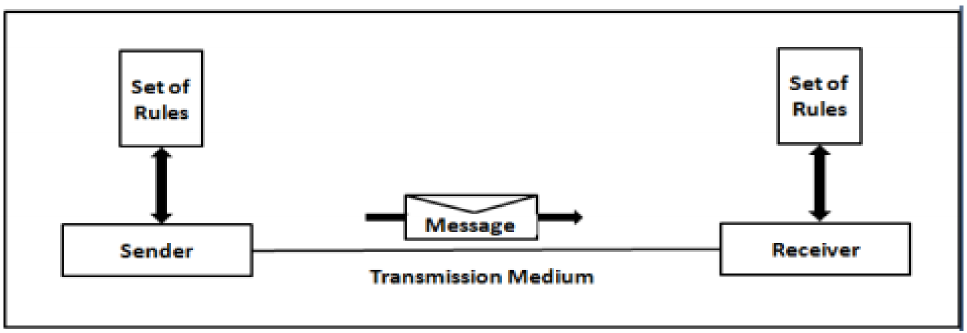
Unit 1
Introduction

Components of data communication:

The term telecommunication means communication at a distance. The word data refers to information presented in whatever form is agreed upon by the parties creating and using the data.

Data Communication are the exchange of data between two devices between two devices via some form of transmission medium such as a wire cable. For data communications to occur, the communicating devices must be part of a communication system made of a combination of hardware and software.

Five components of data communication

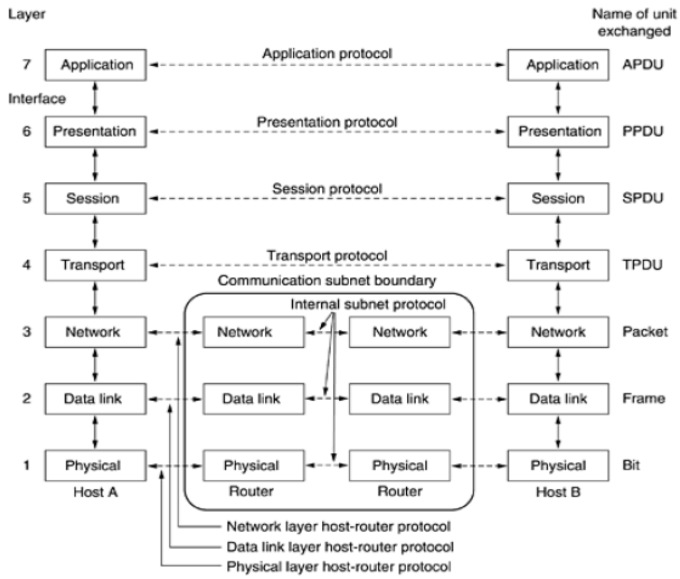


1. **Message -** It is the information to be communicated. Popular forms of information include text, pictures, audio, video etc.
2. **Sender -** It is the device which sends the data messages. It can be a computer, workstation, telephone handset etc.
3. **Receiver -** It is the device which receives the data messages. It can be a computer, workstation, telephone handset etc.
4. **Transmission Medium -** It is the physical path by which a message travels from sender to receiver. Some examples include twisted-pair wire, coaxial cable, radio waves etc.
5. **Protocol -** It is a set of rules that governs the data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating.

The OSI Reference Model

The OSI model (minus the physical medium) is shown in below Figure. 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). Themodel 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. We will just call it the OSI model for short.



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.

Below we will discuss each layer of the model in turn, starting at the bottom layer. Note that the OSI model itself is not a network architecture because it does not specify the exact services and protocols to be used in each layer. It just tells what each layer should do. However, ISO has also produced standards for all the layers, although these are not part of the reference model itself. Each one has been published as a separate international standard.

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 a0 bit. Typical questions here are how many volts should be used to represent a 1 and how many for a 0, how many nanoseconds a bit lasts, whether transmission may proceed simultaneously in both directions, how the initial connection is established and how it is torn down when both sides are finished, and how many pins the network connector has and what each pin is used for. The design issues here largely deal with mechanical, electrical, and timing interfaces, and the physical transmission medium, which lies below the physical layer.

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 transmit 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.

