[CHAPTER-IV]

------🙞------ॐ------🙜----🕮---- TELECOMMUNICATION SYSTEM----🕮-----🙜-----ॐ------🙞-----

4.1 The operation of Electronic Telephone System. (Telephone Set)

The Basic Telephone Set Fundamental Functions

- The basic telephone set connected to the telephone network we are all very comfortable with using, has 4 basic functions:
- To provide a signal to the telephone company that a call is to be made (off-hook) or a call is complete (on-hook).
- To provide the telephone company with the number the caller wishes to call.
- To provide a way for the telephone company to indicate that a call is coming in or ringing.
- To convert voice frequencies to electrical signals that can be transmitted at the transmitter and convert those electrical signals back to voice frequencies at the receiver.

The Federal Communications Commission (FCC) has set standards for the above features and all manufacturers selling telephones in this country must match these standards or the phone will not work properly.

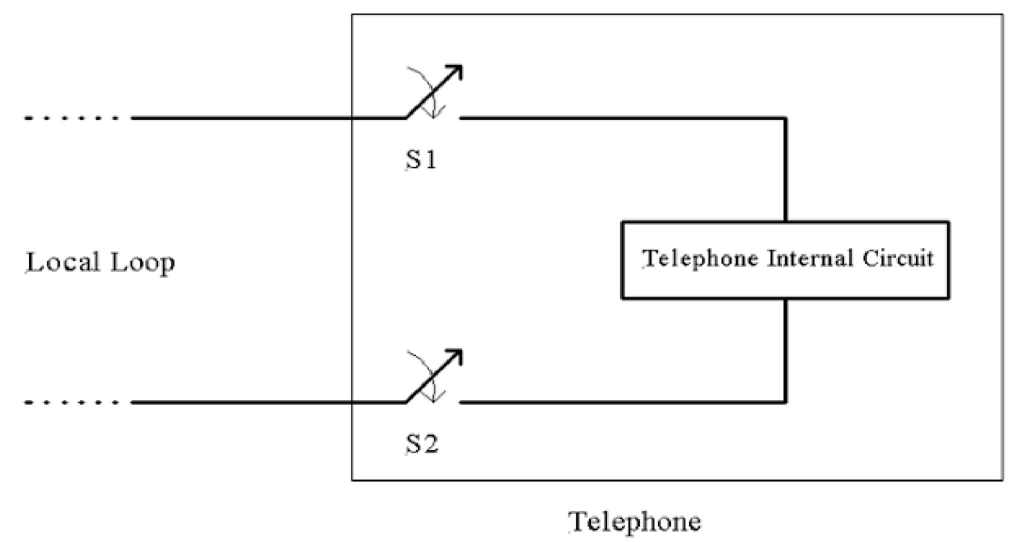
In addition many modern telephones also come with features like speed dial, redial, memory, caller ID, voice mail, etc. These are all additional features that are not necessary to make or receive calls.

Let's look at Telephone Set Function 1: To provide a signal to the telephone company that a call is to be made (off-hook) or a call is complete (on-hook).

The switchhook gets its name from the old telephones that had a hook on the side. On modern phones the switchhook is a button that is depressed when the handset is put on the cradle of the telephone.

According to Telephone Company specifications individual telephone set DC resistance should be 200 Ω but in reality most telephones range between 150 and 1000 Ω of DC resistance.

When a user picks up a connected telephone handset to make a call the switch hooks in the figure below (S_1 and S_2) close (off-hook condition) and the local loop circuit is complete.



When a handset is picked up, a DC current ranging between 20 and 120 mA flows on the pair of wires connecting the telephone to the CO. This current flow causes a relay coil to magnetize and it's contacts close.

In the CO current flows through a relay coil attached to the local loop wire pair. The coil energizes, it's contacts close and the CO switch knows a phone is off hook somewhere.

A line feeder in the CO switch looks for the off-hook signal, finds it and sets up a connection. In the CO switch a dial-tone generator is connected to the line so the caller knows they can dial a number.

4.2 The function of Switching System & Call Procedures

Switching system

When there are many devices, it is necessary to develop suitable mechanism for communication between any two devices. One alternative is to establish point-to-point communication between each pair of devices using **mesh topology.**

However, mesh topology is impractical for large number of devices, because the number of links increases exponentially (n(n-1)/2, where n is the number of devices) with the number of devices. A better alternative is to use switching techniques leading to switched communication network.

In the **switched network** methodology, the network consists of a set of interconnected nodes, among which information is transmitted from source to destination via different routes, which is controlled by the switching mechanism. A basic model of a switched communication is shown in Fig. 4.1.1.

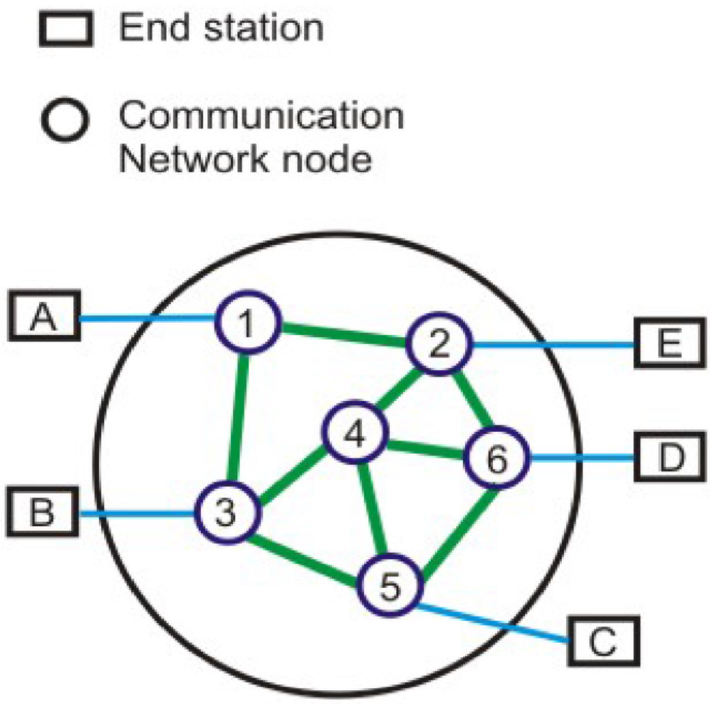
The end devices that wish to communicate with each other are called *stations*. The switching devices are called *nodes*. Some nodes connect to other nodes and some are to connected to some stations.

Key features of a switched communication network are given below:

- Network Topology is not regular.
- Uses FDM or TDM for node-to-node communication.
- There exist multiple paths between a source-destination pair for better network reliability.
- The switching nodes are not concerned with the contents of data.
- Their purpose is to provide a switching facility that will move data from node to node until they reach the destination.

The switching performed by different nodes can be categorized into the following three types:

- Circuit Switching
- Packet Switching
- Message Switching



Basic Call Procedure:

Fig. Shows a simplification diagram illustrating how two telephone sets (subscribers) are interconnected through a central office dial switch. Each subscriber is connected to the switch through a local loop. The switch is most likely some sort of an electronic switching system (ESS machine). The local loop are terminated at the calling and called station s in telephone sets and at the central office ends to switching machines.

