DEPARTMENT OF
ELECTRONICS AND TELECOMMUNICATION ENGINEERING

JADAVPUR UNIVERSITY

KOLKATA - 700032

COMMUNICATION SWITCHING LABORATORY

Name of the Student:________________________________

Roll Number:____________________________________

Year: III Session:_______________ Semester_________

Teacher-in-Charge__________________________________

Co-Workers:

Name	Roll Number

LIST OF EXPERIMENTS

EXPT. No.	DATE	NAME OF THE EXPERIMENT	TEACHERS' INITIAL
1		Study of the Telephone Ringer Circuit	
2		Study of the Dual Tone Multiple Frequency (DTMF) Dialer	
3		Familiarization with EPBX System - Part-I / Ex. No. 3(a) to 3(g)	
4		Familiarization with EPBX System - Part-II / Ex. No. 4(a) to 4(g)	
5		Simulation study of Frequency Division Multiplexing (FDM) using MATLAB	
6.A		PC to PC communication using Serial Port (RS-232)	
6.B		Study of Flow Control in Serial Communication	
7.A		**Experiments Using the LAN Trainer:** (A) Study of PC to PC Communication	
7.B		(B) Study the Performance of Token Ring Protocol	
7.C		(C) Study the Performance of a Star Topology with CSMA/CD Protocol	

Experiment No. 1

Name of the Experiment: Study of the Telephone Ringer Circuit

Objective: |To Design and test the telephone ringer circuit

Theory: The telephone line wires are connected to the ringer section through off hook/on hook/ringer switch and capacitor. When there is a call and telephone is in on-hook position, then exchange supplies an AC signal as the ringer signals to the telephone set. The ringer section is controlled by a special purpose IC for ring signal amplification and energizing buzzer/speaker through the volume control. The capacitor in the input section passes the AC signals to the ringer IC and block DC. The exchange supplies different timed on/off ring signals through the telephone line and ringer section generates different ring signals based on these signals. The on/off timing of these signals are different for local and STD/ISD calls. KA 2418B ringer IC is used in telephone receiver models like Siemens Euroset 802, BPL-2790, Tushaco. On the other hand LS 1240 ringer IC is used in Beetel, ITI, GCEL-501, TATA FONE T 800. Circuit diagram of telephone ringer circuit based on LS 1240 is shown in Figure 4.

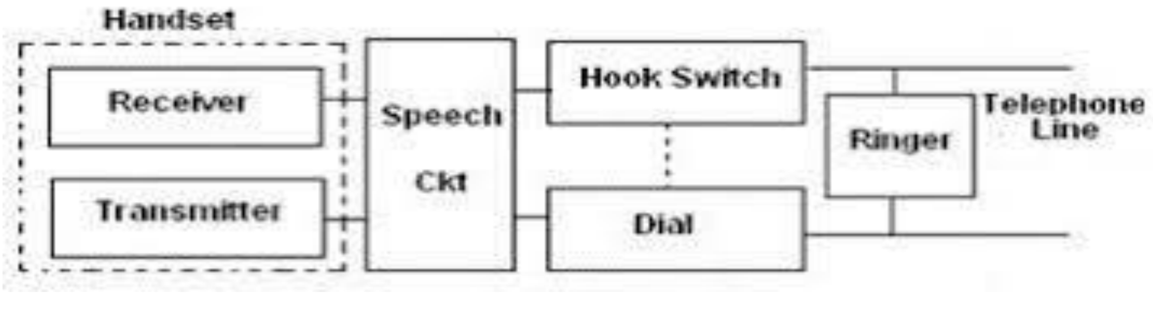


Fig. 1: Block Diagram of Telephone Handset

Apparatus & Components Required

	List of the component	Quantity
1	Bread Board	1
2	IC LS1240	1
3	Resistors (18Ω, 100 Ω, 470 Ω, 2.2 KΩ, 22KΩ, 23.5KΩ, 100 KΩ, 10MΩ)	1
4	Variable Resistor (220KΩ)	1
5	Capacitor (0.1µF, 1.8µF)	1
6	Capacitor (10µF, 63 V)	1
7	Buzzer	1
8	Connecting wires	
9	CRO & its probes	1

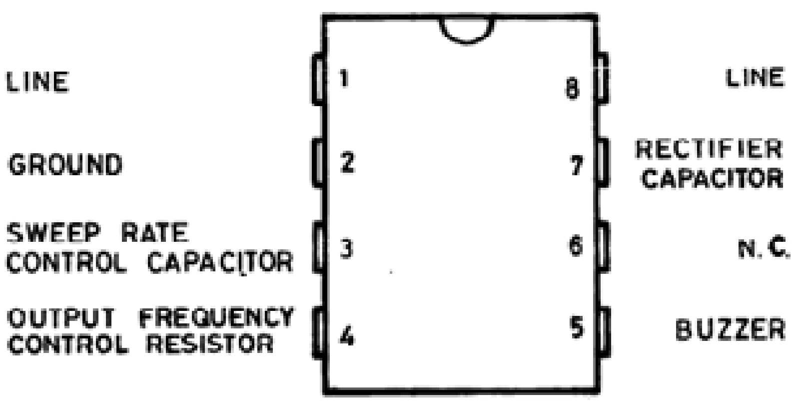


Fig. 2: Pin Configuration of LS1240

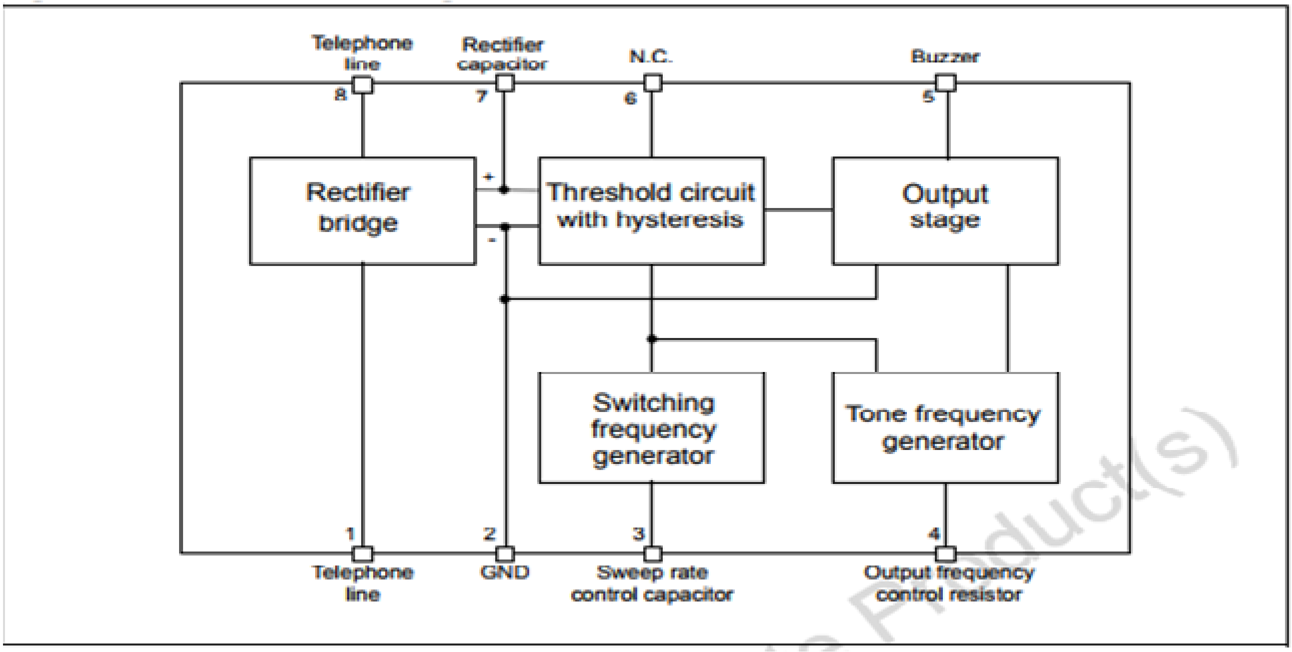


Fig. 3: Internal Circuit of LS1240

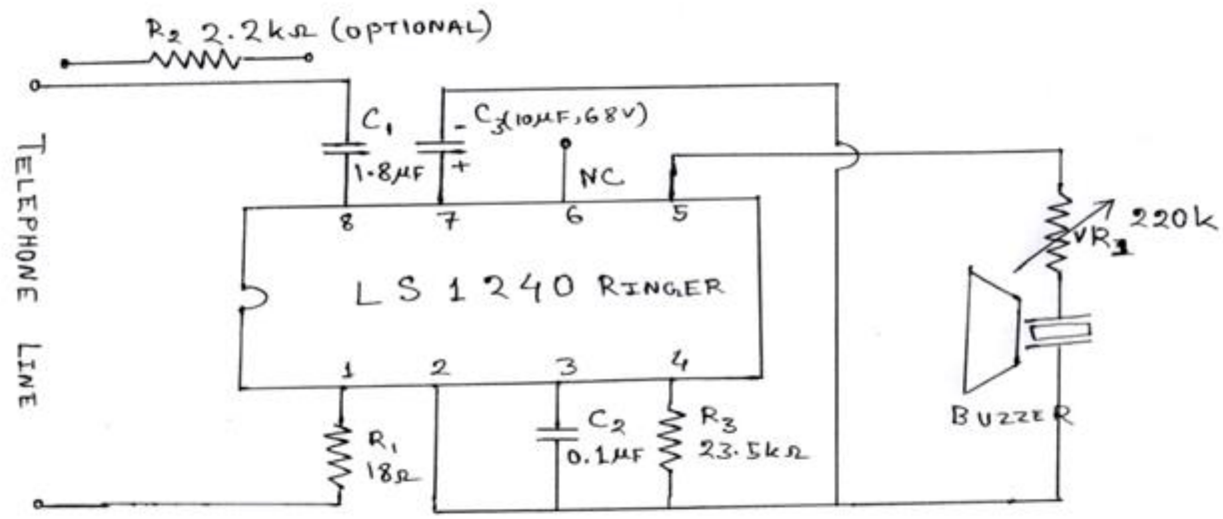


Fig. 4: Circuit Diagram of Telephone Ringer Circuit

Procedure:

1. Connect the circuit and the telephone line as shown in Fig. 4.
2. Dial the number (033-2457-2996) connected with the telephone ringer circuit and observe output at Pin no. 5.
3. Vary the resistance using 220KΩ potentiometer and observe the variation in sound and waveforms.
4. For a fixed value of C_2 = 0.1 μF and VR_1 = 20 Ω, use different values of resistance R3 and comment on the ringing tone.
5. For a fixed potentiometer resistances (say 20 Ω) vary capacitor C_2 and observe ringing frequency.

Resistances R_3	**Comment on Ringing Tone**
100 Ω	
470 Ω	
22 KΩ	
100 KΩ	
10 MΩ	

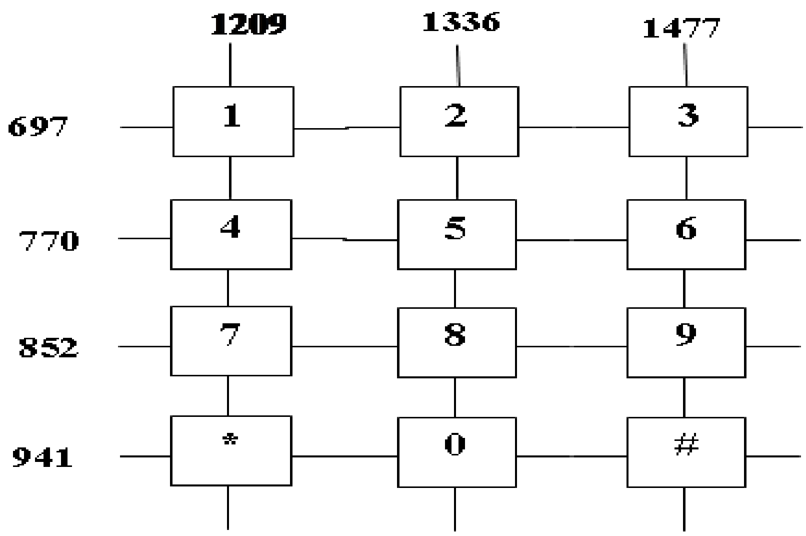
Conclusion / Inference:

Experiment No. 2

Name of the Experiment: Study of the Dual Tone Multiple Frequency (DTMF) Dialer

Objective: To Familiarize with DTMF encoder and decoder circuits

Theory: In Dual Tone Multi Frequency (DTMF) Dialing**,** the rotary dial is replaced by keypad. The principal method uses a pair of tones to signal each digit. The basic keyboard matrix consist of 12 keys (0 to 9, *, #) arranged in four rows and three columns. This arrangement is such that every button is related to one row and one column, where each row has unique low frequency and the column has a high frequency. Thus when a key from the key pad is pressed, then the low frequency of that row and high frequency of that column is generated and send to exchange. Figure shows the basic keyboard matrix. In the following Figure, all frequencies are in Hz. DTMF (Dual Tone Multi Frequency) decoder circuit identifies the dial tone from the telephone line and decodes the key pressed on the remote telephone.



Apparatus Required:

	List of the Apparatus	Make
1	Telephone Trainer Kit	Tesca Technologies pvt.ltd.

2	Telephone tester	NIKKI TELE I
3	CRO & its Probes	GEINSTEK (GOS-622G)

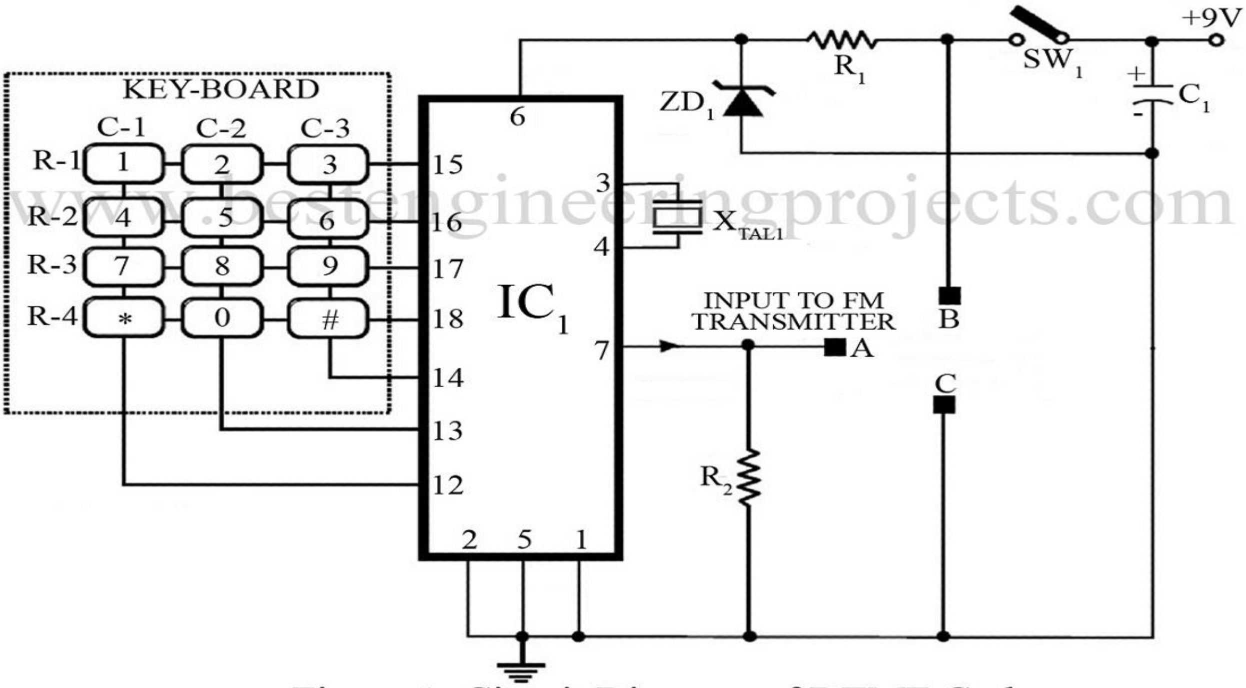


Figure 1: Circuit Diagram of DTMF Coder

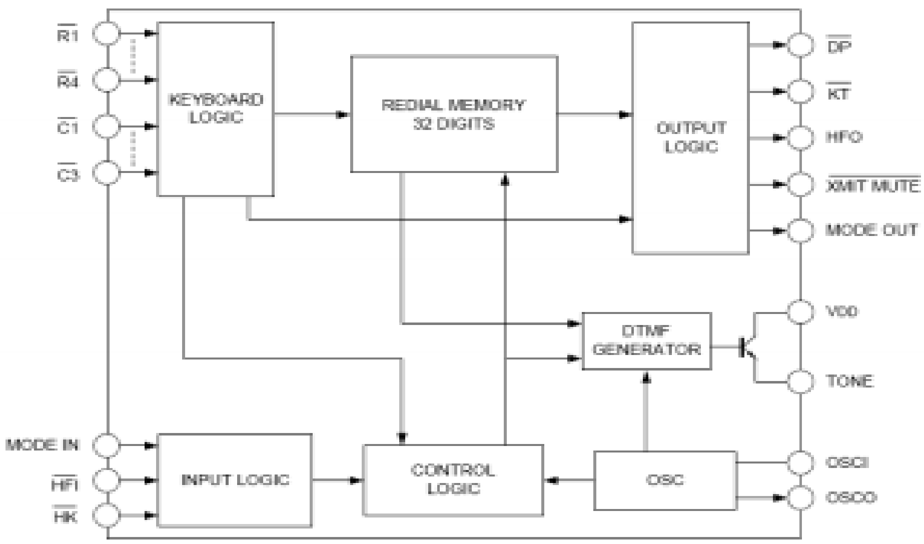


Figure 2: Internal Circuitry of the DTMF Encoder

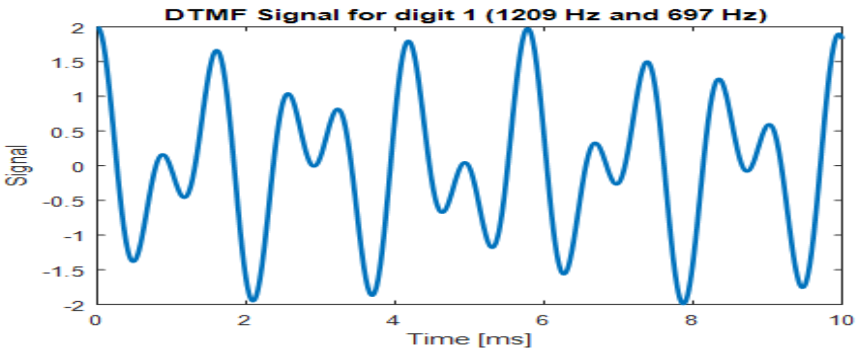
Procedure:

1. Connect the line connector from the telephone tester to telephone trainer board.

2. Trainer is set for Tone Mode.
3. Now if any switch of key pad is pressed, then DTMF output is generated. It is the combination of high frequency signal and low frequency signal.
4. At TP8 crystal oscillator, 3.58 MHz frequency output is obtained if any switch is pressed.
5. Table below is showing DTMF low and high frequency tones and decoded outputs.
6. Compare output frequencies with the frequency given in the Table.
7.

Button	Low DTMF frequency (Hz)	High DTMF frequency (Hz)	Binary coded output			
			Q1	Q2	Q3	Q4
1	697	1209	0	0	0	1
2	697	1336	0	0	1	0
3	697	1477	0	0	1	1
4	770	1209	0	1	0	0
5	770	1336	0	1	0	1
6	770	1477	0	1	1	0
7	852	1209	0	1	1	1
8	852	1336	1	0	0	0
9	852	1477	1	0	0	1
0	941	1336	1	0	1	0
*	941	1209	1	0	1	1
#	941	1477	1	1	0	0

Sample Output Waveform



Conclusion / Inference:

Experiment No. 3 & 4

Understanding Automatic Telephone Exchange / EPABX System

<u>Table of Contents</u>

To study and analyze the received tone and audio signals

- Experiment 4a

  To study the Control Unit and analyze the Ring Enable Control Signals
- Experiment 4b

  To study the Decode unit & DTMF unit and analyze the DTMF signal
- Experiment 4c

  To study the Ringer Unit and analyze the Ringer Voltage generation
- Experiment 4d

  To study the Clock Generation and Tone Generation Units and analyze the role of different tones
- Experiment 4e

  To analyze and measure that ring back tone is addition of Dial Tone and Busy Tone
- Experiment 4f

  To measure the frequencies of Dial Tone and Busy Tone
- Experiment 4g

  To study and analyze the Music Tone

Safety Instructions

Read the following safety instructions carefully before operating the instrument. To avoid any personal injury or damage to the instrument or any product connected to it.

For your safety:

Use proper Mains cord : Use only the mains cord designed for this instrument.

Ensure that the mains cord is suitable for your country.

Ground the Instrument : This instrument is grounded through the protective

earth conductor of the mains cord. To avoid electric shock the grounding conductor must be connected to the earth ground. Before making connections to the input terminals, ensure that the instrument is properly grounded.

Observe Terminal Ratings : To avoid fire or shock hazards, observe all ratings and

marks on the instrument.

Use only the proper Fuse : Use the fuse type and rating specified for this

instrument.

Use in proper Atmosphere : Please refer to operating conditions given in the

manual.

1. Do not operate in wet / damp conditions.
2. Do not operate in an explosive atmosphere.
3. Keep the product dust free, clean and dry.

Operating Instructions : The system has been tested for ruggedness. But

certain safety rules to be followed.

1. All extensions should be on Hook when powering on the system.
2. Avoid leaving the system 'On' for a long time.

Introduction

Scientech 2657 Understanding Automatic Telephone Exchange / EPABX System is ideal for the study of Automatic Telephone Exchange / EPABX. It provides the study of basic fundamentals in a unique way. It has been designed to meet the requirements of technical academics. The analysis of various detection phenomena such as ‘on’ hook /‘Off’ hook, Ring detection etc. and the generation of various signaling tones i.e. ring, dial, busy and ring back tone is well explained.

The basic of exchange technology and the automatic switching is well performed using the system. The EPABX system is of 308 configurations that is 3 direct lines and 8 extensions. The system on board explores the functioning of one direct line circuit, two extension line (with direct line and without direct line respectively).

Technical Specifications

Number of Inputs	:	3 Trunk / Direct Lines
Number of Extension	:	8 Lines
Technology	:	CMOS Cross Point Switching for Speech Communication (Microcontroller based)
Standard Features	:	Tones such as Dial, Busy, Ring etc
Dialing	:	DTMF and Pulse Dialing
		(Speed: 10pps; Ratio 66.33%; IDP 800ms)
Program Memory	:	512 kb EEPROM
Loop Resistance	:	Extension -600 Ohms, Co-line- 1200 ohms
Cross Talk Attenuation	:	>70 dBm
Test Points	:	More than 40 Nos.
Power Supply	:	230 V ± 10%, 50 Hz
Switched Faults	:	8 Nos
Features that can be set	:	Line Hunting, Direct Access to trunk line, Redial,
		Line Status Indication, Automatic call back, Do Not Disturb, Call Transfer, Call Pick Up, Direct Access Trunk Call Forwarding, Call parking, Conference, Hot line System, Extension Privacy, Call Transfer, Barge in
Weight	:	3 Kgs.
Dimension	:	(L) 435 mm × (W) 260 mm × (H) 95 mm
Fuse	:	1Amp

Technology History

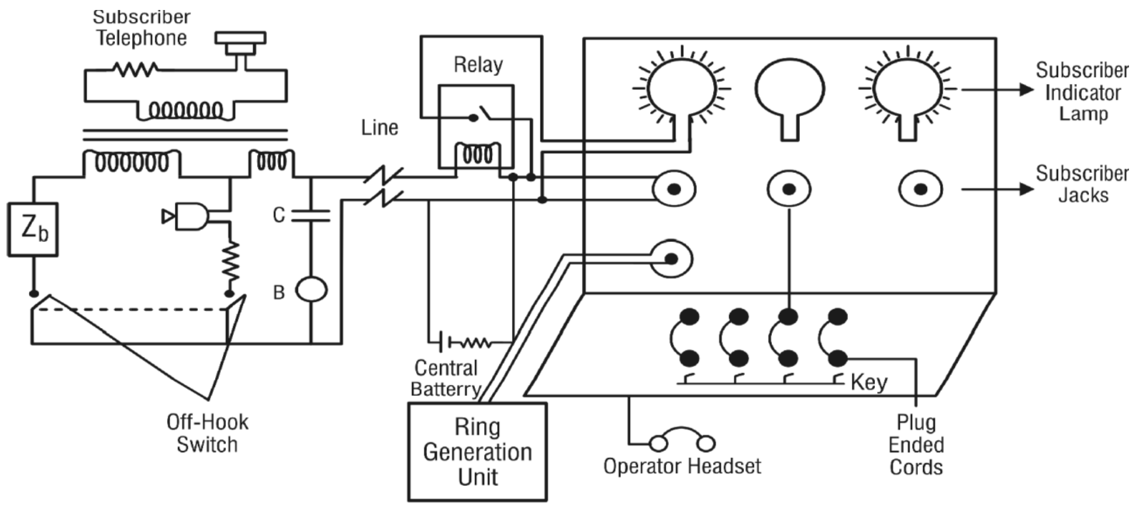
In the 1876's Alexander Graham Bell invented the telephone; a wired system for two way voice communication between remote locations. You spoke into a unit at one location and your voice was heard at the other location, immediately this system was somewhat limited in that. It only allowed communication with one fixed location, so it was an obvious advance to have lines going to other locations. Initially, this is what happened-each telephone had lines going to many other telephones, which meant a lot of wires and there were practical limitations as to the number of phones one could connect to.

