o Syntax. The term *syntax* refers to the structure or format of the data, meaning the order in which they are presented. For example, a simple protocol might expect the first 8 bits of data to be the address of the sender, the second 8 bits to be the address of the receiver, and the rest of the stream to be the message itself.

o Semantics. The word *semantics* refers to the meaning of each section of bits. How is a particular pattern to be interpreted, and what action is to be taken based on that interpretation? For example, does an address identify the route to be taken or the final destination of the message?

o Timing. The term *timing* refers to two characteristics: when data should be sent and how fast they can be sent. For example, if a sender produces data at 100 Mbps but the receiver can process data at only 1 Mbps, the transmission will overload the receiver and some data will be lost.

Standards

Standards are essential in creating and maintaining an open and competitive market for equipment manufacturers and in guaranteeing national and international interoperability of data and telecommunications technology and processes. Standards provide guidelines to manufacturers, vendors, government agencies, and other service providers to ensure the kind of interconnectivity necessary in today's marketplace and in international communications.

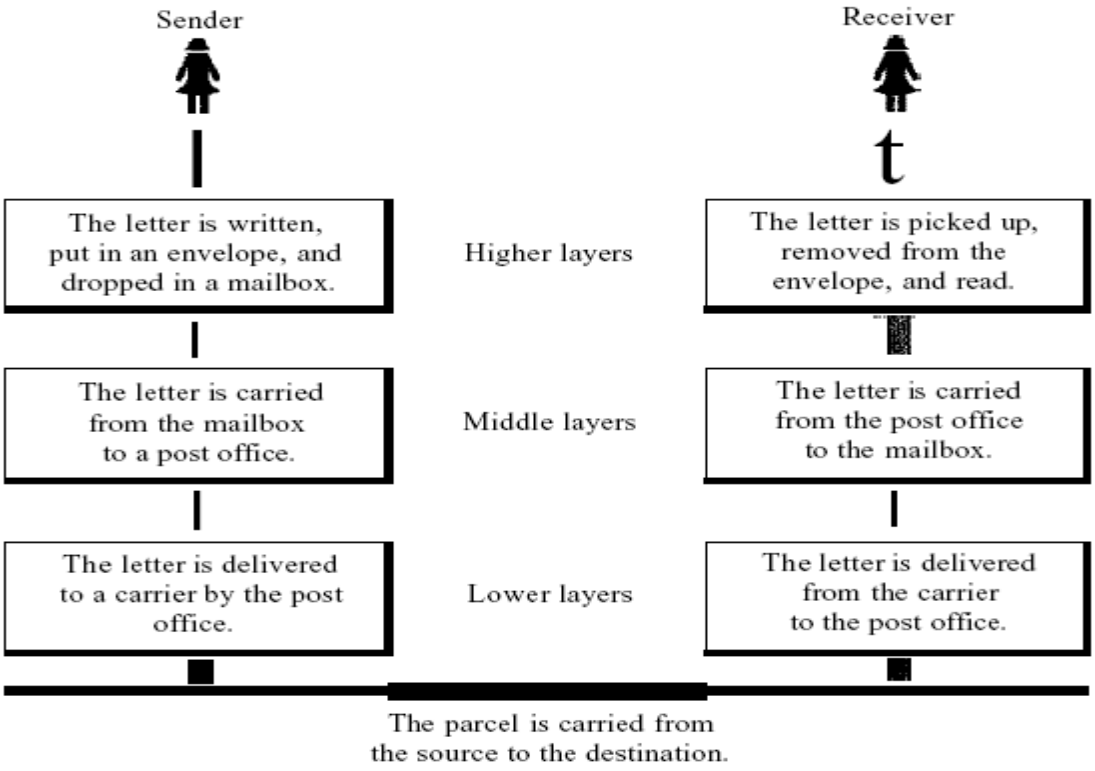
Data communication standards fall into two categories: *de facto* (meaning "by fact" or "by convention") and *de jure* (meaning "by law" or "by regulation").

o De facto. Standards that have not been approved by an organized body but have been adopted as standards through widespread use are de facto standards. De facto standards are often established originally by manufacturers who seek to define the functionality of a new product or technology.

o De jure. Those standards that have been legislated by an officially recognized body are de jure standards.

1.5 LAYERED TASKS

We use the concept of layers in our daily life. As an example, let us consider two friends who communicate through postal maiL The process of sending a letter to a friend would be complex if there were no services available from the post office. Below Figure shows the steps in this task.



Sender, Receiver, and Carrier

In Figure we have a sender, a receiver, and a carrier that transports the letter. There is a hierarchy of tasks.

At the Sender Site

Let us first describe, in order, the activities that take place at the sender site.

o Higher layer. The sender writes the letter, inserts the letter in an envelope, writes the sender and receiver addresses, and drops the letter in a mailbox.

o Middle layer. The letter is picked up by a letter carrier and delivered to the post office.

o Lower layer. The letter is sorted at the post office; a carrier transports the letter.

0n the Way: The letter is then on its way to the recipient. On the way to the recipient's local post office, the letter may actually go through a central office. In addition, it may be transported by truck, train, airplane, boat, or a combination of these.

At the Receiver Site

- o Lower layer. The carrier transports the letter to the post office.
- o Middle layer. The letter is sorted and delivered to the recipient's mailbox.
- o Higher layer. The receiver picks up the letter, opens the envelope, and reads it.

1.6 The OSI Reference Model:

The OSI model (minus the physical medium) is shown in Fig. This model is based on a proposal developed by the International Standards Organization (ISO) as a first step toward international standardization of the protocols used in the various layers (Day and Zimmermann, 1983). It was revised in 1995(Day, 1995). The model is called the ISO-OSI (Open Systems Interconnection) Reference Model because it deals with connecting open systems—that is, systems that are open for communication with other systems.

The OSI model has seven layers. The principles that were applied to arrive at the seven layers can be briefly summarized as follows:

1. A layer should be created where a different abstraction is needed.
2. Each layer should perform a well-defined function.
3. The function of each layer should be chosen with an eye toward defining internationally standardized protocols.
4. The layer boundaries should be chosen to minimize the information flow across the interfaces.
5. The number of layers should be large enough that distinct functions need not be thrown together in the same layer out of necessity and small enough that the architecture does not become unwieldy.

