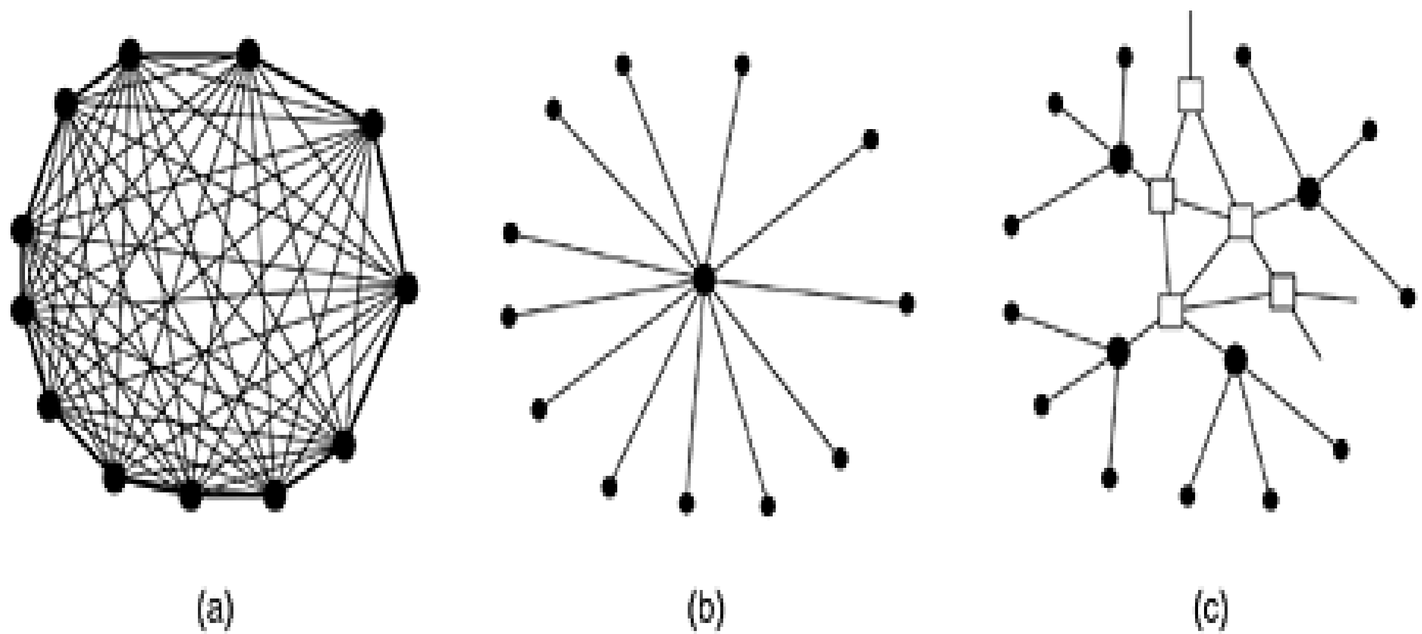
2.5 The Public Switched Telephone Network

The Public Switched Telephone Network

- When two computers owned by the same company or organization and located close to each other need to communicate, it is often easiest just to run a cable between them. LANs work this way.
- However, when the distances are large or there are many computers or the cables have to pass through a public road or other public right of way, the costs of running private cables are usually prohibitive.
- Consequently, the network designers must rely on the existing telecommunication facilities.
- These facilities, especially the PSTN (Public Switched Telephone Network).
- PSTN were usually designed, with a completely different goal in mind: transmitting the human voice in a more-or-less recognizable form.

2.5.1 Structure of the Telephone System

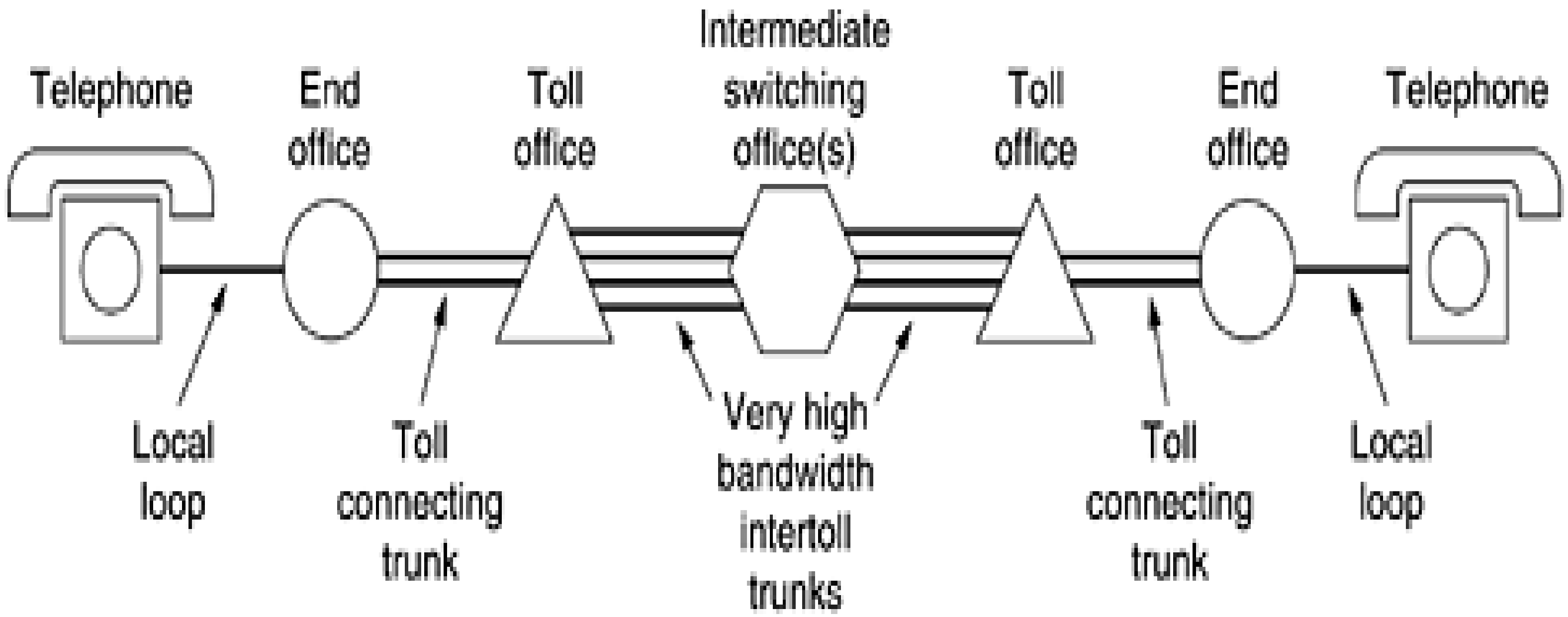
- **Figure 2-20. (a) Fully-interconnected network. (b) Centralized switch. (c) Two-level hierarchy.**



- The three major parts of the telephone system are:

- ❑ The switching offices,
- ❑ The wires between the customers and the switching offices
- ❑ The long-distance connections between the switching offices.