Soon, the idea of central switching that's *Manual Switching* was developed. Each telephone connected to a central hub (the exchange) and from there, the operator would connect your call to another subscriber. Thus switchboards were developed where upon lifting your receiver, an operator was alerted and you would tell her who you wished to speak. Apart from other limitations it was very labour intensive and as the popularity of the telephone grew, the number of operators employed by the Office grew; large switching centres (exchanges) could have many tens of operators, each with their own switchboard. In the early days, each telephone user was known as the 'subscriber' and that term is still used today.

Manual Switching Theory

A simple Central Battery system operated by a human being is shown in Figure 4. The system consists of one switchboard manned by operator. The subscriber lines are terminated on jacks mounted on the switchboard. There is one jack for every subscriber line. Associated with each jack is a light indicator to draw the attention of the operator. When a subscriber lifts the hand set, the off-hook switch is closed, causing a current to flow through the hand set and the lamp relay coil. The lamp relay operates and the indicator corresponding to the subscriber lights up. The operator establishes contact with the subscriber by connecting the head set to the subscriber line via the headset key and a plug-ended cord pair. A plug-ended cord pair has two cords that are connected internally and terminated with a plug each at the external ends. The plugs mate with the jacks. To establish contact, a cord is plugged into the subscriber jack and the key corresponding to the chosen cord is thrown in position to connect the head set. On being told the number required by the subscriber, the operator verifies whether the called party is free, and if so, sends out the ringing current to the called subscriber using a plug-ended cord pair.

The ringing, circuit at the subscriber end is usually a bell shown as B in Figure 4, with a capacitor C, in series. They remain connected to the circuit always. The capacitor allows the alternating ringing current from the exchange to pass through the bell but prevents the loop direct current. If the called party is busy, the calling subscriber is told about the same. When the called party answers, his indicator lamp lights up.



Block Diagram of CB Manual Exchange

Figure 4

The operator then establishes a connection between the calling and the called party by plugging in the cord pair to the called party jack. The supervisory lamps used to glow during conversation and would be extinguished when the subscribers replaced their handsets. This type of clearing signal used to be called negative. There is another method in which during conversation both the lamps are off. At the end of conversation when both the subscribers replace their handsets, both the lamps glow indicating to the operator that the cord circuit is free. The operator then withdraws the plugs from the jacks, and may use the cord circuit for establishing another connection.' The glowing of the supervisory lamp when a subscriber replaces his handset is called the clearing signal, and a positive clearing signal. . This is called positive, as the clearing is indicated by a positive appeal to the sense of the operator. In this particular case the positive appeal is to the eye and the object is glowing lamp. A positive signal maintains a persistent effect on the nerve of the operator, so that the operator is very unlikely to forget to unplug the cord circuit.

In a manual switching system, the operator has full control of a connection. He enables the signaling systems, performs switching and releases a connection after a conversation.

The Invention of Automatic Switching:

Almon B. Strowger was an undertaker in Kansas City, USA. The story goes that there was a competing undertaker locally whose wife was an operator at the local (manual) telephone exchange. Whenever a caller asked to be put through to Strowger, calls were deliberately put through to his competitor. This obviously frustrated Strowger greatly and he set about devising a system for doing away with the human part of the equation.

Strowger developed a system of automatic switching using an electromechanical switch. With the help of his nephew (Walter S. Strowger) he produced a working model in 1888 (US Patent No. 447918 10/6/1891). In this selector, a moving wiper (with contacts on the end) moved up to and around a bank of many other contacts, making a connection with any one of them.

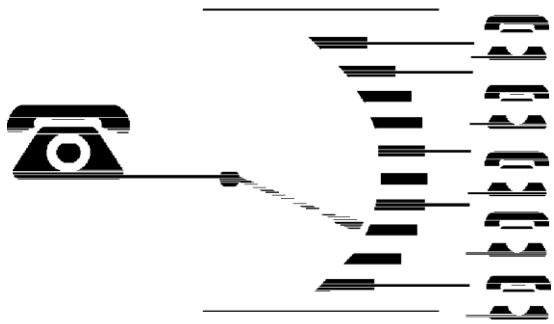


Diagram of a simple Selector

Figure 1

Strowger formed his company 'Strowger Automatic Telephone Exchange' in October 1891.The first automatic exchange appeared in 1892 in La Porte Indiana.Almon B. Strowger died in 1902.

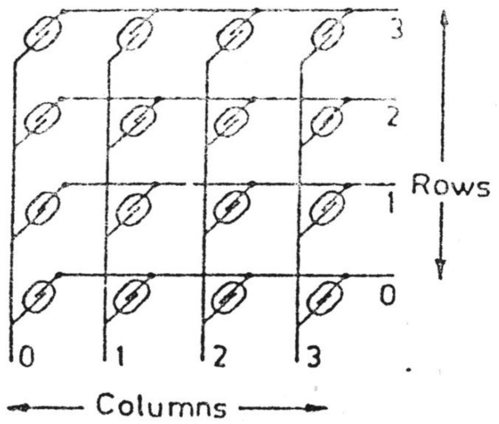
Though it improved upon the performance of a manual exchange, it still had a number of disadvantages, Such as large number of mechanical parts, limited availability, bulky in size etc.

Then there were chronological developments in the field such as Analog switching, Digital switching, Space division switching, Time division switching and the evolution is still on with the IP PBX.

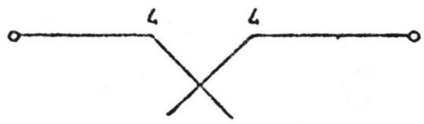
Switching Technology

Matrix:

A basic switching network or switch is made by arranging components in rows and columns in the form of a matrix. A connection is set up between a vertical access point (columns) and a horizontal access point (row) by closing the crosspoint at the intersection of the row and column, corresponding to the access points. The basic switch of inlets are equal to the number of outlets, it is called a square matrix. The matrix and its symbol are shown in figure 2.



(a) BASIC MATRIX OF SIZE 4 X 4



(b) Symbolic Representation of 4 □ 4 matrix

Figure 2

Cross point:

A cross point is a miniaturized switch, used to connect various links of the speech path. It has two stable states viz, closed (conducting) and open (blocking). In the open state, the impedance of the cross point must be as high as possible, over the entire range of the frequencies being switched. In the closed state, the impedance should be as low as possible. Appropriate acceptance range of values for the impedances, are

Closed : 0.1-200 Ohms

Open : 10-25 M ohms

Switching networks may use different types of switches. Various types of cross points have been evolved, keeping in pace with the other developments in SPC exchange systems. The prime continuing priority is to reduce its size, operating time, and cost. There are two technological categories of cross points, viz, electromechanical, i.e. a metallic switch, and electronic device. They are further sub-divided, as shown in Figure 3.

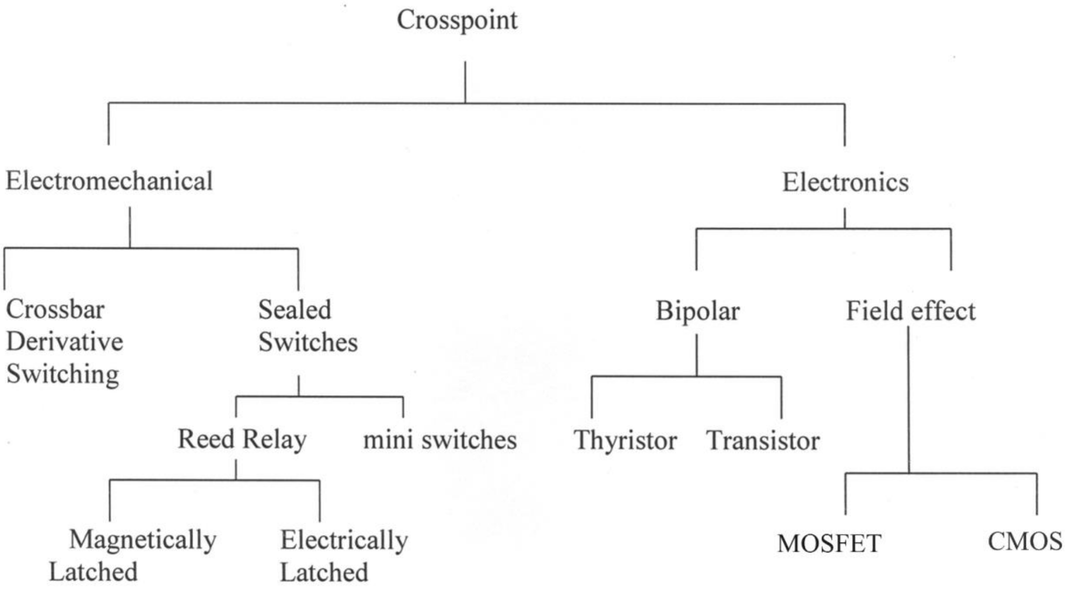


Figure 3

Electronic Cross point:

Electronic cross point offer considerable advantageous of reduced size, weight, and power consumption, compared to their electromechanical counterpart. However, they suffer from high ‘On’-resistance and low power handling capacity, compared to metallic and MOS crosspoints are 0.5 ohms, several ohms, and around a hundred ohms, respectively. Due to these limitations, they have so far been used limitedly, in Analogue switching.

Space Division Switching:

The **Scientech 2657** A.T. Exchange / training system works on space division or domain switching. The space division switching has its own advantages over other switching domains such as more bandwidth, simple circuits and easy voice handling. In space division, each subscriber or extension has it own dedicated path in the exchange. The exchange uses one full physical path and a dedicated cross point for one call this may result in blocking. As the number of extension increases it directly affects the number of crosspoints and circuitry.

Erlang:

1. Erlang is a unit of traffic measurement and represents the continuous use of a line for one hour.
2. One Erlang represents the amount of traffic intensity carried by a channel that is completely occupied.
3. One hour of telephone traffic in an hour of time. For example, if circuit carries 120 minutes of traffic in an hour, that's two Erlang.

A.K. Erlang the Danish mathematician was the first person to study the problem of telephone networks. In 1917 by studying a village telephone exchange he worked out a formula, now known as Erlang’s formula, to calculate the fraction of callers attempting to call someone outside the village that must wait because all of the lines are in use. Although Erlang’s model is a simple one, the mathematics underlying today's complex telephone networks is still based on his work.

$$\text{Erlang} = \frac{\text{Number of calls X Average call duration}}{\text{One hour (60 min or 3600 sec)}}$$

Suppose a system handles 408 calls of 2.5 minutes in duration during busiest 1 hour So,

$$\text{Erlang (BHT)} = \frac{408 \text{ X } 2.5}{60 \text{ (min)}}$$

$$\text{Erlang} = 17$$

There are two types of trunked systems which are commonly used

1. The first type offers no queuing for call requests. This type of trunking is called Blocked calls cleared or Lost call cleared. The probability that a call is blocked is determined by the Erlang B formula.
2. The second type of trunked system is one in which a queue is provided to hold calls which are blocked. This type of trunking is called Blocked calls delayed. The likelihood of a call not having immediate access to a channel is determined by the Erlang C formula.

Erlang B Formula:

The Erlang B model is an established traffic model for predicting Telecommunication performance. This model makes assumptions about the nature of telephone traffic and may prove inaccurate under special circumstances. It is widely used to determine the number of trunks required to handle a known calling load during one hour period. It is also used to determine the probability that a call is blocked and gives the GOS (Grade of Service) for a trunked system. The formula assumes that if callers get busy signals, they go away forever and no retries where as the fact is some callers retry.

Disadvantage:

It under estimates the trunks required.

The Erlang B formula is

$$\text{GoS[blocking]} = \frac{\frac{A^C}{C!}}{\sum_{n=0}^{C} \frac{An}{n!}}$$

Where,

C = number of trunked channel

A = Total offered traffic

Erlang C:

This formula also predicts the waiting time (delay) based on three things No. of lines, No. of people waiting to be served and the average amount of it takes to serve each person. It can also predict the resources required to keep waiting time with in targeted limits.

Disadvantage:

It can over estimate the resources required

Traffic Analysis Definitions:

1. **Busy hour:**

   The busy hour of an exchange is a chosen 60 minute interval in which the telephone traffic is the highest. The busy hour varies from exchange to exchange depending on its location, and the community interest of its subscriber.

2. **Busy hour traffic:**

   This is an amount of call traffic handled by a group of phone lines during the busiest hour of the busiest day for your system. Busy hour traffic is defined in units of Erlangs.

3. **Holding time:**

   Average duration of a typical call. Denoted by H (In seconds)

4. **Grade of service (GOS):**

   The Grade of service (GOS) is a measure of the ability of a user to access a trunked system during the busiest hour.

   It is a measure of congestion which is specified as the probability of a call being blocked, or the probability of a call being delayed beyond a certain amount of time. The grade of service of .002 means 2 calls per 1000 or 1 in 500 has been lost.

Features of EPABX System

The **Scientech 2657** is of 308 configuration, 3 direct lines and 8 extension lines.

Monitoring Tones:

It is necessary to get familiar with the various "tones" of your EPABX System.

1. **Exchange Dial Tone**: The dial tone is a continuous sound and sounds like *uoon* continuously with no silence period but gets interrupted interval. It lasts for 8 seconds during which the exchange waits for dialing to be initiated. If no dialing takes place during this period the EPABX times the user out and starts issuing a busy tone.

2. **DOT, P and T, and Central Exchange Dial Tone:** On accessing a direct line, you will obtain the normal P and T dial tone.

3. **Busy Tone:** The busy tone is a discontinuous sound (Du-Du) with 'On' and 'Off' periods. There are kinds of busy tones with differences in 'On' and 'Off' period or pitch. This tone indicates that the system is too busy to accept any operation. This tone converts to a dial tone as soon as the system is free enough to accept your dialing. The type of busy tone (which is the slow one) is encountered when called subscriber number is busy.

4. **Ring Back Tone:** This sound is continuous with silence period. When you dial extensions, you will hear this ring-back tone till the extension answers.

5. **Ringing Tone:** There are two types of rings that can be heard from the telephone instrument connected to the System.

   a. When your instrument is called by another internal instrument the ring will be a continuous one with a one second 'On' and 3 seconds 'Off' period.

   b. A ring from a Co. In. (Trunk) Line will ring like a normal telephone.

6. **In-Coming Call Queuing Tone:** This tone will be heard when a caller is waiting for your extension to get free. It is similar to the feature mode tone and will be heard repeatedly with a long pause. This tone will only be heard only when there is a Co. In. (Trunk) incoming call.

   Intrusion Tone: (Barge in Tone): This is-a fast beep.

7. **Hook Flash (HF):** The features of your EPABX require the use of a Hook Flash (HF). This is performed by tapping the hook switch of your extension for a period less than 0.5 second. Care should be taken not to press the hook switch for more than 0.5 seconds when a HF is desired. If the exchange hook switch is pressed for longer than 0.5 seconds it will register a "hang up" or "reset".

   Telephone instruments have a built in electronic flash key. It is advisable to use the flash key instead of the hook switch in order to register a "hook flash".

Features/Operations:

1. Access to Trunk Line (0): (Line hunting)

   Extensions are programmed to have access to all trunk lines by dialing;

   - 20 for Trunk 0
   - 21 for Trunk 1
   - 22 for Trunk 2
   - Dialing of 0 will accesses the trunk by line hunting. Extensions can also be denied this access.

2. Extension to Extension Call (Extension)

   - EXT 30
   - EXT 31
   - EXT 32
   - EXT 33

   When one extension user wishes to talk to another extension, the user has to proceed as follows:

   Lift hand set and hear dial tone. Dial Extension Number Wait for the internal ring tone. Speak when called party answers.

3. Redial:

   Any extension user can repeatedly dial the last number (whether internal or external) without pressing all the numbers again. For this follow the procedure below:

   Disconnect previous call. Lift hand set and hear dial tone.

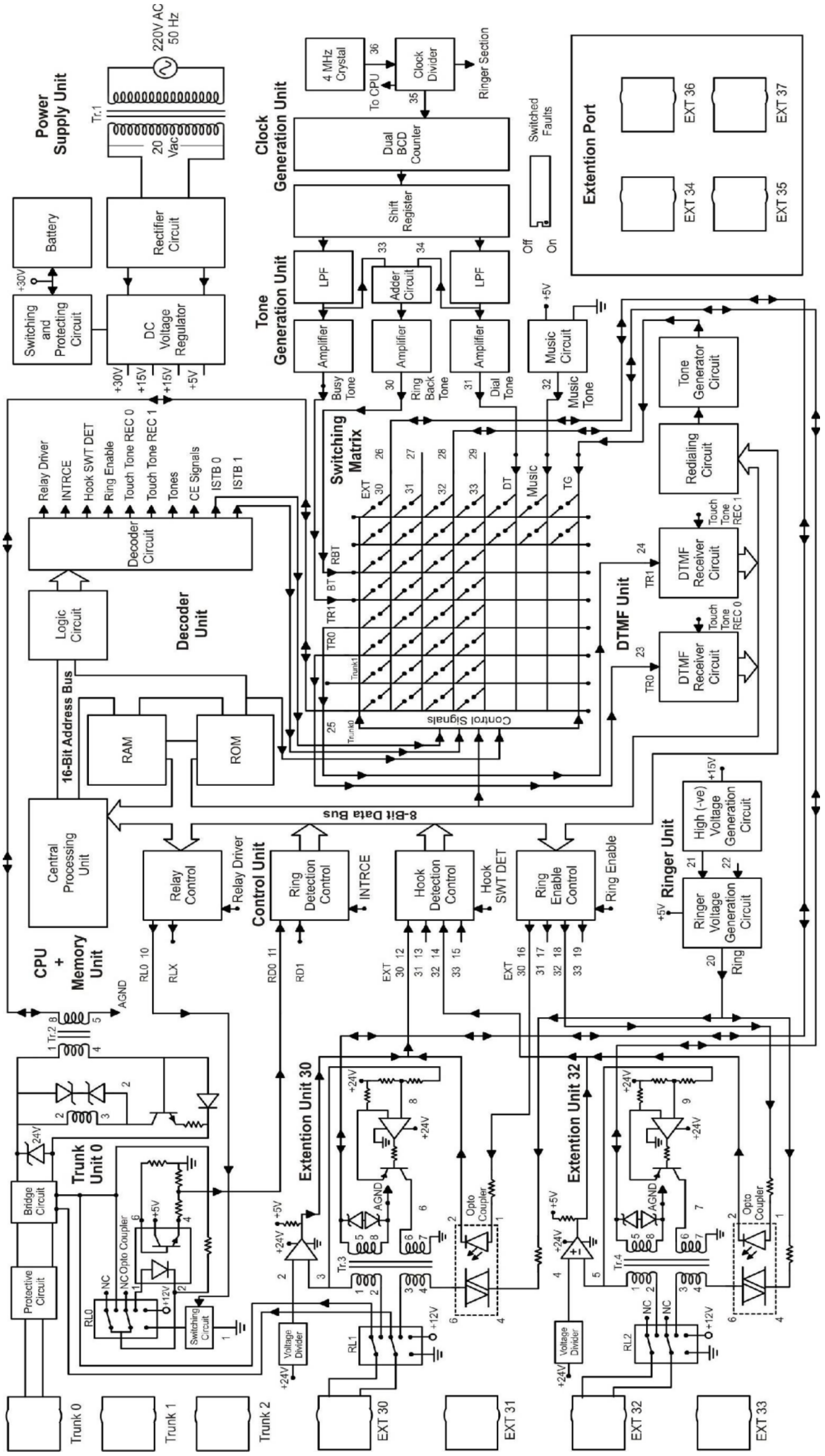
   Dial * asterisk key.

   The last dialed number will be redialed automatically

Introduction to EPABX System

An Automatic Telephone Exchange / EPABX System consist of following units.

1. Power Supply Unit
2. Trunk Unit
3. Extension Unit
4. Control Unit
5. Decoder Unit
6. DTMF Unit
7. Clock Generation Unit
8. Tone Generation Unit
9. Ringer Unit
10. Switching Matrix
11. Memory + CPU Unit



Block Diagram of EPABX System

Experiment 3a

Objective: Getting started with Automatic Telephone Exchange / EPABX System

Theory: This section of the manual will introduce you about ‘how to make a working set up of Automatic Telephone Exchange / EPABX System. By connecting the Telephone Instruments to the Extension ports (EXT 30/ EXT 31/ EXT 32/ EXT 33) one can make and receive the calls among them and can check the working condition of the system. By connecting Telephone line (DOT line/ CO line / EPABX Line) to Trunk Unit (Trunk 0/ Trunk 1/ Trunk 2) one can make / receive call from direct line.

Apparatus required:

- Scientech 2657 – Automatic Telephone Exchange / EPABX System
- Telephone Instruments
- Telephone line: DOT Line / CO Line / Trunk Line

Operating Condition: Before performing this experiment, please ensure that;

- All Switched Faults should be in ‘Of’ position.
- Ensure that the Telephone instruments are properly connected to the Extension ports.
- Ensure that the Telephone line should be alive.

Procedure:

1. Connect the Mains cord to the System and switch the power ‘On’.
2. Observe that the TRO and TR1 LED will glow to indicate that the lines are ready to receive the calls from the direct line.
3. Lift up the handset of telephone instrument connected to EXT 30 and observe that the status LED will glow Green to show the ‘Off hook’ condition of the telephone instrument.
4. Also observe that Hook Detection Control unit and Switching Matrix also indicate the status of the EXT 30 by glowing LEDs.
5. Now Dial EXT 32 from EXT 30 and observe the status of glowing LEDs on board while EXT 32 rings.
6. The status LED of EXT 32 glows Red shows that the telephone has been dialed and it is ringing. At the same time ‘Ring Enable control unit also indicates the status of called EXT 32 by glowing LED.
7. As soon as the handset of EXT 32 is lift up the status of the EXT32 led changes from Red to Green and Ring Enable Control LED goes off.

8. Also Hook Detection Control unit and Switching Matrix indicate the status of the EXT 32 by glowing LEDs.
9. Now the conversation can be done between the two extension lines.
10. Connect the trunk line or direct line to the Trunk 0 jack on the system.
11. Make a call to the trunk line connected number.
12. Observe the LED glowing at Ring Detection Control unit Test point 11.
13. EXT 30 will ring up.
14. Pick up the phone and do the conversation.

Experiment 3b

Objective:

To study the Power Supply Unit of EPABX System

Power Supply Unit

The Power Supply unit generates the Regulated DC Voltages (+30V; +15V; +5V) and High negative Ringer voltage (-12V approximately) for EPABX system. The input to the Power Supply may be 220 – 230V AC or the system can also work on +24 V DC battery.

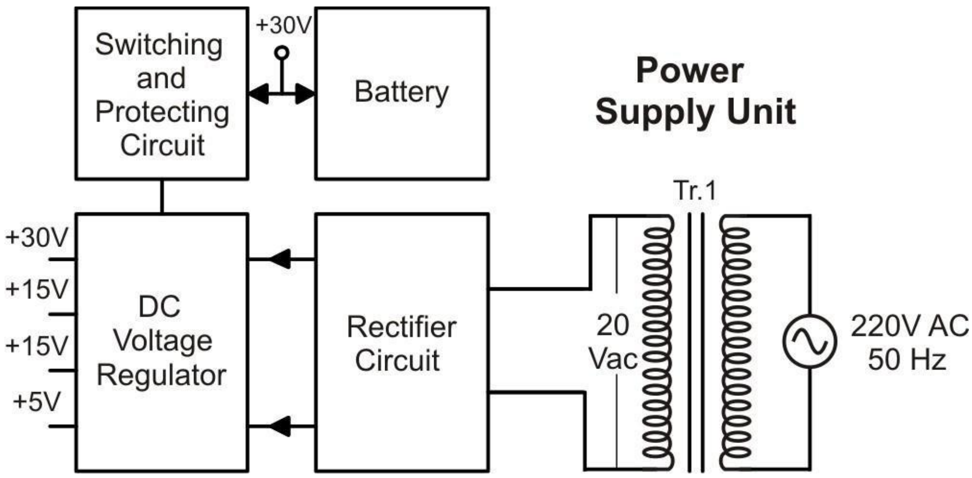


Figure 5: Block diagram of the power supply

Circuit Description:

The AC line is fed to a step down transformer which gives out the AC voltage 20V. This is fed to the Voltage regulator through rectifier. The regulator circuit consists of LM 317 ICs. The regulator circuits provide different voltages (+30V; +15V; +5V) to the various units of the system.

Procedure:

1. Connect the Mains cord to the System and switch the power 'On'.
2. In Power Supply Unit; measure the AC voltage across the Test points given at the output of the Transformer. It will be 20V AC approximately.
3. Measure the regulated DC voltages at the output Test points of 'DC Voltage Regulator' section; +30V; +15V and +5V approximately.
4. Measure the Ringer Voltage at Test point 21 of Ringer Unit. It will be -120V approximately.

Experiment 3c

Objective:

To study Trunk Unit and analyze:

- Ring Detection phenomena and
- Trunk Relay Switching

Trunk Unit:

Trunk unit often termed as junction unit/card. It is interfaced to the lines coming from PSTN or Central exchange or the other exchange.