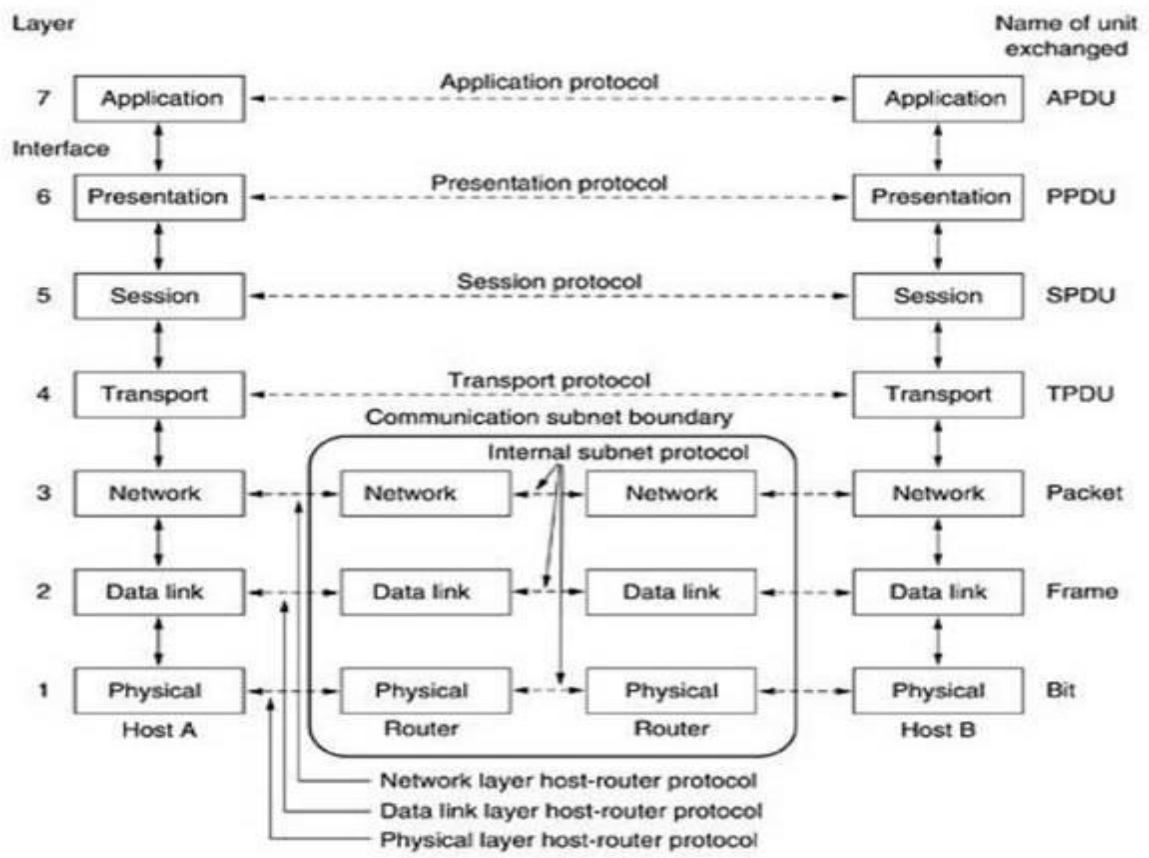


Fig.4: The OSI reference model

The Physical Layer:

The physical layer is concerned with transmitting raw bits over a communication channel. The design issues have to do with making sure that when one side sends a 1 bit, it is received by the other side as a 1 bit, not as a 0 bit.

The Data Link Layer:

The main task of the data link layer is to transform a raw transmission facility into a line that appears free of undetected transmission errors to the network layer. It accomplishes this task by having the sender break up the input data into data frames (typically a few hundred or a few thousand bytes) and transmits the frames sequentially. If the service is reliable, the receiver confirms correct receipt of each frame by sending back an acknowledgement frame.

Another issue that arises in the data link layer (and most of the higher layers as well) is how to keep a fast transmitter from drowning a slow receiver in data. Some traffic regulation mechanism is often needed to let the transmitter know how much buffer space the receiver has at the moment. Frequently, this flow regulation and the error handling are integrated.

The Network Layer:

The network layer controls the operation of the subnet. A key design issue is determining how packets are routed from source to destination. Routes can be based on static tables that are "wired into" the network and rarely changed. They can also be determined at the start of each conversation, for example, a terminal session (e.g., a login to a remote machine). Finally, they can be highly dynamic, being determined anew for each packet, to reflect the current network load.

If too many packets are present in the subnet at the same time, they will get in one another's way, forming bottlenecks. The control of such congestion also belongs to the network layer. More generally, the quality of service provided (delay, transit time, jitter, etc.) is also a network layer issue.

When a packet has to travel from one network to another to get to its destination, many problems can arise. The addressing used by the second network may be different from the first one. The second one may not accept the packet at all because it is too large. The protocols may differ, and so on. It is up to the network layer to overcome all these problems to allow heterogeneous networks to be interconnected. In broadcast networks, the routing problem is simple, so the network layer is often thin or even nonexistent.

The Transport Layer:

The basic function of the transport layer is to accept data from above, split it up into smaller units if need be, pass these to the network layer, and ensure that the pieces all arrive correctly at the other end. Furthermore, all this must be done efficiently and in a way that isolates the upper layers from the inevitable changes in the hardware technology. The transport layer also determines what type of service to provide to the session layer, and, ultimately, to the users of the network. The most popular type of transport connection is an error-free point-to-point channel that delivers messages or bytes in the order in which they were sent. However, other possible kinds of transport service are the transporting of isolated messages, with no guarantee about the order of delivery, and the broadcasting of messages to multiple destinations. The type of service is determined when the connection is established.

The transport layer is a true end-to-end layer, all the way from the source to the destination. In other words, a program on the source machine carries on a conversation with a similar program on the destination machine, using the message headers and control messages. In the lower layers,

the protocols are between each machine and its immediate neighbours, and not between the ultimate source and destination machines, which may be separated by many routers.

The Session Layer:

The session layer allows users on different machines to establish sessions between them. Sessions offer various services, including dialog control (keeping track of whose turn it is to transmit), token management (preventing two parties from attempting the same critical operation at the same time), and synchronization (check pointing long transmissions to allow them to continue from where they were after a crash).

The Presentation Layer:

The presentation layer is concerned with the syntax and semantics of the information transmitted. In order to make it possible for computers with different data representations to communicate, the data structures to be exchanged can be defined in an abstract way, along with a standard encoding to be used "on the wire." The presentation layer manages these abstract data structures and allows higher-level data structures (e.g., banking records), to be defined and exchanged.

The Application Layer:

The application layer contains a variety of protocols that are commonly needed by users. One widely-used application protocol is HTTP (Hypertext Transfer Protocol), which is the basis for the World Wide Web. When a browser wants a Web page, it sends the name of the page it wants to the server using HTTP. The server then sends the page back. Other application protocols are used for file transfer, electronic mail, and network news.

1.7 The TCP/IP Reference Model:

The TCP/IP reference model was developed prior to OSI model. The major design goals of this model were,

1. To connect multiple networks together so that they appear as a single network.
2. To survive after partial subnet hardware failures.
3. To provide a flexible architecture.

Unlike OSI reference model, TCP/IP reference model has only 4 layers. They are,

1. Host-to-Network Layer
2. Internet Layer

3. Transport Layer

4. Application Layer

Application Layer

Transport Layer

Internet Layer

Host-to-Network Layer

Host-to-Network Layer:

The TCP/IP reference model does not really say much about what happens here, except to point out that the host has to connect to the network using some protocol so it can send IP packets to it. This protocol is not defined and varies from host to host and network to network.

