CHAPTER

3 Underlying Technologies

We can think of the Internet as a series of backbone networks that are run by international, national, and regional ISPs. The backbones are joined together by connecting devices such as routers or switching stations. The end users are either part of the local ISP LAN or connected via point-to-point networks to the LANs. Conceptually, the Internet is a set of switched WANs (backbones), LANs, point-to-point WANs, and connecting or switching devices.

Although the TCP/IP Protocol Suite is normally shown as a five-layer stack, it only defines the three upper layers; TCP/IP is only concerned with the network, transport, and application layers. This means that TCP/IP assumes the existence of these WANs, LANs, and the connecting devices that join them.

As a brief review, we touch upon some of these underlying technologies in this chapter.

OBJECTIVES

The chapter has several objectives:

- To briefly discuss the technology of dominant wired LANs, Ethernet, including traditional, fast, gigabit, and ten-gigabit Ethernet.
- To briefly discuss the technology of wireless WANs, including IEEE 802.11 LANs, and Bluetooth.
- To briefly discuss the technology of point-to-point WANs including 56K modems, DSL, cable modem, T-lines, and SONET.
- To briefly discuss the technology of switched WANs including X.25, Frame Relay, and ATM.
- To discuss the need and use of connecting devices such as repeaters (hubs), bridges (two-layer switches), and routers (three-layer switches).

3.1 WIRED LOCAL AREA NETWORKS

A local area network (LAN) is a computer network that is designed for a limited geographic area such as a building or a campus. Although a LAN can be used as an isolated network to connect computers in an organization for the sole purpose of sharing resources, most LANs today are also linked to a wide area network (WAN) or the Internet.

The LAN market has seen several technologies such as Ethernet, token ring, token bus, FDDI, and ATM LAN. Some of these technologies survived for a while, but Ethernet is by far the dominant technology.

In this section, we first briefly discuss the IEEE Standard Project 802, designed to regulate the manufacturing and interconnectivity between different LANs. We then concentrate on the Ethernet LANs.

Although Ethernet has gone through a four-generation evolution during the last few decades, the main concept has remained the same. Ethernet has changed to meet the market needs and to make use of the new technologies.

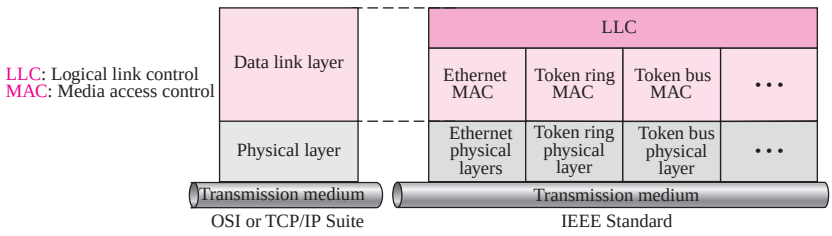
IEEE Standards

In 1985, the Computer Society of the IEEE started a project, called **Project 802,** to set standards to enable intercommunication among equipment from a variety of manufacturers. Project 802 does not seek to replace any part of the OSI or the Internet model. Instead, it is a way of specifying functions of the physical layer and the data link layer of major LAN protocols.

The standard was adopted by the American National Standards Institute (ANSI). In 1987, the International Standards Organization (ISO) also approved it as an international standard under the designation ISO 8802.

The relationship of the 802 Standard to the traditional OSI model is shown in Figure 3.1. The IEEE has subdivided the data link layer into two sublayers: **logical link control (LLC)** and **media access control** (**MAC**)**.** IEEE has also created several physical layer standards for different LAN protocols.

Figure 3.1 *IEEE standard for LANs*



In this text, however, we treat physical and data link layer together as the underlying technology supporting other layers in the TCP/IP protocol suite. For more details about physical and data link layer technology see Forouzan, *Data Communications and Networking,* 4th ed., McGraw-Hill, 2007.

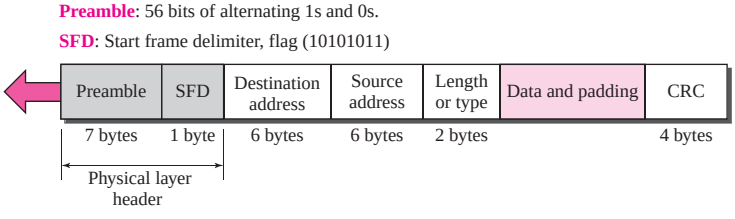
Frame Format

The packet sent in an Ethernet LAN is called a frame. In this section we discuss the format and the length of the frame that is used in our versions of the Ethernet.

Frame Format

The Ethernet frame contains seven fields: preamble, SFD, DA, SA, length or type of data unit, upper-layer data, and the CRC. Ethernet does not provide any mechanism for acknowledging received frames, making it what is known as an unreliable medium. Acknowledgments must be implemented at the higher layers. The format of the MAC frame is shown in Figure 3.2.

Figure 3.2 *Ethernet frame*



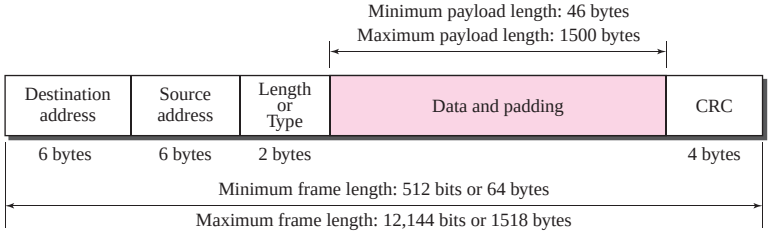
- **Preamble.** The first field of the 802.3 frame contains 7 bytes (56 bits) of alternating 0s and 1s that alerts the receiving system to the coming frame and enables it to synchronize its input timing. The pattern provides only an alert and a timing pulse. The 56-bit pattern allows the stations to miss some bits at the beginning of the frame. The preamble is actually added at the physical layer and is not (formally) part of the frame.
- **Start frame delimiter (SFD).** The second field (1 byte: 10101011) signals the beginning of the frame. The SFD warns the station or stations that this is the last chance for synchronization. The last 2 bits are 11 and alert the receiver that the next field is the destination address. The SFD is also added at the physical layer.
- **Destination address (DA).** The DA field is 6 bytes and contains the physical address of the destination station or stations to receive the packet. We will discuss addressing shortly.
- **Source address (SA).** The SA field is also 6 bytes and contains the physical address of the sender of the packet.
- **Length or type.** This field is defined as a type field or length field. The original Ethernet used this field as the type field to define the upper-layer protocol using the MAC frame. The IEEE standard used it as the length field to define the number of bytes in the data field. Both uses are common today.

- **Data.** This field carries data encapsulated from the upper-layer protocols. It is a minimum of 46 and a maximum of 1500 bytes, as we will see later.
- **CRC.** The last field contains error detection information, in this case a CRC-32 (See Appendix C).

Frame Length

Ethernet has imposed restrictions on both the minimum and maximum lengths of a frame, as shown in Figure 3.3.

Figure 3.3 *Minimum and maximum lengths*



The minimum length restriction is required for the correct operation of CSMA/CD, as we will see shortly. An Ethernet frame needs to have a minimum length of 512 bits or 64 bytes. Part of this length is the header and the trailer. If we count 18 bytes of header and trailer (6 bytes of source address, 6 bytes of destination address, 2 bytes of length or type, and 4 bytes of CRC), then the minimum length of data from the upper layer is 64 − 18 = 46 bytes. If the upper-layer packet is less than 46 bytes, padding is added to make up the difference.

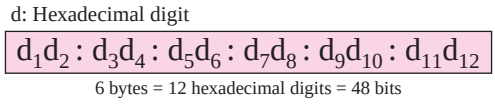
The standard defines the maximum length of a frame (without preamble and SFD field) as 1518 bytes. If we subtract the 18 bytes of header and trailer, the maximum length of the payload is 1500 bytes. The maximum length restriction has two historical reasons. First, memory was very expensive when Ethernet was designed: a maximum length restriction helped to reduce the size of the buffer. Second, the maximum length restriction prevents one station from monopolizing the shared medium, blocking other stations that have data to send.

Minimum length: 64 bytes (512 bits) **Maximum length: 1518 bytes (12,144 bits)**

Addressing

Each station on an Ethernet network (such as a PC, workstation, or printer) has its own **network interface card (NIC).** The NIC fits inside the station and provides the station with a 6-byte physical address. As shown in Figure 3.4, the Ethernet address is 6 bytes (48 bits), normally written in **hexadecimal notation,** with a colon between the bytes. The address normally is referred to as the data link address, physical address, or MAC address.

Figure 3.4 *Ethernet address in hexadecimal notation*



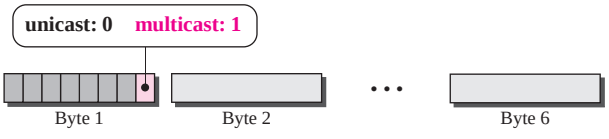
For example, the following shows an Ethernet MAC address:

4A:30:10:21:10:1A

Unicast, Multicast, and Broadcast Addresses

A source address is always a unicast address—the frame comes from only one station. The destination address, however, can be unicast, multicast, or broadcast. Figure 3.5 shows how to distinguish a unicast address from a multicast address. If the least significant bit of the first byte in a destination address is 0, the address is unicast; otherwise, it is multicast.

Figure 3.5 *Unicast and multicast addresses*



The least significant bit of the first byte defines the type of address. If the bit is 0, the address is unicast; otherwise, it is multicast.

A unicast destination address defines only one recipient; the relationship between the sender and the receiver is one-to-one. A multicast destination address defines a group of addresses; the relationship between the sender and the receivers is one-to-many.

The broadcast address is a special case of the multicast address; the recipients are all the stations on the LAN. A broadcast destination address is forty-eight 1s.

The broadcast destination address is a special case of the multicast address in which all bits are 1s.

Example 3.1

Define the type of the following destination addresses:

a. 4A:30:10:21:10:1A
b. 47:20:1B:2E:08:EE
c. FF:FF:FF:FF:FF:FF

Solution

To find the type of the address, we need to look at the second hexadecimal digit from the left. If it is even, the address is unicast. If it is odd, the address is multicast. If all digits are F's, the address is broadcast. Therefore, we have the following:

- **a.** This is a unicast address because A in binary is 1010 (even).
- **b.** This is a multicast address because 7 in binary is 0111 (odd).
- **c.** This is a broadcast address because all digits are F's.

The way the addresses are sent out on line is different from the way they are written in hexadecimal notation. The transmission is left-to-right, byte by byte; however, for each byte, the least significant bit is sent first and the most significant bit is sent last. This means that the bit that defines an address as unicast or multicast arrives first at the receiver.

Example 3.2

Show how the address 47:20:1B:2E:08:EE is sent out on line.

Solution

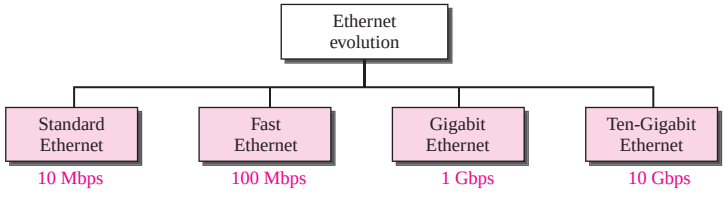
The address is sent left-to-right, byte by byte; for each byte, it is sent right-to-left, bit by bit, as shown below:

```
← 11100010 00000100 11011000 01110100 00010000 01110111
```

Ethernet Evolution

Ethernet was created in 1976 at Xerox's Palo Alto Research Center (PARC). Since then, it has gone through four generations: **Standard Ethernet** (10 Mbps), **Fast Ethernet** (100 Mbps), **Gigabit Ethernet** (1 Gbps), and **Ten-Gigabit Ethernet** (10 Gbps), as shown in Figure 3.6. We briefly discuss all these generations starting with the first, Standard (or traditional) Ethernet.

