College of Electronic Technology

Autumn 2016 **Course: Switching and Networks**

Q-1. What are the major systems of a telecommunication network? Discuss in detail the subscriber loop systems.

Answer:

The major systems of any telecommunication network may consist of the following major systems:

1. Subscriber end instruments or equipments
2. Subscriber loop systems
3. Switching Systems
4. Transmission systems
5. Signalling systems

Subscriber Loop System: Every subscriber in a telephone network is connected generally to the nearest switching office by means of a dedicated pair of wires.

Subscriber loop refers to this pair of wires. It is unwidely to run physically independent pairs from every subscriber premises to the exchange. It is far easier to lay cables containing a number of pairs of wires for different geographical locations and run individual pairs as required by the subscriber premises. Generally four levels of cabling are used as shown in fig. At the subscriber end, the drop wires are taken to a distribution point. The drop wires are the individual pairs that run into the subscriber premises. At the distribution point, the drop wires are connected to wire pairs in the distribution cables.

Many distribution cables from nearby geographical locations are terminated on a feeder point where they are connected to branch feeder cables which, in turn, are connected to the main feeder cable. The main feeder cables carry a larger number of wire pairs, typically 100-2000, than the distribution cables which carry typically 10-500 pairs. The feeder cables are terminated on a main distribution frame (MDF) at the exchange. The subscriber cable pairs emanating from the exchange are also terminated on the MDF.

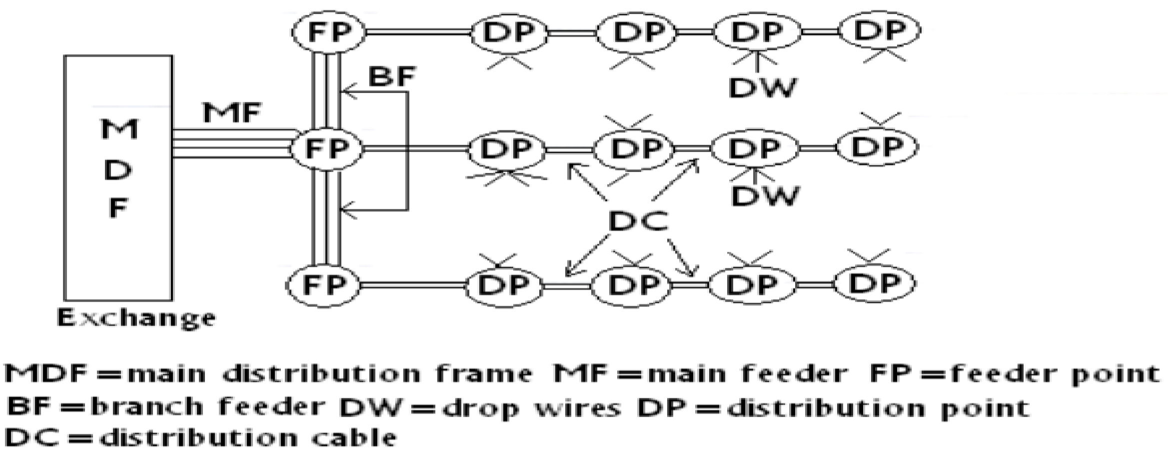


FIG – Cable Hierarchy For Subscriber Loops.

Q2. List the basic functions of a switching system.

Answer

The switching office performs the following basic functions irrespective of the system whether it is a manual or electromechanical or electronic switching system.

1. **Identity.** The local switching center must react to a calling signal from calling subscriber and must be able to receive information to identify the required destination terminal seize.

2. **Addressing.** The switching system must be able to identify the called subscriber from the input information

3. **Finding and path setup.** Once the calling subscriber destination is identified and the called subscriber is available, an accept signal is passed to the switching system and calling subscriber. Based on the availability, suitable path will be selected.

4. **Busy testing.** If number dialled by the calling subscriber is wrong or the called subscriber is busy (not attending the phone) or the terminal may be free (lifting the phone) but no response (not willing to talk or children handling), a switching system has to pass a corresponding voice message or busy tone after waiting for some time (status).

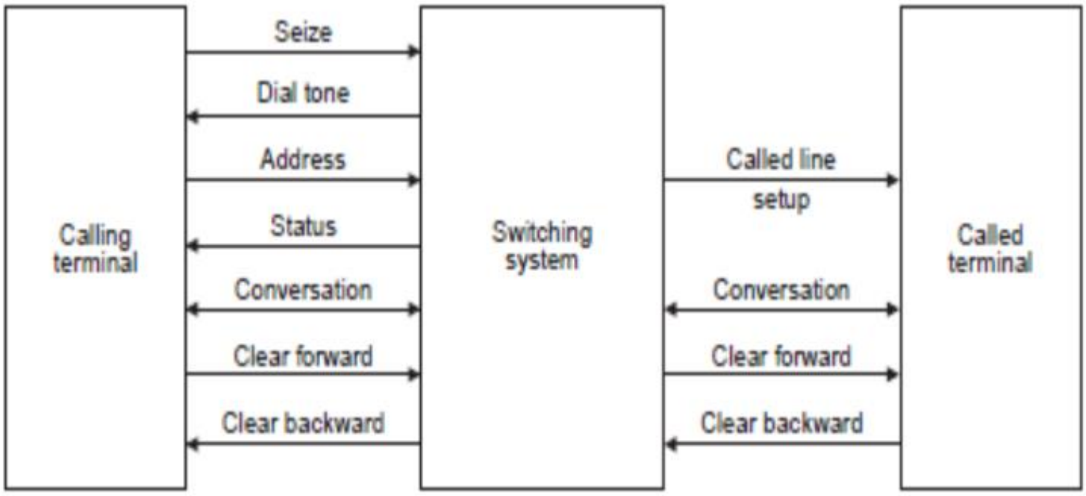
5. **Supervision.** Once the path is setup between calling and called subscriber, it should be supervised in order to detect answer and clear down conditions and recording billing information.

6. **Clear down.** When the established call is completed, the path setup should be disconnected. If the calling subscriber keeps the phone down first, the signal called clear forward is passed to the switching system. If the called subscriber keeps the phone down first, a signal called clear backward signal is passed to the switching system. By clear signal, the switching system must disconnect the path setup between calling and called subscriber.

7. **Billing.** A switching system should have a mechanism to meter to count the number of units made during the conversation. The cumulative number of units made for a particular duration by the calling subscriber is calculated. This information and if any should be sent to the called subscriber.

Q-3. Draw the signal exchange diagram for a local call used to represent the sequence of events between the subscriber and exchanges?