Internet Layer:

This layer, called the internet layer, is the linchpin that holds the whole architecture together. Its job is to permit hosts to inject packets into any network and have they travel independently to the destination (potentially on a different network). They may even arrive in a different order than they were sent, in which case it is the job of higher layers to rearrange them, if in-order delivery is desired. Note that "internet" is used here in a generic sense, even though this layer is present in the Internet.

The internet layer defines an official packet format and protocol called IP (Internet Protocol). The job of the internet layer is to deliver IP packets where they are supposed to go. Packet routing is clearly the major issue here, as is avoiding congestion. For these reasons, it is reasonable to say that the TCP/IP internet layer is similar in functionality to the OSI network layer. Fig. shows this correspondence.

The Transport Layer:

The layer above the internet layer in the TCP/IP model is now usually called the transport layer. It is designed to allow peer entities on the source and destination hosts to carry on a conversation, just as in the OSI transport layer. Two end-to-end transport protocols have been defined here. The first one, TCP (Transmission Control Protocol), is a reliable connection-oriented protocol that allows a byte stream originating on one machine to be delivered without error on any other machine in the internet. It fragments the incoming byte stream into discrete messages and passes each one on to the internet layer. At the destination, the receiving TCP process reassembles the received messages into the output stream. TCP also handles flow control

to make sure a fast sender cannot swamp a slow receiver with more messages than it can handle.

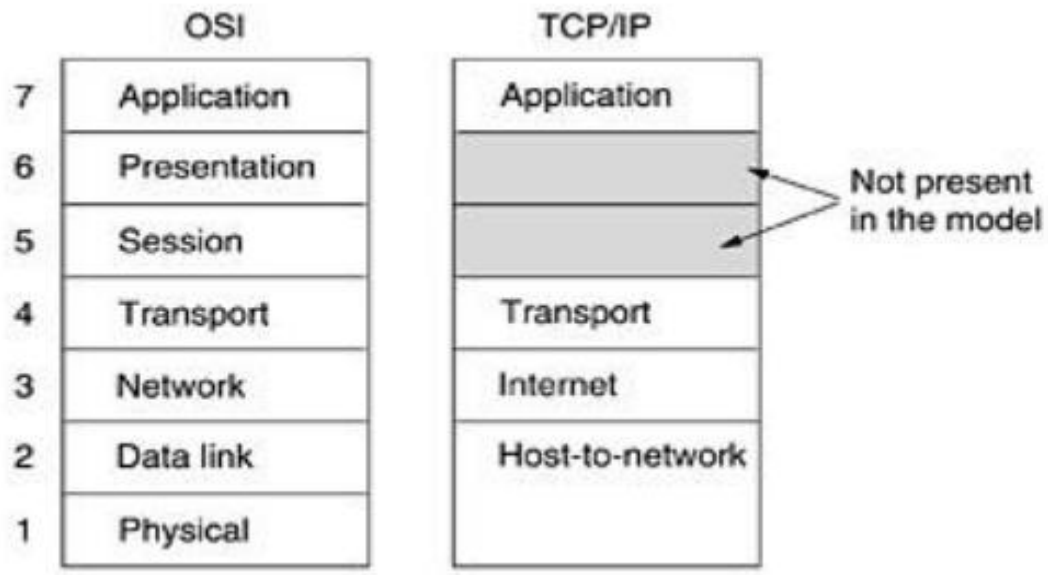


Fig.1: The TCP/IP reference model.

The second protocol in this layer, UDP (User Datagram Protocol), is an unreliable, connectionless protocol for applications that do not want TCP's sequencing or flow control and wish to provide their own. It is also widely used for one-shot, client-server-type request-reply queries and applications in which prompt delivery is more important than accurate delivery, such as transmitting speech or video. The relation of IP, TCP, and UDP is shown in Fig.2. Since the model was developed, IP has been implemented on many other networks.

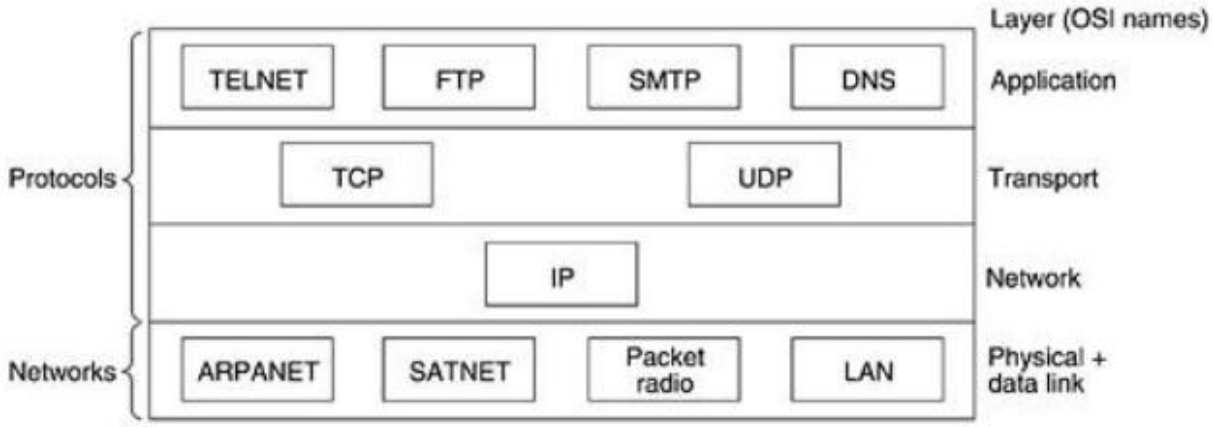


Fig.2: Protocols and networks in the TCP/IP model initially.

The Application Layer:

The TCP/IP model does not have session or presentation layers. On top of the transport layer is the application layer. It contains all the higher-level protocols. The early ones included virtual terminal (TELNET), file transfer (FTP), and electronic mail (SMTP), as shown in Fig.6.2. The virtual terminal protocol allows a user on one machine to log onto a distant machine and work there. The file transfer protocol provides a way to move data efficiently from one machine to another. Electronic mail was originally just a kind of file transfer, but later a specialized protocol (SMTP) was developed for it. Many other protocols have been added to these over the years: the Domain Name System (DNS) for mapping host names onto their network addresses, NNTP, the protocol for moving USENET news articles around, and HTTP, the protocol for fetching pages on the World Wide Web, and many others.

Comparison of the OSI and TCP/IP Reference Models:

The OSI and TCP/IP reference models have much in common. Both are based on the concept of a stack of independent protocols. Also, the functionality of the layers is roughly similar. For example, in both models the layers up through and including the transport layer are there to provide an end-to-end, network-independent transport service to processes wishing to communicate. These layers form the transport provider. Again in both models, the layers above transport are application-oriented users of the transport service. Despite these fundamental similarities, the two models also have many differences Three concepts are central to the OSI model:

1. Services.
2. Interfaces.
3. Protocols.

Probably the biggest contribution of the OSI model is to make the distinction between these three concepts explicit. Each layer performs some services for the layer above it. The service definition tells what the layer does, not how entities above it access it or how the layer works. It defines the layer's semantics.