- **Figure 2-21. A typical circuit route for a medium-distance call.**



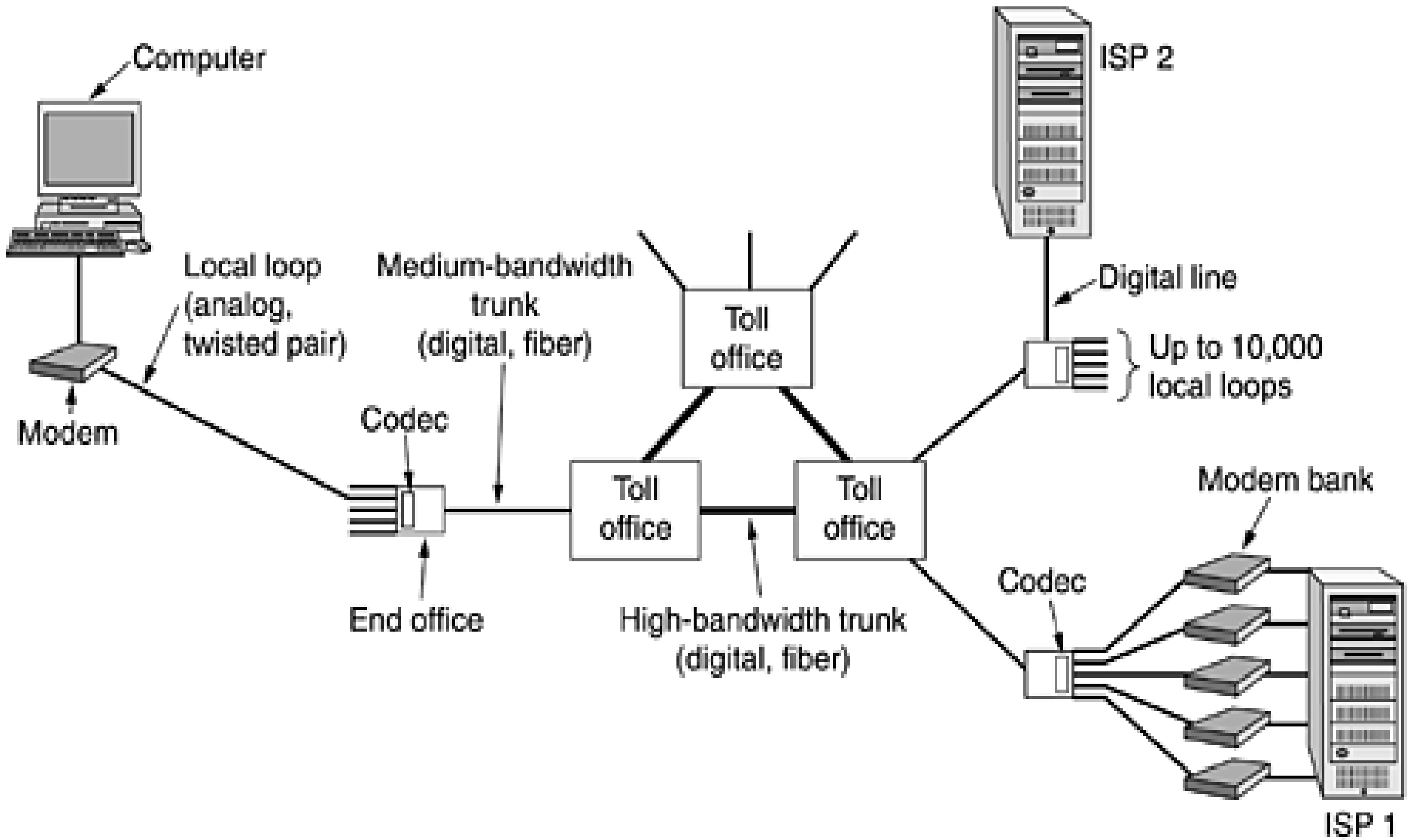
- Each telephone has two copper wires coming out of it that go directly to the telephone company's nearest end office (also called a local central office).
- The two-wire connections between each subscriber's telephone and the end office are known in the trade as the **local loop.**
- If a subscriber attached to a given end office calls another subscriber attached to the same end office, the switching mechanism within the office sets up a direct electrical connection between the two local loops.
- 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.
- These lines are called toll connecting trunks. If both the caller's and callee's end offices happen to have a toll connecting trunk to the same toll office (a likely occurrence if they are relatively close by), the connection may be established within the toll office.

- If the caller and callee do not have a toll office in common, the path will have to be established somewhere higher up in the hierarchy.
- Primary, sectional, and regional offices form a network by which the toll offices are connected.
- The toll, primary, sectional, and regional exchanges communicate with each other via high- bandwidth intertoll trunks (also called interoffice trunks).
- A variety of transmission media are used for telecommunication. Local loops consist of twisted pairs nowadays.
- Between switching offices, coaxial cables, microwaves, and especially fiber optics are widely used.
- In the past, transmission throughout the telephone system was analog, with the actual voice signal being transmitted as an electrical voltage from source to destination.
- With the advent of fiber optics, digital electronics, and computers, all the trunks and switches are now digital.

- In summary, the telephone system consists of three major components:
  - **Local loops** (analog twisted pairs going into houses and businesses).
  - **Trunks** (digital fiber optics connecting the switching offices).
  - **Switching offices** (where calls are moved from one trunk to another).
- The local loops provide everyone access to the whole system.
- For the long-haul trunks, the main issue is how to collect multiple calls together and send them out over the same fiber.

2.5.3 The Local Loop: Modems, ADSL, and Wireless

- **Figure 2-23. The use of both analog and digital transmission for a computer to computer call. Conversion is done by the modems and codecs.**



- An end office has up to 10,000 local loops.
- The area code + exchange indicated the end office.
- Example:(212) 601-xxxx was a specific end office with 10,000 subscribers, numbered 0000 through 9999.
- The two-wire local loop coming from a telephone company end office into houses and small businesses.
- The local loop is also frequently referred to as the ''last mile,'' although the length can be up to several miles.
- It has used analog signaling for over 100 years and is likely to continue doing so for some years to come, due to the high cost of converting to digital.
- When a computer wishes to send digital data over an analog dial-up line, the data must first be converted to analog form for transmission over the local loop. This conversion is done by a device called a modem.

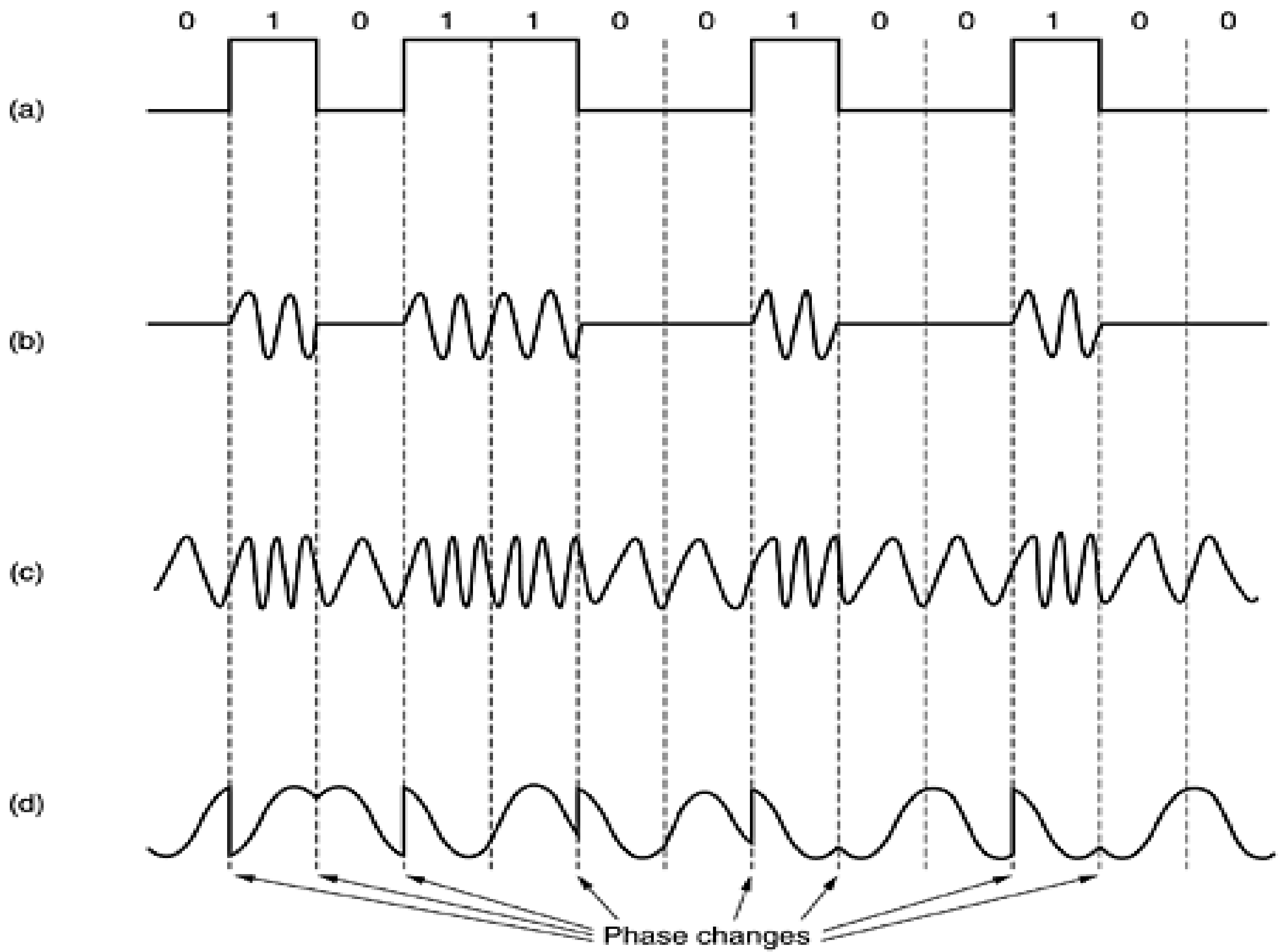
- At the telephone company end office the data are converted to digital form for transmission over the long-haul trunks.
- If the other end is a computer with a modem, the reverse conversion—digital to analog—is needed to traverse the local loop at the destination.
- ISP 1 (Internet Service Provider), which has a bank of modems, each connected to a different local loop.
- This ISP can handle as many connections as it has modems. (assuming its server or servers have enough computing power).
- Analog signaling consists of varying a voltage with time to represent an information stream.
- If transmission media were perfect, the receiver would receive exactly the same signal that the transmitter sent.
- Unfortunately, media are not perfect, so the received signal is not the same as the transmitted signal.
- For digital data, this difference can lead to errors.