Answer



Q-4. What are the three basic steps involved in data communication through circuit switching?

Answer:

The steps are:

i) Circuit establishment (before data transfer)

ii) Circuit maintenance (When data transfer is going on)

iii) Circuit disconnect (When data transfer is over)

Q-5. Define calling rate and holding time.

Answer

Calling rate: This is the average number of requests for connection that are made per unit time. If the instant in time that a call request arises is a random variable, the calling rate may be stated as the probability that a call request will occur in a certain short interval of time.

If '*n*' is the average number of calls to and from a terminal during a period T seconds, the calling rate is defined as

$$\lambda = \frac{n}{T}$$

Holding time: The average holding time or service time '*h*' is the average duration of occupancy of a traffic path by a call. For voice traffic, it is the average holding time per call in hours or 100 seconds and for data traffic, average transmission per message in seconds.

The reciprocal of the average holding time referred to as service rate (μ) in calls per hour is given as

$$\mu = \frac{1}{h}$$

Q-6. List the advantages and disadvantages of in band and out band voice signaling

Answer

In band signaling:

Advantages of In band signalling:

1. In band signalling can be used on any transmission medium.
2. The control signals can be sent to every part where a speech signal can reach.
3. Owing to the flexibility of operation, it is the most widely used signalling system for long distance telephone networks.
4. It is operations are simpler.

Disadvantages of In band signalling:

1. More possibility of speech signals imitating control signals. This problem can be reduced using suitable guard circuit.
2. The in band signals may 'spill-over' from one link to the another and causes error in that signalling system.

This limitation occurs when several transmissions links one connected end-to-end. The spill over problem can be eliminated by operating a line split to disconnect link whenever a signal is detected. The line split is designed generally to operate with in 35 ms.

Out band signalling:

Advantages:

1. The requirement of line splits are not necessary to avoid signal limitation.
2. Signals and speech can be transmitted simultaneously without disturbing the Conversation.
3. Simple and consequently cheap.

Disadvantages:

1. Very narrow bandwidth is available for signalling.
2. Filtering circuits are needed to handle the signalling bands.
3. More dependent on the transmission system.

Q-7. Explain Channel Associated mode, Channel Non-Associated mode and Quasi-Associated mode of common channel signalling networks.

Answer

In **associated CCS signalling mode**, there is a direct link between two exchanges. In this mode, the signalling path passes through the same set of switches as does the speech path.

Network topologies of the signalling network and the speech network are the same.

This mode of operation is simple, economic and easy to control.

This involves in delayed operation for long distance communication.

In non-associated CCS signalling, there are separate controls of the networks from the switching machines themselves. In multi exchange network, signal message passing through several intermediate nodes is referred as non-associated signalling.

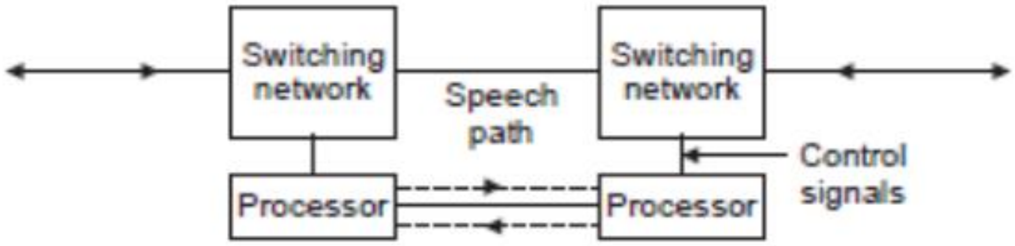
The network topologies for the signalling and the speech networks are different.

Between exchanges,many STP’s are placed.

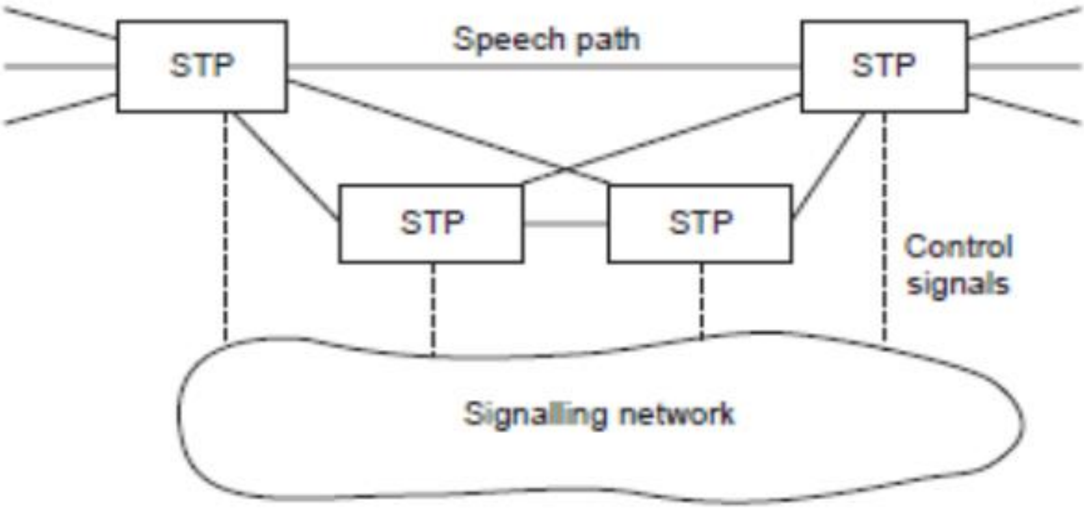
This approach is flexible as far as the routing is concerned. It demands more comprehensive scheme for message addressing than is needed for channel associated signalling.

In practice, CCS messages are routed through one intermediate node for short distance communication. This is known as quasi-associated signalling. It establishes simplified predetermined paths between exchanges.

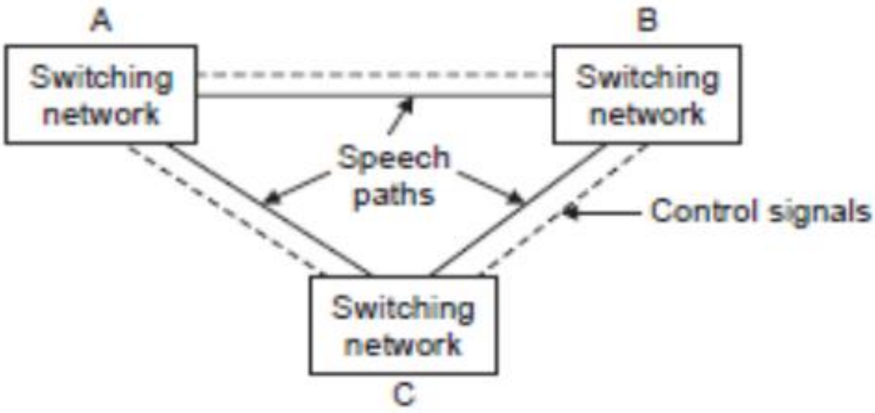
The signalling paths are not associated but are fixed for given speech connections.