Figure 3.6 *Ethernet evolution through four generations*



Standard Ethernet

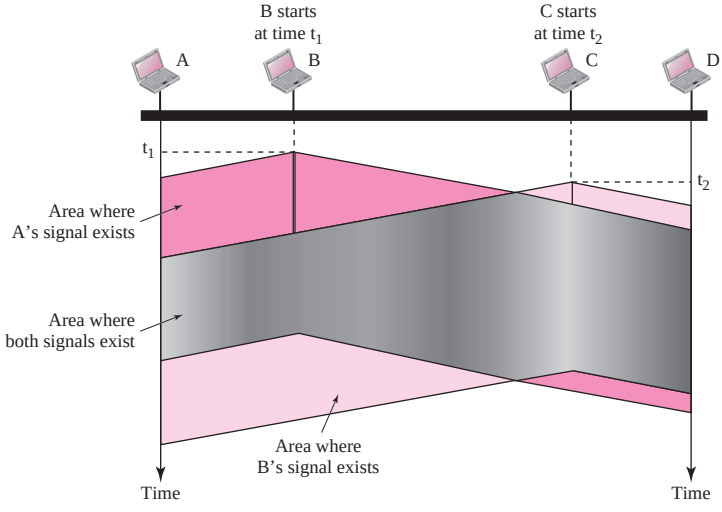
The original Ethernet with 10-Mbps data rate is now history, but we briefly discuss its characteristics to pave the way for understanding other Ethernet versions.

Access Method: CSMA/CD

The IEEE 802.3 standard defines **carrier sense multiple access with collision detection (CSMA/CD)** as the access method for traditional Ethernet. Stations on a traditional Ethernet can be connected together using a physical bus or star topology, but the logical topology is always a bus. By this, we mean that the medium (channel) is shared between stations and only one station at a time can use it. It also implies that all stations receive a frame sent by a station (broadcasting). The real destination keeps the frame while the rest drop it. In this situation, how can we be sure that two stations are not using the medium at the same time? If they do, their frames will collide with each other.

To minimize the chance of collision and, therefore, increase the performance, the CSMA method was developed. The chance of collision can be reduced if a station senses the medium before trying to use it. **Carrier sense multiple access (CSMA)** requires that each station first listen to the medium (or check the state of the medium) before sending. In other words, CSMA is based on the principle "sense before transmit" or "listen before talk." CSMA can reduce the possibility of collision, but it cannot eliminate it. The reason for this is shown in Figure 3.7, a space and time model of a CSMA network. Stations are connected to a shared channel (usually a dedicated medium).

Figure 3.7 *Space/time model of a collision in CSMA*

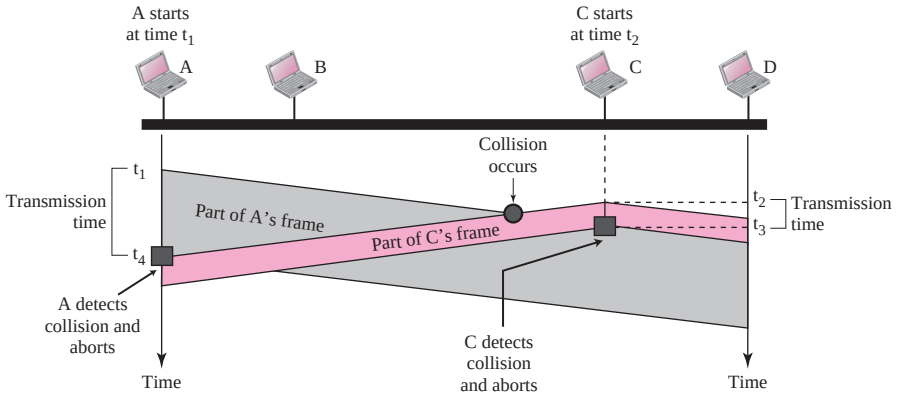


The possibility of collision still exists because of propagation delay; when a station sends a frame, it still takes time (although very short) for the first bit to reach every station and for every station to sense it. In other words, a station may sense the medium and find it idle, only because the first bit sent by another station has not yet been received.

At time t_1, station B senses the medium and finds it idle, so it sends a frame. At time t_2 ($t_2 > t_1$), station C senses the medium and finds it idle because, at this time, the first bits from station B have not reached station C. Station C also sends a frame. The two signals collide and both frames are destroyed.

Carrier sense multiple access with collision detection (CSMA/CD) augments the algorithm to handle the collision. In this method, a station monitors the medium after it sends a frame to see if the transmission was successful. If so, the station is finished. If, however, there is a collision, the frame is sent again. To better understand CSMA/CD, let us look at the first bits transmitted by the two stations involved in the collision. Although each station continues to send bits in the frame until it detects the collision, we show what happens as the first bits collide. In Figure 3.8, stations A and C are involved in the collision.

Figure 3.8 *Collision of the first bit in CSMA/CD*



At time t_1, station A has started sending the bits of its frame. At time t_2, station C has not yet sensed the first bit sent by A. Station C starts sending the bits in its frame, which propagate both to the left and to the right. The collision occurs sometime after time t_2. Station C detects a collision at time t_3 when it receives the first bit of A's frame. Station C immediately (or after a short time, but we assume immediately) aborts transmission. Station A detects collision at time t_4 when it receives the first bit of C's frame; it also immediately aborts transmission. Looking at the figure, we see that A transmits for the duration $t_4 - t_1$; C transmits for the duration $t_3 - t_2$. Later we show that, for the protocol to work, the length of any frame divided by the bit rate in this protocol must be more than either of these durations. At time t_4, the transmission of A's frame, though incomplete, is aborted; at time t_3, the transmission of B's frame, though incomplete, is aborted.

Minimum Frame Size

For CSMA/CD to work, we need a restriction on the frame size. Before sending the last bit of the frame, the sending station must detect a collision, if any, and abort the transmission. This is so because the station, once the entire frame is sent, does not keep a copy of the frame and does not monitor the line for collision detection. Therefore, the frame

transmission time T_{fr} must be at least two times the maximum propagation time T_p. To understand the reason, let us think about the worst-case scenario. If the two stations involved in a collision are the maximum distance apart, the signal from the first takes time T_p to reach the second, and the effect of the collision takes another time T_P to reach the first. So the requirement is that the first station must still be transmitting after $2T_p$.

Example 3.3

In the standard Ethernet, if the maximum propagation time is 25.6 μs, what is the minimum size of the frame?

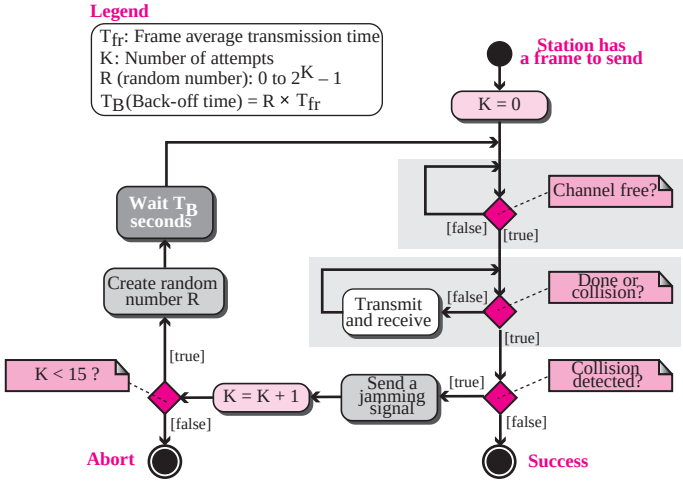
Solution

The frame transmission time is $T_{fr} = 2 \times T_p = 51.2$ μs. This means, in the worst case, a station needs to transmit for a period of 51.2 μs to detect the collision. The minimum size of the frame is 10 Mbps × 51.2 μs = 512 bits or 64 bytes. This is actually the minimum size of the frame for Standard Ethernet, as we discussed before.

Procedure

Figure 3.9 shows the flow diagram for CSMA/CD. We need to sense the channel before we start sending the frame. We do not send the entire frame and then look for a collision. The station transmits and receives continuously and simultaneously (using two different ports). We use a loop to show that transmission is a continuous process. We constantly monitor in order to detect one of two conditions: either transmission is finished or a collision is detected. Either event stops transmission. When we come out of the loop, if a collision has not been detected, it means that transmission is complete; the entire frame is transmitted. Otherwise, a collision has occurred. The diagram also shows a short **jamming signal** that enforces the collision in case other stations have not yet sensed the collision.

Figure 3.9 *CSMA/CD flow diagram*



Implementation

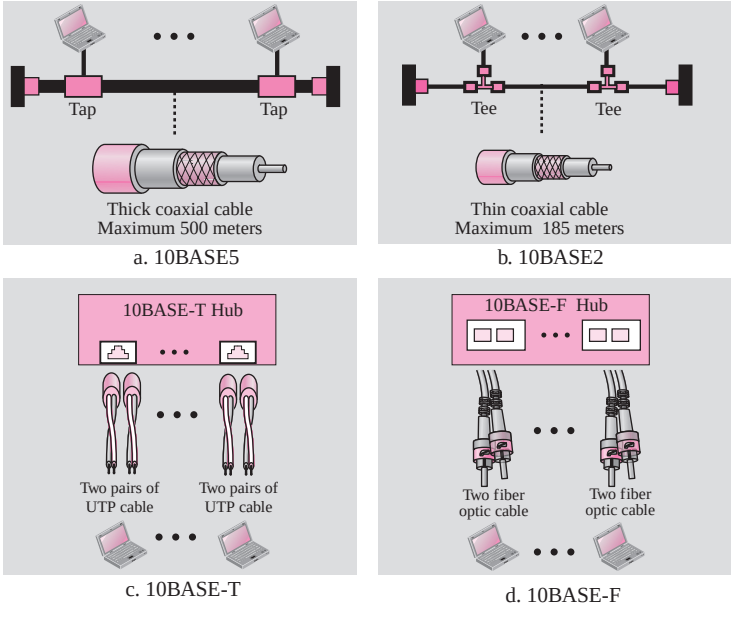
The Standard Ethernet defined several implementations, but only four of them became popular during ’80s. Table 3.1 shows a summary of Standard Ethernet implementations. In the nomenclature 10Base-X, the number defines the data rate (10 Mbps), the term Base means baseband (digital) signal, and X approximately defines either the maximum size of the cable in 100 meters (for example 5 for 500 or 2 for 185 meters) or the type of the cable, T for unshielded twisted pair cable (UTP) and F for fiber-optic.

Table 3.1 *Summary of Standard Ethernet implementations*

Characteristics	*10Base5*	*10Base2*	*10Base-T*	*10Base-F*
Medium	Thick coax	Thin coax	2 UTP	2 Fiber
Maximum length	500 m	185 m	100 m	2000 m

Figure 3.10 shows simplified diagrams of each implementation.

Figure 3.10 *Standard Ethernet Implementation*



Fast Ethernet

Fast Ethernet was designed to compete with LAN protocols such as FDDI or Fiber Channel. IEEE created Fast Ethernet under the name 802.3u. Fast Ethernet is backward-compatible with Standard Ethernet, but it can transmit data 10 times faster at a rate of 100 Mbps. The goals of Fast Ethernet can be summarized as follows:

1. Upgrade the data rate to 100 Mbps.
2. Make it compatible with Standard Ethernet.

3. Keep the same 48-bit address.
4. Keep the same frame format.
5. Keep the same minimum and maximum frame lengths.

MAC Sublayer

A main consideration in the evolution of Ethernet from 10 to 100 Mbps was to keep the MAC sublayer untouched. However, a decision was made to drop the bus topologies and keep only the star topology. For the star topology, there are two choices: half duplex and full duplex. In the half-duplex approach, the stations are connected via a hub; in the full-duplex approach, the connection is made via a switch with buffers at each port (see Section 3.5, Connecting Devices, at the end of the chapter).

The access method is the same (CSMA/CD) for the half-duplex approach; for full-duplex Fast Ethernet, there is no need for CSMA/CD. However, the implementations keep CSMA/CD for backward compatibility with Standard Ethernet.

Autonegotiation

A new feature added to Fast Ethernet is called **autonegotiation.** It allows a station or a hub a range of capabilities. Autonegotiation allows two devices to negotiate the mode or data rate of operation. It was designed particularly for the following purposes:

- To allow incompatible devices to connect to one another. For example, a device with a maximum capacity of 10 Mbps can communicate with a device with a 100 Mbps capacity (but can work at a lower rate).
- To allow one device to have multiple capabilities.
- To allow a station to check a hub's capabilities.

Implementation

Fast Ethernet implementation at the physical layer can be categorized as either two-wire or four-wire. The two-wire implementation can be either shielded twisted pair, STP (**100Base-TX**) or fiber-optic cable (**100Base-FX**). The four-wire implementation is designed only for unshielded twist pair, UTP (**100Base-T4**). Table 3.2 is a summary of the Fast Ethernet implementations.