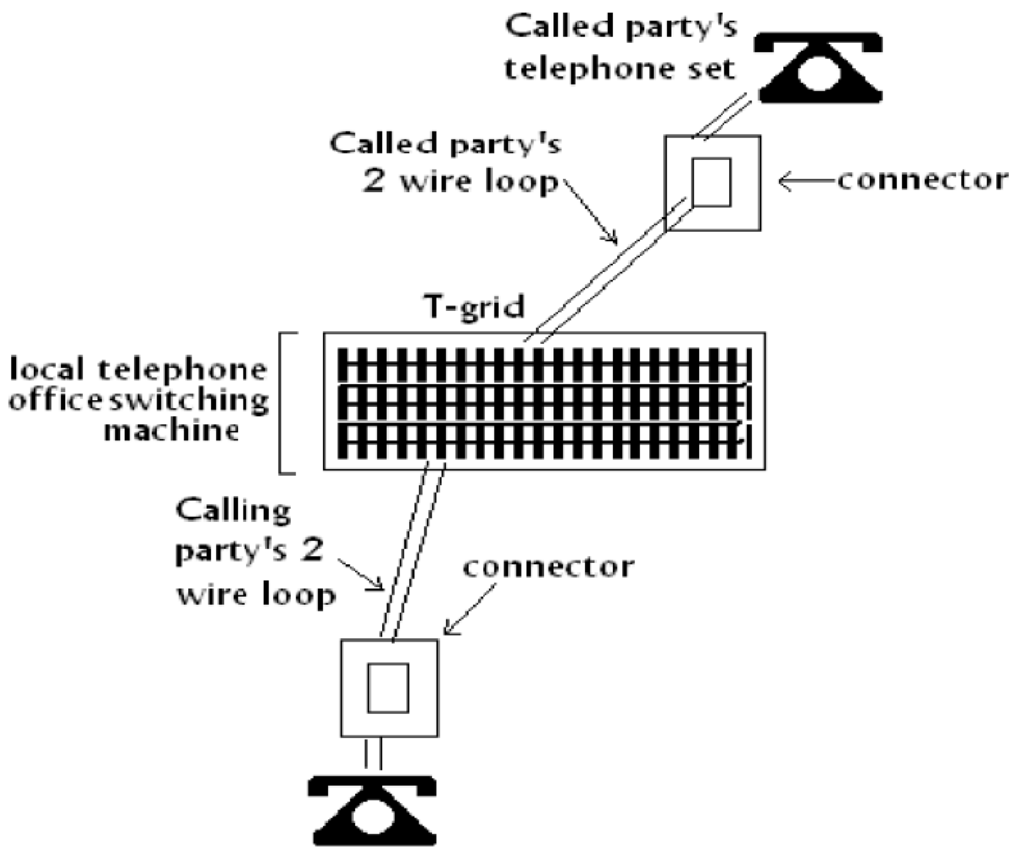


FIG - Telephone Call Procedure

When the calling party's telephone set goes off hook (i.e., lifting the handset off the cradle), the switch hook in the telephone set is released, completing a dc path between the tip and the ring of the loop trough the microphone .

The ESS machine senses a dc current in the loop and recognizes this as an off-hook condition. Completing a local telephone call between two subscribers connected to the same telephone switch is accomplished through a standard set of procedure that includes the 10 steps listed next.

1. Calling station goes off hook.

2. After detecting a dc current flow on the loop, the switching machine returns an audible dial tone to the calling station, acknowledging that the caller has access to the switching machine.

3. The caller dials the destination telephone number using one of the two methods: Mechanical dial pulsing or, more likely, electronic dual-tone multi frequency (Touch-Tone) signals.

4. When the switching machine detects the first dialled number, it removes the dial tone from the loop.

5. The switch interprets the telephone number and then locates the loop for the destination telephone number.

6. Before ringing the destination telephone , the switching machine tests the destination loop for dc current to see if tt is idle (on hook) or in use (off hook). At the same time, the switching machine locates a signal path through the switch between the two local loops.

7. (a) If the destination telephone is off hook, the switching machine sends a station busy signal back to the calling station.

(b) If the destination telephone is on hook, the switching machine sends a ringing signal to the destination telephone on the local loop and the same time sends a ring back signal to the calling station to give the caller some assurance that something is happening.

8. When the destination answers the telephone, it completes the loop, causing dc current to flow.

9. The switch recognizes the dc current as the station answering the telephone. At this time, the switch removes the ringing and ring-back signals and completes the path through the switch, allowing the calling and called parties to begin conversation.

10. When either end goes on hook, the switching machine detects an open circuit on that loop and then drops the connections through the switch.

4.3 The principle of space and time switching.

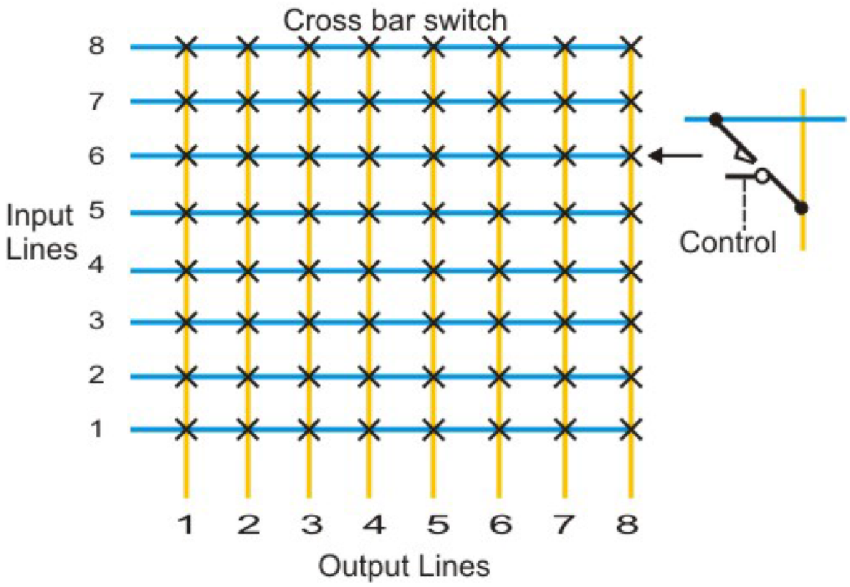
Space Switching

Circuit switching uses any of the three technologies: **Space-Division** switches, **Time-Division** switches or a **Combination of both**.

In Space-division switching, the paths in the circuit are separated with each other spatially, i.e. different ongoing connections, at a same instant of time, uses different switching paths, which are separated spatially.

This was originally developed for the analog environment, and has been carried over to the digital domain. Some of the space switches are crossbar switches, Multi-stage switches (e.g. Omega Switches).

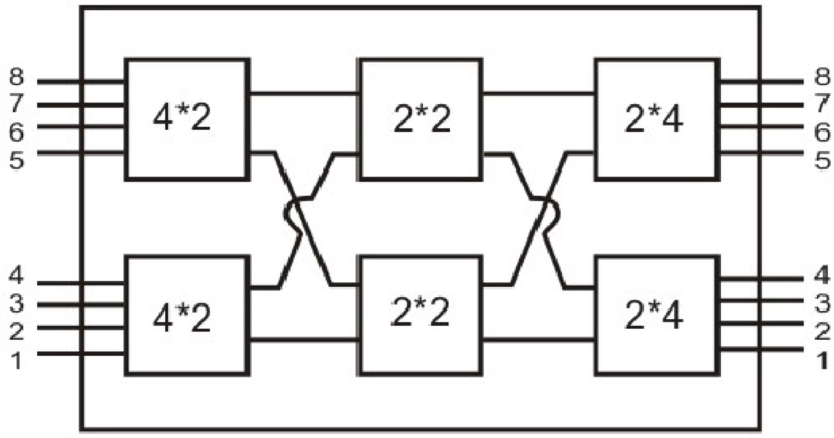
A **Crossbar** switch is shown in Fig. Basic building block of the switch is a metallic crosspoint or semiconductor gate that can be enabled or disabled by a control unit.



[Figure Schematic diagram of a crossbar switch]

Limitations of crossbar switches are as follows:

• The number of cross points grows with the square of the number of attached stations.
• Costly for a large switch.
• The failure of a cross point prevents connection between the two devices whose lines intersect at that cross point.
• The cross points are inefficiently utilized.
• Only a small fraction of cross points are engaged even if all of the attached devices are active. Some of the above problems can be overcome with the help of *multistage space division* switches.
• By splitting the crossbar switch into smaller units and interconnecting them, it is possible to build multistage switches with fewer cross points.