(a) Associated signalling



(b) Non-associated signalling



(c) Quasi-associated signalling

Q-8. What are the various features of CCITT SIGNALLING SYSTEM 7 (SS7)?

Answer

1. Internationally standardized by the ITU.

2. SS7 is suitable for any transmission medium *i.e.*, can be operated over both terrestrial and satellite links.

3. Even though SS7 is optimized to work with digital SPC exchanges utilizing 64 kbps digital channels, it is suitable for operation over analog channels.

4. SS7 is suitable for various communication services such as telephony, text, data, images and video.

5. Transport mechanism is application independent.

Q-9. What is the need of a hybrid in telephone networks? How does it work?

Answer:

Digital exchanges require receive and transmit signals on separate two-wire circuits.
This calls for two-wire to four-wire conversion. Such a conversion is normally
required for trunk transmissions in analog exchanges. The circuit that performs 2-wire to 4-wire conversion is called Hybrid. A transformer based hybrid circuit is shown in Fig. The main function of a hybrid is to ensure that there is no coupling of signal from the input to the output in the 4-wire circuit. The operation of the circuit is as follows: The input signal is coupled to the B and F windings equally. Through the C winding, the input is coupled to the 2-wire circuit. The same signal when it flows through the balanced 2-wire couples the signal to winding D through winding C.
The signal induced in B flows through E and induces a current in D that opposes the current induced by F. If the impedance Z_B exactly matches that of the 2-wire circuit, the effect of input signal on the output winding D is completely nullified. In a similar way, the input signal from the subscriber end is completely nullified from coupling into the winding A.

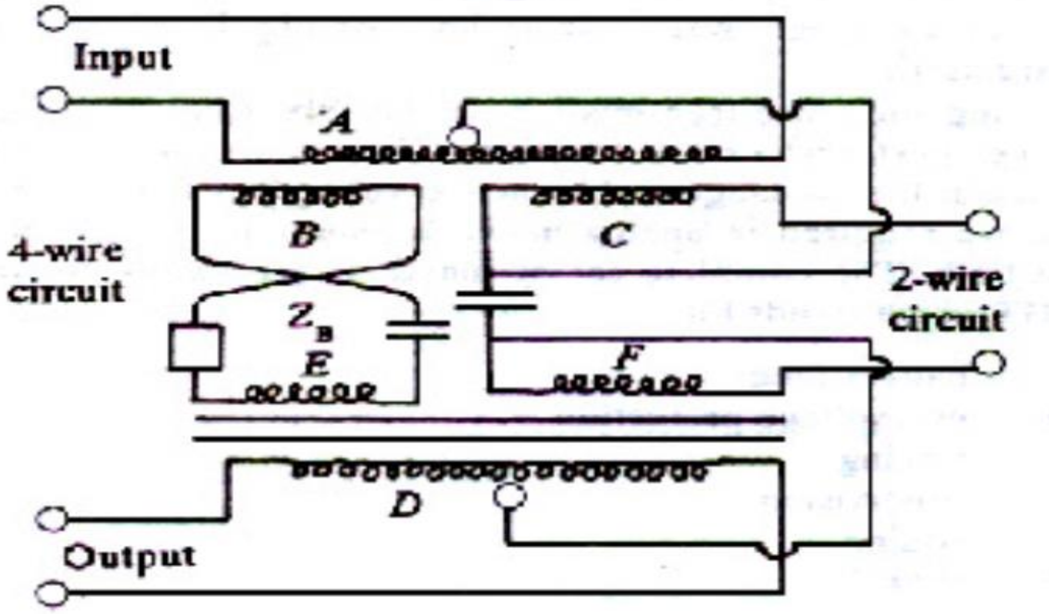


FIG – Two-wire to 4-wire Transformer Hybrid

Q-10. Write short notes on:

i. Stored Program Control.

ii. Congestion.

iii. Common channel signaling.

Answer.

(i) Stored Program Control:

In centralized control, all the control equipment is replaced by a single processor which

must be quite powerful. It must be capable of processing 10 to 100 calls per second, depending on the load on the system, and simultaneously performing many other ancillary tasks.

A typical control configuration of an ESS using centralized SPC is shown in Fig. A centralized SPC configuration may use more than one processor for redundancy purposes. In almost all the present day electronic switching systems using centralized control, only a two-processor configuration is used. A dual processor architecture may be configured to operate in one of three modes: (i) Standby mode (ii) Synchronous duplex mode (iii) Load Sharing mode

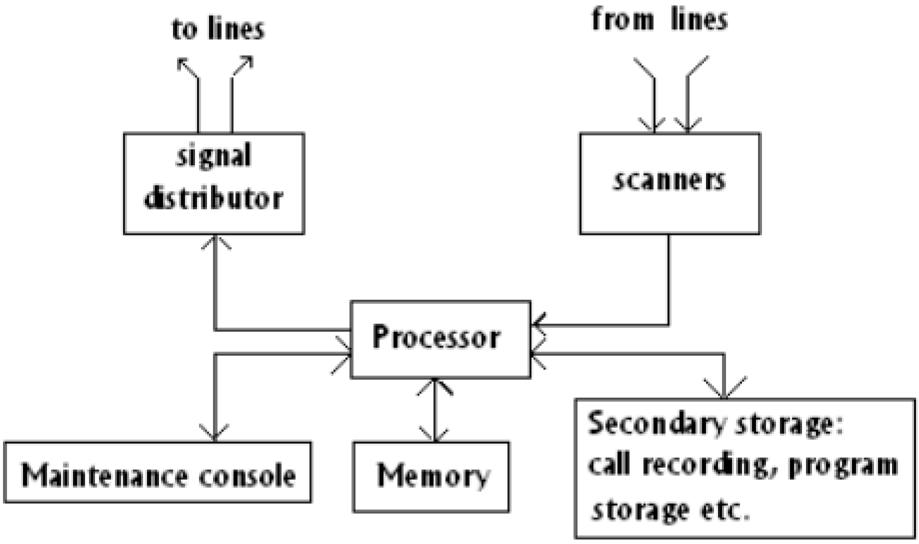


FIG - Typically Centralized SPC Organization

(ii) **CONGESTION:** It is uneconomic to provide sufficient equipment to carry all the traffic that could possibly be offered to a telecommunication system. In a telephone exchange it is theoretically possible for every subscriber to make a call simultaneously. A situation can therefore arise when all the trunks in a group of trunks are busy, and so it can accept further calls. This state is known as congestion. In a message-switched system, calls that arrive during congestion wait in a queue until an outgoing trunk becomes free. Thus, they are delayed but not lost. Such systems are therefore called queuing systems or delay system.