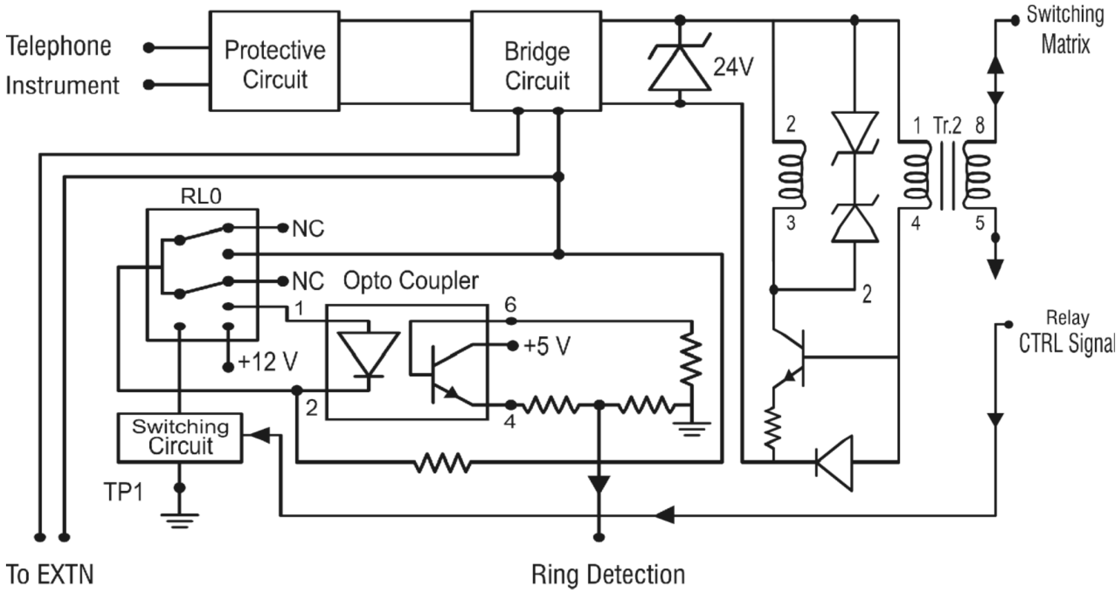


Figure 6: Circuit diagram of Trunk circuit

Circuit Description:

The protective circuit provides protection against over voltages and other hazards. It consists of bridge circuit for polarity protection, similar to that of a telephone instrument. A 24 V Zener diode is used to maintain proper voltage. The opto-coupler is used for incoming ring detection. As the coupler gets the signal at its pin 1 the 5V applied to pin 5 is extended to pin 4 which is used as an indication of an incoming ring .Then after detection the exchange serves its own ring to the extension. When power is ON the relay is connected to NC terminal and hence it is not connected to any extension. When trunk line is accessed using 20 or 0. The ground pin of the relay is grounded and the relay switches and gets connected with the extension through switching matrix. The relay signal is generated by the control unit under the supervision of CPU.

Ring Detection Phenomena:

Procedure:

- Connect the trunk line or direct line to the Trunk 0 jack on the system.
- Power 'On' the system.
- Now to observe Ring detection phenomena at EPABX system; make a call to the connected trunk line number from any other Telephone instrument.
- Observe that the LED glows at Ring Detection Control unit Test point 11.
- Measure the voltage at Test point 11. It will be 3V approximately.

Working:

As the coupler gets the signal at its pin 1 the 5V applied to pin 5 is extended to pin 4 which is used as an indication of an incoming ring and measured at Test point 11. By this the CPU gets the information through control unit.

We conclude that if a subscriber calls a number located in a different exchange, the ring is extended till it reaches the called subscriber exchange where the incoming ring is detected and extends its own ring till the subscriber end.

Trunk Relay Switching:

Procedure:

- Connect the trunk line or direct line to the Trunk 0 jack on the system.
- Power 'On' the system.
- Measure the voltage at Trunk Unit 0 Test point 1. It will be 12 VDC.
- Dial 20 from EXT 30 and listen to the dial tone.
- Observe the Relay Control LED is glowing. Measure the voltage at Test point 10. It will be 1.5V approximately.
- Now measure the voltage Test point1 again it will be 0V or it is actually grounded.

Working:

When power is 'On' the relay is connected to NC terminal (Similar to normal Power off position) because its ground pin is not grounded and hence it is not connected to any extension. When trunk line is accessed using 27 or 0 The ground pin of the relay is grounded and the relay switches and gets connected with the extension through switching matrix. The power for switching circuit is provided by Relay CTRL of control unit.

Experiment 3d

Objective:

To study the Extension Unit and switching mechanism of relay

Extension Unit

Extension unit or Subscriber line circuit or line card are the various terms used. It is the first circuit that is connected to the subscriber among the chain of circuits. It is used as an interface between individual subscribers (Extensions) lines and the rest of the telephone exchange. Being a part of both outside and inside telecommunication network makes it susceptible to transients caused by atmospheric disturbances such as lightening and those induced by the AC mains like power cross apart from the physical hazards, so most of the line units incorporate protective circuits such as varistor etc.

The line unit performs following functions:

- Protection
- Voltage (battery) Feed
- Supervision
- Ringing

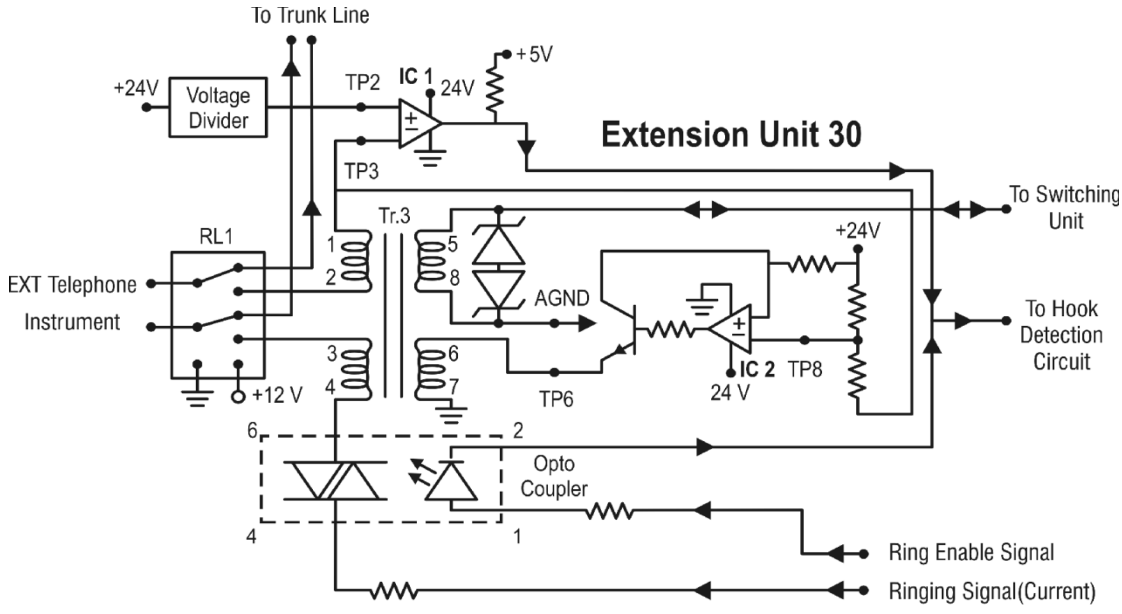


Figure 7: Circuit diagram of line unit with trunk line

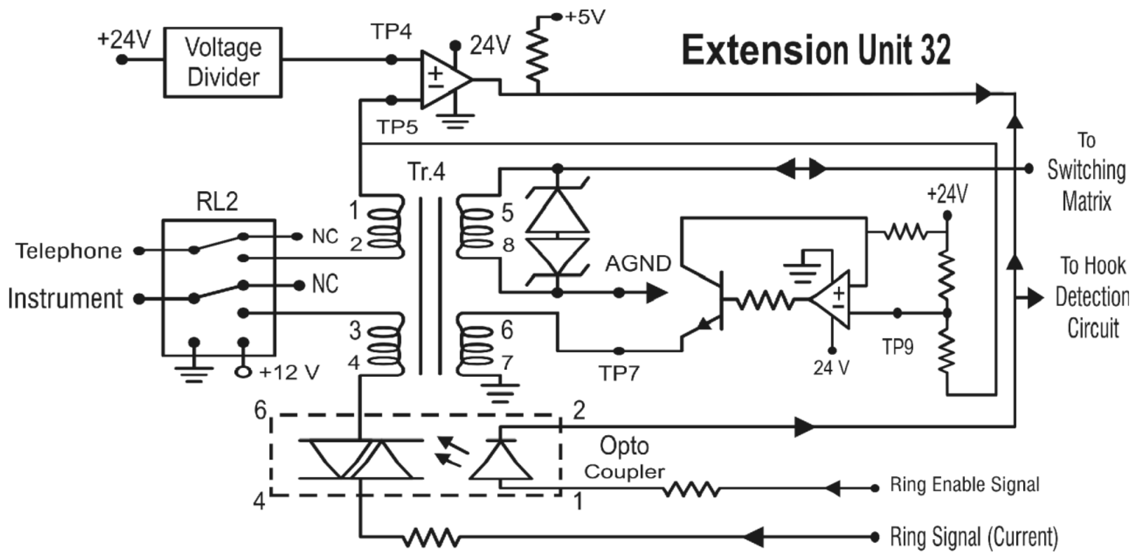


Figure 8: Circuit diagram of an extension unit without trunk line

Circuit description:

The surge protectors and hybrid transformers provides protection against over voltages. The hybrid transformers also help in impedance matching failure of which may result in echo. A voltage of 20V-60V is being fed to the extension loops. Specifically this EPABX system feeds 24V ‘On’ Hook and 8-12V (40 mA) ‘Off’ Hook. When the power is off the EXT 30 is directly connected to Trunk 0, So that no calls are missed. The calls will land only on this EXT no line hunting is done.

The extension line has an op-Amp which senses the ‘On’/‘Off’ Hook condition and passes the information to control unit. Ring is extended to extensions through line unit under the supervision.

Procedure:

- Connect a trunk line or direct line to the system.
- Power ‘Off’ of the system.
- Lift the handset of the EXT 30, listen to the dial tone.
- It will be a trunk line tone.
- Now power on the system.
- Lift the handset of EXT 30 again, Listen to the dial tone; it will be internal dial tone.
- Now, in order to access Trunk 0, dial 20 or 0.
- Power ‘Off’ of the system.
- Lift the handset of the EXT 32 and no dial tone is heard.

- Now power on the system.
- Lift the handset of EXT 32 again, Listen to the dial tone. It will be internal dial tone.
- Now, in order to access Trunk 0, dial 21 or 0.

Working of Extension 30:

The normally closed terminals of the relay RL1 of the EXT 30 is connected between subscriber telephone instrument and the Trunk line 0. So in case of power failure no calls are missed but calls can’t be transferred. As soon as power is switched ‘On’ the relay switches. Now any switching that’s accessing a trunk line or calling other extension is done by Switching Matrix (for more details refer Switching Section).

Working of Extension 32:

The EXT 32 is not directly connected to any trunk line since it’s a 204 the Trunk 0 and 1 are connected to EXT 30 and 31 respectively. So in the power ‘Off’ condition the relay RL2 is in normally closed terminals which are left open. When the power is ‘On’ the relay operates and get in contact with circuitry. Now switching will take place through matrix.

Experiment 3e

To analyze and measure the ‘Off’ Hook detection

Procedure:

- Connect the Mains cord to the System and switch the power ‘On’.
- Measure the voltage at Test point 2; it will be 30V approximately.
- Measure the voltage at Test point 3; the voltage will be between 40 to 60V approximately.
- Lift handset of the EXT 30 and observe the status LED is glowing green.
- Measure the voltage again at Test point 3; it will be 8 – 10V (where as Test point 2 remains the same).

Working:

As soon as the extension goes off hook the current starts flowing in the telephone instrument which consists of voltage dropper or current limiter circuits as a result the voltage present in the line drops down to 8V.This changes the voltage at the negative terminal Test point 3 of the op amp LM 393 that’s IC 1(we already know voltage at Test point 2 is 17 V positive terminal of Op Amp), which in turn changes the state of the op amp. Since LM393 is an open collector Op-Amp the 5 V applied to the collector is forwarded to the hook detection circuit. Which takes it as an ‘Off’ hook indication and the information is also fed to 8 bit bi-directional data bus for CPU.

Experiment 3f

Objective:

To analyze the working of the ‘Flash’ and its use

Procedure:

- Connect the Mains cord to the System and switch the power ‘On’.
- Measure the voltage at Test point 3; the voltage will be between 40 to 55V approximately for ‘On’ Hook condition.
- Lift the handset of EXT 30.
- Measure the ‘Off’ Hook voltage 8 – 10V at Test point 3 and keep the multimeter connected.
- Operate the ‘Flash’ button on the telephone instrument.
- Observe the voltage fall and rise at Test Point 3.

Working:

The ‘Flash’ button is provided on telephone instrument generates a pause. The duration of the pause varies from brand to brand and also with countries. Here it generates a pause of less than 0.5 sec supported by the EPABX. The flash facility is provided for transfer of call or to get to next call while hanging the first call on hold. Flash can also performed by tapping the switch. Some PSTN or central exchanges don’t provide this facility.

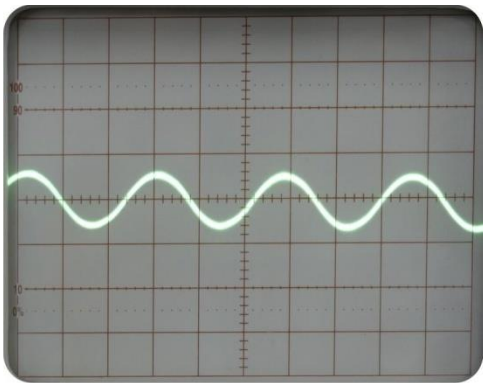
Experiment 3g

Objective:

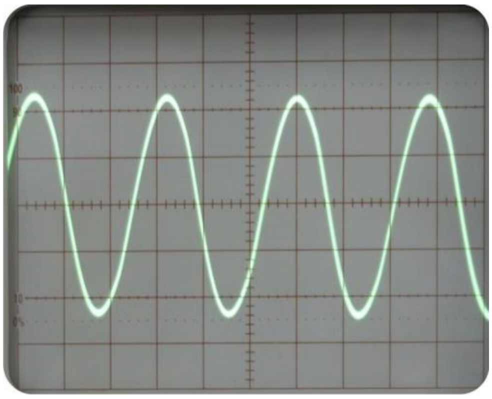
To study and analyze the received tone and audio signals

Procedure:

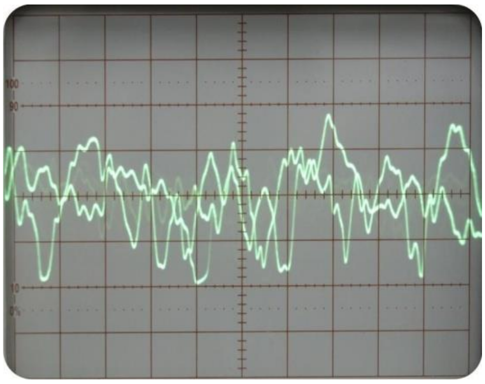
- Connect the Mains cord to the System and switch the power ‘On’.
- Lift the handset of EXT 30, listen to the dial tone.
- Connect the Oscilloscope at Test point 8 of EXT 30 unit, a low amplitude sine wave is observed.
- Now connect at Test point 6 and observe the amplified dial tone.
- Now dial EXT 32 and receive the call. Observe the Audio signals at Test point 8 during the conversation.



Dial Tone



Amplified Dial Tone



Audio signals

Experiment 4a

Objective:

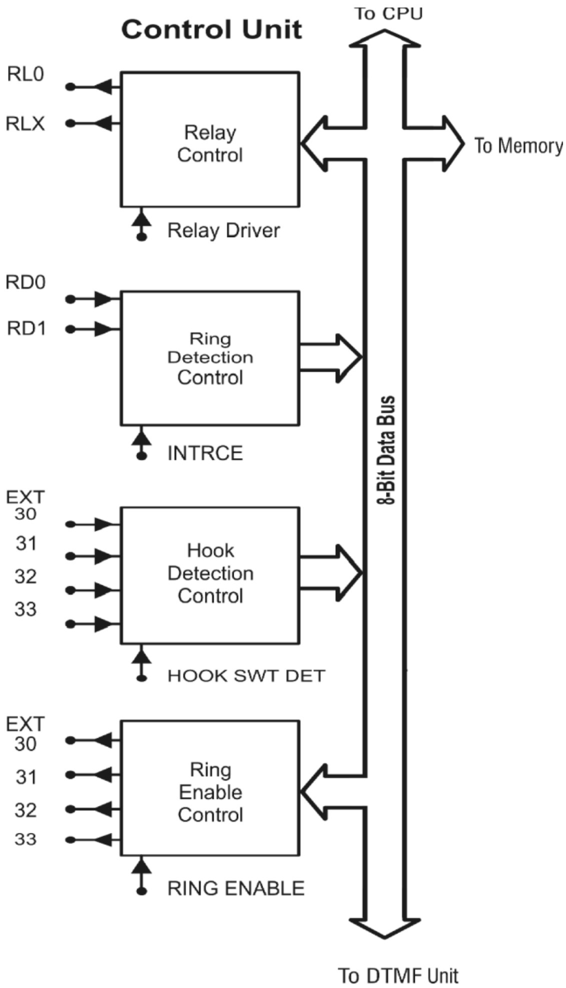
To study Control Unit and analyze the Ring Enable Control Signals

Control Unit / Signal Processor Unit

This unit consists of digital IC's. It serves as an interpreter between CPU and the analog circuitry. The functions performed are.

- Receiving supervisory 'On'/'Off' Hook, Flash signals and decadic dial pulses from extension DC loops.
- Controlling ringing towards extensions and providing automatic ring trip when extension goes off hook.
- Recognizing incoming ring from trunk/ direct lines.
- Controlling out pulsing on out going calls.
- Relay driving signals.

The signaling information received from the terminal card is encoded into bidirectional 8 bit data bus.



Procedure:

Note: We have already performed the experiments related to Relay Control, Ring detection, Hook detection control in the Trunk / Extension unit section.

1. Connect the Mains cord to the System and switch the power ‘On’.
2. Lift the handset of EXT 30 and observe the status of corresponding LED at ‘Hook Detection Control’ section.
3. Dial the EXT 32 and observe the status of corresponding LED at ‘Ring enable Control’ section.
4. Observe the signal at Test point 18 of “Ring Enable Control’ section and also measure the voltage at the same test point; it will be 4V DC approximately.

Working:

The Ring volatge signal consisting of high ‘–ve’ voltage and AC RMS is all the time generated in the exchange and forwarded through control signals.

The signal generated by the Ring Control Unit after sensing the data bus signals is fed to opto-coupler (pin 1) of extension unit. The pin 2 of the coupler is connected with hook detection pin. If the extension is off hook a voltage of 4 V will be available at pin 2 as a result the opto-coupler will not conduct. Whereas when the extension is On hook no voltage will be available at pin 2, The opto-coupler will conduct as soon as signal is available at its pin 1.

Experiment 4b

Objective:

To study Decode unit & DTMF unit and analyze the DTMF signal

Decoder Unit

This unit also termed as demultiplexer unit it is often integrated in the CPU unit or card.

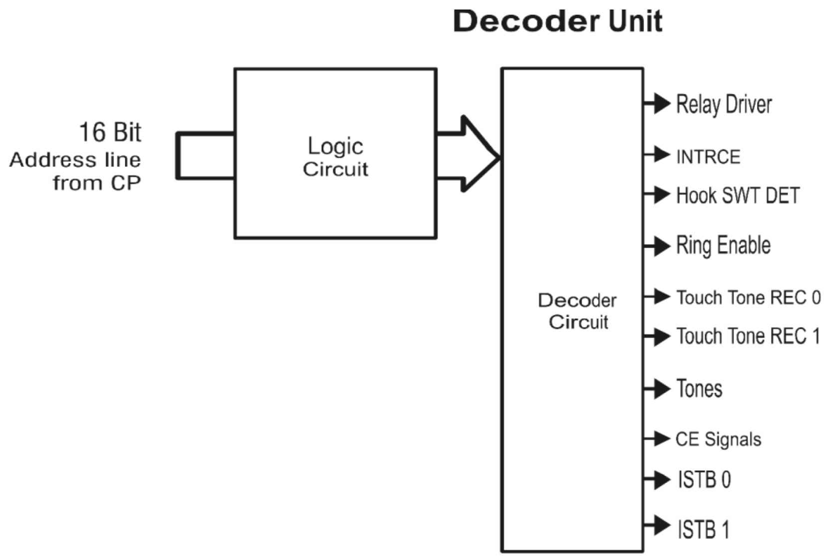


Figure 10: Block Diagram of decoder unit

This unit provides the control and enable signal to the various units. The control signals relay driver, Intrce, Hook switch detection, Ring enable Touch tone rec 0 and 1, tones are signals kept high as soon as power is switched 'On'. Whereas the CE signals that chip enable signals are activated on particular function. The ISTB 0 and 1 are voltages provided to switching matrix through digital circuit.

DTMF Unit

The Dual Tone Multi Frequency Unit it consists of DTMF receiver ICs 2 number the telephone keypad is divided between two sets lower and higher frequencies. When a digit is pressed the two frequencies is sent out to reach the exchange.

Note: More experiments of dialing can be performed with DTMF Telephone Trainer **Scientech 2654**.

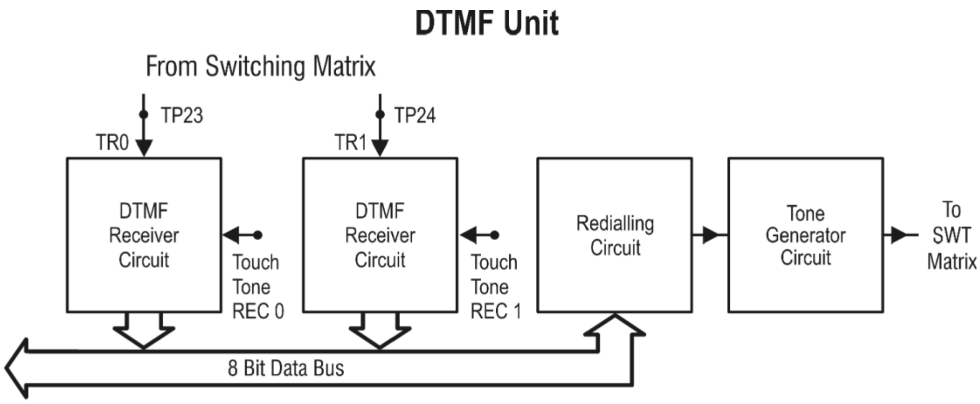


Figure 11: Block diagram of DTMF unit

At the exchange the DTMF unit decodes the frequencies with the help DTMF receiver IC's which consist of a low pass and high pass filter to differentiate the frequencies.

After detection the DTMF tone pairs are given as a 4 bit code to data bus for CPU. The numbers of DTMF receivers are limited to the number of extensions served with the dial tone at the same time.

Procedure:

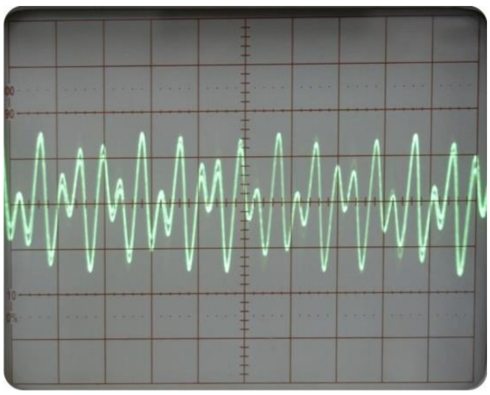
1. Connect the Mains cord to the System and switch the power 'On'.

2. Connect the Oscilloscope at Test Point 30

3. Lift the handset of EXT 30.

4. Dial any one digit say 3.

5. Observe DTMF signal on Oscilloscope which is a combination of two frequencies (keep the digit pressed long for better result).



Showing two frequencies with digit pressed

Experiment 4c

Objective:

To study the Ringer Unit and analyze the Ringer Voltage generation

Ringer Unit

The ringing voltage is all the time generated in the exchange and extended to subscriber whenever ringing is required. The ringing voltage is a 120Vpp, 20 – 40 Hz AC signals.

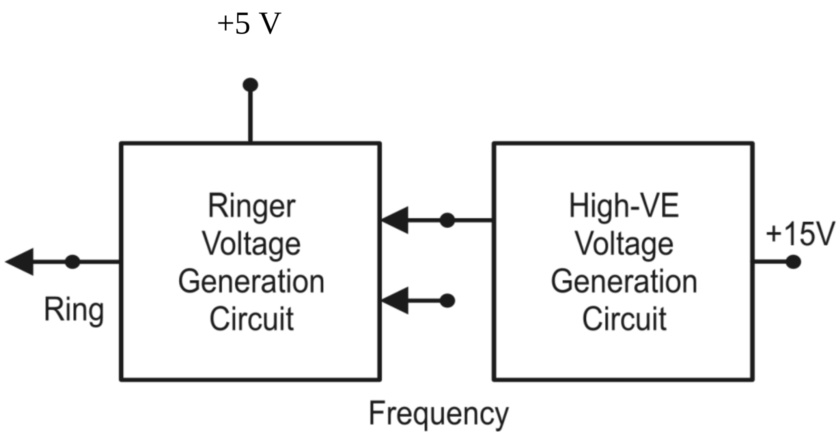
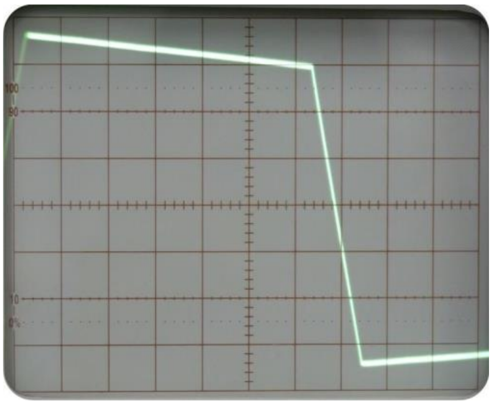


Figure 12: Block diagram of ringer unit

The ringer unit consists of a high negative DC voltage generation circuit -110 to -140V. This is used for generation of AC 120Vpp. A frequency of 20 – 40 Hz is also provided to this unit from the clock generation unit. The ringing signal is available at pin 4 of opto-coupler of each extension circuit. The ring enable signal under the supervision of CPU will make the coupler to conduct only if the extension is ON hook and no voltage is available at pin 2 of opto-coupler.

