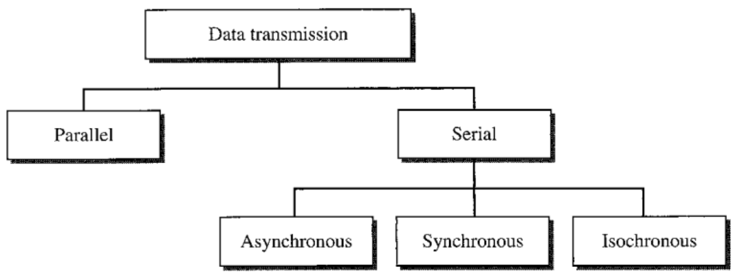
Module 2:PART II

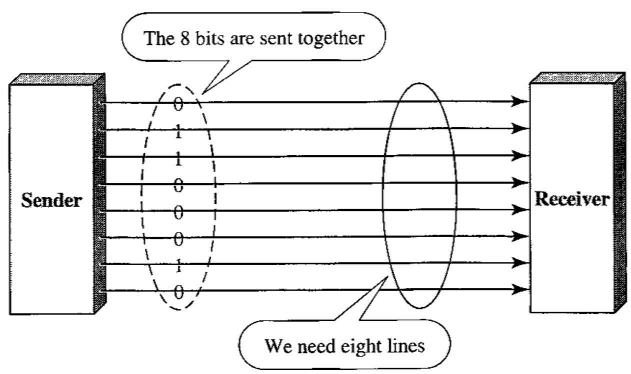
Transmission Modes:



PARALLEL TRANSMISSION

Binary data, consisting of ls and Os, may be organized into groups of n bits each. Computers produce and consume data in groups of bits. By grouping, we can send data n bits at a time instead of 1. This is called **parallel transmission**.

The mechanism for parallel transmission is a conceptually simple one: Use n wires to send n bits at one time. That way each bit has its own wire, and all n bits of one group can be transmitted with each clock tick from one device to another. The following figure shows how parallel transmission works for n = 8. Typically, the eight wires are bundled in a cable with a connector at each end.

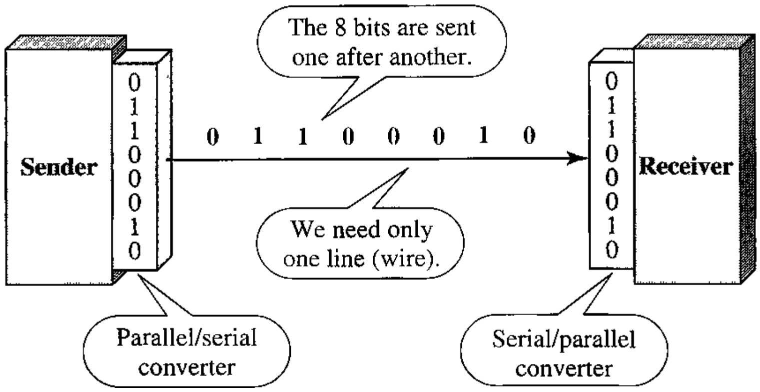


The advantage of parallel transmission is speed. All else being equal, parallel transmission can increase the transfer speed by a factor of n over serial transmission.

But there is a significant disadvantage: cost. Parallel transmission requires n communication lines just to transmit the data stream. Because this is expensive, parallel transmission is usually limited to short distances.

SERIAL TRANSMISSION

In **serial transmission** one bit follows another, so we need only one communication channel rather than n to transmit data between two communicating devices.



The advantage of serial over parallel transmission is that with only one communication channel, serial transmission reduces the cost of transmission over parallel by roughly a factor of n.

Since communication within devices is parallel, conversion devices are required at the interface between the sender and the line (parallel-to-serial) and between the line and the receiver (serial-to-parallel).

Serial transmission occurs in one of three ways: asynchronous, synchronous, and isochronous.