[Fig- A three-stage space division switch]

Figure shows a three-stage space division switch. In this case the number of crosspoints needed goes down from 64 to 40. There is more than one path through the network to connect two endpoints, thereby increasing reliability. Multistage switches may lead to *blocking*.

The problem may be tackled by increasing the number or size of the intermediate switches, which also increases the cost. The blocking feature is illustrated in Fig. 4.1.6. As shown in Fig. 4.1.6, after setting up connections for 1-to-3 and 2-to-4, the switch cannot establish connections for 3-to-6 and 4-to-5.

Time Division Switching

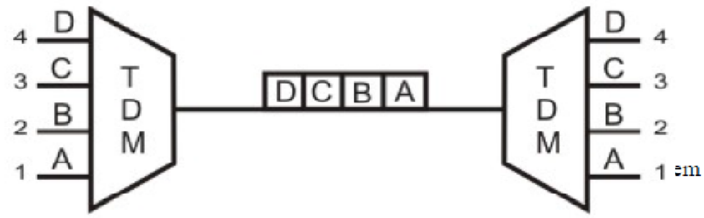
Both voice and data can be transmitted using digital signals through the same switches. All modern circuit switches use digital time-division multiplexing (TDM) technique for establishing and maintaining circuits. Synchronous TDM allows multiple low-speed bit streams to share a high-speed line.

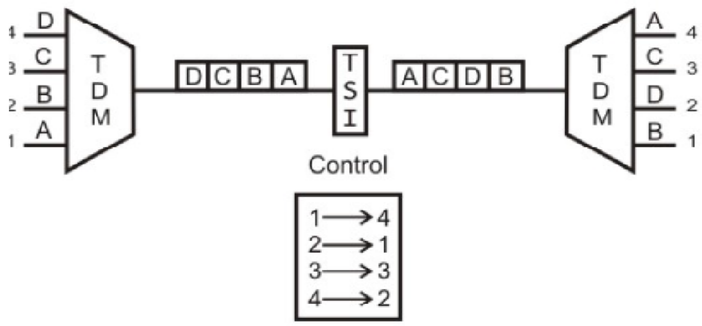
A set of inputs is sampled in a round robin manner. The samples are organized serially into slots (channels) to form a recurring frame of slots.

During successive time slots, different I/O pairings are enabled, allowing a number of connections to be carried over the shared bus. To keep up with the input lines, the data rate on the bus must be high enough so that the slots recur sufficiently frequently.

For 100 full-duplex lines at 19.200 Kbps, the data rate on the bus must be greater than 1.92 Mbps. The source-destination pairs corresponding to all active connections are stored in the control memory.

Thus the slots need not specify the source and destination addresses. Schematic diagram of time division switching is shown in Fig.





Time-division switching uses time-division multiplexing to achieve switching, i.e. different ongoing connections can use same switching path but at different interleaved time intervals.

There are two popular methods of time-division switching namely, Time-Slot Interchange (TSI) and the TDM bus.

TSI changes the ordering of the slots based on desired connection and it has a random-access memory to store data and flip the time slots as shown in Fig. 4.1.8.

The operation of a TSI is depicted in Fig.1 As shown in the figure, writing can be performed in the memory sequentially, but data is read selectively.

In TDM bus there are several input and outputs connected to a high-speed bus. During a time slot only one particular output switch is closed, so only one connection at a particular instant of time as shown in Fig.2

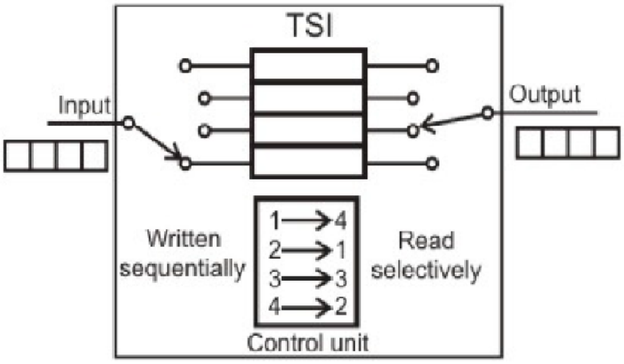


Figure 1 Operation of a **TSI**

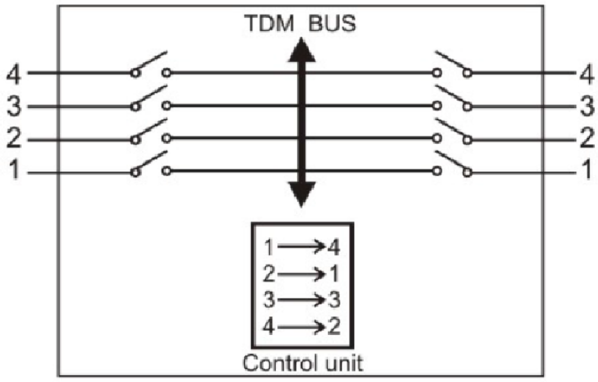


Figure 2 **TDM** bus switching

4.4 The principle of PDH and SDH modes of transmission.

The Plesiochronous Digital Hierarchy (PDH) In a PDH network you have different levels of Multiplexers.

Figure 1 shows three levels of multiplexing:-

· 2Mbit/s to 8Mbit/s

· 8Mbit/s to 34Mbit/s

· 34Mbit/s to 140Mbit/s

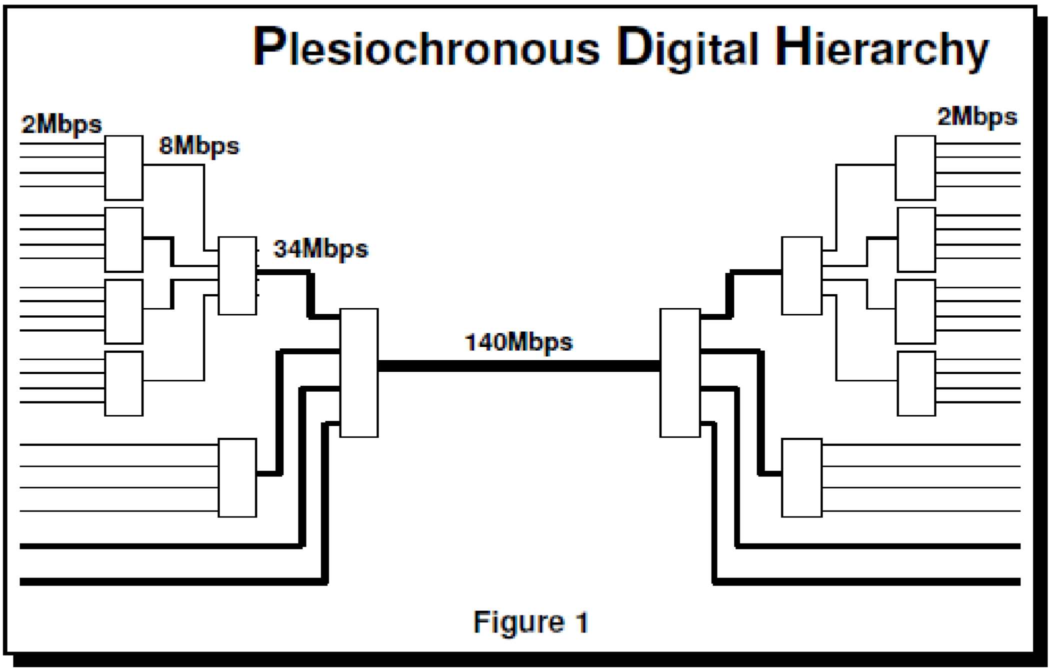


Figure 1

So to carry a 2Mbit/s data stream across the 140Mbit/s trunk requires it to be multiplexed up through the higher order multiplexers into the 140Mbit/s trunk and then to be multiplexed down through the lower order multiplexers.