Procedure:

1. Connect the Mains cord to the System and switch the power ‘On’.
2. Measure the high (-ve) voltage generated at Test point 21. It will be -120V DC supply approximately.
3. Observe the frequency of the signal at Test point 22 on Oscilloscope. The frequency will be 20/ 40 Hz approximately.
4. Observe the Ringer Voltage signal at Test point 20 on Oscilloscope. It will be approximately -120Vpp AC signal with the same frequency 20/ 40 Hz approximately. (See figure below)



Ringer Voltage signal

5. Observe the status of the Ring LED glowing all time showing that the ringing voltage is all the time generated in the exchange and extended to subscriber whenever ringing is required.
6. Lift the handset of EXT 30 and dial EXT 32 and observe the ring tone, Ring Enable Control LED and status LED of EXT 32.
7. Now switch the Fault 3 to ‘on’ position.
8. Dial EXT 32 from EXT 30 and observe that Ring Enable Control LED and status LED of EXT 32 are working same but no ring tone is heard in EXT 32.
9. Also the Ringer Voltage Generation Circuit LED is not blinking and no voltage at Test point 20.
10. **Switch Fault 3** causes to disconnect the 20/ 40 Hz frequency signal from the network.
11. Observe the difference in voltage by switching the Fault 3 to ‘off’ position. Now make the call from EXT 30 to EXT 32 and check the functionality.

Experiment 4d

Objective:

To study the Clock Generation and Tone Generation Units and analyze the role of different tones

Clock Generation Unit

The 24 MHz clock generated by the crystal is divided by using the flip flop IC 74HC 74 (also served to CPU) and the frequency of 4 MHz is generated. The 4 MHz signal is also fed to dual BCD counter, which is used as frequency divider. The output at pin 6 is 400 KHz which is by dividing the input frequency 4 MHz by 10 which is again fed to next counter in the same IC. So 400 KHz again divided by 10 to achieve 40- 50 KHz (approximately) and also by 2 for 20 KHz.

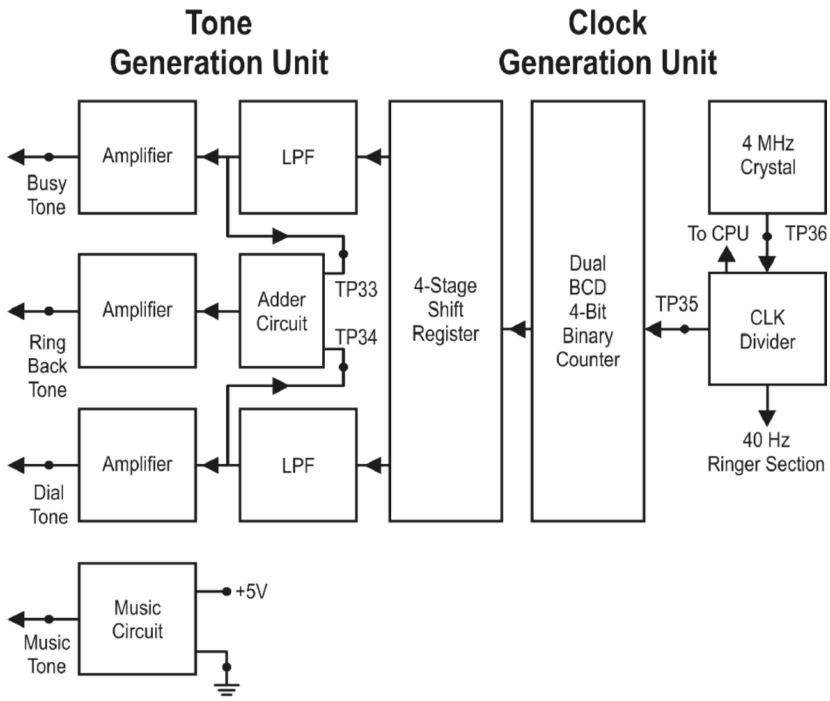


Figure 13: Block Diagram of Clock / Tone generation unit

This 20 KHz signal is fed to next counter as input the output of this counter is 200 Hz achieved by dividing 20 KHz by 10 for 2 KHz and again 2 KHz by 10. The 200 Hz is fed to next counter for dividing it by 5 to achieve final 40 Hz. This 40 Hz is fed to the ringer section for ring generation.

The 40-50 KHz (approximately) is given to the tone generation unit.

Note: In some exchanges even 20 Hz is used a ringer frequency.

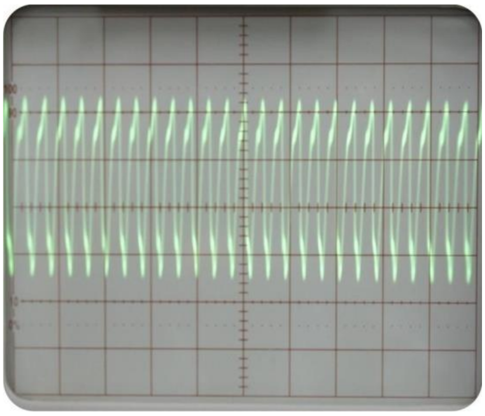
Tone Generation Unit

The 40 – 50 KHz signal is given as input to two synchronous binary counters The counters U1 gives out a signal of 3.3 KHz that's 40 KHz divided by 12 = 3.3 KHz. The counter U2 output is divided by 13 and the output is 3. 07 KHz (approximately).After inverting through a Not gate both the signal are fed to a Dual Shift Register U3, Serial in parallel out. The first shift register gives output by dividing by 8 that's *420-460 Hz* approximately as busy tone frequency .The second shift register output is by dividing by 8 that's *380 - 400 Hz* that's *dial tone* frequency. The LPF has been connected to the outputs after which the signal is amplified using an op- Amp.

The DT and BT are mixed to produce the Ring Back Tone (RBT) *480 Hz* of after amplification. The three tones are every time generated and available at the switching matrix extended to the extensions whenever required.

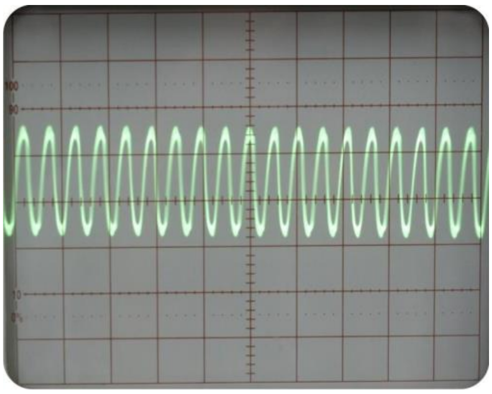
Procedure:

1. Observe the Clock frequency at Test point 36. It will be a 12 MHz clock signal.



Clock Frequency 12 MHz

2. Observe the clock divider frequency at Test point 35. It will be a 4 MHz clock signal.



Clock Frequency 12 MHz

3. Observe the frequency at Test point 22. It will be a 20 Hz signal.
4. Lift the handset of EXT 30 and dial EXT 32 and observe the Ring tone, Ring Enable Control LED and status LED of EXT 32.
5. Also observe the Ringer Voltage Generation Circuit LED and the voltage at Test point 20 (- 56 V DC approx).
6. Now switch the Fault 3 to ‘on’ position and dial EXT 32 from EXT 30.
7. Observe that Ring Enable Control LED and status LED of EXT 32 are working same but no ring tone is heard in EXT 32.
8. Observe only ring back tone is heard at EXT 30 but no ringing at EXT 32.
9. Also the Ringer Voltage Generation Circuit LED is not blinking and no voltage at Test point 20.
10. **Switch Fault 3** causes to disconnect the 20/ 40 Hz frequency signal from the network. Observe the difference in voltage by switching the Fault 3 to ‘off’ position. Now make the call from EXT 30 to EXT 32 and check the functionality.

Experiment 4e

Objective:

To study and analyze that ring back tone is addition of Dial Tone and Busy Tone.

Procedure:

1. Observe the Busy Tone at Tone Generation Unit Test point 33.
2. Observe the Dial tone at Tone Generation Unit Test point 34.
3. Now lift the handset of EXT 30 and dial EXT 32 and observe the functionality of the system.
4. Observe the ring back tone at Test point 30 as shown in figure 14.
5. Put the handsets of EXT 30 and EXT 32 back to their places.
6. Now switch the Fault 4 to ‘On’ position (to remove Dial Tone from Ring Back Tone).
7. Lift the handset of EXT 30 and observe that there is no dial tone because of switch fault.
8. Now dial EXT 32 and observe that it is ringing but Ring back tone is again not heard at EXT 30. The communication can be done between the two properly.
9. **Switch Fault 4** causes to disconnect the dial tone, check the dial tone at test point 34 and as the result no ring back tone will be there at the output of adder and amplifier circuit, check the Ring back tone at test point 30.
10. Observe the difference by switching the Fault 4 to ‘off’ position. Now repeat the action done above in the absence of fault to verify.

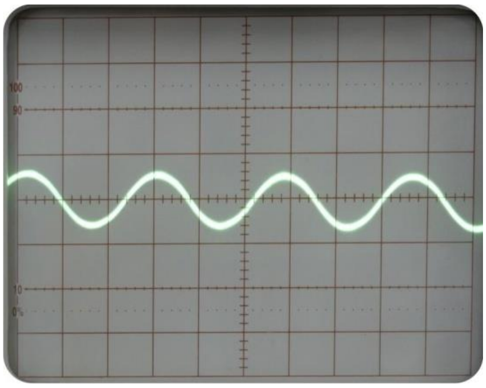
Experiment 4f

Objective:

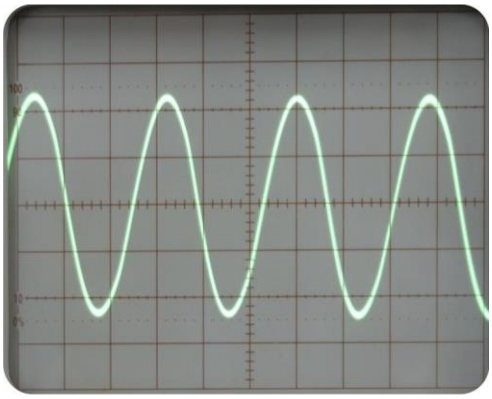
To study and measure the frequencies of Dial Tone and Busy Tone

Procedure:

1. Connect the frequency counter at Test point 31 to measure DT of 380-400 Hz approximately.
2. Connect the frequency counter at Test point 33 to measure BT of 430-460 Hz approximately.
3. Connect the Oscilloscope at Tone Generation Circuit Test point 34 and observe the Dial Tone signal as shown in figure.
4. Now connect the Oscilloscope at Test point 31 and observe the amplified Dial Tone signal.



Showing dial tone before amplification



Showing amplified dial tone

Experiment 4g

Objective:

To study and analyze the Music Tone

Music Circuit:

It can be commercially available melody tone or voice circuit. Or it can be with recording facility to the subscriber for customized service. Like other tones this is also all the time generated in the exchange and extended to the required extension through switching matrix.

Procedure:

1. Lift the handset of EXT 30, dial EXT 32.
2. Answer the EXT 32, and keep ‘On’ hold.
3. Press the ‘Flash’ button on the telephone instrument of EXT 30.
4. Hear the Music Tone at EXT 32, since its ‘On’ hold.
5. Observe the same at Test point 32 by connecting Oscilloscope.

Test points

Test point 1	:	12 VDC normal and when accessed 0V and short to GND.
Test point 2	:	30V
Test point 3	:	‘On’ Hook – between 40 to 60V / ‘Off’ Hook 8-10V
Test point 4	:	30V
Test point 5	:	‘On’ Hook – between 40 to 60V /‘Off’ Hook 8-10 V
Test point 6	:	‘On’ Hook – 8.5V/ ‘Off’ Hook 7.5V and DT Sine wave, Speech Signals.
Test point 7	:	‘On’ Hook 8.5V / ‘Off’ Hook 7.5V and DT Sine wave, Speech Signals.
Test point 8	:	‘On’ Hook 58V/‘Off’ Hook 8.0 V
Test point 9	:	‘On’ Hook 58V/‘Off’ Hook 8.0V
Test point 10	:	1-2 V DC When Trunk Accessed
Test point 11	:	Incoming Call 2-4 V DC
Test point 12	:	‘On’ Hook 0 VDC /‘Off’ Hook 2-5 VDC
Test point 13	:	‘On’ Hook 0 VDC /‘Off’ Hook 2-5 VDC
Test point 14	:	‘On’ Hook 0 VDC /‘Off’ Hook 2-5 VDC
Test point 15	:	‘On’ Hook 0 VDC /‘Off’ Hook 2-5 VDC
Test point 16 to 19	:	‘On’ Hook / ‘Off’ Hook – 0V / Incoming Call 4 V
Test point 20	:	60 – 70 V RMS 20/40 Hz AC Signal (All Time)
Test point 21	:	-120 VDC
Test point 22	:	20-40 Hz Ringer Frequency
Test point 23 – 24	:	DTMF Frequency when digit pressed
Test point 25	:	Speech signals when accessed, 8V Constant ‘On’/‘Off’ Hook
Test point 26-29	:	‘On’ Hook– 8 V, ‘Off’ Hook- 8 VDC and DT Sine wave, Speech signals.
Test point 30	:	Ring Back Tone – 500 Hz (460-480 Hz) approximately
Test point 31	:	Dial Tone 2V Sine wave Amplified 384 Hz
Test point 32	:	Music Tone 740 Hz approximately
Test point 33	:	Busy Tone Sine wave 416 Hz
Test point 34	:	Dial Tone 1V Sine wave before Amp 384 Hz
Test point 35	:	4 MHz Clock
Test point 36	:	Sine wave 12 MHz

Glossary

Central Office: Refers to either a telephone company switching centre or the type of telephone switch used in a telephone company switching centre.

Dual Tone Multi frequency: A signaling system that sends pairs of audio frequencies to represent digits on a telephone keypad. It is often used interchangeably with the term Touchtone.

EPABX / PABX: (Electronic Private Automatic Branch Exchange) is a privately owned telephone switching system for handling multiple telephone lines without having to pay the PSTN Company to lease each line separately.

Erlang B: A formula developed by A.K. Erlang, widely used to determine the number of trunks required to handle a known calling load during a one hour period. The formula assumes that if callers get busy signals, they go away forever, never to retry (lost calls cleared). Since some callers retry, Erlang B can underestimate trunks required. However, Erlang B is generally accurate in situations with few busy signals. **Erlang C**: Calculates predicted waiting times (delay) based on three things: the number of servers (reps); the number of people waiting to be served (callers); and the average amount of time it takes to serve each person. It can also predict the resources required to keep waiting times within targeted limits. Erlang C assumes no lost calls or busy signals, so it has a tendency to overestimate staff required.

Erlang, A.K.: A Danish engineer who worked for the Copenhagen Telephone Company in the early 1900s and developed Erlang B, Erlang C and other telephone traffic engineering formulas.

Erlang: One hour of telephone traffic in an hour of time. For example, if circuits carry 120 minutes of traffic in an hour, that's two Erlangs.

Gateway: A server dedicated to providing access to a network.

Grade of Service: The probability that a call will not be connected to a system because all trunks are busy. Grade of service is often expressed as "p.01" meaning 1% of calls will be "blocked." Sometimes, grade of service is used interchangeably with service level, but the two terms have different meanings. See Service Level.

Experiment No. 5

Name of the Experiment: Simulation study of Frequency Division Multiplexing (FDM) using MATLAB

Objective: To study the principle of FDM system

Theory: Frequency Division Multiplexing (FDM) is a multiplexing technique in which several signals are combined into a composite signal (frequency sharing basis) for transmission over a common transmission medium. Any type of modulation can be used in FDM as long as carrier spacing is sufficient to avoid spectral overlapping. FDM is used in telephone system, telemetry, commercial radio and TV broadcasting and communication networks.

MATLAB code:

```
clear all
close all
%PARAMETERS
bw = 4000; % bandwidth for each frequency band in Hz
guard_band = 300; %
signal_to_noise_ratio = 20; % SNR in dB
modulation_ssb = 1; % 1 for Single SSB modulation, 0 for AM
freq_carrier1 = bw*3; %frequencies in Hz
freq_carrier2 = bw*4;
freq_carrier3 = bw*5;
Fs = freq_carrier3*2+5000; %Fs = 44100;
passband_freq = 2500;
graphics = 1;
sounds = 1;
%
[B,A] = butter(4,passband_freq/(Fs/2));
pass_band = @(S) filter(B,A,S);
[C1,D1] = butter(2,[bw*2+guard_band bw*3-guard_band]/(Fs/2));
filter_band3 = @(S) filter(C1,D1,S);
```

```
[C2,D2] = butter(2,[bw*3+guard_band bw*4-guard_band]/(Fs/2));
filter_band4 = @(S) filter(C2,D2,S);
[C3,D3] = butter(2,[bw*4+guard_band bw*5-guard_band]/(Fs/2));
filter_band5 = @(S) filter(C3,D3,S);
% 3 different input signals
s1 = wavread('weird.wav');
longitude1 = length(s1);
s2 = wavread('bird.wav');
longitude2 = length(s2);
s3 = wavread('wobble.wav');
longitude3 = length(s3);
% beep signal to separate input signals
beep = wavread('squeaky.wav');
playerbeep = audioplayer(beep,44100);
%
longitude_minima = min([longitude1 longitude2]);
t = linspace(0,5, longitude_minima);
s1 = s1(1:longitude_minima);
s2 = s2(1:longitude_minima);
s3 = s3(1:longitude_minima);
FLAG = input('step1. Press Enter to play input signals with pause-beep and spectrums:\n' );
if (sounds > 0)
player = audioplayer(s1,44100);
playblocking(player);
playblocking(playerbeep);
player2 = audioplayer(s2,44100);
playblocking(player2);
playblocking(playerbeep);
player3 = audioplayer(s3,44100);
playblocking(player3);
end
if (graphics > 0)
figure
esps1=abs(fft(s1));
subplot(3,1,1),plot(esps1),grid on,zoom,title('Spectrum of s1');
esps2=abs(fft(s2));
subplot(3,1,2),plot(esps2),grid on,zoom,title('Spectrum of s2');
esps3=abs(fft(s3));
```

```
subplot(3,1,3),plot(esps3),grid on,zoom,title('Spectrum of s3');
end;
%
s1 = pass_band(s1);
s2 = pass_band(s2);
s3 = pass_band(s3);
FLAG = input('step2. Press 1 for SSB modulation OR Press 0 for Standard AM:\n');
if ( modulation_ssb > 0)
s1mod = ssbmod(s1,freq_carrier1,Fs);
s2mod = ssbmod(s2,freq_carrier2,Fs);
s3mod = ssbmod(s3,freq_carrier3,Fs);
else
s1mod = ammod(s1,freq_carrier1,Fs);
s2mod = ammod(s2,freq_carrier2,Fs);
s3mod = ammod(s3,freq_carrier3,Fs);
end
%
fs1 = s1mod;
fs2 = s2mod;
fs3 = s3mod;
fdm_signal = fs1+fs2+fs3;
FLAG = input('step3. Press Enter to see spectrum of Frequency Division Multiplexed (FDM)
signal:\n' );
if (graphics > 0)
figure
esps11=abs(fft(fdm_signal));
subplot(2,1,1),plot(esps11),grid on,zoom,title('Spectrum of transmitted FDM signal');
end
fdm_signal = awgn(fdm_signal, signal_to_noise_ratio );
if (graphics > 0)
esps22=abs(fft(fdm_signal));
subplot(2,1,2),plot(esps22),grid on,zoom,title('Spectrum of received FDM signal(considering
AWGN channel)');
end
FLAG = input('step4. Press Enter to play Transmitted FDM signal and Received FDM signal
after pause-beep:\n' );
if (sounds > 0)
player11 = audioplayer(esps11,44100);
```

```
playblocking(player11);
playblocking(playerbeep);
player22 = audioplayer(esps22,44100);
playblocking(player22);
end
demuxs1 = filter_band3(fdm_signal);
demuxs2 = filter_band4(fdm_signal);
demuxs3 = filter_band5(fdm_signal);
FLAG = input('step5. Press 1 for SSB demodulation OR Press 0 for AM demodulation:\n' );
if ( modulation_ssb > 0)
demods1 = ssbdemod(demuxs1, freq_carrier1,Fs);
demods2 = ssbdemod(demuxs2, freq_carrier2,Fs);
demods3 = ssbdemod(demuxs3, freq_carrier3,Fs);
else
demods1 = amdemod(demuxs1, freq_carrier1,Fs);
demods2 = amdemod(demuxs2, freq_carrier2,Fs);
demods3 = amdemod(demuxs3, freq_carrier3,Fs);
end;
FLAG = input('step6. Press Enter to play reproduced signals with pause-beep and spectrums
at receiver:\n ' );
demods1 = pass_band(demods1);
demods2 = pass_band(demods2);
demods3 = pass_band(demods3);
if (graphics > 0)
figure
esps1=abs(fft(demods1));
subplot(3,1,1),plot(esps1),grid on,zoom,title('Spectrum of reproduced signal s1');
esps2=abs(fft(demods2));
subplot(3,1,2),plot(esps2),grid on,zoom,title('Spectrum of reproduced signal s2');
esps3=abs(fft(demods3));
subplot(3,1,3),plot(esps3),grid on,zoom,title('Spectrum of reproduced signal s3');
end
if (sounds > 0)
player4 = audioplayer(demods1,44100);
playblocking(player4);
playblocking(playerbeep);
player5 = audioplayer(demods2,44100);
playblocking(player5);
```

```
playblocking(playerbeep);
player6 = audioplayer(demods3,44100);
playblocking(player6);
end
```

Procedure:

1. Run the MATLAB Code
2. Observe the spectrum of individual signal
3. Observe the spectrum of multiplexed signal
4. Observe the spectrum with AWGN
5. Observe the spectrum of de-multiplexed signal

Conclusion / Inference:

Experiment No. 6.A

Name of the Experiment: PC to PC communication using serial port (RS-232)

Objective: To study the PC to PC serial communication using RS-232 port

Theory: Serial communications with RS232 is one of the oldest and most widely spread communication methods in computer world. The way this type of communication can be performed is pretty well defined in standards with one exception. The standards show the use of DTE/DCE communication, the way a computer should communicate with a peripheral device like a modem, where DTE means *data terminal equipment* (computers etc.) and DCE is the abbreviation of *data communication equipment* (modems). One of the main uses of serial communication today where no modem is involved i.e. a *serial null modem* configuration with two computers communicate directly.

