COMPUTER NETWORKS

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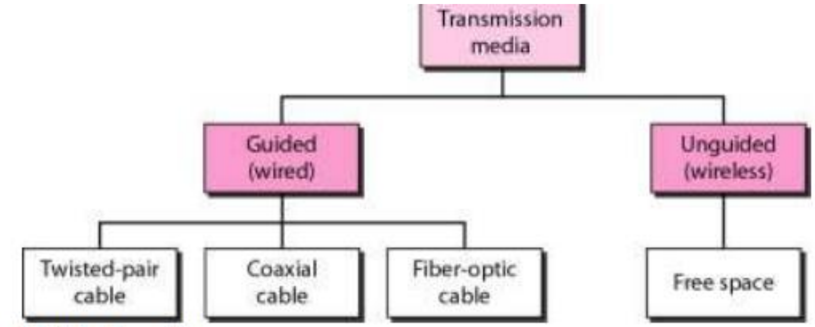
UNIT II

The Physical Layer: Guided Transmission Media – Digital Modulation and Multiplexing: Baseband Transmission – # Frequency Division Multiplexing # – The Public Switched Telephone Network: Structure of the Telephone System – The Politics of Telephones – The Local Loop: Modems, ADSL, and Fiber – Trunks and Multiplexing – Switching.

GUIDED TRANSMISSION MEDIA:



1. The purpose of the physical layer is to transport a raw bit stream from one machine to another.
2. Media are roughly grouped into guided media, such as copper wire and fiber optics, and unguided media, such as radio and lasers through the air.

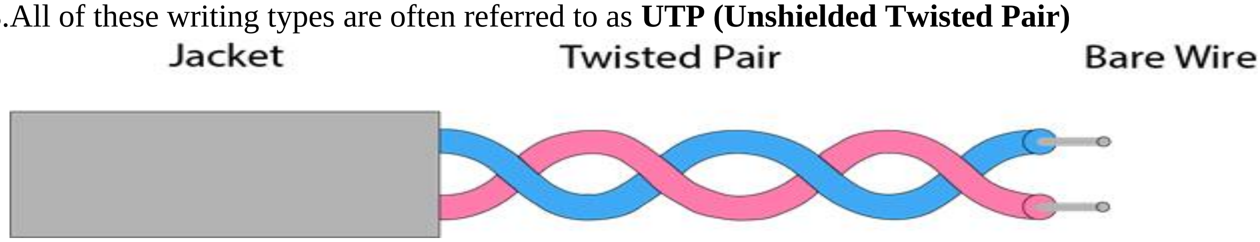
MAGNETIC MEDIA:

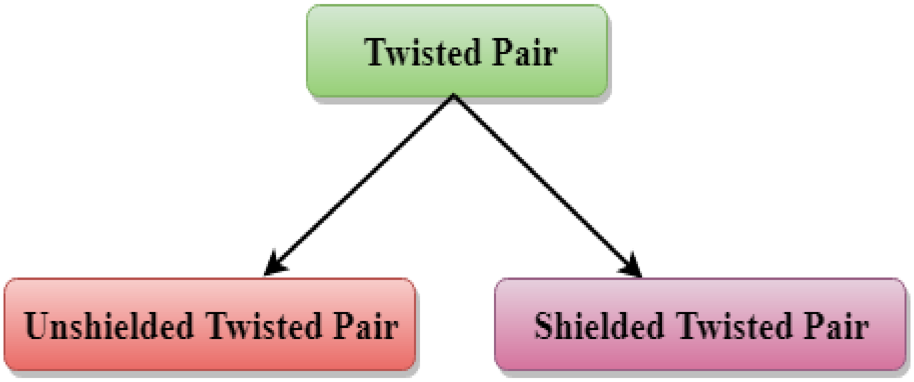
1 .One of the most common ways to transport data from one computer to another is to write them onto magnetic tape or removable media physically transport the tape or disks to the destination machine, and read them back in again.

2. Although this method is not sophisticated as using a geosynchronous communication satellite, it is often more cost effective.

TWISTED PAIR:

1.Most common transmission media is twisted pair.
2.A twisted pair consists of two insulated copper wires, typically about 1mm thick.
3.The wires are twisted together in a helical form, just like a DNA molecule.
4.The most common application of the **twisted pair** is the telephone system. Nearly all telephones are connected to the telephone company office by a twisted pair. Twisted pair can run several kilometers without amplification, but for longer distances, repeaters are needed.
5.Twisted pairs can be used for transmitting either analog or digital signals.
6.**Category 3** twisted pairs consist of two insulted wires gently twisted together.
7.Around 1988, the more advanced **category 5** twisted pairs were introduced.
8.All of these writing types are often referred to as **UTP (Unshielded Twisted Pair)**





Types of Twisted pair:

Unshielded Twisted Pair:

An unshielded twisted pair is widely used in telecommunication. Following are the categories of the unshielded twisted pair cable:

- **Category 1:** Category 1 is used for telephone lines that have low-speed data.
- **Category 2:** It can support upto 4Mbps.
- **Category 3:** It can support upto 16Mbps.
- **Category 4:** It can support upto 20Mbps. Therefore, it can be used for long-distance communication.
- **Category 5:** It can support upto 200Mbps.

Advantages Of Unshielded Twisted Pair:

- It is cheap.
- Installation of the unshielded twisted pair is easy.
- It can be used for high-speed LAN.

Disadvantage:

- This cable can only be used for shorter distances because of attenuation.

Shielded Twisted Pair:

A shielded twisted pair is a cable that contains the mesh surrounding the wire that allows the higher transmission rate.

Characteristics Of Shielded Twisted Pair:

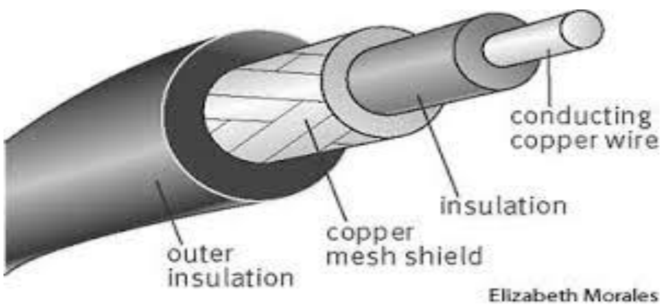
- The cost of the shielded twisted pair cable is not very high and not very low.
- An installation of STP is easy.
- It has higher capacity as compared to unshielded twisted pair cable.
- It has a higher attenuation.
- It is shielded that provides the higher data transmission rate.

Disadvantages

- It is more expensive as compared to UTP and coaxial cable.
- It has a higher attenuation rate.

COAXIABLE CABLE:

1.Another common transmission medium is the **coaxial cable**.
2. It has better shielding than twisted pairs, so it can span longer distances at higher speeds.
3. Two kinds of coaxial cable are widely used.
4. One kind 50-ohm cable, is commonly used when it is intended for digital transmission from the start.
5. The other kind is 75-ohm cable is, commonly used for analog transmission and cable television but becoming more important with the advent of internet over cable.
6. A coaxial cable consists of a stiff copper wire as the core, surrounded by an insulating material.
7. The insulator is encased by a cylindrical conductor, often as a closely woven braided mesh.
8. The outer conductor is covered in a protective plastic sheath.
9. Coaxial cables are used for telephone system for long distance lines



Coaxial cable is of two types:

1. **Baseband transmission:** It is defined as the process of transmitting a single signal at high speed.
2. **Broadband transmission:** It is defined as the process of transmitting multiple signals simultaneously.

Advantages Of Coaxial cable:

- The data can be transmitted at high speed.
- It has better shielding as compared to twisted pair cable.
- It provides higher bandwidth.

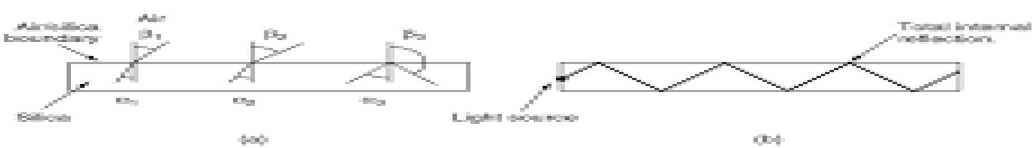
Disadvantages Of Coaxial cable:

- It is more expensive as compared to twisted pair cable.
- If any fault occurs in the cable causes the failure in the entire network.

Fiber Optic:

1. Current fiber technology, the achievable bandwidth is certainly in excess of 50000Gbps.
2. An optical transmission system has three key components: the light source, the transmission medium, and the detector.
3. Conventionally, a pulse of light indicates a 1 bit and the absence of light indicates a 0 bit.
4. The transmission medium is an ultra-thin fiber of glass.
5. The detector generates an electrical pulse when light falls on it.
6. When a light ray passes from one medium to another, example: From fused silica to air,the ray is refracted at the silica/air boundary.
7. A light ray incident at or above the critical angle is trapped inside the fiber,and can propagate for many kilometers with virtually no loss.
8. Each ray is said to have a different **mode**, so a fiber having this property is called a **multimode fiber**.
9. The fibers diameter is reduced to a few wavelengths of light , the fiber acts like a wave guide, and the light can propagate only in a straight line, without bouncing, yielding a **single-mode fiber.**

Fiber Optics



(a) Three examples of a light ray from inside a silica fiber impinging on the air/silica boundary at different angles.
(b) Light trapped by total internal reflection.

TRANSMISSION OF LIGHT THROUGH FIBER:

1. Optical fibers are made of glass, which in turn, is made from sand, an inexpensive raw material available in unlimited amounts.
2. The attenuation of light through glass depends on the wavelength of the light.
3. The attenuation in decibels is given by the formula :

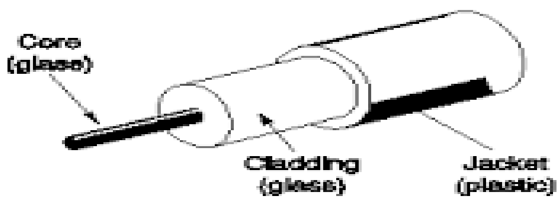
Attenuation in decibels =10 log 10 $\frac{\textit{transmitted power}}{\textit{recieved power}}$

For example: a factor of two loss gives an attenuation of 10log 10 2=3dB.

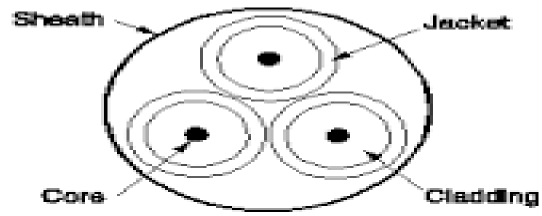
4. Three wave length bands are used for optical communication .They are centered at 0.85, 1.30, and 1.55 microns, respectively. The last two have good attenuation properties(less than 5 percent loss per kilometers). The 0.85 micron band has higher attenuation.
5. Light pulses sent down a fiber spread out in length as they propagate. This spreading is called **chromatic dispersion.**

FIBER CABLES:

1.The fiber cables shows a single fiber viewed from the side. At the center is the glass core through which the light propagates. In multimode fibers the core is typically 50 microns in diameter about the thickness of a human hair. In single-mode fibers the core is 8 to 10 microns.



(a)



(b)

2.The core is surrounded by a glass cladding with a lower index of refraction than the core, to keep all the light in the core. Next comes a thin plastic jacket to protest the cladding. Fibers are typically grouped in bundles, protected by an outer sheath. Sheaths with three fibers.

3.Fibers can be connected in three different ways. First, they can terminate in connectors and plugged into fiber sockets.

4.Second, they can be spliced mechanical. Mechanical splices just lay the two carefully-cut ends next to each other in a special sleeve and clamp them in place.