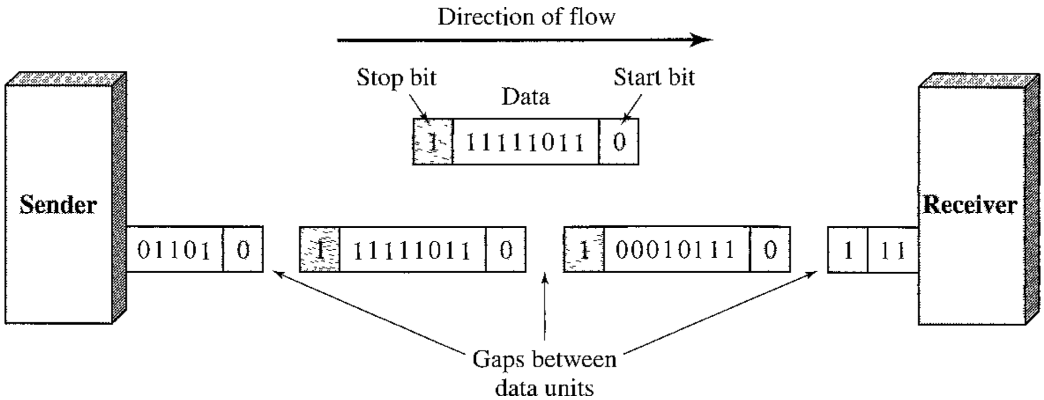
Asynchronous Transmission

Asynchronous transmission is so named because the timing of a signal is unimportant. Instead, information is received and translated by agreed upon patterns. As long as those patterns are followed, the receiving device can retrieve the information without regard to the rhythm in which it is sent. Patterns are based on grouping the bit stream into bytes. Each group, usually 8 bits, is sent along the link as a unit. The sending system handles each group independently, relaying it to the link whenever ready, without regard to a timer.

Without synchronization, the receiver cannot use timing to predict when the next group will arrive. To alert the receiver to the arrival of a new group, therefore, an extra bit is added to the beginning of each byte. This bit, usually a 0, is called the start bit. To let the receiver know that the byte is finished, 1 or more additional bits are appended to the end of the byte. These bits,usually 1 s, are called stop bits. By this method, each byte is increased in size to at least 10 bits, of which 8 bits is information and 2 bits or more are signals to the receiver. In addition, the transmission of each byte may then be followed by a gap of varying duration. This gap can be represented either by an idle channel or by a stream of additional stop bits.

The start and stop bits and the gap alert the receiver to the beginning and end of each byte and allow it to synchronize with the data stream. This mechanism is called asynchronousbecause, at the byte level, the sender and receiver do not have to be synchronized. But within each byte, the receiver must still be synchronized with the incoming bit stream. That is, some synchronization is required, but only for the duration of a single byte. The receiving device resynchronizes at the onset of each new byte. When the receiver detects a start bit, it sets a timer and begins counting bits as they come in. After n bits, the receiver looks for a stop bit. As soon as it detects the stop bit, it waits until it detects the next start bit.

The figure is a schematic illustration of asynchronous transmission. In this example, the start bits are 0s, the stop bits are ls, and the gap is represented by an idle line rather than by additional stop bits.

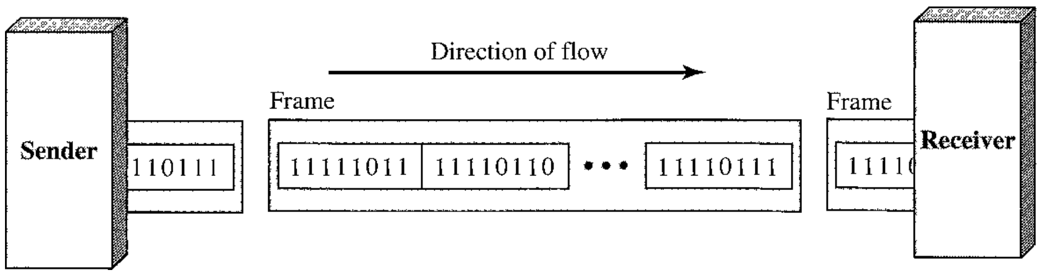


The addition of stop and start bits and the insertion of gaps into the bit stream make asynchronous transmission slower than forms of transmission that can operate without theaddition of control information. But it is cheap and effective, two advantages that make it an attractive choice for situations such as low-speed communication.

Synchronous Transmission

In synchronous transmission, the bit stream is combined into longer "frames," which may contain multiple bytes. Each byte, however, is introduced onto the transmission link without a gap between it and the next one. It is left to the receiver to separate the bit stream into bytes for decoding purposes. In other words, data are transmitted as an unbroken string of 1 s and 0s, and the receiver separates that string into the bytes, or characters, it needs to reconstruct the information.

The figure gives a schematic illustration of synchronous transmission. The sender puts its data onto the line as one long string. If the sender wishes to send data in separate bursts, the gaps between bursts must be filled with a special sequence of 0s and ls that means idle. The receiver counts the bits as they arrive and groups them in 8-bit units.



Without gaps and start and stop bits, there is no built-in mechanism to help the receiving device adjust its bit synchronization midstream. Timing becomes very important, therefore, because the accuracy of the received information is completely dependent on the ability of the receiving device to keep an accurate count of the bits as they come in.

The advantage of synchronous transmission is speed. With no extra bits or gaps to introduce at the sending end and remove at the receiving end, and, by extension, with fewer bits to move across the link, synchronous transmission is faster than asynchronous transmission. For this reason, it is more useful for high-speed applications such as the transmission of data from one computer to another.