Table 3.2 *Summary of Fast Ethernet implementations*

Characteristics	*100Base-TX*	*100Base-FX*	*100Base-T4*
Media	STP	Fiber	UTP
Number of wires	2	2	4
Maximum length	100 m	100 m	100 m

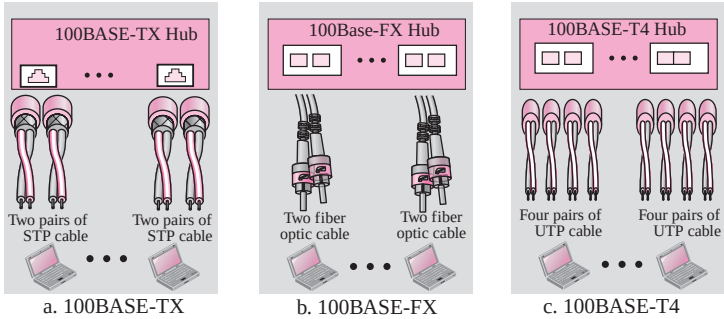
Figure 3.11 shows simplified diagrams of each implementation.

Gigabit Ethernet

The need for an even higher data rate resulted in the design of the Gigabit Ethernet Protocol (1000 Mbps). The IEEE committee calls the Standard 802.3z. The goals of the Gigabit Ethernet design can be summarized as follows:

1. Upgrade the data rate to 1 Gbps.
2. Make it compatible with Standard or Fast Ethernet.

Figure 3.11 *Fast Ethernet Implementation*



a. 100BASE-TX b. 100BASE-FX c. 100BASE-T4

3. Use the same 48-bit address.
4. Use the same frame format.
5. Keep the same minimum and maximum frame lengths.
6. To support autonegotiation as defined in Fast Ethernet.

MAC Sublayer

A main consideration in the evolution of Ethernet was to keep the MAC sublayer untouched. However, to achieve a data rate of 1 Gbps, this was no longer possible. Gigabit Ethernet has two distinctive approaches for medium access: half-duplex and full-duplex. Almost all implementations of Gigabit Ethernet follow the full-duplex approach. However, we briefly discuss the half-duplex approach to show that Gigabit Ethernet can be compatible with the previous generations.

Full-Duplex Mode In full-duplex mode, there is a central switch connected to all computers or other switches. In this mode, each switch has buffers for each input port in which data are stored until they are transmitted. There is no collision in this mode. This means that CSMA/CD is not used. Lack of collision implies that the maximum length of the cable is determined by the signal attenuation in the cable, not by the collision detection process.

In the full-duplex mode of Gigabit Ethernet, there is no collision; the maximum length of the cable is determined by the signal attenuation in the cable.

Half-Duplex Mode Gigabit Ethernet can also be used in half-duplex mode, although it is rare. In this case, a switch can be replaced by a hub, which acts as the common cable in which a collision might occur. The half-duplex approach uses CSMA/CD. However, as we saw before, the maximum length of the network in this approach is totally dependent on the minimum frame size. Three solutions have been defined: traditional, carrier extension, and frame bursting.

- **Traditional.** In the traditional approach, we keep the minimum length of the frame as in traditional Ethernet (512 bits). However, because the length of a bit is 1/100 shorter in Gigabit Ethernet than in 10-Mbps Ethernet, the maximum length of the network is 25 m. This length may be suitable if all the stations are in one room, but it may not even be long enough to connect the computers in one single office.
- **Carrier Extension.** To allow for a longer network, we increase the minimum frame length. The **carrier extension** approach defines the minimum length of a frame as 512 bytes (4096 bits). This means that the minimum length is 8 times longer. This method forces a station to add extension bits (padding) to any frame that is less than 4096 bits. In this way, the maximum length of the network can be increased 8 times to a length of 200 m. This allows a length of 100 m from the hub to the station.
- **Frame Bursting**. Carrier extension is very inefficient if we have a series of short frames to send; each frame carries redundant data. To improve efficiency, **frame bursting** was proposed. Instead of adding an extension to each frame, multiple frames are sent. However, to make these multiple frames look like one frame, padding is added between the frames (the same as that used for the carrier extension method) so that the channel is not idle. In other words, the method deceives other stations into thinking that a very large frame has been transmitted.

Implementation

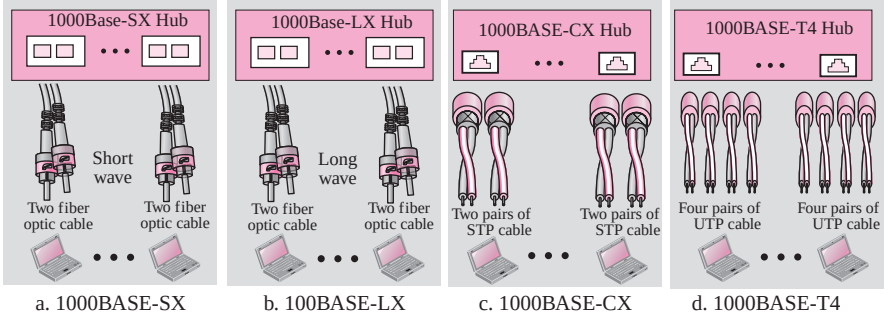
Table 3.3 is a summary of the Gigabit Ethernet implementations.

Table 3.3 *Summary of Gigabit Ethernet implementations*

Characteristics	*1000Base-SX*	*1000Base-LX*	*1000Base-CX*	*1000Base-T4*
Media	Fiber short-wave	Fiber long-wave	STP	Cat 5 UTP
Number of wires	2	2	2	4
Maximum length	550 m	5000 m	25 m	100 m

Figure 3.12 shows the simplified diagrams for Gigabit Ethernet.

Figure 3.12 *Gigabit Ethernet implementation*



a. 1000BASE-SX b. 100BASE-LX c. 1000BASE-CX d. 1000BASE-T4

Ten-Gigabit Ethernet

The IEEE committee created Ten-Gigabit Ethernet and called it Standard 802.3ae. The goals of the Ten-Gigabit Ethernet design can be summarized as follows:

1. Upgrade the data rate to 10 Gbps.
2. Make it compatible with Standard, Fast, and Gigabit Ethernet.
3. Use the same 48-bit address.
4. Use the same frame format.
5. Keep the same minimum and maximum frame lengths.
6. Allow the interconnection of existing LANs into a metropolitan area network (MAN) or a wide area network (WAN).
7. Make Ethernet compatible with technologies such as Frame Relay and ATM.

Implementation

Ten-Gigabit Ethernet operates only in full duplex mode, which means there is no need for contention; CSMA/CD is not used in Ten-Gigabit Ethernet. Three implementations are the most common: 10GBase-S, 10GBase-L, and 10GBase-E. Table 3.4 shows a summary of the Ten-Gigabit Ethernet implementation.

Table 3.4 *Ten-Gigabit Ethernet Implementation*

Characteristics	*10GBase-S*	*10GBase-L*	*10GBase-E*
Media	multi-mode fiber	single-mode fiber	single-mode fiber
Number of wires	2	2	2
Maximum length	300 m	10,000 m	40,000 m

3.2 WIRELESS LANS

Wireless communication is one of the fastest-growing technologies. The demand for connecting devices without the use of cables is increasing everywhere. **Wireless LANs** can be found on college campuses, in office buildings, and in many public areas. In this section, we concentrate on two wireless technologies for LANs: IEEE 802.11 wireless LANs, sometimes called wireless Ethernet, and Bluetooth, a technology for small wireless LANs.

IEEE 802.11

IEEE has defined the specifications for a wireless LAN, called **IEEE 802.11,** which covers the physical and data link layers.

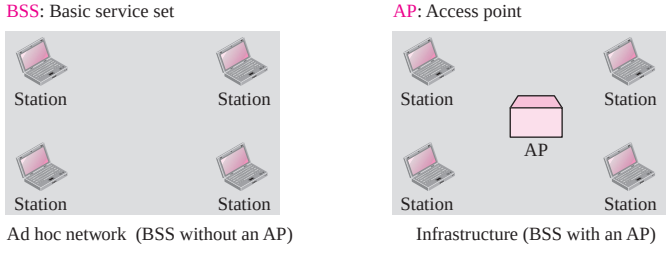
Architecture

The standard defines two kinds of services: the basic service set (BSS) and the extended service set (ESS).

Basic Service Set IEEE 802.11 defines the **basic service set (BSS)** as the building block of a wireless LAN. A basic service set is made of stationary or mobile wireless stations and an optional central base station, known as the **access point (AP).** Figure 3.13 shows two sets in this standard. The BSS without an AP is a stand-alone network and

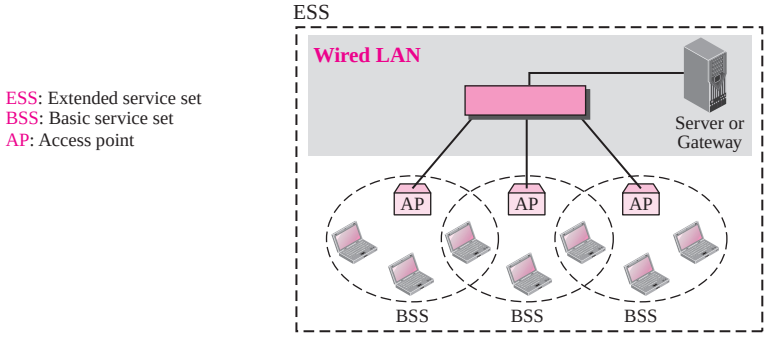
cannot send data to other BSSs. It is called an *ad hoc architecture*. In this architecture, stations can form a network without the need of an AP; they can locate one another and agree to be part of a BSS. A BSS with an AP is sometimes referred to as an *infrastructure* network.

Figure 3.13 *Basic service sets (BSSs)*



Extended Service Set An **extended service set (ESS)** is made up of two or more BSSs with APs. In this case, the BSSs are connected through a *distribution system*, which is usually a wired LAN. The distribution system connects the APs in the BSSs. IEEE 802.11 does not restrict the distribution system; it can be any IEEE LAN such as an Ethernet. Note that the extended service set uses two types of stations: mobile and-stationary. The mobile stations are normal stations inside a BSS. The stationary stations are AP stations that are part of a wired LAN. Figure 3.14 shows an ESS.

Figure 3.14 *Extended service sets (ESSs)*



When BSSs are connected, the stations within reach of one another can communicate without the use of an AP. However, communication between two stations in two different BSSs usually occurs via two APs.

Station Types

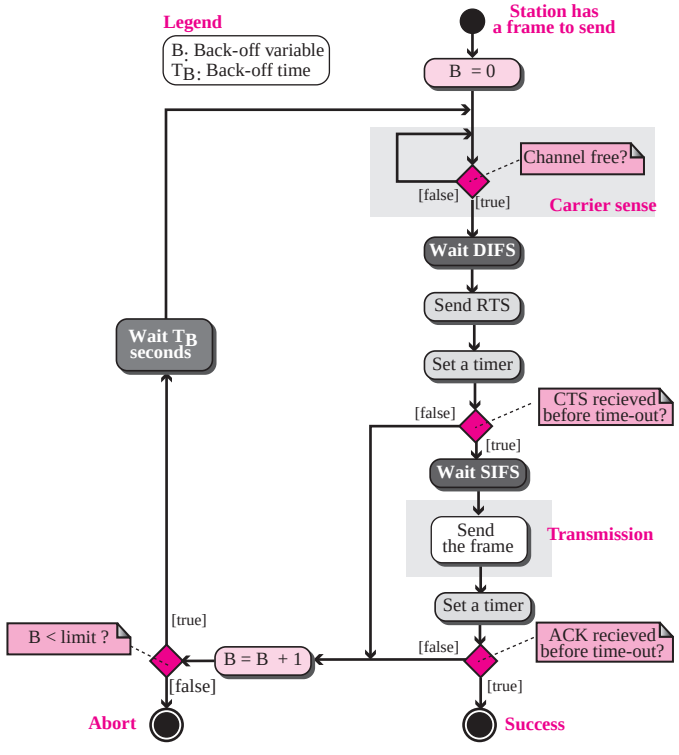
IEEE 802.11 defines three types of stations based on their mobility in a wireless LAN: **no-transition, BSS-transition,** and **ESS-transition mobility.** A station with no-transition

mobility is either stationary (not moving) or moving only inside a BSS. A station with BSS-transition mobility can move from one BSS to another, but the movement is confined inside one ESS. A station with ESS-transition mobility can move from one ESS to another.