5.Third, two pieces of fiber can be fused to form a solid connection.

6.Two kinds of light sources are typically used to do the signaling. LEDs(Light Emitting Diodes)

7.And semiconductor lasers. They have different properties.

Item	LED	Semiconductor laser
Data rate	Low	High
Fiber type	Multimode	Multimode or single mode
Distance	Short	Long
Lifetime	Long life	Short life
Temperature sensitivity	Minor	Substantial
Cost	Low cost	expensive

FIBER OPTIC NETWORKS:

1. Fiber optics can be used for LANs as well as for long-haul transmission.
2. Two types of interfaces are used. A passive interface consists of two tapes fused onto the main fiber.
3. The other interface type is the active repeater. The incoming light is converted to an electrical signal, regenerated to full strength if it has been weakened, and retransmitted as light.

4.If an active repeater fails, the ring is broken and the network goes down. On the other hand, since the signal is regenerated at regenerated at each interface, the individual computer-to-computer links can be kilometers long, with no limit on the total size of the ring. The passive interfaces lose light at each junction, so the number of computers and total ring length are greatly restricted.

Following are the advantages of fiber optic cable over copper:

- **Greater Bandwidth:** The fibre optic cable provides more bandwidth as compared copper. Therefore, the fibre optic carries more data as compared to copper cable.
- **Faster speed:** Fibre optic cable carries the data in the form of light. This allows the fibre optic cable to carry the signals at a higher speed.
- **Longer distances:** The fibre optic cable carries the data at a longer distance as compared to copper cable.
- **Better reliability:** The fibre optic cable is more reliable than the copper cable as it is immune to any temperature changes while it can cause obstruct in the connectivity of copper cable.
- **Thinner and Sturdier:** Fibre optic cable is thinner and lighter in weight so it can withstand more pull pressure than copper cable.

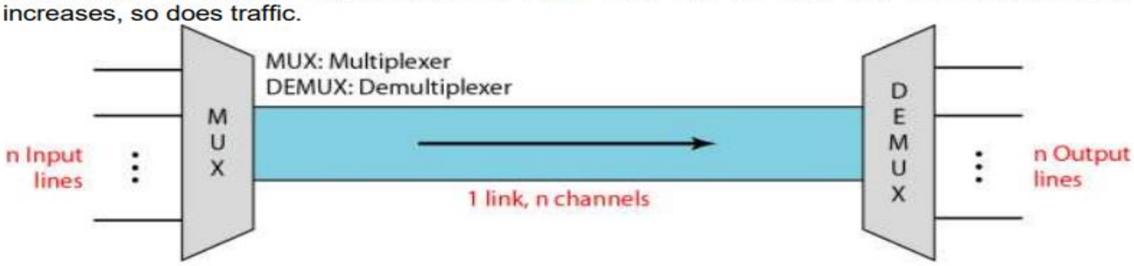
COMPARISON OF FIBER OPTICS AND COPPER WIRE:

1. Fiber has many advantages. To start with, it can handle much **higher bandwidths** than copper. This alone would require its use in high end networks. Due to the **low attenuation**, repeaters are needed only about every **50km on long lines**, versus about every **5km for copper**, a substantial **cost saving**. Fiber also has the advantage of not being affected by power surges, electromagnetic interfaces, or power failures.
2. Telephone companies like fiber for a different reason: it is thin and light weight. Many existing cable ducts are completely full, so there is no room to add new capacity.
    a. Fibers do not leak light are quite difficult to tap. These properties give fiber excellent security against potential wire tappers.

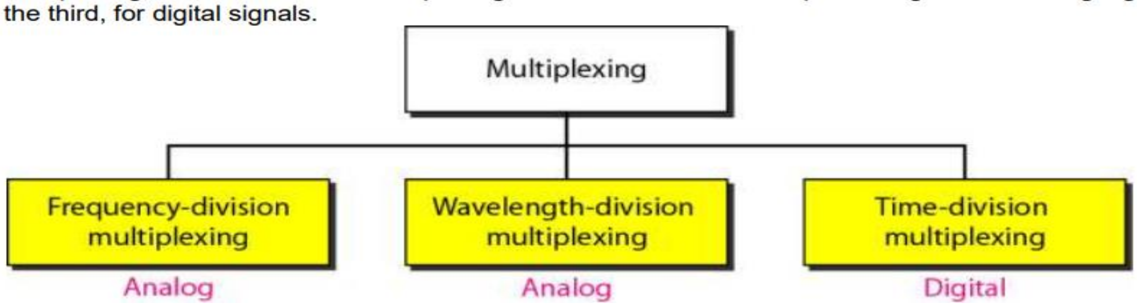
Digital Modulation and Multiplexing:

MULTIPLEXING

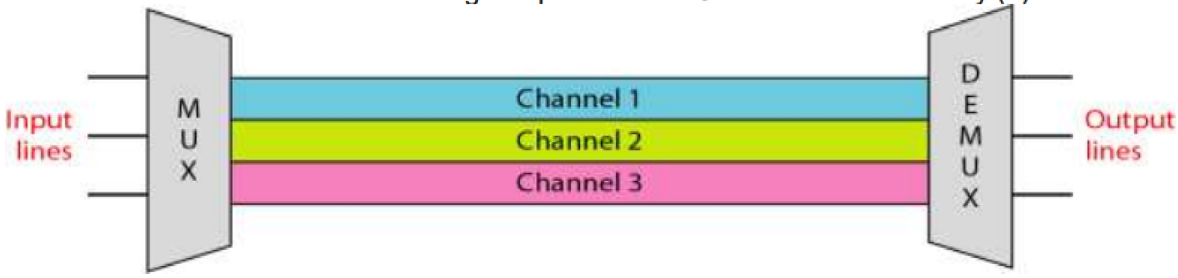
Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allows the (simultaneous) transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic.



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.

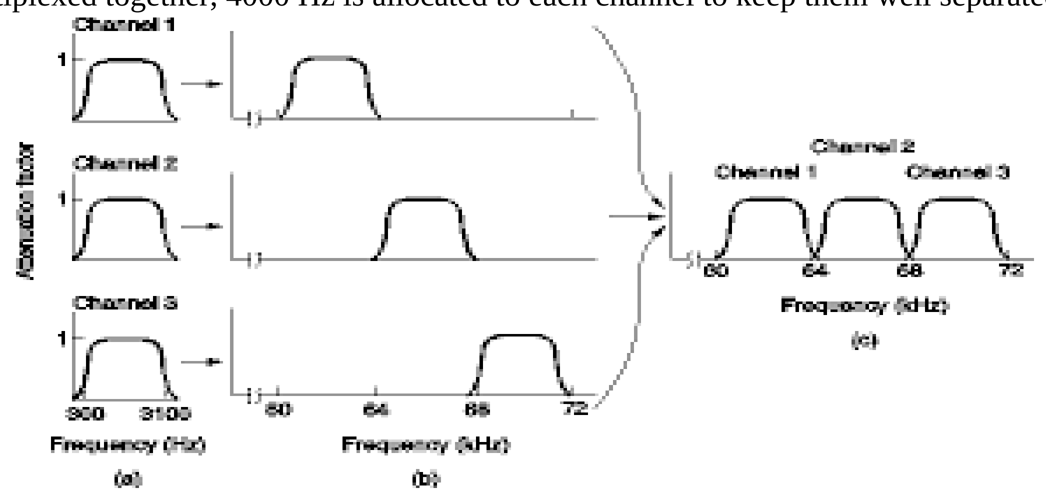


In a multiplexed system, n lines share the bandwidth of one link. The lines on the left direct their transmission streams to a multiplexer (MUX), which combines them into a single stream (many-to-one).
At the receiving end, that stream is fed into a demultiplexer (DEMUX), which separates the stream back 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.



FREQUENCY DIVISION MULTIPLEXING:

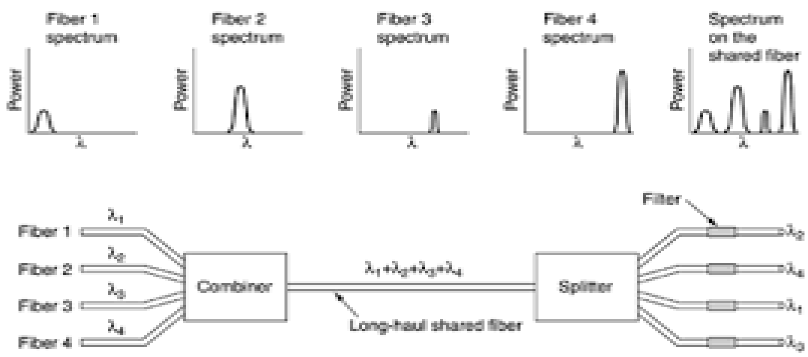
Shows how three voice-grade telephone channels are multiplexed using FDM. Filters limit the usable bandwidth to about 3100 Hz per voice-grade channel. When many channels are multiplexed together, 4000 Hz is allocated to each channel to keep them well separated.



The FDM schemes used around the world are to some degree standardized. A widespread standard is twelve 4000-Hz voice channels multiplexed into the 60-108 kHz band. This unit is called **a group**. The 12 kHz-60 kHz is sometimes used for another group. Many carries offer a 48 to 56-kbps leased line service to customers, based on the group. Five groups (60 voice channels) can be multiplexed to form a**super group**. The next unit is the master group, which are five super groups (CCITT standard) or ten super groups (Bell system). Other standards of up to 230,000 voice channels also exist.

WAVELENGTH DIVISION MULTIPLEXING:

1. For fiber optic channels, a variation of frequency division multiplexing is used. It is called **WDM (Wavelength Division Multiplexing).**
2. The basic principle ofWDM on fibers is depicted. Here four fibers come together at an optical combiner, each with its energy present at a different wavelength.

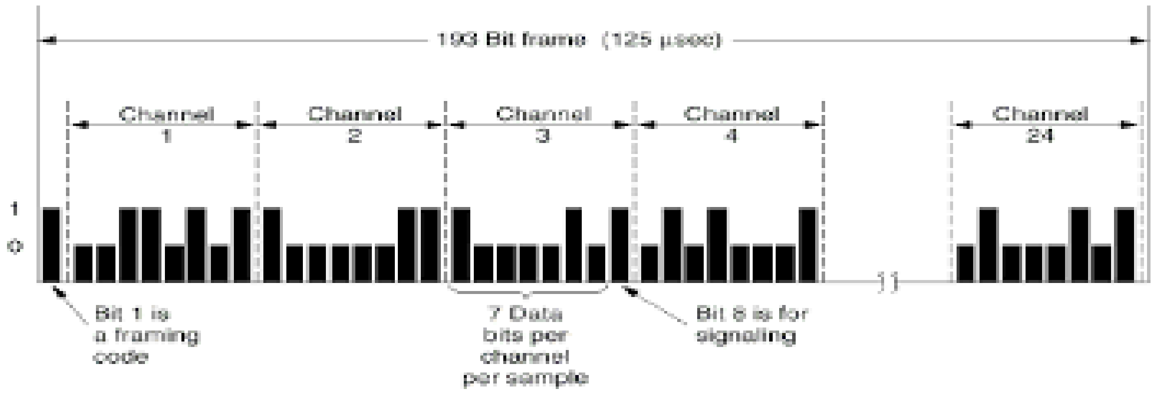


3. The four beams are combined onto a single shared fiber for transmission to a distant destination.
4. WDM was invented around 1990. The first commercial systems had eight channels of 2.5 Gbps per channel.
5. By 2001, there were products with 96 channels of 10 Gbps, for a total of 960Gbps.
6. This is enough bandwidth to transmit 30 full-length movies per second (in MPEG-2).
7. When the number of channels is very large and the wavelengths are spaced closed together, for example, 0.1 nm, the system is often referred to as **DWDM (Dense WDM)**.
8. 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.

