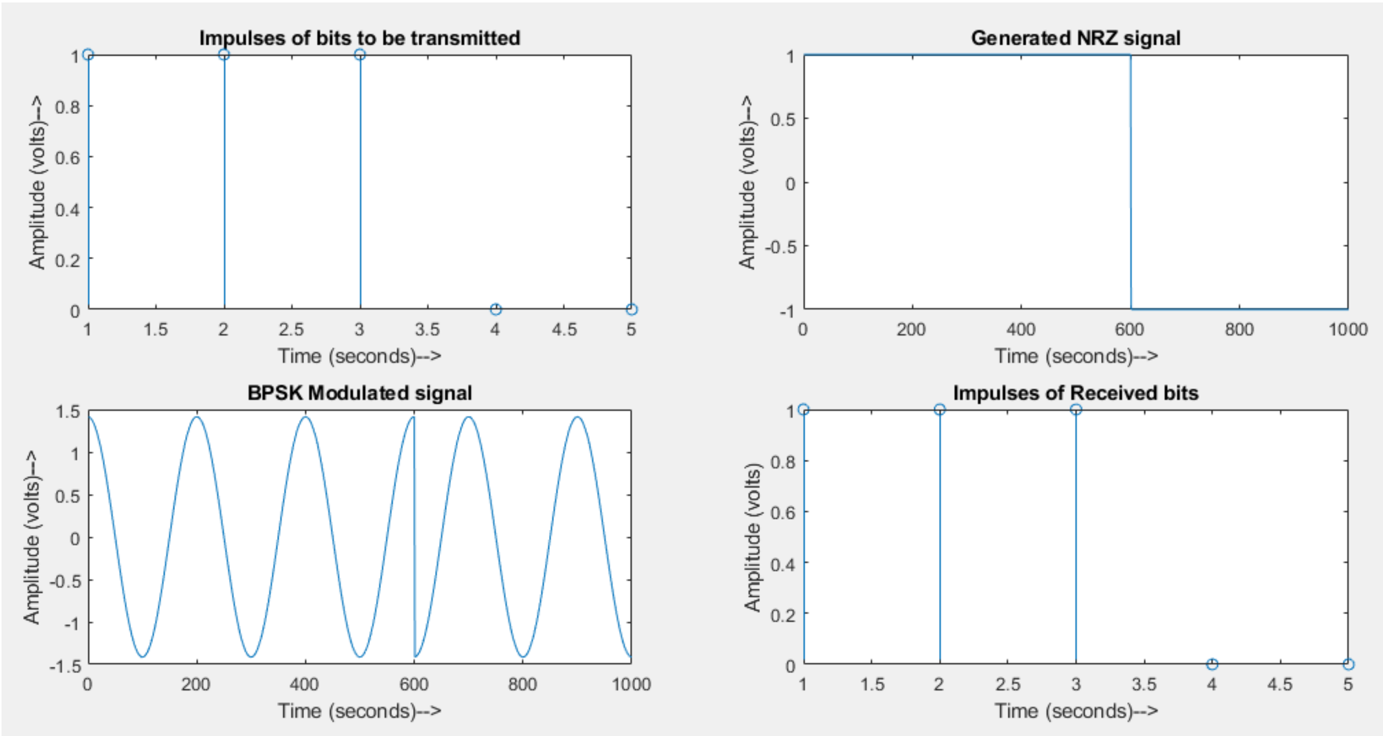
MATLAB Code: BPSK

```
clear all;
close all;
%Nb is the number of bits to be transmitted
T=1;%Bit rate is assumed to be 1 bit/s;
%bits to be transmitted
b=input("bits to be transmitted");
%Rb is the bit rate in bits/second
NRZ_out=[];
%Vp is the peak voltage +v of the NRZ waveform
Vp=1;
%Here we encode input bitstream as Bipolar NRZ-L waveform
for index=1:size(b,2)
 if b(index)==1
 NRZ_out=[NRZ_out ones(1,200)*Vp];
 elseif b(index)==0
 NRZ_out=[NRZ_out ones(1,200)*(-Vp)];
 end
end
```

```
%Generated bit stream impulses
figure(1);
stem(b);
xlabel('Time (seconds)-->')
ylabel('Amplitude (volts)-->')
title('Impulses of bits to be transmitted');
figure(2);
plot(NRZ_out);
xlabel('Time (seconds)-->');
ylabel('Amplitude (volts)-->');
title('Generated NRZ signal');

t=0.001:0.001:1;
%Frequency of the carrier
f=5;
%Here we generate the modulated signal by multiplying it with
%carrier (basis function)
Modulated=NRZ_out.*(sqrt(2/T)*cos(2*pi*f*t));
figure;
plot(Modulated);
xlabel('Time (seconds)-->');
ylabel('Amplitude (volts)-->');
title('BPSK Modulated signal');
y=[];
%We begin demodulation by multiplying the received signal again with
%the carrier (basis function)
demodulated=Modulated.*(sqrt(2/T)*cos(2*pi*f*t));
%Here we perform the integration over time period T using trapz
%Integrator is an important part of correlator receiver used here
for i=1:200:size(demodulated,2)
 y=[y trapz(t(i:i+199),demodulated(i:i+199))];
end
received=y>0;
figure;
stem(received)
title('Impulses of Received bits');
xlabel('Time (seconds)-->');
ylabel('Amplitude (volts)')
```

Observation :

BPSK



Impulses of bits to be transmitted
Amplitude (volts)-->
Time (seconds)-->
Generated NRZ signal
Amplitude (volts)-->
Time (seconds)-->
BPSK Modulated signal
Amplitude (volts)-->
Time (seconds)-->
Impulses of Received bits
Amplitude (volts)
Time (seconds)-->

Conclusion :

Experiment-7

Aim: To Study DTMF Telephone Trainer Kit (2654).

Apparatus: ST 2654 Kit, CRO

Theory:

The basic objective of DTMF Telephone Trainer is to study working of a telephone system. The basic components of a telephone system is shown in fig. 1 and 2. It consist of Handset, Key Board Dialer, Line on / Pulse Dialing Indication, Tone Dialing Indication, and Ringer Volume Control.

When receiver of telephone is in on hook condition then circuit gets + 40V to 60V (approximately) supply from telephone line. But because of blocking capacitor this supply does not reach to ringer section. In this condition, 20 Hz AC ring signals from the exchange are passed through blocking capacitor and goes to ringer section so that ring can be heard from the buzzer.

But when we pick up the receiver, voltage dropper section comes in to the picture. Now +60V (approximately) drops up to +12V and this is used for other operations of telephone circuit. First Tip and Ring signals are passed through polarity protection bridge. So if DC supply connection wires coming from exchange reversely connected then also because of polarity protection bridge right polarity obtained for telephone circuitry.

Key matrix have a special type of arrangement of switches. Here if any switch is pressed then corresponding row & column pins are shorted. This will provide a different function for different switches. Key matrix supports both pulse and Dual Tone multi frequency (DTMF) modes. This mode selection is possible with * switch. For pulse mode dialer section generates the pulses according to the number of that particular switch. For example, if we press '4' then '4' pulses are generated and sent to exchange. According to these pulses exchange got the identity about the pressed switch number.

In the same way in DTMF mode for every switch a row and column frequencies are there. It is the group of high and low frequencies. So when we press a particular switch corresponding High and Low frequency signals are sent to exchange. According to these signals exchange got the identity about the pressed switch number. Exchange have High Pass and Low Pass Filter circuits for identification. High Pass & Low Pass Filter enables us to observe the signals. It is not the part of telephones. It is of exchange. Sound signals are amplified by sound amplifier section also it amplifies the dial tone signals from dialing generator section.

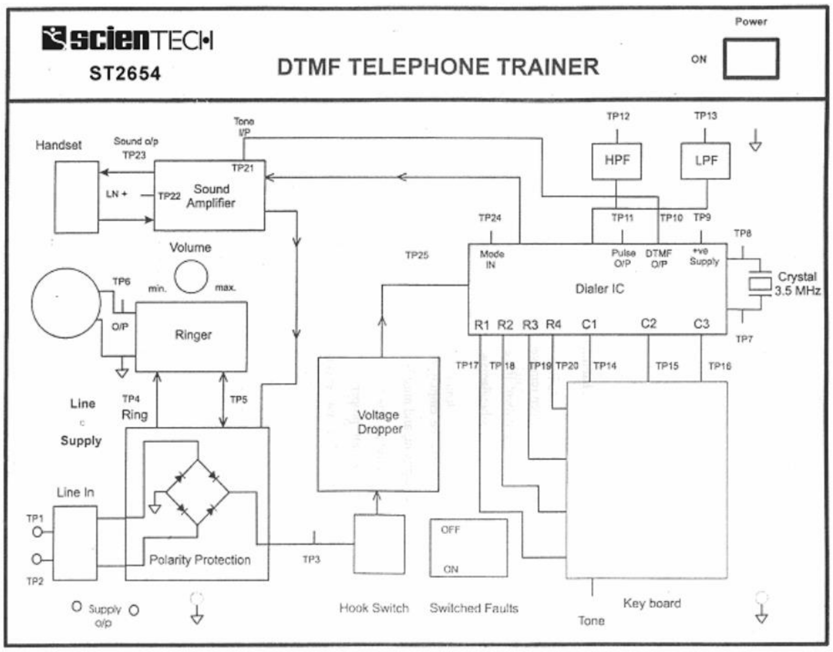


Fig. 1 Telephone trainer kit

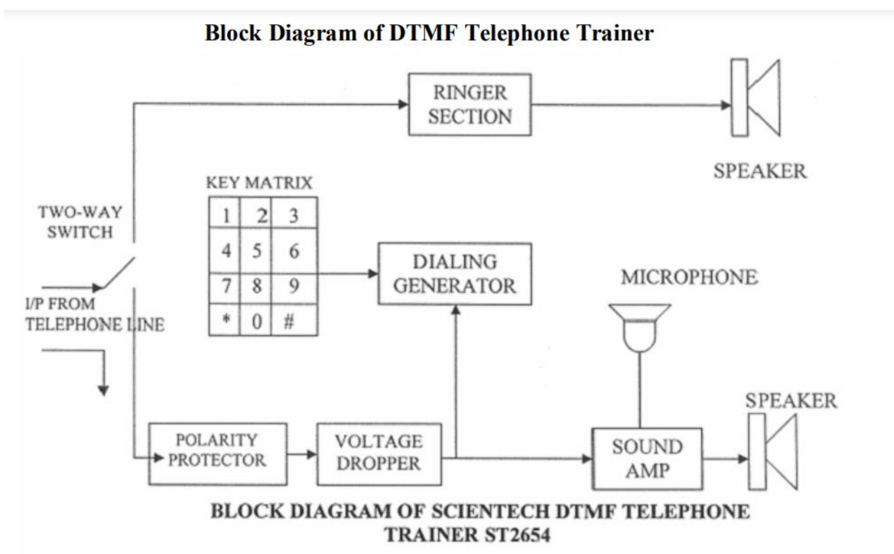


Fig. 2 Block diagram of trainer kit

(1) Study of the working of ringer circuit

Ringer circuit is mainly having a IC 1240. From telephone line one wire is directly connected to pin 8 of IC 1240 & another with a capacitor (105 K) C22 & resistor R25 (2K2) to Pin No. 1. In 'On' Hook condition 40 DC (approximately) coming from line is not going upto ringer circuit because of the blocking capacitor C22 but when ring signals are coming that is of 20 Hz frequency then these AC signals are passed through the capacitor & reaches up to ringer IC's Pin 1. Now these ring signals from pin 1 and 8 are rectified first & filtered by capacitor C23 (10μ) on pin7. Hence with this DC ring signals are generated by IC & obtained at pin5 of IC 1240. These amplified ring signals are given to speaker or buzzer with volume controlled potentiometer RV1 (100K) so that we can hear the ring signals. Between pin 2 & 3 a capacitor C24 (104K) is connected for the discharging of the signal & between 2 & 4 a resistor R26 (22K) is connected for ringing speed control.