Broadcast networks have an additional issue in the data link layer: how to control access to the shared channel. A special sub layer of the data link layer, the medium access control sublayer, deals with this problem.

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 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

Unlike lower layers, which are mostly concerned with moving bits around, 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.

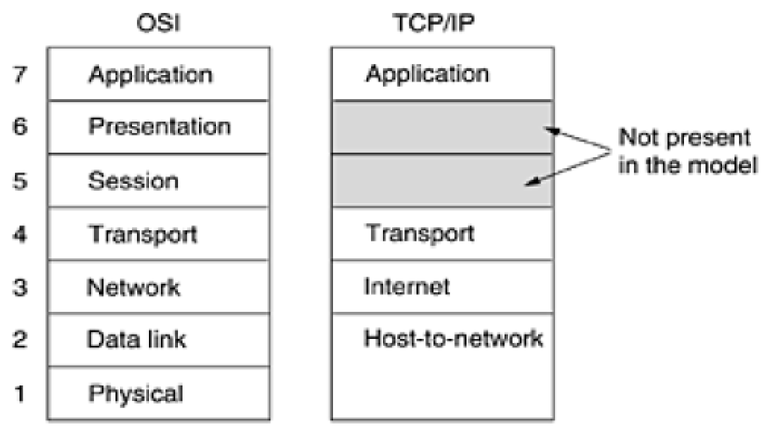
The Application Layer

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.

The TCP/IP Reference Model

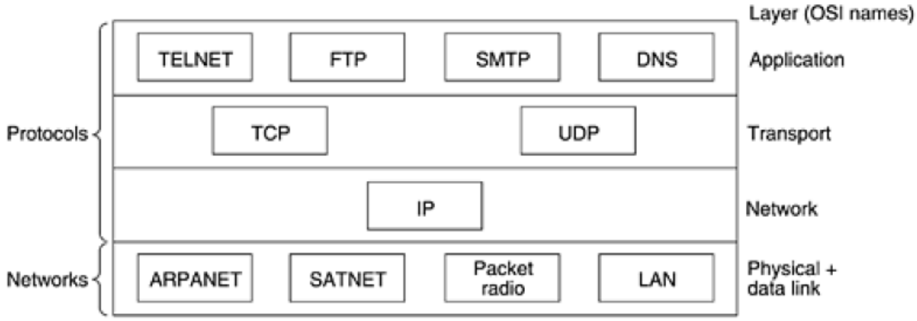
The Internet Layer

All these requirements led to the choice of a packet-switching network based on a connectionless internet work 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 them 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.



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.



The Application Layer

The TCP/IP model does not have session or presentation layers. No need for them was perceived, so they were not included. Experience with the OSI model has proven this view correct: they are of little use to most applications.

A 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. In this section we will focus on the key differences between the two reference models. It is important to note that we are comparing the reference models here, not the corresponding protocol stacks. The protocols themselves will be discussed later. For an entire book comparing and contrasting TCP/IP and OSI, see (Piscitello and Chapin, 1993).

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.

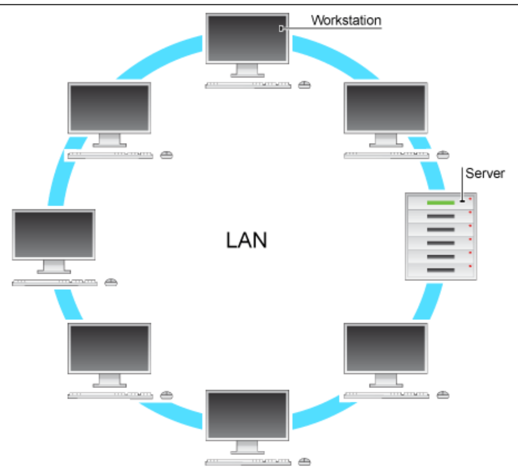
Finally, the peer protocols used in a layer are the layer's own business. It can use any protocols it wants to, as long as it gets the job done (i.e., provides the offered services). It can also change them at will without affecting software in higher layers.

