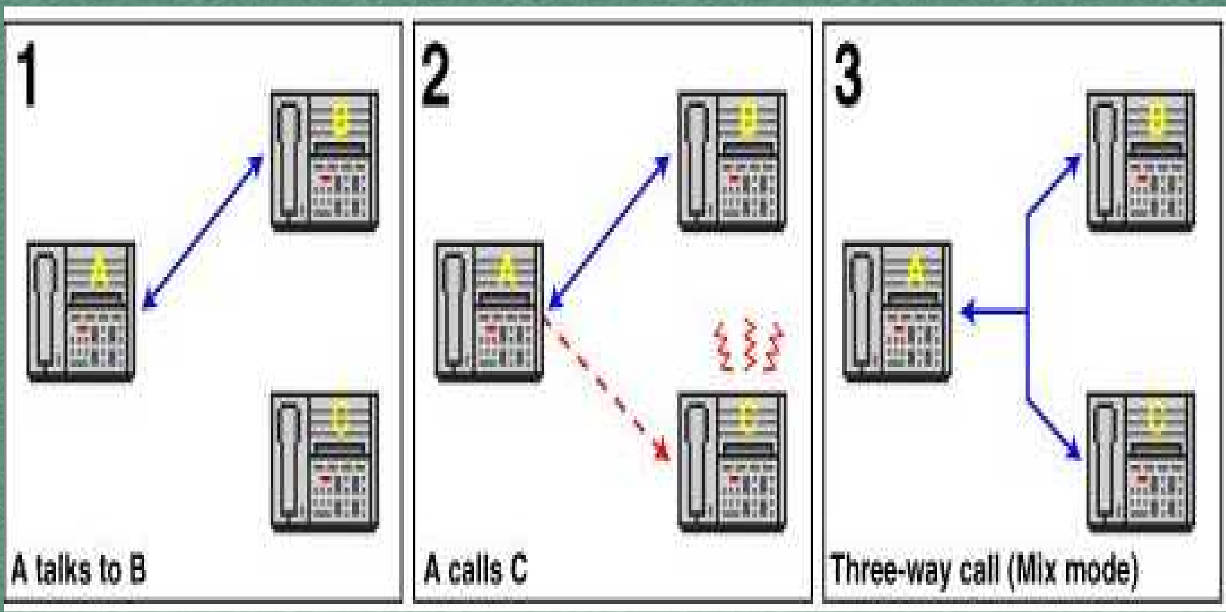
इरिसेट IRISET

TCS1

TELEPHONE INSTRUMENTS



Indian Railways Institute of
Signal Engineering and Telecommunications
SECUNDERABAD - 500 017

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INDIAN RAILWAYS INSTITUTE OF SIGNAL ENGINEERING & TELECOMMUNICATIONS, SECUNDERABAD - 500 017

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CHAPTER 1

CONSTITUENTS OF TELEPHONY

1.1 Principles of Telephony: Telephony provides a means of sending information through human speeches when required between two persons situated at a distance apart. In line telephony the information is sent through the medium of line conductors between them.

Telephones: The apparatus that are used for transmitting and receiving speech signals are called ”Telephones” and the persons who use them for sending information between them are called “Subscribers”.

The telephone transmitter and telephone receiver must be such that the conversion from speech into electrical currents and vice-versa be perfect i.e. free from frequency distortion, amplitude distortion.

Telephone exchange: It is a place where switching between two subscribers is done through either manually or electronically. In addition to switching, signalling and controlling are also done at “Exchange”.

Human speech and its transmission: Human speech consists of a large number of frequency components of different values between 0.3 to 3.4 KHz having different amplitudes and different phase relations between them.

Steps in Telephone transmission:

i) Conversion of speech into the electrical voice frequency currents at transmitting end.
ii) Transmission of speech currents through a media lines to the distant end.
iii) Conversion of voice frequency currents into speech sounds at the receiving end.

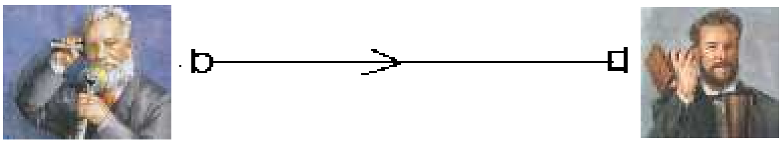


FIG.1.

As shown in the Fig.1 when any subscriber speaks into the “Transmitter” of his telephone set, his speech is converted into oscillatory electrical currents and then it is sent through lines/Transmission lines and then exchange to the other subscriber’s telephone set where it is converted into speech sounds by “Receiver”. One pair of conductors are required between two subscribers. All the line conductors from different subscriber’s premises are concentrated at a place known as the telephone exchange.

1.2 Cables used in Telephony:

In line telephony speech currents are carried by lines first into the exchange and there to the receiver of the called subscriber through separate pair of lines. The type of transmission lines used is generally insulated copper conductors, which are packed into a bunch of 10, 20, 50 or 100 pairs called as “Telephone cables”.

Copper wires are used for Telephony transmission due to, less attenuation and less distortion provided that the insulation resistance of conductor is within the given values.

1.3 Basics of Telephone Exchange:

Telephone Exchange is a place where switching between two subscribers is done through either manually or electronically. In addition to switching, signalling and controlling are also done at exchange.

It consists of the following functional blocks:

a) Main Distribution Frame with protective devices.
b) Card Frame.
c) Mother board.
d) Power supply panel with protective devices.

a) Main Distribution Frame (MDF): In a Telephone exchange different subscribers from different places are terminated on a frame called "Main Distribution Frame" (MDF) in the exchange and from there they are extended to subscriber's line cards/Trunk cards kept in the exchange rack. Protective devices are located in the MDF.

Purpose of MDF:

There are three purposes of MDF:

1) It is the place where both outdoor and indoor cables are terminated
2) The cross connection between the two cables conductors is done on the MDF and this is done by means of jumper wires (Red & White).
3) It carries all the protective devices used in the exchange. They are Fuses, Heat coils & Lightning protectors.
4) The MDF is the most suitable place for testing purposes.

b) Card Frame: It contains different slots in which the nominated cards are to be inserted. It is different in different types of exchanges.

c) Mother board: It provides connectivity between different cards. It is a multilayer PCB.

d) Power supply panel:
It provides power supply to different cards in the exchange at different low D.C. voltages. It also includes protective devices like fuses etc.

1.4 Main functional areas in Telephone Exchange:

a) Switching Function: The switching functions are carried out through the switching network, which provides a temporary path for simultaneous, bi-directional speech between the following:-

- Two subscribers connected to the same exchange. This is called as "Local switching"
- Two subscribers connected to different exchanges. This is known as "Trunk switching".
- Pairs of trunks towards different exchanges. This is known as "Transit switching"

b) Signalling function: The signaling function enables various equipments in a network to communicate with each other in order to establish and supervise the call. It is of two types,

i) Subscriber line signalling: It enables the exchange to identify calling subscriber's line, extend dial tone, receive the dialed digits, extend the ringing voltage to the called subscriber, extend the ring back tone to the calling subscriber to indicate the called subscriber is being connected. In the event the called subscriber is busy, engage tone is sent to the calling subscriber.

ii) Inter exchange signalling: It enables a call to be set up, supervised and cleared between exchanges.

c) Controlling function: The controlling function performs the task of processing the signalling information and controlling the operation of the switching network.

The control functions may be,

i) Wired logic control: In this pre wiring is done between different speech path devices and common control. If any changes are required in facilities of subscribers or introduction of new services require wiring changes.

ii) Stored Program Control (SPC): After introduction of microprocessor, stored program control system came into use. In this system the establishment and supervision of the connections in the exchange is under the control of "Microprocessor", which is suitably programmed.

Exercise Objectives:

1) Switching, Signaling and Controlling are the main functions of Exchange. (T/F)
2) Two Subscribers of two different Exchanges are connected through Trunk Switching. (T/F)
3) Loop Signaling is extended from Subscriber to Exchange. (T/F)
4) Stored Program Control has over all control on the Exchange. (T/F)
5) MDF is meant for connecting indoor and outdoor cable pairs. (T/F)
6) Protective devices are used on MDF for protecting the Exchange SLC/TRK cards from damage. (T/F)
7) Copper wires are used for transmission due to less attenuation & less distortion (T/F)
8) A Trunk line uses 3 pairs of copper wire for its termination. (T/F)
9) Mother Board is a multilayered PCB for placing cards. (T/F)
10) Card Frame is used to suitably connect the card in slots. (T/F)
11) The main function of SPC exchange is signalling, switching and controlling. (T/F)
12) Two subscribers of two different exchanges are connected through local switching.(T/F)
13) Loop signal is extended from subscriber to Exchange. (T/F)
14) Over all control of the exchange is done by Central Control. (T/F)
15) Indoor and outdoor cables are terminated on Main Distribution Frame. (T/F)
16) Protective devices are used on IDF to protect surge voltages entering in exchange.(T/F)

Answer the following in short:

1. What is telephony?
2. What are the functional areas of telephone exchange?

CHAPTER 2

MDF AND PROTECTIVE DEVICES

2.1 Main Distribution Frame (MDF)

Introduction:

MDF stands for **“Main Distribution Frame”** which is the first distribution point (DP) from the Telephone Exchange towards subscriber. All the incoming cables from subscriber telephones and trunk lines from the other Telephone exchanges are terminated on one side of MDF known as **“Cable side”**. The cables coming from the exchange equipment room are terminated on the other side known as **“System side”**. These are terminated on **KRONE TERMINAL BLOCKS**.

MDF use IDC (Insulation Displacement Connector) terminal strips on which pairs of wires can be terminated. IDC terminal strips contain two parallel rows of terminals in the same molded plastic block. The terminals in one top row are internally connected by metallic strips to the corresponding terminals in the second bottom row. However the connection is made through contacts by the metallic strips that may be broken by the insertion of a monitoring plug or a special disconnection plug known as” Wedge Connector”.

Fig.2.1 Professional IDC insertion tool

IDC insertion tool: It is used for making connections to the IDC terminals by punching the Cable pair. The two features of the tool are,

a) It includes a scissor-action cutter that is designed to cut off surplus wire after the termination has been made. This occurs automatically as part of the punch-down action.
b) It is provided with a folded-out metal hook for removing the wires from the terminals. It is shown in the figure.2.1

2.2 Purpose of MDF:

a) All incoming lines from subscribers and outgoing lines towards exchange are terminated at MDF.
b) It is a place suitable for testing the line side and equipment side.
c) It carries protective devices like fuses & Integrated Protection Module (IPM).

2.3 Terminations:

The incoming and outgoing lines are terminated at MDF on Krone Tag Blocks. These are available in capacities of 50 pair &100 pair. Krone block picture is shown below.

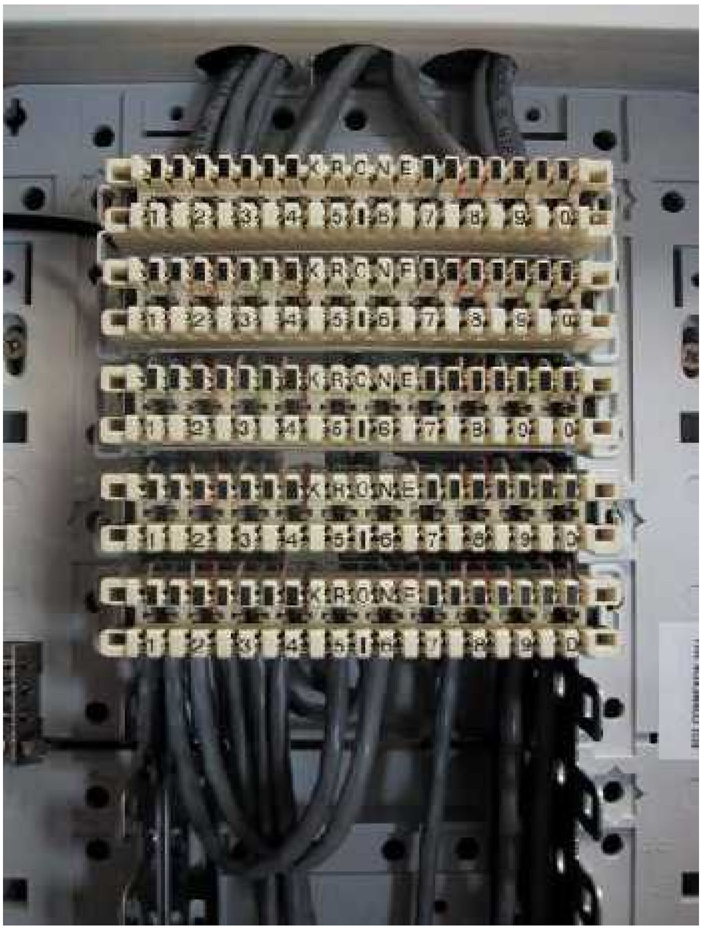


Fig.2.2 Krone Tag Blocks

2.4 Protective devices used in MDF:

The Exchange equipment to be protected against A.C. power across, power induction, and lightning faults on Telecommunication lines. Otherwise such hazards can potentially travel into the exchange modules and severely damage sensitive switching and transmission equipment. To minimize the effects of such occurrences, poly switch resettable devices can be used as over current protection as “Primary protection” at MDF in modular form.