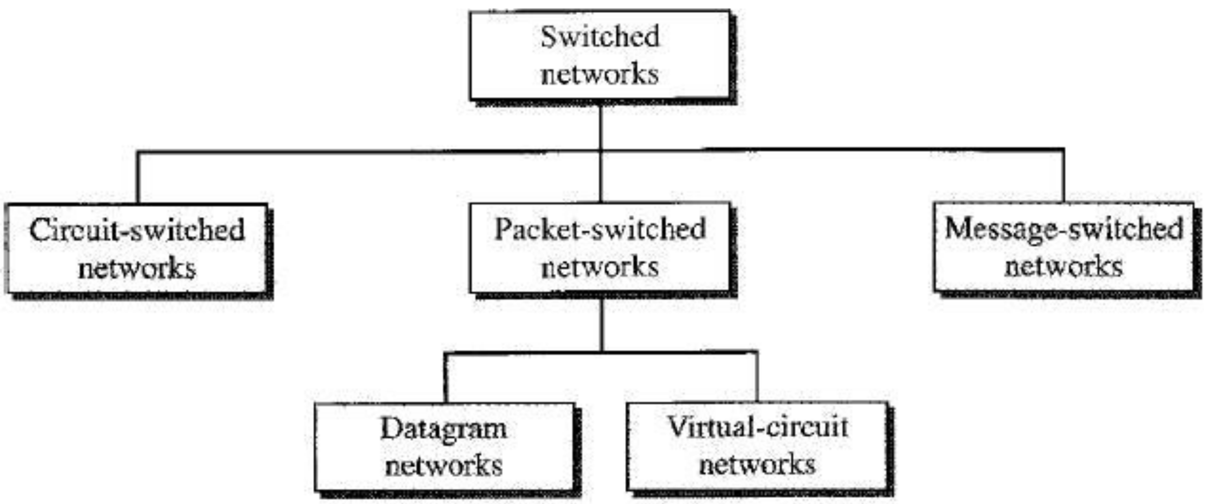
Isochronous

In real-time audio and video, in which uneven delays between frames are not acceptable, synchronous transmission fails. For example, TV images are broadcast at the rate of 30 images per second; they must be viewed at the same rate. If each image is sent by using one or more flames, there should be no delays between frames. For this type of application, synchronization between characters is not enough; the entire stream of bits must be synchronized. The isochronous transmission guarantees that the data arrive at a fixed rate.

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.

Traditionally, three methods of switching have been important: **circuit switching**, **packet switching**, and **message switching**. The first two are commonly used today. The third has been phased out in general communications but still has networking applications. We can then divide today's networks into three broad categories: circuit-switched networks, packet-switched networks, and message-switched. Packet- switched networks can further be divided into two subcategories-virtual-circuit networks and datagram networks as shown in Figure.



Circuit switching and Telephone Network

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.

circuit switching takes place at the physical layer.

Before starting communication, the stations must make a reservation for the resources to be used during the communication. These resources, such as channels (bandwidth in FDM and time slots in TDM), switch buffers, switch processing time, and switch input/output ports, must remain dedicated during the entire duration of data transfer until the teardown phase.

Data transferred between the two stations are not packetized (physical layer transfer of the signal). The data are a continuous flow sent by the source station and received by the destination station, although there may be periods of silence.

There is no addressing involved during data transfer. The switches route the data based on their occupied band (FDM) or time slot (TDM). Of course, there is end-to-end addressing used during the setup phase.

Three Phases

The actual communication in a circuit-switched network requires three phases: connection setup, datatransfer, 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.

Data Transfer Phase

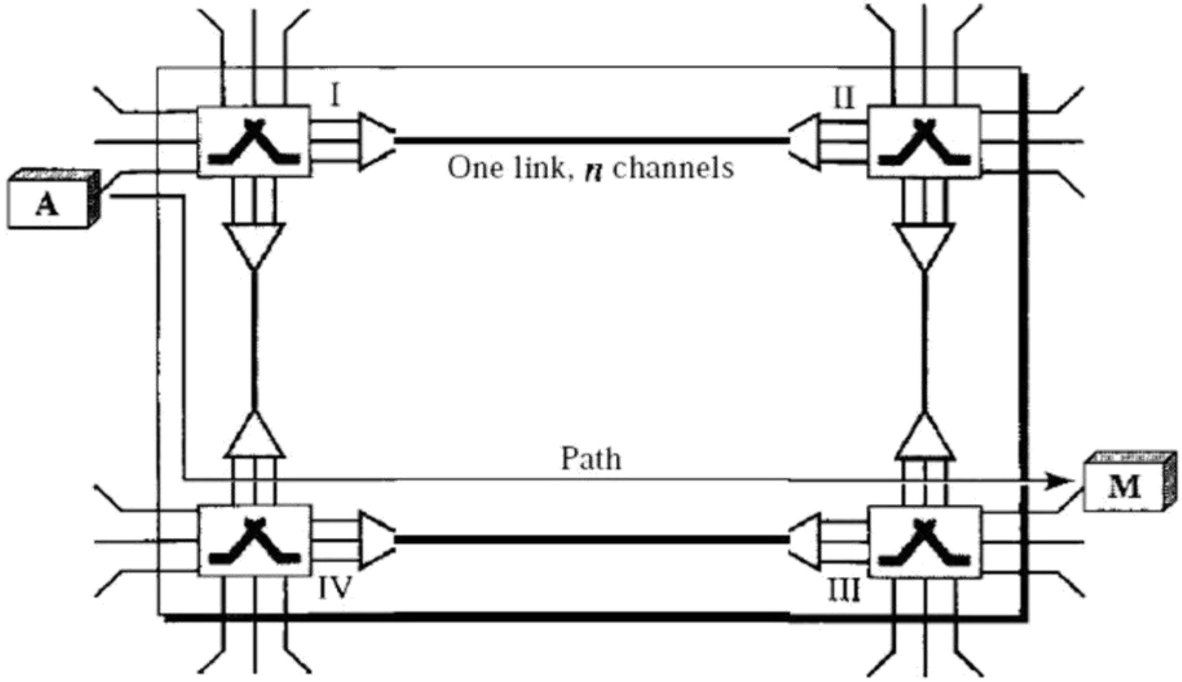
After the establishment of the dedicated circuit (channels), the two parties can transfer data.