Procedure :

1. Make all the fault switches in 'Off' condition.
2. Connect the line connector to telephone trainer board.
3. When trainer is in 'On' hook condition then there is no AC voltage on TP5 (due to blocking capacitor effect).
4. Now call on this telephone from some other telephone.
5. As ring signal are received by trainer board, observe them on TP5. It is 20 Hz AC signal with high amplitude.
6. TP4 will give you the ring signal which is directly coming from exchange.

(2) Study of the working of key matrix section

The key pad connected on trainer kit is of 3 × 4 matrix total 12 push buttons are there which are 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #. These switches are connected in 4 Rows & 3 Columns in such a way that every switch is having relation with one row & one column. When switch is pressed that row is shorted with corresponding column. These row & column tracks are connected with row & column pins of dialer IC 91214B. When switch is pressed signal from column pin of dialer IC goes to row pin through push button. Hence dialer IC gets the information about row & column of that switch which is pressed.

	C1	**C2**	**C3**
R1	1	2	3
R2	4	5	6
R3	7	8	9
R4	*	0	#

When 1 is pressed column pin 15 is shorted with row pin 12 of dialer IC. It means C1 is shorted with R1 & dialer IC gets the information that switch between pin 15 & 12 is pressed. By pressing different Key switches different row and column pins of dialer IC are shorted. These are as follows

Push Button	Pins to be shorted of dialer IC
0	18-13
1	15-12
2	15-13
3	15-14
4	16-12
5	16-13
6	16-14
7	17-12
8	17-13
9	17-14
*	18-12
#	18-14

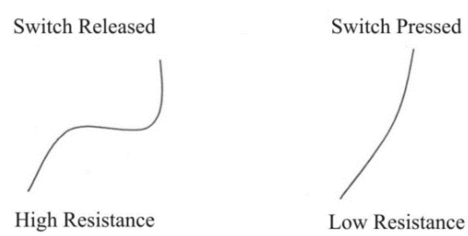
Procedure :

1. Make all the fault switches in 'Off' condition.

2. To understand the working of key pad block there is no need to connect the line or supply input.

3. Now use multimeter (low resistance range) or component tester facility of CRO.

4. Press 1 of keypad since 1 is making the connection of R1 with C1 so measure resistance between TP17 & TP14. It will be low. If released it will be high. If component tester pattern is observed on CRO between TP17 & TP14.



5. In the same way other switches of the key pad can be tested. Now how these are working electrically that we will see in dialer section

(3) Study of the Working DTMF Signals Using High Pass Filter and Low Pass Filter

The dialer circuit in this telephone trainer is made up of the IC UM91214 B and some other related components. This IC has the facility to select the tone or pulse dialing. In the pulse dialing mode, the number key pressed on the keypad to dial number are decoded by this IC and based on the pressed number this works as a pulse circuit.

For the DTMF dialing the DTMF sound tones are provided by this IC to the sound amplifier IC TEA 1062A from where this signal is amplified and passed on to the telephone line. A 3.58 MHz crystal is connected in between Pin 3 & 4 of the dialer IC. This crystal or ceramic resonator works as time base for the oscillator in the dialer IC. This helps the dialer IC to generate accurate DTMF signal. The number dialed from the key matrix is converted into tone or pulse dialing signal by the IC UM91214 in the dialing generator section.

In the pulse dialing mode, based on the key pressed this IC generates number of 'On–Off' pulses for example - If you press number 4 on the key pad this IC will send 4 'On–Off' pulses to telephone exchange. This 'On-Off' signals turns 'On' and 'Off' 10 times in each second and there is a delay of around 1 second between the 'On-Off' pulse of different numbers being dialed.

Pin 1 of this IC is HK i.e. Hook switch input. This inverter input pin detects the state of the Hook switch contact.

Pin 2 is mode input pin. It will select the dialing mode from either tone or pulse.

Pin 3 & 4 are oscillator pins. Crystal is connected between these Pins.

Pin 5 is negative supply (ground here).

Pin 6 is positive supply.

From pin 7 tone dialing output comes out, from pin 11 pulse dialing output comes out, pin 8 is used for mute function and pin 9 is mode indication by LED.

Pin 12, 13, 14 are column pins and pin 15, 16, 17, 18 are row pins.

When any switch from key pad is pressed pin 11 gives pulse output or pin 7 gives tone output (depends on mode selection) at that time exchange get amplified dial tone cut signals through voltage dropper section, hook switch and bridge network.

Upper Band Frequency

HF / LF	1209	1336	1477
687	1	2	3
770	4	5	6
852	7	8	9
941	*	0	#

Lower Band Frequency

Procedure :

1. Connect line connector to telephone trainer board.

2. Press * switch of key pad in ‘Off’-Hook condition, tone LED lits.

3. Trainer is set for tone mode.

4 . Now if any switch of key pad is pressed DTMF output is generated. It is the combination of high frequency signal and low frequency signal.

5. At TP8 crystal oscillator's 3.5 MHz frequency output is obtained if any switch pressed.

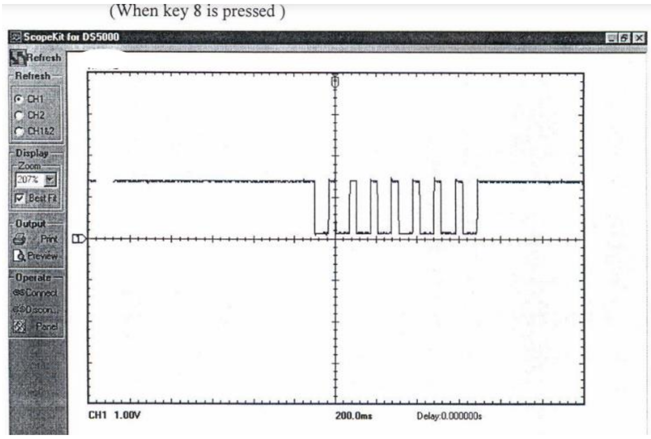
6. Now press hook switch. Trainer comes out of tone mode and now it is in pulse mode.

7. Now if any switch of keypad is pressed corresponding number of ‘On–Off’ pulses are generated and obtained at TP11.

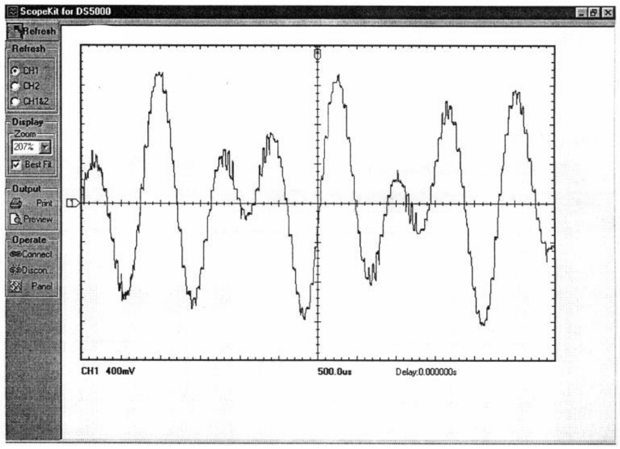
Observations:

(1) Diagram captured on DSO

(When key 8 is pressed)

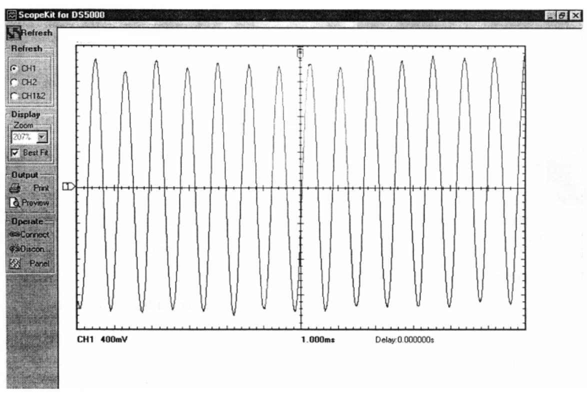


(2)

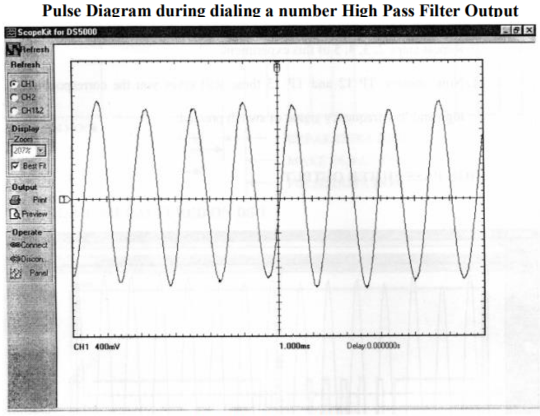


DTMF Output from Dialer

8. Disconnect the actual line connector from mimic board.
9. Connect simulated supply line.
10. Repeat steps 2, 3, 4, 5 of this experiment.
11. Now observe TP12 and TP13 these will gives you the corresponding high and low frequency signal of switch pressed.



Pulse Diagram during dialing a number High Pass Filter Output



Low Pass Filter output

Conclusion:

Experiment-8

Aim: To study setting up of Fiber optic

Apparatus: Oscilloscope, probes, connecting wires, optical cable, Scientech ST 2501 Fiber optics trainer, power supply.

Analog link

Theory:

Fiber optic links can be used for the transmission of digital as well as analog signals. Basically a fiber optic link contains three main elements, a transmitter, an optical fiber link and a receiver. The transmitter module takes the input signal in the electrical form and then transforms it into optical (light) energy containing the same information. The optical fiber is the medium, which takes the energy to the receiver. At the receiver light is converted back into the electrical form with the same pattern as originally fed to the transmitter.

Transmitter:

Fiber optic transmitters are typically composed of a buffer, driver and optical source. The buffer provides both an electrical connection and isolation between the transmitter & the electrical system supplying the data. The driver provides electrical power to the optical source. Finally, the optical source converts the electrical current to the light energy with the same pattern. Commonly used optical sources are light emitting diode (LEDs) and Laser Beam. Simple LED circuits, for digital and analog transmissions are shown below.

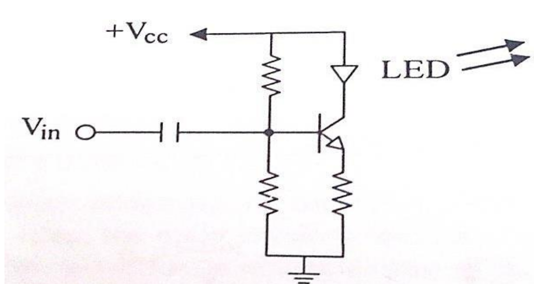


Fig shows Trans conductance drive circuits for analog transmission – common emitter configuration. The transmission section comprises of:
Function Generator
Frequency modulator &
Pulse width modulator block.

The function generator generates the input signals that are going to be used as the information to transmit through the fiber optic link. The output voltage available is 1 KHz sinusoidal signal of adjustable amplitude, and fixed amplitude 1 KHz square wave signal. The modulator section accepts the information signal and converts it into suitable form for transmission through the fiber optic link.