In a circuit-switched system, such as a telephone exchange, all attempts to make calls over a congested group of trunks are successful. Such systems are therefore called lost-call systems. In a lost-call system the result of congestion is that the traffic actually carried is less than the traffic offered to the system.

(iii) **Common channel signalling**. Signaling systems link the variety of switching systems, transmission systems and subscriber equipment's in telecommunication network to enable the network to function as a whole.

Three forms of signaling are involved in a telecommunication network:

1. Subscriber loops signaling.
2. Intra exchange or register signaling
3. Interexchange or inter register signaling

In a telephone network, subscriber loop signaling depends upon the type of a telephone instrument used. The intra exchange signaling is internal to the switching system and is heavily dependent upon the type and design of a switching system. It varies from one model to another even with the same manufacturer. This signaling does not involve signaling system of the type required on the switching network. When interexchange signaling takes place between exchanges with common control subsystems, it is called Inter register signaling.

The main purpose of Inter register signaling is the exchange of address digits which pass from exchange to exchange on a link by link basis. Network wide signaling also involves end to end signaling between the originating exchange and the terminating exchange. Such a form of signaling is called line signaling. CCS does not use the speech or the data path for signaling. It uses a separate common channel for passing control signals for a group of trunks or information paths.

It gives the following advantages:

(i) Information can be exchange between the processors much more rapidly than when channel associated signaling is used.

(ii) As a result, a much wider repertoire of signals can be used and this enables more services to be provided to customers.

(iii) Signals can be added or changed by software modification to provide new services.

(iv) There is no longer any need for line signaling equipments on every junction which results in a considerable cost saving.

(v) Since there is no line signaling, the junctions can be used for calls from B to A in addition to calls from A to B. Both way working requires fewer circuits to carry the traffic than if separate groups of junctions are provided from A to B and from B to A.

(vi) Signals relating to a call can be sent while the call is in progress.

This enables customers to alter connections after they have been set up.

For example a customer can transfer a call elsewhere, or request a third party to be connected in to an existing connection.

Q11. What are single stage and multistage networks? Compare the strengths and weaknesses of each.

OR

List the major difference in single stage, two stage and three stage Networks. Also discuss their blocking characteristics.

Answer:

Single Stage Vs. Multistage Network

Sr. No.	Single Stage	Multi Stage
1.	Inlet to outlet connection is through a single cross point.	Inlet to Outlet connection is through multiple cross points
2.	Use of single cross point per connection results in better quality link.	Use of multiple cross points may degrade the quality of a connection.
3.	Each individual cross point can be used for only one inlet/outlet pair connection.	Same cross point can be used establish connection between a number of inlet/outlet pairs.
4.	A specific cross point is needed for each specific connection.	A specific connection may be established by using sets of cross points.
5.	If a cross points fails, associated connection cannot be establish- There is no redundancy.	Alternative cross-points and paths are available.
6.	Cross points are inefficiently used. Only one cross point in each row or column of a square or triangular switch matrix is even in use , even if all the lines are active.	Cross points are used Efficiently
7.	Number of cross points is Prohibitive	Number of cross points is reduced significantly
8.	A large number of cross points in each inlet/outlet leads to capacitive loading.	There is no capacitive loading problem
9.	The network is non blocking in character	The network is blocking in character
10.	Time for establishing a call is less.	Time for establishing a call is more.

Q12. A three stage switching structure is to accomodate N = 128 input and 128 output terminals. For 16 first stage and 16 last stage, determine the number of cross points for nonblocking.

If the number of crosspoints in the example is to be reduced by the factor of 3 with nonblocking what is the probability that a call will be blocked? Assume the utilization probability p = 15%

Answer

Sol. The number of matrices at first and last stage is given by $\alpha = \frac{N}{n}$.

Hence $n = \frac{N}{\alpha} = \frac{128}{16} = 8$

To avoid blocking $k = 2n - 1 = 2 \times 8 - 1 = 15.$

Number of crosspoints is calculated by

$$N_x = k\left[2N + \left(\frac{N}{n}\right)^2\right] = 15\left[2 \times 128 + \left(\frac{128}{8}\right)^2\right]$$

N_x = 7680 cross points.

Number of cross points = 7680

Number of cross points reduced by factor 3 = $\frac{7680}{3}$ = 2560.

For the cross point 2560, the number of k matrices is calculated from

$$N_x = k\,(2N + (N/n)^2]$$

$$k = \frac{N}{[2N + (N/n)^2]} = \frac{2560}{256 + (128/8)^2}$$

$$k = 5$$

$$P = np/k = 8 \times 0.15/5 = 0.24$$

The probability that k links are busy is

$$B = [1 - (1 - P)^2]^k$$

$$B = [1 - (1 - 0.24)^2]^5 = 1.34\%$$

Q-13. Define grading in telecommunication switching networks.

Answer:

Grading: for a route switch it is not necessary for each incoming trunk to have access to every outgoing trunk. It is adequate if each incoming trunk has access to a sufficient number of trunks on each route to give the required grade of service. The technique of interconnecting the multiples of switches is called Grading.

Q-14. Design a strictly non blocking network for 1000 incoming and 1000 outgoing trunks. Also calculate the total cross points.