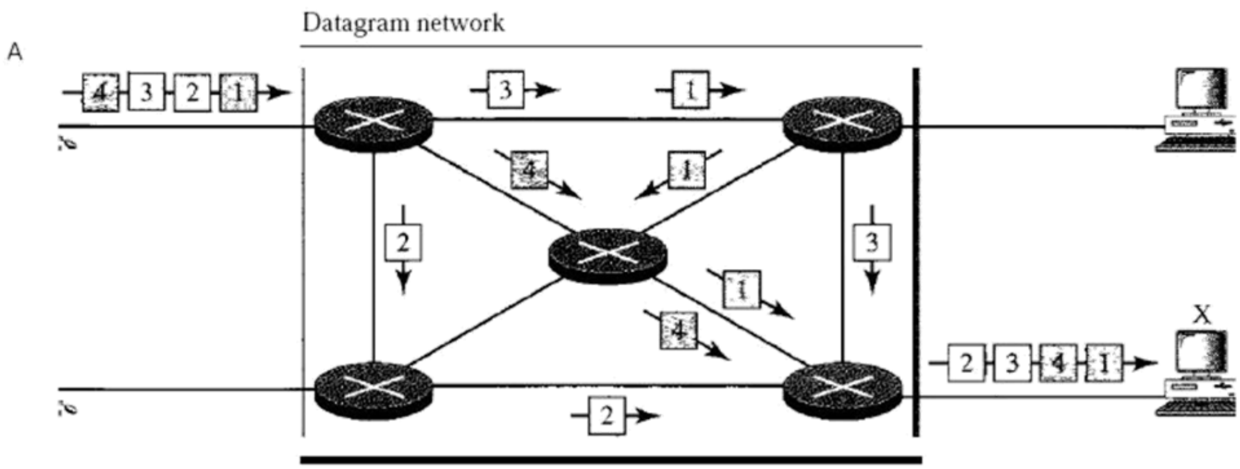
Teardown Phase

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

Circuit-Switched Technology in Telephone Networks

The telephone companies have previously chosen the circuit switched approach to switching in the physical layer; today the tendency is moving toward other switching techniques. For example, the telephone number is used as the global address, and a signaling system (called SS7)is used for the setupand teardown phases.





DATAGRAM NETWORKS

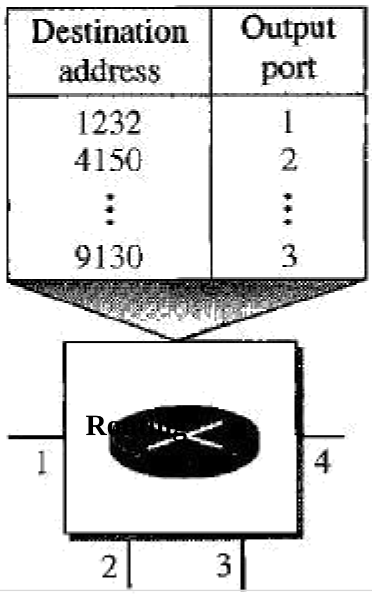
In a datagram network, each packet is treated independently of all others. Even if a packet is part of a multipacket transmission, the network treats it as though it existed alone. Packets in this approach are referred to as datagrams. Datagram switching is normally done at the networklayer.

The datagram networks are sometimes referred to as connectionless networks. The term connectionless here means that the switch (packet switch) does not keep information about the connection state. There are no setup or teardown phases. Each packet is treated the same by a switch regardless of its source or destination.

Routing Table

If there are no setup or teardown phases, how are the packets routed to their destinations in a datagram network? In this type of network, each switch (or packet switch) has a routing table which is based on the destination address. The routing tables are dynamic and are updated periodically. The destination addresses and the corresponding forwarding output ports are recorded in the tables. This is different from the table of a circuit switched network in which

phase is completed and deleted when the teardown phase is over.