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.
- 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.
- The analog signals are digitized in the end office by a device called a codec (coder-decoder), producing a series of 8-bits numbers.
- 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.

- The T1 carrier consists of 24 voice channels multiplexed together.
- Each of the 24 channels, in turn, gets to insert 8 bits into the output stream. Seven bits are data and one is for control, yielding 7x8000=56000 bps of data, and 1x8000=8000 bps of signaling information per channel.



The T1 carrier (1.544 Mbps)

- A frame consists of 24x8=192 bits plus one extra bit for framing, yielding 193 bits every 125µsec. This gives a gross data rate of 1.544 Mbps. The 193^{rd} bit is used for frame synchronization.
- When a T1 system is being used entirely for data, only 23 of the channels are used for data. The 24^{th} one is used for a special synchronization pattern, to allow faster recovery in the event that the frame slips.
- A delta modulation is an analog to digital and digital-to-analog signal conversion technique used for transmission of voice information where quality is not of primary.

THE PUBLIC SWITCHED TELEPHONE NETWORK:

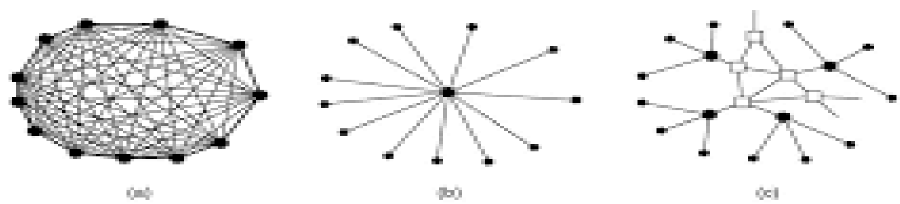
1.The PSTN(Public Switched Telephone Network), were usually designed many years ago, with a completely different goal in mind: transmitting the human voice in more-or-less recognizable form.
2. PSTN is the world collection of inter connected voice-oriented public telephone networks, both commercial and government owned. It's also referred to as the plain-old telephone system(POTS)

STRUCTURE OF THE TELEPHONE SYSTEM:

The following modal of the connecting every telephone to every,
other telephone was not going to work:

- By 1890, the three major parts of the telephone system were in place:the switching offices, the wires between the customers and the switching offices (by now balanced, insulated, twisted pairs instead of open wires with an earth return), and the long distance connections between the switching offices.
- Prior to the 1984 breakup of AT&T, the telephone system was organized as a highly redundant, multilevel hierarchy.
- Each telephone has two copper wires coming out of it that go directly to the telephone company's nearest **end office**(also called as **Local central office**). The distance is typically 1 to 10 km, being shorter in cities than in rural areas.

Structure of the Telephone System

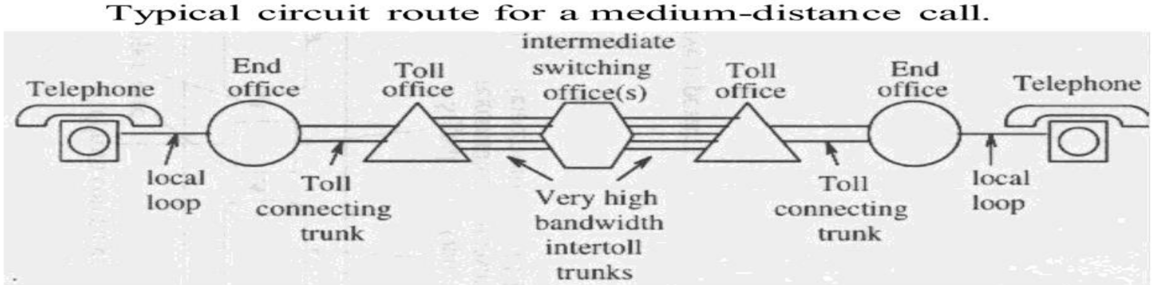


(a) Fully-interconnected network.
(b) Centralized switch.
(c) Two-level hierarchy.

- The two wire connections between each subscriber's telephones at the end office are known in the trade as the **local loop**.
- If the called telephone is attached to another end office, a different procedure has to be used. Each end office has a number of outgoing lines to one or more nearby switching centers, called **toll offices** (or if they are within the same local area , **tandem offices**). These lines are called **toll connecting trunks**.
- The number of different kinds of switching centers and their topology varies from country to country depending on the country's density. The following figure shows how a medium-distance connection might be routed:

Example of Circuit Route

Typical circuit route for a medium-distance call.



- A variety of transmission media are used for telecommunication.
- Between switching offices, coaxial cables, microwaves, and especially fiber optics are specially used.
- The telephone system consists of three major components:
  1. Local loops (analog twisted pairs going into houses and business).
  2. Trunks (digital fiber optics connecting the switching offices).
  3. Switching offices (where calls are moved from one trunk to another).

The Politics of Telephones

For decades prior to 1984, the Bell System provided both local and long-distance service throughout most of the United States. In the 1970s, the U.S. Federal Government came to believe that this was an illegal monopoly and sued to break it up. The government won, and on January 1, 1984, AT&T was broken up into AT&T Long Lines, 23 **BOCs** (**Bell Operating Companies**), and a few other pieces. The 23 BOCs were grouped into seven regional BOCs (RBOCs) to make them economically viable. The entire nature of telecommunication in the United States was changed overnight by court order (*not* by an act of Congress).

The exact specifications of the divestiture were described in the so-called **MFJ** (**Modified Final Judgment**), an oxymoron if ever there was one—if the judgment could be modified, it clearly was not final. This event led to increased competition, better service, and lower long-distance rates for consumers and businesses. However, prices for local service rose as the cross subsidies from long-distance calling were eliminated and local service had to become self supporting. Many other countries have now introduced competition along similar lines.

Of direct relevance to our studies is that the new competitive framework caused a key technical feature to be added to the architecture of the telephone network. To make it clear who could do what, the United States was divided up into 164 **LATAs** (**Local Access and Transport Areas**). Very roughly, a LATA is about as big as the area covered by one area code. Within each LATA, there was one **LEC** (**Local Exchange Carrier**) with a monopoly on traditional telephone service within its area. The most important LECs were the BOCs, although some LATAs contained one or more of the 1500 independent telephone companies operating as LECs.

The new feature was that all inter-LATA traffic was handled by a different kind of company, an **IXC** (**IntereXchange Carrier**). Originally, AT&T Long Lines was the only serious IXC, but now there are well-established competitors such as Verizon and Sprint in the IXC business. One of the concerns at the breakup was to ensure that all the IXCs would be treated equally in terms of line quality, tariffs, and the number of digits their customers would have to dial to use them. The way this is handled is illustrated in Fig. 2-31. Here we see three example LATAs, each with several end offices. LATAs 2 and 3 also have a small hierarchy with tandem offices (intra-LATA toll offices).

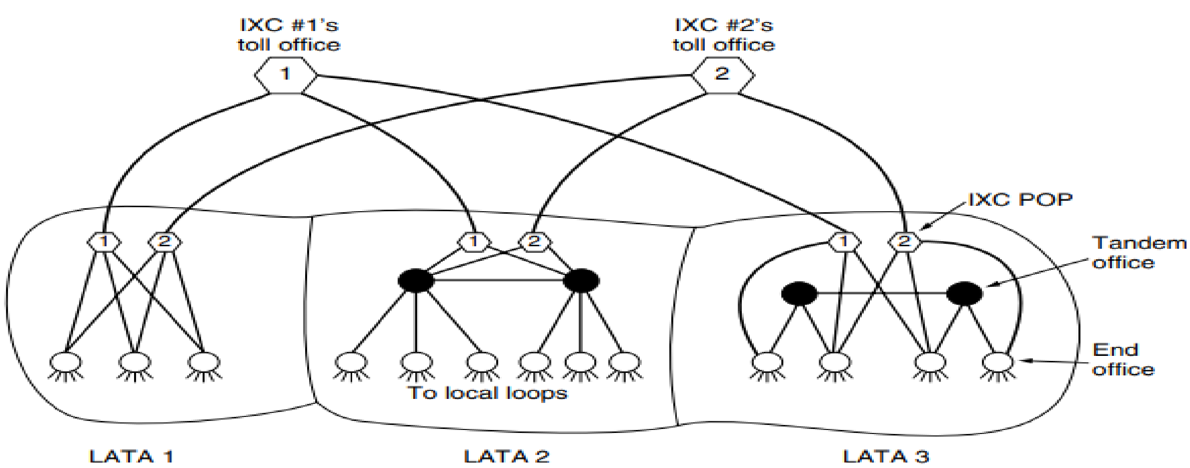


Figure 2-31. The relationship of LATAs, LECs, and IXCs. All the circles are LEC switching offices. Each hexagon belongs to the IXC whose number is in it.

Any IXC that wishes to handle calls originating in a LATA can build a switching office called a **POP** (**Point of Presence**) there. The LEC is required to connect each IXC to every end office, either directly, as in LATAs 1 and 3, or indirectly, as in LATA 2. Furthermore, the terms of the connection, both technical and financial, must be identical for all IXCs. This requirement enables, a subscriber in, say, LATA 1, to choose which IXC to use for calling subscribers in LATA 3.

As part of the MFJ, the IXCs were forbidden to offer local telephone service and the LECs were forbidden to offer inter-LATA telephone service, although both were free to enter any other business, such as operating fried chicken restaurants. In 1984, that was a fairly unambiguous statement. Unfortunately, technology has a funny way of making the law obsolete. Neither cable television nor mobile phones were covered by the agreement. As cable television went from one way to two way and mobile phones exploded in popularity, both LECs and IXCs began buying up or merging with cable and mobile operators.

By 1995, Congress saw that trying to maintain a distinction between the various kinds of companies was no longer tenable and drafted a bill to preserve accessibility for competition but allow cable TV companies, local telephone companies, long-distance carriers, and mobile operators to enter one another's businesses. The idea was that any company could then offer its customers a single integrated package containing cable TV, telephone, and information services and that different companies would compete on service and price. The bill was enacted into law in February 1996 as a major overhaul of telecommunications regulation. As a result, some BOCs became IXCs and some other companies, such as cable television operators, began offering local telephone service in competition with the LECs.

One interesting property of the 1996 law is the requirement that LECs implement **local number portability**. This means that a customer can change local telephone companies without having to get a new telephone number. Portability for mobile phone numbers (and between fixed and mobile lines) followed suit in 2003. These provisions removed a huge hurdle for many people, making them much more inclined to switch LECs. As a result, the U.S. telecommunications landscape became much more competitive, and other countries have followed suit. Often other countries wait to see how this kind of experiment works out in the U.S. If it works well, they do the same thing; if it works badly, they try something else.

The Local Loop: Modems, ADSL, and Fiber – Trunks and

Telephone Modems

To send bits over the local loop, or any other physical channel for that matter, they must be converted to analog signals that can be transmitted over the channel. This conversion is accomplished using the methods for digital modulation that we studied in the previous section. At the other end of the channel, the analog signal is converted back to bits.

A device that converts between a stream of digital bits and an analog signal that represents the bits is called a **modem**, which is short for "*mo*dulator *dem*odulator." Modems come in many varieties: telephone modems, DSL modems, cable modems, wireless modems, etc. The modem may be built into the computer (which is now common for telephone modems) or be a separate box (which is common for DSL and cable modems). Logically, the modem is inserted between the (digital) computer and the (analog) telephone system, as seen in Fig. 2-32.