A layer's interface tells the processes above it how to access it. It specifies what the parameters are and what results to expect. It, too, says nothing about how the layer works inside.

Finally, the peer protocols used in a layer are the layer's own business. It can use any protocols it wants to, as long as it gets the job done (i.e., provides the offered services). It can also change them at will without affecting software in higher layers.

The TCP/IP model did not originally clearly distinguish between service, interface, and protocol, although people have tried to retrofit it after the fact to make it more OSI-like. For example, the only real services offered by the internet layer are SEND IP PACKET and RECEIVE IP PACKET.

As a consequence, the protocols in the OSI model are better hidden than in the TCP/IP model and can be replaced relatively easily as the technology changes. Being able to make such changes is one of the main purposes of having layered protocols in the first place. The OSI reference model was devised before the corresponding protocols were invented. This ordering means that the model was not biased toward one particular set of protocols, a fact that made it quite general. The downside of this ordering is that the designers did not have much experience with the subject and did not have a good idea of which functionality to put in which layer.

Another difference is in the area of connectionless versus connection-oriented communication. The OSI model supports both connectionless and connection-oriented communication in the network layer, but only connection-oriented communication in the transport layer, where it counts (because the transport service is visible to the users). The TCP/IP model has only one mode in the network layer (connectionless) but supports both modes in the transport layer, giving the users a choice. This choice is especially important for simple request-response protocols.

UNIT- 2

Physical Layer and Overview of PL Switching

2.1 MULTIPLEXING

Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link.

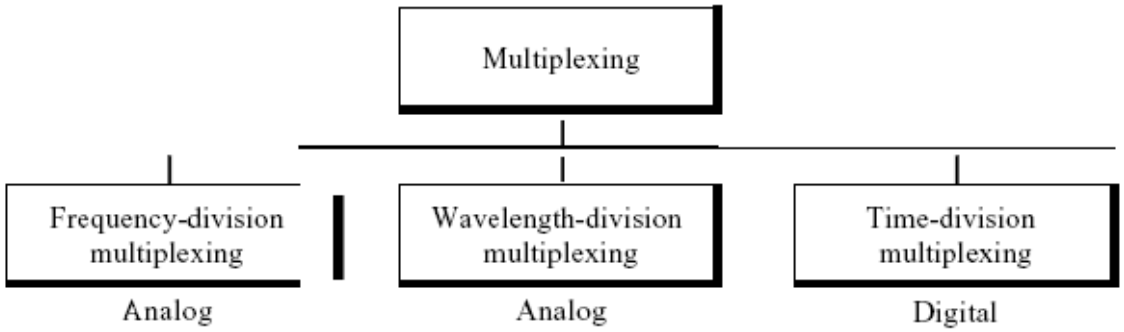


Figure *Categories of multiplexing*

2.1.1 Frequency-Division Multiplexing

Frequency-division multiplexing (FDM) is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted. In FOM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. These bandwidth ranges are the channels through which the various signals travel. Channels can be separated by strips of unused bandwidth-guard bands-to prevent signals from overlapping. In addition, carrier frequencies must not interfere with the original data frequencies.

Figure 1 gives a conceptual view of FDM. In this illustration, the transmission path is divided into three parts, each representing a channel that carries one transmission.

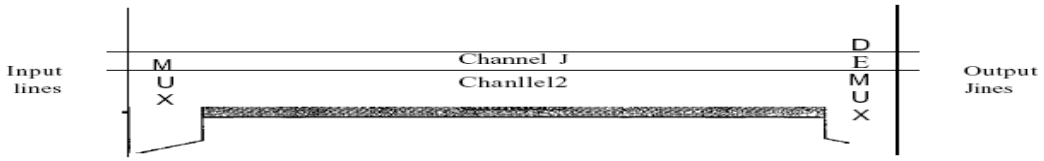


Fig: FDM

We consider FDM to be an analog multiplexing technique; however, this does not mean that FDM cannot be used to combine sources sending digital signals. A digital signal can be converted to an analog signal before FDM is used to multiplex them.

Multiplexing Process

Figure 2 is a conceptual illustration of the multiplexing process. Each source generates a signal of a similar frequency range. Inside the multiplexer, these similar signals modulates different carrier frequencies *(/1,12,* and *h).* The resulting modulated signals are then combined into a single composite signal that is sent out over a media link that has enough bandwidth to accommodate it.

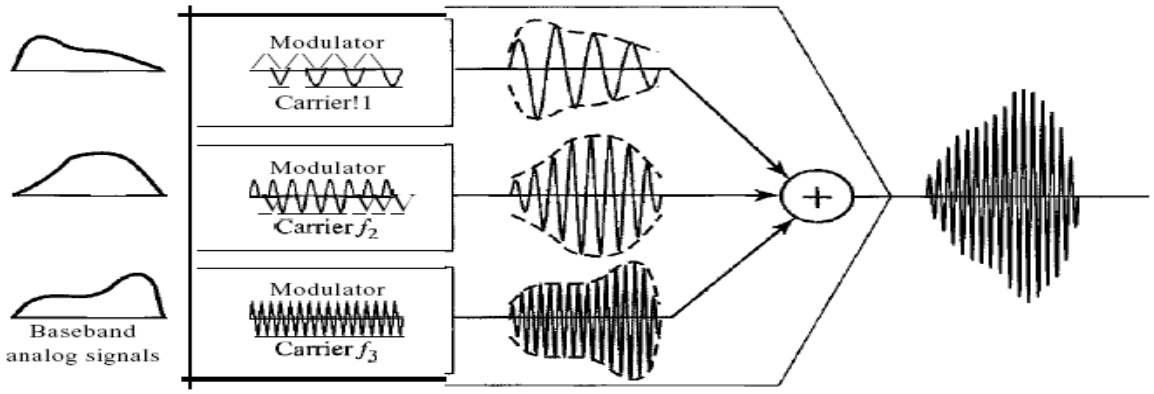


Figure 2 *FDM process*

Demultiplexing Process

The demultiplexer uses a series of filters to decompose the multiplexed signal into its constituent component signals. The individual signals are then passed to a demodulator that separates them from their carriers and passes them to the output lines. Figure 3 is a conceptual illustration of demultiplexing process.