MAC Sublayer

There are two different MAC sublayers in this protocol, however; the one that is used most of the time is based on CSMA/CA (**carrier sense multiple access with collision avoidance**). Figure 3.15 shows the flow diagram.

Figure 3.15 *CSMA/CA flow diagram*



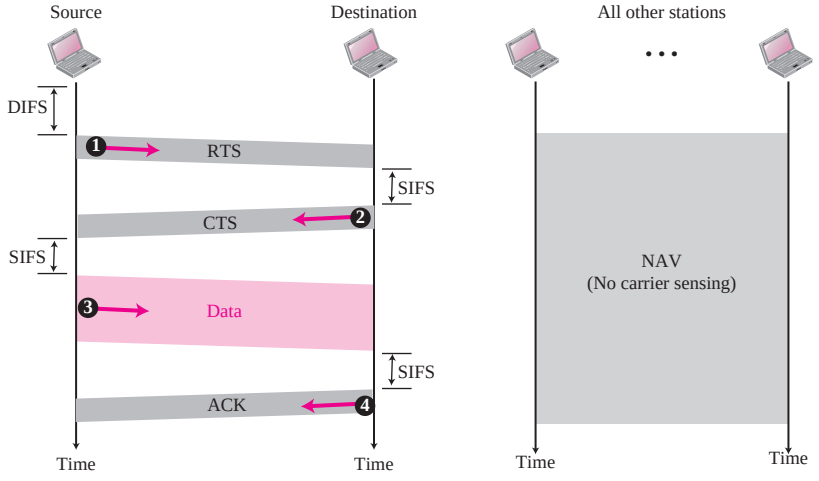
Wireless LANs cannot implement CSMA/CD for three reasons:

1. For collision detection a station must be able to send data and receive collision signals at the same time. This can mean costly stations and increased bandwidth requirements.
2. Collision may not be detected because of the hidden station problem. We will discuss this problem later in the chapter.
3. The distance between stations can be great. Signal fading could prevent a station at one end from hearing a collision at the other end.

Frame Exchange Time Line

Figure 3.16 shows the exchange of data and control frames in time.

Figure 3.16 *CSMA/CA and NAV*



1. Before sending a frame, the source station senses the medium by checking the energy level at the carrier frequency.
   a. The channel uses a persistence strategy with back-off until the channel is idle.
   b. After the station is found to be idle, the station waits for a period of time called the **distributed interframe space (DIFS);** then the station sends a control frame called the request to send (RTS).
2. After receiving the RTS and waiting a period of time called the **short interframe space (SIFS),** the destination station sends a control frame, called the clear to send (CTS), to the source station. This control frame indicates that the destination station is ready to receive data.
3. The source station sends data after waiting an amount of time equal to SIFS.
4. The destination station, after waiting an amount of time equal to SIFS, sends an acknowledgment to show that the frame has been received. Acknowledgment is needed in this protocol because the station does not have any means to check for the successful arrival of its data at the destination. On the other hand, the lack of collision in CSMA/CD is a kind of indication to the source that data have arrived.

Network Allocation Vector How do other stations defer sending their data if one station acquires access? In other words, how is the *collision avoidance* aspect of this protocol accomplished? The key is a feature called NAV.

When a station sends an RTS frame, it includes the duration of time that it needs to occupy the channel. The stations that are affected by this transmission create a timer called a **network allocation vector (NAV)** that shows how much time must pass before these stations are allowed to check the channel for idleness. Each time a station

accesses the system and sends an RTS frame, other stations start their NAV. In other words, each station, before sensing the physical medium to see if it is idle, first checks its NAV to see if it has expired. Figure 3.16 also shows the idea of NAV.

What happens if there is collision during the time when RTS or CTS control frames are in transition, often called the **handshaking period**? Two or more stations may try to send RTS frames at the same time. These control frames may collide. However, because there is no mechanism for collision detection, the sender assumes there has been a collision if it has not received a CTS frame from the receiver. The back-off strategy is employed, and the sender tries again.

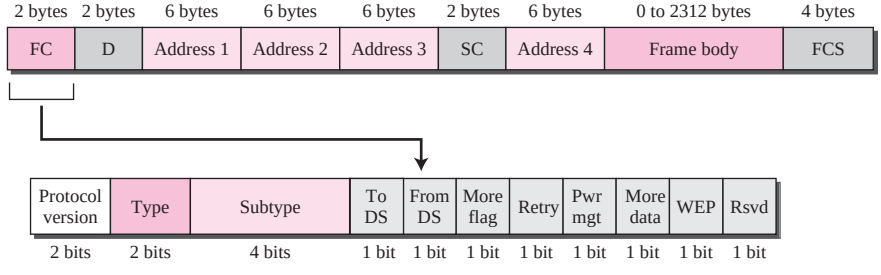
Fragmentation

The wireless environment is very noisy; a corrupt frame has to be retransmitted. The protocol, therefore, recommends fragmentation—the division of a large frame into smaller ones. It is more efficient to resend a small frame than a large one.

Frame Format

The MAC layer frame consists of nine fields, as shown in Figure 3.17.

Figure 3.17 *Frame format*



- **Frame control (FC).** The FC field is 2 bytes long and defines the type of frame and some control information. Table 3.5 describes the subfields. We will discuss each frame type later in this chapter.

Table 3.5 *Subfields in FC field*

Field	*Explanation*
Version	Current version is 0
Type	Type of information: management (00), control (01), or data (10)
Subtype	Subtype of each type (see Table 3.6)
To DS	Defined later
From DS	Defined later
More flag	When set to 1, means more fragments
Retry	When set to 1, means retransmitted frame
Pwr mgt	When set to 1, means station is in power management mode
More data	When set to 1, means station has more data to send
WEP	Wired equivalent privacy (encryption implemented)
Rsvd	Reserved

- **D.** In all frame types except one, this field defines the duration of the transmission that is used to set the value of NAV. In one control frame, this field defines the ID of the frame.
- **Addresses.** There are four address fields, each 6 bytes long. The meaning of each address field depends on the value of the *To DS* and *From DS* subfields and will be discussed later.
- **Sequence control.** This field defines the sequence number of the frame to be used in flow control.
- **Frame body.** This field, which can be between 0 and 2312 bytes, contains information based on the type and the subtype defined in the FC field.
- **FCS.** The FCS field is 4 bytes long and contains a CRC-32 error detection sequence.

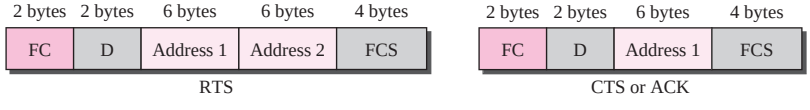
Frame Types

A wireless LAN defined by IEEE 802.11 has three categories of frames: management frames, control frames, and data frames.

Management Frames Management frames are used for the initial communication between stations and access points.

Control Frames Control frames are used for accessing the channel and acknowledging frames. Figure 3.18 shows the format.

Figure 3.18 *Control frames*



For control frames the value of the type field is 01; the values of the subtype fields for frames we have discussed are shown in Table 3.6.

Table 3.6 *Values of subfields in control frames*

Subtype	*Meaning*
1011	Request to send (RTS)
1100	Clear to send (CTS)
1101	Acknowledgment (ACK)

Data Frames Data frames are used for carrying data and control information.

Addressing Mechanism

The IEEE 802.11 addressing mechanism specifies four cases, defined by the value of the two flags in the FC field, *To DS* and *From DS*. Each flag can be either 0 or 1, resulting in four different situations. The interpretation of the four addresses (address 1 to address 4) in the MAC frame depends on the value of these flags, as shown in Table 3.7.

Table 3.7 *Addresses*

To DS	*From DS*	*Address 1*	*Address 2*	*Address 3*	*Address 4*
0	0	Destination	Source	BSS ID	N/A
0	1	Destination	Sending AP	Source	N/A
1	0	Receiving AP	Source	Destination	N/A
1	1	Receiving AP	Sending AP	Destination	Source

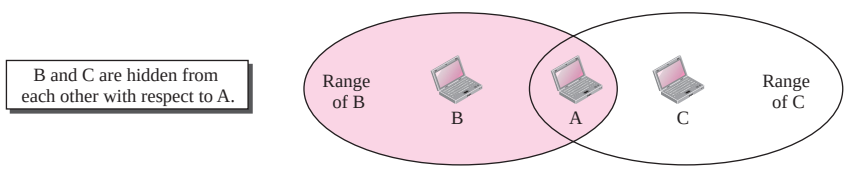
Note that address 1 is always the address of the next device. Address 2 is always the address of the previous device. Address 3 is the address of the final destination station if it is not defined by address 1. Address 4 is the address of the original source station if it is not the same as address 2.

Hidden and Exposed Station Problems

We referred to hidden and exposed station problems in the previous section. It is time now to discuss these problems and their effects.

Hidden Station Problem Figure 3.19 shows an example of the hidden station problem. Station B has a transmission range shown by the left oval (sphere in space); every station in this range can hear any signal transmitted by station B. Station C has a transmission range shown by the right oval (sphere in space); every station located in this range can hear any signal transmitted by C. Station C is outside the transmission range of B; likewise, station B is outside the transmission range of C. Station A, however, is in the area covered by both B and C; it can hear any signal transmitted by B or C.

Figure 3.19 *Hidden station problem*



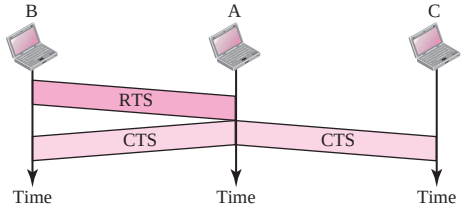
Assume that station B is sending data to station A. In the middle of this transmission, station C also has data to send to station A. However, station C is out of B's range and transmissions from B cannot reach C. Therefore C thinks the medium is free. Station C sends its data to A, which results in a collision at A because this station is receiving data from both B and C. In this case, we say that stations B and C are hidden from each other with respect to A. Hidden stations can reduce the capacity of the network because of the possibility of collision.

The solution to the hidden station problem is the use of the handshake frames (RTS and CTS) that we discussed earlier. Figure 3.20 shows that the RTS message from B reaches A, but not C. However, because both B and C are within the range of A, the CTS message, which contains the duration of data transmission from B to A reaches C.

Station C knows that some hidden station is using the channel and refrains from transmitting until that duration is over.

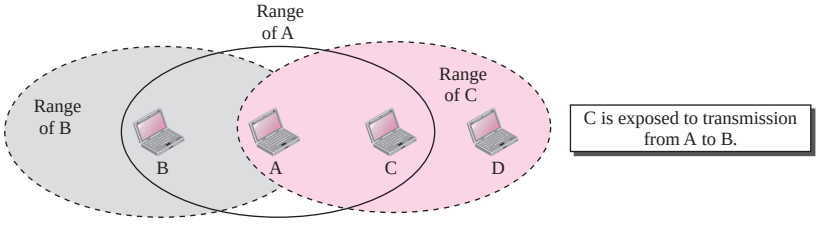
The CTS frame in CSMA/CA handshake can prevent collision from a hidden station.

Figure 3.20 *Use of handshaking to prevent hidden station problem*



Exposed Station Problem Now consider a situation that is the inverse of the previous one: the exposed station problem. In this problem a station refrains from using a channel when it is, in fact, available. In Figure 3.21, station A is transmitting to station B. Station C has some data to send to station D, which can be sent without interfering with the transmission from A to B. However, station C is exposed to transmission from A; it hears what A is sending and thus refrains from sending. In other words, C is too conservative and wastes the capacity of the channel.

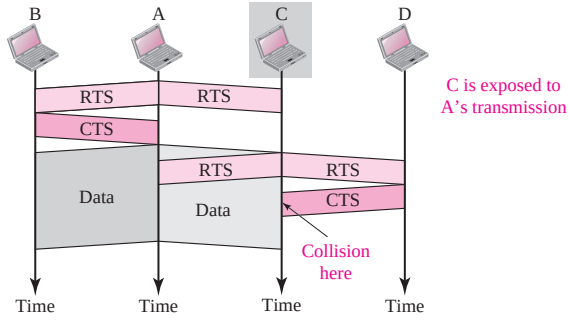
Figure 3.21 *Exposed station problem*



The handshaking messages RTS and CTS cannot help in this case, despite what we might think. Figure 3.22 shows the situation.