In MDF primary protection is provided. Primary protectors are Gas Discharge Tube (GDT) OR Carbon blocks or Integrated Protection Modules (IPM). The primary protectors limit the very high energy transients.

The secondary protectors are incorporated on Telecom line card itself, which limit the voltage and current to acceptable levels.

Here are other types of over voltage protectors such as “Metal Oxide Veristor (MOV) AND Thyristors Surge protection devices (TSPD) preferred choice for Telecom protection applications.

a) Gas Discharge Tube

Gas Discharge Tube (GDT), operates on the principle of the arc discharge phenomenon. Electrically, GDT act as voltage-dependent switches. As soon as the voltage applied to the GDT exceeds the spark over voltage (70 volts to a few KV, depending on the type), an arc is formed in the hermetically sealed discharge region within nanoseconds. When the discharge has died down, the GDT extinguishes and the internal resistance immediately returns to values of several hundred mega ohms.

Under normal operating conditions the high insulation resistance and the low self-capacitance contribute to the fact that a gas-filled surge arrestor has virtually no effect on the system to be protected.

- **Applications:**

Typical applications are,

✓ Telephone exchange substations
✓ Cable distribution system
✓ Telephone MDF
✓ Subscriber's terminals
✓ Satellite reception system

b) Metal Oxide Veristor (MOV):

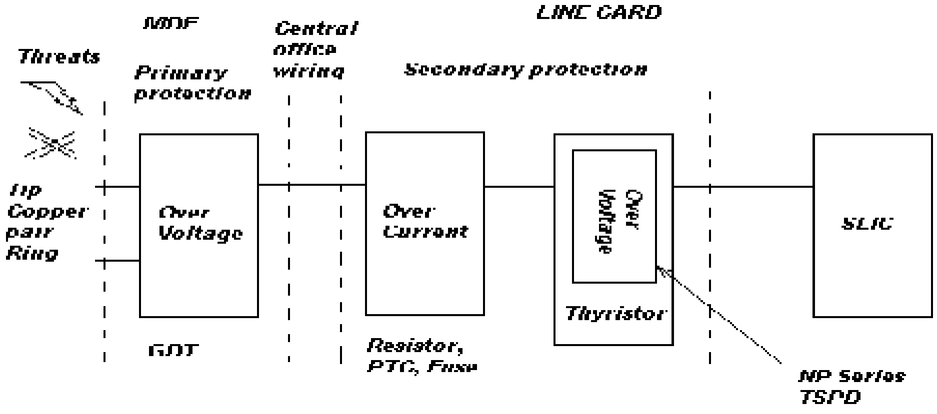
Metal oxide Veristor are voltage dependent, non- linear devices which are manufactures using a semiconducting metal-oxide.

- **Applications:**

✓ Consumer electronics (video recorders, color TV, amplifiers, and car radios)
✓ Domestic applications (heating controls, washing machines, and microwave ovens).
✓ Telecommunications (telephone handsets, telephone exchanges, fax, telex, and modems).
✓ General industrial applications (machine control, air conditioning, medical analysis, transformers etc).
✓ Lighting (electric ballasts)
✓ Power supplies (washing machines, power switches etc).
✓ Automotive electronics (protection of an ignition circuit output transistor, protection of the solenoid driver circuit).

- **Typical Protection design:**

Typical Protection design is shown in the figure shown below. It has 'Primary protection' in MDF by GD tubes and 'Secondary protection in the Line Card by providing over current and over voltage devices like Resistor, PTC, and Fuse & Thyristors.



Typical Protection Design

Answer the following:

1. What is MDF (Main Distribution Frame)? What are the requirements of MDF?
2. What are the different types of protections used on MDF?
3. What care should be taken while installing the IDF and MDF?

Objective:

1. Normally an IDC can accommodate 10 pairs of copper cable on it. (T/F)
2. IDF are to be provided with protective devices on it. (T/F)
3. Gas discharge tubes works on the principle of Arc Discharge phenomenon (T/F)
4. MOV are Non Linear devices. (T/F)
5. IDC are more reliable terminations than other termination. (T/F)

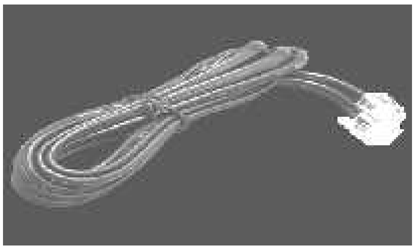
CHAPTER 3

TELEPHONE INSTRUMENTS

3.0 To fulfill the need of communication, a device which is used, is a telephone instrument. Various types of instruments are available according to their type, function they perform, features available with them and much more.

Following are the types of telephone instruments used in Railways.

Push button Telephones: push button telephones uses a dial pad containing button to press digits from 0 to 9 and special characters * and #.they comes in the variant of single line and double line. A picture of line cord is shown below. it is a 2 wire line cord with RJ11 connector at both end.



Main & Extension (1+1) type Telephones: these telephones are used to connect between boss and secretary or may be between operator as a main phone and other users as extension.

Hand free Telephone with caller ID (CLIP phone): these telephones come with inbuilt speaker to enable the user to talk without lifting the handset and to display the identity of the caller, a display unit is also provided.
Two lines with speaker phone facility: these telephones come with 2 lines selectable. 2 different dial tones can be connected on it.

Two lines CLIP with speaker phone facility: as the name indicates these telephone instruments are available with 2 lines terminated on it with hands free and caller identity display unit.

Cord less phone: these telephones come with a handset connected to base telephone by a radio transmission. Uses trans-receiver frequencies for Trans and receive.

Control telephones: this phone is used for train traffic control at various stations which are suppose to intimate the information of all the passing trains through that station to the central control at control office. These phones are connected in Omni bus fashion to achieve all time communication between stations with central control and selective calling facility.

Magneto Telephones: magneto Telephones are used in Indian Railways for connecting point to point communication without talking the help of any exchange switch. Generally used for Level crossing gate communication with nearest cabin. Now a day's other communication ways are available so this type of magneto Telephones are not in fashion.

3.1 PUSH BUTTON TELEPHONE:

A simple push button telephone circuit consists of the following stages:

a) Ringer stage – produces audible ringing sound.
b) Key board matrix stage – to dial the required number or to perform the function of dialing the digits.
c) Dialer stage – to generate the tone or pulse corresponding to the key dialed.
d) Sound amplifier stage – to produce the speech sounds during transmitting and receiving.
e) Voltage dropper stage – to drop the from -48V DC to 5V to 12V when the hand set is lifted.
f) Rectifier and protection stage – to give voltage of proper polarity, to limit any high voltage peaks due to induction on lines.

All the above stages are inter connected as shown in the given block diagram (Fig.3.1)

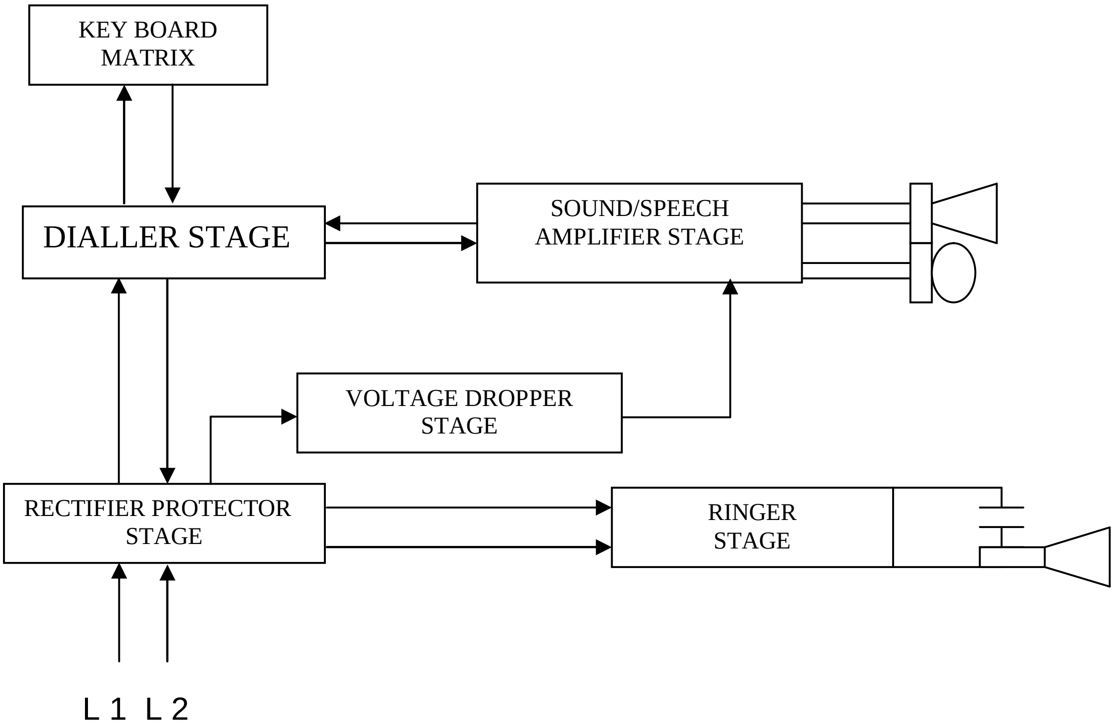


Fig.3.1 BLOCK DIAGRAM OF PUSH BUTTON TELEPHONE

a) Ringer Stage:

It consists of a Ringer IC, which operates an “Piezo electric buzzer’ or ‘Speaker’. The IC operates on AC ringing signal of 20Hz frequency with its associated circuit.

Ringer IC: It consists of 8 terminals IC. It includes bridge rectifier circuit, power supply control circuit, output amplifier, low and high frequency oscillator circuit.

Piezo electric buzzer – It gives audible sound. It is operated by the tone produced by the output of the oscillator. It is connected between the output terminal of output stage of IC, and ground (+ve) through a volume control which regulates the sound volume.

Cradle hook switch – When the hand set is on the cradle the entire telephone circuit is disconnected from the lines and only ringer section is kept connected to the lines L1 and L2.

When the hand set is OFF the cradle hooks, the ringer section is disconnected by the cradle switch contacts and the bridge rectifier is connected to the voltage dropper stage, and further to dialer, sound amplifier stage.

Other components: The other components of ringer section are,

- Capacitor (C1) of value 0.1uF. It is to change the frequency of oscillator.
- Resistance (R5) of value about 212Ω. It decides the frequency of oscillations. To change the frequency of oscillator, the resistance value to be changed.
- Capacitor (C4) of value 10Uf 63V, is used to filter the rectifier output.

Description of ringer stage circuit used in GCEL 501 telephone is given below. The ringer section consists of LS 1240 (8 pin) IC, which consumes less current. This IC has an internal oscillator to generate two tone frequencies and connects them across the output amplifier.

The tone and oscillator frequencies can be adjusted externally with the help of resistor R3 and capacitor C2. This stage consists of a ringer switch having S1, S2, and S3 switches.

Working: When the hand set is on the cradle hooks, switch S1 and S2 are OFF and S3 contact on ‘B’ side. So the ringing current of 20 Hz is applied to the Bridge rectifier inside the IC as input. This AC ringing current is rectified by the bridge rectifier and the DC output is used for the working of oscillator IC. The oscillations produced by the oscillator are connected between pin.No.5 and 2 via output stage in the IC. These are connected to the piezo electric buzzer via volume control. The volume of the buzzer can be adjusted by the volume control. IC pin.No.3 and 4 are connected with a capacitor of 0.1uF and a resistance of 212Ω which decides the frequency of oscillations (tone). The 10 uF, 63V capacitor is a filter capacitor which filters the output of the bridge rectifier. As soon as the hand set is lifted, the ringer switch operates and S3 changes over its position to ‘A’ side and the input to IC pin.No1 is disconnected and ring stops.