Because Plesiochronous is not quite Synchronous each of the multiplexers need a little bit of overhead on their high speed trunks to cater for the slight differences in data rates of the streams on the low speed ports.

Some of the data from low speed ports (that are running too fast) can be carried in the trunk overhead, and this can happen at all multiplexing levels. This is known as Justification or Bit Stuffing.

PDH Multiplexing Hierarchy Figure 2 shows that there are two totally different hierarchies, one for the US and Japan and another for the rest of the world.

PDH Multiplexing Levels

Multiplexing Level	United States & Japan Name	# calls	Rate (Mbps)	Europe & Australia Name	# calls	Rate (Mbps)
1	DS1	24	1.544	CEPT1	30	2.048
2	DS2	96	6.312	CEPT2	120	8.448
3	DS3	672	44.736	CEPT3	480	34.368
4	DS4	4032	274.176	CEPT4	1920	139.264

Figure 2

The other thing to notice is that the different multiplexing levels are not multiples of each other. For example CEPT2 supports 120 Calls but it requires more than 4 times the bandwidth of CEPT1 to achieve this.

This is because PDH is not exactly synchronous and each multiplexing level requires extra bandwidth to perform Bit Stuffing. So the Plesiochronous Hierarchy requires “Bit Stuffing”, at all levels, to cater for the differences in clocks.

This makes it particularly difficult to locate a particular 2Mbit/s stream in the 140Mbit/s trunk unless you fully de-multiplex the 140Mbit/s stream all the way down to 2Mbit/s. Drop & Insert a 2Mbit/s stream To drop & insert a 2Mbit/s stream from a 140Mbit/s trunk you need to break the 140Mbit/s trunk and insert a couple of “34Mbit/s to 140Mbit/s” multiplexers.

We can then isolate the appropriate 34Mbit/s stream and multiplex the other 34Mbit/s streams back into the 140Mbit/s trunk. Then you de-multiplex the 34Mbit/s stream, isolate the appropriate 8Mbit/s Stream and multiplex the other 8Mbit/s streams through the higher layer multiplexer, into the 140Mbit/s trunk.

The Limitations of PDH:-

- ***PDH is not very flexible***

As previously explained, it is not easy to identify individual channels in a higher order bit stream. You must multiplex the high rate channel down through all multiplexing levels to find a particular lower speed channel. This requires an expensive and complex “multiplexer mountain”.

- ***Lack of Performance***

It is not easy to provide good performance if you can’t monitor the performance in the first place. For PDH there is no international standard for performance monitoring and no agreed management channels.

There are some spare overhead bits that are being used for management but they have limited bandwidth and are hard to locate in a 140 Meg stream without de-multiplexing.

Lack of standards

Not only does PDH have two totally different multiplexing hierarchies but it is quite weak on standards. For example there are no standards for data rates above 140Mbit/s and no standards for the line side of a “Line Transmission Terminal”.

The Synchronous Digital Hierarchy (SDH)

SDH, like PDH is based on a hierarchy of continuously repeating, fixed length frames designed to carry isochronous traffic channels. SDH was specifically designed in such a way that it would preserve a smooth interworking with existing PDH networks.

The developers of SDH also addressed the weaknesses of PDH. They recognised that it was necessary to adopt not only a Synchronous frame structure but one that also preserves the byte boundaries in the various traffic bit streams.

Because SDH is synchronous it allows single stage multiplexing and de-multiplexing. This eliminates hardware complexity. You don’t need multiplexer mountains.

SDH Multiplexing levels

Figure 4 shows the SDH multiplexing levels.

The US and Japan use SONET while most of the rest of the world use SDH.

SDH Multiplexing Hierarchy

Data Rate (Mbps)	SONET (USA)	SDH (Europe)
51.84	STS-1, OC-1	*(not defined)*
155.52	STS-3, OC-3	STM-1
466.56	STS-9, OC-9	STM-3
622.08	STS-12, OC-12	STM-4
933.12	STS-18, OC-18	STM-6
1244.16	STS-24, OC-24	STM-8
1866.24	STS-36, OC-36	STM-12
2488.32	STS-48, OC-48	STM-16
9953.28	STS-192, OC-192	STM-64

STS = Synchronous Transport Signal

OC = Optical carrier

STM = Synchronous Transport Module

Figure 4

Apart from using some different terminology, thereis very little difference between SONET and SDH. You can see that the data rates are the same except SDH doesn’t specify a 51 Meg rate.

STM-1 forms the basis of the SDH frame structure. For example an STM-4 is a frame consisting of 4 x STM-1s. In Sonet, the STS levels refer to the speed of the bit stream. When these bits are converted to a train of optical pulses in a fibre, they are called an Optical Carrier (OC).

You may also see “OC-3c” referred to. This is simply the same bit rate as OC-3, but interpreted as one channel instead of 3 multiplexed OC-1s. The “c” stands for “Concatenated”.

SDH Network Elements

Figure 5 shows the elements that make up an SDH network.

□ ***Path Terminating Element***

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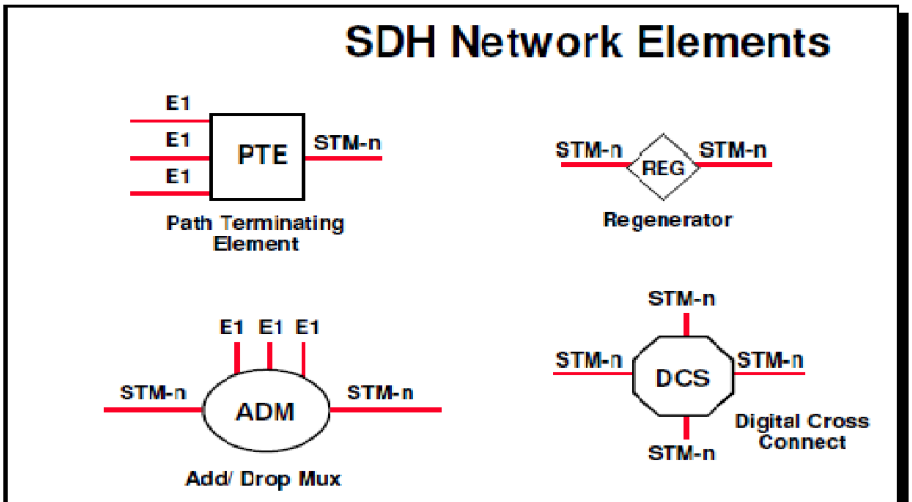


Figure 5

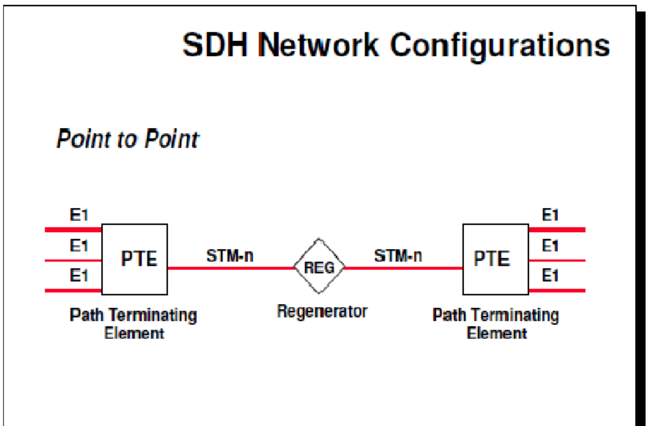


Figure 6

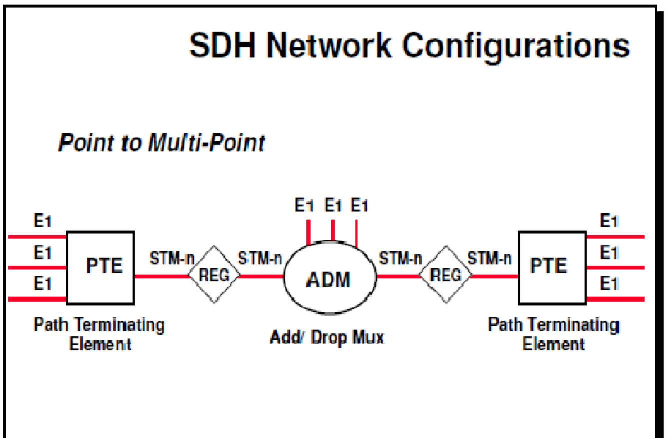


Figure 7

4.5 The operation of ATM , ISDN network.

ATM

Asynchronous transfer mode (ATM) is one of many network transmission protocols included in Windows Server 2003. The most commonly used transmission protocol included in Windows Server 20003 is TCP/IP, which is a connectionless protocol.