- Transmission lines suffer from three major problems: **attenuation, delay distortion, and noise.**
- Attenuation is the loss of energy as the signal propagates outward. The loss is expressed in decibels per kilometer. The amount of energy lost depends on the frequency.
- To see the effect of this frequency dependence, imagine a signal not as a simple waveform, but as a series of Fourier components.
- Each component is attenuated by a different amount, which results in a different Fourier spectrum at the receiver.
- The different Fourier components also propagate at different speeds in the wire. This speed difference leads to distortion of the signal received at the other end.
- Another problem is noise, which is unwanted energy from sources other than the transmitter.
- Thermal noise is caused by the random motion of the electrons in a wire and is unavoidable.
- Crosstalk is caused by inductive coupling between two wires that are close to each other.
- Finally, there is impulse noise, caused by spikes on the power line or other causes. For digital data, impulse noise can wipe out one or more bits.

Modems

- Both attenuation and propagation speed are frequency dependent, it is undesirable to have a wide range of frequencies in the signal.
- Unfortunately, the square waves used in digital signals have a wide frequency spectrum and thus are subject to strong attenuation and delay distortion.
- These effects make baseband (DC) signaling unsuitable except at slow speeds and over short distances.
- To get around the problems associated with DC signaling, especially on telephone lines, AC signaling is used.
- A continuous tone in the 1000 to 2000-Hz range, called a sine wave carrier, is introduced.

- **Figure 2-24. (a) A binary signal. (b) Amplitude modulation. (c) Frequency modulation. (d) Phase modulation.**

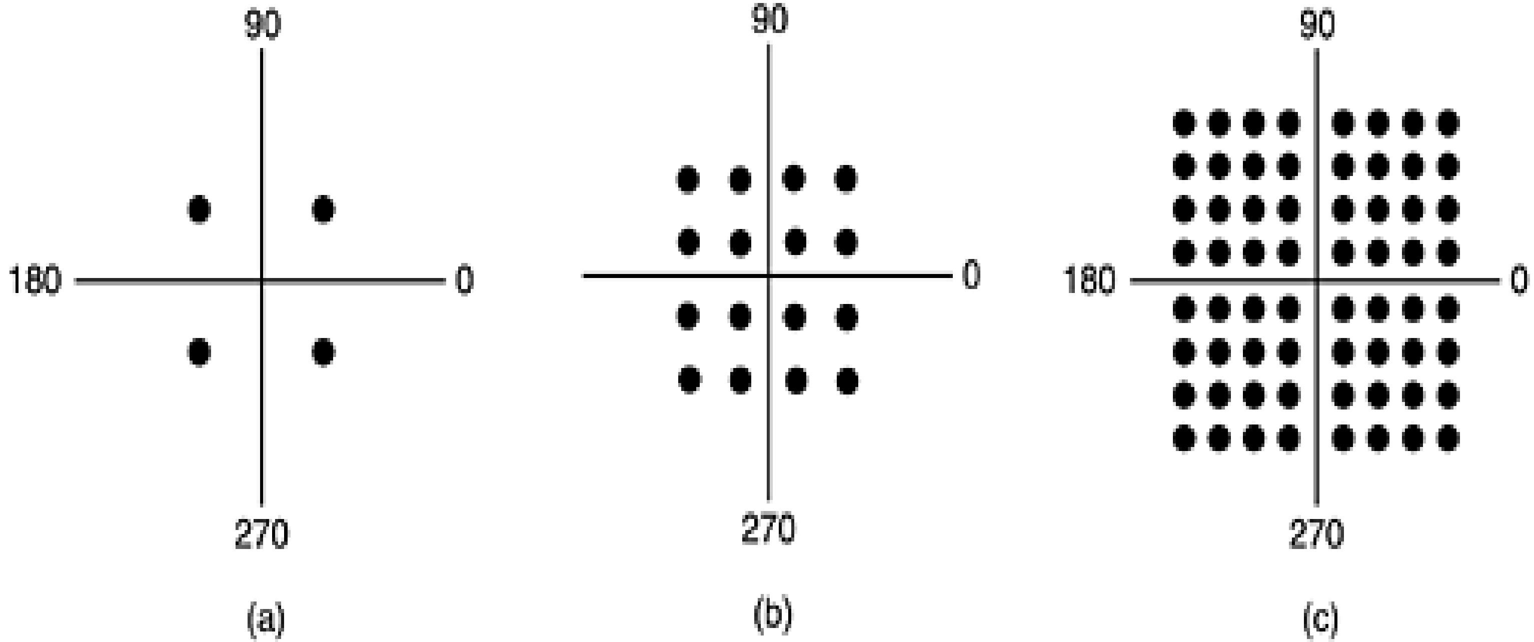


- Its amplitude, frequency, or phase can be modulated to transmit information.
- In amplitude modulation, two different amplitudes are used to represent 0 and 1, respectively.
- In frequency modulation, also known as frequency shift keying, two (or more) different tones are used. (The term keying is also widely used in the industry as a synonym for modulation.)
- In the simplest form of phase modulation, the carrier wave is systematically shifted 0 or 180 degrees at uniformly spaced intervals.
- A device that accepts a serial stream of bits as input and produces a carrier modulated by one (or more) of these methods (or vice versa) is called a modem (for modulator-demodulator).
- The modem is inserted between the (digital) computer and the (analog) telephone system.

- To go to higher and higher speeds, it is not possible to just keep increasing the sampling rate.
- The **Nyquist theorem** says that even with a perfect 3000-Hz line , there is no point in sampling faster than 6000 Hz.
- **In practice, most modems sample 2400 times/sec and focus on getting more bits per sample.**
- **The number of samples per second is measured in baud.**
- In telecommunication and electronics, baud is a common unit of measurement of symbol rate,
- Baud is one of the components that determine the speed of communication over a data channel.
- During each baud, one symbol is sent. Thus, an n- baud line transmits n symbols/sec.
- For example, a 2400-baud line sends one symbol about every 416.667 μsec.
- If the symbol consists of 0 volts for a logical 0 and 1 volt for a logical 1, the bit rate is 2400 bps.
- If, however, the voltages 0, 1, 2, and 3 volts are used, every symbol consists of 2 bits, so a 2400-baud line can transmit 2400 symbols/sec at a data rate of 4800 bps.

- Similarly, with four possible phase shifts, there are also 2 bits/symbol, so again here the bit rate is twice the baud rate. This technique is widely used and called **QPSK (Quadrature Phase Shift Keying).**
- The concepts of **bandwidth, baud, symbol, and bit rate** are commonly confused.
- The **bandwidth** of a medium is the range of frequencies that pass through it with minimum attenuation. It is a physical property of the medium (usually from 0 to some maximum frequency) and measured in Hz.
- The **baud rate** is the number of samples/sec made. Each sample sends one piece of information, that is, **one symbol**. The baud rate and symbol rate are thus the same.
- The modulation technique (e.g., QPSK) determines the number of bits/symbol.
- The bit rate is the amount of information sent over the channel and is equal to the number of symbols/sec times the number of bits/symbol.
- All advanced modems use a combination of modulation techniques to transmit multiple bits per baud.
- Often multiple amplitudes and multiple phase shifts are combined to transmit several bits/symbol.