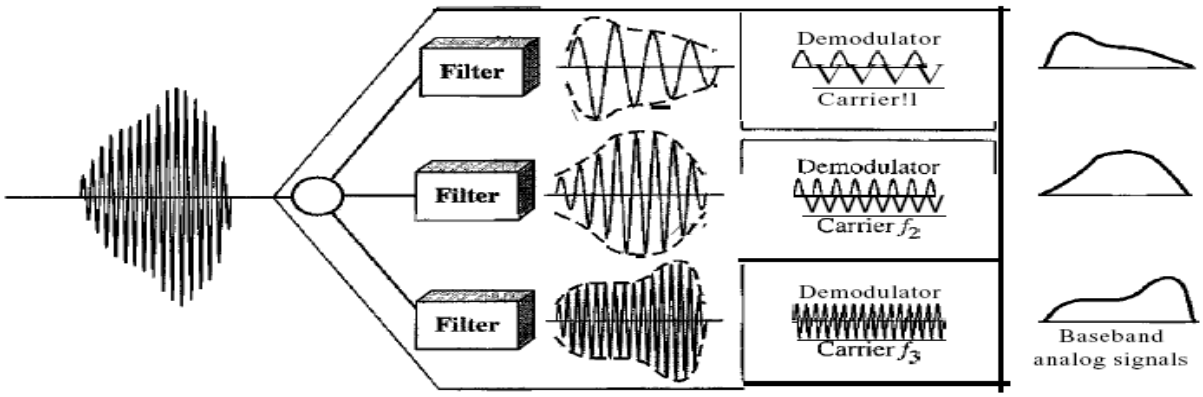


Figure 3 *FDM de-multiplexing example*

2.1.2 Wavelength-Division Multiplexing

Wavelength-division multiplexing (WDM) is designed to use the high-data-rate capability of fiber-optic cable. The optical fiber data rate is higher than the data rate of metallic transmission cable. Using a fiber-optic cable for one single line wastes the available bandwidth. Multiplexing allows us to combine several lines into one.

WDM is conceptually the same as FDM, except that the multiplexing and demultiplexing involve optical signals transmitted through fiber-optic channels. The idea is the same: We are combining different signals of different frequencies. The difference is that the frequencies are very high.

Figure 4 gives a conceptual view of a WDM multiplexer and demultiplexer. Very narrow bands of light from different sources are combined to make a wider band of light. At the receiver, the signals are separated by the demultiplexer.

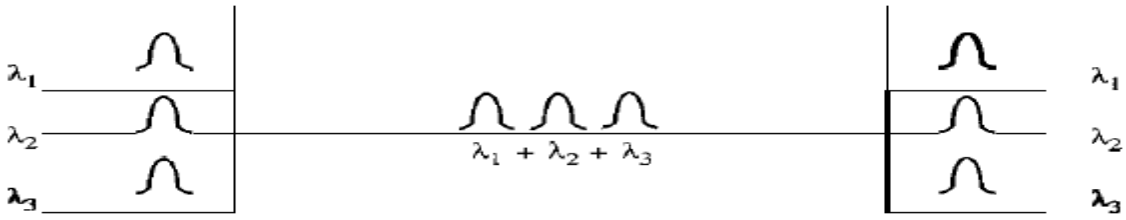


Figure 4 *Wavelength-division multiplexing*

Although WDM technology is very complex, the basic idea is very simple. We want to combine multiple light sources into one single light at the multiplexer and do the reverse at the demultiplexer. The combining and splitting of light sources are easily handled by a prism. Recall from basic physics that a prism bends a beam of light based on the angle of incidence and the frequency. Using this technique, a multiplexer can be made to combine several input beams of light, each containing a narrow band of frequencies, into one output beam of a wider band of frequencies. A demultiplexer can also be made to reverse the process. Figure 5 shows the concept.

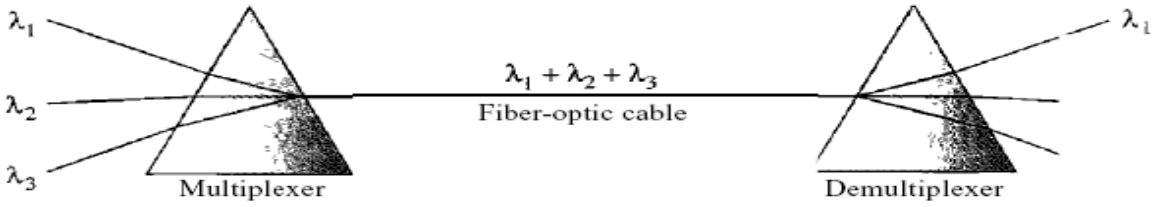


Figure 5 *Prisms in wavelength-division multiplexing and demultiplexing*

2.1.3 Synchronous Time-Division Multiplexing

Time-division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a linle Instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link. Figure 6 gives a conceptual view of TDM. Note that the same link is used as in FDM; here, however, the link is shown sectioned by time rather than by frequency. In the figure, portions of signals 1,2,3, and 4 occupy the link sequentially.

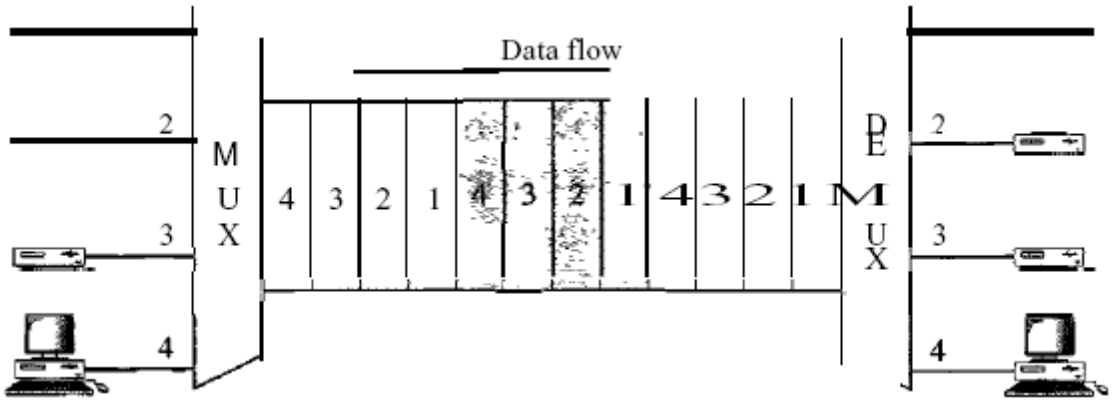


Figure 6 *TDM*

Note that in Figure 6 we are concerned with only multiplexing, not switching. This means that all the data in a message from source 1 always go to one specific destination, be it 1, 2, 3, or 4. The delivery is fixed and unvarying, unlike switching. We also need to remember thatTDM is, in principle, a digital multiplexing technique. Digital data from different sources are combined into one timeshared link. However, this does not mean that the sources cannot produce analog data; analog data can be sampled, changed to digital data, and then multiplexed by using TDM.

We can divide TDM into two different schemes: synchronous and statistical.

In synchronous TDM, each input connection has an allotment in the output even if it is not sending data.