Equipments Needed:

- Scientech 5001 Data Communication Techniques TechBook
- Scientech 5001 Software
- 3 Patch Cords (2mm)
- 2 Serial Port Cables
- TechBook Power Supply
- Mains Cord

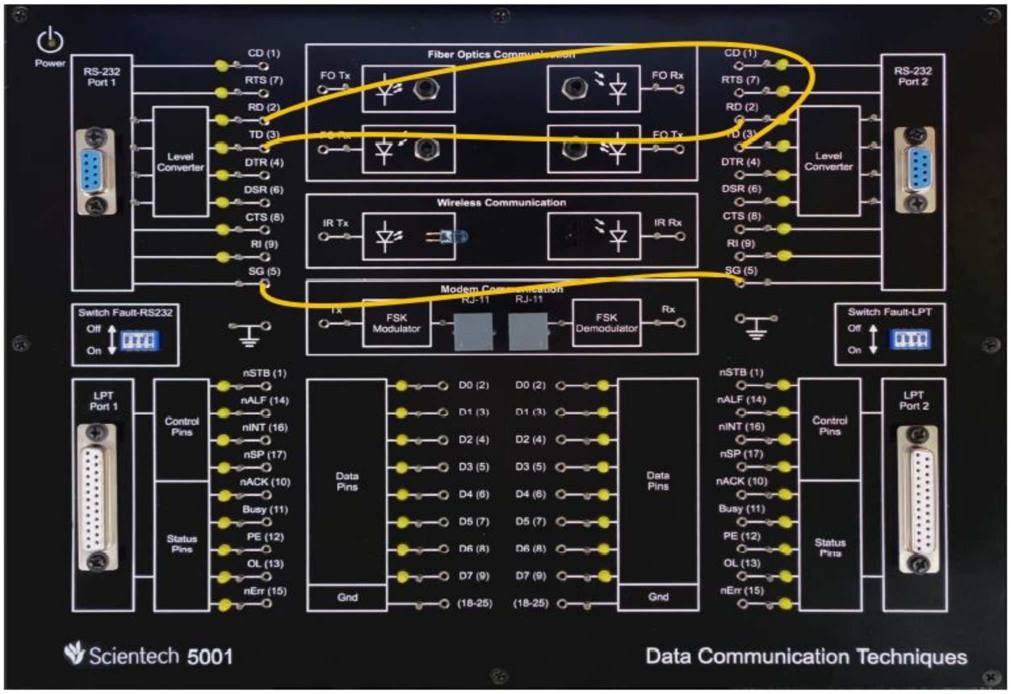
Procedure:

1. Make the connections according to the diagram given below

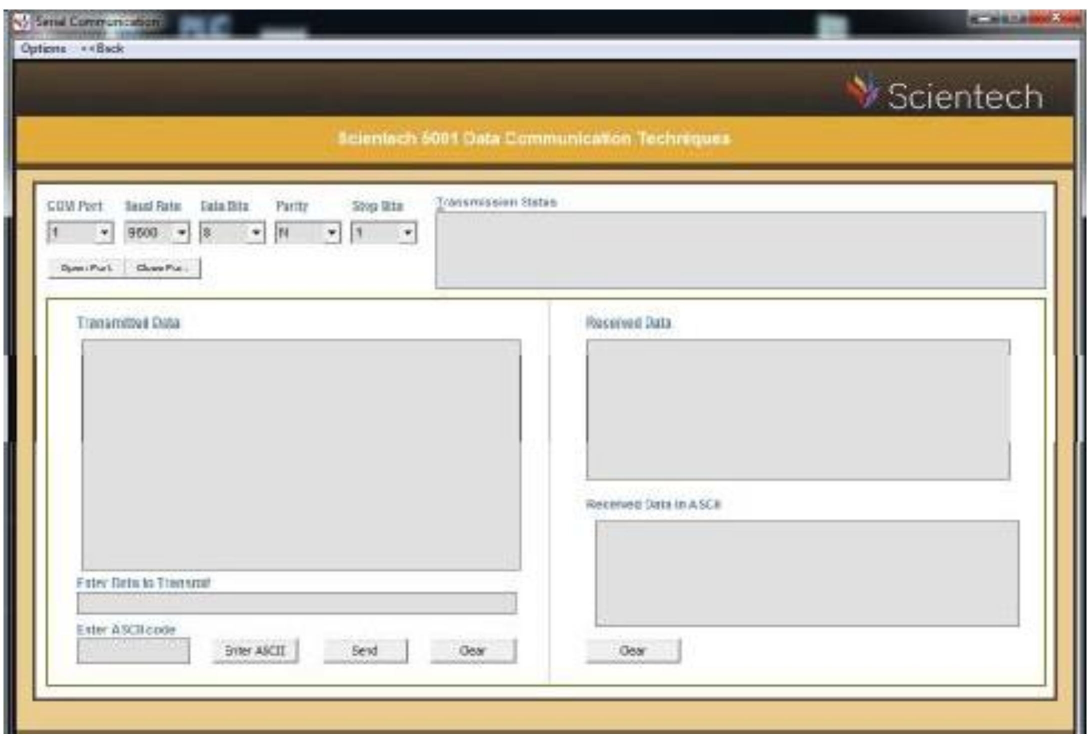
2. Connect the TechBook Power Supply to the Scientech 5001. Turn ON the Rocker switch

3. Connect the Serial port cables between one PC to RS232 Port1 & another PC to RS232 Port2.

4. Open Scientech 5001 Software

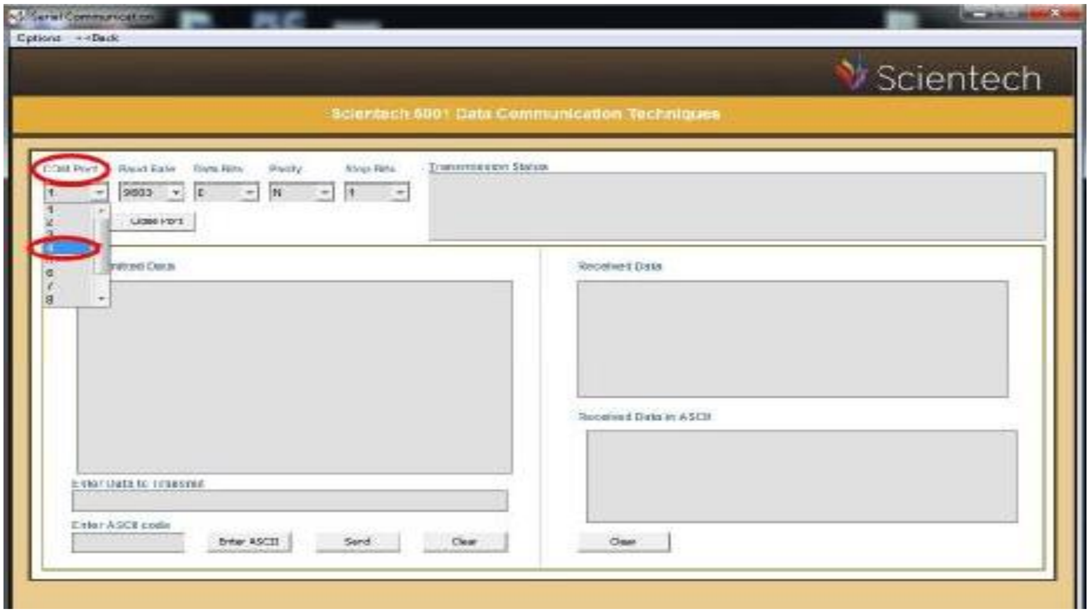


6. The software will open with default settings. Select the desired port settings.

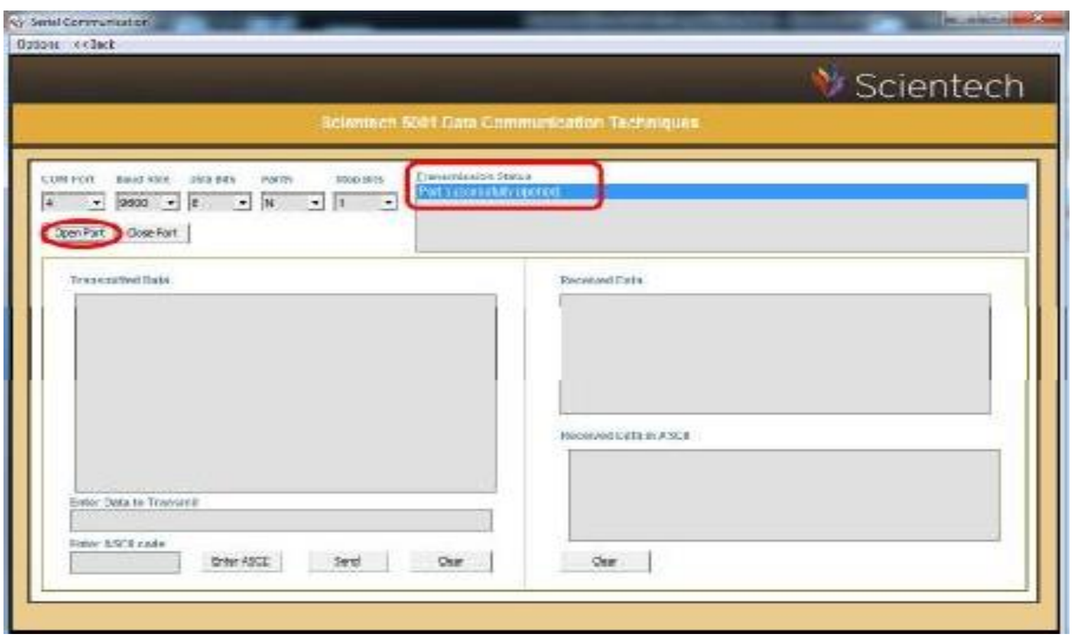


Serial Communication Window

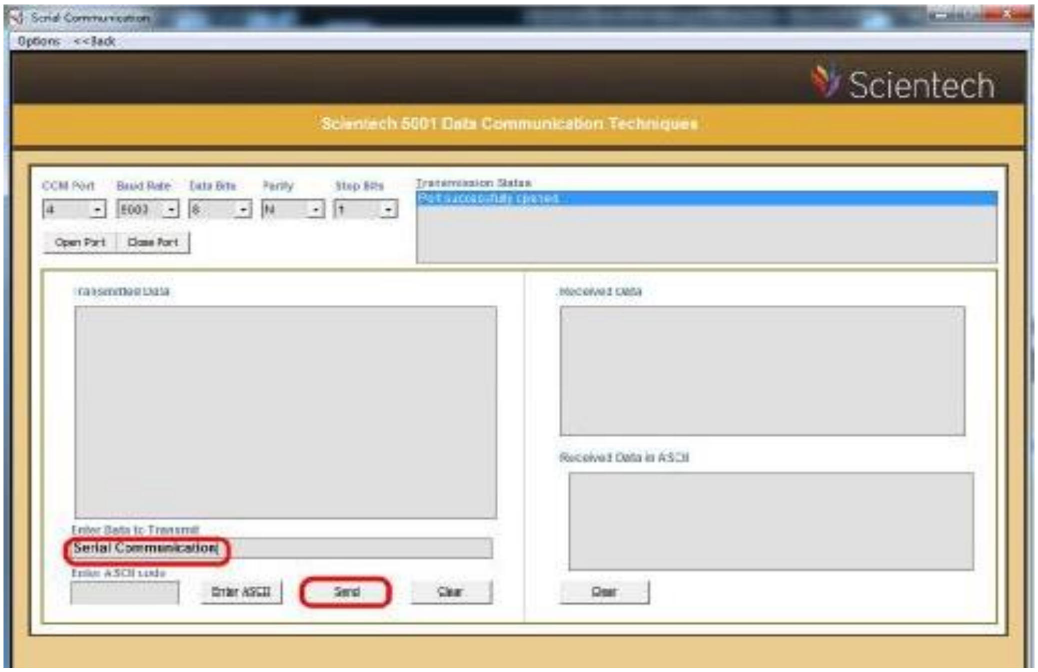
7. Select desired Com Port no. at both Sender and Receiver Node as shown below.



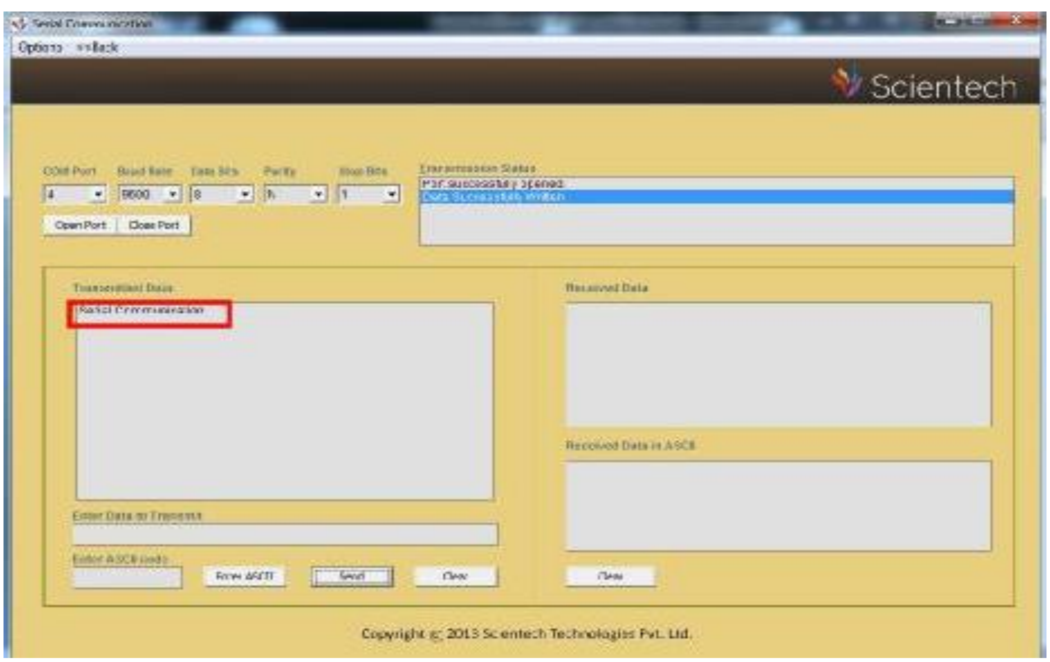
8. After the COM Port No. selection click on Open Port button. When you click on Open Port button, then Port successfully opened message will comes in Transmission Status window as shown below.



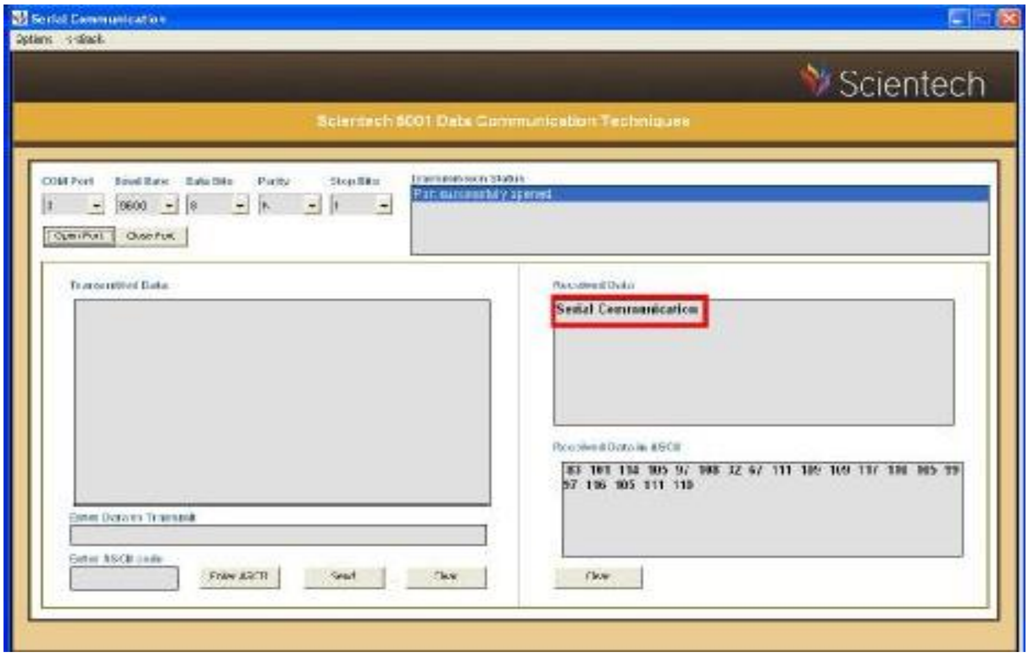
9. To send data to other node enter the text in the window titled as “Enter Data to Transmit”. Click on “Send” button to transmit.



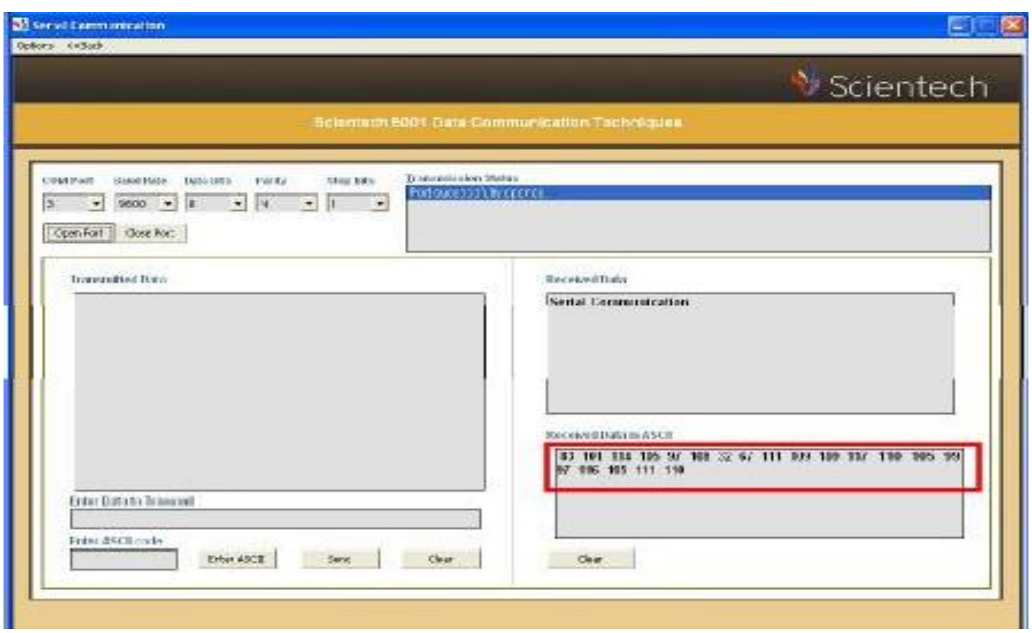
10. Transmit data shown in Transmitted Data Window at Sender Node.



11. Receive Data shown in Received Data window at Receiver Node.

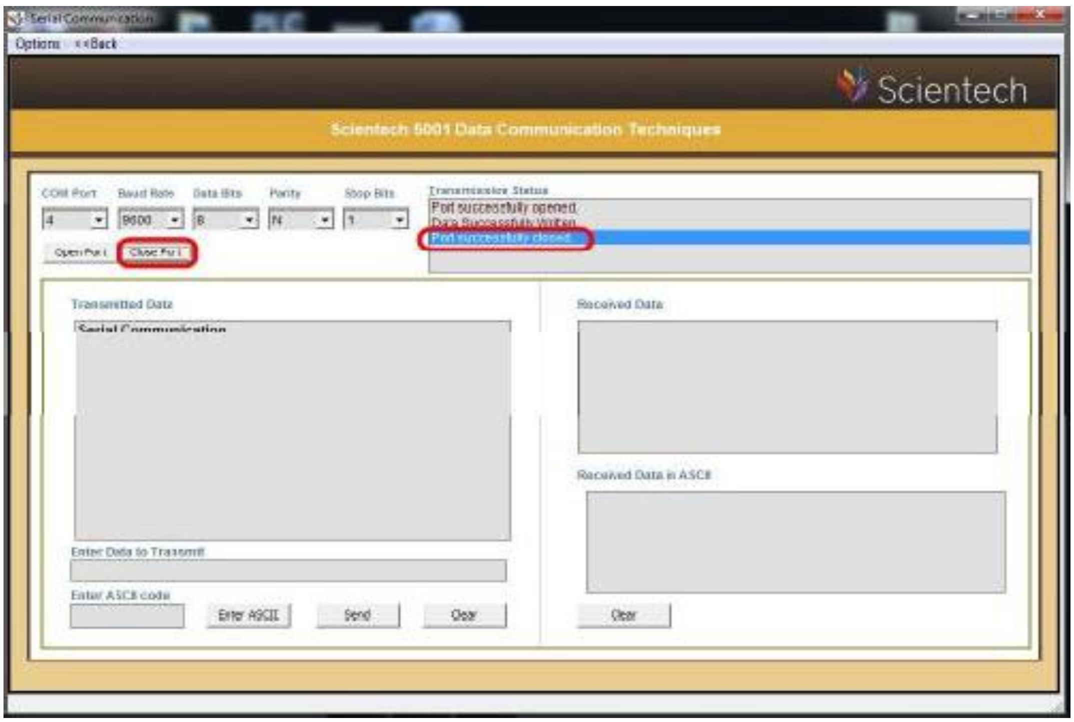


12. Transmitted data will be displayed in the “Received Data” window and the ASCII codes of each byte received is displayed in “Received Data in ASCII” window at Receiver Node.



13. “Clear” buttons have been given to clear all windows.

14. Before changing the port settings close the port by clicking “Close Port” button and after changing the parameters click “Open Port” button to reopen the port.



15. For hardware safety of the CPU, it is very important to connect serial port cables only after making all the connections on board and after switching ON the TechBook.

Conclusion / Inference:

Experiment: 6.B

Objective: Study of Flow Control in Serial Communication

Equipments Needed:

- Scientech 5001 Data Communication Techniques TechBook
- Scientech 5001 Software
- 2 Patch Cords (2mm)
- 2 Serial Port Cables
- TechBook Power Supply
- Mains Cord

Theory:

Because a sender and receiver can't always process data at the same rate, some method of negotiating when to start and stop transmission is required. The Serial Driver supports two methods of controlling serial data flow. One method relies on the serial port hardware, while the other is implemented in software.

Hardware data flow control:

Hardware flow control uses two of the serial port signal lines to control data transmission. When the Serial Driver is ready to accept data from an external device, it asserts the Data Terminal Ready (DTR) signal on pin 1 of the serial port, which the external device receives through its Clear to Send (CTS) input.

RTS/CTS and DTR/DSR Flow Control:

This is "hardware" flow control. Only RTS/CTS flow control will be discussed since DTR/DSR flow control works the same way. To get RTS/CTS flow control one needs to either select hardware flow control in an application program. When a DTE (such as a PC) wants to stop the flow into it, it negates RTS. Negated "Request To Send" (-12 volts) means "request NOT to send to me" (stop sending). When the PC is ready for more bytes it asserts RTS (+12 volts) and the flow of bytes to it resumes. Flow control signals are always sent in a direction opposite to the flow of bytes that is being controlled. DCE equipment (modems) works the same way but sends the stop signal out the CTS pin. Thus its RTS/CTS flow control using 2 lines. On what pins is this stop signal received? That depends on whether we have a DCEDTE connection or a DTE-DTE connection. For DCE-DTE its a straight-through connection so obviously the signal is received on a pin with the same name as the pin its sent out from. Its RTS-->RTS (PC to modem) and CTS<--CTS (modem to PC). For DTE-to-DTE the connection is also easy to figure out. The RTS pin always sends and the CTS pin always receives. Assume that we connect two PCs (PC1 and PC2) together via their serial ports. Then its RTS (PC1) -->CTS (PC2) and CTS (PC1) <-- RTS (PC2). In other words RTS and CTS cross over. Such a cable (with other signals crossed over as well) is called a "null modem" cable

The DTR and DSR Pins:
Just like RTS and CTS, these pins are paired. For DTE-to-DTE connections they are likely to cross over. There are two ways to use these pins. One way is to use them as a substitute for RTS/CTS flow control. The DTR pin is just like the RTS pin while the DSR pin behaves like the CTS pin. DTR flow control is the same as DTR/DSR flow control but its only one-way and only uses the DTR pin at the device. Many text terminals and some printers use DTR/DSR (or just DTR) flow control.

Software data flow control:
Flow control can also be handled in software by using an agreed-upon set of characters as start and stop signals. The Serial Driver supports XON/XOFF flow control, which typically assigns the ASCII DC1 character (also known as control-Q) as the start signal and the DC3 character (control-S) as the stop signal, although you can choose different characters.

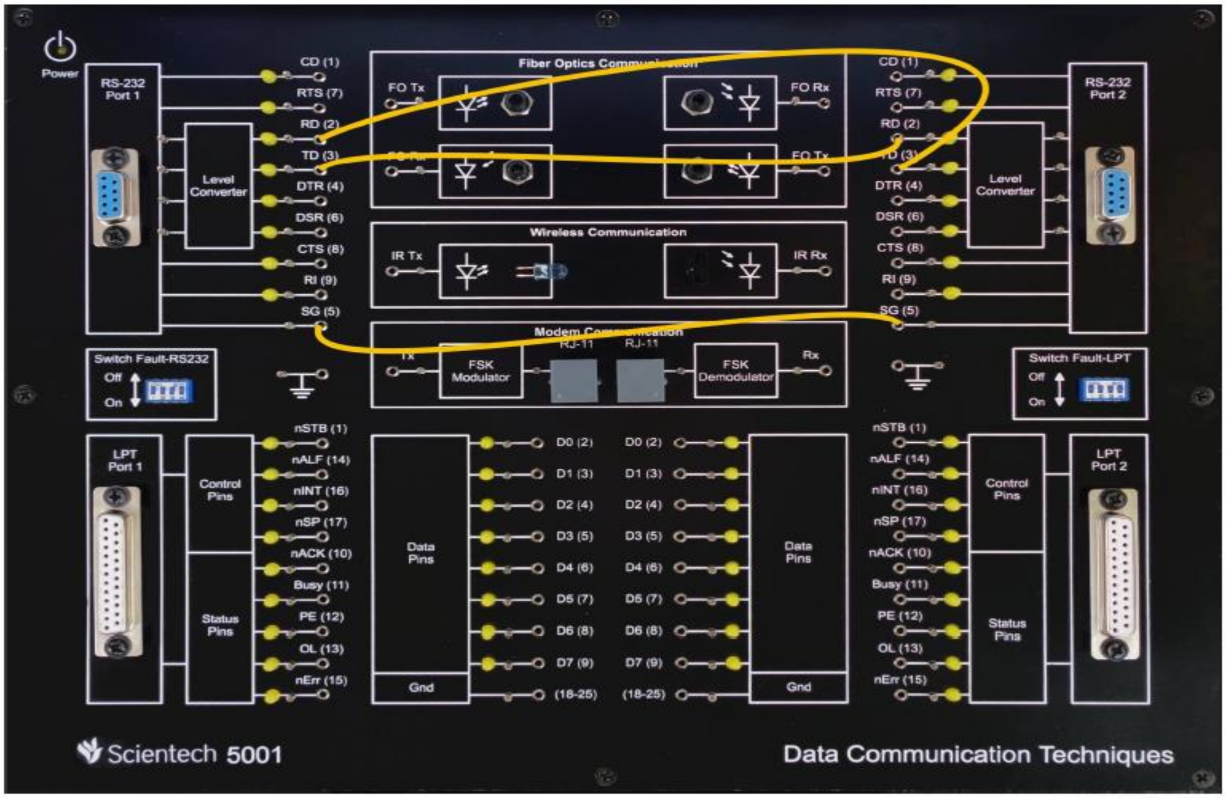
Handshaking:

comXon-Xoff Handshaking:
In this type of handshaking two pins are important DTR & DSR; these are used by both nodes for handshaking while communicating with each other. Basically this is a software type handshaking. These pins are used to indicate X on i.e. transmission ON and Xoff i.e. transmission OFF. When transmission is on DTR pin is set as enabled, as start signal.

comRTS Handshaking:
RTS & CTS pins of Serial port are used as handshaking signal. This is hardware type handshaking.

Procedure:

comXon-Xoff Handshaking:



• Connect the TechBook Power Supply to the Scientech 5001. Turn ON the Rocker switch

• Connect the serial port cables between one **PC to RS232 Port1** & another **PC to RS232 Port2.**

• Run **Scientech 5001** software on both PCs Run **Scientech 5001** software on both PCs and then **>> Click on "Protocols" "Ctrl + T"/click** "Options" on menu bar.