The Fiber Optic Link:

Emitter and Detector circuit on board form the fiber optic link. This section provides the light source for the optic fiber and the light detector at the far end of the optic fiber links. The optic fiber plugs into the connectors provided in this part of the board. Two separate links are provided.

The Receiver:

The Comparator circuit, Low Pass Filter, Phase Locked Loop, AC Amplifier Circuits form receiver on the board. It is able to undo the modulation process in order to recover the original information signal. In this experiment the trainer board is used to illustrate One-Way Communication between digital transmitter and receiver circuits.

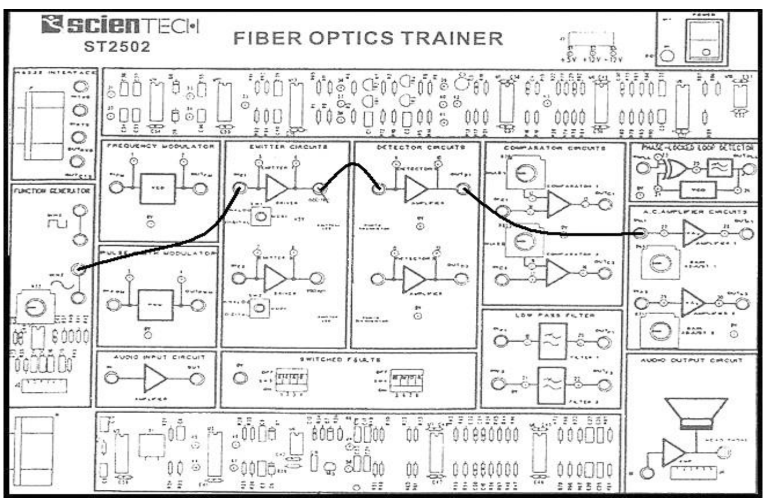


Figure:

Procedure:

1. Connect the power supply to the board.
2. Ensure that all switched faults are off.
3. Make the following connections. (As shown in the below figure)
   a. Connect the F.G. 1 KHz sine wave to the emitter's input.

b. Connect the F.O. cable between emitter output and detector’s input.
c. Detector Output to AC Amplifier input.

4. On the board, switch emitter driver to analog mode.
5. Switch ON the power.
6. Observe the input to emitter (t.p. 5) with the output from AC Amplifier (t.p. 19) and notethat the two signals are same.

Conclusion:

After performing this practical we can study about analog link, receive signal, transmitted signal and the relationship between the analog input signal and received signal.

a) To study setting up of Fiber optic Digital link.

Apparatus: Oscilloscope, probes, connecting wires, optical cable, Scientech ST 2501 Fiber optics trainer, power supply.

Theory:

Fiber optic links can be used for the transmission of digital as well as analog signals. Basically a fiber optic link contains three main elements, a transmitter, an optical fiber link and a receiver. The transmitter module takes the input signal in the electrical form and then transforms it into optical (light) energy containing the same information. The optical fiber is the medium, which takes the energy to the receiver. At the receiver light is converted back into the electrical form with the same pattern as originally fed to the transmitter.

Transmitter:

Fiber optic transmitters are typically composed of a buffer, driver and optical source. The buffer provides both an electrical connection and isolation between the transmitter & the electrical system supplying the data. The driver provides electrical power to the optical source. Finally, the optical source converts the electrical current to the light energy with the same pattern. Commonly used optical sources are light emitting diode (LEDs) and Laser Beam. Simple LED circuits, for digital and analog transmissions are shown below.

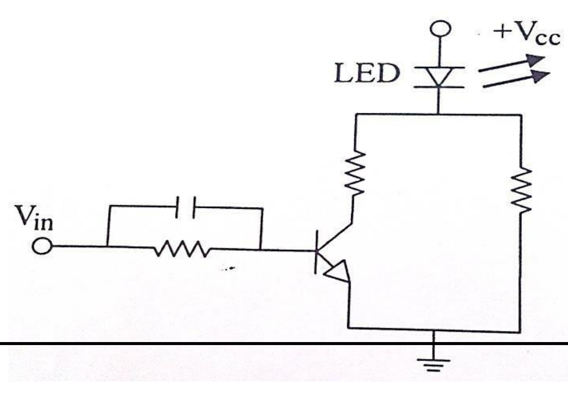
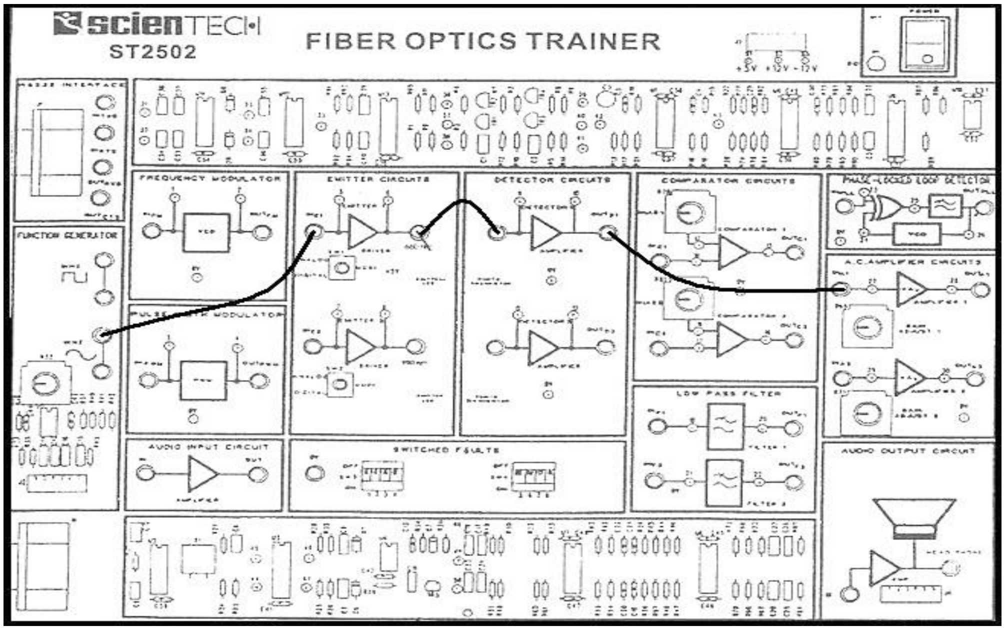


Figure shows a simple drive circuit for the binary digital transmission consisting a common emitter-saturating switch.



Procedure:

1. Connect the power supply to the board.
2. Ensure that all switched faults are off.
3. Make the following connections. (As shown in the below figure)
    a. Connect the F.G. 1 KHz square wave output to the emitter's input.
    b. Connect the F.O. cable between emitter output and detector's input.
    c. Detector Output to Comparator's input.
    d. Comparators output to the AC Amplifier input.
4. On the board, switch emitter's driver to digital mode.
5. Switch ON the power.
6. Monitor both the inputs to the comparator (t.p. 9 & 10). Slowly adjust the comparators bias preset, until DC Level on the input (t.p. 9) lies mid way between the high and the low levelof the signal on the positive input (t.p.11).
7. Observe the input to emitter (t.p. 5) with the output from AC Amplifier (t.p. 19) and notethat the two signals are same.

Conclusion:

After performing this practical, we can study about digital link and the relationship between the analog input signal and received signal.

Experiment-9(A)

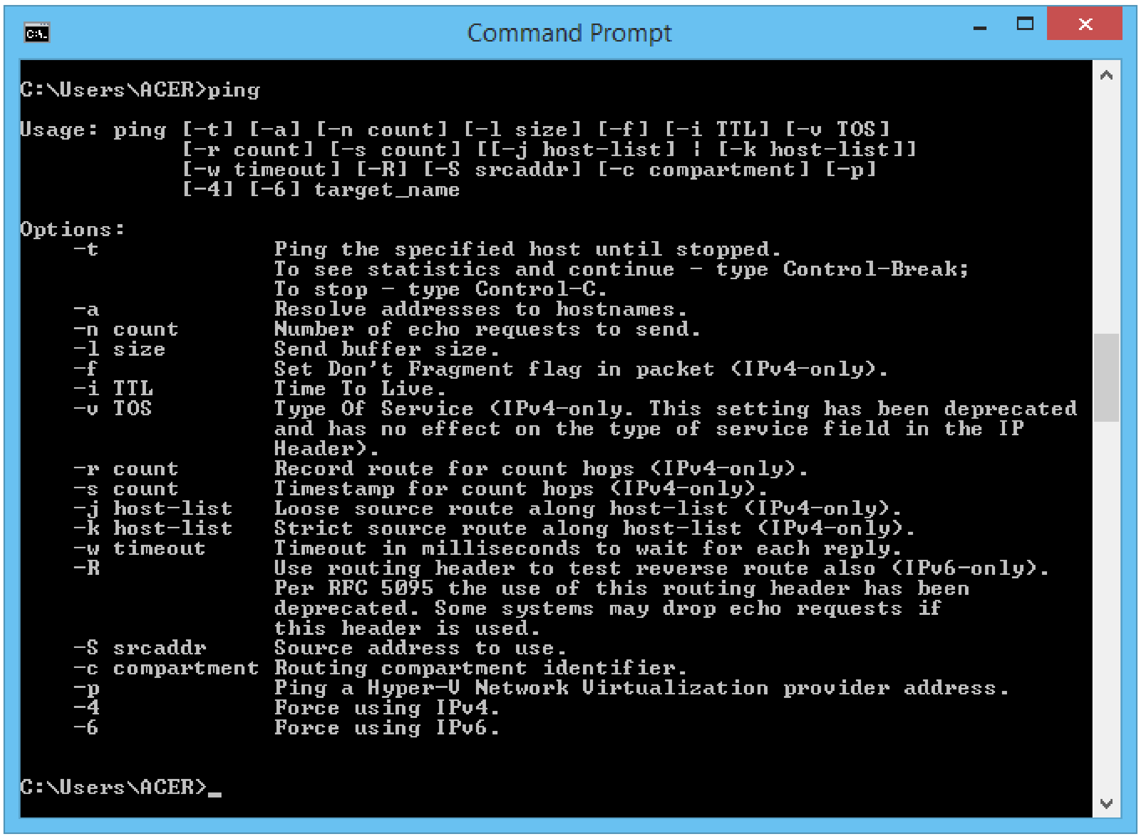
Aim: Introduction to different Network Commands in DOS.

Theory:

Important Network commands:

1. ping:

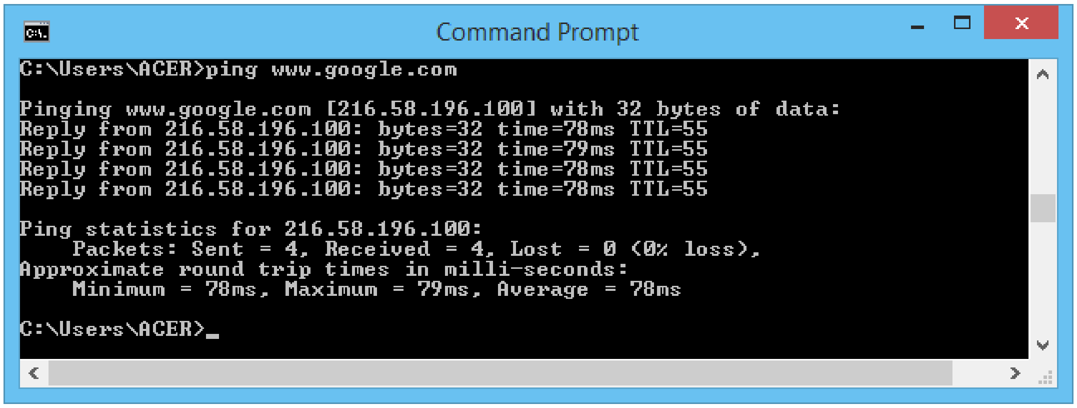
In computer networking, ping is a process for sending test messages from one computer to another to check the health of network connections. Ping tests are run using various software commands and utility programs.