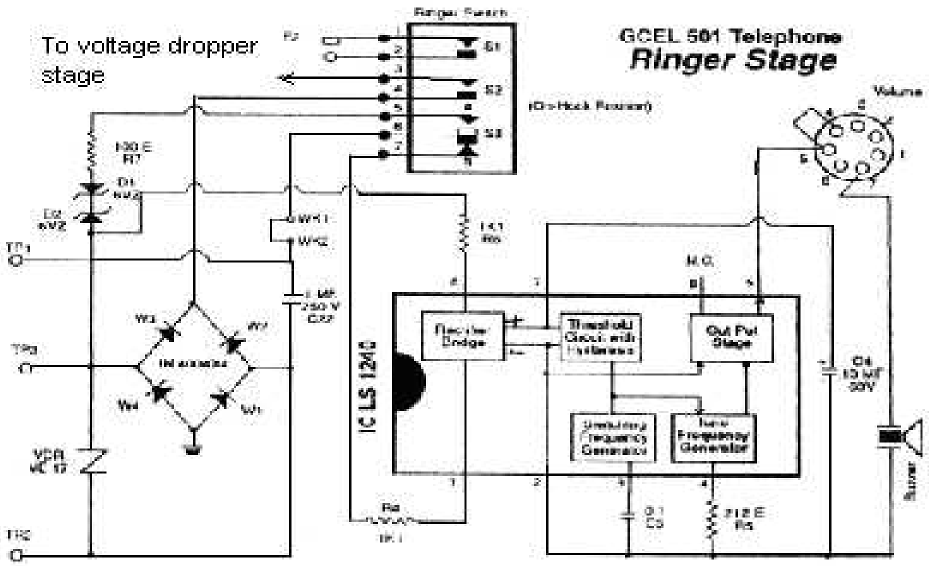


Fig.3.2 Ringer circuit

b) Key board matrix stage:

It is provided with a “Key pad” to enter the subscriber’s telephone numbers.
It has 4 Rows R1, R2, R3 & R4 and 3C columns C1, C2 & C3. The key pad is connected to dialer IC. Minimum 12 numbers of keys are provided on key pad.

Number buttons -10 No’s (0-9 numbers)
Special keys – 2 to 6 No’s

Function of some special buttons is:

- Redialing ® - To repeat the last dialed number.
- Tone-Pulse dialing
- Star key(*) – or Ashtreik key (#) – To get the Tone mode dialing.
- Mute (M) – To prevent outgoing speech.
- Pause (P) to get multiples of 2.2 seconds delay between two digits of a number.

A row and the column are shorted when a number on the key pad is pressed. This generates equal number of pulses or DTMF tone pair which will be available at dialer IC.

Refer Fig.3.3 for key board matrix connection diagram**.**

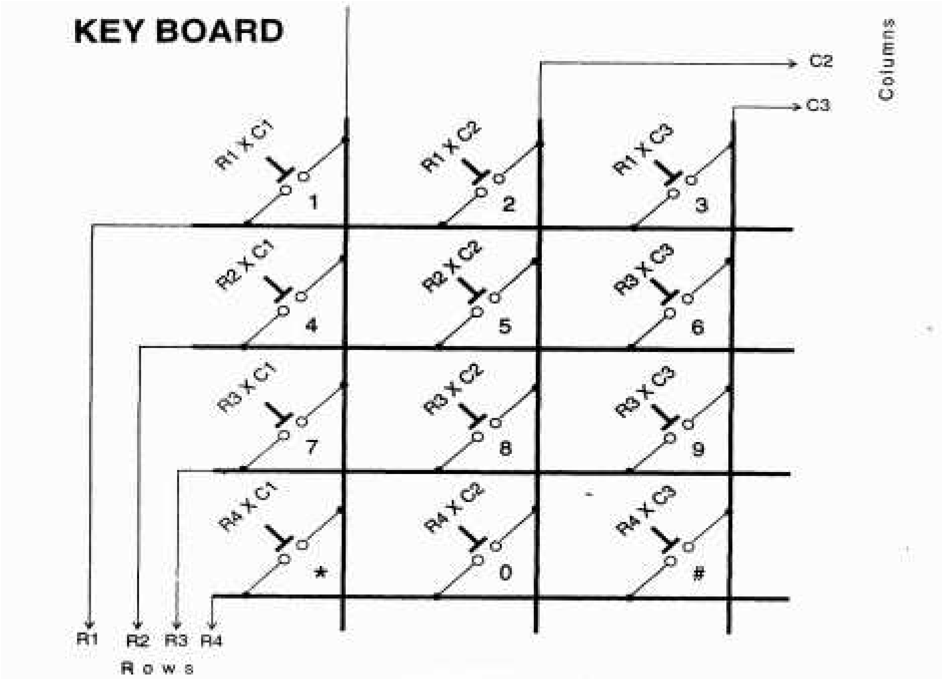


Fig.3.3 Key-board matrix connection diagram

c) Dialer Stage:

This section is connected to the key board matrix stage on one side, and other side it is connected to rectifier and protection stage, and sound amplifier.

When any key on the key board matrix is pressed the corresponding pins of dialer IC are stored and pulses are produced. These are connected with the lines via rectifier protection stage.

It sends the dial pulses at speed of 10 IPS. It can store the dial pulses, when the keys are operated fast to an extent of 1720 digits.

The dialer section has the following parts:

(i) Dialer IC: - It is 16 pin IC. It consists of, Key board interface circuit, Oscillatory circuit, Transmit Amplifier circuit, Mute circuit, Memory circuit etc.

Working of stage: When ay key of the keyboard is pressed corresponding pins are shorted. Thus the corresponding pulses are generated by UIC via keyboard interface, control logic these are conned to pin.5 of IC. Then these are connected to telephone lines, through Q10 Transistor and ringer switch contacts. These generated pulses are stored in 20 bit memory if the dialing is fast. If any key is pressed in ON-HOOK position, it is not considered and the oscillator will not function.

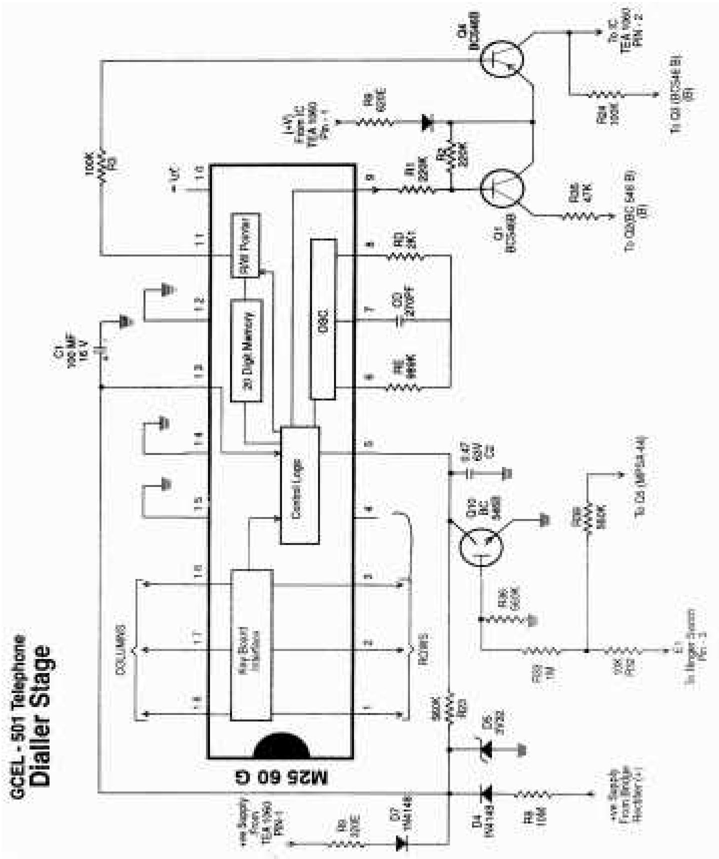


FIG.3.4 Dialer stage

d) Sound Amplifier Stage:

This is also called as “Speech stage”. It consists of a speech IC having 18 or 20 pins in special cases. This IC performs al interfacing functions between the Microphone, the ear phone, the dialer IC and the Telephone lines.

This stage performs 3 functions:

- It receives the incoming speech signals, amplifies and are connected to the loud speaker, which reconverts them to sound signals by sound output circuit in the IC.
- The outgoing speech sounds are converted to corresponding speech currents by microphone, amplified by microphone amplifier and connected to telephone lines.
- Side tone suppression is done by “Side Tone’ circuit in the IC.

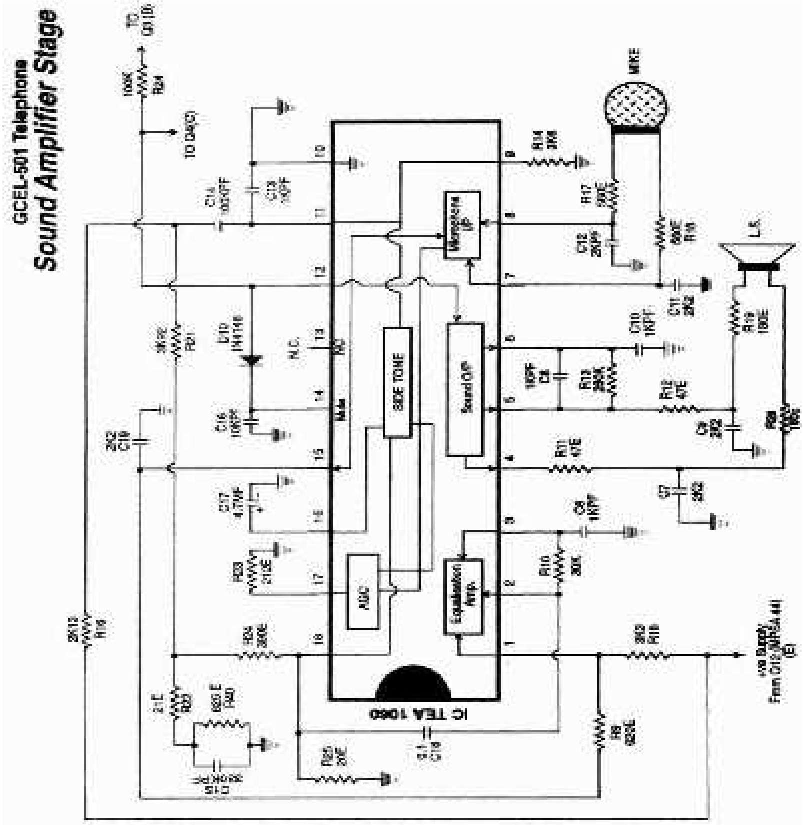


Fig. 3.5 Sound Amplifier Stage

- ➢ The gain of the sound amplifier is controlled by "Automatic gain control (AGC) circuit" provided in the IC.
- ➢ Impedance matching between the Telephone lines, IC circuit, Mike, and Loud speaker is done by "Equalization Amplifier" provided in the IC. These IC's are bipolar integrated circuits performing all speech and line interface functions required in fully electronic circuits.
- ➢ The circuits internally perform electronic switching between dialing and speech.

Working of Sound Amplifier:

Speech Transmission: The microphone output is connected to input of microphone amplifier. The output of the amplifier is connected to telephone lines.

Speech Reception: The incoming signal received on Pin.No1 is internally applied to pin11 and pin18. The output of the speech output amplifier is connected to loudspeaker.

e) Voltage Dropper Section:

Telephone instrument gets 40 to 50V DC supply from the telephone exchange through *telephone lines. Till the ring comes, this high voltage is maintained. But as soon as the* receiver is lifted, with two way switch, this high voltage goes to voltage dropper section, where it is converted into +9V or +12V as per the telephone needs. Then dial tone signals are given and are heard after amplification.