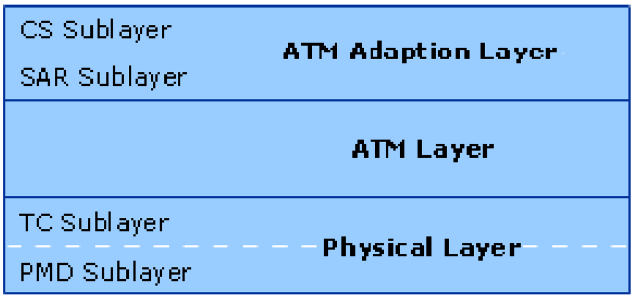
As such, TCP/IP cannot offer some of the advantages that a connection-oriented, virtual circuit, packet-switching technology, such as ATM, can. Unlike most connectionless networking protocols, ATM is a deterministic networking system — it provides predictable, guaranteed quality of service.

The ideal environment in which to use ATM is one that combines computer, voice, and video networking into a single network, and the combination of existing networks into a single infrastructure.

ATM Architecture

ATM is a combination of hardware and software that can provide either an end-to-end network or form a high-speed backbone. The structure of ATM and its software components comprise the ATM architecture, as the following illustration shows. The primary layers of ATM are the physical layer, the ATM layer, and the ATM Adaptation layer.

ATM Architectural Diagram



Each layer and sublayer is described briefly in the following table, "ATM Layers."

ATM LAYERS

Physical Layer

The physical layer provides for the transmission and reception of ATM cells across a physical medium between two ATM devices. This can be a transmission between an ATM endpoint and an ATM switch, or it can be between two ATM switches. The physical layer is subdivided into a Physical Medium Dependent sublayer and Transmission Convergence sublayer.

PMD Sublayer

The Physical Medium Dependent (PMD) sublayer is responsible for the transmission and reception of individual bits on a physical medium. These responsibilities encompass bit timing, signal encoding, interacting with the physical medium, and the cable or wire itself.

ATM does not rely on any specific bit rate, encoding scheme or medium and various specifications for ATM exist for coaxial cable, shielded and unshielded twisted pair wire, and optical fiber at speeds ranging from 64 kilobits per second to 9.6 gigabits per second.

In addition, the ATM physical medium can extend up to 60 kilometers or more by using single-mode fiber and long-reach lasers.

Thus it can readily support wide-range connectivity, including a private metropolitan area network. The independence of ATM from a particular set of hardware constraints has allowed it to be implemented over radio and satellite links.

Transmission Convergence Sublayer

The Transmission Convergence (TC) sublayer functions as a converter between the bit stream of ATM cells and the PMD sublayer. When transmitting, the TC sublayer maps ATM cells onto the format of the PDM sublayer, such as the DS-3 interface or Synchronous Optical Network (SONET) frames.

Because a continuous stream of bytes is required, unused portions of the ATM cell stream are filled by idle cells. These idle cells are identified in the ATM header and are silently discarded by the receiver. They are never passed to the ATM layer for processing.

The TC sublayer also generates and verifies the Header Error Control (HEC) field for each cell. On the transmitting side, it calculates the HEC and places it in the header. On the receiving side, the TC sublayer checks the HEC for verification.

If a single bit error can be corrected, the bit is corrected, and the results are passed to the ATM layer. If the error cannot be corrected (as in the case of a multibit error) the cell is silently discarded.

Finally, the TC sublayer delineates the ATM cells, marking where ATM cells begin and where they end. The boundaries of the ATM cells can be determined from the PMD layer formatting or from the incoming byte stream using the HEC field.

The PMD performs the HEC validation per byte on the preceding 4 bytes. If it finds a match, the next ATM cell boundary is 48 bytes away (corresponding to the ATM payload). The PMD performs this verification several times to ensure that the cell boundaries have been determined correctly.

ATM Layer

The ATM layer provides cell multiplexing, demultiplexing, and VPI/VCI routing functions. The ATM layer also supervises the cell flow to ensure that all connections remain within their negotiated cell throughput limits.

If connections operate outside their negotiated parameters, the ATM layer can take corrective action so the misbehaving connections do not affect connections that are obeying their negotiated connection contract. The ATM layer also maintains the cell sequence from any source.

The ATM layer multiplexes and demultiplexes and routes ATM cells, and ensures their sequence from end to end. However, if a cell is dropped by a switch due to congestion or corruption, it is not the responsibility of the ATM layer to correct the dropped cell by means of retransmission or to notify other layers of the dropped cell. Layers above the ATM layer must detect the lost cell and decide whether to correct it or disregard it.

In the case of interactive voice or video, a lost cell is typically disregarded because it takes too long to resend the cell and place it in the proper sequence to reconstruct the audio or video signal. A significant number of dropped cells in time-dependent services, such as voice or video, results in a choppy audio or video playback, but the ATM layer cannot correct the problem unless a higher Quality of Service is specified for the connection.

In the case of data (such as a file transfer), the upper layer application must sense the absence of the cell and retransmit it. A file with randomly missing 48-bytes chunks is a corrupted file that is unacceptable to the receiver. Because operations, such as file transfers, are not time dependent, the contents of the cell can be recovered by incurring a delay in the transmission of the file corresponding to the recovery of the lost cell.

ATM Layer Multiplexing and Demultiplexing

ATM layer multiplexing blends all the different input types so that the connection parameters of each input are preserved. This process is known as traffic shaping.

ATM layer demultiplexing takes each cell from the ATM cell stream and, based on the VPI/VCI, either routes it (for an ATM switch) or passes the cell to the ATM Adaptation Layer (AAL) process that corresponds to the cell (for an ATM endpoint).

ATM Adaptation Layer

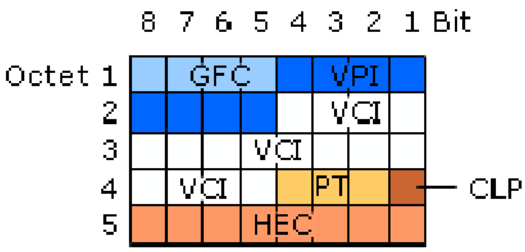
The ATM Adaptation Layer (AAL) creates and receives 48-byte payloads through the lower layers of ATM on behalf of different types of applications. Although there are five different types of AALs, each providing a distinct class of service, Windows Server 2003 supports only AAL5.

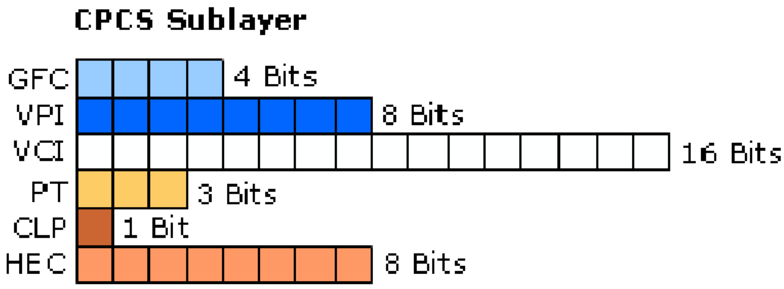
ATM Adaptation is necessary to link the cell-based technology at the ATM Layer to the bit-stream technology of digital devices (such as telephones and video cameras) and the packet-stream technology of modern data networks (such as frame relay, X.25 or LAN protocols such as TCP/IP or Ethernet).