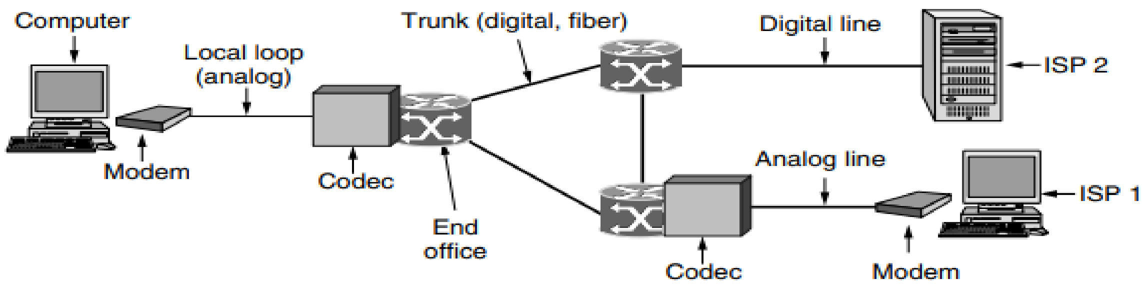


Figure 2-32. The use of both analog and digital transmission for a computer-to-computer call. Conversion is done by the modems and codecs.

Telephone modems are used to send bits between two computers over a voice-grade telephone line, in place of the conversation that usually fills the line. The main difficulty in doing so is that a voice-grade telephone line is limited to 3100 Hz, about what is sufficient to carry a conversation. This bandwidth is more than four orders of magnitude less than the bandwidth that is used for Ethernet or 802.11 (WiFi). Unsurprisingly, the data rates of telephone modems are also four orders of magnitude less than that of Ethernet and 802.11.

Let us run the numbers to see why this is the case. The Nyquist theorem tells us that even with a perfect 3000-Hz line (which a telephone line is decidedly not), there is no point in sending symbols at a rate faster than 6000 baud. In practice, most modems send at a rate of 2400 symbols/sec, or 2400 baud, and focus on getting multiple bits per symbol while allowing traffic in both directions at the same time (by using different frequencies for different directions).

The humble 2400-bps modem uses 0 volts for a logical 0 and 1 volt for a logical 1, with 1 bit per symbol. One step up, it can use four different symbols, as in the four phases of QPSK, so with 2 bits/symbol it can get a data rate of 4800 bps.

A long progression of higher rates has been achieved as technology has improved. Higher rates require a larger set of symbols or **constellation**. With many symbols, even a small amount of noise in the detected amplitude or phase can result in an error. To reduce the chance of errors, standards for the higher-speed modems use some of the symbols for error correction. The schemes are known as **TCM (Trellis Coded Modulation)** (Ungerboeck, 1987).

The **V.32** modem standard uses 32 constellation points to transmit 4 data bits and 1 check bit per symbol at 2400 baud to achieve 9600 bps with error correction. The next step above 9600 bps is 14,400 bps. It is called **V.32 bis** and transmits 6 data bits and 1 check bit per symbol at 2400 baud. Then comes **V.34**, which achieves 28,800 bps by transmitting 12 data bits/symbol at 2400 baud. The constellation now has thousands of points. The final modem in this series is **V.34 bis** which uses 14 data bits/symbol at 2400 baud to achieve 33,600 bps.

Why stop here? The reason that standard modems stop at 33,600 is that the Shannon limit for the telephone system is about 35 kbps based on the average length of local loops and the quality of these lines. Going faster than this would violate the laws of physics (department of thermodynamics).

However, there is one way we can change the situation. At the telephone company end office, the data are converted to digital form for transmission within the telephone network (the core of the telephone network converted from analog to digital long ago). The 35-kbps limit is for the situation in which there are two local loops, one at each end. Each of these adds noise to the signal. If we could get rid of one of these local loops, we would increase the SNR and the maximum rate would be doubled.

This approach is how 56-kbps modems are made to work. One end, typically an ISP, gets a high-quality digital feed from the nearest end office. Thus, when one end of the connection is a high-quality signal, as it is with most ISPs now, the maximum data rate can be as high as 70 kbps. Between two home users with modems and analog lines, the maximum is still 33.6 kbps.

The reason that 56-kbps modems (rather than 70-kbps modems) are in use has to do with the Nyquist theorem. A telephone channel is carried inside the telephone system as digital samples. Each telephone channel is 4000 Hz wide when the guard bands are included. The number of samples per second needed to reconstruct it is thus 8000. The number of bits per sample in the U.S. is 8, one of which may be used for control purposes, allowing 56,000 bits/sec of user data. In Europe, all 8 bits are available to users, so 64,000-bit/sec modems could have been used, but to get international agreement on a standard, 56,000 was chosen.

The end result is the **V.90** and **V.92** modem standards. They provide for a 56-kbps downstream channel (ISP to user) and a 33.6-kbps and 48-kbps upstream channel (user to ISP), respectively. The asymmetry is because there is usually more data transported from the ISP to the user than the other way. It also means that more of the limited bandwidth can be allocated to the downstream channel to increase the chances of it actually working at 56 kbps.

ADSL

Asymmetric Digital Subscriber Line

The most promising of the DSL technologies is ADSL or Asymmetric Digital Subscriber Line. ADSL looks to make the most impact in residential access and the SOHO (Small Office Home Office) market. Just like the name implies ADSL is asymmetric, meaning that the downstream bandwidth is higher than the upstream bandwidth. Downstream refers to traffic in the direction towards the subscriber, and upstream refers to data sent from the subscriber back to the network. This is done because of the kinds traffic that ADSL is designed to carry. Asymmetry is used to increase the downstream bandwidth. This works because all of the downstream signals can be of the same amplitude thus eliminating crosstalk between downstream channels. Upstream signals would have to put up with more interference because the amplitude of the upstream signals would be of smaller amplitude because the are originating from different distances. The asymmetric nature of ADSL lends itself well to applications like the web and client server applications.

To achieve the asymmetry ADSL divides its bandwidth into four classes of transport.

- higher bandwidth simplex channel
- lower bandwidth duplex channel
- duplex control channel
- POTS channel

Transmission on the high bandwidth simplex channel and the lower bandwidth duplex channel do not interfere in any way with the POTS channel. So ADSL can carry both data a POTS on the same medium, which makes it ideal for residential and small office use.

A typical ADSL arrangement is shown in Fig. 2-35. In this scheme, a telephone company technician must install a **NID (Network Interface Device)** on the customer's premises. This small plastic box marks the end of the telephone company's property and the start of the customer's property. Close to the NID (or sometimes combined with it) is a **splitter**, an analog filter that separates the

0–4000-Hz band used by POTS from the data. The POTS signal is routed to the existing telephone or fax machine. The data signal is routed to an ADSL modem, which uses digital signal processing to implement OFDM. Since most ADSL modems are external, the computer must be connected to them at high speed. Usually, this is done using Ethernet, a USB cable, or 802.11.

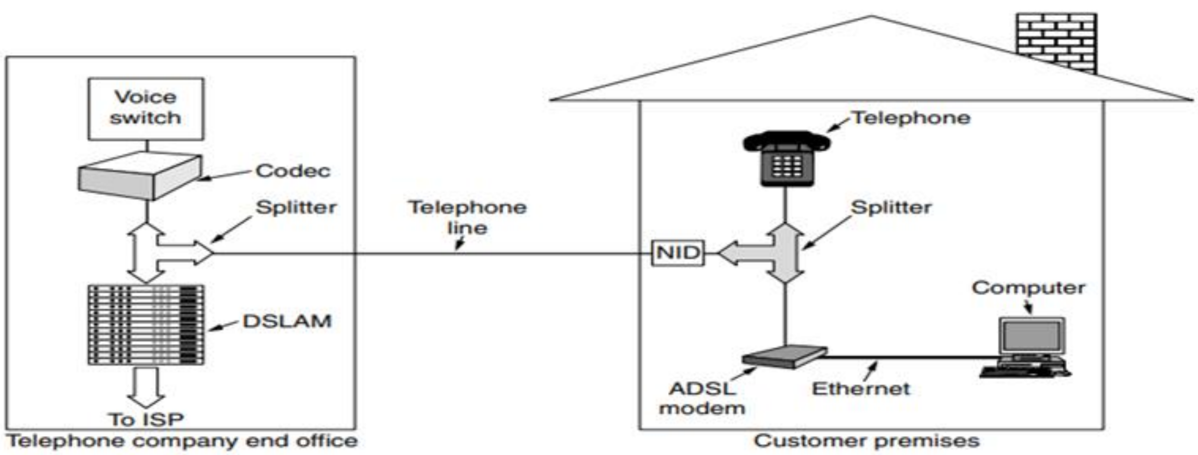


Figure 2-35. A typical ADSL equipment configuration.

At the other end of the wire, on the end office side, a corresponding splitter is installed. Here, the voice portion of the signal is filtered out and sent to the normal voice switch. The signal above 26 kHz is routed to a new kind of device called a **DSLAM** (**Digital Subscriber Line Access Multiplexer**), which contains the same kind of digital signal processor as the ADSL modem. Once the bits have been recovered from the signal, packets are formed and sent off to the ISP.

This complete separation between the voice system and ADSL makes it relatively easy for a telephone company to deploy ADSL. All that is needed is buying a DSLAM and splitter and attaching the ADSL subscribers to the splitter. Other high-bandwidth services (e.g., ISDN) require much greater changes to the existing switching equipment.

One disadvantage of the design of Fig. 2-35 is the need for a NID and splitter on the customer's premises. Installing these can only be done by a telephone company technician, necessitating an expensive "truck roll" (i.e., sending a technician to the customer's premises). Therefore, an alternative, splitterless design, informally called **G.lite**, has also been standardized. It is the same as Fig. 2-35 but without the customer's splitter. The existing telephone line is used as is. The only difference is that a microfilter has to be inserted into each telephone jack between the telephone or ADSL modem and the wire. The microfilter for the telephone is a low-pass filter eliminating frequencies above 3400 Hz; the microfilter for the ADSL modem is a high-pass filter eliminating frequencies below 26 kHz. However, this system is not as reliable as having a splitter, so G.lite can be used only up to 1.5 Mbps (versus 8 Mbps for ADSL with a splitter). For more information about ADSL, see Starr (2003).

Fiber To The Home

Deployed copper local loops limit the performance of ADSL and telephone modems. To let them provide faster and better network services, telephone companies are upgrading local loops at every opportunity by installing optical fiber all the way to houses and offices. The result is called **FttH** (**Fiber To The Home**). While FttH technology has been available for some time, deployments only began to take off in 2005 with growth in the demand for high-speed Internet from customers used to DSL and cable who wanted to download movies. Around 4% of U.S. houses are now connected to FttH with Internet access speeds of up to 100 Mbps.

Several variations of the form "FttX" (where *X* stands for the basement, curb, or neighborhood) exist. They are used to note that the fiber deployment may reach close to the house. In this case, copper (twisted pair or coaxial cable) provides fast enough speeds over the last short distance. The choice of how far to lay the fiber is an economic one, balancing cost with expected revenue. In any case, the point is that optical fiber has crossed the traditional barrier of the "last mile." We will focus on FttH in our discussion.

Like the copper wires before it, the fiber local loop is passive. This means no powered equipment is required to amplify or otherwise process signals. The fiber simply carries signals between the home and the end office. This in turn reduces cost and improves reliability.

Usually, the fibers from the houses are joined together so that only a single fiber reaches the end office per group of up to 100 houses. In the downstream direction, optical splitters divide the signal from the end office so that it reaches all the houses. Encryption is needed for security if only one house should be able to decode the signal. In the upstream direction, optical combiners merge the signals from the houses into a single signal that is received at the end office.

This architecture is called a **PON** (**Passive Optical Network**), and it is shown in Fig. 2-36. It is common to use one wavelength shared between all the houses for downstream transmission, and another wavelength for upstream transmission.