Station C hears the RTS from A, but does not hear the CTS from B. Station C, after hearing the RTS from A, can wait for a time so that the CTS from B reaches A; it then sends an RTS to D to show that it needs to communicate with D. Both stations B and A may hear this RTS, but station A is in the sending state, not the receiving state. Station B, however, responds with a CTS. The problem is here. If station A has started sending its data, station C cannot hear the CTS from station D because of the collision; it cannot send its data to D. It remains exposed until A finishes sending its data.

Figure 3.22 *Use of handshaking in exposed station problem*



Bluetooth

Bluetooth is a wireless LAN technology designed to connect devices of different functions such as telephones, notebooks, computers (desktop and laptop), cameras, printers, coffee makers, and so on. A Bluetooth LAN is an ad hoc network, which means that the network is formed spontaneously; the devices, sometimes called gadgets, find each other and make a network called a piconet. A Bluetooth LAN can even be connected to the Internet if one of the gadgets has this capability. A Bluetooth LAN, by nature, cannot be large. If there are many gadgets that try to connect, there is chaos.

Bluetooth technology has several applications. Peripheral devices such as a wireless mouse or keyboard can communicate with the computer through this technology. Monitoring devices can communicate with sensor devices in a small health care center. Home security devices can use this technology to connect different sensors to the main security controller. Conference attendees can synchronize their laptop computers at a conference.

Bluetooth was originally started as a project by the Ericsson Company. It is named for Harald Blaatand, the king of Denmark (940–981) who united Denmark and Norway. *Blaatand* translates to *Bluetooth* in English.

Today, Bluetooth technology is the implementation of a protocol defined by the IEEE 802.15 standard. The standard defines a wireless personal area network (PAN) operable in an area the size of a room or a hall.

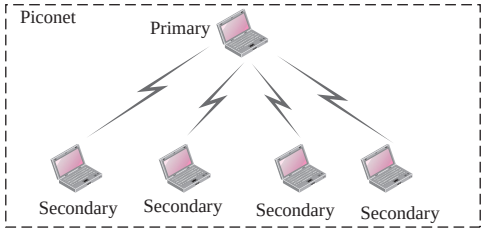
Architecture

Bluetooth defines two types of networks: piconet and scatternet.

Piconets A Bluetooth network is called a **piconet,** or a small net. A piconet can have up to eight stations, one of which is called the **primary;** the rest are called **secondaries.** All the secondary stations synchronize their clocks and hopping sequence with the primary. Note that a piconet can have only one primary station. The communication between the primary and the secondary can be one-to-one or one-to-many. Figure 3.23 shows a piconet.

Although a piconet can have a maximum of seven secondaries, an additional eight secondaries can be in the *parked state*. A secondary in a parked state is synchronized

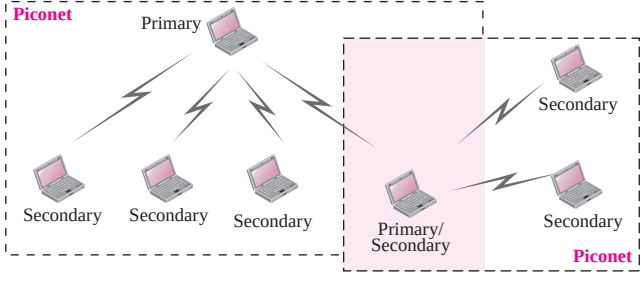
Figure 3.23 *Piconet*



with the primary, but cannot take part in communication until it is moved from the parked state. Because only eight stations can be active in a piconet, activating a station from the parked state means that an active station must go to the parked state.

Scatternet Piconets can be combined to form what is called a **scatternet.** A secondary station in one piconet can be the primary in another piconet. This station can receive messages from the primary in the first piconet (as a secondary) and, acting as a primary, deliver them to secondaries in the second piconet. A station can be a member of two piconets. Figure 3.24 illustrates a scatternet.

Figure 3.24 *Scatternet*



Bluetooth Devices

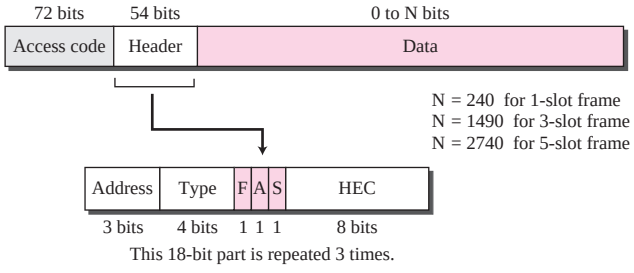
A Bluetooth device has a built-in short-range radio transmitter. The current data rate is 1 Mbps with a 2.4-GHz bandwidth. This means that there is a possibility of interference between the IEEE 802.11b wireless LANs and Bluetooth LANs.

Frame Format

A frame in the baseband layer can be one of three types: one-slot, three-slot, or five-slot. A slot, as we said before, is 625 μs. However, in a one-slot frame exchange, 259 μs is needed for hopping and control mechanisms. This means that a one-slot frame can last only 625 – 259, or 366 μs. With a 1-MHz bandwidth and 1 bit/Hz, the size of a one-slot frame is 366 bits.

A three-slot frame occupies three slots. However, since 259 μs is used for hopping, the length of the frame is $3 \times 625 - 259 = 1616$ μs or 1616 bits. A device that uses a three-slot frame remains at the same hop (at the same carrier frequency) for three slots. Even though only one hop number is used, three hop numbers are consumed. That means the hop number for each frame is equal to the first slot of the frame. A five-slot frame also uses 259 bits for hopping, which means that the length of the frame is $5 \times 625 - 259 = 2866$ bits. Figure 3.25 shows the format of the three frame types.

Figure 3.25 *Frame format types*



The following describes each field:

- **Access code.** This 72-bit field normally contains synchronization bits and the identifier of the primary to distinguish the frame of one piconet from another.
- **Header.** This 54-bit field is a repeated 18-bit pattern. Each pattern has the following subfields:
  - **a. Address.** The 3-bit address subfield can define up to seven secondaries (1 to 7). If the address is zero, it is used for broadcast communication from the primary to all secondaries.
  - **b. Type.** The 4-bit type subfield defines the type of data coming from the upper layers. We discuss these types later.
  - **c. F.** This 1-bit subfield is for flow control. When set (1), it indicates that the device is unable to receive more frames (buffer is full).
  - **d. A.** This 1-bit subfield is for acknowledgment. Bluetooth uses stop-and-wait ARQ; 1 bit is sufficient for acknowledgment.
  - **e. S.** This 1-bit subfield holds a sequence number. Bluetooth uses stop-and-wait ARQ; 1 bit is sufficient for sequence numbering.
  - **f. HEC.** The 8-bit header error correction subfield is a checksum to detect errors in each 18-bit header section.

  The header has three identical 18-bit sections. The receiver compares these three sections, bit by bit. If each of the corresponding bits is the same, the bit is accepted; if not, the majority opinion rules. This is a form of forward error correction (for the header only). This double error control is needed because the nature of the communication, via air, is very noisy. Note that there is no retransmission in this sublayer.

- **Data.** This subfield can be 0 to 2740 bits long. It contains data or control information coming from the upper layers.

a. The sending station, after sensing that the medium is idle, sends a special small frame called request to send (RTS). In this message, the sender defines the total time it needs the medium.

b. The receiver acknowledges the request (broadcast to all stations) by sending a small packet called clear to send (CTS).

c. The sender sends the data frame.

d. The receiver acknowledges the receipt of data.

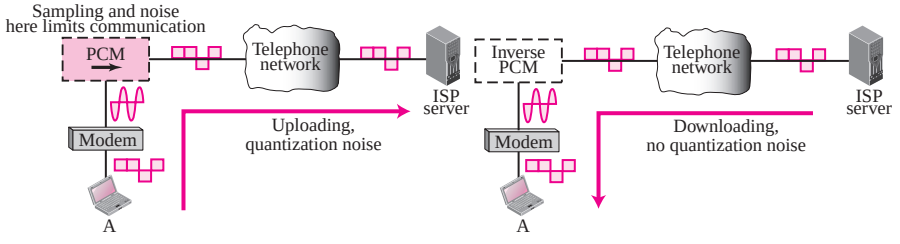
3.3 POINT-TO-POINT WANS

A second type of network we encounter in the Internet is the point-to-point wide area network. A point-to-point WAN connects two remote devices using a line available from a public network such as a telephone network. We discuss traditional modem technology, DSL line, cable modem, T-lines, and SONET.

56K Modems

People still use traditional modems to upload data to the Internet and download data from the Internet, as shown in Figure 3.26.

Figure 3.26 *56K modem*



In **uploading,** the analog signal must be sampled at the switching station, which means the data rate in uploading is limited to 33.6 kbps. However, there is no sampling in **downloading.** The signal is not affected by quantization noise and not subject to the Shannon capacity limitation. The maximum data rate in the uploading direction is 33.6 kbps, but the data rate in the downloading direction is 56 kbps.

One may wonder why 56 kbps. The telephone companies sample voice 8000 times per second with 8 bits per sample. One of the bits in each sample is used for control purposes, which means each sample is 7 bits. The rate is therefore 8000×7, or 56,000 bps or 56 kbps.

The **V.90** and **V.92** standard modems operate at 56 kbps to connect a host to the Internet.

DSL Technology

After traditional modems reached their peak data rate, telephone companies developed another technology, DSL, to provide higher-speed access to the Internet. **Digital subscriber line (DSL)** technology is one of the most promising for supporting high-speed digital communication over the existing local loops (telephone line). DSL technology is a set of technologies, each differing in the first letter (ADSL, VDSL, HDSL, and SDSL). The set is often referred to as *x*DSL, where *x* can be replaced by A, V, H, or S.

ADSL

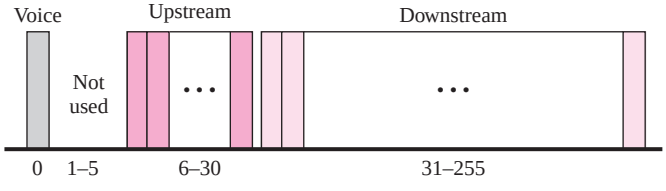
The first technology in the set is **asymmetric DSL (ADSL).** ADSL, like a 56K modem, provides higher speed (bit rate) in the downstream direction (from the Internet to the resident) than in the upstream direction (from the resident to the Internet). That is the reason it is called asymmetric. Unlike the asymmetry in 56K modems, the designers of ADSL specifically divided the available bandwidth of the local loop unevenly for the residential customer. The service is not suitable for business customers who need a large bandwidth in both directions.

ADSL is an asymmetric communication technology designed for residential users; it is not suitable for businesses.

Figure 3.27 shows how the bandwidth is divided:

- **Voice.** Channel 0 is reserved for voice communication.
- **Idle.** Channels 1 to 5 are not used, to allow a gap between voice and data communication.
- **Upstream data and control.** Channels 6 to 30 (25 channels) are used for upstream data transfer and control. One channel is for control, and 24 channels are for data transfer. If there are 24 channels, each using 4 kHz (out of 4.312 kHz available) with 15 bits per Hz, we have $24 \times 4000 \times 15$, or 1.44-Mbps bandwidth, in the upstream direction.
- **Downstream data and control.** Channels 31 to 255 (225 channels) are used for downstream data transfer and control. One channel is for control, and 224 channels are for data. If there are 224 channels, we can achieve up to $224 \times 4000 \times 15$, or 13.4 Mbps.

Figure 3.27 *Bandwidth division*

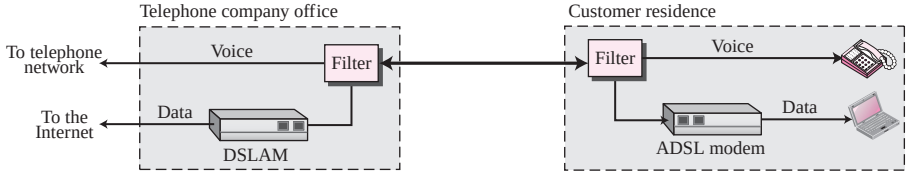


Because of the high signal/noise ratio, the actual bit rate is much lower than the above-mentioned rates. The bit rates are as follows:

Upstream: 64 kbps to 1 Mbps	Downstream: 500 kbps to 8 Mbps

Figure 3.28 shows an ADSL modem installed at a customer's site. The local loop connects to the filter which separates voice and data communication. The ADSL modem modulates the data and creates downstream and upstream channels.