Destination

each entry is created when the setup

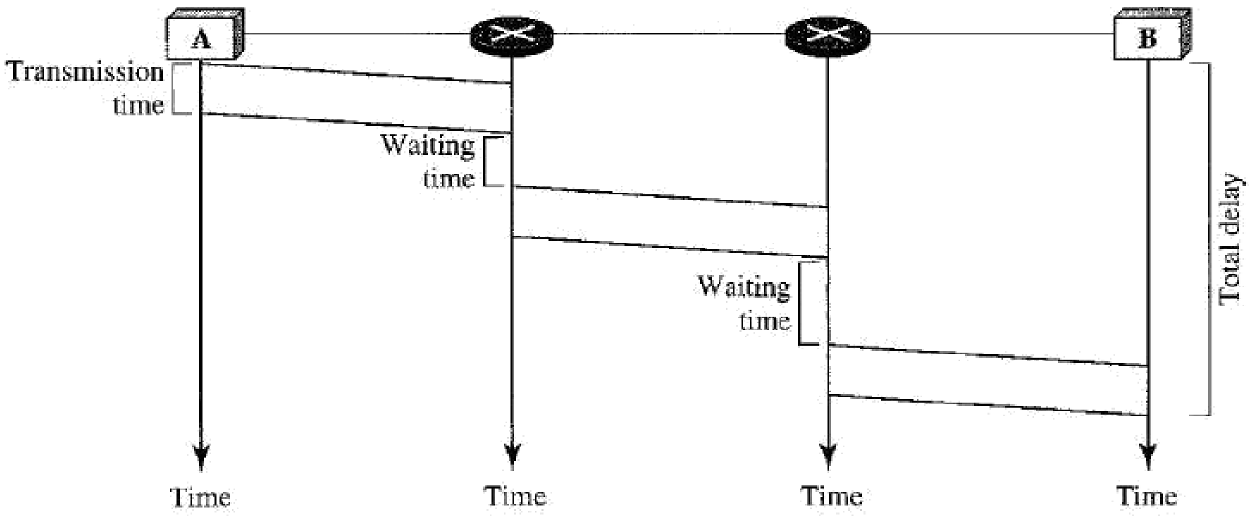
Every packet in a datagram network carries a header that contains, among other information, the destination address of the packet. When the switch receives the packet, this destination address is examined; the routing table is consulted to find the corresponding port through which the packet should be forwarded. This address, unlike the address in a virtual-circuit-switched network, remains the same during the entire journey of the packet.

Efficiency

The efficiency of a datagram network is better than that of a circuit-switched network; resources are allocated only when there are packets to be transferred. If a source sends a packet and there is a delay of a few minutes before another packet can be sent, the resources can be reallocated during these minutes for other packets from other sources.

Delay

There may be greater delay in a datagram network than in a virtual-circuit network. Although there are no setup and teardown phases, each packet may experience a wait at a switch before it is forwarded.



Packet Switching

Packet switching is a digital networking communications method that groups all transmitted data – regardless of content, type, or structure – into suitably sized blocks, called packets. Packet switching features delivery of variable-bit-rate data streams (sequences of packets) over a shared network. When

traversing network adapters, switches, routers and other network nodes, packets are buffered and queued, resulting in variable delay and throughput depending on the traffic load in the network.

Packet switching contrasts with another principal networking paradigm, circuit switching, a method which sets up a limited number of dedicated connections of constant bit rate and constant delay between nodes for exclusive use during the communication session. In case of traffic fees (as opposed to flat rate), for example in cellular communication services, circuit switching is characterized bya fee per time unit of connection time, even when no data is transferred, while packet switching is characterized by a fee per unit of information.

Two major packet switching modes exist:

(1) Connectionless packet switching, also known as datagram switching, and

(2) Connection-oriented packet switching, also known as virtual circuit switching.

In the first case each packet includes complete addressing or routing information. The packets arerouted individually, sometimes resulting in different paths and out-of-order delivery.

In the second case a connection is defined and preallocated in each involved nodeduring a connectionphase before any packet is transferred.

Virtual-Circuit Identifier:

The identifier that is actually used for data transfer is called the virtual-circuit identifier (VCI). A VCI, unlike a global address, is a small number that has only switch scope. It is used by a frame between two switches. When a frame arrives at a switch, it has a VCI; when it leaves, it has a different VCI. The following figure show how the VCI in a data frame changes from one switch to another. Note that a VCI does not need to be a large number since each switch can use its own unique set of VCIs.

Switched network, a source and destination need to go through three phases in a virtual-circuit network: setup, data transfer, and teardown.

Three phases:

Setup phase, the source and destination use their global addresses to help switches make table entries for the connection.

In the teardown phase, the source and destination inform the switches to delete the corresponding entry.

Data transfer occurs between these two phases.

Data Transfer Phase

To transfer a frame from a source to its destination, all switches need to have a table entry for this virtual circuit. The table, in its simplest form, has four columns. This means that the switch holds four pieces of information for each virtual circuit that is already set up. We show later how the switches make their table entries, but for the moment we assume that each switch has a table with entries for all active virtual circuits.