These ideas fit very nicely with modern ideas about object-oriented programming. An object, like a layer, has a set of methods (operations) that processes outside the object can invoke. The semantics of these methods define the set of services that the object offers. The methods' parameters and results from the object's interface. The code internal to the object is its protocol and is not visible or of any concern outside the object.

Types of networks:

Local Area Network

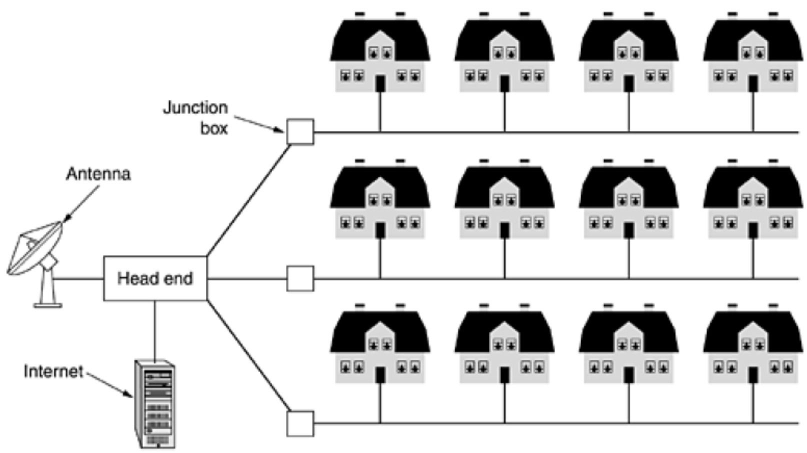
Local area network (**LAN**) is a computer network that interconnects computers within a limited area such as a residence, school, laboratory, or office building. A local area network is contrasted in principle to a wide area network (WAN), which covers a larger geographic distance and may involve leased telecommunication circuits, while the media for LANs are locally managed.
Ethernet over twisted pair cabling and Wi-Fi are the two most common transmission technologies in use for local area networks. Historical technologies include ARCNET, Token Ring, and AppleTalk.



Metropolitan Area Networks

A metropolitan area network, or MAN, covers a city. The best-known example of a MAN is the cable television network available in many cities. This system grew from earlier community antenna systems used in areas with poor over-the-air television reception. In these early systems, a large antenna was placed on top of a nearby hill and signal was then piped to the subscribers' houses.

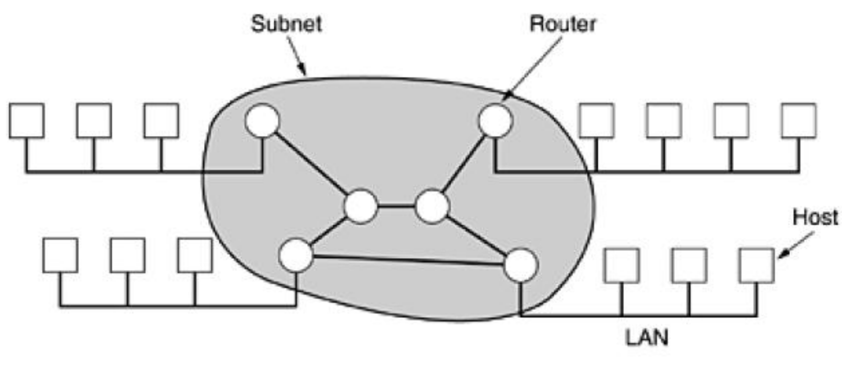
At first, these were locally-designed, ad hoc systems. Then companies began jumping into the business, getting contracts from city governments to wire up an entire city. The next step was television programming and even entire channels designed for cable only. Often these channels were highly specialized, such as all news, all sports, all cooking, all gardening, and so on. But from their inception until the late 1990s, they were intended for television reception only.