Figure 3.28 *ADSL and DSLAM*



At the telephone company site, the situation is different. Instead of an ADSL modem, a device called a **digital subscriber line access multiplexer (DSLAM)** is installed that functions similarly to an ADSL modem. In addition, it packetizes the data to be sent to the Internet. Figure 3.28 shows the configuration.

Other DSL Technologies

ADSL provides asymmetric communication. The downstream bit rate is much higher than the upstream bit rate. Although this feature meets the needs of most residential subscribers, it is not suitable for businesses that send and receive data in large volumes in both directions. The **symmetric digital subscriber line (SDSL)** is designed for these types of businesses. It divides the available bandwidth equally between the downstream and upstream directions.

The **high bit rate digital subscriber line (HDSL)** was designed as an alternative to the T-1 line (1.544 Mbps). The T-1 line (discussed later) uses alternate mark inversion (AMI) encoding, which is very susceptible to attenuation at high frequencies. This limits the length of a T-1 line to 1 km. For longer distances, a repeater is necessary, which means increased costs.

The **very high bit rate digital subscriber line (VDSL),** an alternative approach that is similar to ADSL, uses coaxial, fiber-optic, or twisted-pair cable for short distances (300 to 1800 m). The modulating technique is discrete multitone technique (DMT) with a bit rate of 50 to 55 Mbps downstream and 1.5 to 2.5 Mbps upstream.

Cable Modem

Cable companies are now competing with telephone companies for the residential customer who wants high-speed access to the Internet. DSL technology provides high-data-rate connections for residential subscribers over the local loop. However, DSL uses the existing unshielded twisted-pair cable, which is very susceptible to

interference. This imposes an upper limit on the data rate. Another solution is the use of the cable TV network.

Traditional Cable Networks

Cable TV started to distribute broadcast video signals to locations with poor or no reception. It was called **community antenna TV (CATV)** because an antenna at the top of a high hill or building received the signals from the TV stations and distributed them, via coaxial cables, to the community.

The cable TV office, called the **head end,** receives video signals from broadcasting stations and feeds the signals into coaxial cables. The traditional cable TV system used coaxial cable end to end. Because of attenuation of the signals and the use of a large number of amplifiers, communication in the traditional network was unidirectional (one-way). Video signals were transmitted downstream, from the head end to the subscriber premises.

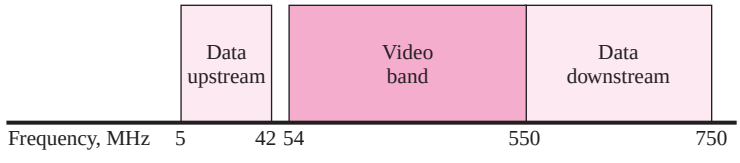
HFC Network

The second generation of cable networks is called a **hybrid fiber-coaxial (HFC) network.** The network uses a combination of fiber-optic and coaxial cable. The transmission medium from the cable TV office to a box, called the **fiber node,** is optical fiber; from the fiber node through the neighborhood and into the house, the medium is still coaxial cable. One reason for moving from traditional to hybrid infrastructure is to make the cable network bidirectional (two-way).

Bandwidth

Even in an HFC system, the last part of the network, from the fiber node to the subscriber premises, is still a coaxial cable. This coaxial cable has a bandwidth that ranges from 5 to 750 MHz (approximately). The cable company has divided this bandwidth into three bands: video, downstream data, and upstream data, as shown in Figure 3.29.

Figure 3.29 *Cable bandwidth*



- **Video Band.** The downstream-only **video band** occupies frequencies from 54 to 550 MHz. Since each TV channel occupies 6 MHz, this can accommodate more than 80 channels.
- **Downstream Data Band.** The downstream data (from the Internet to the subscriber premises) occupies the upper band, from 550 to 750 MHz. This band is also divided into 6-MHz channels. The downstream data can be received at 30 Mbps. The standard specifies only 27 Mbps. However, since the cable modem is connected to the computer through a 10BASE-T cable, this limits the data rate to 10 Mbps.

- **Upstream Data Band.** The upstream data (from the subscriber premises to the Internet) occupies the lower band, from 5 to 42 MHz. This band is also divided into 6-MHz channels. The **upstream data band** uses lower frequencies that are more susceptible to noise and interference. Theoretically, downstream data can be sent at 12 Mbps (2 bits/Hz × 6 MHz). However, the data rate is usually less than 12 Mbps.

Sharing

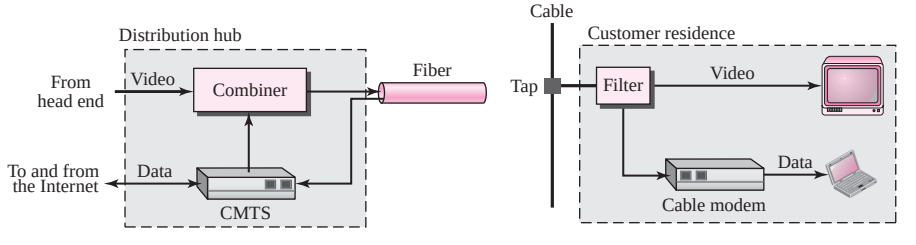
Both upstream and downstream bands are shared by the subscribers. The upstream data bandwidth is only 37 MHz. This means that there are only six 6-MHz channels available in the upstream direction. A subscriber needs to use one channel to send data in the upstream direction. The question is, How can six channels be shared in an area with 1000, 2000, or even 100,000 subscribers? The solution is time-sharing. The band is divided into channels; these channels must be shared between subscribers in the same neighborhood. The cable provider allocates one channel, statically or dynamically, for a group of subscribers. If one subscriber wants to send data, she or he contends for the channel with others who want access; the subscriber must wait until the channel is available. The situation is similar to CSMA discussed for Ethernet LANs.

We have a similar situation in the downstream direction. The downstream band has 33 channels of 6 MHz. A cable provider probably has more than 33 subscribers; therefore, each channel must be shared between a group of subscribers. However, the situation is different for the downstream direction; here we have a multicasting situation. If there are data for any of the subscribers in the group, the data are sent to that channel. Each subscriber is sent the data. But since each subscriber also has an address registered with the provider, the cable modem for the group matches the address carried with the data to the address assigned by the provider. If the address matches, the data are kept; otherwise, they are discarded.

Devices

To use a cable network for data transmission, we need two key devices: a CM and a CMTS. The **cable modem (CM)** is installed on the subscriber premises. It is similar to an ADSL modem. Figure 3.30 shows its location. The **cable modem transmission system (CMTS)** is installed inside the distribution hub by the cable company. It receives data from the Internet and passes them to the combiner, which sends them to the subscriber. The CMTS also receives data from the subscriber and passes them to the Internet. Figure 3.30 shows the location of the CMTS.

Figure 3.30 *Cable modem configurations*



T Lines

T lines are standard digital telephone carriers originally designed to multiplex voice channels (after being digitized). Today, however, T lines can be used to carry data from a residence or an organization to the Internet. They can also be used to provide a physical link between nodes in a switched wide area network. T lines are commercially available in two data rates: T-1 and T-3 (see Table 3.8).

Table 3.8 *T line rates*

Line	*Rate (Mbps)*
T-1	1.544
T-3	44.736

T-1 Line

The data rate of a **T-1 line** is 1.544 Mbps. Twenty-four voice channels are sampled, with each sample digitized to 8 bits. An extra bit is added to provide synchronization. This makes the frame 193 bits in length. By sending 8000 frames per second, we get a data rate of 1.544 Mbps. When we use a T-1 line to connect to the Internet, we can use all or part of the capacity of the line to send digital data.

T-3 Line

A **T-3 line** has a data rate of 44.736 Mbps. It is equivalent to 28 T-1 lines. Many subscribers may not need the entire capacity of a T line. To accommodate these customers, the telephone companies have developed fractional T line services, which allow several subscribers to share one line by multiplexing their transmissions.

SONET

The high bandwidths of fiber-optic cable are suitable for today's highest data rate technologies (such as video conferencing) and for carrying large numbers of lower-rate technologies at the same time. ANSI created a set of standards called **Synchronous Optical Network (SONET)** to handle the use of fiber-optic cables. It defines a high-speed data carrier.

SONET first defines a set of electrical signals called **synchronous transport signals (STSs).** It then converts these signals to optical signals called **optical carriers (OCs).** The optical signals are transmitted at 8000 frames per second.

Table 3.9 shows the data rates for STSs and OCs. Note that the lowest level in this hierarchy has a data rate of 51.840 Mbps, which is greater than that of a T-3 line (44.736 Mbps).

Table 3.9 *SONET rates*

STS	*OC*	*Rate (Mbps)*	*STS*	*OC*	*Rate (Mbps)*
STS-1	OC-1	51.840	STS-24	OC-24	1244.160
STS-3	OC-3	155.520	STS-36	OC-36	1866.230
STS-9	OC-9	466.560	STS-48	OC-48	2488.320
STS-12	OC-12	622.080	STS-96	OC-96	4976.640
STS-18	OC-18	933.120	STS-192	OC-192	9953.280

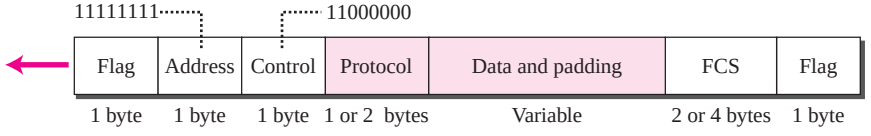
PPP

The telephone line or cable companies provide a physical link, but to control and manage the transfer of data, there is a need for a special protocol. The **Point-to-Point Protocol (PPP**) was designed to respond to this need.

PPP Layers

PPP has only physical and data link layers. No specific protocol is defined for the physical layer by PPP. Instead, it is left to the implementer to use whatever is available. PPP supports any of the protocols recognized by ANSI. At the data link layer, PPP defines the format of a frame and the protocol that are used for controlling the link and transporting user data. The format of a PPP frame is shown in Figure 3.31.

Figure 3.31 *PPP frame*



The descriptions of the fields are as follows:

1. **Flag field.** The flag field identifies the boundaries of a PPP frame. Its value is 01111110.
2. **Address field.** Because PPP is used for a point-to-point connection, it uses the broadcast address used in most LANs, 11111111, to avoid a data link address in the protocol.
3. **Control field.** The control field is assigned the value 11000000 to show that, as in most LANs, the frame has no sequence number; each frame is independent.
4. **Protocol field.** The protocol field defines the type of data being carried in the data field: user data or other information.
5. **Data field.** This field carries either user data or other information.
6. **FCS.** The frame check sequence field is simply a 2-byte or 4-byte CRC used for error detection.

Link Control Protocol (LCP)

The **Link Control Protocol (LCP)** is responsible for establishment, maintenance, and termination of the link. When the data field of a frame is carrying data related to this protocol, it means that PPP is handling the link; it does not carry data.

Network Control Protocol (NCP)

The **Network Control Protocol (NCP)** has been defined to give flexibility to PPP. PPP can carry data from different network protocols, including IP. After establishment of the link, PPP can carry IP packets in its data field.

PPPoE

PPP was designed to connect a single user to the Internet via a conventional modem and a telephone line. Today, DSL, cable modem, and wireless technology allow a group of users, on an Ethernet LAN, to access the Internet through a single physical line. In other words, the hosts connected to the LAN can share one single physical line to access the Internet. **PPP over Ethernet (PPPoE)** is a new protocol that uses a discovery technique to find the Ethernet address of the host to be connected to the Internet. After address discovery, a regular PPP session can be used to provide the connection.

3.4 SWITCHED WANS

The backbone networks in the Internet can be switched WANs. A switched WAN is a wide area network that covers a large area (a state or a country) and provides access at several points to the users. Inside the network, there is a mesh of point-to-point networks that connects switches. The switches, multiple port connectors, allow the connection of several inputs and outputs.

Switched WAN technology differs from LAN technology in many ways. First, instead of a star topology, switches are used to create multiple paths. Second, LAN technology is considered a connectionless technology; there is no relationship between packets sent by a sender to a receiver. Switched WAN technology, on the other hand, is a connection-oriented technology. Before a sender can send a packet, a connection must be established between the sender and the receiver. After the connection is established, it is assigned an identifier (sometimes called a label) used during the transmission. The connection is formally terminated when the transmission is over. The connection identifier is used instead of the source and destination addresses in LAN technology.