ATM Cell Structure

At either a private or a public User-Network Interface (UNI), an ATM cell always consists of a 5-byte header followed by a 48-byte payload. The header is composed of six elements, each detailed in the following figure, "Cell Header Structure."

Cell Header Structure





ISDN

Integrated Service Digital Network, or ISDN, is the original high-speed internet service. It sparked the high-speed internet development between service providers during the 1990's and, of course, revolutionized internet use. Much like its predecessor, the dial-up internet service, ISDN utilizes a phone line. In fact, it set the standard for telephone data service.

ISDN internet service was the improvement upon dial-up, and it also paved the way for DSL and cable-modem internet service thereafter. It can be considered the step of internet evolution that lies between dial-up and DSL/Cable. Modernizing internet use and bringing high-speed access inside the home, ISDN became the standard by which rival broadband internet service providers competed.

Although ISDN internet service still exists, like the dial-up connection it is being replaced by faster and cheaper services that the broadband companies are providing. Regardless, broadband high-speed internet service is still compared with ISDN today since they both represent the standard of their times.

ISDN internet service is basically a telephone-based network system that operates by a circuit switch, or dedicated line. It can transmit data and phone conversations digitally over normal telephone wires. This makes it both faster and of higher quality than dial-up internet service.

During the 1990's this revolutionized the way people did business. No longer would you have to miss a call in order to access your internet, or shut down the internet to make a telephone call. As such, ISDN internet service made video teleconferencing not only possible, but very popular at this time as well.

There are two different types, or lines, of ISDN internet service. The first is a basic rate ISDN line. Called a Basic Rate Interface (BRI), this line has two data, or bearer, channels that operate at 64 kbit/sec. Two or more ISDN-BRI lines can be combined as well, yielding speeds of 256 kbit/sec. Combining these lines is common for video conferencing use or for transmitting data at higher speeds.

The second type of ISDN line is called a primary rate line, or Primary Rate Interface (PRI). This line had 23 bearer channels and has a total speed of 1,544 kbit/sec. It is used mostly for telephone communication rather than data transmission, particularly within companies that have large, private telephone exchange systems operating inside their business.

The advantages of having ISDN internet service definitely lies in the data lines themselves. Not only do you have constant data speed via these lines, each bearer channel runs at 64 kbit/sec with the ability to be combined to reach greater speeds. ISDN internet serviced also allows for multiple data transmission, so telephone calls and data downloading are no longer mutually exclusive.

The disadvantages, however, is that the digital clarity of ISDN voice communication and its speedy data transmission come at an extra cost. ISDN is billed like a phone line, but with an added cost for service. And although its operational distance from the ISDN central office is greater than that for DSL, its terminal adaptor (similar to a modem) is more expensive than DSL or cable modems.

While this equipment and service continue to remain costly, it is leaving the way open for other internet services, like broadband, to quickly replace ISDN's share of the marketplace.

4.6 The numbering plan of telephone networks (National Schemes & International Numbering)

INTRODUCTION

The National Numbering Plan was last reviewed during 1993. The plan covered basic as well as other services like cellular mobile, paging etc. Though the 1993 Numbering Plan could cater to the needs of existing and new services for another few years, yet it was felt to rationalise and review the existing National Numbering Plan because of introduction of a large number of new telecom services and opening up of the entire telecom sector for private participation.

The existing Numbering Plan was formulated at a time when there was no competition in the basic telecom services and the competition in cellular mobile services had just started, paging services were in a stage of infancy and Internet services were not available in the country.

The main objectives of the plan are –

i) To plan in conformity with relevant and applicable ITU standards to the extent possible.

ii) To meet the challenges of the changing telecom environment.

iii) To reserve numbering capacity to meet the undefined future needs.

iv) To support effective competition by fair access to numbering resources.

v) To meet subscriber needs for a meaningful and user-friendly scheme. Only the decimal character set 0-9 has been used for all number allocations. Letters and other non-decimal characters shall not form part of the National (Significant) Number [N(S)N]. Dialling procedure as per ITU Recommendation E.164 has been followed.

The Short Distance Charging Area (SDCA) based linked numbering scheme with 10-digit N(S)N has been followed. This would expand the existing numbering capacity to ten times.

NATIONAL NUMBERING SCHEME

Level '0':

Sub level '000':

The prefix '000' shall be used for home country direct service (Bilateral) and international toll free service (Bilateral). The format used is: '000 + Country Code + Operator Code' except '000800' which is used for bilateral international toll free service.

Sub level '0010' - INTERNATIONAL CARRIER ACCESS (Prefix) CODE: The prefix '0010' shall be used for selection of international carrier. It will be followed by International Carrier Identification Code (ICIC), Country Code (CC) and N(S)N. The format shall be as under:

Prefix	International Carrier Identification Code	Country Code	National(Significant)Number
0010	ICIC	CC	N(S)N

Initially ICIC shall be a two-digit code. This will be sufficient for allotment to 50 international long distance service providers considering that maximum of two codes may be allotted to each service provider depending upon toll quality and non-toll quality network. However, to take care of all possible future requirements, length of ICIC may be reviewed and changed to 3- digit code as and when required. The allotment of ICIC may start from '10' and codes '00' to '09' may be kept reserved.

Sub level '00' - INTERNATIONAL PREFIX:

The prefix '00' shall be used for International dialling. It will be followed by country code and the N(S)N of the country to which that call is attempted. The format is as per ITU Recommendation E.164:

Prefix	Country Code	National(Significant)Number
00	CC	N(S)N

Sub level '010' - NATIONAL CARRIER ACCESS (Prefix) CODE:

The prefix '010' shall be used for selection of national long distance carrier. It will be followed by (National) Carrier Identification Code (CIC) and N(S) N. The format shall be as under:

Prefix	Carrier Identification Code	National(Significant)Number
010	CIC	N(S)N

Initially CIC shall be a two-digit code. This will be sufficient for allotment to 40 NLDOs (including NLDOs licensed for basic services) and 10 BSOs licensed only for basic services, considering that maximum of two codes may be allotted to each service provider depending upon toll quality and non-toll quality network. However to take care of all possible future requirements, length of CIC may be reviewed and changed to 3-digit code in future.

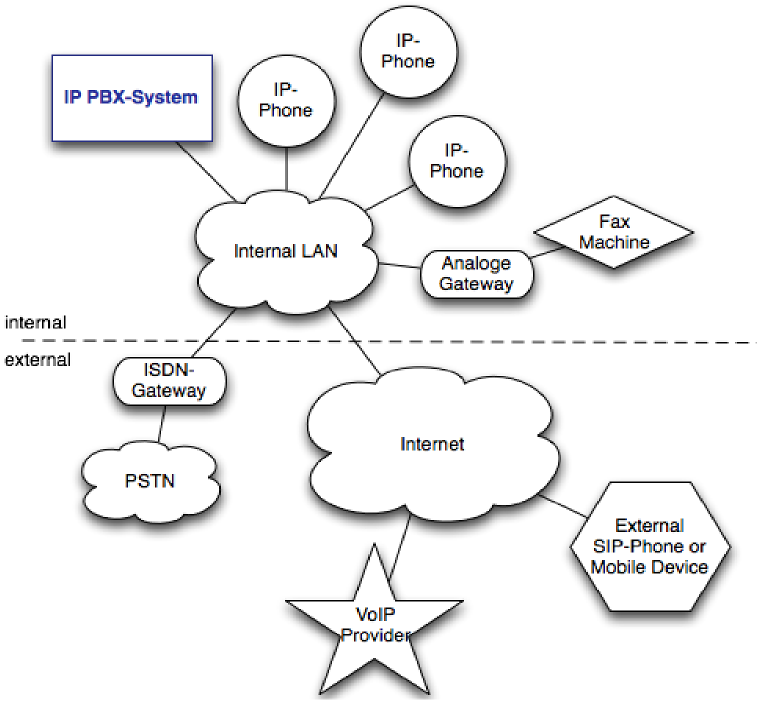
The allotment of CIC may start from '10' and codes '00' to '09' may be kept reserved. For intra circle long distance service, the carrier access code shall be the same as applicable for NLD service. The CIC from '10' to '79' shall be allotted to NLD service providers. For the NLD service providers, who are also Basic Service Operators (BSOs), same CIC shall be applicable for intra circle (service area) calls. CIC from '80' to '99' shall be allocated to the BSOs who are not licensed to provide NLD service.