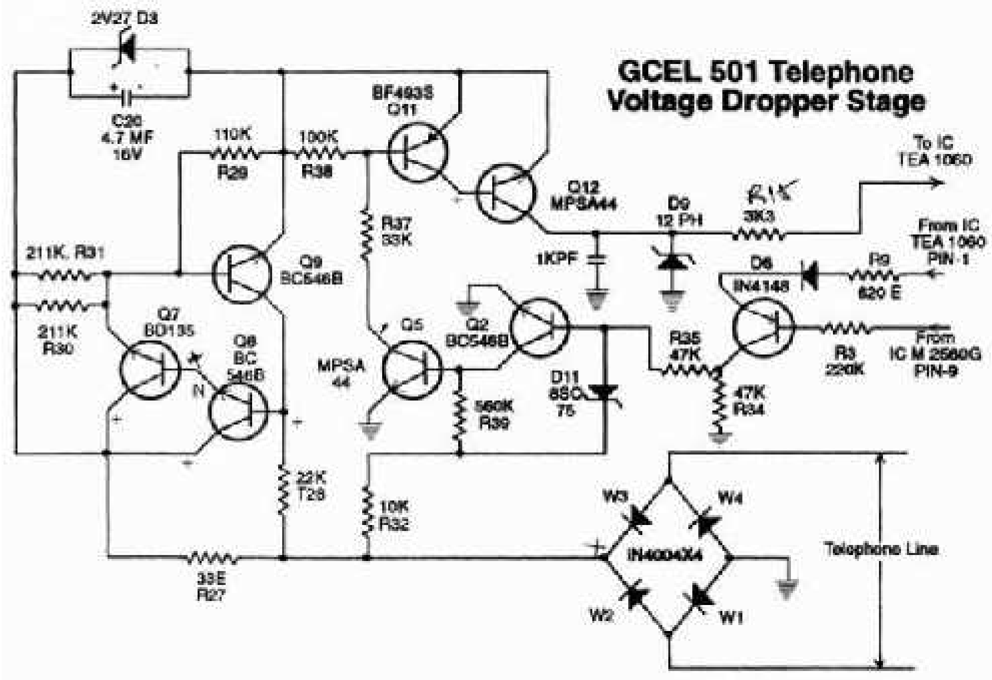


Fig.3.6 Voltage dropper stage

f) Rectifier / Protection Section (fig.3.7):

The rectifier section is also called as "Polarity guard".

Functions of this section are,

1. To supply voltage of the proper polarity to the input of the telephone circuit for proper functioning and avoids damage to electronic circuit. This is done by bridge rectifier.
2. A Metal Oxide Veristor (MOV) or Voltage Dependent Resistor (VDR) of 95V rating is connected across the input of the bridge circuit to limit any high peaks of voltages due to induction on telephone lines.
3. The output of the bridge rectifier is connected to the dialer section, voltage dropper section and Ringer section. IN4004 diodes are used in bridge rectifier.

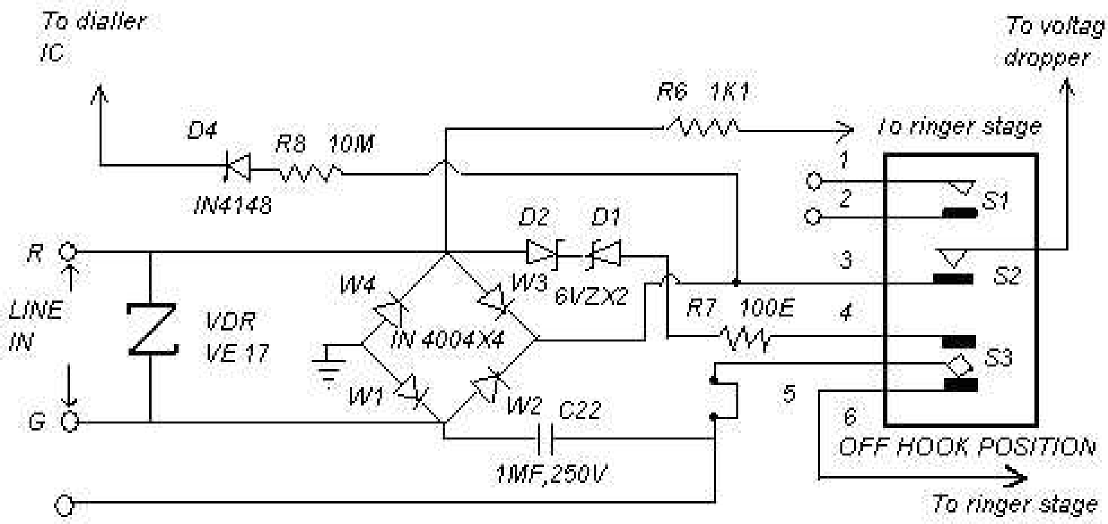


Fig.3.7 Rectifier protection stage

3.2 DTMF Phone

DTMF means “Duel Tone Multi Frequency” signaling is used in telephony transmission system due to the following advantages over pulse signaling.

1) Dialing speed is fast – each digit takes just a few ms to transmit. However, pulse digits will take between 1 and 2 seconds each.
2) It is more reliable because the decoding and switching operations are accomplished electronically, so sensitive electromechanical devices can be eliminated.
3) The tones may be used for signaling purposes after connection has been made.
   This method of signaling system uses 16 distinct voice band frequency signals, each consisting of sinusoidal signals one from a ‘low group” and other from a “high group”

	Higher group Tones			
	H1	H2	H3	H4
Lower group tones	1209 HZ	1336 HZ	1477 HZ	1633 HZ
L1= 697 HZ	1	2	3	A
L2 = 770 HZ	4	5	6	B
L3 = 852 HZ	7	8	9	C
L4 = 941 HZ	*	0	#	D

Fig.3.8 DTMF Key-pad & frequency selection

The A, B, C and D buttons are used in special applications and are not part of the common telephone key board.

Column H4 is normally available on a telephone keypad and reversed for special signaling.

DTMF Decoder

DTMF decoder is necessary in any system using DTMF signaling. The purpose of decoder is to decode a valid pair of signaling tone and provide output data corresponding to DTMF signaling received.

The DTMF decoder must have the following characteristics: -

1) The decoder should decode DTMF signals within ±1.5% of their frequencies and should not decode signals with either frequency deviation more than 3.5% from the standard.
2) The decoder should decode DTMF tone bursts on short as 40 mS and recognize inter digital intervals as short as 40ms. It should not recognize tone burst or inter digital intervals shorter than 20 ms.
3) The decoder should decode DTMF signals in the presence of a dial tone that has each of its frequencies at a level of –16 dB to +3dB.
4) The decoder should decode DTMF signals that have a power per frequency of 25 to 0 dBm, with the high frequency tone –4 to –8 dB relative to the low frequency tone.
5) The decoder, in the presence of message circuit noise, should miss decoding less than 1 in 10,000 valid DTMF tone bursts.

3.3 Main and Extension (1+1) Telephones:

It is also called as Twin set. It is used between boss and his secretary. The Instrument which is kept at Boss is known as “Extension” and that kept at secretary is known as “Main”.

a) Installation and Connection

The sets can be installed as shown in the connection and wiring diagram. The lines L1&L2 coming from the exchange are connected to the jack of the main telephone instrument. The same lines are extended to the extension telephone through a SB cable. Power supply of about 12V DC is extended to the two telephones by an adaptor. “Control and “Buzzer” lines are also extended between the two telephones. A total 3 pairs are required between the two telephones. Both the telephones are identical and can be used in place of each other. In case of no power both the instruments will become parallel on exchange lines. The inter connection wiring diagram is shown below (Fig.3.9)

HOW TO USE:

This is a tone / Pulse Switchable system which can be used to access both Pulse dialing (Loop signaling) and Tone Dialing (DTMF Signaling). The instruments are also having the facility of dynamic change over to Tone mode by pressing ‘*’ key while in Pulse mode.

Main & Extension interconnection

The figure below is showing wiring for main and extension interconnection.

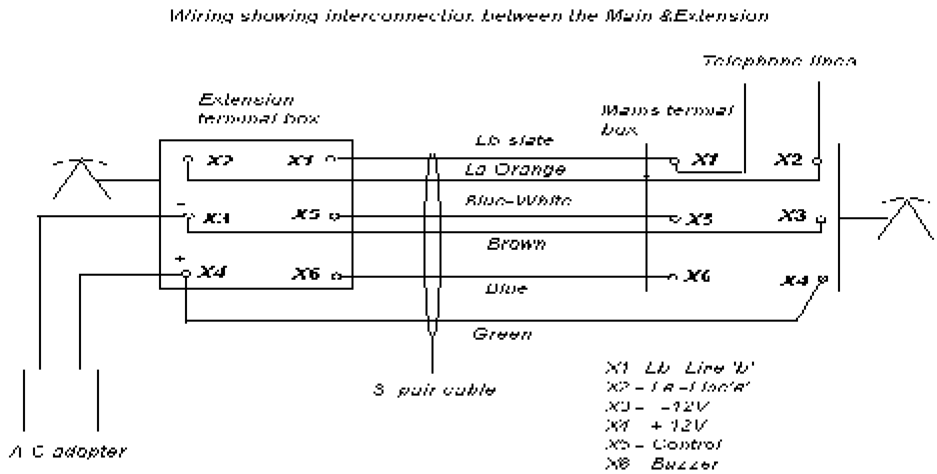


Fig.3.9 Main & Extension interconnection

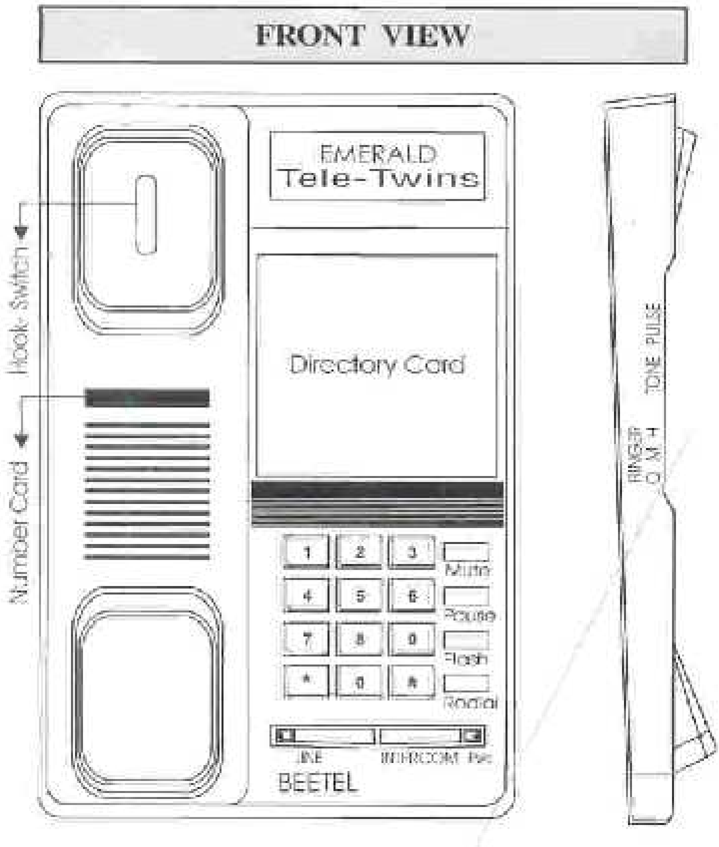


Fig.3.10 PB Phone Front View

OUTGOING CALLS:

PULSE DIALING (Tone/Pulse switch on Pulse)

Go off hook.

Press ’LINE’

Dial the desired number.

After conversation, replace the handset. REDAIL (In case the line is not connected to desired number after dialing)

Go off hook.

Press ‘LINE’.

Press ‘REDIAL’.

After conversation, replace the handset.

DTMF DIALING (Tone/Pulse switch is on Pulse)

Go off hook.

Dial’*’ and then the desired number.

After conversation, replace the handset.

RDIAL

Go off hook.

Press ‘LINE’.

Press ‘REDIAL’.

After conversation, replace handset.

DTMF DIALING (Tone/Pulse switch is on Tone)
Go off hook.
Press ‘LINE’.
Dial the desired number.
After conversation, replace the handset.

REDIAL
Go off hook
Press ‘LINE’.
Press ‘REDIL’.
After conversation, replace the handset.

INCOMING CALLS: Whenever there is any incoming call, there comes ring on both the instruments. The volume of the ring can be adjusted as per your requirement. Press ‘LINE’ to receive incoming calls from any of the instruments.

INTERCOME FACILITY: If you want to consult your companion on other side, press ‘INTERCOME’ to call from any of the extension to other extension.

CALL TRANSFER: To transfer the call, press ‘INTERCOM’ from the extension which is on line. The ‘BEEP’ will indicate the transfer. The outer party will get the music. Press ‘LINE’ from the other extension to complete the transfer.

HOLDING THE LINE: To hold the line, press ‘INTERCOME’. The line will be put to hold. You can speak through the intercom without the caller over-hearing your conversation. Press ‘LINE’ to get back the line.