Typically, when you issue the ping command, one of four things will happen, each of which has its own meaning:

The first thing that can happen is that the specified machine will produce four replies. This indicates that the workstation is able to communicate with the specified host at the TCP/IP level.

The second thing that can happen is that all four requests time out, as shown in Figure B. If you look at Figure A, you will notice that each response ends in TTL=128. TTL stands for "time to live." What this means is that each of the four queries and responses must be completed within 128 milliseconds. The TTL is also decremented once for each hop on the way back. A hop occurs when a packet moves from one network to another. I will be talking a lot more about hops later on in this series.



At any rate, if all four requests have timed out, it means that the TTL expired before the reply was received. This can mean one of three things:

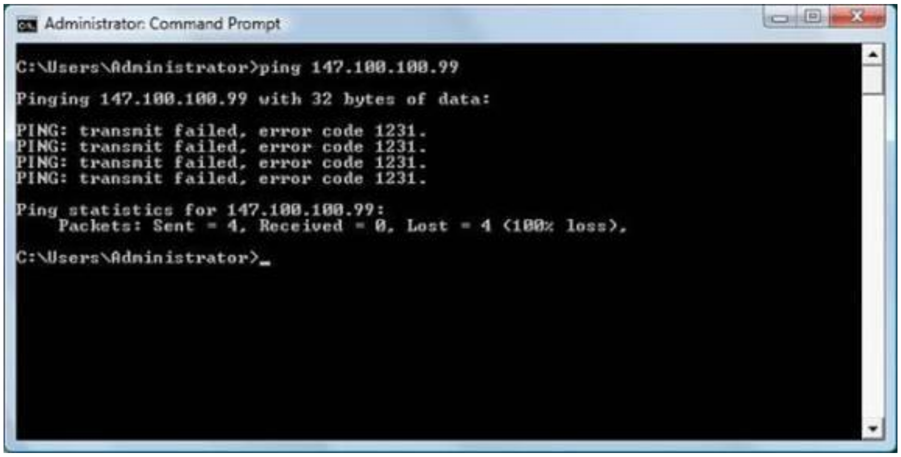
Communication problems are preventing packets from flowing between the two machines: This could be caused by a disconnected cable, a bad routing table, or a number of other issues.

Communications are occurring, but are too slow for ping to acknowledge: This can be caused by extreme network congestion, or by faulty network hardware or wiring.

Communications are functional, but a firewall is blocking ICMP traffic: Ping will not work unless the destination machine's firewall (and any firewalls between the two machines) allow ICMP echoes.

A third thing that can happen when you enter the ping command is that some replies are received, while others time out. This can point to bad network cabling, faulty hardware, or extreme network congestion.

The fourth thing that can occur when pinging a host is that you receive an error similar to the one that is shown in Figure.



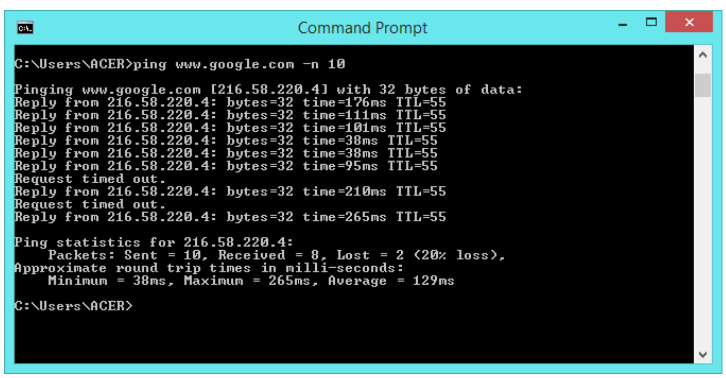
Error message indicating that TCP/IP is not configured correctly.

Commands:

ping xxx.xxx.xx.xx -t

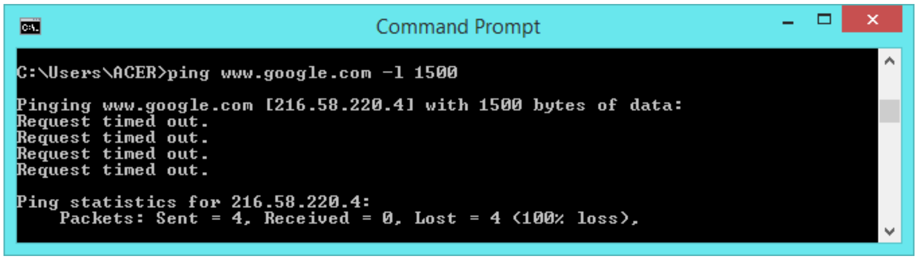
Continuous Ping - It will keep on pinging forever until you hit Ctrl + C to stop it. This can be great for troubleshooting intermittent connections. Just open up a Command Prompt (or 3) and run the continuous ping command on a site like Google.com to see when you're dropping packets. I would also run one on your router and on another site like site.com for a better data sample control group.

ping xxx.xxx.xx.xx -n 10



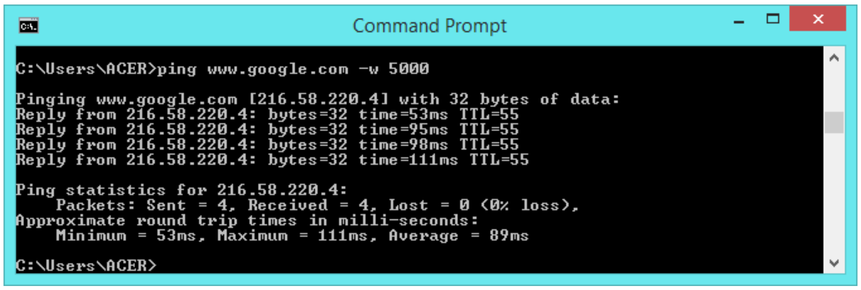
Number of Pings - The N Switch is simply for setitng the number of pings. By default the ping cmd sends out 4 packets at 32 bytes each.

ping xxx.xxx.xx.xx -l 1500



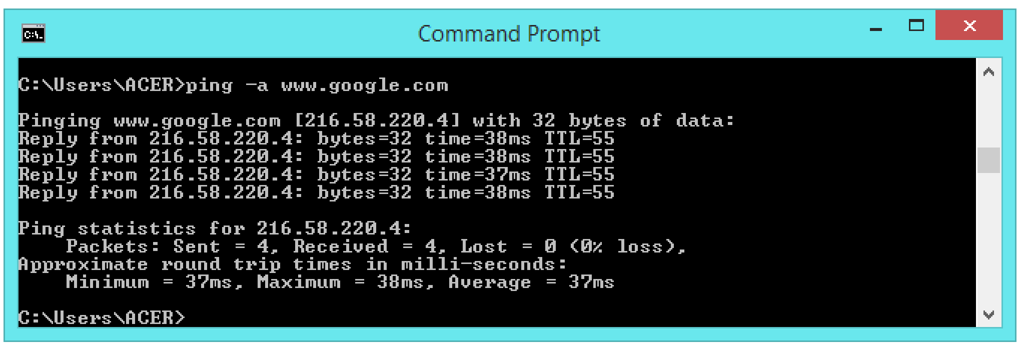
Size of Packet - By default the packets sent are a small 32 bytes. You can set your own size up to the max 65500 bytes. This can really help for stress testing your local network.

ping xxx.xxx.xx.xx -w 5000



Time Out - This is in milliseconds. The timeout by default is 4,000 milliseconds which amounts to 4 minutes. Just seeing if you were still paying attention. It really is only 4 seconds.

ping -a xxx.xxx.xx.xx



Resolves Hostname Address - This is a great one if you are helping someone else and need to find out what router model they are using. You can resolve the host of an IP Address with this command. Try pinging your router or your local computer with it. ping -a 127.0.0.1. Not this switch will only work in front of the IP Address.

2. ipconfig:

Is used to find out your current TCP/IP settings. With IPCONFIG you can find out your IP Address, find your Default Gateway and find your Subnet Mask. This is a very handy network tool for finding your local IP address.

Commands:

ipconfig /all

To display all your IP information for all adapters. With ipconfig /all you can also find out your DNS Server and MAC Address. This will show your full TCP/IP configuration for all adapters on your Windows machine. You can find out your own IP Address as well as your default gateway.

ipconfig /release

To release your current IP information and obtain a new IP Address from the DHCP server.

ipconfig /renew

Used to renew your IP Address if you have it set to obtain IP Address automatically.

ipconfig /displaydns

This shows your current DNS Resolver Cache Logs.

ipconfig /flushdns

This flushes or clears your current DNS Resolver Cache Logs.

ipconfig /registerdns

The register DNS command updates the DNS settings on the Windows computer. It doesn't just access the local DNS cache, it initiates communication with the DNS server and the DHCP server so it can re-register the network address. You can use this for troubleshooting problems with connection to the ISP (Internet Service Provider), like failing to obtain a dynamic IP address from the DHCP Server or failing to connect to the ISP DNS server.