4.7 Operation PBX & Digital EPABX.

PBX

What is a PBX Phone System?

PBX stands for Private Branch Exchange, which is a private telephone network used within a company or organization. The users of the PBX phone system can communicate within their company or organization and the outside world, using different communication channels like Voice over IP, ISDN or analog. A PBX also allows you to have more phones than physical phone lines (PTSN) and allows free calls between users. It also provides features like transfers, voicemail, call recording, interactive voice menus (IVRs) and ACD call queues.



PBX phone systems are available as Hosted or Virtual solutions (sometimes also called Centrix), and as inhouse solutions to be used on your own hardware.

PBX phone systems are usually much more flexible than proprietary systems, as they are using open standards and interfaces. Modern PBX phone systems are based on standard hardware, which is cheaper and can easier be replaced than a closed systems.

Switching to an IP PBX offers many benefits

With an IP phone system all your internal telephony is routed through the existing LAN (local computer network). This way a separate network for telephony is not required.

Even though the internal telephony is routed through the LAN, it is also possible to connect your IP-PBX via gateways to the PSTN. Of course, VoIP (Voice over IP, telephony via the internet) is also possible.

Since IP telephony is mostly using the open SIP standard, an IP phone system gives you a lot more freedom in your choice of phones.

Basically any SIP compatible phone (VoIP phone) will work with an IP PBX. Furthermore an IP PBX doesn’t limit the growth of a company.

Since VoIP phones don’t have to be connected physically to the phone system, it doesn’t require a free port in the phone system like it used to be with traditional phone systems.

IP phones can not only be connected via the LAN but also via the internet, using for example a VPN connection. Because of this, multiple locations and offices can easily be connected.

There is a huge variety of VoIP providers on the internet which provide SIP trunking (telephony services) for cheaper call rates than traditional telephony providers. Internal calls via an IP phone system are free general.

Practical advantages of IP telephony

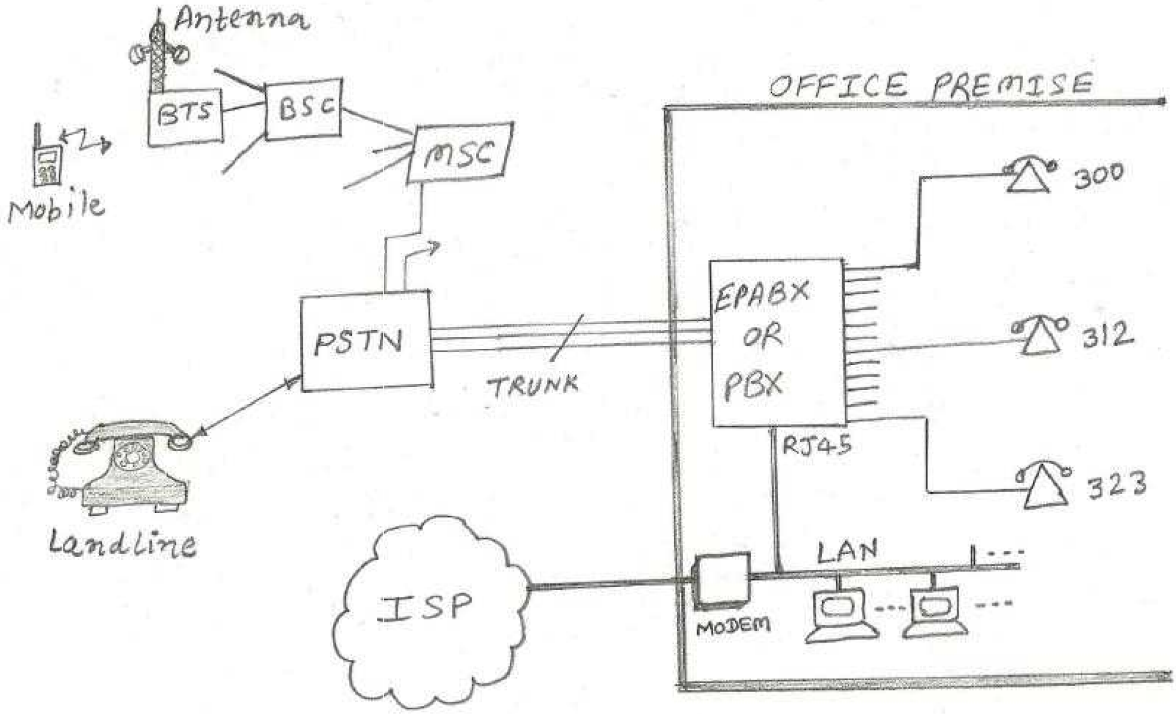
Interconnecting teams and mobile working is one of the huge advantages of IP phone systems. No matter if team members are on the road, are located in a different country or work from home, they can connect via IP desk phone, smart phone or laptop to the PBX in the office.

This way all calls within the company are free and clients will not realize if en employee is in the office or somewhere else around the world. The same also applies for conferences, these can be hosted directly on the own IP PBX with as many participants as required. This safes traveling time and money.

Digital EPABX

As shown in the figure EPABX/PBX facilitates use of one external telephone line by many internal users in the office premises. In the office each employee is provided one telephone set and all the telephones are connected with PBX.

All the employees within the office premises can communicate using 3-digit or 4-digit number programmed in EPABX/PBX without any charge.



EPABX/PBX is connected to PSTN (Public switched Telephone network) via trunk lines; hence all can use one external voice line in time shared basis.

PSTN is connected with MSC (Mobile switching centre) of cellular networks such as GSM/CDMA/UMTS. By this mobile cell phone user can connect to any telephone set in the office premises using extension number.

Similar to voice line EPABX/PBX can be used for Data applications. As shown in figure Data port of PBX is connected to LAN where so many PCs are connected and are using same external internet connection line from ISP via Modem/router. The same facility of PBX can extended for WLAN users too.

4.8 Define units of Power Measurement.

The watt (symbol: W) is a derived unit of power in the International System of Units (SI), named after the Scottish engineer James Watt (1736–1819). The unit is defined as joule per second[1] and can be used to express the rate of energy conversion or transfer with respect to time. It has dimensions of

L2MT-3

When an object's velocity is held constant at one meter per second against constant opposing force of one newton the rate at which work is done is 1 watt.

$$W = \frac{J}{s} = \frac{N \cdot m}{s} = \frac{kg \cdot m^2}{s^3}$$

In terms of electromagnetism, one watt is the rate at which work is done when one ampere (A) of current flows through an electrical potential difference of one volt (V).

$$W = V \cdot A$$

Two additional unit conversions for watt can be found using the above equation and Ohm's Law.

$$W = \frac{V^2}{\Omega} = A^2 \cdot \Omega$$

Where ohm (Ω) is the SI derived unit of electrical resistance.

Femtowatt

The femtowatt is equal to one quadrillionth (10−15) of a watt. Technologically important powers that are measured in femtowatts are typically found in reference(s) to radio and radar receivers. For example, meaningful FM tuner performance figures for sensitivity, quieting and signal-to-noise require that the RF energy applied to the antenna input be specified.