RINGER VOLUME CONTROL: The extensions are provided with a3- position ringer volume control switch on the right side. The volume can be adjusted at OFF, MEDIUM & HIGH as per requirement. The OFF option is primarily given for the situations in which the Manager/Boss does not want to be disturbed on incoming calls.

PAUSE: When the Pause button is pressed, it introduces a delay of about 2.2 Sec. between two digits dialed before and after the PAUSE.

MUTE: As long as the MUTE button is pressed, no sound will be transmitted to other party.

FLASH: The FLASH button can be used for call waiting. It may be used to access facilities offered by some EPABXs or Network services provided by the telephone company.

3.4 Hand free telephone with caller ID compatible:

This type of telephone is compatible with a caller ID service offered by your telephone company. The calling part’s information will be displayed after the first ring.

The unit can record information of up to 50 different callers for both lines combined, including the time and date received, the called line, and the number of times called, in the caller list.

Integrated telephones (Panasonic make)

(a) Single line with Data connection with speaker phone facility.

It is a programmable telephone. It is provided with a LCD. For the function of LCD and speaker phone 3 AA type Alkaline or Manganese cells are to be installed into the space provided on the back side of the instrument. The location of controls is shown tin the figure **3.11.**

It is provided with TWO line jacks, one is to connect with single-line Telephone Jack and the other is Data jack. Data jack can be connected to,

- Computer
- Modem
- FAX
- Answering machine

It is provided with a PROGRAM button. We can program different functions. For each setting the PROGRAM button to be pressed at the starting and ending of each setting.

(i) Different types of programs are: Selecting the Dialing mode either TONE or PULSE, Time adjustment, Setting the LCD Contrast, Storing Phone Numbers in Memory.

- You can store 10 phone numbers in memory 0 to 9 number buttons function as memory stations.
- You can store up to 20 phone numbers in the one-touch auto dial buttons (10 number in UPPER memory locations, 10 numbers in LOWER memory locations.
- Other facilities are erasing the stored numbers, Turning Music Played during the Hold ON/OFF, Selecting the flash time, Setting the PIN code to prevent unauthorized persons from using your unit, Dial lock, Call Restriction.

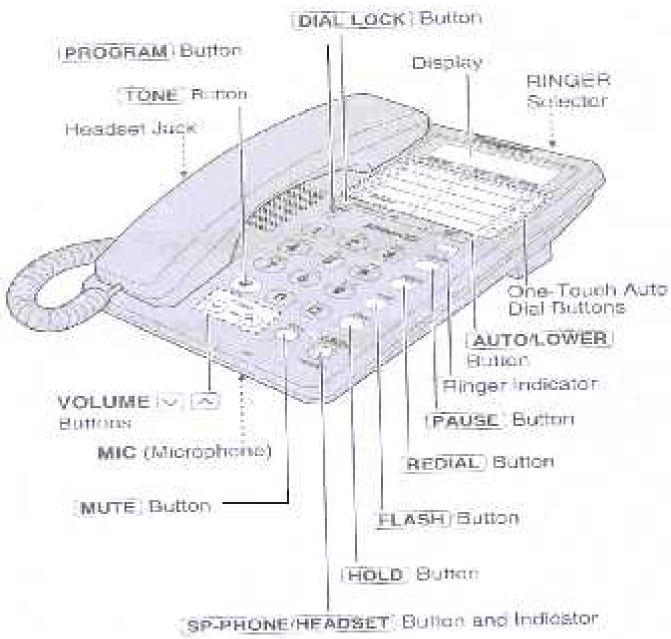


Fig.3.11 Location of controls on Panasonic phone

(ii) Special Features: Call waiting service**,** Temporary Tone dialing**,** Using PAUSE BUTTON**,** Muting the conversation.

(b) Two line with speaker phone facility: It is provided with 2 line jacks. One is used to connect telephone line and the other is to connect telephone line or Data line. It is not provided with caller ID. Location of controls is shown in the **Fig 3.12.**

(c) Caller ID compatible (Fig.3.12): It is a telephone with Caller ID, Call waiting service and Voice Mail Service. The different controls are shown in the figure 2.11. It is having a LCD display. It requires 3 'AA' size Alkaline cells (LR6) or Manganese cells (R6, UM-3) for the LCD working.

(i) Settings: Time and Date setting, Dialing mode setting, LCD contrast setting, Ringer volume setting

(ii) Making calls: Outgoing calls**,** Answering incoming calls.

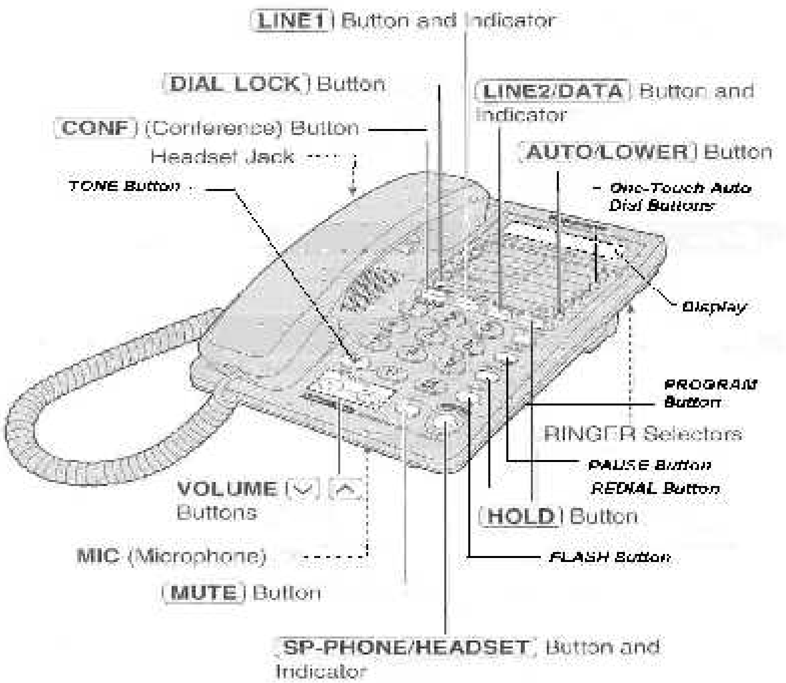


Fig.3.12 Location of controls on caller ID phone

(iii) Features:

- ✓ Caller Identification feature.
- ✓ Viewing the caller list.
- ✓ Calling back from the caller list
- ✓ Editing the Caller's Phone Number.
- ✓ Storing Caller List Information in the Phone Book List.
- ✓ Erasing Caller List Information.
- ✓ Storing Names and Phone Numbers in Phone book list.
- ✓ Dialing from the Phone book.
- ✓ Phone book Editing.
- ✓ Phone book Erasing.

(iv) Special Features:

- ✓ Voice mail service- Listening to voice mail services.
- ✓ Temporary tone dialing
- ✓ Using the PAUSE button if pause is required.
- ✓ Answering call waiting service.
- ✓ Setting the password.
- ✓ Dial Lock.

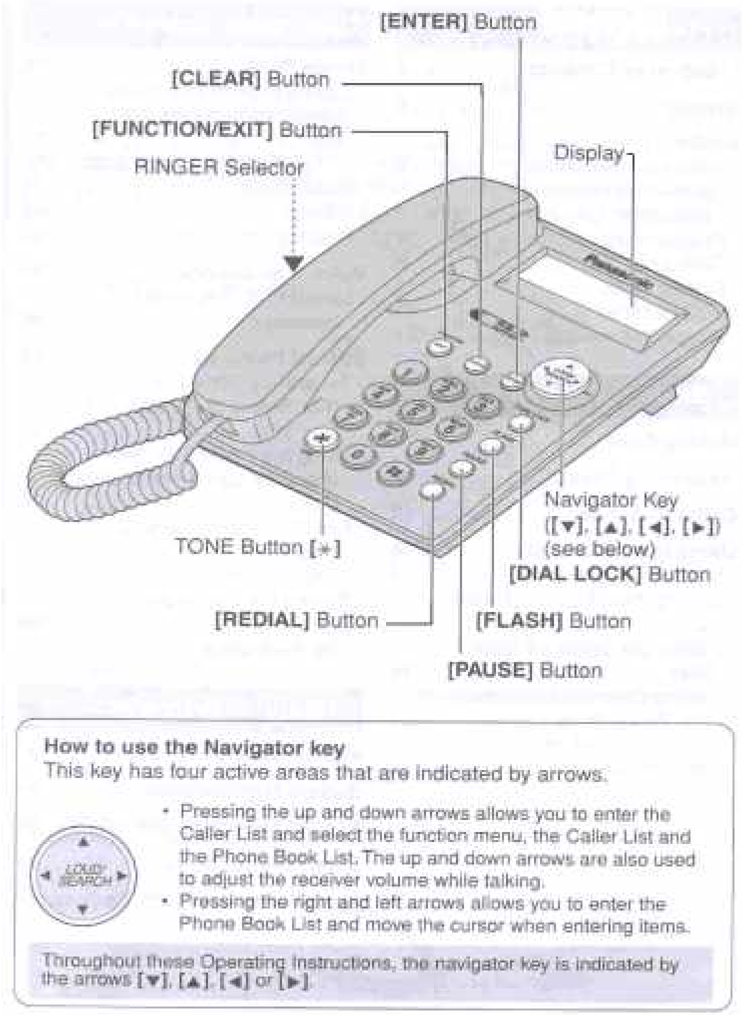


Fig.3.13 Caller-ID phone with navigator key

(v) Battery replacement procedure:

If “ ” flashes. The battery power is low. Install new batteries as soon as possible.

- ✓ Disconnect the telephone line cord from the unit.
- ✓ Press down in the direction of the arrow and remove the battery cover.
- ✓ Replace the batteries with new ones using correct polarity (+,-)
- ✓ Connect the telephone line cord to the unit.

- After battery replacement, the information stored in the Redial List will be cleared. Store the desired item in the Phone Book List and Caller List.
- The time will be shown as “12:00AM 31/12 or “0:00 31/12” after replacing the batteries, readjust the time and date.

3.5 Two line CLIP with speaker phone facility.

It also 2 line telephone, in addition it is provided with CLIP (Caller Line Identification Program). Location of different controls is shown in the **Figure 3.14**

(i) Battery Installation:

- Install the three “AA” size Alkaline(LR6) or Manganese (R6,UM-3) batteries in battery compartment. They work as emergency power during a power failure.
- During power failure the battery will retain the clock memory and redial memory. If you do not install the battery, the data in the memory will be lost during a power failure.

If “ ” flashes. The battery power is low. Replace all the cells with new ones.

Disconnect the telephone line cords before opening the battery cover.
- You do not need to disconnect the AC adaptor, otherwise the clock memory and the redial memory will be lost. If ” O “ flashes on the display, adjust the clock.

(ii) Programmable settings:

- ✓ Time and Date adjustment.
- ✓ Assigning the extension number.
- ✓ Dialing Mode (PULSE/TONE).
- ✓ LCD Contrast.
- ✓ Ringer volume adjustment.
- ✓ Ringer pattern selection.