Time Slots and Frames

In synchronous TDM, the data flow of each input connection is divided into units, where each input occupies one input time slot. A unit can be 1 bit, one character, or one block of data. Each input unit becomes one output unit and occupies one output time slot. However, the duration of an output time slot is *n* times shorter than the duration of an input time slot. If an input time slot is *T* s, the output time slot is *Tin* s, where *n* is the number of connections. In other words, a unit

in the output connection has a shorter duration; it travels faster. Figure 7 shows an example of synchronous TDM where *n* is 3.

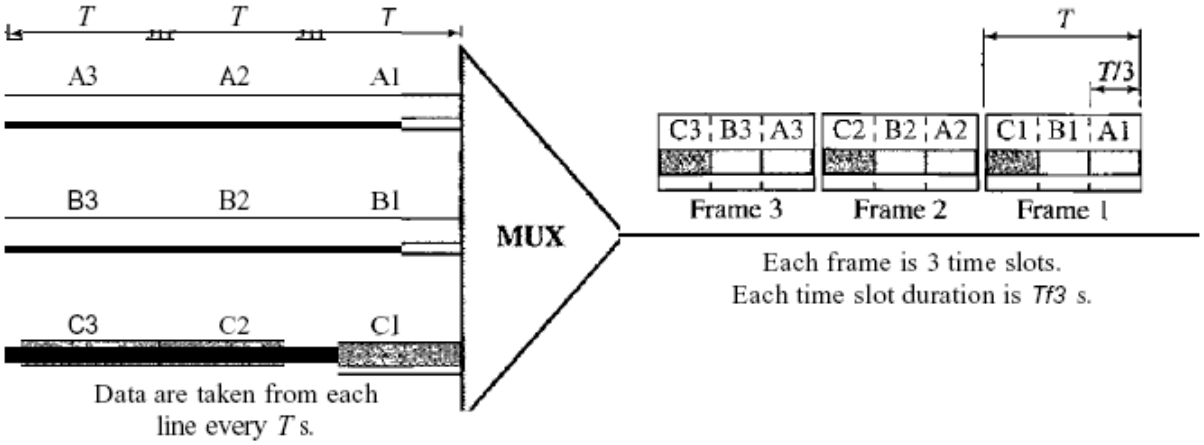


Figure 7 *Synchronous time-division multiplexing*

In synchronous TDM, a round of data units from each input connection is collected into a frame (we will see the reason for this shortly). If we have *n* connections, a frame is divided into *n* time slots and one slot is allocated for each unit, one for each input line. If the duration of the input unit is *T*, the duration of each slot is *Tin* and the duration of each frame is *T* (unless a frame carries some other information, as we will see shortly).

The data rate of the output link must be *n* times the data rate of a connection to guarantee the flow of data. In Figure 7, the data rate of the link is 3 times the data rate of a connection; likewise, the duration of a unit on a connection is 3 times that of the time slot (duration of a unit on the link). In the figure we represent the data prior to multiplexing as 3 times the size of the data after multiplexing. This is just to convey the idea that each unit is 3 times longer in duration before multiplexing than after. Time slots are grouped into frames. A frame consists of one complete cycle of time slots, with one slot dedicated to each sending device. In a system with *n* input lines, each frame has *n* slots, with each slot allocated to carrying data from a specific input line.

Interleaving

TDM can be visualized as two fast-rotating switches, one on the multiplexing side and the other on the demultiplexing side. The switches are synchronized and rotate at the same speed, but in opposite directions. On the multiplexing side, as the switch opens in front of a connection, that

connection has the opportunity to send a unit onto the path. This process is called **interleaving.** On the demultiplexing side, as the switch opens in front of a connection, that connection has the opportunity to receive a unit from the path.

Figure 8 shows the interleaving process for the connection shown in Figure 7.

In this figure, we assume that no switching is involved and that the data from the first connection at the multiplexer site go to the first connection at the demultiplexer

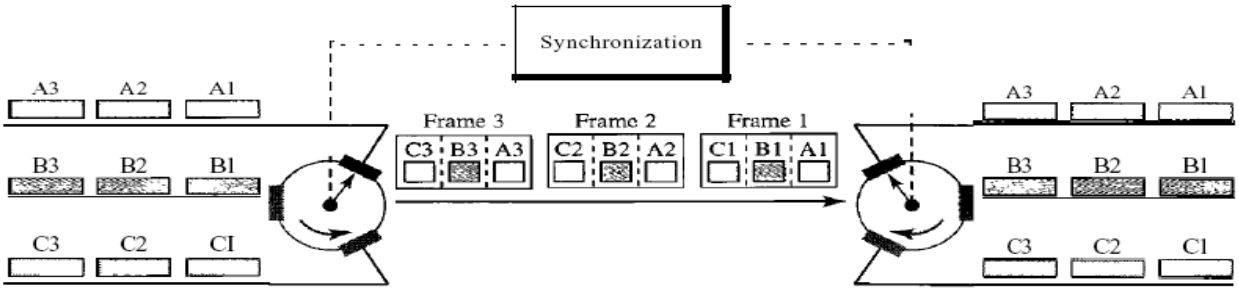


Figure 8 *Interleaving*

- *Empty Slots*
- *Data Rate Management*
    - **Multilevel Multiplexing**
    - **Multiple-Slot Allocation**
    - **Pulse Stuffing**
- *Frame Synchronizing*
- *Digital Signal Service*

2.1.4 Statistical Time-Division Multiplexing

Addressing

Figure a also shows a major difference between slots in synchronous TDM and statistical TDM. An output slot in synchronous TDM is totally occupied by data; in statistical TDM, a slot needs to carry data as well as the address of the destination.

In synchronous TDM, there is no need for addressing; synchronization and preassigned relationships between the inputs and outputs serve as an address. We know, for example, that input 1 always goes to input 2. If the multiplexer and the demultiplexer are synchronized, this is guaranteed. In statistical multiplexing, there is no fixed relationship between the inputs and outputs because there are no preassigned or reserved slots. We need to include the address of the

receiver inside each slot to show where it is to be delivered. The addressing in its simplest form can be *n* bits to define *N* different output lines with *n* =10g2 *N*. For example, for eight different output lines, we need a 3-bit address.