Answer

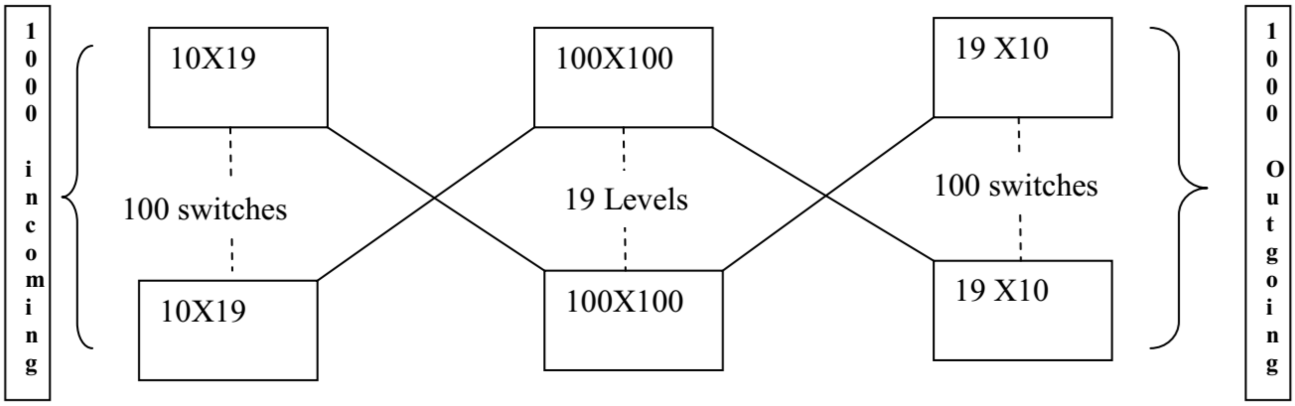
Choose a five stage network. The minimum number of cross point is obtained when

n= (2N) ^1/3 = 2000^1/3=12.599

Use n=10

The number of levels needed is 2n-1=19

No of switches in 1st stage = No of switches in 5th stage= 1000/10=100



The number of cross points is

= 100 X 10 X 19 + 19 X 5400 + 100 X 19 X10= 140600

Q-15. Mention the key advantages and disadvantages of circuit switching technique.

Answer:

Advantages:

i) After path is established, data communication without delay.

ii) Very suitable for continuous traffic.

iii) It establishes a dedicated path.

iv) No overhead after call setup.

v) It is transparent and data passes in order.

Disadvantages:

i) Provide initial delay for setting up the call.

ii) Inefficient for bursty traffic.

iii) Data rate should be same because of fixed bandwidth.

iv) When load increases, some calls may be blocked.

Q-16. Why data communication through circuit switching is not efficient?

Answer

In data communication, traffic between terminal and server are not continuous.

Sometimes more data may come or sometimes there is no data at all.

Circuit switching is not efficient because of its fixed bandwidth.

Q-17.What are the three functional blocks of a conventional time-slot Interchanger (i.e., a time switch), explain with neat diagram?

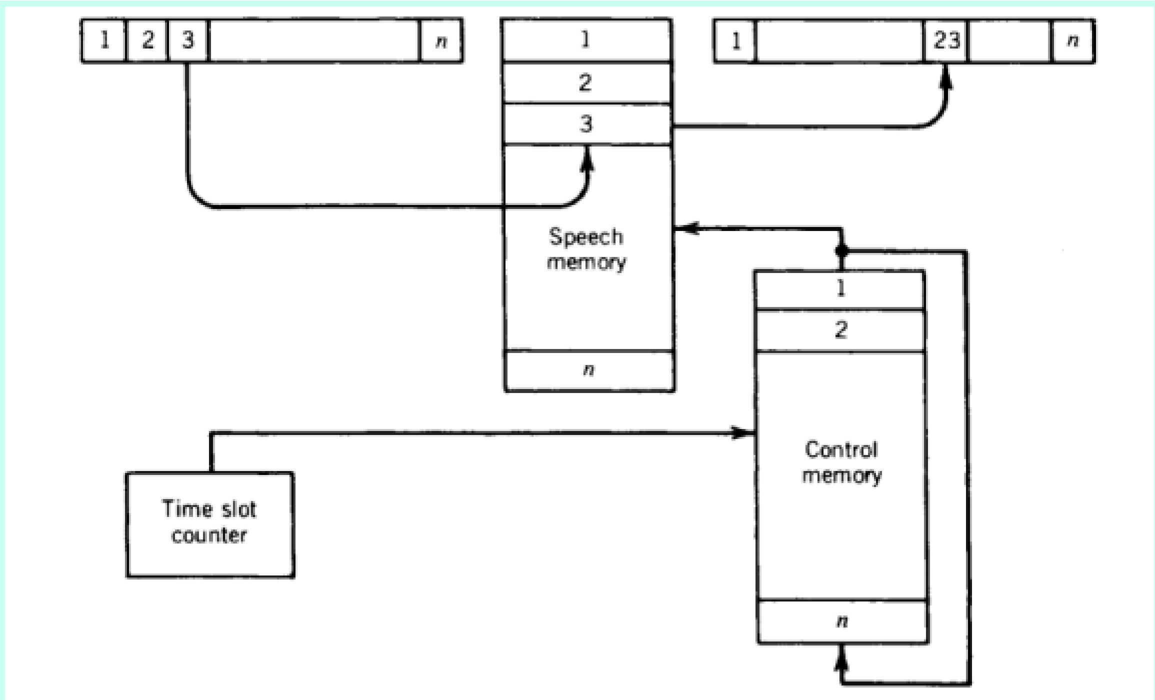
Answer

The three functional blocks are:

(i)- Speech memory

(ii) Control memory

(iii) Time slot counter



Q-18. Compare the performance of space-division single-stage switch with multi-stage switch.

Answer

Space-division single-stage switch requires more number of crosspoints, nonblocking in nature but provides no redundant path. On the other hand multi-stage switches require lesser number of crosspoints, blocking in nature but provides redundant path

Q-19 Draw and explain time division space switching in detail.

Answer:

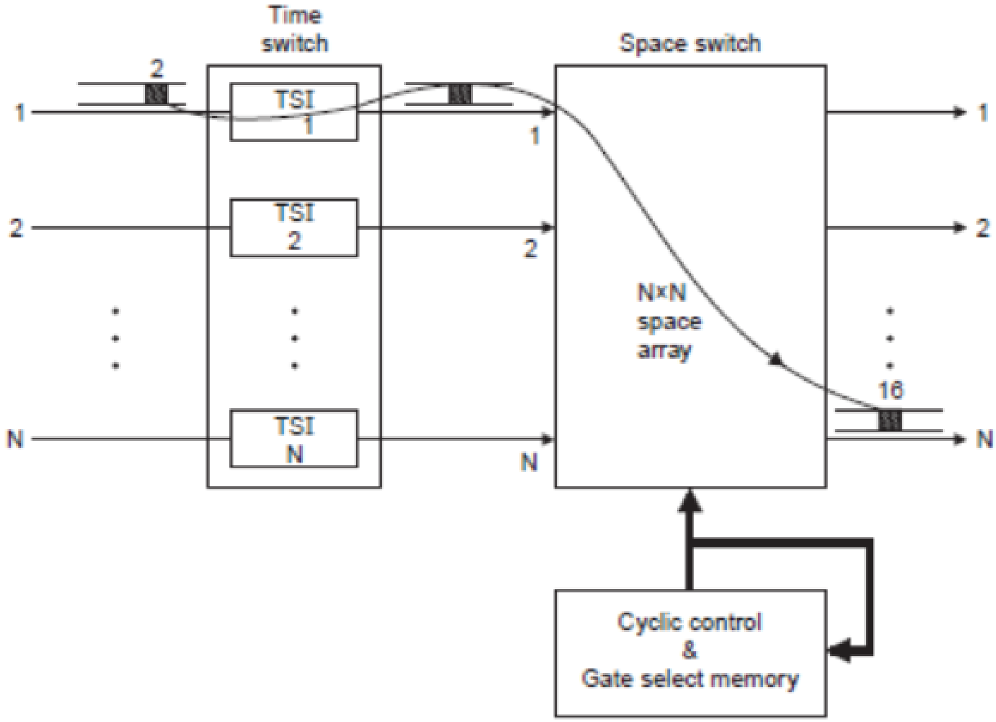
This switch consists of only two stages. This structure contains a time stage T followed by a space stage S as shown in Figure. Thus this structure is referred to as time-space (TS) switch.

The space arrays have N inlets and N outlets.

For each inlet line, a time slot interchanger with T slots is introduced. Each TSI is provided with a time slot memories. Similarly a gate select memory needs to be provided for the space array.

The transmission of signals carried out from sender to receiver through multiplexer input and de multiplexer output. The reverse communication also similar.

Thus a hybrid arrangement is needed to isolate the transmitted signal from the received signal. The basic function of the time switch is to delay information in arriving time slots until the desired output time slot occurs.



Q20- Determine the implementation complexity of 2048 channel TST switch with 16 TDM links and 128 channels. Let the time slot of space switch is 25.

Answer

$$IC = N^2 + \frac{NL \log_2 N + 2NT(8) + 2NL \log_2 T}{100}$$

Sol. Given $N = 16$

$T = 128$

$L = 25$

$$IC = 16^2 + \frac{16 \times 25 \times \log_2 16 + 2 \times 16 \times 128 \times 8 + 2 \times 16 \times 25 \times \log_2 128}{100}$$

IC = 656 cross points.

Q21- List any four important features of T-S-T (time space time) switching.

Answer

Some important features of TST switches are:

(*i*) **Low blocking probability.** An incoming channel time slot may be connected to an outgoing channel time slot using any possible space array time slot. Thus there are many alternative paths between two subscribers.

This concept reduces the blocking probability of a three stage combination switch.

(*ii*) **Stage independency.** The space stage operates in a time-divided fashion, independently of the external TDM links. The number of space stage time slots L does not coincide with the number of external TDM time slots T.

(*iii*) **Implementation advantage.** The factors to be considered for switching design and implementation are traffic loads, modularity, testability, expandability and simple control requirements. For large switches with heavy traffic loads, the TST have good implementation advantage.

(*iv*) **More cost effective.** If the input channel loading is high, the time expansion of TST and space expansion of STS are required.

Time expansion of TST can be achieved at less cost than space expansion of STS

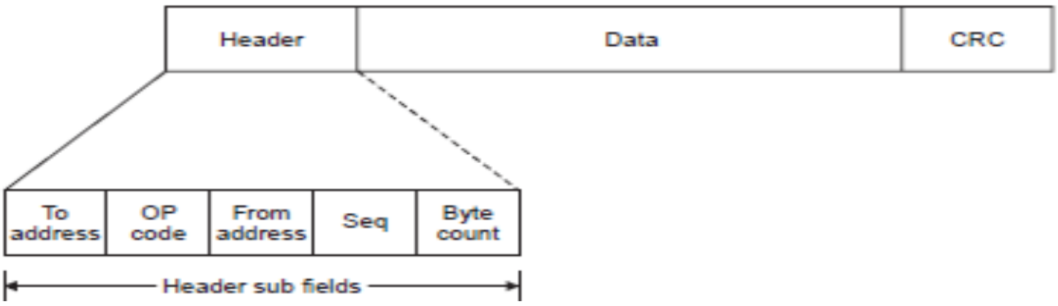
Q22- **What are some differences between circuit switching, datagram packet switching and virtual circuit packet switching?**

Answer

Circuit Switching	Datagram Packet	Virtual Circuit Packet
Dedicated path	No dedicated path	No dedicated path
Path established for entire conversation	Route established for each packet	Route established for entire conversation
Call set up delay	Packet transmission delay	Call set up delay, Packet transmission delay
Overload may block call set up	Overload increases packet delay	Overload may block call set up and increases packet delay
No speed or code conversion	Speed or code conversion	Speed or code conversio
Fixed bandwidth	Dynamic bandwidth	Dynamic bandwidth
No overhead bits after call set up	Overhead bits in each packet	Overhead bits in each packet

Q23. Draw the Frame format of typical packet switching and explain various fields?

Answer



A packet contains 3 major fields.

1. Header. It contains sub fields in addition to the necessary address fields. Other than the to and from address field, the following are the useful control information.

(*a*) Op code. It designates whether the packet is a message (text) packet or control packet.

(*b*) A sequence number (Seq) to reassemble messages at the destination node, detect faults and facilitates recovery procedures.

(*c*) Byte count. Used to indicate the length of a packet.

2. Data. A portion of a data stream to be transferred in the data field. Some packets may not contain a message field if they are being used strictly for control purposes.

3. CRC. The cyclic redundancy checks (CRC) field contains a set of parity bits that cover overlapping fields of message bits. The fields overlap in such a way that small numbers of errors are always detected. The probability of not detecting the occurrence of 2 large number of errors is 1 in 2M, where M is the number of bits in the cheek code.

Q24- Explain three types of ISDN channels. Tabulate the specifications of all the channels

Answer

ISDN consists of three types of communications channels. They are:

1. Bearer channel (B channel)
2. Delta channel (D channel), and
3. Hybrid channels (H channel).

These three ISDN channels are described below.

B channel B channels are logical digital ‘‘pipes’’ which exist on a single ISDN line.

B channel carry data and services at 64 kbps. It carries data in full duplex mode.

Each B channels provide a 64 kbps clear channel, clear meaning that the entire bandwidth is available for data, B channels typically form circuit switched connections. B channel connection is an end-to-end physical circuit that is temporarily dedicated to transferring data between two devices. The circuit switched nature of B channel connections; combined with their reliability and relatively high bandwidth makes ISDN suitable for a range of applications including voice, video, fax and data.