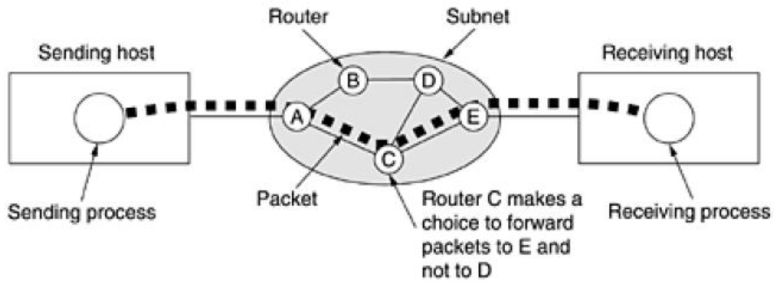


Wide Area Networks

A wide area network, or WAN, spans a large geographical area, often a country or continent. It contains a collection of machines intended for running user (i.e., application) programs. We will follow traditional usage and call these machines hosts. The hosts are connected by a communication subnet, or just subnet for short. The hosts are owned by the customers (e.g., people's personal computers), whereas the communication subnet is typically owned and operated by a telephone company or Internet service provider. The job of the subnet is to carry messages from host to host, just as the telephone system carries words from speaker to listener. Separation of the pure communication aspects of the network (the subnet) from the application aspects (the hosts), greatly simplifies the complete network design.

In most wide area networks, the subnet consists of two distinct components: transmission lines and switching elements. Transmission lines move bits between machines. They can be made of copper wire, optical fiber, or even radio links. Switching elements are specialized computers that connect three or more transmission lines. When data arrive on an incoming line, the switching element must choose an outgoing line on which to forward them. These switching computers have been called by various names in the past; the name router is now most commonly used. Unfortunately, some people pronounce it "rooter" and others have it rhyme with "doubter." Determining the correct pronunciation will be left as an exercise for the reader.





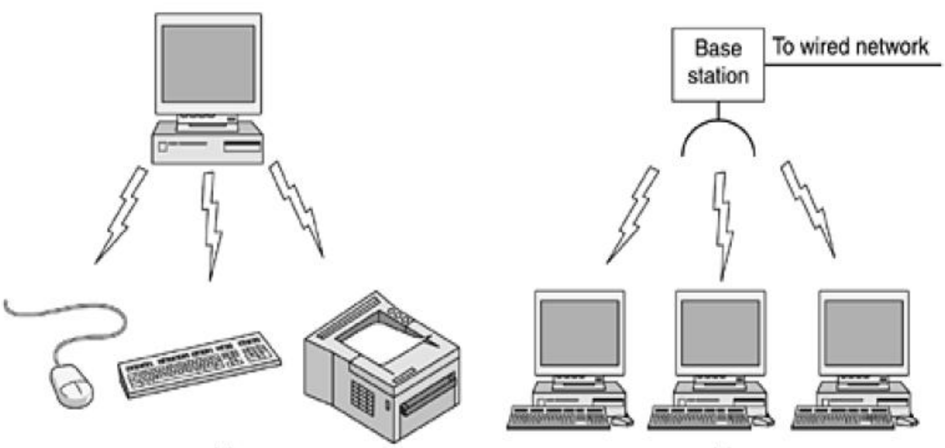
Wireless Networks

Digital wireless communication is not a new idea. As early as 1901, the Italian physicist Guglielmo Marconi demonstrated a ship-to-shore wireless telegraph, using Morse Code (dots and dashes are binary, after all). Modern digital wireless systems have better performance, but the basic idea is the same.

To a first approximation, wireless networks can be divided into three main categories:

1. System interconnection.
2. Wireless LANs.
3. Wireless WANs.

System interconnection is all about interconnecting the components of a computer using short-range radio. Almost every computer has a monitor, keyboard, mouse, and printer connected to the main unit by cables. So many new users have a hard time plugging all the cables into the right little holes (even though they are usually color coded) that most computer vendors offer the option of sending a technician to the user's home to do it. Consequently, some companies got together to design a short-range wireless network called Bluetooth to connect these components without wires. Bluetooth also allows digital cameras, headsets, scanners, and other devices to connect to a computer by merely being brought within range. No cables, no driver installation, just put them down, turn them on, and they work. For many people, this ease of operation is a big plus.