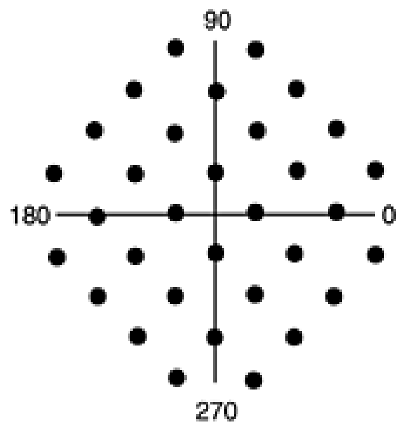
- **Figure 2-25. (a) QPSK. (b) QAM-16. (c) QAM-64.**



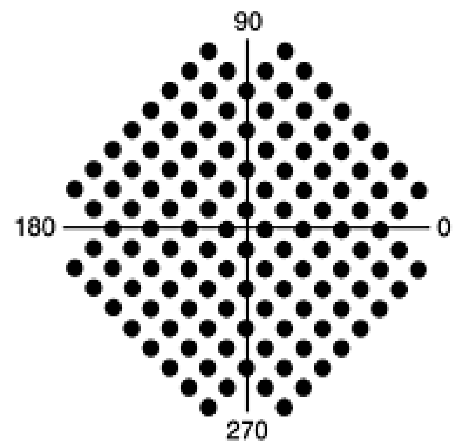
- Fig. 2-25(a) has four valid combinations and can be used to transmit 2 bits per symbol. It is QPSK.
- Fig. 2-25(b) has four amplitudes and four phases are used, for a total of 16 different combinations. This modulation scheme can be used to transmit 4 bits per symbol. It is called QAM-16 (Quadrature Amplitude Modulation).
- Figure 2-25(c) allows 64 different combinations, so 6 bits can be transmitted per symbol. It is called QAM-64. Higher-order QAMs also are used.
- Diagrams such as those of Fig. 2-25, which show the legal combinations of amplitude and phase, are called **constellation diagrams.**

- Each high-speed modem standard has its own constellation pattern and can talk only to other modems that use the same one (although most modems can emulate all the slower ones).
- With many points in the constellation pattern, even a small amount of noise in the detected amplitude or phase can result in an error and, potentially, many bad bits.
- To reduce the chance of an error, standards for the higher speeds modems do error correction by adding extra bits to each sample. The schemes are known as TCM (Trellis Coded Modulation).
- The V.32 modem standard uses 32 constellation points to transmit 4 data bits and 1 parity bit per symbol at 2400 baud to achieve 9600 bps with error correction.
- Its constellation pattern is shown in Fig. 2-26(a). The decision to ''rotate'' around the origin by 45 degrees was done for engineering reasons; the rotated and unrotated constellations have the same information capacity.
- The next step above 9600 bps is 14,400 bps. It is called V.32 bis. This speed is achieved by transmitting 6 data bits and 1 parity bit per sample at 2400 baud.

- **Figure 2-26. (a) V.32 for 9600 bps. (b) V32 bis for 14,400 bps.**



(b)



(c)

- Its constellation pattern has 128 points when QAM-128 is used and is shown in Fig. 2-26(b).
- Fax modems use this speed to transmit pages. QAM-256 is not used in any standard telephone modems, but it is used on cable networks.
- The next telephone modem after V.32 bis is V.34, which runs at 28,800 bps at 2400 baud with 12 data bits/symbol.
- The final modem in this series is V.34 bis which uses 14 data bits/symbol at 2400 baud to achieve 33,600 bps.

- To increase the effective data rate further, many modems compress the data before transmitting it, to get an effective data rate higher than 33,600 bps.
- All modern modems allow traffic in both directions at the same time (by using different frequencies for different directions).
- A connection that allows traffic in both directions simultaneously is called full duplex. A two-lane road is full duplex.
- A connection that allows traffic either way, but only one way at a time is called half duplex. A single railroad track is half duplex.
- A connection that allows traffic only one way is called simplex. A one-way street is simplex. Another example of a simplex connection is an optical fiber with a laser on one end and a light detector on the other end.

- The reason that standard modems stop at 33,600 is that the Shannon limit for the telephone system is about 35 kbps, so going faster than this would violate the laws of physics.
- whether 56-kbps modems are theoretically possible?
- But why is the theoretical limit 35 kbps? It has to do with the average length of the local loops and the quality of these lines. The 35 kbps is determined by the average length of the local loops.
- A call originating at the computer on the left and terminating at ISP1 goes over two local loops as an analog signal, once at the source and once at the destination. Each of these adds noise to the signal. If we could get rid of one of these local loops, the maximum rate would be doubled.

- ISP 2 does precisely that. It has a pure digital feed from the nearest end office. The digital signal used on the trunks is fed directly to ISP 2, eliminating the codecs, modems, and analog transmission on its end.
- Thus, when one end of the connection is purely digital, 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 33.6 kbps.
- The reason that 56 kbps modems are in use has to do with the Nyquist theorem.
- The telephone channel is about 4000 Hz wide (including the guard bands). The maximum number of independent samples per second is thus 8000. The number of bits per sample in the U.S. is 8, one of which is used for control purposes, allowing 56,000 bit/sec of user data.

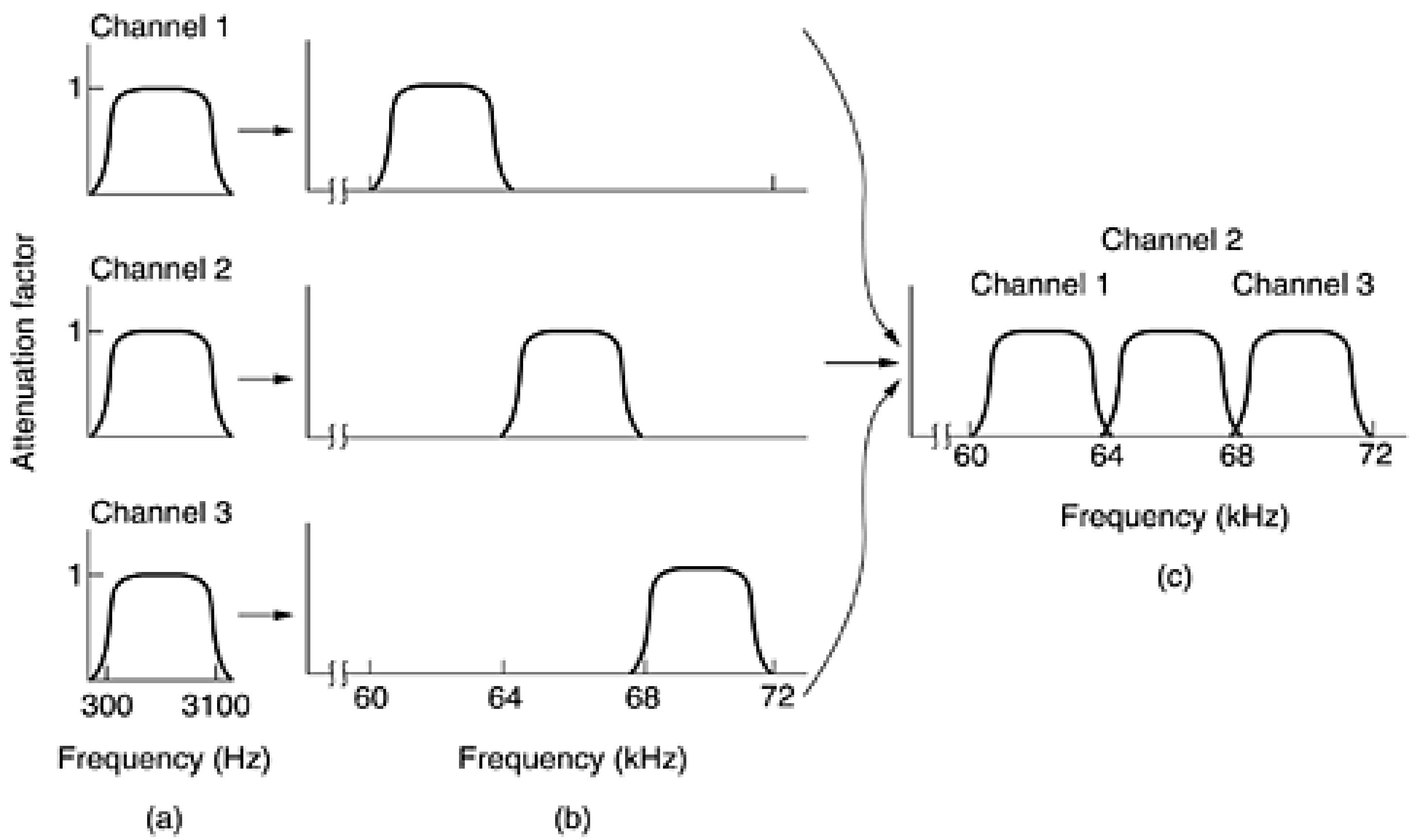
- In Europe, all 8 bits are available to users, so 64,000-bit/sec modems could have been used, but to get international agreement on a standard, 56,000 was chosen.
- This modem standard is called V.90. It provides for a 33.6-kbps upstream channel (user to ISP), but a 56 kbps downstream channel (ISP to user) because there is usually more data transport from the ISP to the user than the other way (e.g., requesting a Web page takes only a few bytes, but the actual page could be megabytes).
- In theory, an upstream channel wider than 33.6 kbps would have been possible, but since many local loops are too noisy for even 33.6 kbps, it was decided to allocate more of the bandwidth to the downstream channel to increase the chances of it actually working at 56 kbps.
- The next step beyond V.90 is V.92. These modems are capable of 48 kbps on the upstream channel if the line can handle it.