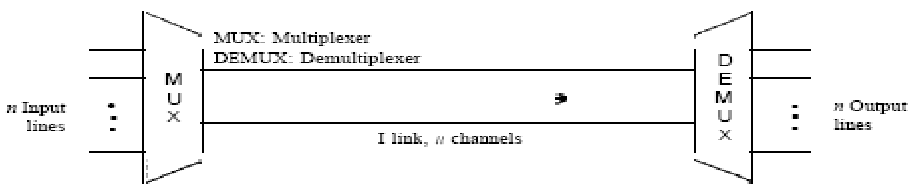
The following figure shows a frame arriving at port 1 with a VCI of 14. When the frame arrives, the switch looks in its table to find port 1 and a VCI of 14. When it is found, the switch knows to change the VCI to 22 and send out the frame from port 3.

Teardown Phase: In this phase, source A, after sending all frames to B, sends a special frame called a teardown request. Destination B responds with a teardown confirmation frame. All switches delete the corresponding entry from their table

MULTIPLEXING

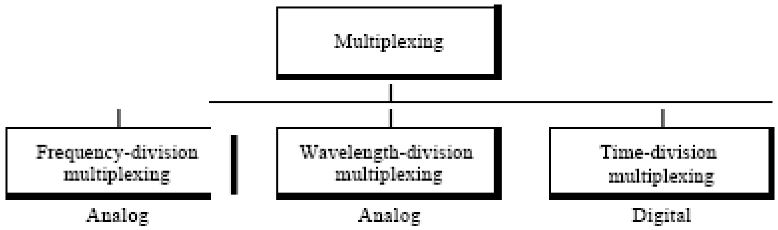
Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic. We can accommodate this increase by continuing to add individual links each time a new channel is needed; or we can install higher-bandwidth links and use each to carry multiple signals. As described in Chapter 7, today's technology includes high-bandwidth media such as optical fiber and terrestrial and satellite microwaves. Each has a bandwidth far in excess of that needed for the average transmission signal. If the bandwidth of a link is greater than the bandwidth needs of the devices connected to it, the bandwidth is wasted. An efficient system maximizes the utilization of all resources; bandwidth is one of the most precious resources we have in data communications. In a multiplexed system, *n* lines share the bandwidth of one link. Figure 6.1 shows the basic format ofa multiplexed system. The lines on the left direct their transmission streams to a multiplexer (MUX), which combines them into a single stream (many-toone). At the receiving end, that stream is fed into a demultiplexer (DEMUX), which separates the stream back into its component transmissions (one-to-many) and directs them to their corresponding lines. In the figure, the word link refers to the physical path. The word channel refers to the portion of a link that carries a transmission between a given pair of lines. One link can have many *(n)* channels.

Figure 6.1 *Dividing a link into channels*



There are three basic multiplexing techniques: frequency-division multiplexing, wavelength-division multiplexing, and time-division multiplexing. The first two are techniques designed for analog signals, the third, for digital signals (see Figure 6.2).

Figure 6.2 *Categories of multiplexing*

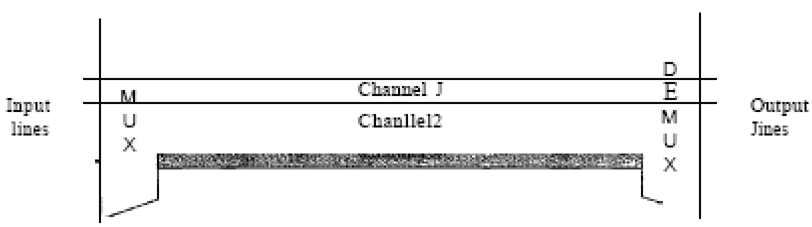


Although some textbooks consider *carrier division multiple access* (COMA) as a fourth multiplexing category, we discuss COMA as an access method

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 6.3 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.

Figure 6.3 *Frequency-division multiplexing*



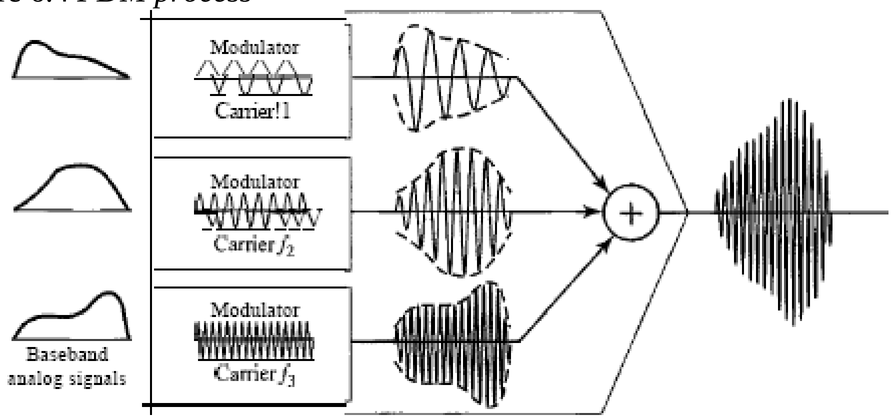
We consider FDM to be an analog multiplexing technique; however, this does not mean that FDM cannot be used to combine sources sending digital signals. A digital signal can be converted to an analog signal before FDM is used to multiplex them.