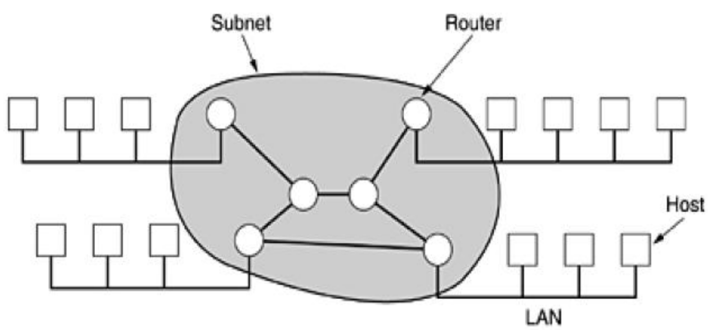
(a) (b)

Internetworks

Many networks exist in the world, often with different hardware and software. People connected to one network often want to communicate with people attached to a different one. The fulfillment of this desire requires that different, and frequently incompatible networks, be connected, sometimes by means of machines called gateways to make the connection and provide the necessary translation, both in terms of hardware and software. A collection of interconnected networks is called an internetwork or internet. These terms will be used in a generic sense, in contrast to the worldwide Internet (which is one specific internet), which we will always capitalize.

A common form of internet is a collection of LANs connected by a WAN. In fact, if we were to replace the label "subnet" in Fig. 1-9 by "WAN," nothing else in the figure would have to change. The only real technical distinction between a subnet and a WAN in this case is whether hosts are present. If the system within the gray area contains only routers, it is a subnet; if it contains both routers and hosts, it is a WAN. The real differences relate to ownership and use. Subnets, networks, and internetworks are often confused. Subnet makes the most sense in the context of a wide area network, where it refers to the collection of routers and communication lines owned by the network operator. As an analogy, the telephone system consists of telephone switching offices connected to one another by high-speed lines, and to houses and businesses by low-speed lines. These lines and equipment, owned and managed by the telephone company, form the subnet of the telephone system. The telephones themselves (the hosts in this analogy) are not part of the subnet. The combination of a subnet and its hosts forms a network. In the case of a LAN, the cable and the hosts form the network. There really is no subnet.

An internetwork is formed when distinct networks are interconnected. In our view, connecting a LAN and a WAN or connecting two LANs forms an internetwork, but there is little agreement in the industry over terminology in this area. One rule of thumb is that if different organizations paid to construct different parts of the network and each maintains its part, we have an internetwork rather than a single network. Also, if the underlying technology is different in different parts (e.g., broadcast versus point-to-point), we probably have two networks.



Data communication fundamentals

Analog vs. Digital Data

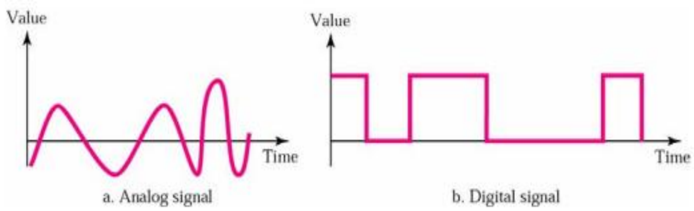
• analog data – representation variable takes on continuous values in some interval, e.g. voice, temperature, etc.

• digital data – representation variable takes on discrete (a finite & countable number of) values in a given interval, e.g. text, digitized images, etc.

Analog vs. Digital Signal

•analog signal – signal that is continuous in time and can assume an infinite number of values in a given range (continuous in time and value)

•discrete (digital) signal – signal that is continuous in time and assumes only a limited number of values (maintains a constant level and then changes to another constant level)