Trunks and Multiplexing

- Many conversations over a single physical trunk.
- 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 (in a round-robin fashion), each one periodically getting the entire bandwidth for a little burst of time.
- AM radio broadcasting provides illustrations of both kinds of multiplexing.

Frequency Division Multiplexing

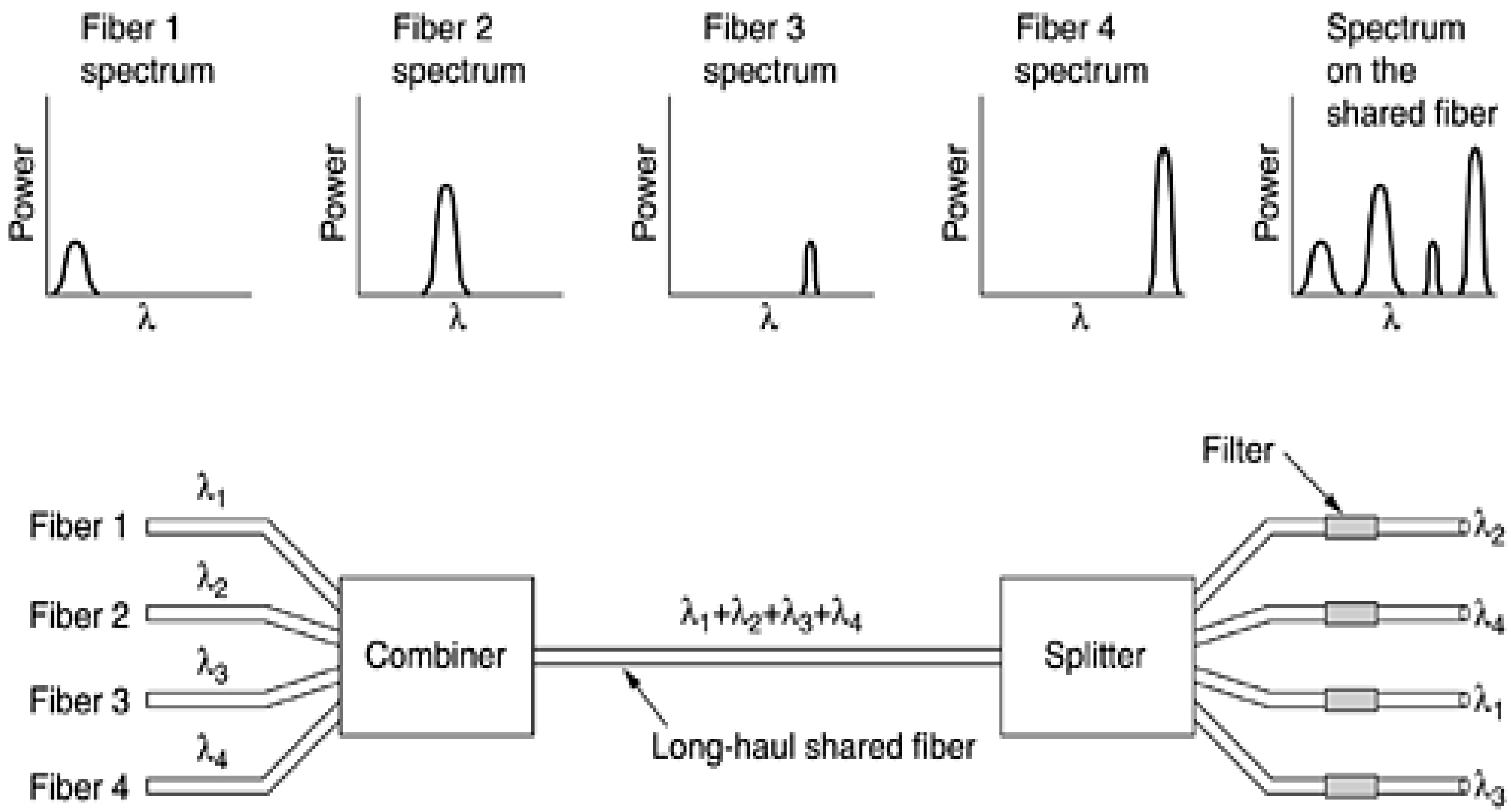
- **Figure 2-31. Frequency division multiplexing. (a) The original bandwidths. (b) The bandwidths raised in frequency. (c) The multiplexed channel.**



- 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.
- 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 to 108 kHz band. This unit is called a group.
- The 12-kHz to 60- kHz band is sometimes used for another group. Many carriers 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 supergroup.
- The next unit is the mastergroup, which is five supergroups (CCITT standard) or ten supergroups (Bell system). Other standards of up to 230,000 voice channels also exist.

Wavelength Division Multiplexing

- **Figure 2-32. Wavelength division multiplexing.**



- For fiber optic channels, a variation of frequency division multiplexing is used. It is called WDM (Wavelength Division Multiplexing).
- Many fibers come together at an optical combiner, each with its energy present at a different wavelength.
- Many 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.

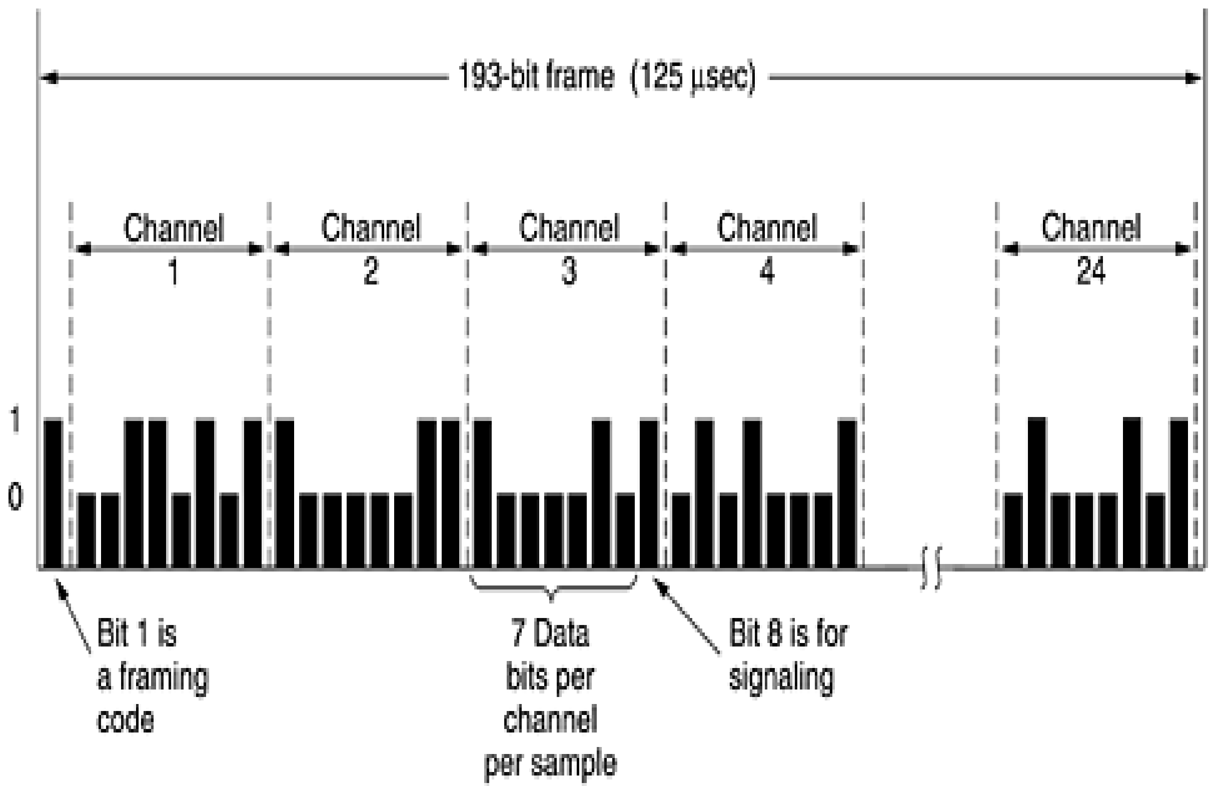
Time Division Multiplexing

- Although FDM is still used over copper wires or microwave channels, it requires analog circuitry and is not suitable for digital data.
- 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..
- 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).