These input levels are often stated in dBf (decibels referenced to 1 femtowatt). This is 0.2739 microvolt across a 75-ohm load or 0.5477 microvolt across a 300 ohm load; the specification takes into account the RF input impedance of the tuner.

Picowatt

The picowatt is equal to one trillionth (10−12) of a watt. Technologically important powers that are measured in picowatts are typically used in reference to radio and radar receivers, acoustics and in the science of radio astronomy.

Nanowatt

The nanowatt is equal to one billionth (10^{-9}) of a watt. Important powers that are measured in nanowatts are also typically used in reference to radio and radar receivers.

Microwatt

The microwatt is equal to one millionth (10−6) of a watt. Important powers that are measured in microwatts are typically stated in medical instrumentation systems such as the EEG and the ECG, in a wide variety of scientific and engineering instruments and also in reference to radio and radar receivers. Compact solar cells for devices such as calculators and watches are typically measured in microwatts.

Milliwatt

The milliwatt is equal to one thousandth (10−3) of a watt. A typical laser pointer outputs about five milliwatts of light power, whereas a typical hearing aid for people uses less than one milliwatt.

Kilowatt

The kilowatt is equal to one thousand (103) watts, or one sthene-metre per second. This unit is typically used to express the output power of engines and the power of electric motors, tools, machines, and heaters. It is also a common unit used to express the electromagnetic power output of broadcast radio and television transmitters.

One kilowatt is approximately equal to 1.34 horsepower. A small electric heater with one heating element can use 1.0 kilowatt, which is equivalent to the power of a household in the United States averaged over the entire year.

Also, kilowatts of light power can be measured in the output pulses of some lasers.

A surface area of one square meter on Earth receives typically about one kilowatt of sunlight from the sun (the solar irradiance) (on a clear day at mid day, close to the equator).

Megawatt

The megawatt is equal to one million (106) watts. Many events or machines produce or sustain the conversion of energy on this scale, including lightning strikes; large electric motors; large warships such as aircraft carriers, cruisers, and submarines; large server farms or data centers; and some scientific research equipment, such as supercolliders, and the output pulses of very large lasers.

A large residential or commercial building may use several megawatts in electric power and heat. On railways, modern high-powered electric locomotives typically have a peak power output of 5 or 6 MW, although some produce much more.

The Eurostar, for example, uses more than 12 MW, while heavy diesel-electric locomotives typically produce/use 3 to 5 MW. U.S. nuclear power plants have net summer capacities between about 500 and 1300 MW.

The earliest citing of the megawatt in the Oxford English Dictionary (OED) is a reference in the 1900 Webster's International Dictionary of English Language. The OED also states that megawatt appeared in a 28 November 1947 article in the journal Science

Gigawatt

The gigawatt is equal to one billion (109) watts or 1 gigawatt = 1000 megawatts. This unit is often used for large power plants or power grids. For example, by the end of 2010 power shortages in China's Shanxi province were expected to increase to 5–6 GW and the installed capacity of wind power in Germany was 25.8 GW. The largest unit (out of four) of the Belgian Nuclear Plant Doel has a peak output of 1.04 GW. HVDC converters have been built with power ratings of up to 2 GW.[11] The London Array, the world's largest offshore wind farm, is designed to produce a gigawatt of power

Terawatt

The terawatt is equal to one trillion (1012) watts. The total power used by humans worldwide (about 16 TW in 2006) is commonly measured in this unit. The most powerful lasers from the mid-1960s to the mid-1990s produced power in terawatts, but only for nanosecond time frames. The average lightning strike peaks at 1 terawatt, but these strikes only last for 30 microseconds.

Petawatt

The petawatt is equal to one quadrillion (1015) watts and can be produced by the current generation of lasers for time-scales on the order of picoseconds (10−12 s). One such laser is the Lawrence Livermore's Nova laser, which achieved a power output of 1.25 PW (1.25 × 1015 W) by a process called chirped pulse amplification. The duration of the pulse was about 0.5 ps (5 × 10−13 s), giving a total energy of 600 J, or enough energy to power a 100 W light bulb for six seconds

4.9 OPERATION AND PRINCIPLE OFInternet Protocol Telephony (IP Telephony)

Internet Protocol Telephony (IP Telephony) is the use of IP-based networks to build, provide and access voice, data or other forms of telephonic communications. IP telephony provides traditional telephonic communication over an IP-based network, the Internet - via an Internet service provider (ISP) - or directly from a telecommunications service provider.

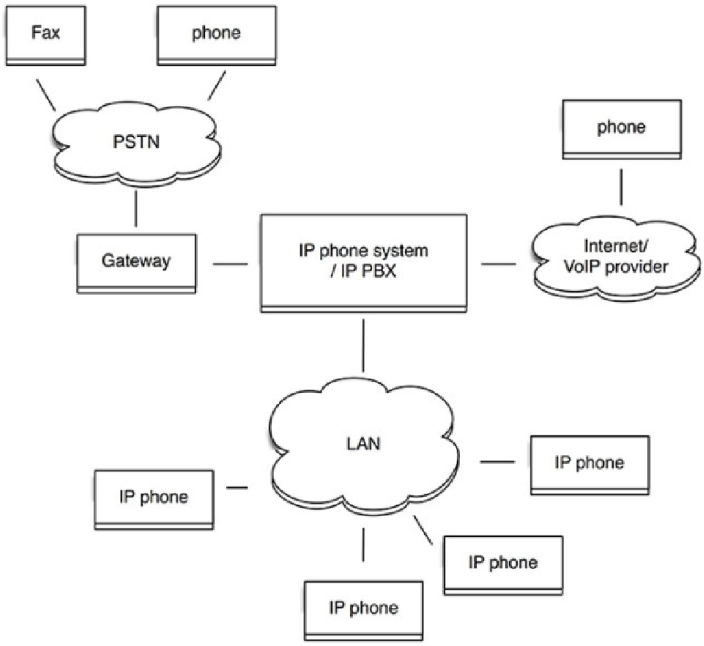
IP telephony is designed to replace the telecommunications' infrastructure of circuit switched public data networks (CSPDN) and public switched telephone networks (PSTN) with packet switched IP communication networks. In a consumer IP telephony solution, a soft IP phone application and backend Internet connection enable voice and data communication, such as calling and faxing.

A user may call other soft phone users, send or receive faxes and even communicate with circuit switched and cellular communication services. In an enterprise environment, IP telephony is implemented through physical IP phones that work on top of an IP network infrastructure. An IP phone's built-in firmware provides the complete functionality for initiating and managing telephonic communications.

Moreover, IP telephony also supports video communication between two or more users. Voice over Internet Protocol (VoIP), a popular IP telephony implementation, only supports voice communication over IP.

How do IP phone systems work?

The "IP" in IP phone system refers to Voice over IP, or having your phone calls routed over the internet or your local network (LAN). This is great for many reasons. First of all, you don't have to use the telephone network of your telephony service provider for making calls, which will reduce your costs for phone calls.



At the same time you are gaining many technical advantages by using IP technology for your telephony. Users of an VoIP phone system simply plug their IP phone into the nearest LAN port. Then, the IP phone registers automatically at the VoIP phone system.

The IP phone always keeps its number, and behaves exactly the same way, no matter where you plug it in – on your desk, in the office next door or on a tropical island. All of this works because of the SIP protocol.

It is a standard widely used by ISPs, VoIP phone systems and VoIP phones world-wide. It makes expensive proprietary phones obsolete, and helps that all devices can talk to each other.

IP phone systems are usually built on standard PC or embedded hardware which are more cost-effective and powerful than the hardware of the traditional phone manufacturers.

At the same time, ip phone systems are scalable, as they are not limited to a certain number of physical phone ports. That means you don't need to replace your phone system when your company grows.

---- ALL THE BEST ---- ALL THE BEST ----