FDM is an analog multiplexing technique that combines analog signals.

Multiplexing Process

Figure 6.4 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.

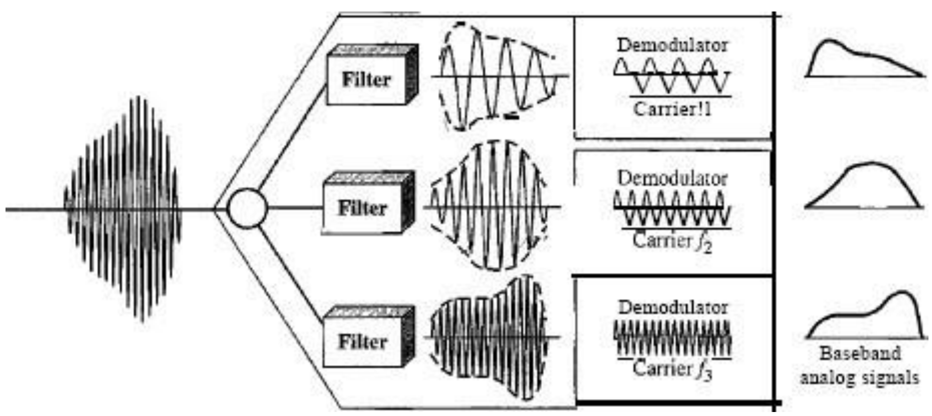
Figure 6.4 *FDM process*



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 6.5 is a conceptual illustration of demultiplexing process.

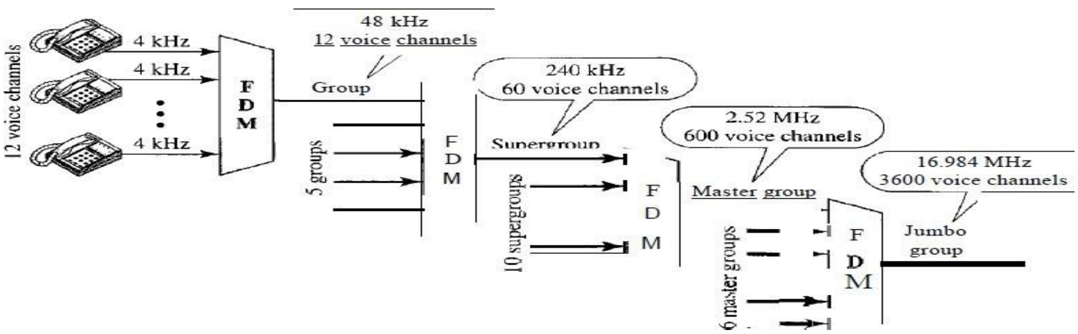
Figure 6.5 *FDM demultiplexing example*



The Analog Carrier System

To maximize the efficiency of their infrastructure, telephone companies have traditionally multiplexed signals from lower-bandwidth lines onto higher-bandwidth lines. In this way, many switched or leased lines can be combined into fewer but bigger channels. For analog lines, FDM is used. One of these hierarchical systems used by AT&T is made up of groups, super groups, master groups, and jumbo groups

Figure 6.9 *Analog hierarchy*



In this analog hierarchy, 12 voice channels are multiplexed onto a higher-bandwidth line to create a group. A group has 48 kHz of bandwidth and supports 12 voice channels. At the next level, up to five groups can be multiplexed to create a composite signal called a supergroup. A supergroup has a bandwidth of 240 kHz and supports up to 60 voice channels. Supergroups can be made up of either five groups or 60 independent voice channels. At the next level, 10 supergroups are multiplexed to create a master group. A master group must have 2.40 MHz of bandwidth, but the need for guard bands between the supergroups increases the necessarybandwidth to 2.52 MHz. Master groups support up to 600 voice channels. Finally, six master groups can be combined into a jumbo group. A jumbo group must have 15.12 MHz (6 x 2.52 MHz) but is augmented to 16.984 MHz to allow for guard bands between the master groups.