Even with the splitting, the tremendous bandwidth and low attenuation of fiber mean that PONs can provide high rates to users over distances of up to 20 km. The actual data rates and other details depend on the type of PON. Two kinds are common. **GPONs** (**Gigabit-capable PONs**) come from the world of telecommunications, so they are defined by an ITU standard. **EPONs** (**Ethernet PONs**)

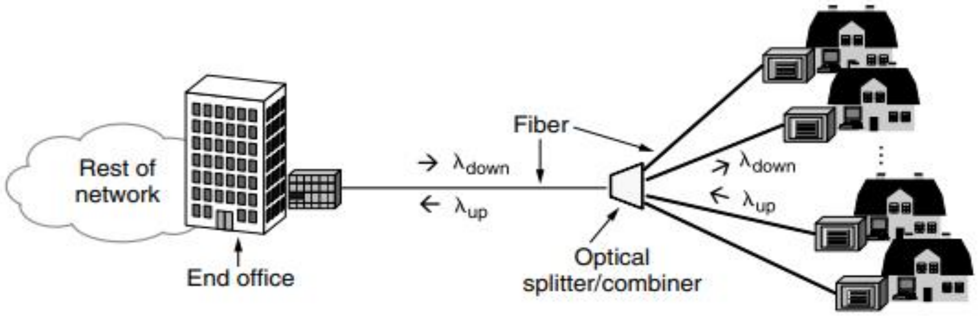


Figure 2-36. Passive optical network for Fiber To The Home.

are more in tune with the world of networking, so they are defined by an IEEE standard. Both run at around a gigabit and can carry traffic for different services, including Internet, video, and voice. For example, GPONs provide 2.4 Gbps downstream and 1.2 or 2.4 Gbps upstream.

Some protocol is needed to share the capacity of the single fiber at the end office between the different houses. The downstream direction is easy. The end office can send messages to each different house in whatever order it likes. In the upstream direction, however, messages from different houses cannot be sent at the same time, or different signals would collide. The houses also cannot hear each other's transmissions so they cannot listen before transmitting. The solution is that equipment at the houses requests and is granted time slots to use by equipment in the end office. For this to work, there is a ranging process to adjust the transmission times from the houses so that all the signals received at the end office are synchronized. The design is similar to cable modems, which we cover later in this chapter. For more information on the future of PONs, see Grobe and Elbers (2008).

TRUNKS AND MULTIPLEXING:

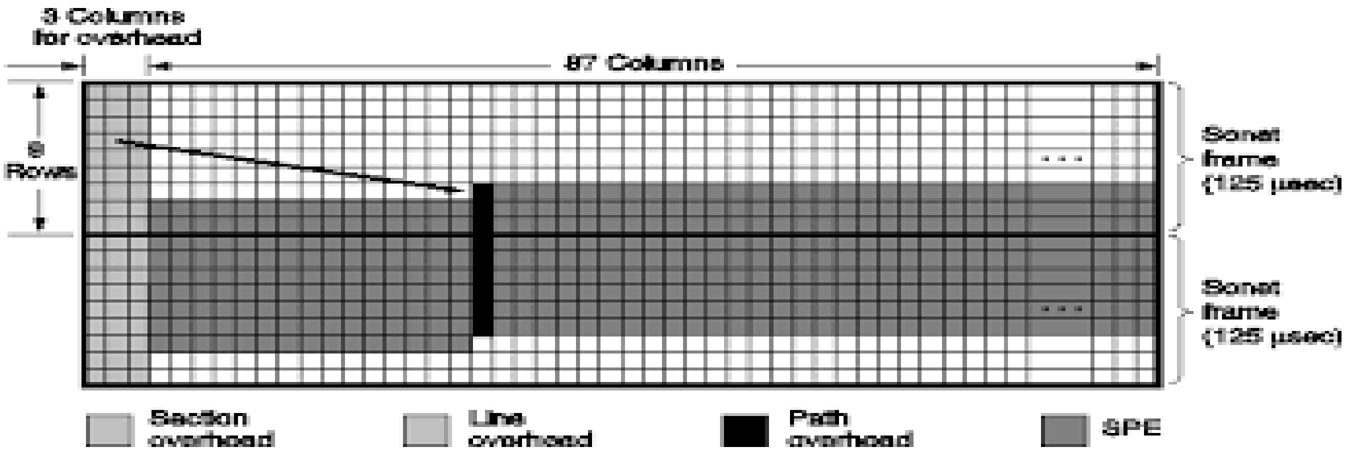
- ✓ Telephone companies have developed elaborate schemes for multiplexing many conversations over a single physical trunk.
- ✓ These multiplexing schemes can be divided into two basic categories: **FDM (Frequency Division Multiplexing**) and **TDM (Time Division Multiplexing**).
- ✓ In FDM, the frequency spectrum is divided into frequency bands, with each user having exclusive possession of some band.
- ✓ In TDM, the users take turns each one periodically getting the entire bandwidth for a little burst of time.

DELTA MODULATION:

- ✓ The transmitter and receiver must use the same **predication algorithm**. Such schemes are called predictive encoding. They are useful because they reduce the size of the numbers to be encoded, hence the number of bits to be sent.
  - ✓ Time division multiplexing allows multiple T1 carriers to be multiplexed into high order carriers.
  - ✓ Four T1 channels being multiplexed onto one T2 channel.
  - ✓ Four T1 streams at 1.544 Mbps should generate 6.176 Mbps, but T2 is actually 6.312 Mbps. The extra bits are used for framing and recovery in case the carrier slips. T1 and T3 are widely used by customers, whereas T2 and T4 are only used within the telephone system itself.
  - ✓ At the next level, seven T2 streams are combined bitwise to form a T3 stream. Then six streams are joined to form a T4 stream. At each step a small amount of overhead is added for framing and recovery in case the synchronization between sender and receiver is lost.

SONET/SDH (Synchronous Optical Network/Synchronous Digital Hierarchy):

- A standard developed by ANSI for fiber optic technology that can transmit high-speed data. It can be used to deliver text, audio and video.
- The SONET design had four major goals.
- First and foremost, SONET had to make it possible for different carriers to interwork.
- Second, some means was needed to unify the U.S., European, and Japanese digital systems, all of which were based on 64-kbps PCM channels, but all of which combined them in different ways.
- Third, SONET had to provide a way to multiplex multiple digital channels.
- Fourth, SONET had to provide support for the operations, administration, and maintenance (OAM).
- SONET is a synchronous system. It is controlled by a master clock with an accuracy of about 1 part in 10^9.
- The basic SONNET frame is a block of 810 bytes put out every 125µsec.
- The 810-byte SONNET frames are best described as a rectangle of bytes, 90 columns wide by 9 rows high. Thus, 8x810=6480 bits are transmitted 8000 times per second, for a gross data rate of 51.84 Mbps. This the basic SONET channel, called **STS-1 (Synchronous Transport Signal-1)**.
- All SONET trunks are a multiple of STS-1.
- The first three columns of each frame are reserved for the system management information, as illustrated in the figure.
- The first three rows contain the section overhead; the next six contain the line overhead. The section overhead is generated and checked at the start and end of each line.



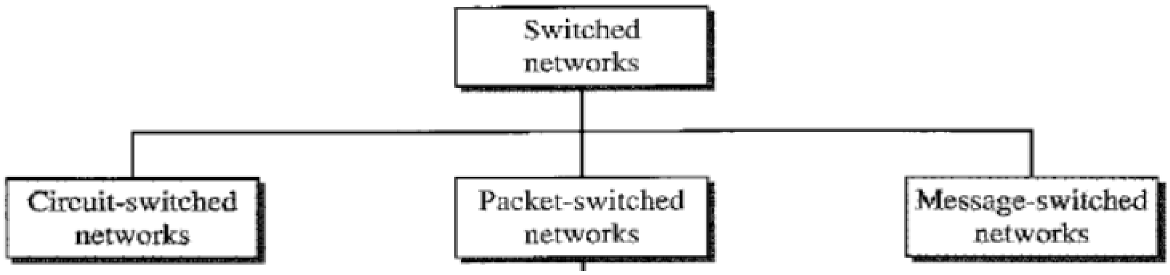
Two back-to-back SONET frames.

- The remaining 87 columns hold 87x9x8x8000=50.112 Mbps of user data. However, the user data, called the **SPE (Synchronous Payload Envelope),** do not always begin in row 1, column 4. The SPE can begin anywhere within the frame.
- The SONET multiplexing hierarchy is shown below. Rates from STS-1 to STS-192 have been defined.
- The SDH names are different, and they start at OC-3 because CCITT-based systems do not have a rate near 51.84 Mbps.

SONET		SDH	Data rate (Mbps)		
Electrical	**Optical**	**Optical**	**Gross**	**SPE**	**User**
STS-1	OC-1		51.84	50.112	49.536
STS-3	OC-3	STM-1	155.52	150.336	148.608
STS-9	OC-9	STM-3	466.56	451.008	445.824
STS-12	OC-12	STM-4	622.08	601.344	594.432
STS-18	OC-18	STM-6	933.12	902.016	891.648
STS-24	OC-24	STM-8	1244.16	1202.688	1188.864
STS-36	OC-36	STM-12	1866.24	1804.032	1783.296
STS-48	OC-48	STM-16	2488.32	2405.376	2377.728

SWITCHING:

Taxonomy of switched networks



- The phone system is divided into two principal parts: outside plant (the local loops and trunks, since they are physically outside the switching offices) and inside plant (the switches), which are inside the switching offices.
- Two different switching techniques are used nowadays: circuit switching and packet switching.

CIRCUIT SWITCHING:

- When you or your computer places a telephone call, the switching equipment within the telephone system seeks out a physical path all the way from your telephone to the receiver's telephone.
- This technique is called circuit switching.
- A switching technology that establishes an electrical connection between stations using a dedicated path.
- Each of the six rectangles represents carrier **switching office(end office, toll office etc.).**
- When a call passes through a switching office, a physical connection is (conceptually) established between the line on which the call came in and one of the output lines, as shown below:

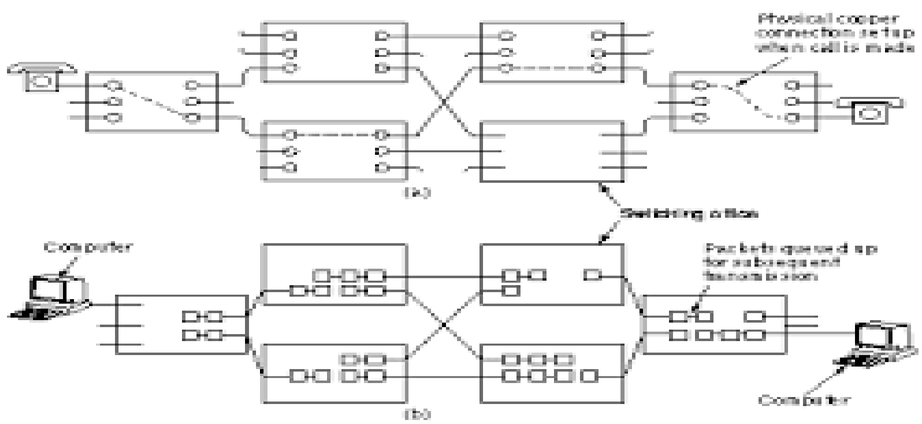


Fig. (a)Circuit switching (b) Packet switching.

- Parts of the physical path between the two telephones may, in fact, be microwave or fiber links onto which thousands of calls multiplexed.
- An important property of circuit switching is the need to set up an end-to-end path before any data can be sent. The elapsed time between the end of dialing and the start of ringing can be easily be 10 sec, more on long distance or international calls.