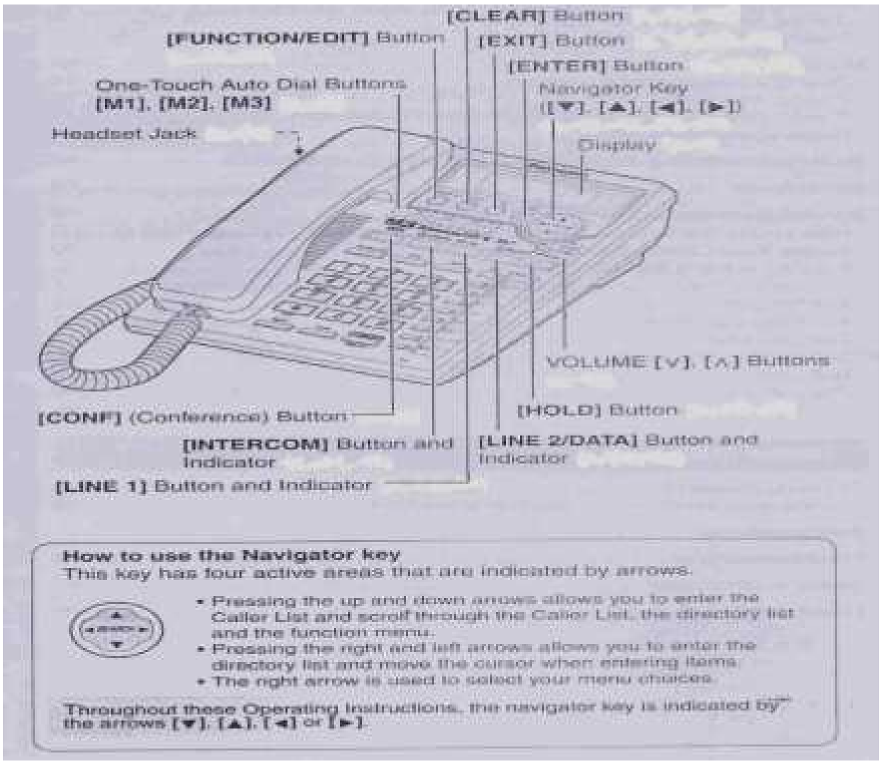


Fig. 3.14 a Two line CLIP with speaker phone

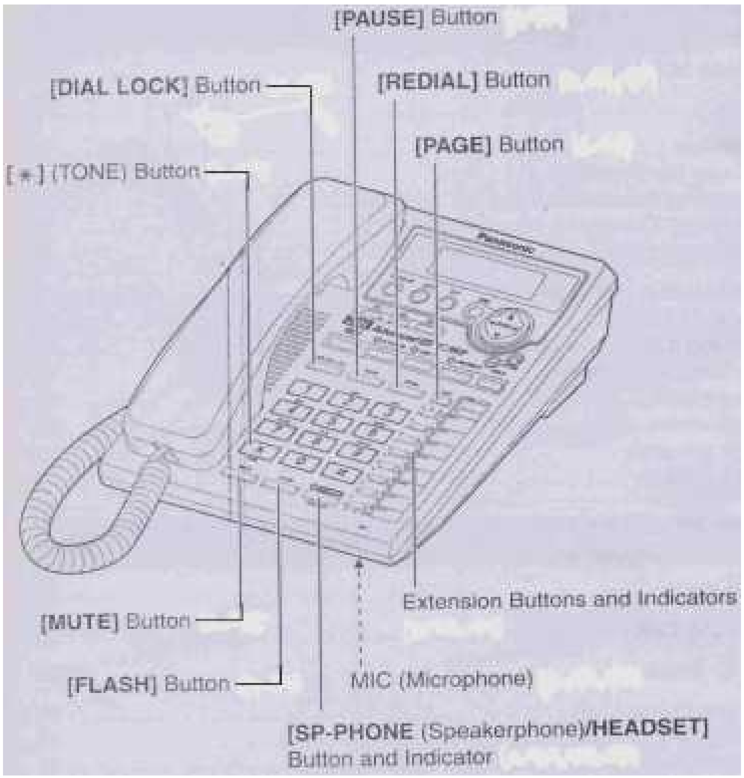


Fig.3.14 b Two line CLIP with speaker phone

(iii) Making outgoing calls:

- ✓ Using the speakerphone
- ✓ Adjust the speaker volume
- ✓ To redial the last number dialed
- ✓ To redial using the redial list (Memory redial)
- ✓ To release the hold

(iv) Answering Incoming calls:

- ✓ Using the speakerphone.
- ✓ Using the Caller List
- ✓ Calling Back from the Caller List
- ✓ Editing the Caller’s Phone Number
- ✓ Storing the Caller List Information in the Directory or in One-Touch Dialer Memory
- ✓ Erasing Caller List Information

(v) Directory facility:

- ✓ Storing Names and Numbers
- ✓ Finding stored items
- ✓ Editing
- ✓ Erasing

(vi) One-Touch Dialer:

- ✓ Storing Names and Numbers
- ✓ Dialing a stored number

(vii) Intercom facility;

- ✓ Paging a Designated Extension
- ✓ Paging all extensions
- ✓ Transferring an External Call to another Extension
- ✓ Room Monitor Feature
- ✓ Making/Answering another call during a conversation
- ✓ Conference facility

(ix) Special Features:

- How to use the PAUSE button
- Muting your conversation
- For call waiting service users
- Temporary Tone dialing
- Incoming call tone
- Line selection
- Setting a password
- Dial lock
- Call restriction
- Call Privacy Features
- Connecting on optional Handset to the Unit.
- Wall mounting facility.

(e) Cordless phone

Introduction: It is a **TWO** line cordless Telephone having a base unit & a hand set. Calls are transmitted between the two using wireless radio waves.

For maximum distance and noise-free operation, the recommended base unit location is:

- Away from electrical appliances such as TV, radio or personal computer.
- In HIGH and CENTRAL location with no obstructions such as walls.
  It works on a rechargeable Ni-Cd battery power for handset. Charge the battery for about **10 hours** before initial use.
- The IN USE/CHARGE indicator lights.

An AC adaptor is required for working between the base unit and handset.

- USE ONLY WITH Panasonic AC ADAPTOR PQLV14BX.
- The AC adaptor must remain connected at all times. (It is normal for the adaptor to feel warm during use)
- When more than one unit is used, the units may interfere with each other. To prevent or reduce interference, please leave ample space between the base units.

Connecting the Telephone Line Cord:

- One 4-wire Telephone Line Cord can be connected to a two-line telephone jack.
- One number 4-wire telephone line cord & One number of 2- wire telephone line cord can be connected to the respective line jacks.

✓ To use as single line telephone One 2-wire telephone line cord can be connected to single line telephone jack.

The location of controls are shown in the below figure **3.14 a&b.**

Settings:

You can program the following:

- ✓ Selecting the dial mode i.e., TONE or PULSE.
- ✓ Selecting the line mode: Line 1 and Line 2 can be set to select “A” or select “B”.
- ✓ Ringer volume can be selected to HIGH or LOW
- ✓ Outgoing calls can be made either from base unit or hand set.
- ✓ 10 Phone numbers can be stored in the handset.
- ✓ 10 Phone numbers can be stored in memory of base unit.
- ✓ The stored numbers can be dialed from either Hand set or Base unit,
- ✓ The handset and base unit can be used at the same time on separate telephone lines.

Conference Call:

While having a conversation on one line, you can make or answer a second call on the other line and then combine the calls to make a conference call.

A **two way intercom** is possible between the handset and the base unit.
The intercom can be used during a call. This feature enables you to transfer a call between the handset and the base unit.

SPECIAL FEATUERES:

- ✓ Call waiting service.
- ✓ Temporary Tone Dialing.
- ✓ Automatic security code setting.
- ✓ Using the PAUSE button.
- ✓ Using the FLASH button to use special features.
- ✓ Selecting the line automatically.
- ✓ Indicating another incoming call by TONE.
- ✓ Wall mounting facility.

Recharging the Battery:

When the RECHARG indicator flashes, or the unit beeps intermittently, recharge the battery for about 10 hours.

If you do not recharge the handset battery for more than 30 minutes, the RECHARGE indicator will continue to flash.

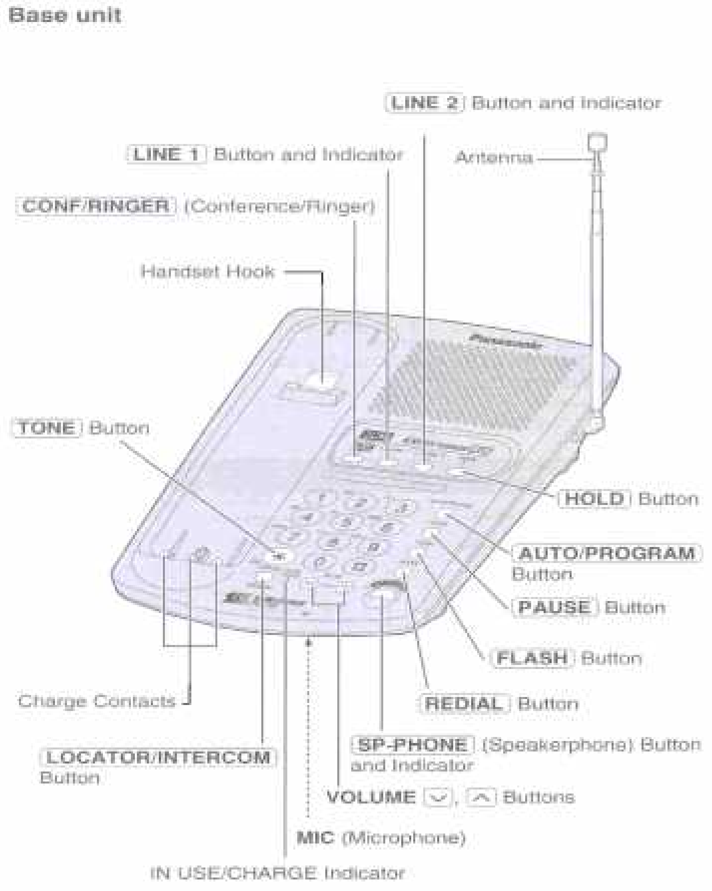


Fig.3.15 a 2-Line Cordless phone base station

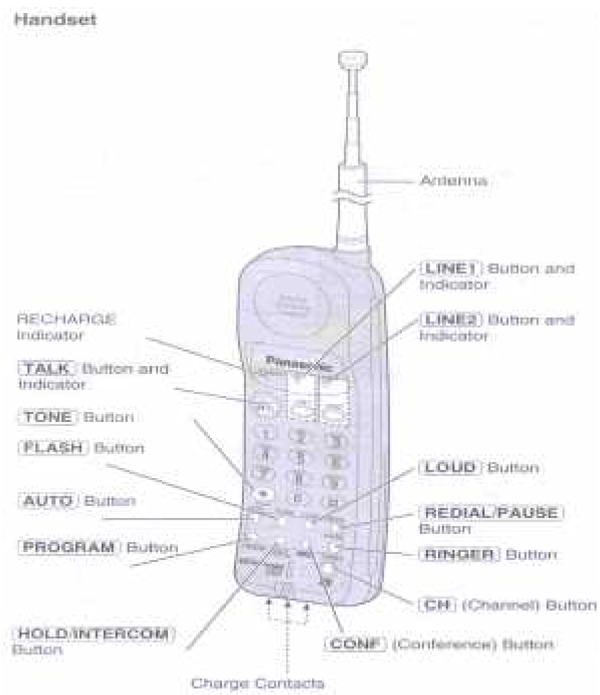


Fig.3.15 b 2-Line Cordless phone handset

Battery information: After Panasonic battery is fully charged:

Operation	Operating time
While in use (TALK)	Up to about 6 hours
While not in use (Stand-by)	Up to about 30 hours

- The battery operating time may vary depending on usage conditions and ambient temperature.
- **Clean the hand set and the base unit charge contacts with a soft, dry cloth once in a month. Clean more often if the unit is subject to grease, dust or high humidity.** Otherwise the battery may not charge properly.
- If the battery is fully charged, you do not have to place the handset on the base unit until the RECHARGE indicator flashes. This will maximize the battery life.
- The battery cannot be overcharged.

ISDN TELEPHONE:

Introduction:

An **ISDN telephone** is an all-digital **telephone** designed to take full advantage of the many **features of ISDN**

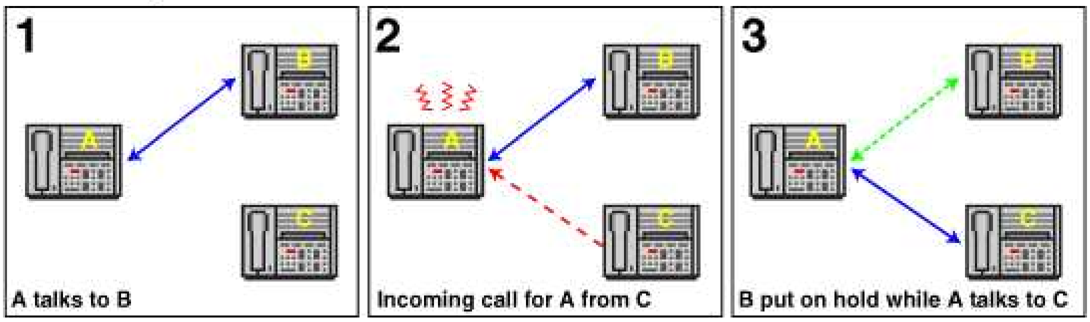
Features of ISDN Telephone:

Features of ISDN Phone are listed below:

- PABX functionality
- password protection
- Calling / Connected Line Identification
- Call Hold, Toggling (HOLD)
- Terminal Portability (TP)
- Call Waiting (CW)
- Call Forwarding (all CF, CFB, CFNR)
- Callback (CCBS, CCNR)
- Three-Party Conference (3PTY)
- Explicit Call Transfer(ECT)
- DTMF and keypad dialing possible
- Display of list of Missed calls, receive calls, redial numbers, and Hotkey numbers.