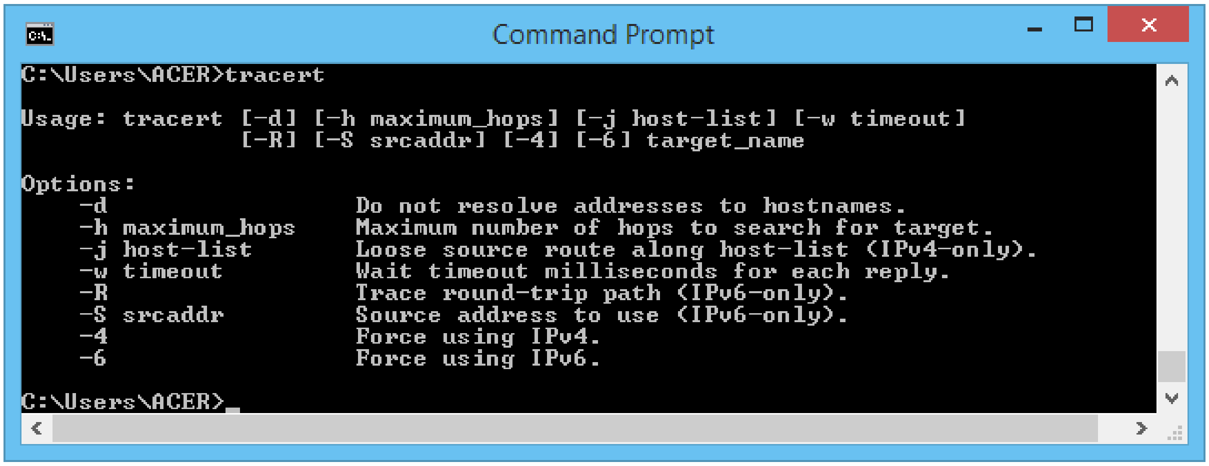
If you ever wonder "what your IP Address is" you would run an ipconfig as shown above. If you need to find your IP address, default gateway(router login) or subnet mask ip config is the tool to use. These numbers can be very helpful when trouble shooting your local network connection. If you have changed your settings but they are not taking place you may try a ipconfig release and renew. If you're having problems resolving to a website you may try flushing your DNS Resolver Logs.

```
C:\Users\ACER>ipconfig /?

USAGE:
    ipconfig [/allcompartments] [/? | /all |
                                 /renew [adapter] | /release [adapter] |
                                 /renew6 [adapter] | /release6 [adapter] |
                                 /flushdns | /displaydns | /registerdns |
                                 /showclassid adapter |
                                 /setclassid adapter [classid] |
                                 /showclassid6 adapter |
                                 /setclassid6 adapter [classid] ]

where
    adapter             Connection name
                       (wildcard characters * and ? allowed, see examples)

    Options:
       /?               Display this help message
       /all             Display full configuration information.
       /release         Release the IPv4 address for the specified adapter.
       /release6        Release the IPv6 address for the specified adapter.
       /renew           Renew the IPv4 address for the specified adapter.
       /renew6          Renew the IPv6 address for the specified adapter.
       /flushdns        Purges the DNS Resolver cache.
       /registerdns     Refreshes all DHCP leases and re-registers DNS names
       /displaydns      Display the contents of the DNS Resolver Cache.
       /showclassid     Displays all the dhcp class IDs allowed for adapter.
       /setclassid      Modifies the dhcp class id.
       /showclassid6    Displays all the IPv6 DHCP class IDs allowed for adapter
.
       /setclassid6     Modifies the IPv6 DHCP class id.


The default is to display only the IP address, subnet mask and
default gateway for each adapter bound to TCP/IP.

For Release and Renew, if no adapter name is specified, then the IP address
leases for all adapters bound to TCP/IP will be released or renewed.

For Setclassid and Setclassid6, if no ClassId is specified, then the ClassId is
removed.

Examples:
    > ipconfig                       ... Show information
    > ipconfig /all                  ... Show detailed information
    > ipconfig /renew                ... renew all adapters
    > ipconfig /renew EL*            ... renew any connection that has its
                                         name starting with EL
    > ipconfig /release *Con*        ... release all matching connections,
                                         eg. "Wired Ethernet Connection 1" or
                                             "Wired Ethernet Connection 2"
    > ipconfig /allcompartments      ... Show information about all
                                         compartments
    > ipconfig /allcompartments /all ... Show detailed information about all
                                         compartments

C:\Users\ACER>_
```

```
C:\Users\ACER>ipconfig

Windows IP Configuration


Wireless LAN adapter Local Area Connection* 13:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Wireless LAN adapter Local Area Connection* 2:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Ethernet adapter Ethernet:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Wireless LAN adapter Wi-Fi:

   Connection-specific DNS Suffix  . :
   Link-local IPv6 Address . . . . . : fe80::d084:3e0f:941f:c21e%3
   IPv4 Address. . . . . . . . . . . : 192.168.1.103
   Subnet Mask . . . . . . . . . . . : 255.255.255.0
   Default Gateway . . . . . . . . . : 192.168.1.1

Ethernet adapter VMware Network Adapter VMnet1:

   Connection-specific DNS Suffix  . :
   Link-local IPv6 Address . . . . . : fe80::1d40:109f:353f:afb4%21
   IPv4 Address. . . . . . . . . . . : 192.168.18.1
   Subnet Mask . . . . . . . . . . . : 255.255.255.0
   Default Gateway . . . . . . . . . :

Ethernet adapter VMware Network Adapter VMnet8:

   Connection-specific DNS Suffix  . :
   Link-local IPv6 Address . . . . . : fe80::98d:71e8:4d77:aca7%22
   IPv4 Address. . . . . . . . . . . : 192.168.22.1
   Subnet Mask . . . . . . . . . . . : 255.255.255.0
   Default Gateway . . . . . . . . . :

Tunnel adapter isatap.{CDCF685A-AB3A-4D69-AD32-D0FA182191CC}:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Tunnel adapter isatap.{EA947DBC-3CA3-47B8-9F24-C91E04909123}:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Tunnel adapter isatap.{59C47111-B4F5-4EE6-B339-7112AD4784DD}:

   Media State . . . . . . . . . . . : Media disconnected
```

3. tracert:

Traceroute is a utility that records the route through the Internet between your computer and a specified destination computer. It also calculates and displays the amount of time each hop took. Traceroute is a handy tool both for understanding where problems are in the Internet network and for getting a detailed sense of the Internet itself. Another utility, PING, is often used prior to using traceroute to see whether a host is present on the network.



tracert site.com

With Trace route you can trace the path your packets take across the internet from you to your destination. Along the way you can determine the time from hop to hop. You can identify server problems and latency with this tool. It really helps see where the failure is between you and a destination.

4. netstat:

In computing, netstat (network statistics) is a command-line tool that displays network connections for the Transmission Control Protocol (both incoming and outgoing), routing tables, and a number of network interface and network protocol statistics.

```
Command Prompt
C:\Users\ACER>netstat /?

Displays protocol statistics and current TCP/IP network connections.

NETSTAT [-a] [-b] [-e] [-f] [-n] [-o] [-p proto] [-r] [-s] [-x] [-t] [interval]

  -a            Displays all connections and listening ports.
  -b            Displays the executable involved in creating each connection or
                listening port. In some cases well-known executables host
                multiple independent components, and in these cases the
                sequence of components involved in creating the connection
                or listening port is displayed. In this case the executable
                name is in [] at the bottom, on top is the component it called,
                and so forth until TCP/IP was reached. Note that this option
                can be time-consuming and will fail unless you have sufficient
                permissions.
  -e            Displays Ethernet statistics. This may be combined with the -s
                option.
  -f            Displays Fully Qualified Domain Names (FQDN) for foreign
                addresses.
  -n            Displays addresses and port numbers in numerical form.
  -o            Displays the owning process ID associated with each connection.
  -p proto      Shows connections for the protocol specified by proto; proto
                may be any of: TCP, UDP, TCPv6, or UDPv6.  If used with the -s
                option to display per-protocol statistics, proto may be any of:
                IP, IPv6, ICMP, ICMPv6, TCP, TCPv6, UDP, or UDPv6.
  -r            Displays the routing table.
  -s            Displays per-protocol statistics.  By default, statistics are
                shown for IP, IPv6, ICMP, ICMPv6, TCP, TCPv6, UDP, and UDPv6;
                the -p option may be used to specify a subset of the default.
  -t            Displays the current connection offload state.
  -x            Displays NetworkDirect connections, listeners, and shared
                endpoints.
  -y            Displays the TCP connection template for all connections.
                Cannot be combined with the other options.
  interval      Redisplays selected statistics, pausing interval seconds
                between each display.  Press CTRL+C to stop redisplaying
                statistics.  If omitted, netstat will print the current
                configuration information once.
```

```
Command Prompt
C:\Users\ACER>netstat

Active Connections

  Proto  Local Address          Foreign Address        State
  TCP    127.0.0.1:4573         ACER-PC:54329          ESTABLISHED
  TCP    127.0.0.1:49192        ACER-PC:65001          ESTABLISHED
  TCP    127.0.0.1:54329        ACER-PC:4573           ESTABLISHED
  TCP    127.0.0.1:65001        ACER-PC:49192          ESTABLISHED

C:\Users\ACER>netstat
```

-a: Displays all connections and listening ports.

-b: Displays the executable involved in creating each connection or listening port. In some cases well-known executables host multiple independent components, and in these cases the sequence of components involved in creating the connection or listening port is displayed. In this case the executable name is at the bottom, on top is the component it called, and so forth until TCP/IP was reached. Note that this option can be time-consuming and will fail unless you have sufficient permissions.

-e: Displays Ethernet statistics. This may be combined with the -s option.

-n: Displays addresses and port numbers in numerical form.

-o: Displays the owning process ID associated with each connection.

-p proto: Shows connections for the protocol specified by proto; proto may be any of: TCP, UDP, TCPv6, or UDPv6. If used with the -s option to display per-protocol statistics, proto may be any of: IP, IPv6, ICMP, ICMPv6, TCP, TCPv6, UDP, or UDPv6.

-r: Displays the routing table.

-s: Displays per-protocol statistics. By default, statistics are shown for IP, IPv6, ICMP, ICMPv6, TCP, TCPv6, UDP, and UDPv6; the -p option may be used to specify a subset of the default.

-v: When used in conjunction with -b, will display sequence of components involved in creating the connection or listening port for all executables.

Interval: Redisplays selected statistics, pausing interval seconds between each display. Press CTRL+C to stop redisplaying statistics. If omitted, netstat will print the current configuration information once.

5. arp:

ARP stands for Address Resolution Protocol. When you try to ping an IP address on your local network, say 192.168.1.1, your system has to turn the IP address 192.168.1.1 into a MAC address. This involves using ARP to resolve the address, hence its name.

Systems keep an ARP look-up table where they store information about what IP addresses are associated with what MAC addresses. When trying to send a packet to an IP address, the system will first consult this table to see if it already knows the MAC address. If there is a value cached, ARP is not used.

If the IP address is not found in the ARP table, the system will then send a broadcast packet to the network using the ARP protocol to ask "who has 192.168.1.1". Because it is a broadcast packet, it is sent to a special MAC address that causes all machines on the network to receive it. Any machine with the requested IP address will reply with an ARP packet that says "I am 192.168.1.1", and this includes the MAC address which can receive packets for that IP.

```
C:\Users\ACER>arp

Displays and modifies the IP-to-Physical address translation tables used by
address resolution protocol (ARP).

ARP -s inet_addr eth_addr [if_addr]
ARP -d inet_addr [if_addr]
ARP -a [inet_addr] [-N if_addr] [-v]

  -a            Displays current ARP entries by interrogating the current
                protocol data.  If inet_addr is specified, the IP and Physical
                addresses for only the specified computer are displayed.  If
                more than one network interface uses ARP, entries for each ARP
                table are displayed.
  -g            Same as -a.
  -v            Displays current ARP entries in verbose mode.  All invalid
                entries and entries on the loop-back interface will be shown.
  inet_addr     Specifies an internet address.
  -N if_addr    Displays the ARP entries for the network interface specified
                by if_addr.
  -d            Deletes the host specified by inet_addr. inet_addr may be
                wildcarded with * to delete all hosts.
  -s            Adds the host and associates the Internet address inet_addr
                with the Physical address eth_addr.  The Physical address is
                given as 6 hexadecimal bytes separated by hyphens. The entry
                is permanent.
  eth_addr      Specifies a physical address.
  if_addr       If present, this specifies the Internet address of the
                interface whose address translation table should be modified.
                If not present, the first applicable interface will be used.
Example:
  > arp -s 157.55.85.212   00-aa-00-62-c6-09  .... Adds a static entry.
  > arp -a                                    .... Displays the arp table.
```

```
C:\Users\ACER>arp -a

Interface: 192.168.1.102 --- 0x3
  Internet Address      Physical Address      Type
  192.168.1.1           c0-4a-00-c6-01-e4     dynamic
  192.168.1.255         ff-ff-ff-ff-ff-ff     static
  224.0.0.2             01-00-5e-00-00-02     static
  224.0.0.22            01-00-5e-00-00-16     static
  224.0.0.251           01-00-5e-00-00-fb     static
  224.0.0.252           01-00-5e-00-00-fc     static
  239.255.255.250       01-00-5e-7f-ff-fa     static
  255.255.255.255       ff-ff-ff-ff-ff-ff     static

Interface: 192.168.18.1 --- 0x15
  Internet Address      Physical Address      Type
  192.168.18.255        ff-ff-ff-ff-ff-ff     static
  224.0.0.2             01-00-5e-00-00-02     static
  224.0.0.22            01-00-5e-00-00-16     static
  224.0.0.251           01-00-5e-00-00-fb     static
  224.0.0.252           01-00-5e-00-00-fc     static

Interface: 192.168.22.1 --- 0x16
  Internet Address      Physical Address      Type
  192.168.22.255        ff-ff-ff-ff-ff-ff     static
  224.0.0.2             01-00-5e-00-00-02     static
  224.0.0.22            01-00-5e-00-00-16     static
  224.0.0.251           01-00-5e-00-00-fb     static
  224.0.0.252           01-00-5e-00-00-fc     static
```

Conclusion: By this practical, various networking commands are studied and seen them practically how it works.