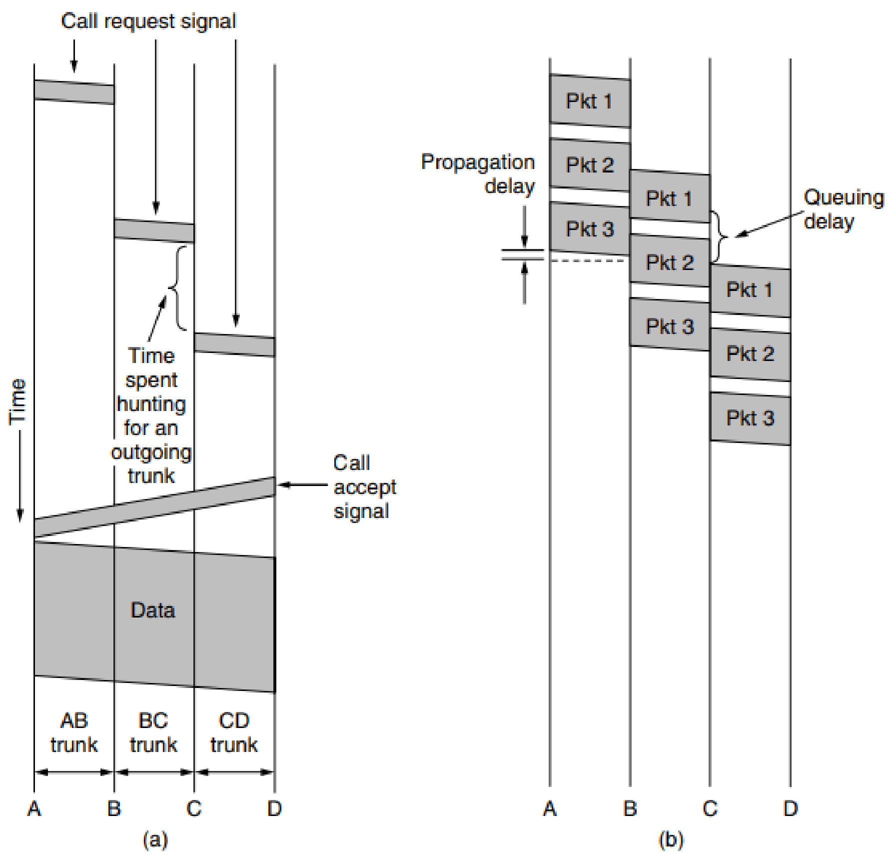


Figure 2-43. Timing of events in (a) circuit switching, (b) packet switching.

PACKET SWITCHING:

- Packet switching networks place a tight on upper limit on block size, allowing packets to be buffered in router main memory instead of on disk.
- By making sure that no user can monopolize any transmission line very long, packets-switching networks are well suited for handling interactive traffic.
- A further advantage of packet switching over message switching is shown the first packet of a multi-packet message can be forwarded before the second one has fully arrived, reducing delay and improving throughput.
- For these reasons, computer networks are usually packet switched, occasionally circuit switched, but never message switched.
- Circuit switching requires that a circuit be set up end-to-end before communication begins. Packet switching does not requires any advance setup.
- Result of the connection setup with circuit switching is the reservation of bandwidth all the way from the sender to receiver. All packets follow this path.
- With packet switching there is no path, so different packets can follow different paths depending on network conditions at the time they are sent.
- Packet switching is more fault tolerant than circuit switching.

- If a circuit has been reserved for a particular user and there is no traffic to send the bandwidth of that circuit is wasted. It cannot be used for other traffic. Packet switching does not waste bandwidth and thus is more efficient from a system-wide perspective.
- Packet switching uses store and forward transmission.
- With circuit switching, the bits just flow through the wire continuously.
- Another difference is that circuit switching is completely transparent.
- The sender and receiver can use any bit rate, format, or framing method they want to, the carrier does not know or care. With packet switching, the carrier determines the basic parameters.
- A final difference between circuit and packet switching is the charging algorithm.
- With circuit switching, charging has historically been based on distance and time.
- For mobiles phones, distance usually does not play a role, except for international calls, and time plays only a minor role.
- With packet switching connect time is not an issue but the volume of traffic sometimes is.

Item	Circuit switched	Packet switched
Call setup	Required	Not needed
Dedicated physical path	Yes	No
Each packet follows the same route	Yes	No
Packets arrive in order	Yes	No
Is a switch crash fatal	Yes	No
Bandwidth available	Fixed	Dynamic
Time of possible congestion	At setup time	On every packet
Potentially wasted bandwidth	Yes	No
Store-and-forward transmission	No	Yes
Transparency	Yes	No
Charging	Per minute	Per packet

MESSAGE SWITCHING:

- An alternative switching strategy is **message switching**. When this form of switching is used, no physical path is established in advance between sender and receiver.
- Instead, when the sender has a block of data to be sent, it is stored in the first switching office and then forwarded later, one hop at a time.
- Each block is received in its entirely, inspected for errors, and then transmitted. A network using this technique is called **a store-and-forward** network.
- The first electromechanical telecommunication systems used message switching, namely, for telegrams.
- The message was punched on paper tape (off-line) at the sending office, and then read in and transmitted over a communication line to the next office along the way, where it was punched out on a paper tape. An operator there tore the tape off and read it in on one of the many tape readers, one reader per outgoing trunk. Such a switching office was called a **torn tape office.**

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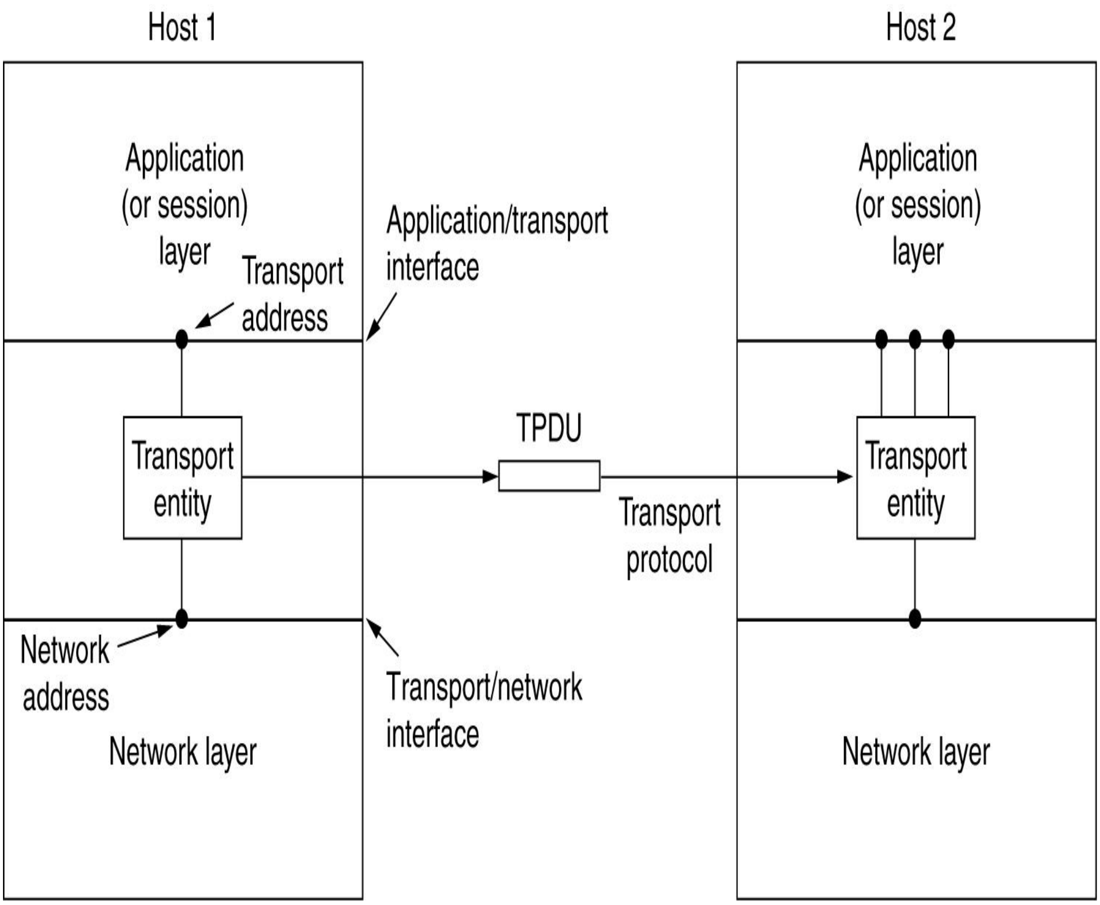
UNIT – V

Transport Layer: Transport Service: Services Provided to the Upper Layers –Transport Service Primitives – Berkeley Sockets – Elements of Transport Protocols – Application Layer: DNS: Domain Name System – DNS Name Space – Domain Resource Records – Electronic Mail: Architecture and Services – User Agent – Message Format.

TRANSPORT LAYER

SERVICES PROVIDED TO THE UPPER LAYER:

The goal of the transport layer is to provide efficient, reliable and cost effective service to its users. The hardware and/or software within the transport layer that does the work are called the transport entity. The transport entity can be located in the operating system. The logical relationship of the network, transport and application layers is shown in figure.



The transport layer is possible for the transport service to be more reliable than the network service. It is a key position for designing layers and the major boundary between the provider and user. The bottom four layers is known as transport service provider and upper layer as transport service user.

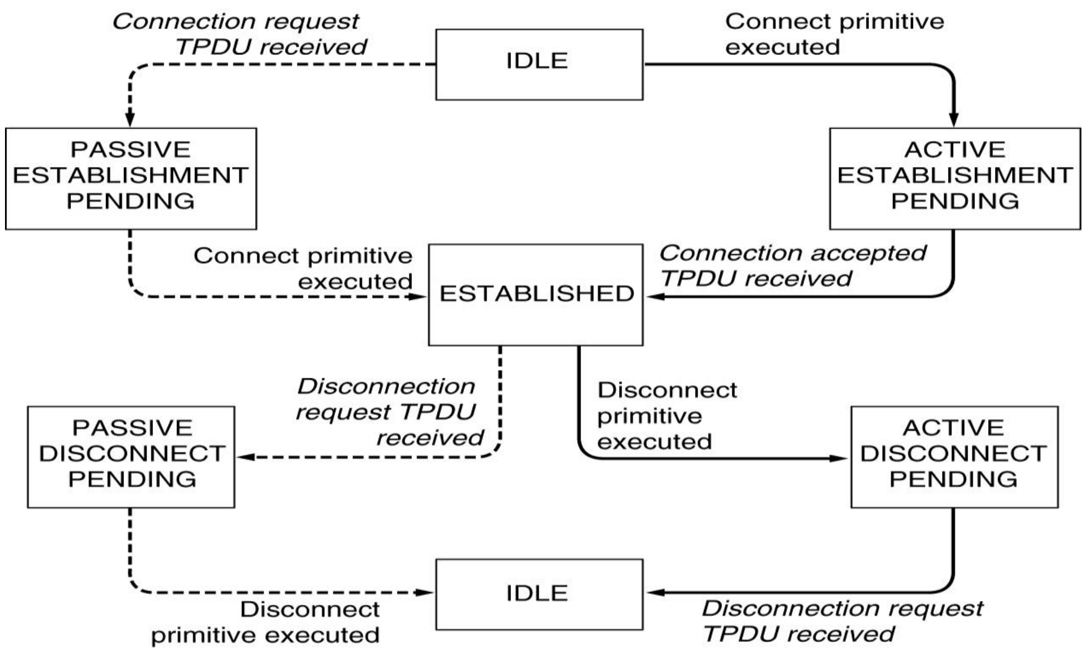
TRANSPORT SERVICE PRIMITIVES:

The transport service is reliable and processes the error free bit stream in connection oriented service. In datagram service it provides unreliable service. Compare to the network service, transport service is convenient and easy to use.