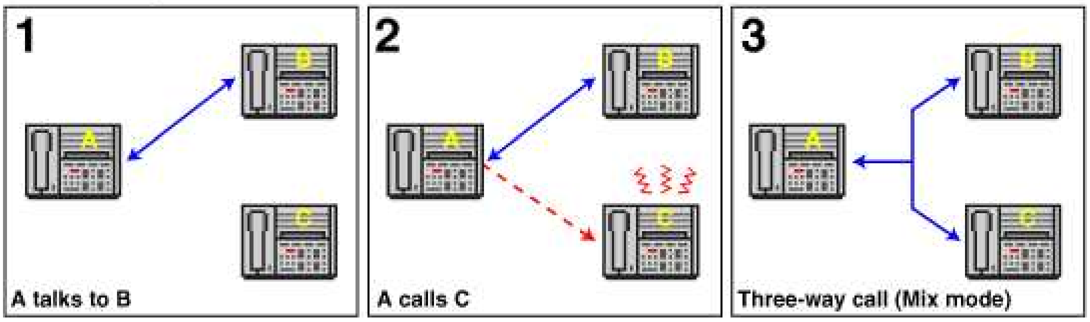
Call Waiting allows you to put a call on temporary hold and answer an incoming call.

Call Waiting



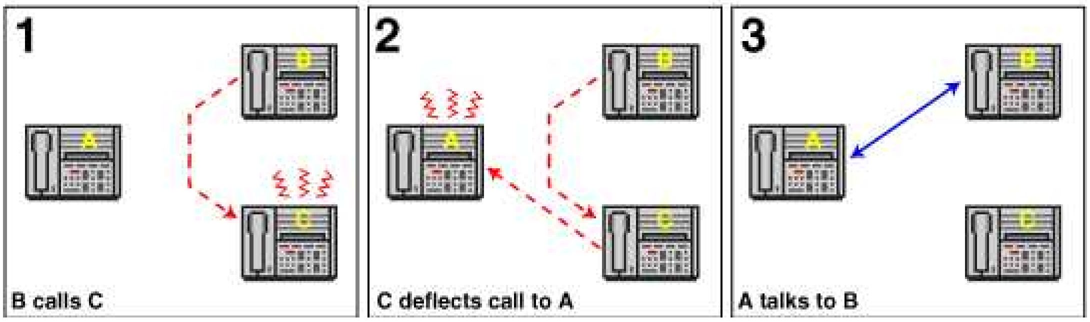
Three-party Service allows you to hold a three-way telephone conversation.

Three Party Service



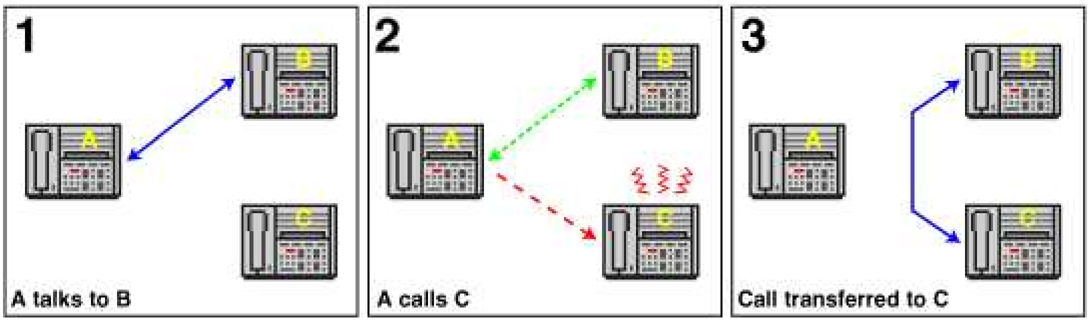
The Three-party service has two modes of operation: Mixing and Switching. In Mixing mode, all three callers can talk simultaneously. In Switching mode, only two callers can talk simultaneously. For example, A can talk to either B or C, but B cannot talk to C and versa.

Call re-direction redirects incoming calls to a specified telephone.



Call Transfer allows you to transfer a call in progress to another telephone.

Call Transfer



3.5 IP PHONES and VIDEO PHONES

IP Phones and Video Phones

As the advancements are going on in communication technology, the trend of using the same is ever more growing among the users. Merger of various services like voice, data, text and video including value added services are available in a single window and the products are available worldwide with international standards.

In today's world, it's necessary to adopt new technology. This is also one of the symbolic representations for development in the field of industry, business, trade, education, medical science and never ending list of such things. No doubt the communication technology has changed the status and standard of human life.

IP Phone:

The Phone instrument which can be connected on internet is called as IP phone for voice, data and video calling. These instruments are different from our conventional telephone instruments. IP Phones are sometimes called VoIP telephones, SIP phones or soft phones. These are all the exact same thing and are based on the principle of transmission of voice over the internet, or what is better known as VoIP (or voice over internet protocol) technology.

SIP Phones are the same thing as VoIP Phones or soft phones. These are telephones that allow phone calls to be made using VoIP (voice over internet protocol) technology.

There are two types of SIP Phones. The first type is the hardware SIP phone, which resembles the common telephone but can receive and make calls using the internet instead of the traditional PSTN system.

SIP Phones can also be software-based. These allow any computer to be used as a telephone by means of a headset with a microphone and/or a sound card. A broadband connection and connection to a VOIP provider or a SIP server are also required.

SoftPhone for Windows, Android and Iphone

Softphone are those which can use to make and receive VoIP phone calls from your PC, Iphone or Android based smartphone. The advantage of using soft phone is that you can leverage low cost or free VoIP calls and you can connect to the company VoIP PBX and work remotely.

To use softphone you will need:

- ✓ Microsoft Windows XP, Vista and 7 OS on your PC
- ✓ For Android 1.6, 2.1, 2.2 based devices such as Google Nexus, Sony Xperia, Motorola Droid or Samsung Galaxy.
- ✓ An account with a VOIP provider or a SIP server / VoIP PBX
- ✓ On Windows a headset
- ✓ An internet connection or a mobile provider
- ✓ Ensure that your mobile provider **allows VoIP** and has 3G enabled. Some mobile operators block VoIP calls purposely to avoid revenue loss

Soft phone benefits

- ✓ Work remotely by registering the softphone to your company PBX.
- ✓ Quick and simple installation.
- ✓ Free download from Android market or Apple appstore.
- ✓ Open Standards based next generation softphone.
- ✓ Easy to use, intuitive user interface with dial pad and buttons.
- ✓ Identical user experience on Windows, Android and Iphone.
- ✓ MSI installation allows for easy network wide installation
- ✓ Completely free – saving licensing costs and licensing administration
- ✓ Environmentally friendly - Softphones save electricity!
- ✓ Substantial savings on telephone bills
- ✓ 3CXphone also works seamlessly as an IP phone connected to 3CX Phone System for Windows, an award winning IP PBX that helps businesses break free from obsolete proprietary hardware based PBXs.

VOIP, which stands for Voice over Internet Protocol, is basically the transmission of voice traffic over IP-based networks. Initially designed for data networking, the Internet Protocol (IP) was adapted to voice networking following its successful positioning as the global standard for data networking.

With VOIP phone systems users are not limited to making and receiving calls through the IP network, traditional phone lines can be used to guarantee a higher call quality and availability. With the use of a VOIP gateway incoming PSTN/telephone lines can be converted to VOIP/SIP. This way the VOIP gateway allows the user to receive and place calls on the regular telephony network.

VOIP PBX systems provide mobility to employees, flexibility when a business expands as they are much easier to manage than the traditional PBX, and can also considerably reduce telephony administration costs.

Some of the models which are used in Coral Flexicom Exchanges with display units, soft keys and hands free calling

Flexicom-soft phone can be installed on computer
FlexiAir- wireless handset which is used based on DECT technology.

IP-Video Phones

As the voice is to be transmitted on Internet Protocol, it is needed to have a precise speech transmission between various devices used for the purpose. Improved, noise free quality of voice is the ultimate goal on VoIP. For this purpose various ITU (T) recommended and other patent VoIP CODEC (Code and Decode) are used.

G.711: this codec helps in delivering precise speech transmission and not using much processor. it is basically known as pulse code modulation which samples the voice signal in to 8 voice samples and 8000 samples per second resulting in 64kbps data. It needs at least 128kbps for two way communication.

G.722: G.722 is an ITU standard codec that provides 7 kHz wideband audio at data rates from 48, 56 and 64 kbit/s. This is useful for voice over IP applications, such as on a local area network where network bandwidth is readily availablethis provides improved voice quality due to wide speech bandwidth. G.722 sample audio data at a rate of 16 KHz (using 14 bits), double that of traditional telephony interfaces, which results in superior audio quality and clarity.

G.723.1: **G.723.1** is an audio codec for voice that compresses voice. G.723.1 is mostly used in Voice over IP (VoIP) applications due to its low bandwidth requirement.

G.726: **G.726** is an ITU-T ADPCM (adaptive differential pulse code modulation) speech codec standard covering the transmission of voice at rates of 16, 24, 32, and 40 kbit/s. It is primarily used on international trunks in the phone network and are the standard codec used in DECT wireless phone systems.

G729: **G.729** is an audio data compression algorithm for voice that compresses digital voice in packets of 10 milliseconds duration. Because of its low bandwidth requirements, G.729 is mostly used in Voice over Internet Protocol (VoIP) applications where bandwidth must be conserved, such as conference calls.

GSM: The GSM standard was developed as a replacement for first generation (1G) analog cellular networks, and originally described a digital, circuit switched network optimized for full duplex voice telephony. "GSM" is a trademark owned by the GSM Association. **GSM** (**Global System for Mobile Communications**, originally ***Groupe Spécial Mobile***), is a standard set developed by the European Telecommunications Standards Institute (ETSI) to describe protocols for second generation (2G) digital cellular networks used by mobile phones

iLBC: Internet Low Bitrate Codec (**iLBC**) is an open source royalty-free narrowband speech codec, developed by Global IP Solutions (GIPS). It is suitable for VoIP applications, streaming audio, archival and messaging. iLBC handles the case of lost frames through graceful speech quality degradation. Lost frames often occur in connection with lost or delayed IP packets.

Speex: Speex is an Open Source/Free Software patent-free audio compression format designed for speech. Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs. Speex is based on CELP and is designed to compress voice at bitrates ranging from 2 to 44 kbps.

Objective:

1. Ringer stage produces audible ringing sound. (T/F)
2. Lifting the handset results in voltage drop from -48volt to +5v or +12v. (T/F)
3. One row and one column frequency is selected for pressing one digit on key pad. (T/F)
4. # and * key are special function key. (T/F)
5. DTMF stands for Dual Tone Multi Frequency. (T/F)
6. The purpose of decoder is to decode a valid pair of signaling tone. (T/F)

Subjective:

1. What are different types of telephone instruments used in Indian Railways?
2. What is DTMF? How they are produced?
3. Write short note on Cordless Telephone instrument.
4. What are the main features of ISDN telephone?
5. Write short note on IP phones and Video phones.

CHAPTER 4

TELEPHONE EXCHANGES IN INDIAN RAILWAYS

4.0 Indian Railways is the largest organization fulfilling the need of Passenger transportation and goods transportation in our country. To facilitate the smooth and proper functioning of traffic, real time information is communicated among various departments of Indian Railways. To achieve this, Railways have its own communication system such as wireless communication, optical fiber communication and land line communication for base telephones.

Exchanges in Indian Railways are providing communication between various functionaries such as operating department, commercial department, engineering department, electrical department, S&T department and so on.

Exchanges have been installed at All Zonal Head Quarters and Division Head Quarters.
Exchange is a switch which connects two subscribers in local exchange and as well as connecting two subscribers on a trunk.

Exchanges are installed according to the requirement: such as,

- ✓ Number of subscribers
- ✓ Trunking between various Zones and division
- ✓ Services and features available for subscribers

Classification of Exchange on the basis of mode of operation: Telephone exchanges can be classified as Manual type and Automatic type.

Manual Exchanges are “operator” dependent. In these types of exchange, operator is connecting calls between subscribers.

Automatic Exchanges are programmable “switch” which carry out all the call process and execution according to pre defined instructions and switching is done automatically. Therefore Automatic Exchanges are also called as Stored Program Controlled Exchanges.