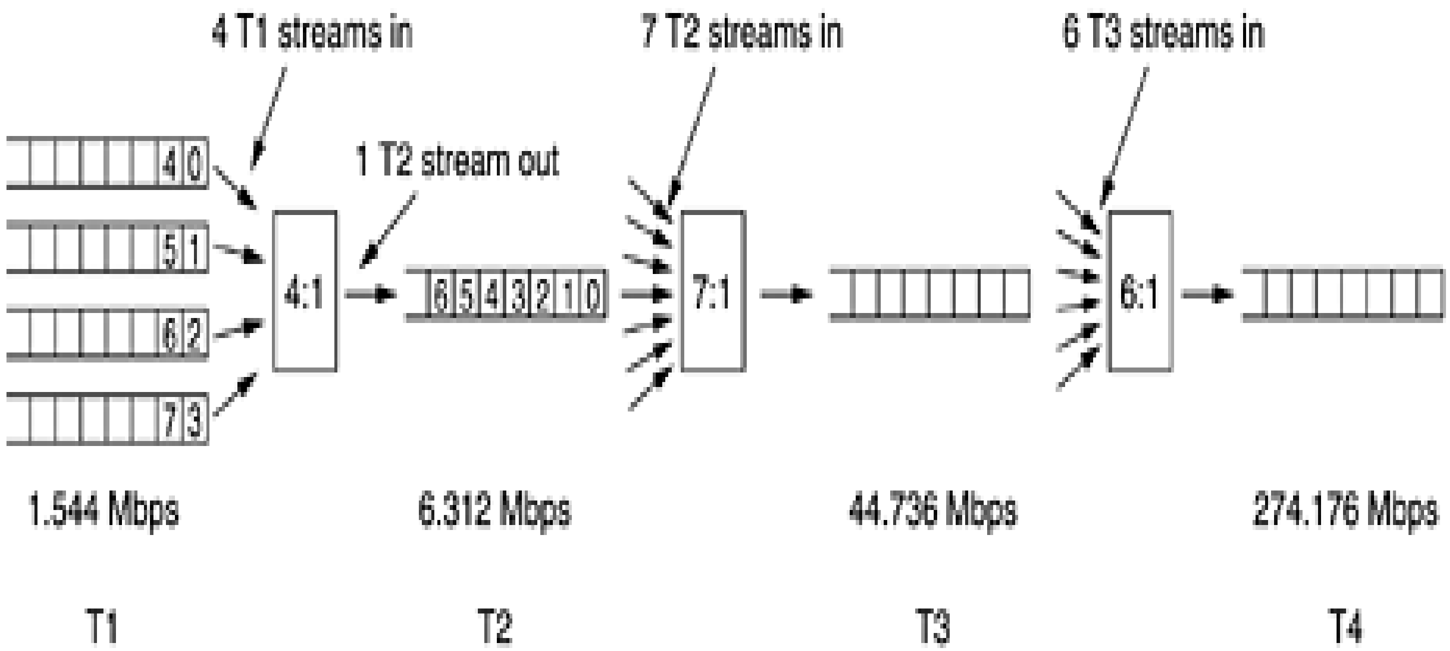
- PCM forms the heart of the modern telephone system. As a consequence, virtually all time intervals within the telephone system are multiples of 125 µsec.
- No international standard for PCM, A variety of incompatible schemes are now in use in different countries around the world.
- The method used in North America and Japan is the T1 carrier.
- The T1 carrier consists of 24 voice channels multiplexed together.
- Usually, the analog signals are sampled on a round-robin basis with the resulting analog stream being fed to the codec rather than having 24 separate codecs and then merging the digital output.

- 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 7 x 8000 = 56,000 bps of data, and 1 x 8000 = 8000 bps of signaling information per channel.
- 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 193/125 µsec = 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.

- **Figure 2-33. The T1 carrier (1.544 Mbps).**



- **Figure 2-35. Multiplexing T1 streams onto higher carriers.**



- Time division multiplexing allows multiple T1 carriers to be multiplexed into higher-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, so they are not well known.
- At the next level, seven T2 streams are combined to form a T3 stream. Then six T3 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

- In the early days of fiber optics, every telephone company had its own proprietary optical TDM system.
- After AT&T was broken up in 1984, local telephone companies had to connect to multiple long-distance carriers, all with different optical TDM systems, so the need for standardization became obvious.

❑ SONET (Synchronous Optical NETwork) - 1985

❑ SDH (Synchronous Digital Hierarchy) -1989

- Virtually all the long-distance telephone traffic in the United States, and much of it elsewhere, now uses trunks running SONET in the physical layer.

Switching

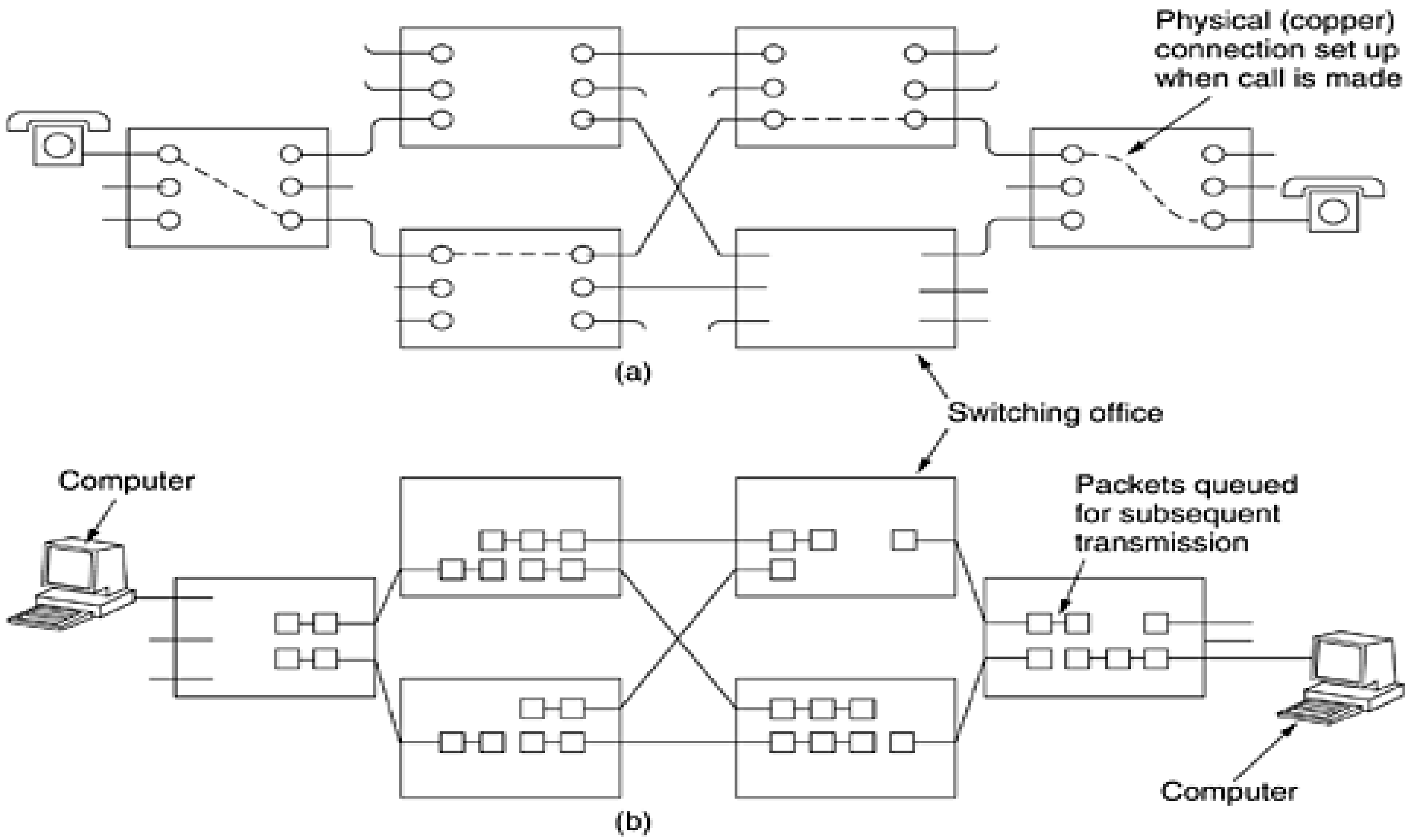
- The phone system is divided into two principal parts:

- ❑ outside plant (the local loops and trunks, since they are physically outside the switching offices)
- ❑ inside plant (the switches), which are inside the switching offices.