· Click on **Flow Control** button as shown below.

• After Click on **Flow Control** button, Flow **Control in Serial Communication** window will be open. Go to **Option b**utton (in Menu Bar) then go **Select COM** and **Select desired COM Port** at both Sender and Receiver Node.

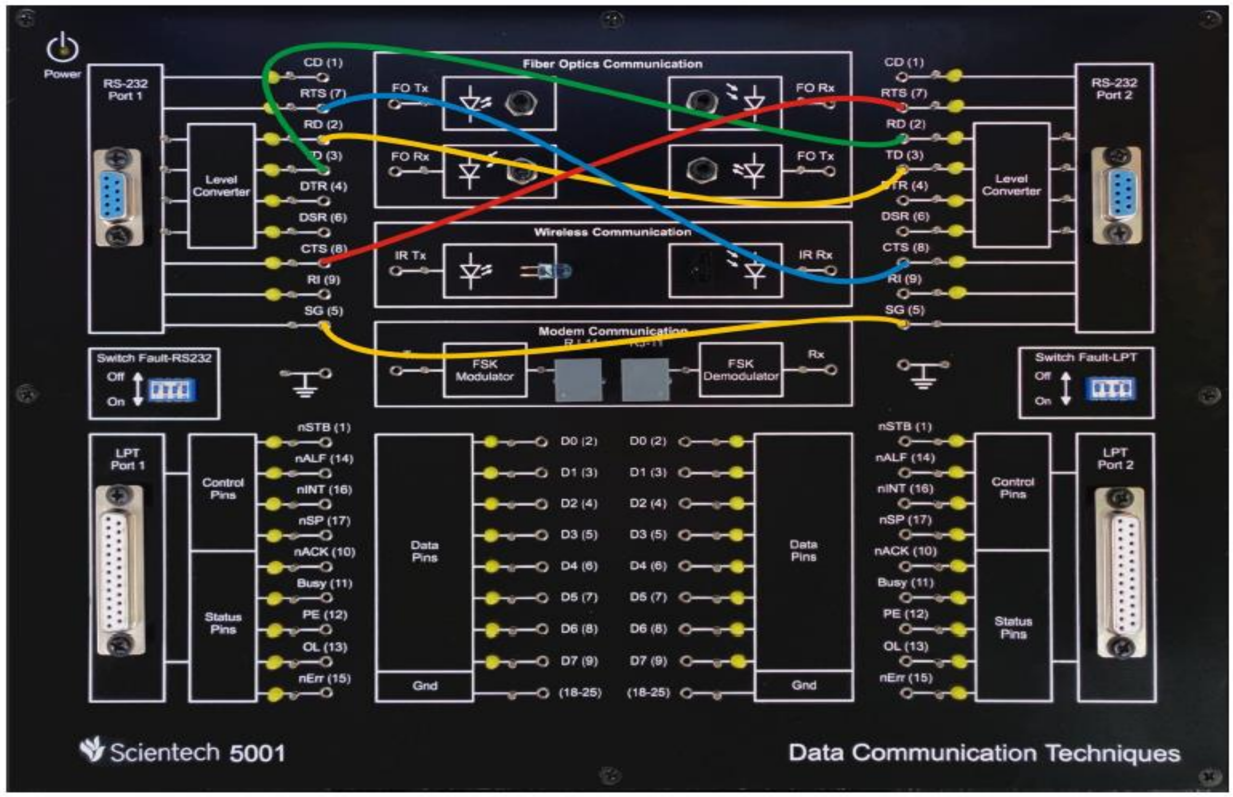
• After Select a **COM Port** No. , **Port Successfully Opened** message will comes as shown below.

• Enter data in the window of sender Node for transmission. After selections click on **"Send"** button.

• Transmit message will comes in Sender window as shown .

• Click on **Receive** button then Message will comes in **Receiver window** as shown .

comRTS Handshaking:
Make a connection according to below given connection diagram.



• Connect the TechBook Power Supply to the **Scientech 5001**. Turn ON the Rocker switch

• Connect the serial port cables between one **PC to RS232 Port1** & another **PC to RS232 Port2.**

• Run **Scientech 5001** software on both PCs. Run **Scientech 5001** software on both PCs and then >> Click on "Protocols" "Ctrl + T"/click "Options" on menu bar. Click on **Flow Control** button.

• After Click on **Flow Control** button, Flow **Control in Serial Communication** window will be open. Go to **Option b**utton (in Menu Bar) then go **Select COM** and **Select desired COM Port** at both Sender and Receiver Node.

• After Select a **COM Port** No. , **Port Successfully Opened** message will comes as shown.

• Click on .comRTS check box and RTS Check box. Enter data in the window of sender side for transmission. After write a message click on **"Send"** button.

• Select **"Write all bytes"** and **"Never Timeout"** and **"Read all bytes"** and **"Never Timeout"** then click on **Receive** button as shown.

• Click on **Receive** button at **Receiver Node**, message will comes in **Receiver** window as shown.

Conclusion / Inference:

Experiment 7.A

Objective: To study the PC to PC Communication using LAN Trainer

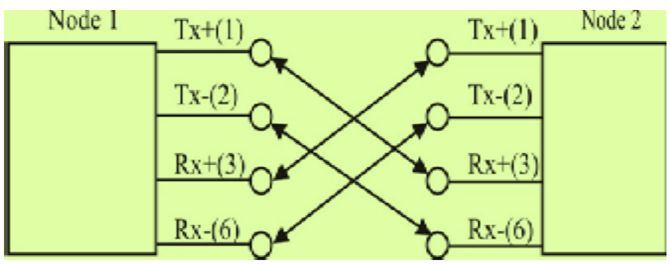
Equipment Needed:

- Scientech 5002A
- Scientech 5002A Software
- 2-CAT5 Cables
- 42 mm Patch Cords

Procedure:

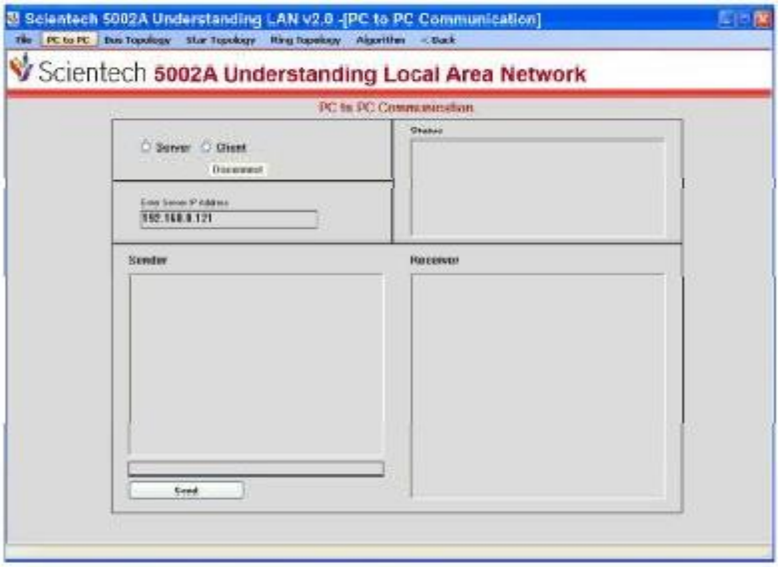
- Connect CAT5 cables between computer (RJ-45 is on back panel of the computer) and RJ-45 connectors on Scientech 5002A.

PC to PC Communication

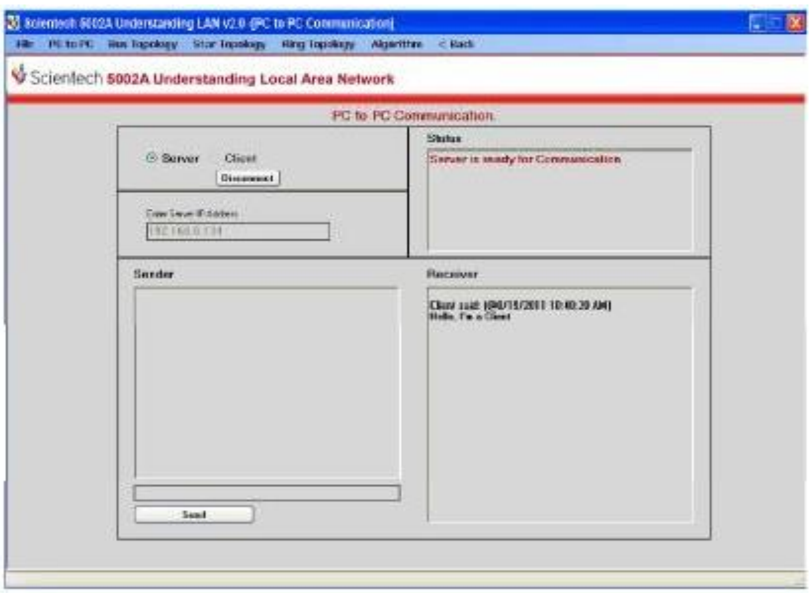


- Connect patch cords between Tx & Rx (Tx+ to Rx+; Tx- to Rx-) for
- both ends. Run **Scientech 5002A** installed software on both nodes
- and follow the steps Click on 'PC to PC Communication'

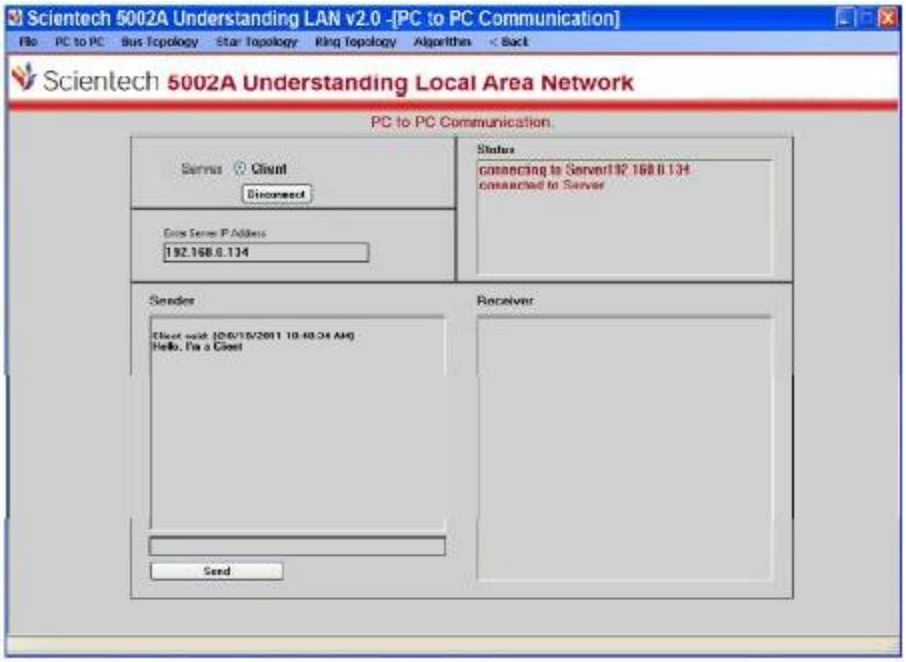
- A window will appear on screen as shown in next figure



- Select ‘Server’ on PC1

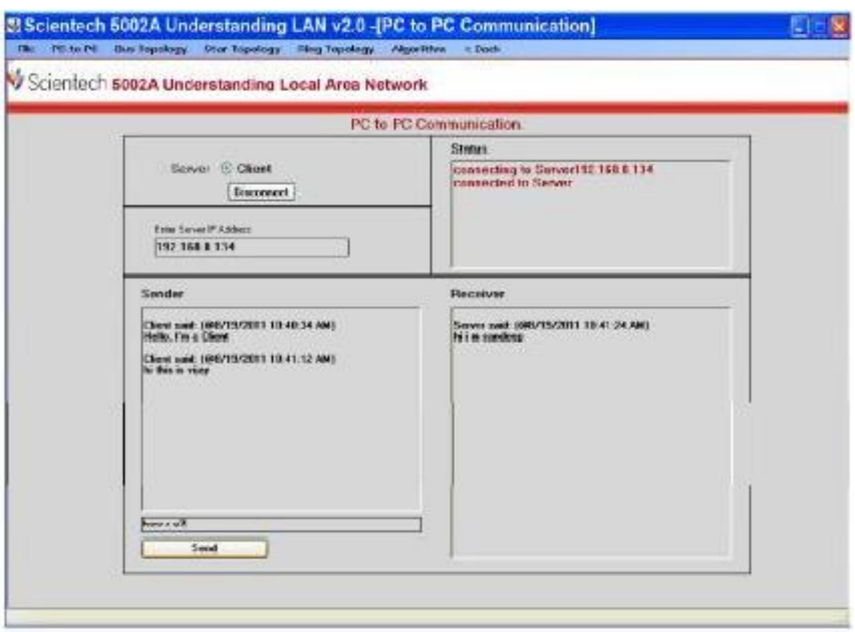


- Enter the IP Address of PC1 into Client Textbox of PC2



- Now you can have a chat application between server
- and client. Enter data here and press 'Enter ↵' or click

on 'Send Data'



Experiment 7.B

Objective: To study the performance of token ring protocols

Equipments Needed:

- Scientech 5002A
- Scientech 5002A software
- 4- DB9 Cables
- 4- 2mm patch cords

Theory:

Token Ring is a network technology first developed by IBM in the 1980s. Although Token Ring is less common but still it is important network access method. In the early 1990s, the Token Ring architecture competed strongly with Ethernet to be the most popular access method. Since that time, the economics, speed, and reliability of Ethernet have improved, leaving Token Ring behind. Because IBM developed Token Ring, a few IBM-centric IT Departments continue to use it. Other network managers have changed their former Token Ring networks into Ethernet networks.

Token Ring networks have traditionally been more expensive to implement than Ethernet networks. Proponents of the Token Ring technology argue that, although some of its connectivity hardware is more expensive, its reliability results in less downtime and lower network management costs than Ethernet. On a practical level, Token Ring has probably lost the battle for superiority because its developers were slower to develop high-speed standards. Token Ring networks can run at 4, 16, or 100 Mbps. The 100-Mbps Token Ring standard, finalized in 1999, is known as *HSTR (High-Speed Token Ring).* HSTR can use either twisted-pair or fiber-optic cable as its transmission medium. Although it is as reliable and efficient, it is still less common than Ethernet because of its higher cost and lagging speed.

Token Ring networks use the token-passing routine and a star-ring hybrid physical topology. In *token passing*, a 3-byte packet, called a token, is transmitted from one node to another in a circular fashion around the ring. When a station has something to send, it picks up the token, changes it to a frame, and then adds the header, information, and trailer fields. The header includes the address of the destination node. All nodes read the frame as it traverses the ring to determine whether they are the intended recipient of the message. If they are, they pick up the data, and then retransmit the frame to the next station on the ring. When the frame finally reaches the originating station, the originating workstation reissues a free token that can then be used by another station. The token-passing control scheme avoids the possibility for collisions. This fact makes Token Ring more reliable and efficient than Ethernet. It also does not impose distance limitations on the length of a LAN segment, unlike CSMA/CD.

On a Token Ring network, one workstation, called the active monitor, acts as the controller for token passing. Specifically, the *active monitor* maintains the timing for ring passing, monitors token and frame transmission, detects lost tokens, and corrects errors when a timing error or other disruption occurs. Only one workstation on the ring can act as the active monitor at any given time.

Token Ring Operation:

Token Ring and IEEE 802.5 are two principal examples of token-passing networks (FDDI is the other). *Token-passing networks* move a small frame, called a token, around the network. Possession of the token grants the right to transmit. If a node receiving the token has no information to send, it passes the token to the next end station. Each station can hold the token for a maximum period of time.

If a station possessing the token does have information to transmit, it seizes the token, alters 1 bit of the token (which turns the token into a start-of-frame sequence), appends the information that it wants to transmit, and sends this information to the next station on the ring. While the information frame is circling the ring, no token is on the network (unless the ring supports early token release), which means that other stations wanting to transmit must wait. Therefore, collisions cannot occur in Token Ring networks. If early token release is supported, a new token can be released when frame transmission is complete.

The information frame circulates the ring until it reaches the intended destination station, which copies the information for further processing. The information frame continues to circle the ring and is finally removed when it reaches the sending station. The sending station can check the returning frame to see whether the frame was seen and subsequently copied by the destination.

Unlike CSMA/CD networks (such as Ethernet), token-passing networks are *deterministic*, which means that it is possible to calculate the maximum time that will pass before any end station will be capable of transmitting. This feature and several reliability features, which are discussed in the section "Fault-Management Mechanisms," later in this chapter, make Token Ring networks ideal for applications in which delay must be predictable and robust network operation is important. Factory automation environments are examples of such applications.

Priority System:

Token Ring networks use a sophisticated priority system that permits certain user-designated, high-priority stations to use the network more frequently. Token Ring frames have two fields that control priority: the priority field and the reservation field.

Only stations with a priority equal to or higher than the priority value contained in a token can seize that token. After the token is seized and changed to an information frame, only stations with a priority value higher than that of the transmitting station can reserve the token for the next pass around the network. When the next token is generated, it includes the higher priority of the reserving station. Stations that raise a token's priority level must reinstate the previous priority after their transmission is complete.

Fault-Management Mechanisms:

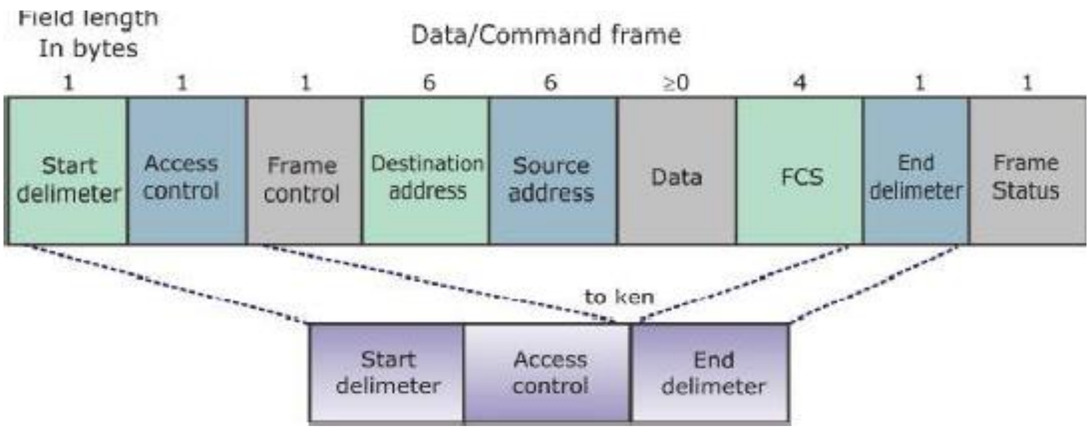
Token Ring networks employ several mechanisms for detecting and compensating for network faults. For example, one station in the Token Ring network is selected to be the *active monitor*. This station, which potentially can be any station on the network, acts as a centralized source of timing information for other ring stations and performs a variety of ring- maintenance functions. One of these functions is the removal of continuously circulating frames from the ring. When a sending device fails, its frame may continue to circle the ring. This can prevent other stations from transmitting their own frames and essentially can lock up the network. The active monitor can detect such frames, remove them from the ring, and generate a new token.

The IBM Token Ring network's star topology also contributes to overall network reliability. Because all information in a Token Ring network is seen by active MSAUs, these devices can be programmed to check for problems and selectively remove stations from the ring, if necessary.

A Token Ring algorithm called *beaconing* detects and tries to repair certain network faults. Whenever a station detects a serious problem with the network (such as a cable break), it sends a beacon frame, which defines a failure domain. This domain includes the station reporting the failure, its nearest active upstream neighbour (NAUN), and everything in between. Beaconing initiates a process called *auto reconfiguration*, in which nodes within the failure domain automatically perform diagnostics in an attempt to reconfigure the network around the failed areas. Physically, the MSAU can accomplish this through electrical reconfiguration.

Frame Format:

Token Ring and IEEE 802.5 support two basic frame types: tokens and data/command frames. Tokens are 3 bytes in length and consist of a start delimiter, an access control byte, and an end delimiter. Data/command frames vary in size, depending on the size of the Information field. Data frames carry information for upper-layer protocols, while command frames contain control information and have no data for upper-layer protocols. Both formats are shown in next figure



Frame Format for Token Ring

The three token frame fields illustrated in next figure are summarized in the descriptions that follow**:**

Start delimiter Alerts each station of the arrival of a token (or data/command frame). This field includes signals that distinguish the byte from the rest of the frame by violating the encoding scheme used elsewhere in the frame.

Starting delimiter	J K 0 J K 0	0	0	J, K non-data symbols (line code)

Access-control byte Contains the Priority field (the most significant 3 bits) and the Reservation field (the least significant 3 bits), as well as a token bit (used to differentiate a token from a data/command frame) and a monitor bit (used by the active monitor to determine whether a frame is circling the ring endlessly).

Access Control

PPP	T	M	RRR

PPP Priority; T Token bit
M Monitor bit; RRR Reservation

End delimiter Signals the end of the token or data/command frame. This field also contains bits to indicate a damaged frame and identify the frame that is the last in a logical sequence.

Data/Command Frame Fields:

Data/command frames have the same three fields as Token Frames, plus several others. The Data/command frame fields illustrated in are described in the following summaries:

Start delimiter Alerts each station of the arrival of a token (or data/command frame). This field includes signals that distinguish the byte from the rest of the frame by violating the encoding scheme used elsewhere in the frame.

Access-control byte Contains the Priority field (the most significant 3 bits) and the Reservation field (the least significant 3 bits), as well as a token bit (used to differentiate a token from a data/command frame) and a monitor bit (used by the active monitor to determine whether a frame is circling the ring endlessly).

Frame-control bytes Indicates whether the frame contains data or control information. In control frames, this byte specifies the type of control information.

Frame Control

F F	Z	Z	Z	Z	Z	Z

FF:frame type
ZZZZZZ: control bit

Destination and source addresses Consists of two 6-byte address fields that identify the destination and source station addresses.

Data Indicates that the length of field is limited by the ring token holding time, which defines the maximum time a station can hold the token.

Frame-check sequence (FCS) Is filed by the source station with a calculated value dependent on the frame contents. The destination station recalculates the value to determine whether the frame was damaged in transit. If so, the frame is discarded.

End Delimiter Signals the end of the token or data/command frame. The end delimiter also contains bits to indicate a damaged frame and identify the frame that is the last in a logical sequence.

Ending delimiter	J K 1 J K	I	E	I: intermediate-frame bit E: error-detection bit

Frame Status Is a 1-byte field terminating a command/data frame. The Frame Status field includes the address-recognized indicator and frame-copied indicator.

Frame Status

A	C	x x	A	C	x x

A: address-recognized bit
xx: undefined
C: frame-copied bit

Summary:

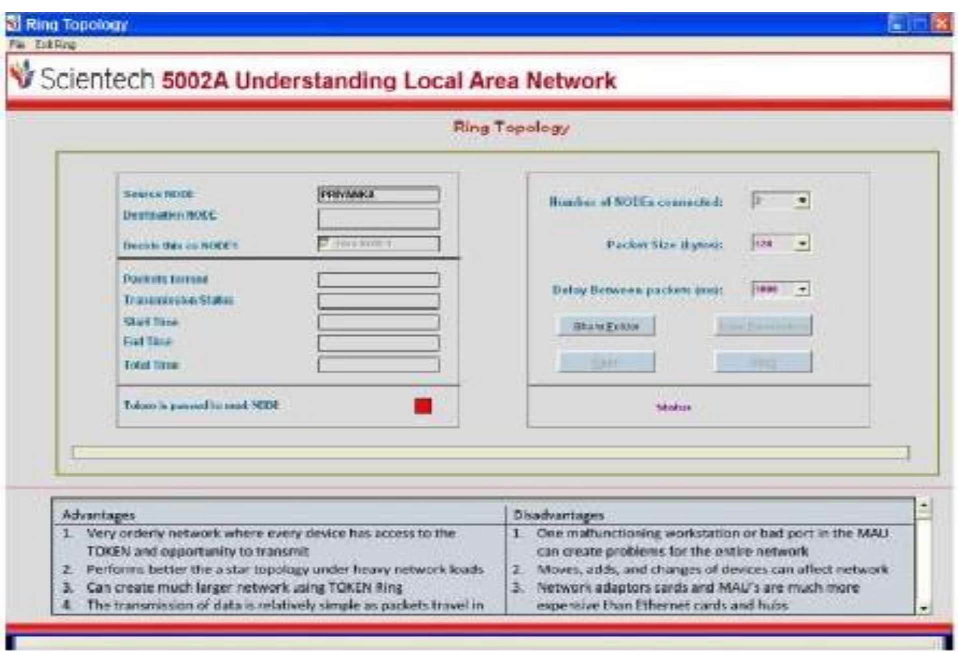
Token Ring technology was developed in the 1970s by IBM. Token-passing networks move a small frame, called a token, around the network. Possession of the token grants the right to transmit. If a node receiving the token has no information to send, it passes the token to the next end station. Each station can hold the token for a maximum period of time.

If a station possessing the token does have information to transmit, it seizes the token, alters 1 bit of the token (which turns the token into a start-of-frame sequence), appends the information that it wants to transmit, and sends this information to the next station on the ring.