In case of “Strowger exchanges” the switching is done by “Electro-mechanical switches”.

Facilities provided by Electronic exchanges:

Facilities are THREE types-

- ✓ Facilities to subscribers
- ✓ Facilities to administration
- ✓ Facilities to the maintenance personal

Some of the Subscriber’s facilities are,

- ✓ MFB push button dialing (Multi Frequency Buttons) - Priority subscriber lines
- ✓ Toll (outgoing calls) restriction
- ✓ Service Interception
- ✓ Abbreviated dialing
- ✓ Call Forwarding
- ✓ Do not Disturb
- ✓ Conference calls
- ✓ Camp on busy
- ✓ Call waiting

- ✓ Malicious call identification
- ✓ Call charge print out/immediate billing
- ✓ Interception or announcement
- ✓ Hot line
- ✓ Automatic wake up
- ✓ Denied Incoming calls
- ✓ Instrument locking / dynamic locking
- ✓ Free of charge calls / tool free

Facilities to administration:

- ✓ Reduce switch room accommodation
- ✓ Faster installation and easy extension
- ✓ Versatility
- ✓ Economic consideration
- ✓ Automatic test of subscriber line

Maintenance Facilities:

- ✓ Fault processing is automatic.
- ✓ Diagnostics: Location of fault by maintenance staff on demand can be done by programme.
- ✓ Statistical programs - traffic condition, trunk occupancy rate etc. - Blanking - Stopping of calls from subscribers.

Some of the Exchanges which are used in Indian Railways are:

- ➢ **C-DOT** Exchanges (128 port/256 port PABX): mainly used for Intercom for GM, COM, CEE, CSTE etc.

- ➢ **IRIS- IVDX**: it is used as main PBX in Zone or Division Head Quarter and connected with other zones on trunk.

- ➢ **Coral Flexicom series**: this Exchange is also used as Main Exchange in Zone and Division.

- ➢ Various series of coral Flexicom are available. Coral ISBX 5000, Flexicom 6000, IPX 500 and IPX 3000

- ➢ **Siemens Hipath 3800 and Siemens Hipath 4000** also used as Main Exchange in Zone/ division Head Quarter.

<u>C-DOT Exchanges:</u>

C-DOT means “Center for Development of Telemetrics”

This is a central government organization of India set up to develop the necessary equipments (infrastructure) suitable for Indian climatic and environmental conditions.

Special features of this exchange:-

- ✓ Single frame terminal unit capable of 128 ports
- ✓ Standard programme control
- ✓ It is modular in design and hence expandable
- ✓ Man machine communication

- ✓ Low power consumption
- ✓ No air condition is requires
- ✓ No single LCC fault effects more than 8 terminations
- ✓ It is reliable
- ✓ The digital switching system of C-DOT technology will offer 100% non-blocking voice and data network.
- ✓ Easy system starts up.
- ✓ Faults are indicated through maintenance panel in the form of audio and visual alarms.
- ✓ Calling party release or called party release (affected after 60 seconds).
- ✓ Automatic system alarms in case of duplicate unit failure, battery low and power supply unit failure.
- ✓ Remote testing facility to check system status.

4.3 Types of C-DOT exchanges:

There are two types of C-dot exchanges, used in Indian Railways.

C-DOT 128 port RAX (Rural Automatic Exchange): It is designed to meet the telecommunication needs of small sized rural areas. These exchanges are also suitable for Indian Railway applications, where the Telephone line capacity is less. It can be expandable up to 400 lines.

It is a cabinet type exchange. Its front side is provided with hinged doors.

The cabinet contains equipment frame, in which there will be 26 card slots guides filled with 26 cards.

The Mother Board is the Multilayer PCB which connects all the cards. The cards in C-DOT exchange are classified as,

Peripheral cards or Terminal cards: These include subscriber’s cards (LCC), Trunk and Tone cards. These cards are not duplicated.

Control Cards: These cards control the working of the exchange. These cards are duplicated.

Power Supply Unit (PSU) CARDS: This generates necessary D.C voltages required for the exchange. This is also duplicated.

Maintenance: The maintenance and system administration is done by a ”Maintenance panel” which is connected to RAX control processor card, through RS 232C. Microprocessor 65C02 is provided for performing the above functions.

Some key points:

Switching: Digital PCM’A’ law CCITT standards non-blocking
Control: Microprocessor based Stored Program Control (SPC).
Capacity: Up to 96 subscribers with 8 Trunks.

C-DOT 128 Port PBX: PBX stands for “Private Branch Exchange”. This is normally used within the office to connect a number of internal users to a smaller number of outside lines on the public network.

The extensions may be served by a conventional analog telephone network. The block diagram is same as RAX.

The cabinet contains equipment frame, in which there will be 26 slots guides. In this only 23 cards can be housed. The motherboard is Multilayer PCB, which connects all the cards, external voice, and data connections are made from Main distribution frame.

With this exchange:

Maximum number of Extensions can be provided are -94 (Analog)
Maximum number of Junctions can be provided are - 16
Maximum number of Data terminations can be provided are -32

Features:- Some provisions are made in C -DOT exchange to use the system effectively ,

These are,

- ✓ System Features
- ✓ Extension Features
- ✓ Operator Features

Some of the Basic features are provided along with the system are,

- ✓ Class of service: The user is allowed or disallowed the facilities like, ‘0’ dialing, STD facility, conference facility, paging access facility, tie line accesses facility etc.
- ✓ Extensive Automatic Diagnostics: The diagnostic information regarding the status of cards, junctions, positions, links are displayed by LED’s.
- ✓ Metering: Counting the number of calls made by each extension, junction and Operator is called as ‘Metering’.
- ✓ Power Failure cut through: In case of power failure, the PBX will not work, During this period 2 pre determined extensions are immediately connected to the main exchange through the junction lines.
- ✓ So they can pass the information to the main exchange.
- ✓ Redundancy Scheme: Providing the duplicate cards in the exchange is called as “Redundancy scheme“.
- ✓ Tones: - These are necessary in the exchange to progress a call between two extensions and to know the condition of the exchange. The TGD card generates different tones,

Coral Flexicom-6000(ISDN Exchange)

Coral Flexicom-6000 series exchange is manufactured by Tadiran Telecom, ISRAEL. Fully digital, ISDN compatible, VOIP support and distribution of 6000 port on the basis of Memory allocation for physical as well as virtual ports in this exchange. This distribution is standardized by a Size list default in the system design.

Some key points about this exchange:

1. Modular in design. (cards are modular)
2. Easy up gradation of hardware and software.
3. 512 timeslot processing from peripheral to control, those 4 times more than the normal exchange.
4. Redundant control to enable hot standby feature in case of active copy failure.

5. Automatic hardware detection in a slot.
6. Remote access to the system through dial up modem
7. Various features for subscribers, trunks and Tenant groups.
8. Easy partitioning of exchange on the basis of class of services.
9. Full back up support

Hardware: it includes Peripheral cards, service cards and control cards

Peripheral cards:

- ✓ 24 SA/24 SLS 24 port analog subscriber card
- ✓ 24 SFT/24 SDT 24 port digital subscriber card
- ✓ PRI 30 30 voice channel multiplexed= one digital signalling channel, which works on QSig protocol
- ✓ 30 CEPT 30 voice channel for communication works on MFC signaling protocol.
- ✓ 4TEM 4 port E&M (2 wire/ 4 wire) trunk
- ✓ 8T-C 8 port CO trunk card (2 wire)

Service cards:

- ✓ MFC Multi Frequency Receive card
- ✓ iDSP identity display card
- ✓ 8DRCF digital resources and CONF card
- ✓ 8DTR DTMF Trans receive card
- ✓ PUGW VoIP card to connect analog extensions on IP network
- ✓ CONF 8 no of 3 way/ 2 multi party Conferences 15 party in each.

Control cards:

- ✓ **MCP** this is the Main Control card, includes a flash disc of 128 MB with default configuration, routing, number plan of the exchange.
- ✓ **32 GC** this is a Group Control card, connecting 16 peripheral shelves for inter peripheral switching, includes a Software Authorization Unit.
  The SAU contains all the Licenses for hardware and software in it.

IRIS-IVDX Exchange:

- ✓ It is fully digital, ISDN compatible, VoIP support.
- ✓ Single control card with distributed processors for switching, signaling, synchronization of clocks, tone generation, ringing voltage generation, data base management and accessibility to configuration by serial communication port.
- ✓ Contains 128 MB bootable flash disc, with default configuration for the exchange.
- ✓ Each peripheral shelf is having 16 universal slots and one fix slot for control card. Main rack contains MCC32 and other peripheral shelves contain PCC card on fixed slot.
- ✓ Each shelf can process 512 time slot other than the shelf which is connected as Main Rack, generates only 480 time slots

Hardware: the hardware includes peripheral cards, service cards control cards.

Peripheral cards:

- ✓ FLC-32 32 port analog subscriber card
- ✓ DCC-16 16 port digital subscriber card
- ✓ PRI30 30 voice and one digital channel, follows QSig protocol
- ✓ 30 CEPT 30 voice channels with R2 MFC signaling protocol

- ✓ 8ENM	8 port E&M trunks of 2 wire or 4 wire type
- ✓ FTC16	16 port 2 wire analog CO trunk lines

Service cards:

- ✓ MFR	multi Frequency Receive card
- ✓ VoIP	to connect analog subscribers on IP network
- ✓ ADSL	asynchronous digital subscriber lines to connect data circuits from exchange

Control card:

- ✓ MCC32 is the Main Control card, with distributed processor for carrying out switching, signaling, clock synchronization, data base control, system alarm generation, tone generation and other functions.
- ✓ There is no Power supply card in this exchange, MCC 32 card has a DC-DC converter circuitry to generate all subsidiary voltage required for other peripheral cards
- ✓ It has got inverter circuit to generate ringing voltage.

Siemens Hipath 3800:

- ✓ Siemens Hipath 3800 is a small exchange with 500 ports capacity,
- ✓ support for DECT (digital enhanced cordless telephony)
- ✓ Widely used in ARTs (Accident Relief Trains) on Indian Railways.
- ✓ The exchange is housed in a compact cabinet with ten slots.
- ✓ Self standing cabinet and Modular construction
- ✓ 256 DECT handsets can be operated in the system
- ✓ Menu driven , user friendly programming feature Remote login feature, through dialup modem
- ✓ SDRAM for core program and user database storage
- ✓ Various Telephony features

Hardware: includes peripheral cards and Control cards

- ✓ **CBSAP: Main Control Card**

  Central Board System Application Program - This is the main control card in Hipath 3800 system, it is in the dedicated slot No. 6. This card facilitates in programming the exchange. A PC can be connected to the COM port of this card. Hipath Manager Software is used for the programming. The core operating system, user database is stored on a SDRAM on this card. Core exchange requirements of switching, monitoring, programming and feature authorizations are designed on this card.

Peripheral cards:

- ✓ **SLMO-8**	Single Line Module Digital with 8 port.
- ✓ **SLMA-8**	Single Line Module analog with 8 port
- ✓ **TMEW-4**	Trunk Module E&M with 4 port
- ✓ **TMANI-8**	Trunk Module Analog Network Interface with 8 port CO line.
- ✓ **DIUN**	Digital Interface Unit Network - This is a PRI digital trunk card with 30 ports.
- ✓ **DIUT**	Digital Interface Unit Trunk - This is a E1 digital trunk card with 30 ports.
- ✓ **SLCN**	Single Line Cordless Network - This is a DECT card with 16 ports. Radio base stations are connected to this card with single pair.

Objective:

1. Automatic connection between two subscribers is done by switching system. (T/F)
2. Electronic exchanges are easy to upgrade. (T/F)
3. SPC stands for Stored Program Control. (T/F)
4. Digital switch provides 100% Non Blocking voice and data network. (T/F)
5. RAX stands for Rural Automatic Exchange. (T/F)
6. User cards and trunk cards are located in Terminal group. (T/F)
7. All control cards are duplicated in RAX. (T/F)
8. TGD card generates different types of Tones. (T/F)
9. Total number of slots in RAX exchange is 26. (T/F)
10. MCP card is main control card in coral Flexicom 6000 (T/F)
11. SLMA card in Siemens has 8 ports analog (T/F)
12. CBSAP is the Main Control Card in Siemens Exchange. (T/F)

Subjective:

1. What are the features of electronic exchange?
2. Write in short about various types of Exchanges used Indian Railways.
3. What are the peripheral cards available in Siemens hipath exchange?