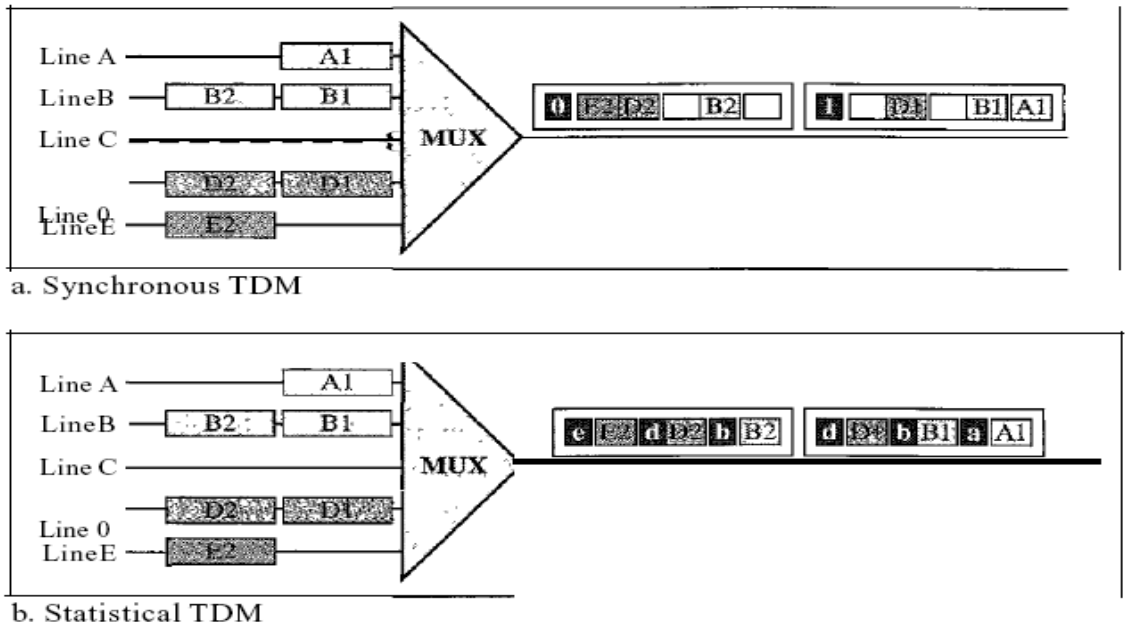


Figure a *TDM slot comparison*

Slot Size

Since a slot carries both data and an address in statistical TDM, the ratio of the data size to address size must be reasonable to make transmission efficient. For example, it would be inefficient to send 1 bit per slot as data when the address is 3 bits. This would mean an overhead of 300 percent. In statistical TDM, a block of data is usually many bytes while the address is just a few bytes.

No Synchronization Bit

There is another difference between synchronous and statistical TDM, but this time it is at the frame level. The frames in statistical TDM need not be synchronized, so we do not need synchronization bits.

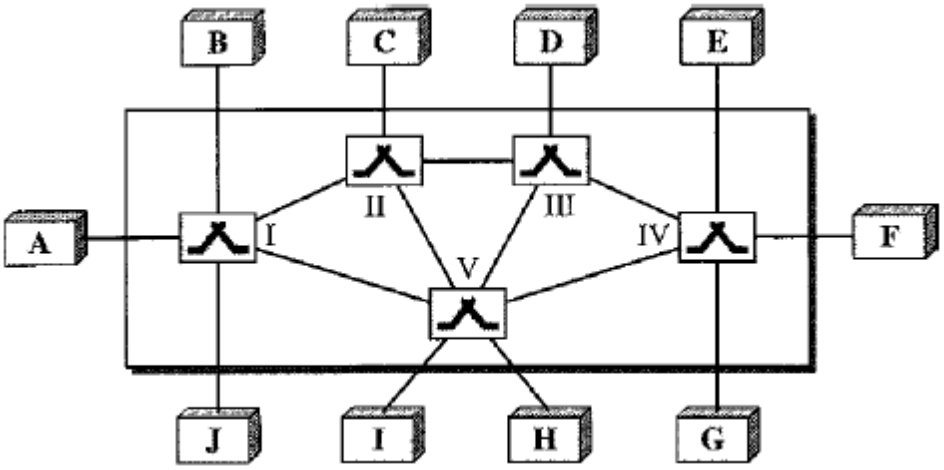
Bandwidth

In statistical TDM, the capacity of the link is normally less than the sum of the capacities of each channel. The designers of statistical TDM define the capacity of the link based on the statistics of the load for each channel. If on average only *x* percent of the input slots are filled, the capacity of the link reflects this. Of course, during peak times, some slots need to wait.

2.2 Switching

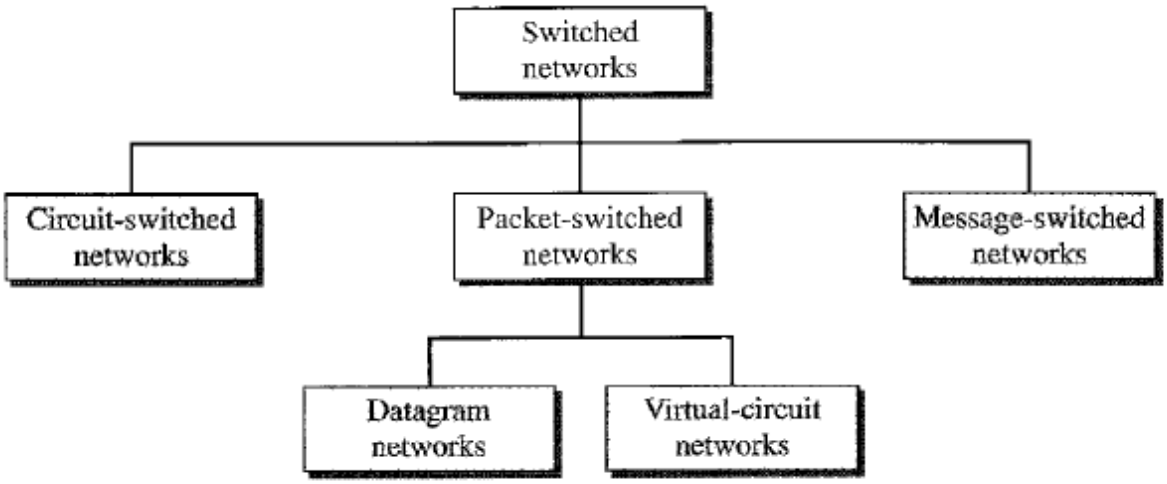
A network is a set of connected devices. Whenever we have multiple devices, we have the problem of how to connect them to make one-to-one communication possible. One solution is to make a point-to-point connection between each pair of devices (a mesh topology) or between a central device and every other device (a star topology). These methods, however, are impractical and wasteful when applied to very large networks. The number and length of the links require too much infrastructure to be cost-efficient, and the majority of those links would be idle most of the time. Other topologies employing multipoint connections, such as a bus, are ruled out because the distances between devices and the total number of devices increase beyond the capacities of the media and equipment.

A better solution is switching. A switched network consists of a series of interlinked nodes, called switches. Switches are devices capable of creating temporary connections between two or more devices linked to the switch. In a switched network, some of these nodes are connected to the end systems (computers or telephones, for example). Others are used only for routing. Figure shows a switched network.



The end systems (communicating devices) are labeled A, B, C, D, and so on, and the switches are labeled I, II, III, IV, and V. Each switch is connected to multiple links.

Taxonomy of switched networks



2.2.1 CIRCUIT-SWITCHED NETWORKS

A circuit-switched network consists of a set of switches connected by physical links. A connection between two stations is a dedicated path made of one or more links. However, each connection uses only one dedicated channel on each link. Each link is normally divided into *n* channels by using FDM or TDM

Figure shows a trivial circuit-switched network with four switches and four links. Each link is divided into *n (n* is 3 in the figure) channels by using FDM or TDM.