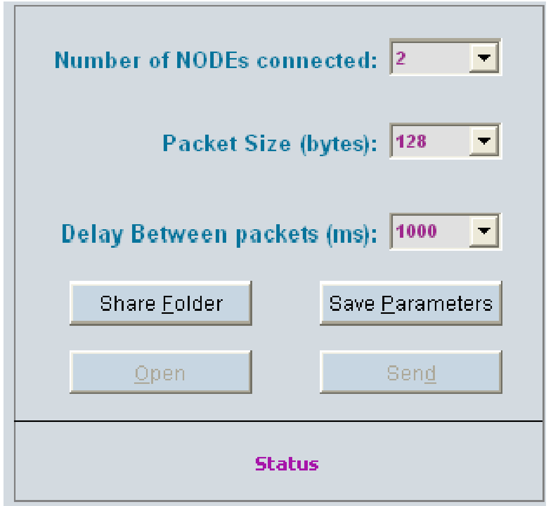
Procedure:

- Connect DB9 Cables to Scientech 5002A as well as each computer node connected to the Scientech 5002A
- Connect patch cords to ‘Ring In’ & ‘Ring Out’ terminals on the Scientech 5002A to form a RING
- Run the Scientech 5002A software on each computer connected to the Scientech 5002A & follow the steps
- Click on “Ring Topology”

1. Select a node to treat him as Node1 by checking the 'Decide this as NODE1' checkbox, automatically other nodes as decided as NODE2, 3, & 4 according to connection

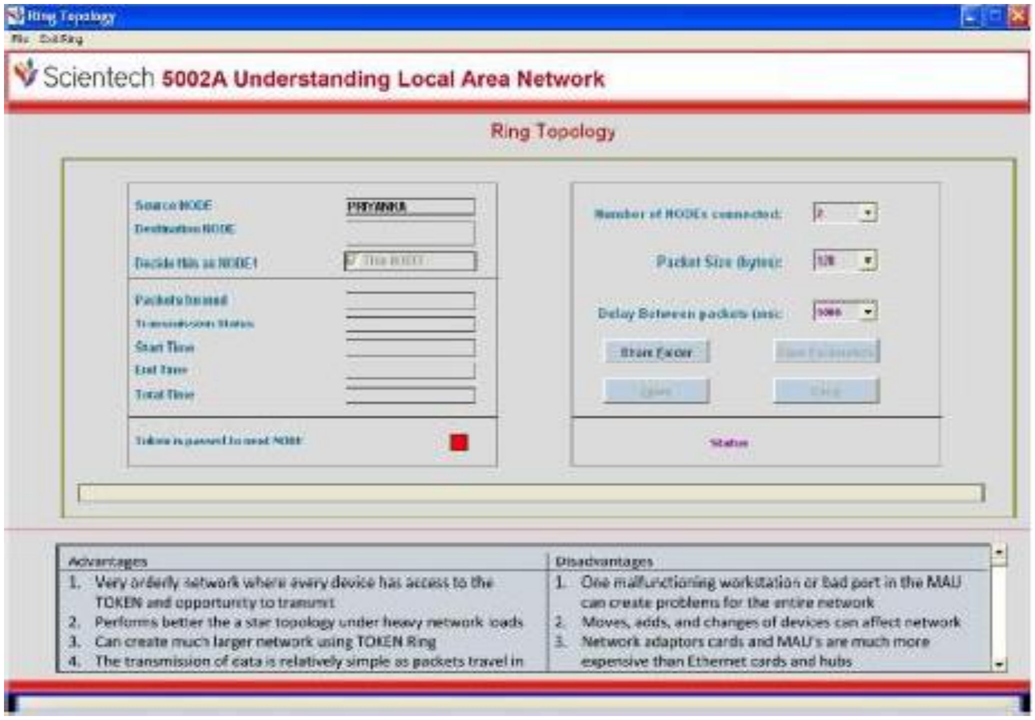


2. The Node which have Token is having authority to transmit the file
3. Before Transmission click “Sharing Folder” to share the folder for RING



topology

4. For transmission select all the parameters & then browse the file for transmission
5. Click “Send” to transmit the file
6. Frame format for Ring topology according to IEEE standards is shown at the bottom side of the window
7. Screen shot is as shown as shown below.



Ring Topology
Scientech 5002A Understanding Local Area Network
Ring Topology
PRIYANKA
Advantages
1. Very orderly network where every device has access to the TOKEN and opportunity to transmit
2. Performs better the a star topology under heavy network loads
3. Can create much larger network using TOKEN Ring
4. The transmission of data is relatively simple as packets travel in
Disadvantages
1. One malfunctioning workstation or bad port in the MAU can create problems for the entire network
2. Moves, adds, and changes of devices can affect network
3. Network adaptors cards and MAU's are much more expensive than Ethernet cards and hubs

Experiment 7.C

Objective: To study the performance of a star topology with CSMA/CD (Carrier Sense Multiple Access with Collision Detection) Protocol

Equipments Needed:

- Scientech 5002A
- Scientech 5002A Software
- 4- CAT5 cables

Theory:

A network's *access method* is its method of controlling how network nodes access the communications channel. In comparing a network to a highway, the on-ramps would be one part of the highway's access method. A busy highway might use stoplights at each on-ramp to allow only one person to merge into traffic every five seconds. After merging, cars are restricted to lanes and each lane is limited as to how many cars it can hold at one time. All of these highway controls are designed to avoid collisions and help drivers get to their destinations. On networks, similar restrictions apply to the way in which multiple computers share a finite amount of bandwidth on a network. These controls make up the network's access method.

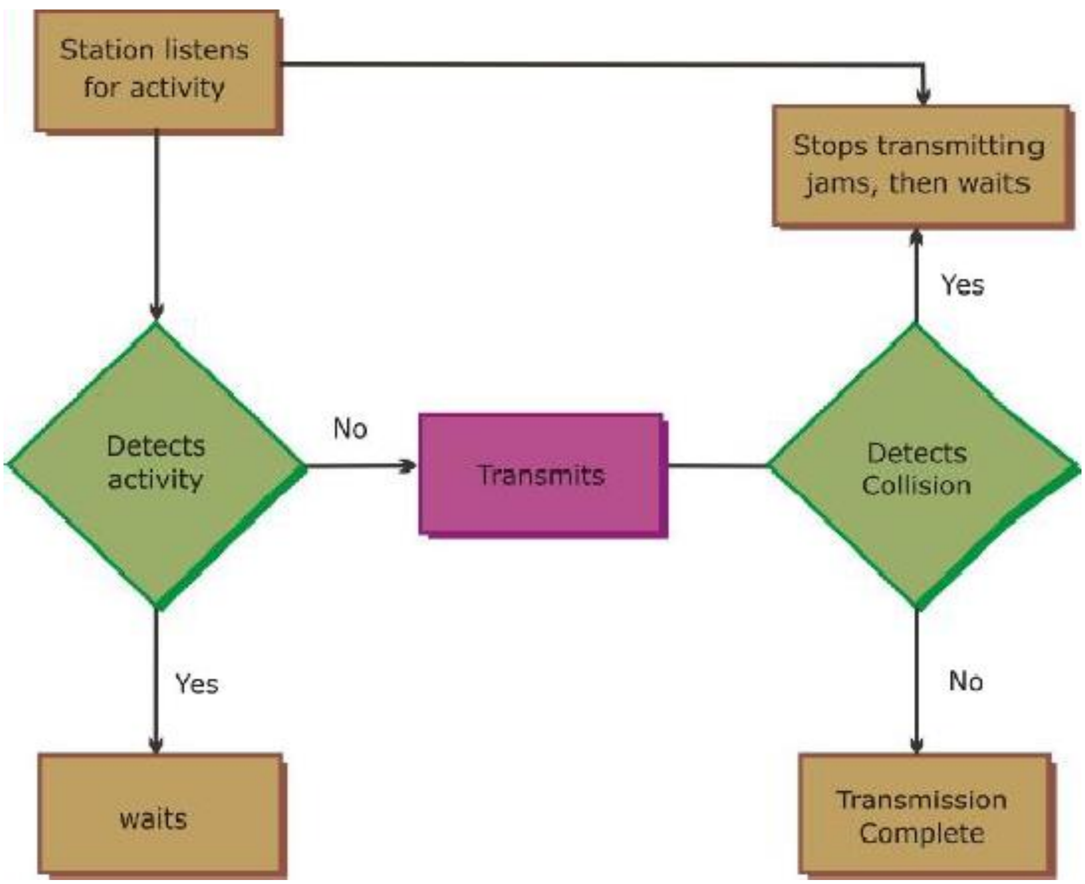
The access method used in Ethernet is called *CSMA/CD (Carrier Sense Multiple Access with Collision Detection)*. All Ethernet networks, independent of their speed or frame type, rely on CSMA/CD. To understand Ethernet, you must first understand CSMA/CD. Take a minute to think about the full name "Carrier Sense Multiple Access with Collision Detection." The term "Carrier Sense" refers to the fact that Ethernet NICs listen on the network and wait until they detect (or sense) that no other nodes are transmitting data over the signal (or carrier) on the communications channel before they begin to transmit. The term "Multiple Access" refers to the fact that several Ethernet nodes can be connected to a network and can monitor traffic, or access the media, simultaneously.

In CSMA/CD, when a node wants to transmit data it must first access the transmission media and determine whether the channel is free. If the channel is not free, it waits and checks again after a very brief amount of time. If the channel is free, the node transmits its data. Any node can transmit data after it determines that the channel is free. But what if two nodes simultaneously check the channel, determine that it's free, and begin to transmit? When this happens, their two transmissions interfere with each other; this is known as a *collision.*

The last part of the term CSMA/CD, "collision detection," refers to the way nodes respond to a collision. In the event of a collision, the network performs a series of steps known as the collision detection routine. If a node's NIC determines that its data has been involved in a collision, it immediately stops transmitting. Next, in a process called *jamming*, the NIC issues a special 32-bit sequence that indicates to the rest of the network nodes that its previous transmission was faulty and that those data frames are invalid. After waiting, the NIC determines if the line is again available; if it is available, the NIC retransmits its data.

On heavily trafficked networks, collisions are fairly common. It is not surprising that the more nodes there are transmitting data on a network, the more collisions that will take place.

(Although a collision rate greater than 5% of all traffic is unusual and may point to a problematic NIC or poor cabling on the network.) When an Ethernet network grows to include a particularly large number of nodes, you may see performance suffer as a result of collisions. This "critical mass" number depends on the type and volume of data that the network regularly transmits. Collisions can corrupt data or truncate data frames, so it is important that the



network detect and compensate for them. As shown in next *figure* depicts the way CSMA/CD regulates data flow to avoid and, if necessary, detect collisions.

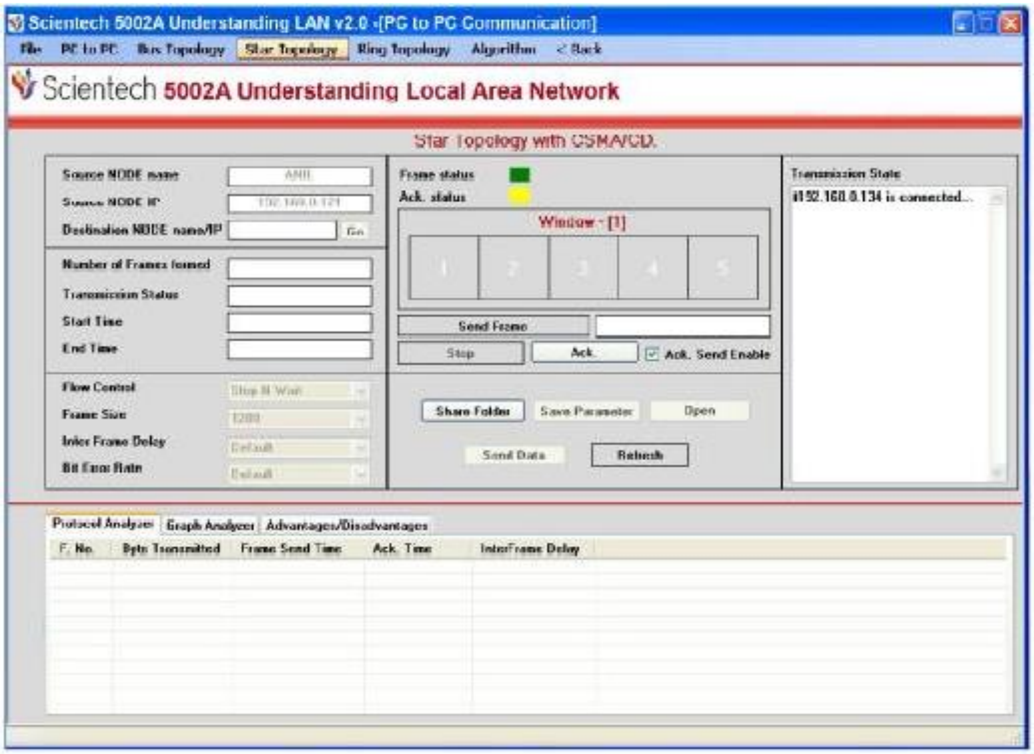
On an Ethernet network, a *collision domain* is the portion of a network in which collisions occur if two nodes transmit data at the same time. When designing an Ethernet network, it's important to note that because repeaters simply regenerate any signal they receive, they repeat collisions just as they repeat data. Thus, connecting multiple parts of a network with repeaters results in a larger collision domain. Higher-layer connectivity devices, such as switches and routers, however, can separate collision domains.

Collision domains play a role in the Ethernet cabling distance limitations. For example, if there is more than 100 meters distance between two nodes on a segment connected to the same 100BASE-TX network bus, data propagation delays will be too long for CSMA/CD to be effective. A *data propagation delay* is the length of time data takes to travel from one point on the segment to another point. When data takes a long time, CSMA/CD's collision detection routine cannot identify collisions accurately. In other words, one node on the segment might begin its CSMA/CD routine and determine that the channel is free even though a second node has begun transmitting, because the second node's data is taking so long to reach the first node.

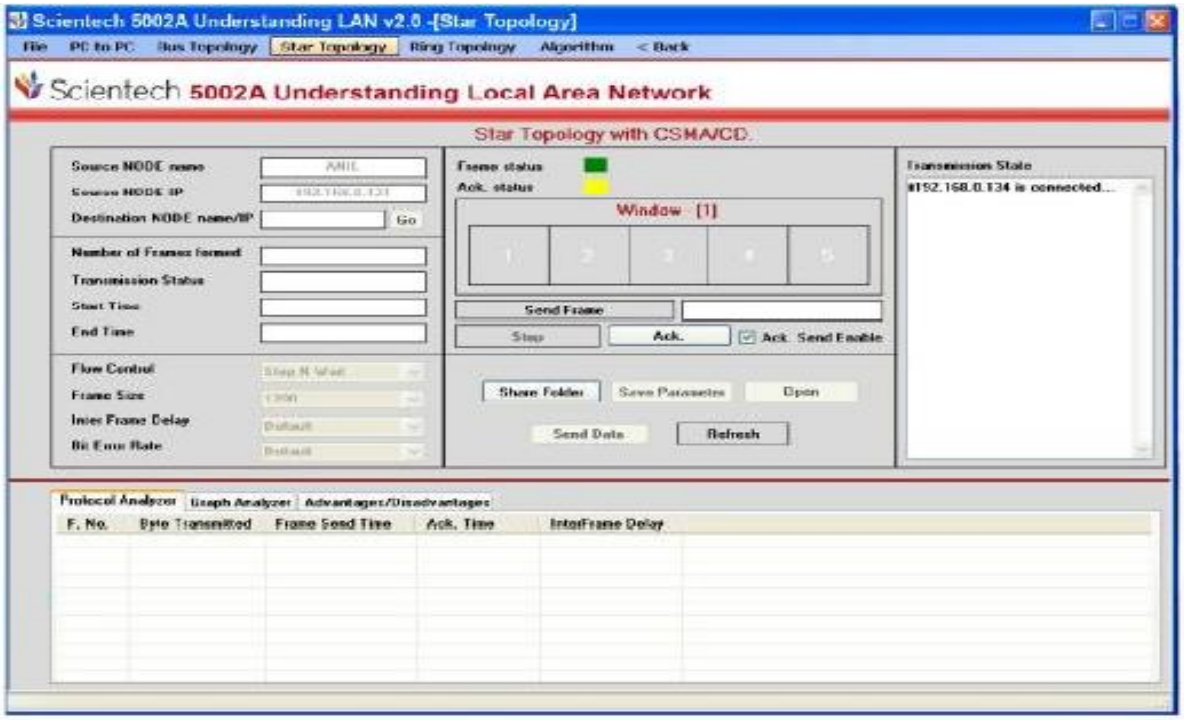
At rates of 100 or 1000 Mbps, data travels so quickly that NICs can't always keep up with the collision detection and retransmission routines. For example, because of the speed employed on a 100BASE-TX network, the window of time for the NIC to both detect and compensate for the error is much less than that of a 10BASE-T network. To minimize undetected collisions, 100BASE-TX networks can support only a maximum of three network segments connected with two hubs, whereas 10BaseT buses can support a maximum of five network segments connected with four hubs. This shorter path reduces the highest potential propagation delay between nodes.

Procedure:

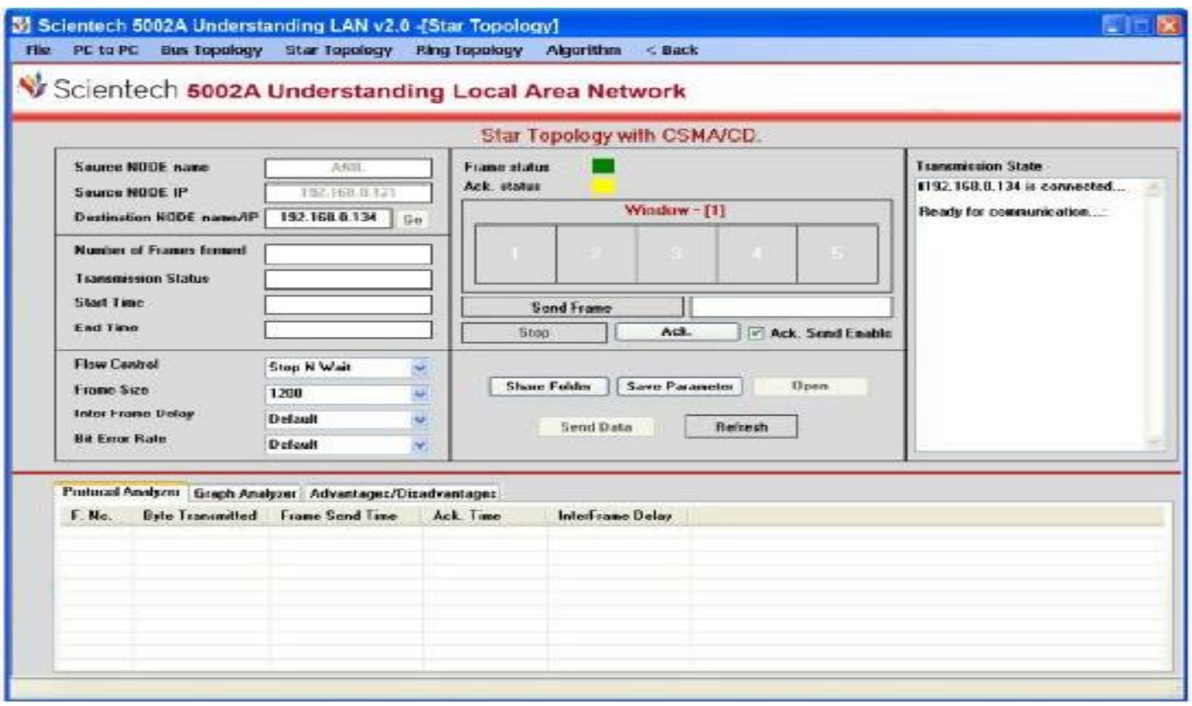
- Connect the computers to the Scientech 5002A using RJ45 cables Switch on the power supply.
- Open Scientech 5002A software Click on 'Star Topology'



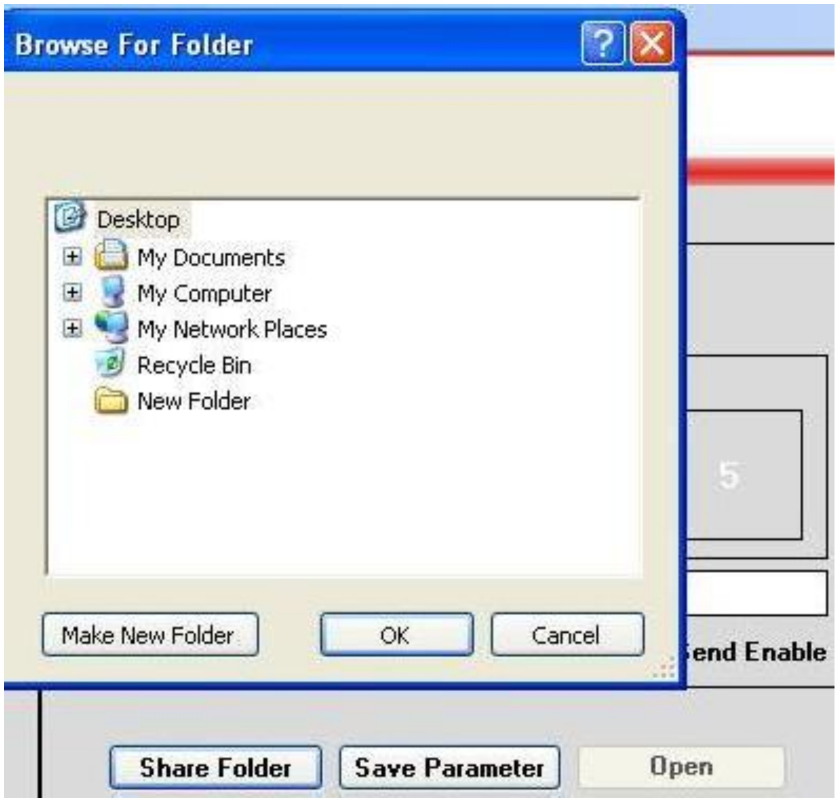
- A window will appear on screen as shown



You will find IP address and Computer name of each connected node.

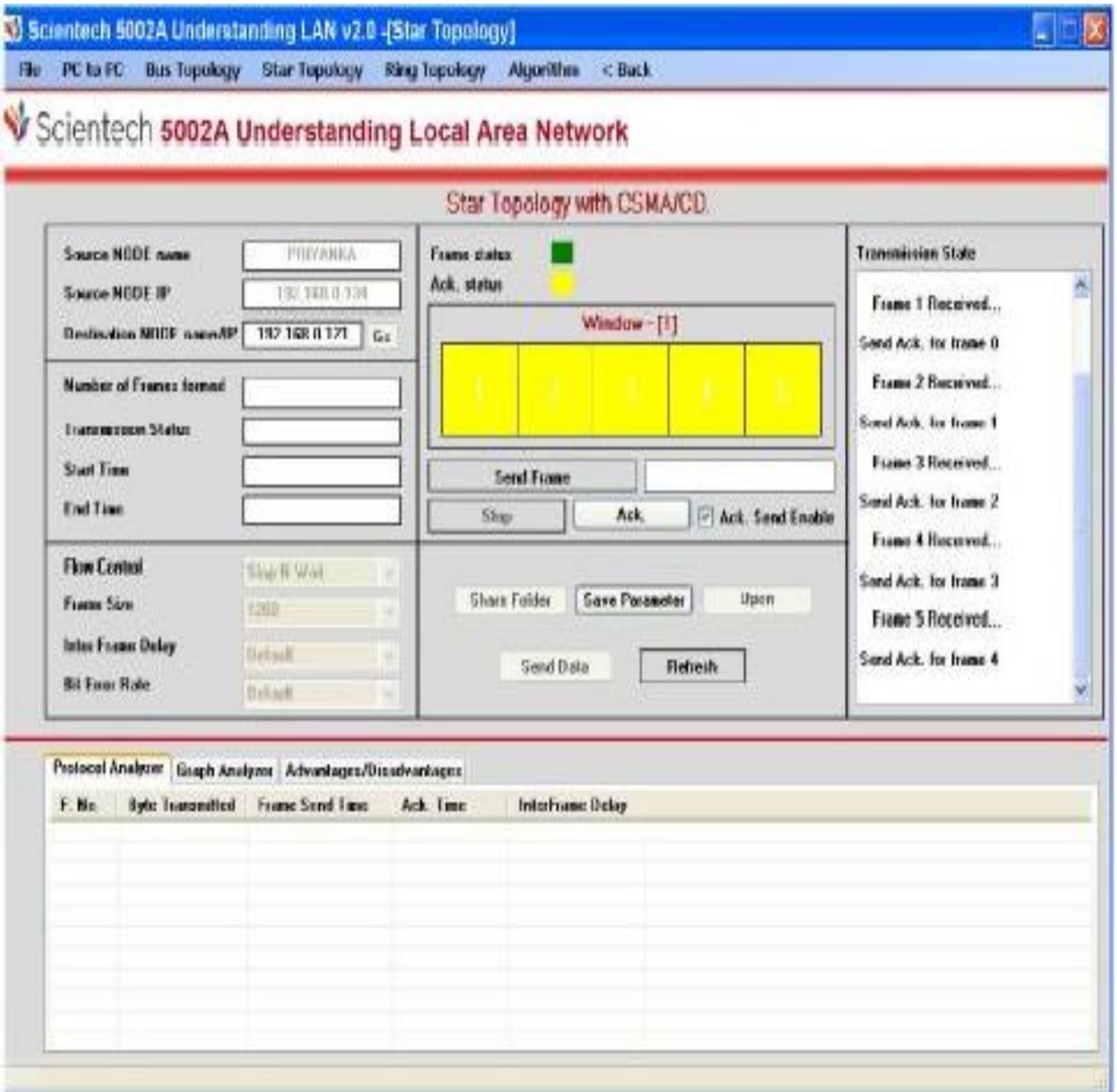


Click on ‘Sharing Folder’ and select a folder except ‘Desktop’ and Drives. You can also make a new folder.