Experiment-9(B)

Aim: Introduction to Packet Tracer.

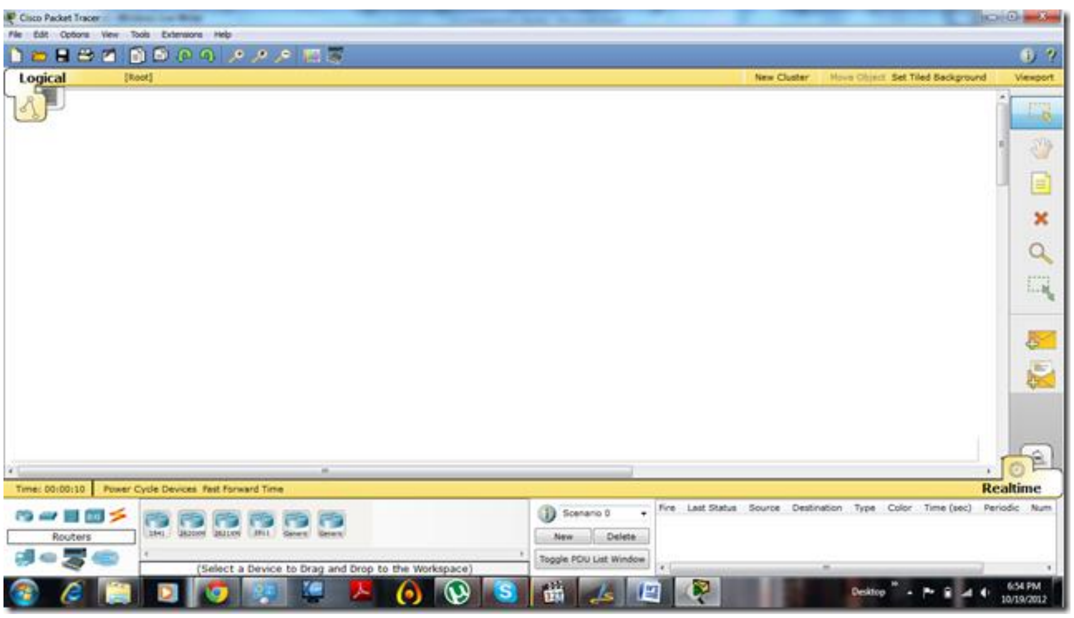
Theory:

Cisco Packet Tracer 6.1:

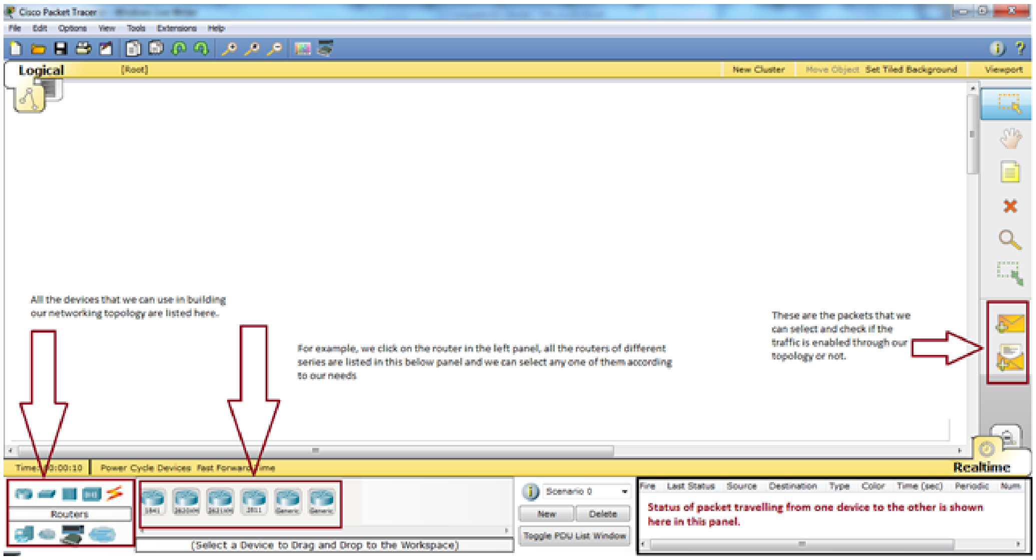
Cisco Packet Tracer 6.1 is a utility that can be used by students to train themselves in CCNA and CCNP by permitting them to create networks by utilizing unlimited number of devices. By using this application you don’t need to buy Cisco routers for experiencing troubleshooting. Cisco Packet Tracer 6.1 has different capabilities of visualization, authoring and collaboration and provides facilities for learning some complicated technological concepts.

Packet Tracer supplements physical equipment in the classroom by allowing students to create a network with an almost unlimited number of devices, encouraging practice, discovery, and troubleshooting. The simulation-based learning environment helps students develop 21st century skills such as decision making, creative and critical thinking, and problem solving. Packet Tracer complements the Networking Academy curricula, allowing instructors to easily teach and demonstrate complex technical concepts and networking systems design.

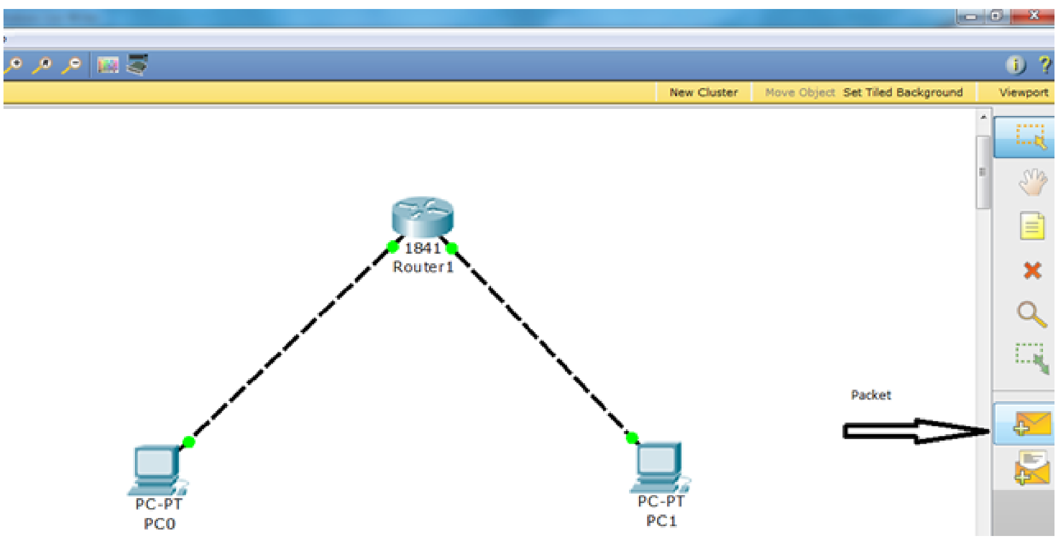
Here how it looks like after we start it:



We have different modules and panels available in the packet tracer. Some important modules, which are important to understand for the working in Packet Tracer, are mentioned in the following diagram.



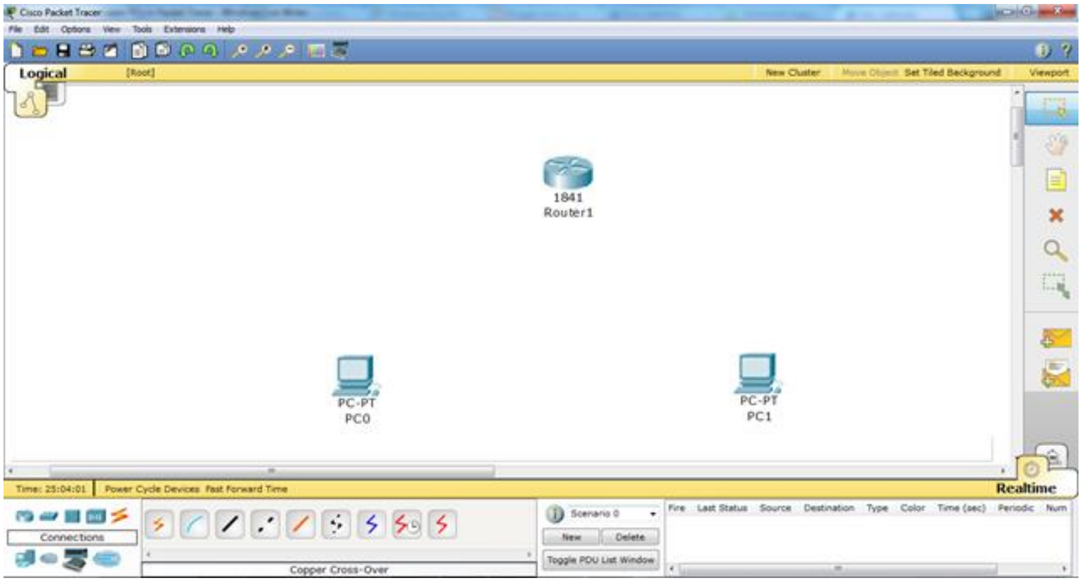
Now, in order to create a topology, we will have to select some of the devices and put them in our main window i.e. the white portion of packet tracer. and here how it looks after we add the devices.



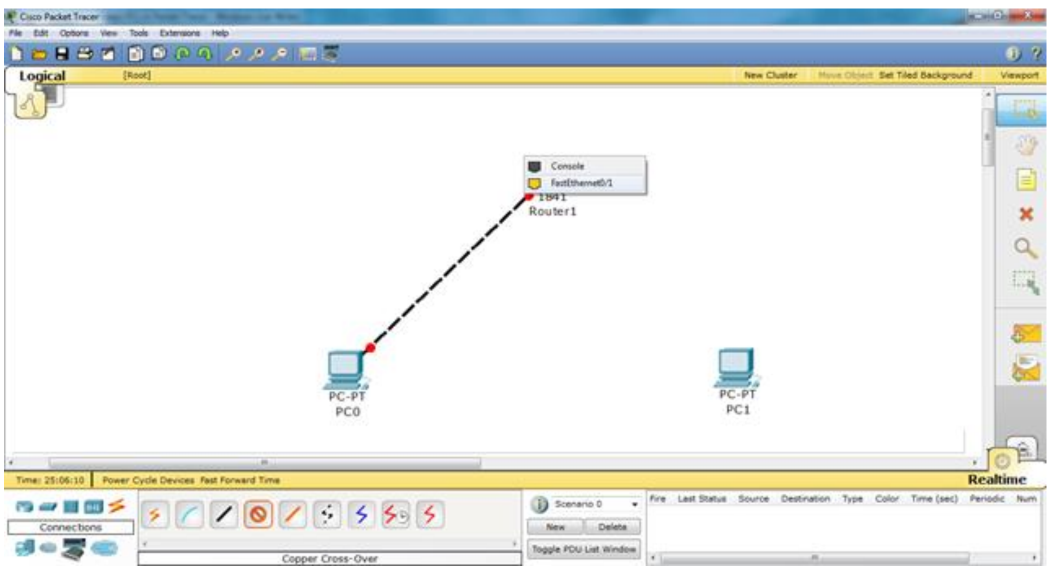
Here, we will see communication enabled between PCs via Router in Packet Tracer.