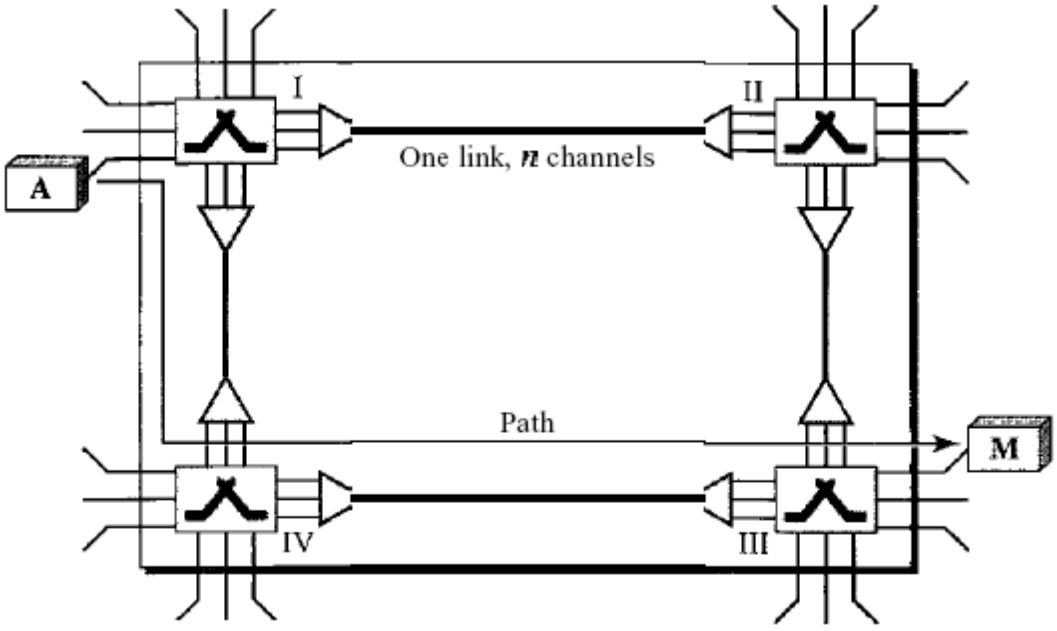


Fig: A trivial circuit-switched network

Three Phases

The actual communication in a circuit-switched network requires three phases: connection setup, data transfer, and connection teardown.

Setup Phase*:*

Before the two parties (or multiple parties in a conference call) can communicate, a dedicated circuit (combination of channels in links) needs to be established. The end systems are normally connected through dedicated lines to the switches, so connection setup means creating dedicated channels between the switches. For example, in Figure, when system A needs to connect to system M, it sends a setup request that includes the address of system M, to switch I. Switch I finds a channel between itself and switch IV that can be dedicated for this purpose. Switch I then sends the request to switch IV, which finds a dedicated channel between itself and switch III. Switch III informs system M of system A's intention at this time.

In the next step to making a connection, an acknowledgment from system M needs to be sent in the opposite direction to system A. Only after system A receives this acknowledgment is the connection established. Note that end-to-end addressing is required for creating a connection between the two end systems. These can be, for example, the addresses of the computers assigned by the administrator in a TDM network, or telephone numbers in an FDM network.

Data Transfer Phase*:*

After the establishment of the dedicated circuit (channels), the two parties can transfer data.

Teardown Phase:

When one of the parties needs to disconnect, a signal is sent to each switch to release the resources.

Efficiency:

It can be argued that circuit-switched networks are not as efficient as the other two types of networks because resources are allocated during the entire duration of the connection. These resources are unavailable to other connections. In a telephone network, people normally terminate the communication when they have finished their conversation. However, in computer networks, a computer can be connected to another computer even if there is no activity for a long time. In this case, allowing resources to be dedicated means that other connections are deprived.

Delay

Although a circuit-switched network normally has low efficiency, the delay in this type of network is minimal. During data transfer the data are not delayed at each switch; the resources are allocated for the duration of the connection. Figure 8.6 shows the idea of delay in a circuit-switched network when only two switches are involved. As Figure shows, there is no waiting time at each switch. The total delay is due to the time needed to create the connection, transfer data, and disconnect the circuit.

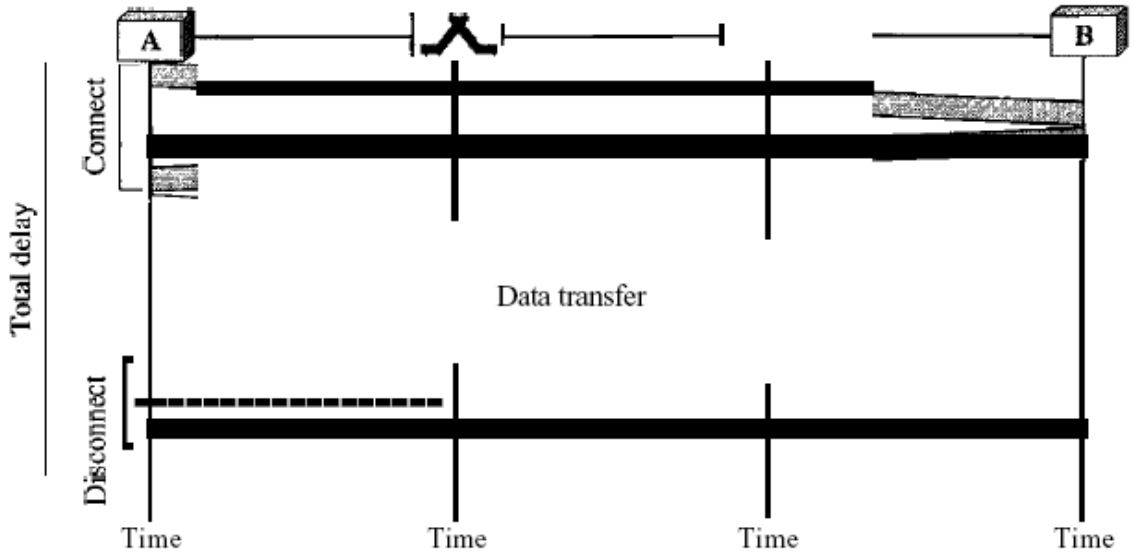


Fig: Delay in a circuit-switched network

The delay caused by the setup is the sum of four parts: the propagation time of the source computer request (slope of the first gray box), the request signal transfer time (height of the first gray box), the propagation time of the acknowledgment from the destination computer (slope of the second gray box), and the signal transfer time of the acknowledgment (height of the second gray box). The delay due to data transfer is the sum of two parts: the propagation time (slope of the colored box) and data transfer time (height of the colored box), which can be very long. The third box shows the time needed to tear down the circuit. We have shown the case in which the receiver requests disconnection, which creates the maximum delay.