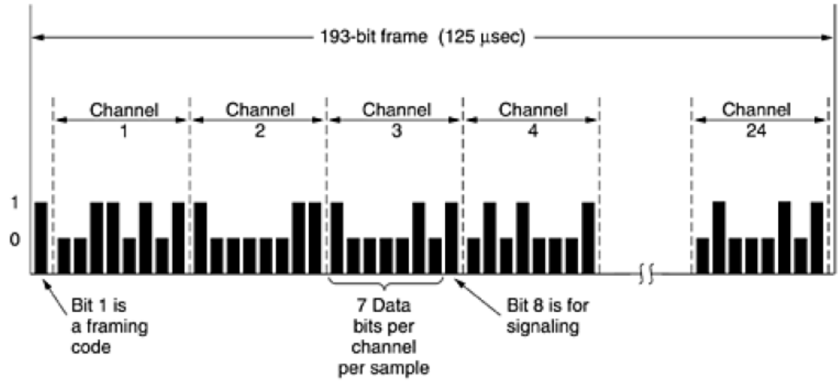


Time Division Multiplexing

WDM technology is wonderful, but there is still a lot of copper wire in the telephone system, so let us turn back to it for a while. Although FDM is still used over copper wires or microwave channels, it requires analog circuitry and is not amenable to being done by a computer. In contrast, TDM can be handled entirely by digital electronics, so it has become far more widespread in recent years. Unfortunately, it can only be used for digital data. Since the local loops produce analog signals, a conversion is needed from analog to digital in the end office, where all the individual local loops come together to be combined onto outgoing trunks.

We will now look at how multiple analog voice signals are digitized and combined onto a single outgoing digital trunk. Computer data sent over a modem are also analog, so the following description also applies to them. The analog signals are digitized in the end office by a device called a codec (coder-decoder), producing a series of 8-bit numbers. The codec makes 8000 samples per second (125 μsec/sample) because the Nyquist theorem says that this is sufficient to capture all the information from the 4-kHz telephone channel bandwidth. At a lower sampling rate, information would be lost; at a higher one, no extra information would be gained. This technique is called PCM (Pulse Code Modulation). PCM forms the heart of the modern telephone system. As a consequence, virtually all time intervals within the telephone system are multiples of 125 μsec.

When digital transmission began emerging as a feasible technology, CCITT was unable to reach agreement on an international standard for PCM. Consequently, a variety of incompatible schemes are now in use in different countries around the world.

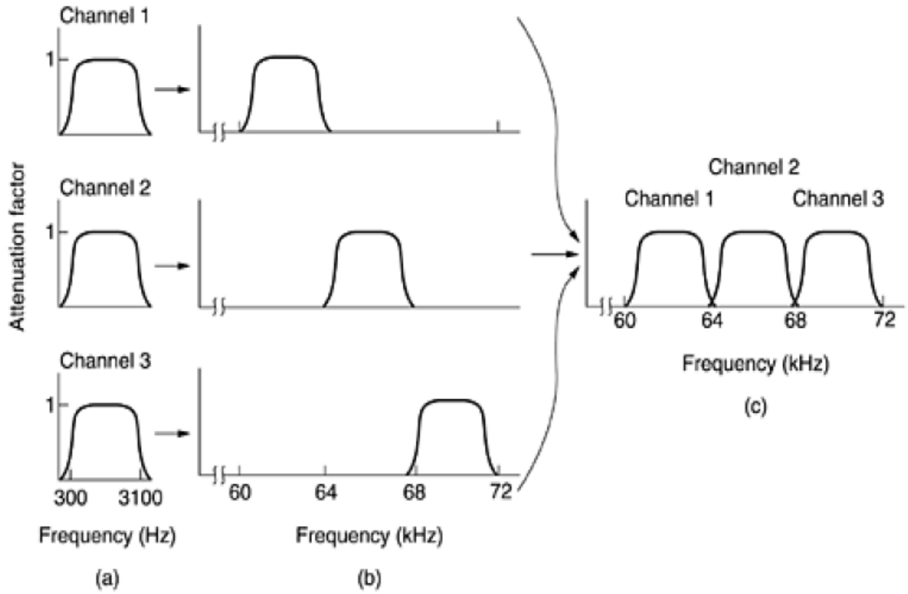


A frame consists of 24 x 8 = 192 bits plus one extra bit for framing, yielding 193 bits every 125 µsec. This gives a gross data rate of 1.544 Mbps. The 193rd bit is used for frame synchronization. It takes on the pattern 0101010101 . . . . Normally, the receiver keeps checking this bit to make sure that it has not lost synchronization. If it does get out of sync, the receiver can scan for this pattern to get resynchronized. Analog customers cannot generate the bit pattern at all because it corresponds to a sine wave at 4000 Hz, which would be filtered out. Digital customers can, of course, generate this pattern, but the odds are against its being present when the frame slips. When a T1 system is being used entirely for data, only 23 of the channels are used for data. The 24th one is used for a special synchronization pattern, to allow faster recovery in the event that the frame slips.