So, for this we need two PCs, a router ,and two cross over cables to connect them.

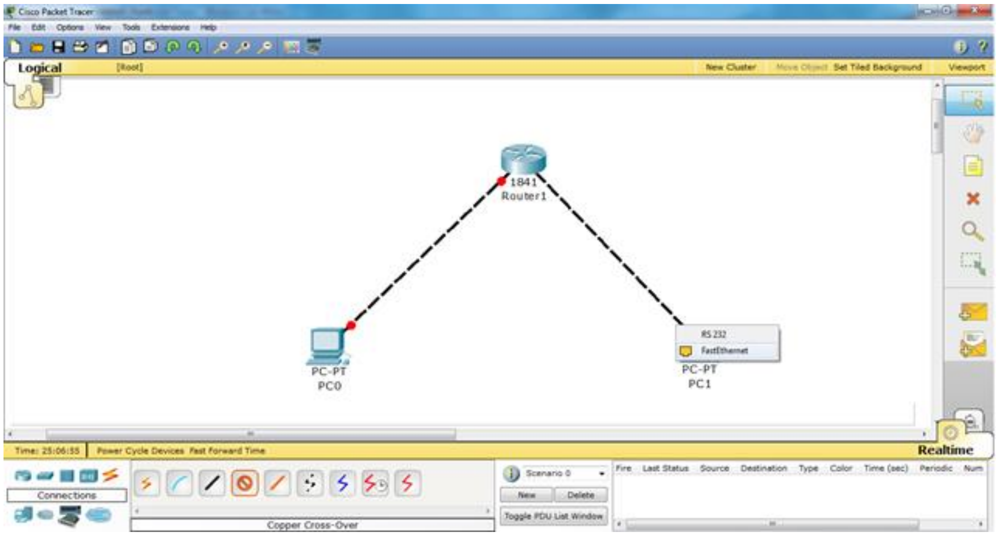
Important point is that we use cross over cable to connect PC to a router because they both use the same pins for transmission and receiving of data.



Now, we will connect them by selecting fast ethenet interfaces on both ends.



Similarly, on the PC side we will select fast Ethernet interface.



Now, we have connect the devices. Further, we will go to the router CLI mode and enter the following commands.

Step by step, we will have to do the following things.

i. Access the interfaces one by one

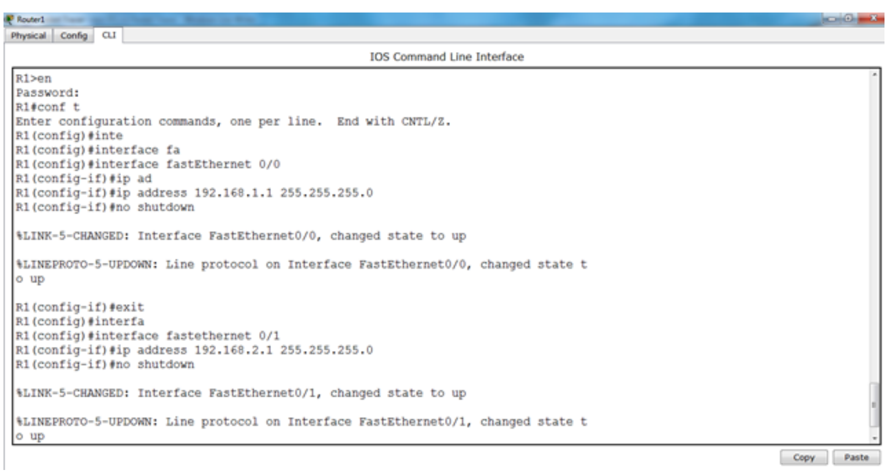
ii. Assign IP addresses to interfaces

iii. Change the status of the interfaces i.e. from Down to Up.

iv. Assign IP addresses to PCs.

v. Assign Default GateWay to PCs. FYI fast ethernet ip address is the gateway address to the PC.

Now, commands of the Router CLI mode are as follows,

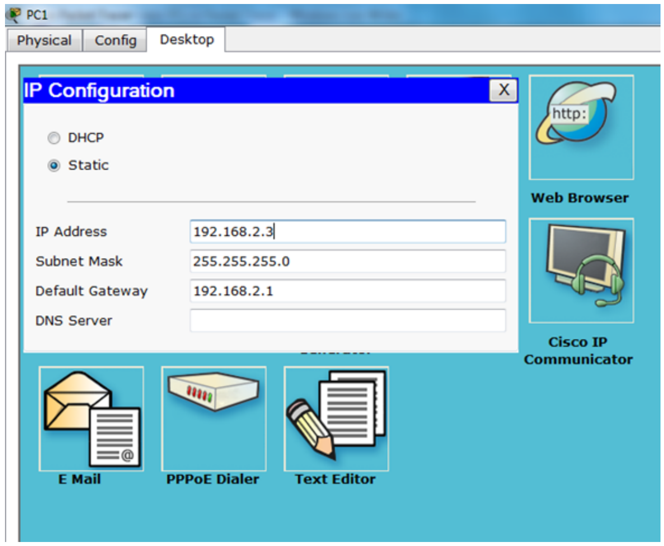


See the difference the lights have changed the color from Red to Green :)

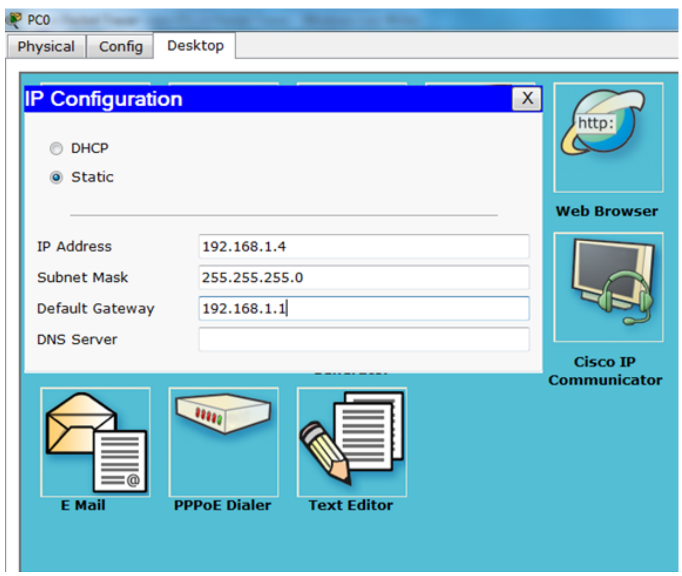
Now, lets assign IP addresses to the PCs.

Click on PC1, go to Desktop, then click IP Configuration.

PC1:

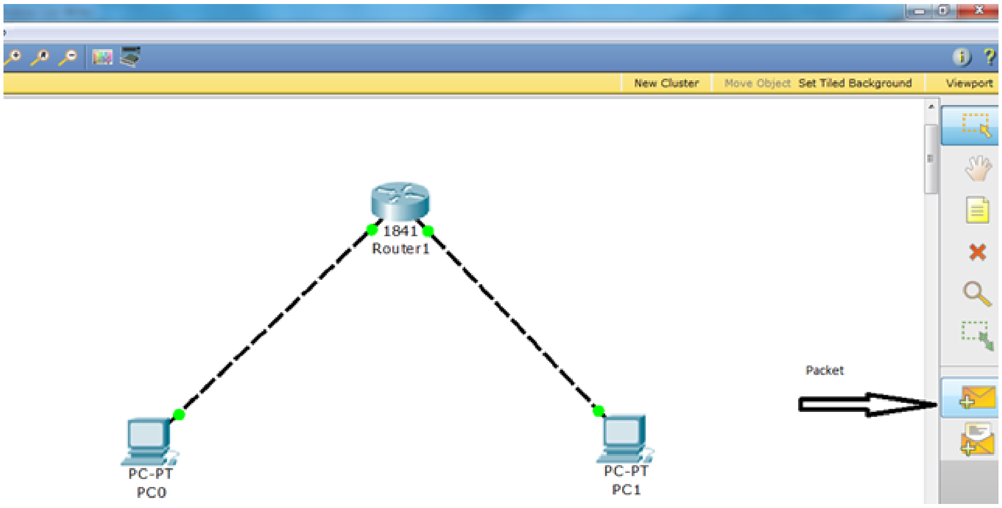


PC2:



Now, our communication is enabled and we are able to communicate from PC1 to PC2 via Router.

Click on the packet in the right panel on the packet tracer, then click on PC1 and then click on PC2. You will see the successful packet tracer (status is shown in the bottom right corner).



Your communication is successful.

Conclusion: By this practical, it is seen that we can establish virtual connections using packet tracer and check the connectivity for the same. Also, we can rectify the errors in this software so that it cannot repeat in physical connection.

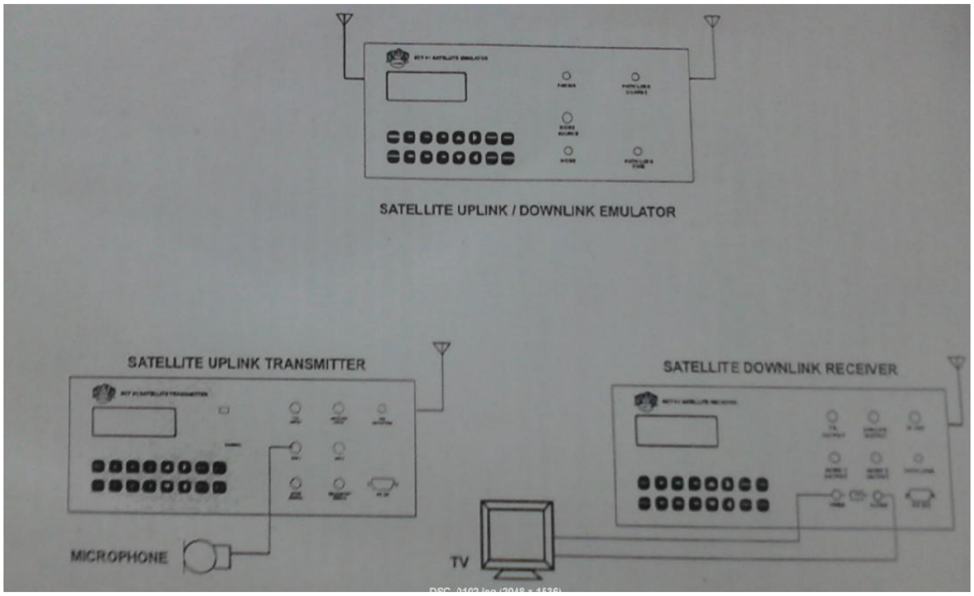
EXPERIMENT-10(A)

Aim:

To set up an Audio link.

Apparatus:

Satellite trainer kit(SCT 01) consisting of transmitter, receiver and emulator modules, RHCP & LHCP axial helix antenna.