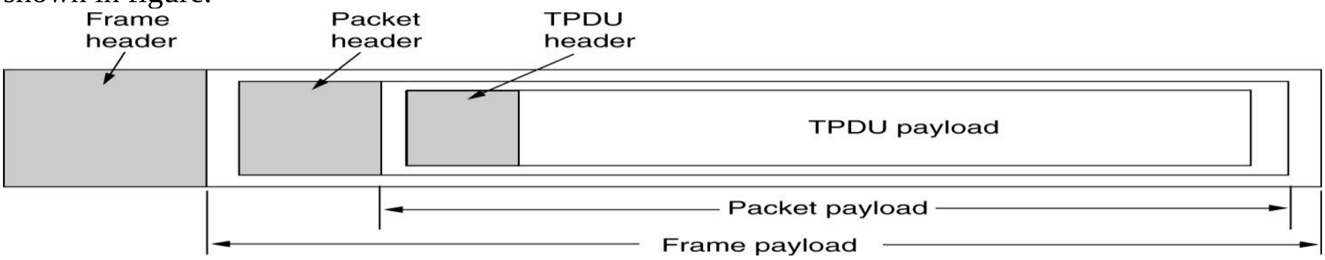
PRIMITIVES ARE:

Primitive	Packet sent	Meaning
LISTEN	(none)	Block until some process tries to connect
CONNECT	CONNECTION REQ.	Actively attempt to establish a connection
SEND	DATA	Send information
RECEIVE	(none)	Block until a DATA packet arrives
DISCONNECT	DISCONNECTION REQ.	This side wants to release the connection

A diagram for connection establishment and release for the primitives is shown below.



Transitions labeled in italics are caused by packet arrivals. The solid lines show the client's state sequence. The dashed lines show the server's state sequence. The use of TPDU (Transport Protocol Data Unit) for messages sent from transport entity to transport entity. TPDU's are exchanged by the transport layer are contained in packets where exchanged by network layer. Packets are contained in frames exchanged by data link layer. When frame arrives the data link layer processes the frame header and passes the contents of the frame payload field up to the network entity. This entity processes the packet header and passes the contents of the payload to the transport entity. This is shown in figure.



BERKELEY SOCKETS:

These primitives are widely used for internet programming. The socket primitives for TCP is shown in table.

Primitive	Meaning
SOCKET	Create a new communication end point
BIND	Attach a local address to a socket
LISTEN	Announce willingness to accept connections; give queue size
ACCEPT	Block the caller until a connection attempt arrives
CONNECT	Actively attempt to establish a connection
SEND	Send some data over the connection
RECEIVE	Receive some data from the connection
CLOSE	Release the connection

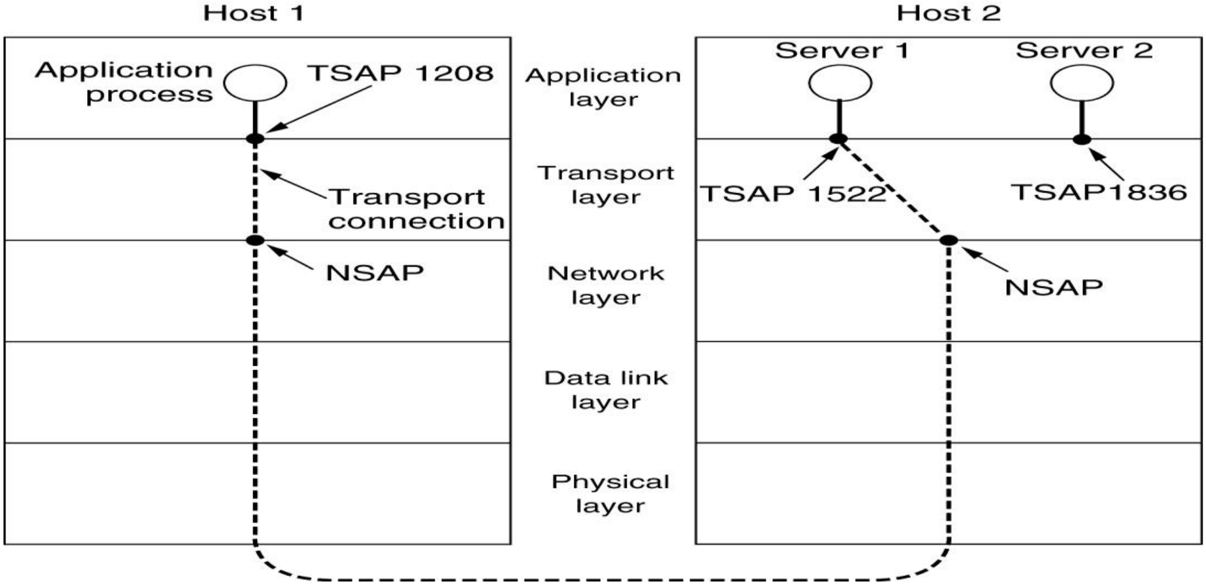
- The socket primitives create a new end point and allocate table space. The parameters specify the addressing format, types of service and the protocol.
- The bind primitive assign the network addresses.
- The listen call which allocates space to queue incoming calls from the several clients.
- The server executes the accept primitive to block waiting for an incoming connection. It returns a file descriptor for reading and writing.
- The connect primitive blocks the caller and starts the connection.
- Both sides can now use send and receive primitive to transmit and receive data over the full-duplex connection. When both sides execute the close primitive the connection is released.

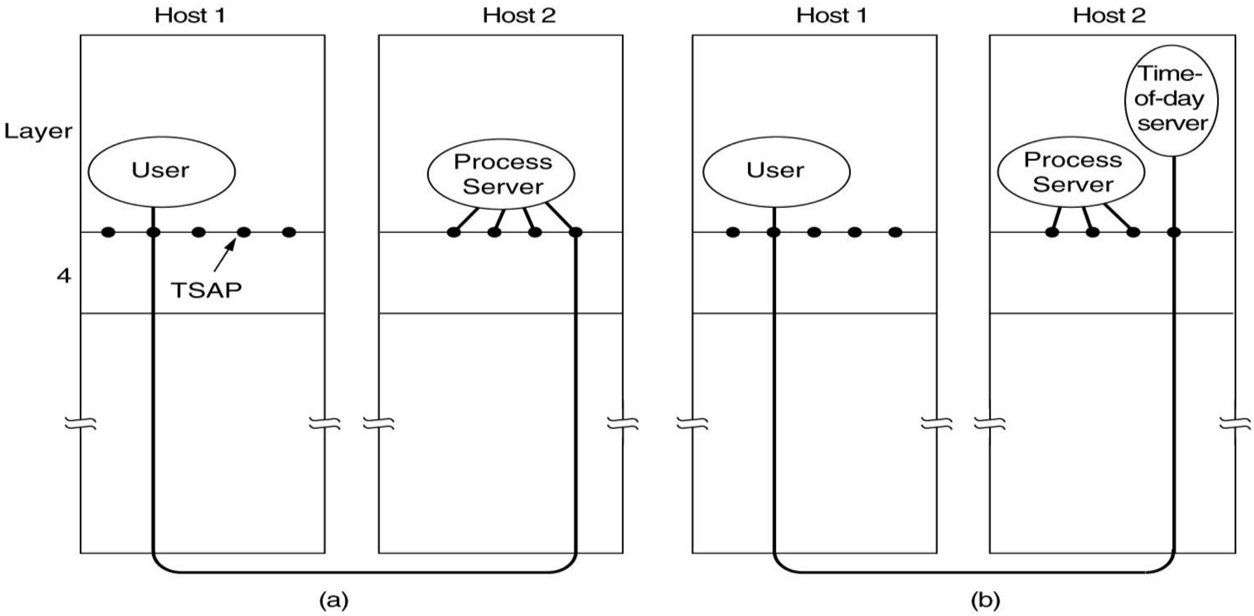
ELEMENTS OF TRANSPORT PROTOCOLS:

ADDRESSING

The method used to define transport addresses to which processes can establish connection is TSAP (Transport service access point). The network layer addresses are called NSAP (Network service access point). Both client and server attach TSAP to establish connection. The connections are run through NSAP's on each host. TSAP is needed to distinguish multiple transport end points that share that NSAP. The relationship between NSAP, TSAP and transport connections is shown below.

The host2 attaches TSAP 1522 to wait for incoming call. On host 1 it issues the connect request specifying the TSAP 1208 as source and TSAP 1522 as destination. The transport connection being established between the application process on host1 and sever1 on host2. The application process then sends over a request for the time. The time server process responds with the current time. The transport connection is then released. To solve the problem of stable TSAP address a simplified scheme is used.





The scheme is known as initial connection protocol. The process server act as a proxy for less heavily used servers. It listens to a set of ports at the same time for a connection request. If no server is waiting for them, they get a connection to process server. The process server executes the requested server to inherit the connection with the user. The new server does the requested work; the process server goes back to listening for new requests.

APPLICATION LAYER

DNS [DOMAIN NAME SYSTEM] Generally host names, mailboxes and other resources are represented by using ASCII sting such as rgm@vsnl.net.in.But the network itself only understands binary address i.e., the address written in the binary form. So we need some mechanism to convert the ASCII strings to network addresses in binary. It is easy to maintain the host names and their IP addresses in file for a network of few hundred hosts. For a network of thousand hosts it is very difficult.

The Domain Name System, DNS is a distributes data that is used by TCP/IP application to map between host names and IP addresses, and to provide electronic mail routing information. We use the term distributed because no single site on the Internet knows all the information. Each site maintains its own data base information and runs a server program that other systems (clients) across the Internet can query. It is a good example of a TCP/IP client-server application. The DNS provides the protocol that allows client and server to communicate with each other. DNS is defined in RFC's 1034 and 1035.

The DNS identifies each host on the internet with a unique name that identifies it as unambiguously as its IP address as follows. To map a name onto an IP address, an application program calls a library procedure called the resolver, passing it the name as a parameter. The 'resolver' sends a UDP packet to a local DNS server, which then looks up the name and returns the IP address to the resolver, which then returns it to the caller. To create names that are unique and at the same time decentralized and easy to change, the TCP/IP designers have chosen a hierarchical system made up of a number of labels separated by dots.

THE DNS NAME SPACE Internet is divided it several hundred top level domains, where each domain covers many hosts. Each domain is partitioned into sub domains, these are further partitioned and so on. Thus DNS is implemented using a tree in which each node represents one possible label of up to 63 characters. The root of the tree is a special node with new label as shown in fig. Any comparison of label considers uppercase and lower-case characters the same i.e., Domain names are case insensitive.

The leaves of the tree represent a company/organization and contain thousands of hosts. Each domain is named by the path from it to the unnamed root.

The components in the name are separated by periods (dots), that is domain name of any node in the tree is the list of labels starting at the node, working up to the root using the period (dot) separate the labels.

The domain names that ends with a period is called an absolute domain name or fully qualified domain name(FQDN).An example is vax.ugc,central.edu. If domain does not end with a period, it is assumed that the name needs to be completed.

How the name is completed on the DNS software being used. If the incomplete names consist of two or more labels, it might be considered to be complete. Otherwise, local addition might be added to the right of the name. The name vax might be completed by adding the local suffix.ugc.central.edu. The right most label in the name corresponds to the level of the tree closest to the root (lowest), and left-most to the level farthest from the root(highest).The tree is divided into three domains: generic, country and reverse as shown in fig.