Frequency Division Multiplexing (FDM)

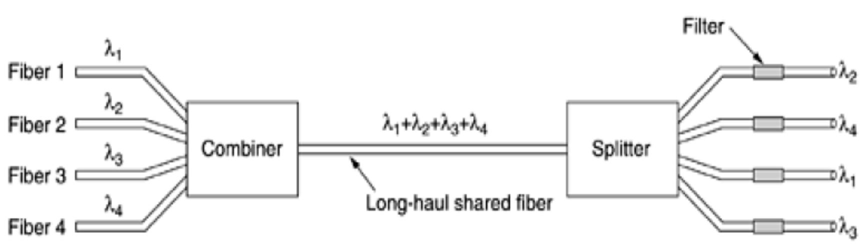
The traditional way of allocating a single channel, such as a telephone trunk, among multiple competing users is Frequency Division Multiplexing (FDM). If there are *N* users, the bandwidth is divided into *N* equal-sized portions each user being assigned one portion. Since each user has a private frequency band, there is no interference between users. When there is only a small and constant number of users, each of which has a heavy (buffered) load of traffic (e.g., carriers' switching offices), FDM is a simple and efficient allocation mechanism.

Figure: shows how three voice-grade telephone channels are multiplexed using FDM. Filters limit the usable bandwidth to about 3100 Hz per voice-grade channel. When many channels are multiplexed together, 4000 Hz is allocated to each channel to keep them well separated. First the voice channels are raised in frequency, each by a different amount. Then they can be combined because no two channels now occupy the same portion of the spectrum. Notice that even though there are gaps (guard bands) between the channels, there is some overlap between adjacent channels because the filters do not have sharp edges. This overlap means that a strong spike at the edge of one channel will be felt in the adjacent one as non thermal noise.



Wavelength Division Multiplexing

For fiber optic channels, a variation of frequency division multiplexing is used. It is called WDM (Wavelength Division Multiplexing). The basic principle of WDM on fibers is depicted in Fig. 2-32. Here four fibers come together at an optical combiner, each with its energy present at a different wavelength. The four beams are combined onto a single shared fiber for transmission to a distant destination. At the far end, the beam is split up over as many fibers as there were on the input side. Each output fiber contains a short, specially-constructed core that filters out all but one wavelength. The resulting signals can be routed to their destination or recombined in different ways for additional multiplexed transport.



WDM technology has been progressing at a rate that puts computer technology to shame. WDM was invented around 1990. The first commercial systems had eight channels of 2.5 Gbps per channel. By 1998, systems with 40 channels of 2.5 Gbps were on the market. By 2001, there were products with 96 channels of 10 Gbps, for a total of 960 Gbps. This is enough bandwidth to transmit 30 full-length movies per second (in MPEG-2). Systems with 200 channels are already working in the laboratory. When the number of channels is very large and the wavelengths are spaced close together, for example, 0.1 nm, the system is often referred to as DWDM (Dense WDM).

It should be noted that the reason WDM is popular is that the energy on a single fiber is typically only a few gigahertz wide because it is currently impossible to convert between electrical and optical media any faster. By running many channels in parallel on different wavelengths, the aggregate bandwidth is increased linearly with the number of channels.

Orthogonal Frequency Division Multiplexing

The first of the high-speed wireless LANs, **802.11a**, uses **OFDM** (**Orthogonal Frequency Division Multiplexing**) to deliver up to 54 Mbps in the wider 5-GHz ISM band. As the term FDM suggests, different frequencies are used—52 of them, 48 for data and 4 for synchronization—not unlike ADSL. Since transmissions are present on multiple frequencies at the same time, this technique is considered a form of spread spectrum, but different from both CDMA and FHSS. Splitting the signal into many narrow bands has some key advantages over using a single wide band, including better immunity to narrowband interference and the possibility of using noncontiguous bands. A complex encoding system is used, based on phase-shift modulation for speeds up to 18 Mbps and on QAM above that. At 54 Mbps, 216 data bits are encoded into 288-bit symbols. Part of the motivation for OFDM is compatibility with the European HiperLAN/2 system (Doufexi et al., 2002). The technique has a good spectrum efficiency in terms of bits/Hz and good immunity to multipath fading.