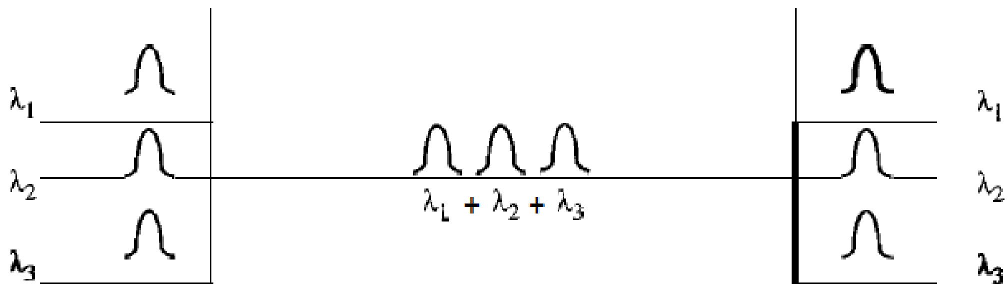
Other Applications of FDM

A very common application of FDM is AM and FM radio broadcasting. Radio uses the air as the transmission medium. A special band from 530 to 1700 kHz is assigned to AM radio. All radio stations need to share this band. As discussed in Chapter 5, each AM station needs 10kHz of bandwidth. Each station uses a different carrier frequency, which means it is shifting its signal and multiplexing. The signal that goes to the air is a combination of signals. A receiver receives all these signals, but filters (by tuning) only the one which is desired. Without multiplexing, onlyone AM station could broadcast to the common link, the air. However, we need to know that there is physical multiplexer or demultiplexer here. As we will see in Chapter 12 multiplexing is done at the data link layer. The situation is similar in FM broadcasting. However, FM has awider band of 88 to 108 MHz because each station needs a bandwidth of 200 kHz. Another common use of FDM is in television broadcasting. Each TV channel has its own bandwidth of 6 MHz. The first generation of cellular telephones (still in operation) also uses FDM. Each user is assigned two 30-kHz channels, one for sending voice and the other for receiving. The voice signal, which has a bandwidth of 3 kHz (from 300 to 3300 Hz), is modulated by using FM. Remember that an FM signal has a bandwidth 10 times that of the modulating signal, which means each channel has 30 kHz (10 x 3) of **bandwidth. Therefore, each** user is given, by the base station, a 60-kHz bandwidth in a range available at the time of the call.

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 6.10 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

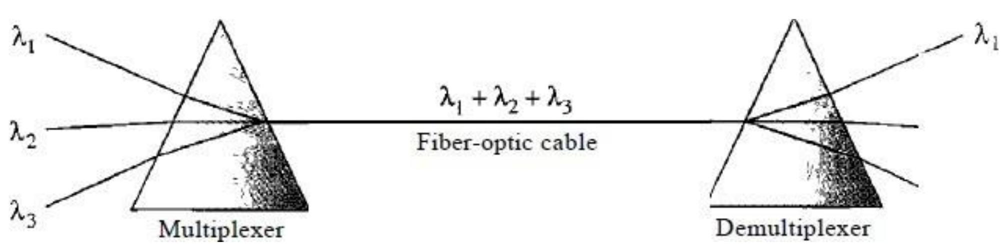
Figure 6.10 ***Wavelength-division multiplexing***



WDM is an analog multiplexing technique to combine optical signals.

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 6.11 shows the concept.

Figure 6.11 *Prisms in wavelength-division multiplexing and demultiplexing*

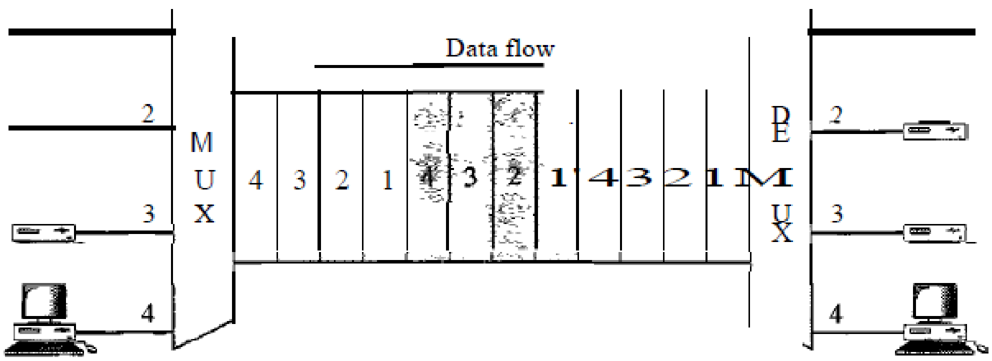


One application of WDM is the SONET network in which multiple optical fiber lines are multiplexed and demultiplexed. We discuss SONET in Chapter 17. A new method, called dense WDM (DWDM), can multiplex a very large number of channels by spacing channels very close to one another. It achieves even greater efficiency

Synchronous Time-Division Multiplexing

Time-division multiplexing (TDM) is a digital process that allows severalconnections 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 6.12 gives a conceptual view of TDM. Note that the same link isused 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 6.12 *TDM*



Note that in Figure 6.12 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

TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

sampled, changed to digital data, and then multiplexed by using TDM.

We can divide TDM into two different schemes: synchronous and statistical. We firstdiscuss synchronous TDM and then show how statistical TDM differs. In synchronous TDM, each input connection has an allotment in the output even if it is not sending data.