- Two different switching techniques are used nowadays: circuit switching and packet switching.

Circuit Switching

Figure 2-38. (a) Circuit switching. (b) Packet 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.
- Each of the six rectangles represents a carrier switching office (end office, toll office, etc.).
- When a call passes through a switching office, a physical connection is established between the line on which the call came in and one of the output lines.
- In the early days of the telephone, the connection was made by the operator plugging a jumper cable into the input and output sockets.
- Now it is replaced by automatic telephone switching equipment.
- The alternative to circuit switching is packet switching.
- With this technology, individual packets are sent as need be, with no dedicated path being set up in advance. It is up to each packet to find its way to the destination on its own
- An important property of circuit switching is the need to set up an end-to-end path before any data can be sent.
- Once a call has been set up, a dedicated path between both ends exists and will continue to exist until the call is finished.

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 (i.e., router) and then forwarded later, one hop at a time.
- Each block is received in its entirety, inspected for errors, and then retransmitted. A network using this technique is called a store-and-forward network.
- The first electromechanical telecommunication systems used message switching, namely, for telegrams.

Packet Switching

- With message switching, there is no limit at all on block size, which means that routers (in a modern system) must have disks to buffer long blocks.
- It also means that a single block can tie up a router-router line for minutes, rendering message switching useless for interactive traffic.
- To get around these problems, packet switching was invented.
- Packet-switching networks place a tight 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 (milliseconds), packet-switching networks are well suited for handling interactive traffic.
- A further advantage of packet switching over message switching is: the first packet of a multipacket 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.

A comparison of circuit-switched and packet-switched networks.

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

- circuit switching requires that a circuit be set up end to end before communication begins. Packet switching does not require any advance setup. The first packet can just be sent as soon as it is available.
- The result of the connection setup with circuit switching is the reservation of bandwidth all the way from the sender to the receiver. All packets follow this path. Among other properties, having all packets follow the same path means that they cannot arrive out of order. With packet switching there is no path, so different packets can follow different paths, depending on network conditions at the time they are sent. They may arrive out of order.
- Packet switching is more fault tolerant than circuit switching.
- Setting up a path in advance also opens up the possibility of reserving bandwidth in advance. If bandwidth is reserved, then when a packet arrives, it can be sent out immediately over the reserved bandwidth. With packet switching, no bandwidth is reserved, so packets may have to wait their turn to be forwarded.
- congestion can occur at different times with circuit switching (at setup time) and packet switching (when packets are sent).
- 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.

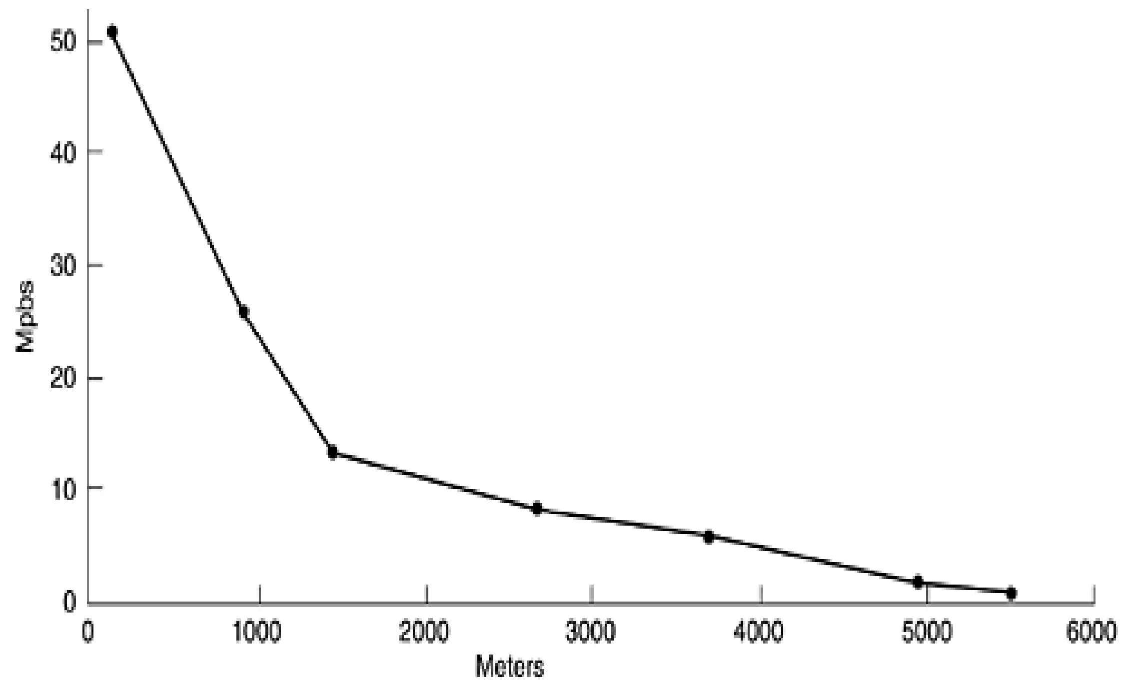
- Packet switching uses store-and-forward transmission. A packet is accumulated in a router's memory, then sent on to the next router. With circuit switching, the bits just flow through the wire continuously. The store- and- forward technique adds delay.
- Another difference is that circuit switching 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.

Digital Subscriber Lines

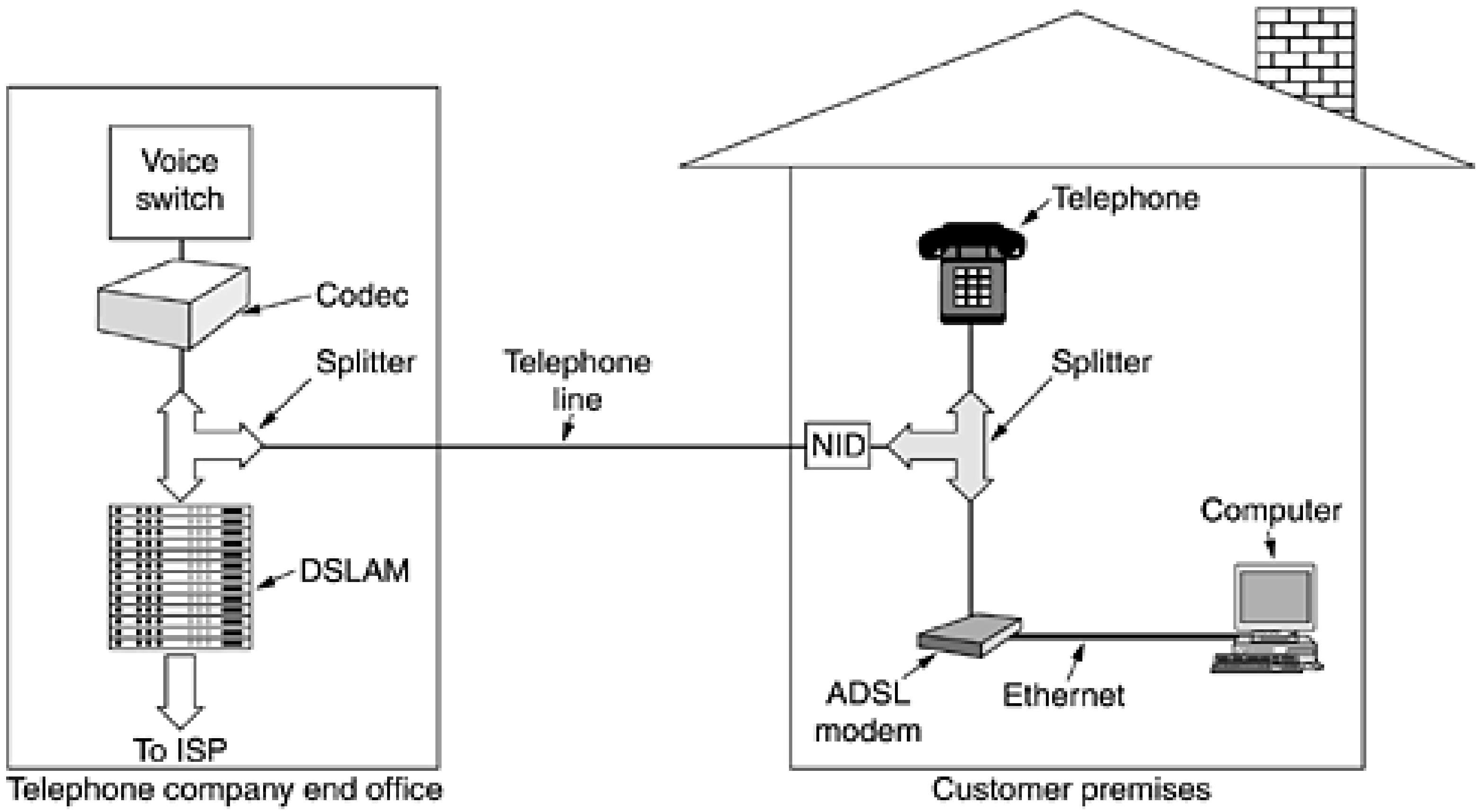
- When the telephone industry finally got to 56 kbps. Meanwhile, the cable TV industry was offering speeds up to 10 Mbps on shared cables, and satellite companies were planning to offer upward of 50 Mbps.
- As Internet access became an increasingly important part of their business, the telephone companies (LECs) began to realize they needed a more competitive product. Their answer was to start offering new digital services over the local loop.
- Services with more bandwidth than standard telephone service are sometimes called broadband, although the term really is more of a marketing concept than a specific technical concept.
- Initially, there were many overlapping offerings, all under the general name of xDSL (Digital Subscriber Line), for various x.
- The most popular of these services, ADSL (Asymmetric DSL).