Domain Name System Architecture

The Domain name system comprises of Domain Names, Domain Name Space, Name Server that have been described below:

Domain Names

Domain Name is a symbolic string associated with an IP address. There are several domain names available; some of them are generic such as com, edu, gov, net etc, while some country level domain names such as au, in, za, us etc.

The following table shows the **Generic** Top-Level Domain names:

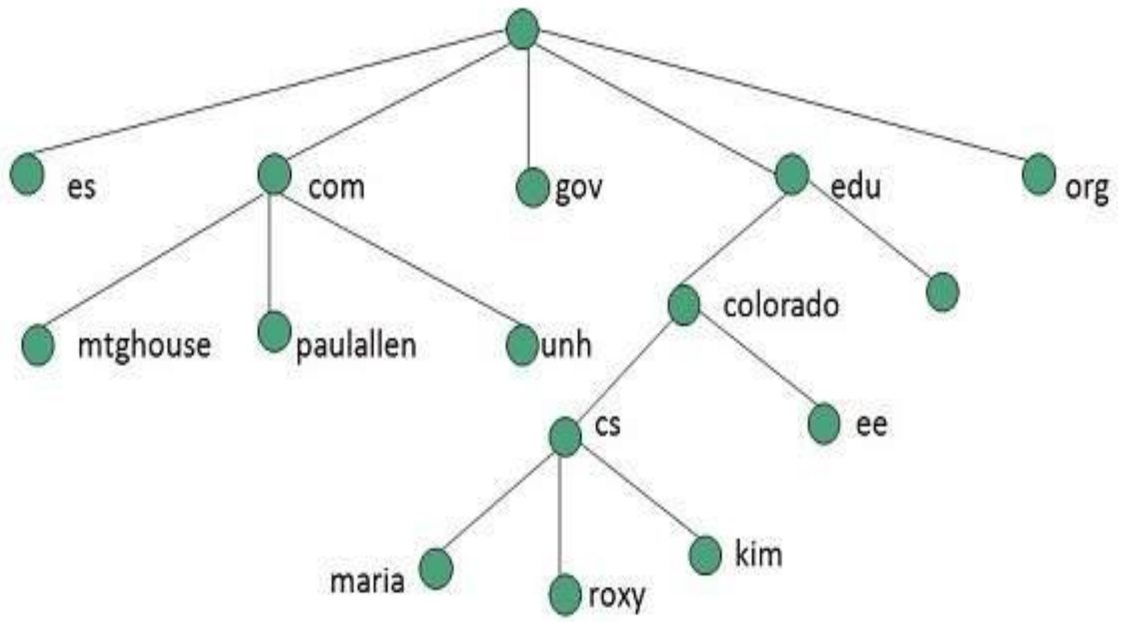
Domain Name	**Meaning**
Com	Commercial business
Edu	Education
Gov	U.S. government agency
Int	International entity
Mil	U.S. military
Net	Networking organization
Org	Non profit organization

The following table shows the **Country top-level** domain names:

Domain Name	Meaning
Au	Australia
In	India
Cl	Chile
Fr	France
Us	United States
Za	South Africa
Uk	United Kingdom
Jp	Japan
Es	Spain
De	Germany
Ca	Canada
Ee	Estonia
Hk	Hong Kong

Domain Name Space

The domain name space refers a hierarchy in the internet naming structure. This hierarchy has multiple levels (from 0 to 127), with a root at the top. The following diagram shows the domain name space hierarchy:



In the above diagram each subtree represents a domain. Each domain can be partitioned into sub domains and these can be further partitioned and so on.

RESOURCE RECORDS

- Resource Records define data types in the Domain Name System(DNS).
- Resource Records are stored in binary format internally for use by DNS software.
- But resource records are sent across a network in text format while they perform zone transfers.
- Every domain whether it is a single host or a top level domain can have a set of resource records associated with it
- Whenever a resolver (this will be explained later) gives the domain name to DNS it gets the resource record associated with it.
- So DNS can be looked upon as a service which maps domain names to resource records.
- Each resource has five fields & looks as below :-

Domain Name	Class	Type	Time to Live	Value

1. **Domain Name** :- the domain to which this record applies.
2. **Class** :- set to IN for internet information. For other information other codes may be specified.
3. **Type** :- tells what kind of record it is.
4. **Time to live** :- Upper Limit on the time to reach the destination.
5. **Value** :- can be an IP address, a string or a number depending on the record type.

EMAIL

Architecture and Services

In this section, we will provide an overview of how email systems are organized and what they can do. The architecture of the email system is shown in Fig. 7-7. It consists of two kinds of subsystems: the **user agents**, which allow people to read and send email, and the **message transfer agents**, which move the messages from the source to the destination. We will also refer to message transfer agents informally as **mail servers**.

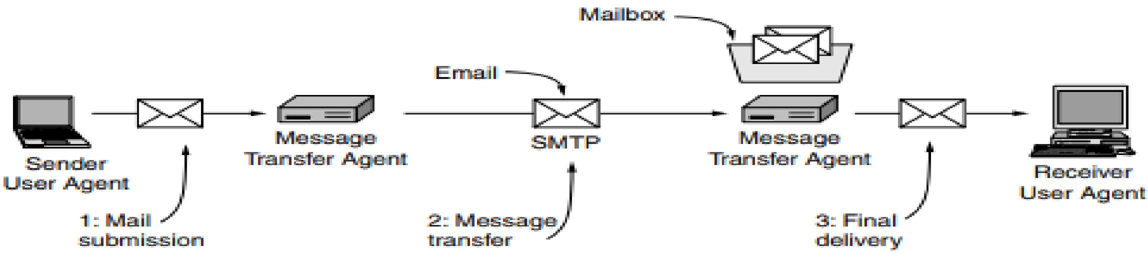


Figure 7-7. Architecture of the email system.

The user agent is a program that provides a graphical interface, or sometimes a text- and command-based interface that lets users interact with the email system. It includes a means to compose messages and replies to messages, display incoming messages, and organize messages by filing, searching, and discarding them. The act of sending new messages into the mail system for delivery is called **mail submission**.

Some of the user agent processing may be done automatically, anticipating what the user wants. For example, incoming mail may be filtered to extract or deprioritize messages that are likely spam. Some user agents include advanced features, such as arranging for automatic email responses ("I'm having a wonderful vacation and it will be a while before I get back to you"). A user agent runs on the same computer on which a user reads her mail. It is just another program and may be run only some of the time.

The message transfer agents are typically system processes. They run in the background on mail server machines and are intended to be always available. Their job is to automatically move email through the system from the originator to the recipient with **SMTP (Simple Mail Transfer Protocol)**. This is the message transfer step.

SMTP was originally specified as RFC 821 and revised to become the current RFC 5321. It sends mail over connections and reports back the delivery status and any errors. Numerous applications exist in which confirmation of delivery is important and may even have legal significance ("Well, Your Honor, my email system is just not very reliable, so I guess the electronic subpoena just got lost somewhere").

Message transfer agents also implement **mailing lists**, in which an identical copy of a message is delivered to everyone on a list of email addresses. Other advanced features are carbon copies, blind carbon copies, high-priority email, secret (i.e., encrypted) email, alternative recipients if the primary one is not currently available, and the ability for assistants to read and answer their bosses' email.

Linking user agents and message transfer agents are the concepts of mailboxes and a standard format for email messages. **Mailboxes** store the email that is received for a user. They are maintained by mail servers. User agents simply present users with a view of the contents of their mailboxes. To do this, the user agents send the mail servers commands to manipulate the mailboxes, inspecting their contents, deleting messages, and so on. The retrieval of mail is the final delivery (step 3) in Fig. With this architecture, one user may use different user agents on multiple computers to access one mailbox.

E-mail :

E-mail system consists of two subsystems
- The user agent, and
- The message transfer agents

User Agents :
They allow people to read and send e-mail they are local programs that provide a command based, menu based, or graphical method for interacting with e-mail system.

Message transfer agents :
They are responsible for moving the messages from the source to the destination. They are typically system daemons that run in the background and move e-mail throughthe system.

E-mail systems support five basic functions given below.

Composition refers to the process of creating messages and answers. For example, when answering a message, the e-mail system can extract the originator's address from the incoming e-mail and automatically insert it into the proper place in the reply.

Transfer refers to moving messages from the originator to the recipient. In large part, this requires establishing a connection to the destination or some intermediate machine, outputting the message, and releasing the connection. The e-mail system should do this automatically, without bothering the user.

Reporting has to do with telling the originator what happened to the message. Was it delivered? Was it rejected? Was it lost?

Displaying incoming messages is needed so people can read their e-mail. Sometimes conversion is required or a special viewer must be invoked, for example, if the message is a PostScript file or digitized voice. Simple conversions and formatting are sometimes attempted as well.

Disposition is the final step and concerns what the recipient does with the message after receiving it. Possibilities include throwing it away before reading, throwing it away after reading, saving it, and so on. It should also be possible to retrieve and reread saved messages, forward them, or process them in other ways.

Other Services of E-mail include:

Mailboxes : Used for storing incoming E-mail.
Mailing List = List of e-mail addresses to whom,
identical copies of messages need to be sent.
Registered E-mail = It allows the originator to know that his mail has arrived.
High priority E-mail = Secret E-mail etc.

User Agent : A user agent is normally a program that accepts a variety of commands for composing, receiving and replying to messages as well as manipulating mail boxes.

Sending E-mail : To send an e-mail a user must provide the message, the destination address and some other parameters. The message can be produced in any text editor (or) the one built in user agent. The destination address must be in the format that the user agent can deal with i.e., either DNS address (or) X.400 address. Most e-mail systems support mailing list, so that a user can send the same message to a list of people with a single command.

Reading E-mail : When a user agent is started up, it will look at the user’s mailbox for incoming e-mail before displaying anything on the screen. It then announces the number of messages in the mail box(or) a one line summary of each one. In a sophisticated system the user can specify the fields to be displayed by providing the display format.

Eg: 1. Message numbers
2. Flag etc.

Message format:

Message consist of a primitive envelope, some number of header field, blank line followed by message body. In normal usage, the user agent builds a message and passes it. To the message transfer agent which then uses some of the header fields to construct the actual envelope.

Principal header include:

To :
DNS address of primary recipient.

CC :
DNS address of secondary recipient. In terms of delivery there is no distinction between primary
and secondary (carbon copies).

BCC :
Similar to CC, allows people to send copies to third parties without primary and secondary knowing it.

From :
Who wrote the message.

Sender :
The one who sent the message.

Received :
Added by each message transfers agent along the way used for finding bugs in routing system.

Return path :

Added by final message transfer agent intended to tell how to get back to the sender etc.

MESSAGE FORMATS

Header	Meaning
To:	Email address(es) of primary recipient(s)
Cc:	Email address(es) of secondary recipient(s)
Bcc:	Email address(es) for blind carbon copies
From:	Person or people who created the message
Sender:	Email address of the actual sender
Received:	Line added by each transfer agent along the route
Return-Path:	Can be used to identify a path back to the sender

Header	Meaning
Date:	The date and time the message was sent
Reply-To:	Email address to which replies should be sent
Message-Id:	Unique number for referencing this message later
In-Reply-To:	Message-Id of the message to which this is a reply
References:	Other relevant Message-Ids
Keywords:	User-chosen keywords
Subject:	Short summary of the message for the one-line display

MESSAGE HEADERS BY MIME (Multipurpose Internet Mail Extensions)

Header	Meaning
MIME-Version:	Identifies the MIME version
Content-Description:	Human-readable string telling what is in the message
Content-Id:	Unique identifier
Content-Transfer-Encoding:	How the body is wrapped for transmission
Content-Type:	Type and format of the content

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THANK YOU