X.25

The **X.25** protocol, introduced in the 1970s, was the first switched WAN to become popular both in Europe and the United States. It was mostly used as a public network to connect individual computers or LANs. It provides an end-to-end service.

Although X.25 was used as the WAN to carry IP packets from one part of the world to another, there was always a conflict between IP and X.25. IP is a third-(network) layer protocol. An IP packet is supposed to be carried by a frame at the second (data link) layer. X.25, which was designed before the Internet, is a three-layer protocol; it has its own network layer. IP packets had to be encapsulated in an X.25 network-layer packet to be carried from one side of the network to another. This is analogous to a person who has a car but has to load it in a truck to go from one point to another.

Another problem with X.25 is that it was designed at a time when transmission media were not very reliable (no use of optical fibers). For this reason, X.25 performs extensive error control. This makes transmission very slow and is not popular given the ever increasing demand for speed.

Frame Relay

The **Frame Relay** protocol, a switched technology that provides low-level (physical and data link layers) service, was designed to replace X.25. Frame Relay has some advantages over X.25:

- **High Data Rate.** Although Frame Relay was originally designed to provide a 1.544-Mbps data rate (equivalent to a T-1 line), today most implementations can handle up to 44.736 Mbps (equivalent to a T-3 line).
- **Bursty Data.** Some services offered by wide area network providers assume that the user has a fixed-rate need. For example, a T-1 line is designed for a user who wants to use the line at a consistent 1.544 Mbps. This type of service is not suitable for the many users today who need to send **bursty data** (non-fixed-rate data). For example, a user may want to send data at 6 Mbps for 2 seconds, 0 Mbps (nothing) for 7 seconds, and 3.44 Mbps for 1 second for a total of 15.44 Mb during a period of 10 seconds. Although the average data rate is still 1.544 Mbps, the T-1 line cannot fulfill this type of demand because it is designed for fixed-rate data, not bursty data. Bursty data requires what is called **bandwidth on demand.** The user needs different bandwidth allocations at different times. Frame Relay accepts bursty data. A user is granted an average data rate that can be exceeded when needed.
- **Less Overhead Due to Improved Transmission Media.** The quality of transmission media has improved tremendously since the last decade. They are more reliable and less error prone. There is no need to have a WAN that spends time and resources checking and double-checking potential errors. X.25 provides extensive error checking and flow control. Frame Relay does not provide error checking or require acknowledgment in the data link layer. Instead, all error checking is left to the protocols at the network and transport layers that use the services of Frame Relay.

ATM

Asynchronous Transfer Mode (ATM) is the *cell relay* protocol designed by the ATM Forum and adopted by the ITU-T.

Design Goals

Among the challenges faced by the designers of ATM, six stand out. First and foremost is the need for a transmission system to optimize the use of high-data-rate transmission media, in particular optical fiber. Second is the need for a system that can interface with existing systems, such as the various packet networks, and provide wide area interconnectivity between them without lowering their effectiveness or requiring their replacement. Third is the need for a design that can be implemented inexpensively so that cost would not be a barrier to adoption. If ATM is to become the backbone of international communications, as intended, it must be available at low cost to every user who wants it. Fourth, the new system must be able to work with and support the existing telecommunications hierarchies (local loops, local providers, long-distance carriers, and so on). Fifth, the new system must be connection-oriented to ensure accurate and predictable delivery. And last but not least, one objective is to move as many of the functions to hardware as possible (for speed) and eliminate as many software functions as possible (again for speed).

Cell Networks

ATM is a *cell network*. A **cell** is a small data unit of fixed size that is the basic unit of data exchange in a cell network. In this type of network, all data are loaded into identical cells that can be transmitted with complete predictability and uniformity. Cells are multiplexed with other cells and routed through a cell network. Because each cell is the same size and all are small, any problems associated with multiplexing different-sized packets are avoided.

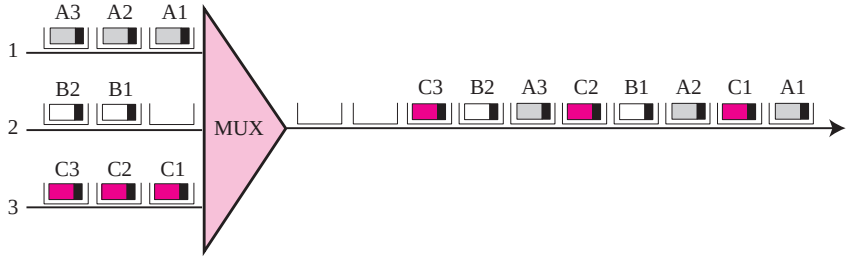
A cell network uses the cell as the basic unit of data exchange.
A cell is defined as a small, fixed-size block of information.

Asynchronous TDM

ATM uses **asynchronous time-division multiplexing**—that is why it is called Asynchronous Transfer Mode—to multiplex cells coming from different channels. It uses fixed-size slots the size of a cell. ATM multiplexers fill a slot with a cell from any input channel that has a cell; the slot is empty if none of the channels has a cell to send.

Figure 3.32 shows how cells from three inputs are multiplexed. At the first tick of the clock, channel 2 has no cell (empty input slot), so the multiplexer fills the slot with a cell from the third channel. When all the cells from all the channels are multiplexed, the output slots are empty.

Figure 3.32 *ATM multiplexing*

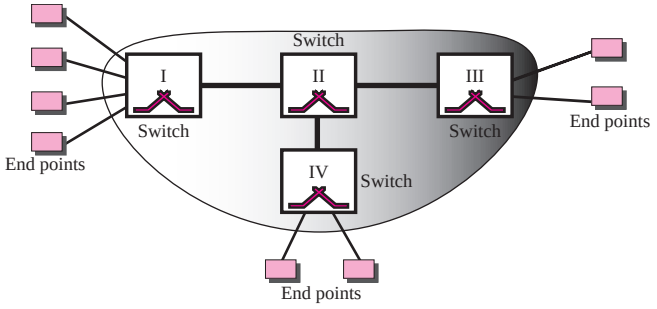


ATM Architecture

ATM is a switched network. The user access devices, called the end points, are connected to the switches inside the network. The switches are connected to each other using high-speed communication channels. Figure 3.33 shows an example of an ATM network.

Virtual Connection Connection between two end points is accomplished through transmission paths (TPs), virtual paths (VPs), and virtual circuits (VCs). A **transmission path (TP)** is the physical connection (wire, cable, satellite, and so on) between an end point and a switch or between two switches. Think of two switches as two cities. A transmission path is the set of all highways that directly connects the two cities.

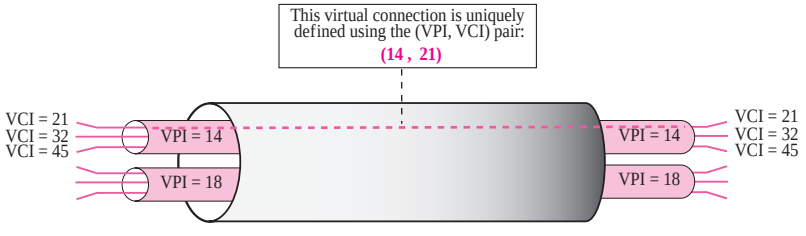
Figure 3.33 *Architecture of an ATM network*



A transmission path is divided into several virtual paths. A **virtual path (VP)** provides a connection or a set of connections between two switches. Think of a virtual path as a highway that connects two cities. Each highway is a virtual path; the set of all highways is the transmission path.

Cell networks are based on **virtual circuits (VCs).** All cells belonging to a single message follow the same virtual circuit and remain in their original order until they reach their destination. Think of a virtual circuit as the lanes of a highway (virtual path) as shown in Figure 3.34.

Figure 3.34 *Virtual circuits*



The figure also shows the relationship between a transmission path (a physical connection), virtual paths (a combination of virtual circuits that are bundled together because parts of their paths are the same), and virtual circuits that logically connect two points together.

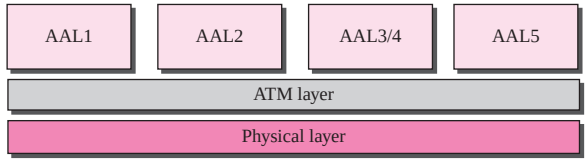
In a virtual circuit network, to route data from one end point to another, the virtual connections need to be identified. For this purpose, the designers of ATM created a hierarchical identifier with two levels: a **virtual path identifier (VPI)** and a **virtual circuit identifier (VCI).** The VPI defines the specific VP and the VCI defines a particular VC inside the VP. The VPI is the same for all virtual connections that are bundled (logically) into one VP.

A virtual connection is defined by a pair of numbers: the VPI and the VCI.

ATM Layers

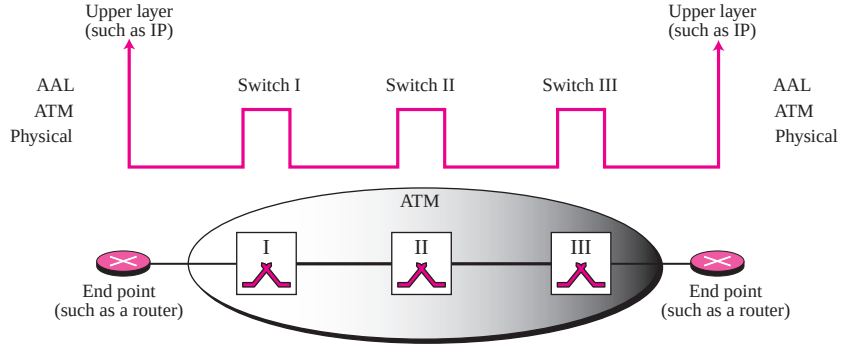
The ATM standard defines three layers. They are, from top to bottom, the application adaptation layer, the ATM layer, and the physical layer as shown in Figure 3.35.

Figure 3.35 *ATM layers*



The physical and ATM layer are used in both switches inside the network and end points (such as routers) that use the services of the ATM. The application adaptation layer (AAL) is used only by the end points. Figure 3.36 shows the use of these layers inside and outside an ATM network.

Figure 3.36 *Use of the layers*



AAL Layer

The **application adaptation layer (AAL)** allows existing networks (such as packet networks) to connect to ATM facilities. AAL protocols accept transmissions from upper-layer services (e.g., packet data) and map them into fixed-sized ATM cells. These transmissions can be of any type (voice, data, audio, video) and can be of variable or fixed rates. At the receiver, this process is reversed—segments are reassembled into their original formats and passed to the receiving service. Although four AAL layers have been defined the one which is of interest to us is AAL5, which is used to carry IP packets in the Internet.

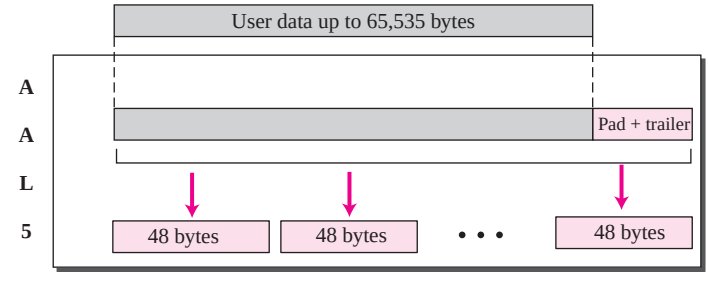
AAL5, which is sometimes called the **simple and efficient adaptation layer (SEAL)**, assumes that all cells belonging to a single message travel sequentially and that control functions are included in the upper layers of the sending application. AAL5

is designed for connectionless packet protocols that use a datagram approach to routing (such as the IP protocol in TCP/IP).

The IP protocol uses the AAL5 sublayer.

AAL5 accepts an IP packet of no more than 65,535 bytes and adds an 8-byte trailer as well as any padding required to ensure that the position of the trailer falls where the receiving equipment expects it (at the last 8 bytes of the last cell). See Figure 3.37. Once the padding and trailer are in place, AAL5 passes the message in 48-byte segments to the ATM layer.