- In telephone system, At the point where each local loop terminates in the end office, the wire runs through a filter that attenuates all frequencies below 300 Hz and above 3400 Hz.
- The cutoff is not sharp—300 Hz and 3400 Hz are the 3 dB points—so the bandwidth is usually quoted as 4000 Hz even though the distance between the 3 dB points is 3100 Hz.
- Data are thus also restricted to this narrow band.
- The trick that makes xDSL work is that when a customer subscribes to it, the incoming line is connected to a different kind of switch, one that does not have this filter, thus making the entire capacity of the local loop available.
- Unfortunately, the capacity of the local loop depends on several factors, including its length, thickness, and general quality.
- A plot of the potential bandwidth as a function of distance is given in Fig.

- The xDSL services have all been designed with certain goals in mind.
  - First, the services must work over the existing category 3 twisted pair local loops.
  - Second, they must not affect customers' existing telephones and fax machines.
  - Third, they must be much faster than 56 kbps.
  - Fourth, they should be always on, with just a monthly charge but no per-minute charge.



A typical ADSL equipment configuration.

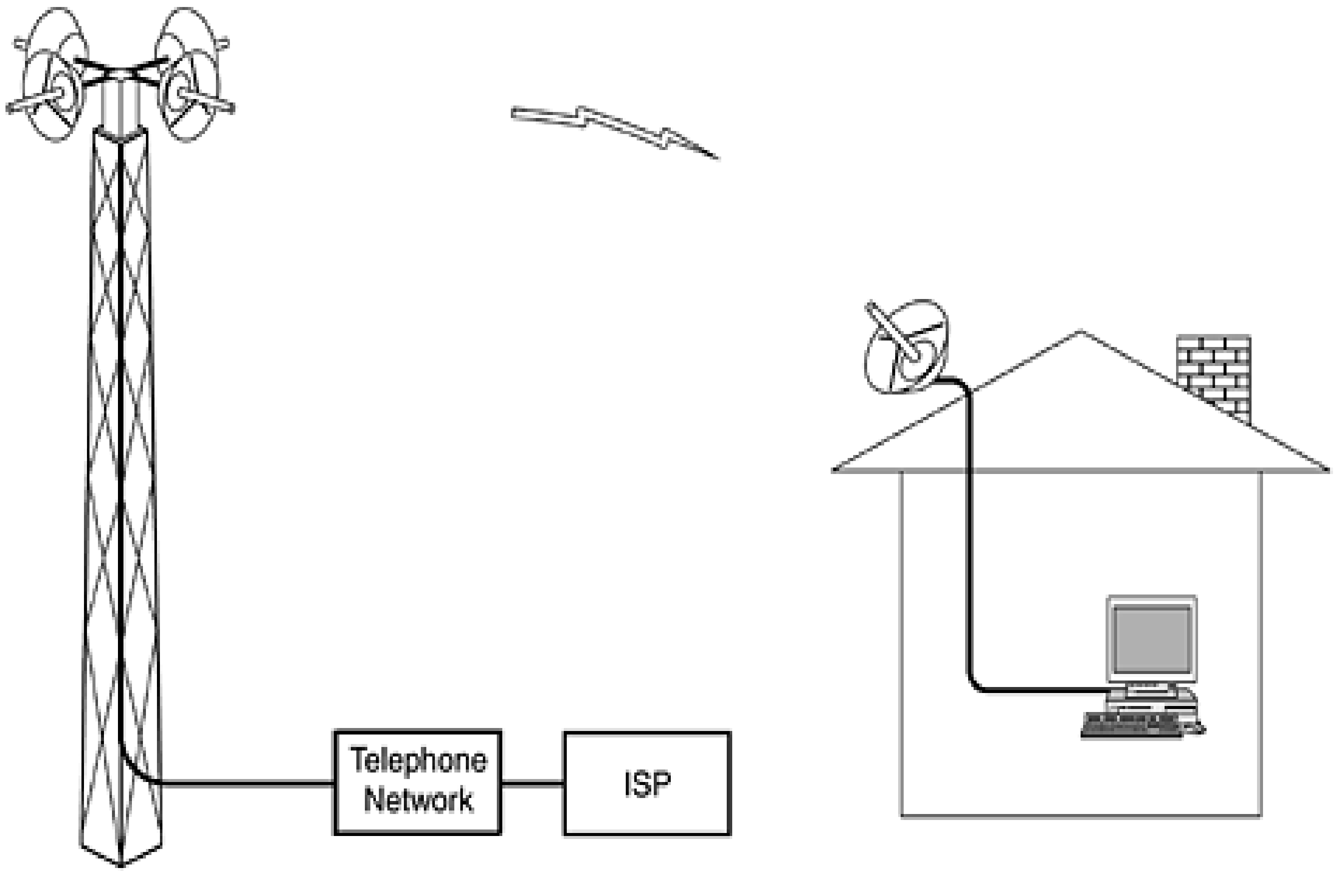


- A typical ADSL arrangement is shown in <u>Fig. 2-29</u>.
- In this scheme, a telephone company technician must install a NID (Network Interface Device) on the customer's premises. Close to the NID (or sometimes combined with it) is a splitter, an analog filter that separates the 0-4000 Hz band.
- The signal is routed to the existing telephone or fax machine, and the data signal is routed to an ADSL modem.
- The ADSL modem is actually a digital signal processor that has been set up to act as 250 QAM modems operating in parallel at different frequencies.
- 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 digital signal has been recovered into a bit stream, packets are formed and sent off to the ISP.

Wireless Local Loops

- Local loop is a circuit line from a subscriber’s phone to the local end office.
- But the implementation of local loop of wires is risky for the operators, especially in rural and remote areas due to less number of users and increased cost of installation.
- Hence, the solution for it is the usage of wireless local loop (WLL) which uses wireless links rather than copper wires to connect subscribers to the local central office.

Architecture of an LMDS (Local Multipoint Distribution Service) system.



- Tower with multiple antennas on it, each pointing in a different direction.
- Since millimeter waves are highly directional, each antenna defines a sector, independent of the other ones.
- At this frequency, the range is 2–5 km, which means that many towers are needed to cover a city.
- With current technology, each sector can have 36 Gbps downstream and 1 Mbps upstream, shared among all the users in that sector.