Procedure:

- Setup the transmitter, receiver and emulator modules in a triangular manner imitating an active satellite link.
- Connect transmitter to RHCP helix antenna and receiver to LHCP helix antenna.
- In the emulator module connect the receiver LHCP helix antenna and transmitter to RHCP helix antenna.
- Select the 2481MHz frequency in Transmitter for uplink, 2400MHz in Receiver for downlink and in Emulator 2400MHz downlink frequency and 2481MHz uplink frequency.
- Set the transmitter and receiver as follows:
- Transmitter:
  - Uplink frequency=2481MHz
  - Noise and path loss knob fully anti clockwise.
  - Input Channel: CH1- MIC1 CH2- OFF CH3 –OFF

- Receiver:

    - Downlink frequency=2400MHz
    - Output Channel: CH1- MIC1
    - CH2- OFF
    - CH3- VIDEO
- Emulator:
    - Uplink frequency=2481MHz
    - Downlink frequency=2400MHz

- Now select a particular uplink and downlink frequency out of the four given frequencies given in transmitter, receiver and emulator.

  NOTE:- UPLINK FREQ. > DOWNLINK FREQ.
- Setup the link in a TRIANGULAR fashion with Transmitter, Receiver and Satellite link emulator at 3 vertices of a triangular. Make sure that RHCP Helix antenna of Transmitter should point towards RHCP Helix antenna of uplink satellite link emulator and LHCP Helix antenna of downlink satellite link emulator. Set the distances between antennas to approx.3 meters.
- Connect the speaker to the Audio post of the Downlink Receiver. Speak out on the microphone in MIC 1 post all the Transmitter side and try to listen it on the TV Speaker at the Audio post on the Receiver, a successful satellite link is said to be established. This is a satellite link using active satellite link emulator.
- In case of passive satellite, no frequency translation and power amplification takes place. Set Transmitter & Receiver at same frequency and switch off the satellite emulator. Point the Transmitter and Receiver antennas towards the reflector sheet in same triangular fashion as explained above. The only difference being that instead of satellite there is a reflection from wall. The transmitted signal is reflected back to receiver without the power being increased and frequency remaining the same. Here, the reflecting surface is functioning like a passive satellite.
- Up linking to a satellite is normally carried out at a higher frequency because of narrow beam width, for pinpointing distance satellites, at higher frequency. There are two up linking frequency channels 2.481 GHz and 2.454 GHz.
- The satellite link emulator consists of transponder(transmit-receive pair). It receive frequency in 2.4-2.5 GHz band and has the capability to retransmit after amplification in 2.4-2.5GHz band. It can be set to receive at one particular frequency and transmit at same different frequency.

- Down linking from a satellite is carried out at lower frequencies because wider beam width gives more footprint coverage. There are two down linking frequency channels 2.4 GHz and 2.427 GHz.
- Repeat the experiment by selecting a different up linking and down linking channel frequencies.

Conclusion:

A clear sound at the receiver indicates that a microwave satellite communication link has been set up successfully.

EXPERIMENT-10(B)

Aim: To set up an Digital link.

Apparatus:

Satellite trainer kit(SCT 01) consisting of transmitter, receiver and emulator modules, RHCP & LHCP axial helix antenna.

Procedure:

- Setup the transmitter, receiver and emulator modules in a triangular manner imitating an active satellite link.
- Connect transmitter to RHCP helix antenna and receiver to LHCP helix antenna.
- In the emulator module connect the receiver LHCP helix antenna and transmitter to RHCP helix antenna.
- Select the 2481MHz frequency in Transmitter for uplink, 2400MHz in Receiver for downlink and in Emulator 2400MHz downlink frequency and 2481MHz uplink frequency.
- Set the transmitter and receiver as follows:Transmitter:
    - Uplink frequency=2481MHz
    - Noise and path loss knob fully anti clockwise.
    - Input Channel:

      CH1- 0FF

      CH2- OFF

      CH3 –TTL

  Receiver:
    - Downlink frequency=2400MHz
    - Output Channel: CH1- OFF
    - CH2- OFF CH3- TTL

  Emulator:
    - Uplink frequency=2481MHz
    - Downlink frequency=2400MHz

- Now select a particular uplink and downlink frequency out of the four given frequencies given in transmitter, receiver and emulator.

NOTE:- UPLINK FREQ. > DOWNLINK FREQ.

- Setup the link in a TRIANGULAR fashion with Transmitter, Receiver and Satellite link emulator at
- 3 vertices of a triangular. Make sure that RHCP Helix antenna of Transmitter should point towards RHCP Helix antenna of uplink satellite link emulator and LHCP Helix antenna of downlink satellite link emulator. Set the distances between antennas to approx.3 meters.
- Connect the speaker to the Audio post of the Downlink Receiver. Speak out on the microphone in MIC 1 post all the Transmitter side and try to listen it on the TV Speaker at the Audio post on the Receiver, a successful satellite link is said to be established. This is a satellite link using active satellite link emulator.
- In case of passive satellite, no frequency translation and power amplification takes place. Set Transmitter & Receiver at same frequency and switch off the satellite emulator. Point the
- Transmitter and Receiver antennas towards the reflector sheet in same triangular fashion as explained above. The only difference being that instead of satellite there is a reflection from wall. The transmitted signal is reflected back to receiver without the power being increased and frequency remaining the same. Here, the reflecting surface is functioning like a passive satellite.
- Up linking to a satellite is normally carried out at a higher frequency because of narrow beam width, for pinpointing distance satellites, at higher frequency. There are two up linking frequency channels 2.481 GHz and 2.454 GHz.
- The satellite link emulator consists of transponder(transmit-receive pair). It receive frequency in
- 2.4-2.5 GHz band and has the capability to retransmit after amplification in 2.4-2.5GHz band. It can be set to receive at one particular frequency and transmit at same different frequency.
- Down linking from a satellite is carried out at lower frequencies because wider beam width gives
- more footprint coverage. There are two down linking frequency channels 2.4 GHz and 2.427 GHz.
- Repeat the experiment by selecting a different up linking and down linking channel frequencies.

Conclusion:

The Function Generator TTL O/P waveforms can be transmitted over a distance via a satellite communication link and same TTL O/P waveform can be received at Receiver input.

EXPERIMENT: 11

Aim: To verify AT commands and introduction to GSM trainer kit ST-2133.

Apparatus: GSM trainer kit ST-2133, RS232 cable, Active SIM card, Power supply

Theory:

GSM

The Global System for Mobile Communications (GSM) is an international digital cellular telecommunications standard. The GSM standard was released by ETSI (European Standard and Technology Institute) back in 1989. First commercial services were launched in 1991. After its early introduction in Europe, the standard went global in 1992 when GSM services were introduced in Australia. Since then, GSM has become the most widely adopted and fastest-growing digital cellular standard, and it is positioned to become the world's dominant cellular standard. In fact, as of January 1999, GSM accounted for more than 120 million subscribers, according to the GSM memorandum of understanding (MoU) Association. With 324 GSM networks in operation in 129 countries, GSM provides almost complete coverage around the globe.

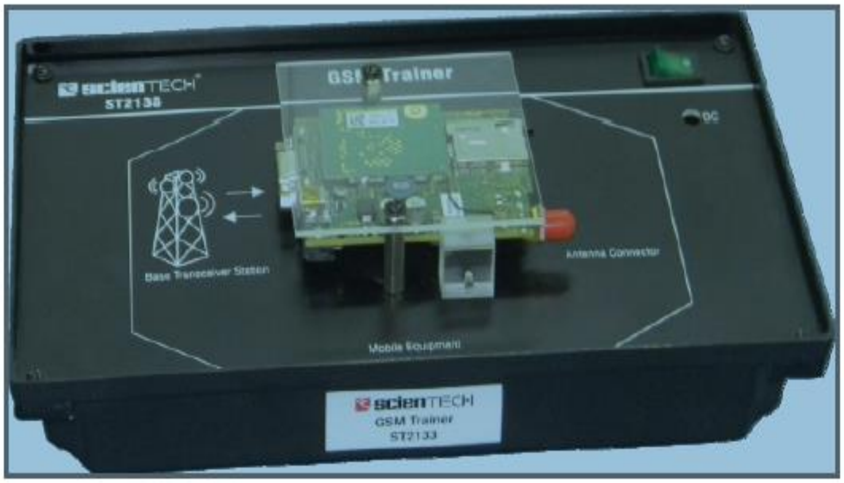
AT commands

AT commands are instructions used to control a modem. AT is the abbreviation of ATtention. Every command line starts with "AT" or "at". That's why modem commands are called AT commands. Many of the commands that are used to control wired dial-up modems, such as ATD (Dial), ATA (Answer), ATH (Hook control) and ATO (Return to online data state), are also supported by GSM/GPRS modems and mobile phones. Besides this common AT command set, GSM/GPRS modems and mobile phones support an AT command set that is specific to the GSM technology, which includes SMS-related commands like AT+CMGS (Send SMS message), AT+CMSS (Send SMS message from storage), AT+CMGL (List SMS messages) and AT+CMGR (Read SMS messages).

GSM Trainer ST-2133

The GSM Trainer ST-2133 is a modem or mobile equipment for transmission of voice and data calls as well as SMS (Short Message Service) in GSM

Network. To control the GSM modem there is an advanced set of AT commands according to GSM ETSI (European Telecommunications Standards Institute) 07.07 and 07.05 implemented. The GSM standard has established itself across continents. The trainer is well suited for studying AT commands by camping to real networks using SIM card.



The features of this kit are Low Cost, Simple / Easy Operation, Easy understanding of AT commands, Real Time operation, External Antenna.

The important technical specifications are

GSM capability**:** GSM 900 /1800, E-GSM
GSM data services: Asynchronous**,** Transparent & Non Transparent modes. 14.4 K bits/s
SIM Interface: 3 V

<u>Procedure:</u>

Hardware Settings:

1. Connect RS232 cable from com port of the pc to the 232 interface of the MODEM directly on
   the trainer kit.
2. Connect the power supply adapter to power source .
3. Make sure that the active SIM card with sufficient credit on it is inserted in the MODEM. Turn

on the power supply. Green LED on GSM Modem will turn on after few seconds and will main
continuously on till its searching for the network .Once the search is over it starts flashing.

Software Settings:

1. Click on start-all programs-accessories-communication-HyperTerminal.
2. Enter name and click on ok.
3. Select appropriate com port and click on ok.
4. Run Hypertrm.exe-A “connection Description “box will appear.
5. Name the connection something that you want to associate with talking to GSM modem.
6. Click <ok>--this brings up the phone number box.
7. The drop-down menu off the “connect using “line to select “direct to com[n]”””, where n is
 the communications port you will use to talk to the GSM modem.
8. Click <ok>-- this brings up the “COM[n] properties‟‟ box.
9. Use the drop-down menus to select Bits per second=9600, data bits=8, parity=none, stop
 bits=1 and flow control=none. Click on ok.
10. Now start the practical by first entering the command “AT” in hyperterminal and press
 enter and if the GSM modules is ready it will give back the “OK” signal.
11. Now start entering various commands like “AT+CIMI” which will give you SIM IMSI number then when someone calls then it will show “RING” in hyperterminal which can be received by “ATA” command.
12. Also enter some other commands which can be obtained from the manual and hence
 observe how much versatility is there in the module.

Observation:

Applying following AT commands the outputs obtained are shown below

Command	Output
AT	OK

AT+CIMI 404781010000682 OK IDEA
SIM IMSI
ATDL 9893091237
redial last number
AT+CGSN 354056000851034
IMEI command

If there is a incoming call then following statements will be seen in hyperterminal

RING Incoming call
+CLIP: "+917314032286", 145
ATA Command to accept the call
ATH Command to disconnect call

Conclusion:

By performing this experiment we can learn how the GSM module and GSM architecture works also we can understand the AT commands while camping it to real network using SIM card.