B channels are normally used for on-demand connection.

As B channel operation based on circuit switching, it can be configured as semi permanent or ‘‘nailed up’’ connections.

D channel D channel can be either 16 or 64 kbps, depending on the needs of the user.

The primary function of the D channel is to carry control signalling and administrative information for B channels to set up and tear down the calls. The D channel uses packet switched connection. The packet switched connection are best adapted to the intermittent but latency sensitive nature of signalling traffic, accounting for the highly reduced call setup time of 1 to 2 seconds on ISDN calls. Unlike the B-channel, which can function as a simple ‘pipe’, the D channel is associated with higher level protocols at layers 2 and 3 of OSI model which form the packet switched connections.

The D channel provides the signalling information that is required for caller identification. It also includes low-rate data transfer and applications such as telemetry and alarm transmission.

H channels H channels are suitable for high data rate applications such as video, teleconferencing and so on.

Table gives ISDN channel and its specifications.

Channel	Bit rate (kbps)	Interface	Purpose
B	64	BRI	Bearer services
H0	384	PRI	6 B channels
H11	1536	PRI	24 B channels
H12	1920	PRI	30 B channels
D	16	BRI	Administrative and control signalling
D	64	PRI	,,

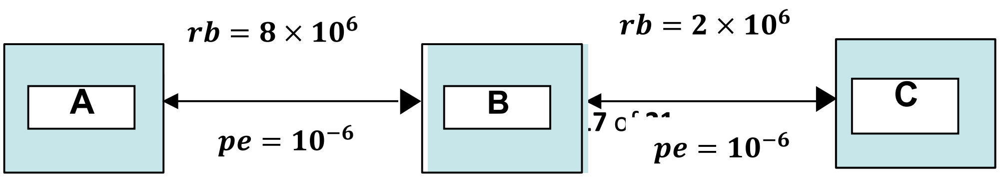
Q25. A file of size 2 Mbits is to be transmitted over two links in packet switching network as shown in figure below. If the link speed between A and B is 8Mbps and between B and C is 2Mbps, and the probability of bit errors in both links is 10^{-6}.

(a)- How many bits need to be transmitted to deliver file correctly if the file is sent all at once.

(b)- How many bits need to be transmitted to deliver file correctly if the file is sent as packets of size 500 Kbits.

(c)- Comment on the results of (a) and (b).

(d)- Compare the transmission delay of the above two cases in (a) and (b).



Answer

a) (3 points)

Packet has $2 \times 10^6\ bits$

$P_{correct} = (1 - P_e)^{2\times10^6}$= 0.135

Number of bits to be transmitted to deliver file correctly = $\frac{1}{0.135} \times 2 \times 2 \times 10^6 = 29.55$ Mbits

b) (2 points)

$P_{correct} = (1 - P_e)^{500000} = 0.666$

Number of bits to be transmitted to deliver file correctly = $\frac{1}{0.666} \times 2 \times 2 \times 10^6$

= 6.59 Mbits

C) (2 points)

This shows the importance of Packetization , the smaller the packet size, the less number of bits needs to be transmitted to get file send correctly.

c) (3 points)

$t_{transmission} = \frac{packet\ lenght}{rb} \times number\ of\ bops$

For case (a) $t_{transmission} = \frac{2\times10^6}{\mathbf{8\times10^6}} + \frac{2\times10^6}{\mathbf{2\times10^6}} = 1.25$ sec

For case (b) $t_{transmission} = \frac{500\times10^3}{\mathbf{8\times10^6}} + \frac{500\times10^3}{\mathbf{2\times10^6}} = 0.3125$ sec

Q26. What are the three basic steps involved in data communication through circuit switching?

Answer

The steps are:

i) Circuit establishment (before data transfer)

ii) Circuit maintenance (When data transfer is going on)

iii) Circuit disconnect (When data transfer is over)

Q27. Explain the difference between the basic rate and the primary rate in ISDN, and what is the best application for each one of them?

Answer

The difference is

- Basic rate of ISDN=2B+D = 2x(64 kbps)+ 16 kbps= 144 kbps

  It is suitable for home use.

- Primary Rate = 30B +D= 30 x 64kbps + 64kbps or 23B+D,

  It is suitable for big businesses use .

Q28. How ATM technology supports real time communication?

Answer

By considering fixed small packet size and high speed switch, it yields small packet delay and if any packet is lost, it will not affect much the quality of voice

Q29. Explain what is DTMF signalling.

Answer:

Dual Tone Multi Frequency (DTMF) was first introduced in 1963 with 10 buttons in Western Electric 1500 –type telephones. DTMF was originally called TouchTone. DTMF is a more efficient means than dial pulsing for transferring telephone numbers from a subscriber's location to the central office switching machine.

DTMF is a simple two-of –eight encoding scheme where each digit is represented by the linear addition of two frequencies. DTMF is strictly for signaling between a subscriber's location and the nearest telephone office or message switching center.

DTMF is sometimes confused with another two-tone signaling system called multi frequency signaling (Mf), which is a two-of-six code designed to be used only to convey information between two electronic switching machines.

Fig. shows the four-row-by-four column keypad matrix used with a DTMF keypad. As the figure shows, the keypad is comprised of 16 keys and eight frequencies.

Most household telephones, however, are not equipped with the special-purpose keys located in the fourth column (i.e., the A, B, C, and D keys). Therefore, most household telephones actually use two-of-seven tone encoding scheme. The four vertical frequencies (called the low group frequencies) are 697 Hz, 852 Hz, and 941 Hz, and the four horizontal frequencies (called the high group frequencies) are 1209 Hz, 1336 Hz,1477 Hz, and 1633 Hz. The frequency tolerance of the oscillators is + .5%. As shown in Figure, the digits 2 through 9 can also be used to represent 24 of the 26 letters (Q and Z are omitted). The letters were originally used to identify one local telephone exchange from another.