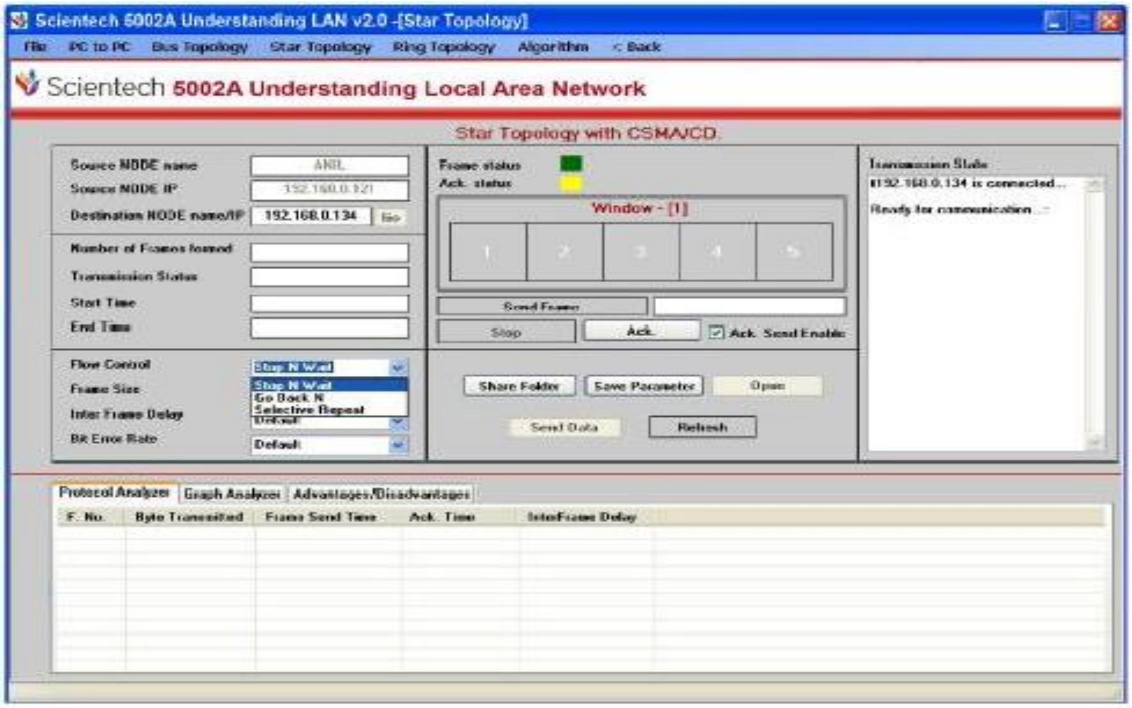


Repeat above step on each node connected to Scientech 5002A

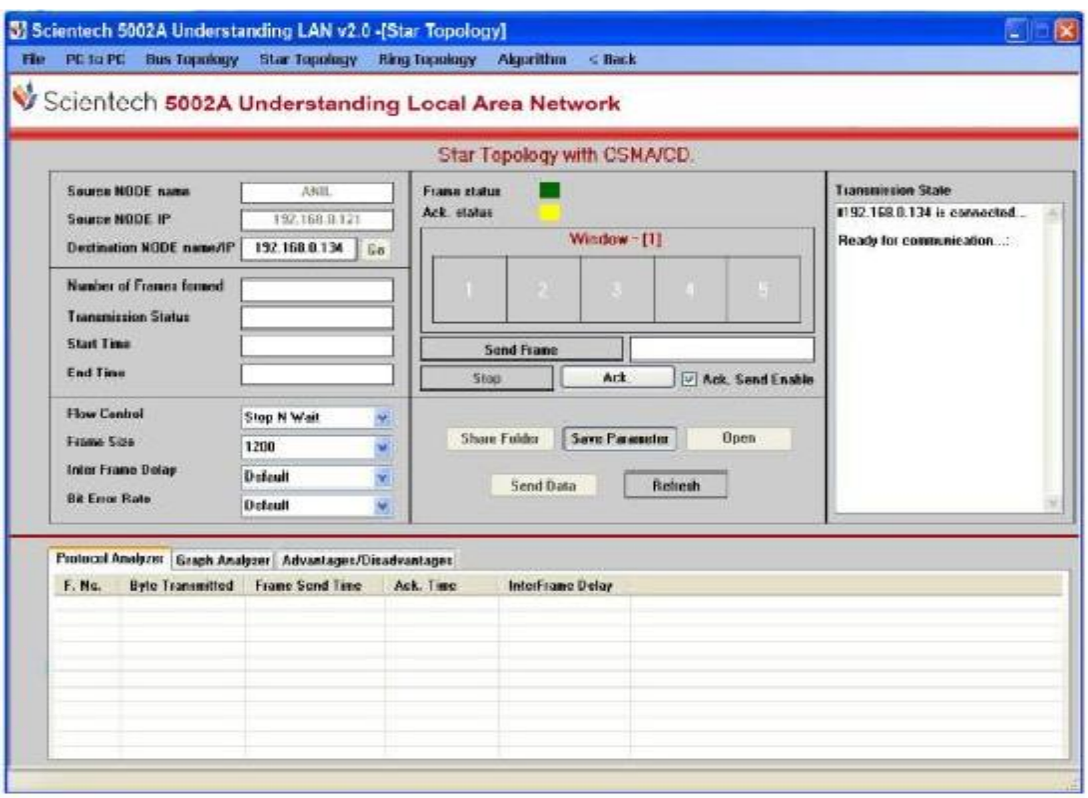
Now enter the destination node Name/IP Address.


Scientech 5002A Understanding LAN v2.0 -[Star Topology]
File
PC to PC
Bus Topology
Star Topology
Ring Topology
Algorithm
< Back
Scientech 5002A Understanding Local Area Network
Star Topology with CSMA/CD
Source NODE name
Source NODE IP
Number of Frames formed
Start Time
End Time
Flow Control
Frame Size
Inter Frame Delay
Bit Error Rate
Frame status
Ack. status
Window - [1]
Send Frame
Stop
Ack
Ack. Send Enable
Share Folder
Save Parameter
Open
Send Data
Refresh
Transmission State
Frame 1 Received...
Send Ack. for frame 0
Frame 2 Received...
Send Ack. for frame 1
Frame 3 Received...
Send Ack. for frame 2
Frame 4 Received...
Send Ack. for frame 3
Frame 5 Received...
Send Ack. for frame 4
Protocol Analyzer
Graph Analyzer
Advantages/Disadvantages
F. No.
Frame Send Time
Ack. Time
InterFrame Delay

Now choose the parameters (The parameters should be same on each node connected to the Scientech 5002A)

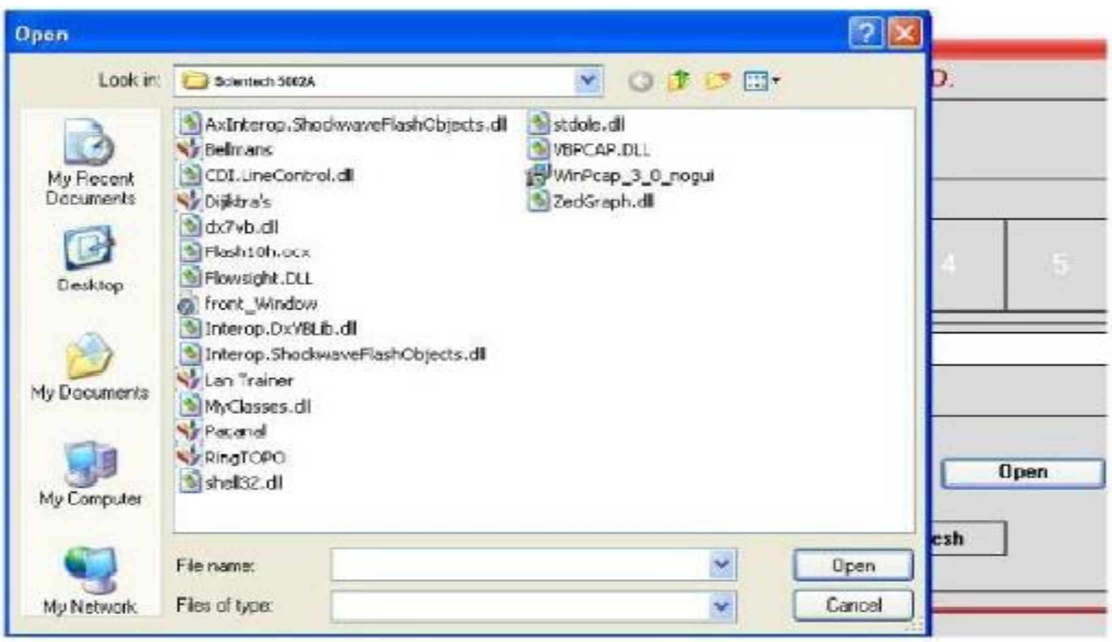


Click on 'Save Parameters'

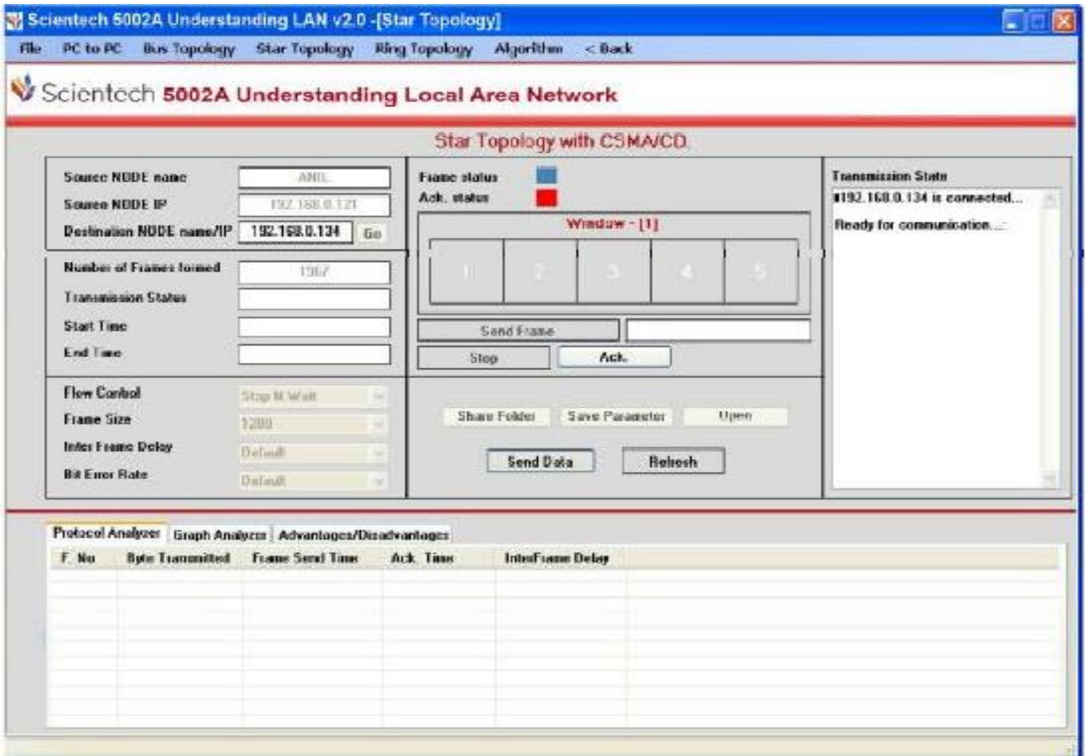


‘Ready for communication’ message will be displayed in status window of both source node & destination node

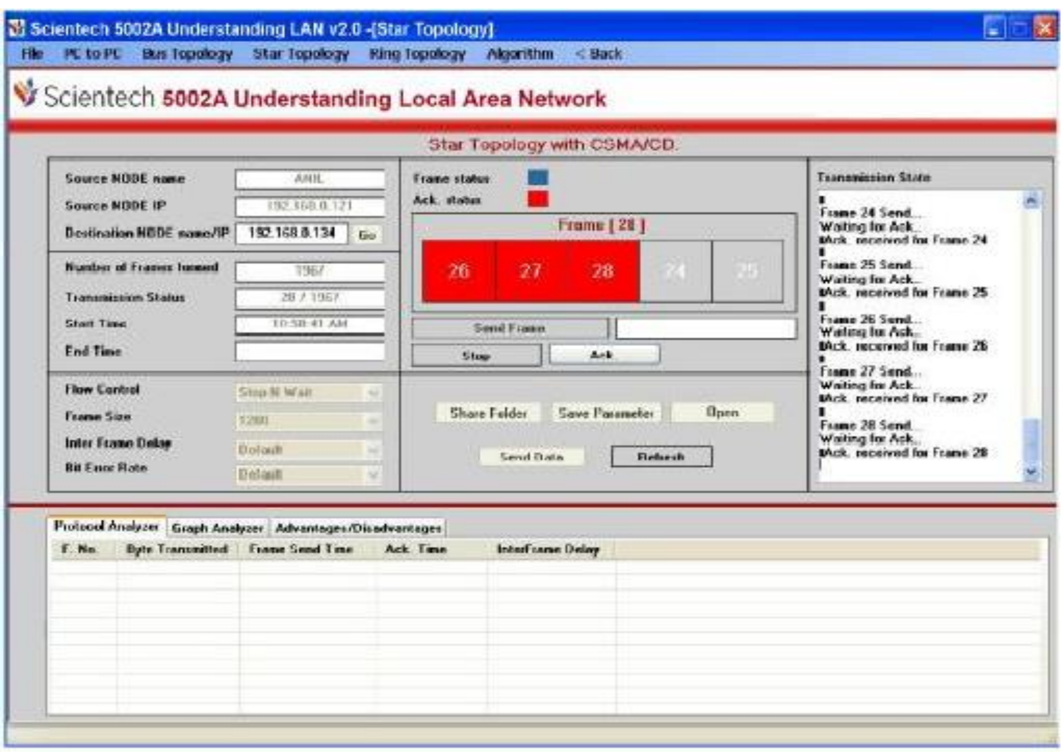
Click on ‘Open’ to open a .txt file to transmit



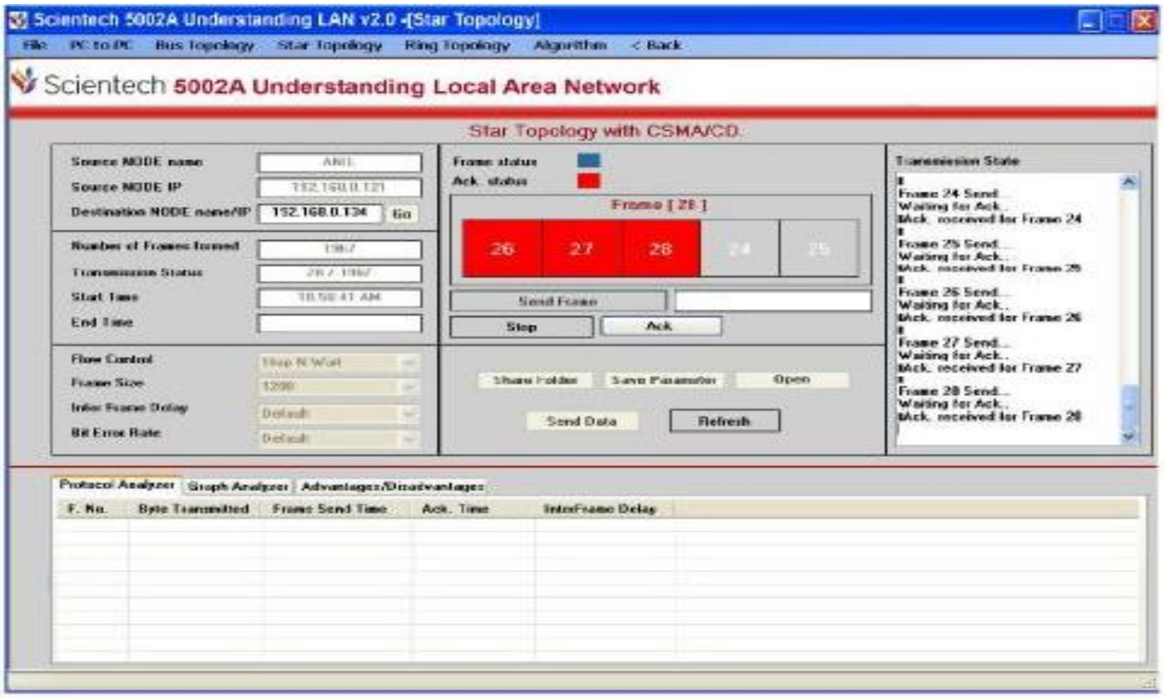
Now click on open then click on sends data



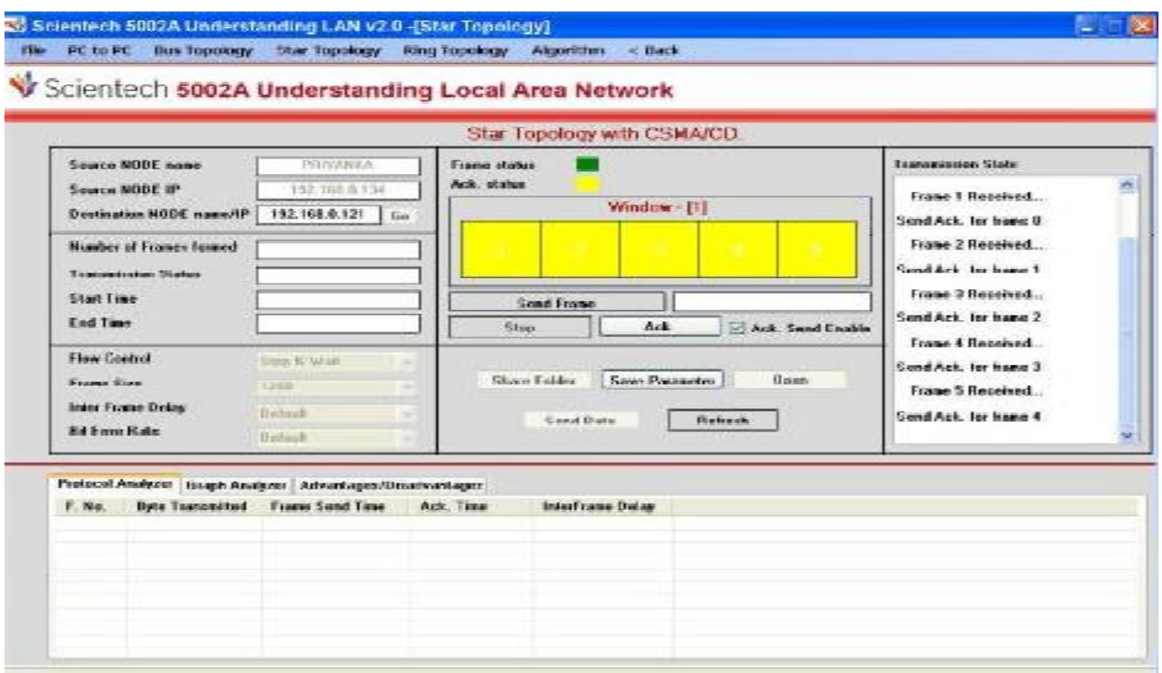
You can see a frame status with blue color and Acknowledge status with red color



You can see number of frames formed

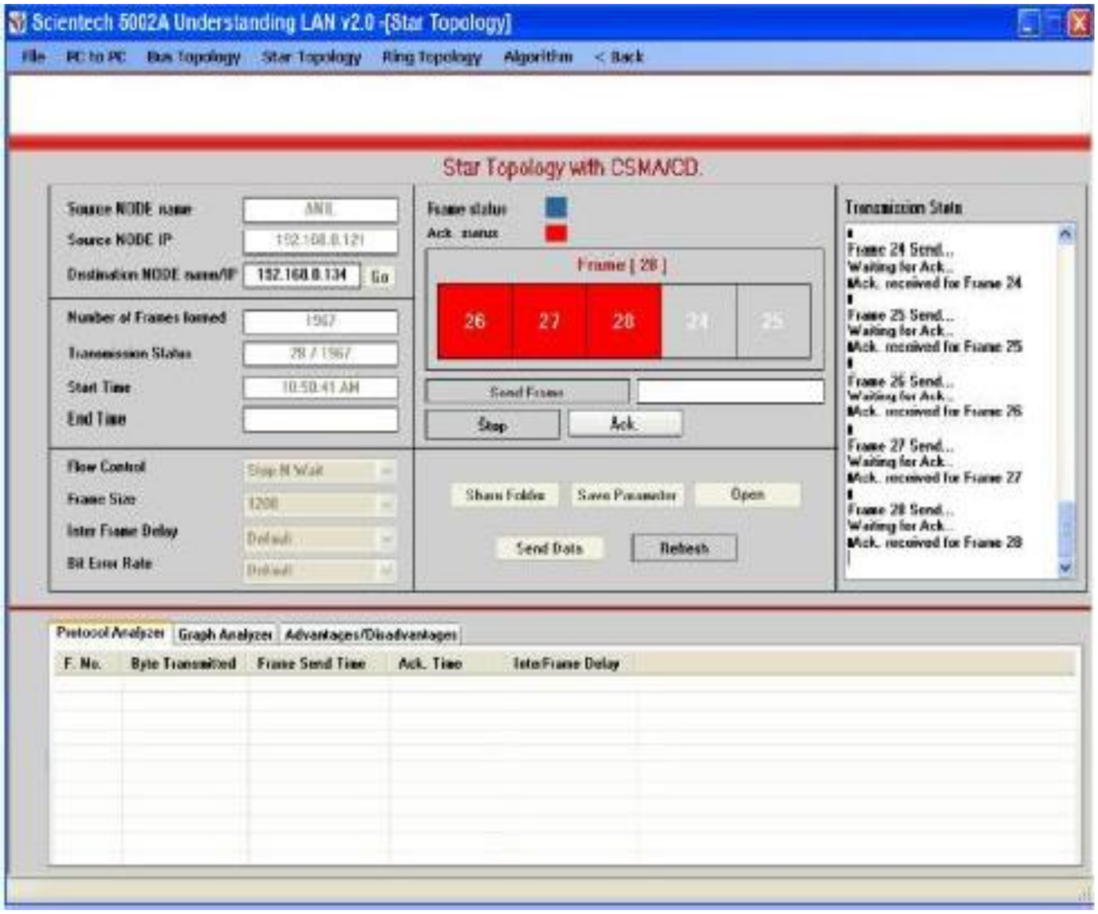


You can stop transfer of file by removing check from Ack. Send enable button on receiver side here green color represents frame received and yellow color represents



Acknowledgement sent

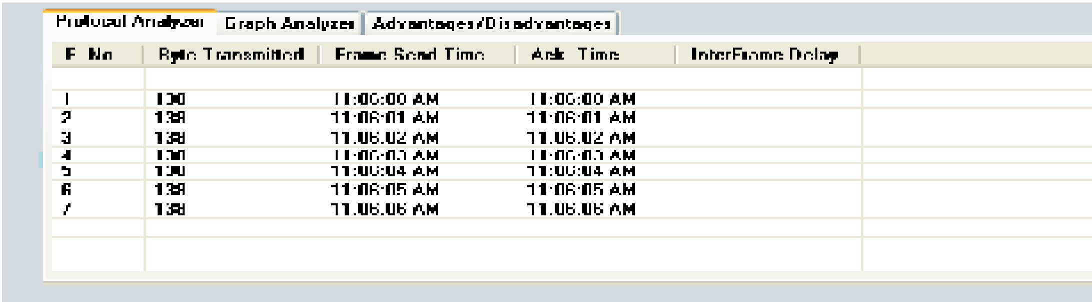
Observe on sender side same status waiting for Acknowledgement



To start transfer of file continuously check on Ack. Send enable button on receiver

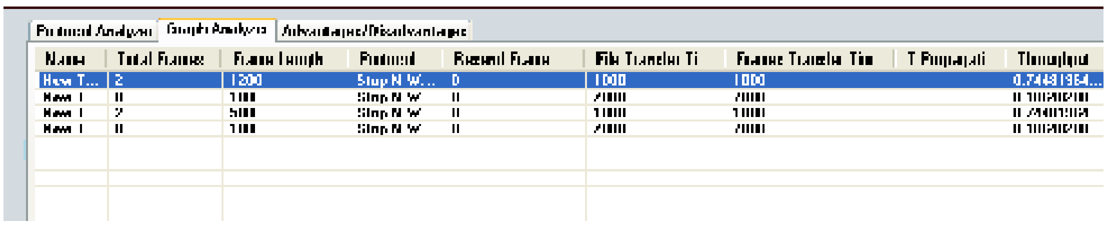
side

You can see the detail timings of each packet.

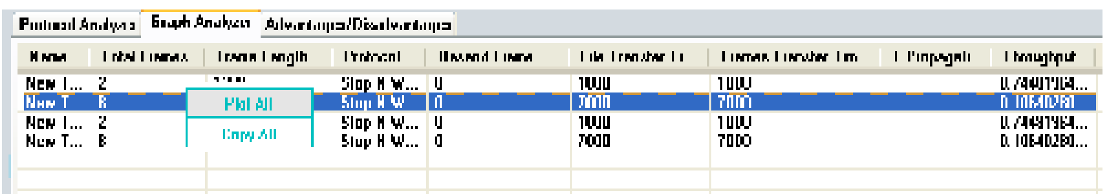


Status window will show, 'File Transmitted…..'

You can also observe throughput of same file when using different frame size



Right click with mouse and user will get listing plot all



And select plot all a graph will be displayed user can also zoom particular area by selecting that area