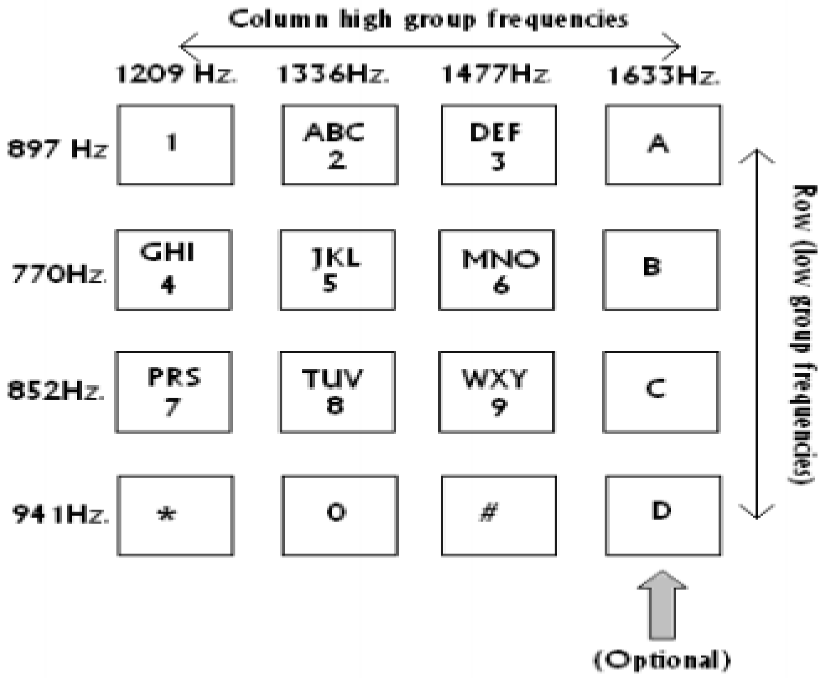


FIG - DTMF Keypad Layout.

Q30- Write short notes on Telephone hand set and its working.
Answer

A standard telephone set is comprised of a transmitter, a receiver, and electrical network for equalization, associated circuitry to control side tone levels and to regulate signal power, and necessary signalling circuitry.

In essence, a telephone set is an apparatus that creates an exact likeness of sound waves with an electric current. Fig 8-4 shows the functional block diagram of a telephone set. The essential components of a telephone set are the ringer circuit, on/off hook circuit, equalizer circuit, hybrid circuit, speaker, microphone, and a dialing circuit.

Ringer Circuit: The purpose of the ringer is to alert the destination party of incoming calls. The audible tone from the ringer must be load enough to be heard from a reasonable distance and offensive enough to make a person want to answer the telephone as soon as possible. In modem telephones, the bell has been placed with an electronic oscillator connected to the speaker..

On/off hook circuit: The on/off hook circuit (some times called a switch hook) is nothing more than a simple single-throw, double-pole (STDP) switch placed across the tip and ring. The switch is mechanically connected to the telephone handset so that when the telephone is idle (on hook), the switch is open. When the telephone is in use (off hook), the switch is closed completing and electrical path through the microphone between the tip and ring of the local loop.

Equalizer circuit: Equalizers are combination of passive components (resistors, capacitors and so on) that are used to regulate the amplitude and frequency response of the voice signals.

Speaker: In essence, the speaker is the receiver for the telephone. The speaker converts electrical signals received from the local loop to acoustical signals(sound waves) that can be heard and understood by a human being. The speaker is connected to the local loop through the hybrid network. The speaker is typically enclosed in the handset of the telephone along with the microphone.

Microphone: For all practical purposes, the microphone is the transmitter for the telephone. The microphone converts acoustical signals in the form of sound pressure waves from the caller to electrical signals that are transmitted into telephone net-work through the hybrid network. Both the microphone and the speaker are transducers, as they convert one form of energy into another form of energy. A microphone converts acoustical energy first to mechanical energy and then to electrical energy.

Hybrid network : The hybrid network (sometimes called a hybrid coil or duplex coil) in a telephone set is a special balanced transformer used to convert a two-wire circuit(the local loop) in to a four wire circuit(the telephone set) and the vice-versa, thus enabling full duplex operation over a two wire circuit. In essence, the hybrid network separates the transmitted signals from the received signals. Outgoing voice signals are typically in the 1-V to 2-V range, while incoming voice signals are typically half that value. Another function of the hybrid network is to allow a small portion of the transmit signal to be returned to the receiver in the form of a sidetone.

In sufficient sidetone causes the speaker to raise his voice, making the telephone conversation seem unnatural. Too much sidetone causes the speaker to talk too softly, thereby reducing the volume that the listener receives.

Dialing Circuit: The dialing circuit enables the subscriber to output signals representing digits, and this enables the caller to enter the destination telephone number. The dialing circuit could be a rotary dialer, which is nothing more than a switch connected to a mechanical rotating mechanism that controls the number and duration of the on/off condition of a switch. However, more than likely, the dialing circuit is either an electronic dial-pulsing circuit or a Touch-Tone keypad, which sends various combinations of tones representing the called digits.

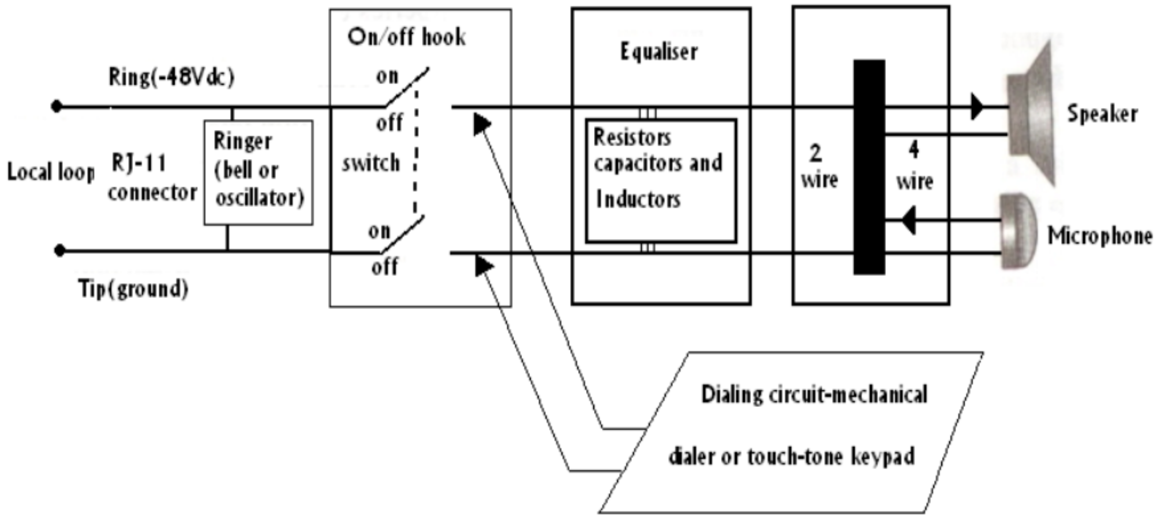


FIG – Functional Block Diagram Of a Standard Telephone Set.