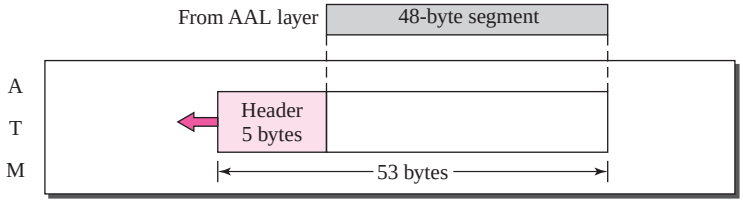
Figure 3.37 *AAL5*



ATM Layer

The ATM layer provides routing, traffic management, switching, and multiplexing services. It processes outgoing traffic by accepting 48-byte segments from the AAL sublayer. The addition of a 5-byte header transforms the segment into a 53-byte cell (see Figure 3.38).

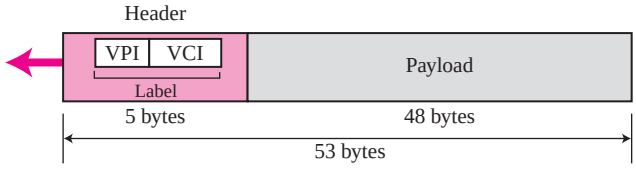
Figure 3.38 *ATM layer*



A cell is 53 bytes in length with 5 bytes allocated to header and 48 bytes carrying payload (user data may be less than 48 bytes). Most of the header is occupied by the VPI and VCI. Figure 3.39 shows the cell structure.

The combination of VPI and VCI can be thought of as a *label* that defines a particular virtual connection.

Figure 3.39 *An ATM cell*



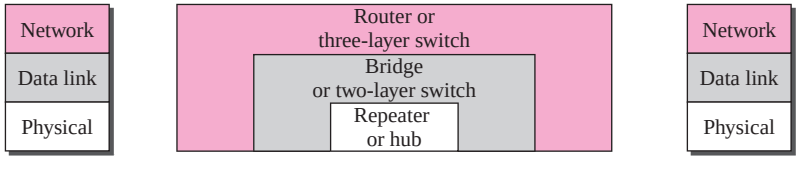
Physical Layer

The physical layer defines the transmission medium, bit transmission, encoding, and electrical to optical transformation. It provides convergence with physical transport protocols, such as SONET and T-3, as well as the mechanisms for transforming the flow of cells into a flow of bits.

3.5 CONNECTING DEVICES

LANs or WANs do not normally operate in isolation. They are connected to one another or to the Internet. To connect LANs and WANs together we use connecting devices. Connecting devices can operate in different layers of the Internet model. We discuss three kinds of **connecting devices:** repeaters (or hubs), bridges (or two-layer switches), and routers (or three-layer switches). Repeaters and hubs operate in the first layer of the Internet model. Bridges and two-layer switches operate in the first two layers. Routers and three-layer switches operate in the first three layers. Figure 3.40 shows the layers in which each device operates.

Figure 3.40 *Connecting devices*

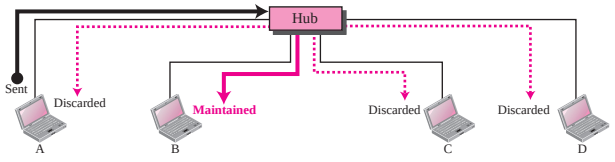


Repeaters

A **repeater** is a device that operates only in the physical layer. Signals that carry information within a network can travel a fixed distance before attenuation endangers the integrity of the data. A repeater receives a signal and, before it becomes too weak or corrupted, *regenerates* and *retimes* the original bit pattern. The repeater then sends the refreshed signal. In the past, when Ethernet LANs were using bus topology, a repeater was used to connect two segments of a LAN to overcome the length restriction of the

coaxial cable. Today, however, Ethernet LANs use star topology. In a star topology, a repeater is a multiport device, often called a **hub,** that can be used to serve as the connecting point and at the same time function as a repeater. Figure 3.41 shows that when a packet from station A to B arrives at the hub, the signal representing the frame is regenerated to remove any possible corrupting noise, but the hub forwards the packet from all outgoing port to all stations in the LAN. In other words, the frame is broadcast. All stations in the LAN receive the frame, but only station B keeps it. The rest of the stations discard it. Figure 3.41 shows the role of a repeater or a hub in a switched LAN.

Figure 3.41 *Repeater or hub*



The figure definitely shows that a hub does not have a filtering capability; it does not have the intelligence to find from which port the frame should be sent out.

A repeater forwards every bit; it has no filtering capability.

A hub or a repeater is a physical-layer device. They do not have any data-link address and they do not check the data-link address of the received frame. They just regenerate the corrupted bits and send them out from every port.

Bridges

A **bridge** operates in both the physical and the data link layers. As a physical-layer device, it regenerates the signal it receives. As a data link layer device, the bridge can check the MAC addresses (source and destination) contained in the frame.

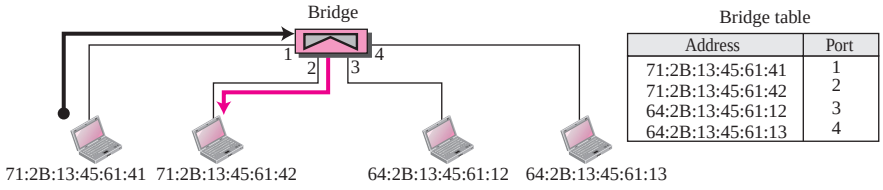
Filtering

One may ask what is the difference in functionality between a bridge and a repeater. A bridge has **filtering** capability. It can check the destination address of a frame and can decide from which outgoing port the frame should be send out.

A bridge has a table used in filtering decisions.

Let us give an example. In Figure 3.42, we have a LAN with four stations that are connected to a bridge. If a frame destined for station 71:2B:13:45:61:42 arrives at port 1, the bridge consults its table to find the departing port. According to its table, frames for 71:2B:13:45:61:42 should be sent out only through port 2; therefore, there is no need for forwarding the frame through other ports.

Figure 3.42 *Bridge*



A bridge does not change the physical (MAC) addresses in a frame.

Transparent Bridges

A **transparent bridge** is a bridge in which the stations are completely unaware of the bridge's existence. If a bridge is added or deleted from the system, reconfiguration of the stations is unnecessary. According to the IEEE 802.1d specification, a system equipped with transparent bridges must meet three criteria:

1. Frames must be forwarded from one station to another.
2. The forwarding table is automatically made by learning frame movements in the network.
3. Loops in the system must be prevented.

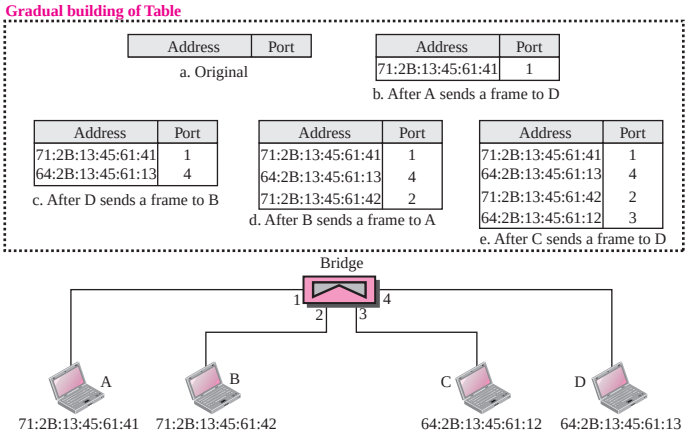
Forwarding A transparent bridge must correctly forward the frames, as discussed in the previous section.

Learning The earliest bridges had forwarding tables that were static. The system administrator would manually enter each table entry during bridge setup. Although the process was simple, it was not practical. If a station was added or deleted, the table had to be modified manually. The same was true if a station's MAC address changed, which is not a rare event. For example, putting in a new network card means a new MAC address.

A better solution to the static table is a dynamic table that maps addresses to ports automatically. To make a table dynamic, we need a bridge that gradually learns from the frame movements. To do this, the bridge inspects both the destination and the source addresses. The destination address is used for the forwarding decision (table lookup); the source address is used for adding entries to the table and for updating purposes. Let us elaborate on this process using Figure 3.43.

1. When station A sends a frame to station D, the bridge does not have an entry for either D or A. The frame goes out from all three ports; the frame floods the network. However, by looking at the source address, the bridge learns that station A must be connected to port 1. This means that frames destined for A, in the future, must be sent out through port 1. The bridge adds this entry to its table. The table has its first entry now.
2. When station D sends a frame to station B, the bridge has no entry for B, so it floods the network again. However, it adds one more entry to the table.
3. The learning process continues until the table has information about every port.

Figure 3.43 *Learning bridge*



However, note that the learning process may take a long time. For example, if a station does not send out a frame (a rare situation), the station will never have an entry in the table.

Two-Layer Switch

When we use the term *switch*, we must be careful because a switch can mean two different things. We must clarify the term by adding the level at which the device operates. We can have a two-layer switch or a three-layer switch. A **two-layer switch** performs at the physical and data link layer; it is a sophisticated bridge with faster forwarding capability.

Routers

A **router** is a three-layer device; it operates in the physical, data link, and network layers. As a physical layer device, it regenerates the signal it receives. As a data link layer device, the router checks the physical addresses (source and destination) contained in the packet. As a network layer device, a router checks the network layer addresses (addresses in the IP layer). Note that bridges change collision domains, but routers limit broadcast domains.

A router is a three-layer (physical, data link, and network) device.

A router can connect LANs together; a router can connect WANs together; and a router can connect LANs and WANs together. In other words, a router is an internetworking device; it connects independent networks together to form an internetwork. According to this definition, two networks (LANs or WANs) connected by a router become an internetwork or an internet.

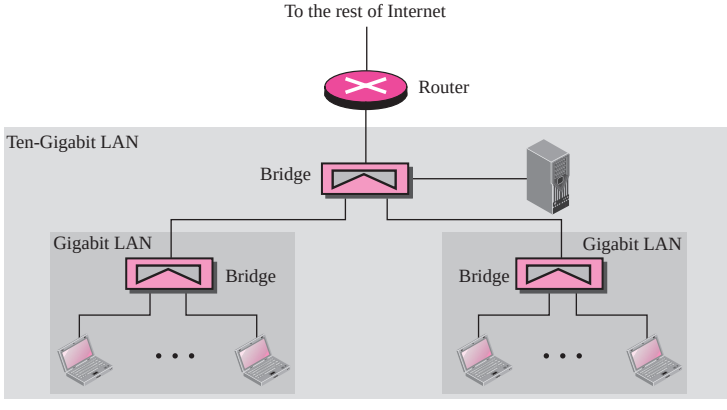
A repeater or a bridge connects segments of a LAN.
A router connects independent LANs or WANs to create an internetwork (internet).

There are three major differences between a router and a repeater or a bridge.

1. A router has a physical and logical (IP) address for each of its interfaces.
2. A router acts only on those packets in which the physical destination address matches the address of the interface at which the packet arrives.
3. A router changes the physical address of the packet (both source and destination) when it forwards the packet.

Let us give an example. In Figure 3.44, assume an organization has two separate buildings with a Gigabit Ethernet LANs installed in each building. The organization uses bridges in each LAN. The two LANs can be connected together to form a larger LAN using Ten-Gigabit Ethernet technology that speeds up the connection to the Ethernet and the connection to the organization server. A router then can connect the whole system to the Internet.

Figure 3.44 *Routing example*



A router as we saw in Chapter 2, will change the MAC address it receives because the MAC addresses have only local jurisdictions.

We will learn more about routers and routing in future chapters after we have discussed IP addressing.

Three-Layer Switch

A **three-layer switch** is a router; a router with an improved design to allow better performance. A three-layer switch can receive, process, and dispatch a packet much faster than a traditional router even though the functionality is the same. In this book, to avoid confusion, we use the term router for a three-layer switch.

A router changes the physical addresses in a packet.

3.6 FURTHER READING

For more details about subjects discussed in this chapter, we recommend the following books: [For 07], [For 03], [Tan 03], and [Gar &Wid 04]. The items enclosed in brackets refer to the reference list at the end of the book.

3.7 KEY TERMS

AAL5
access point (AP)
application adaptation layer (AAL)
asymmetric digital subscriber line (ADSL)
asynchronous time-division multiplexing
Asynchronous Transfer Mode (ATM)
autonegotiation
bandwidth on demand
basic service set (BSS)
Bluetooth
bridge
BSS-transition mobility
cable modem (CM)
cable modem transmission system (CMTS)
cable TV
carrier extension
carrier sense multiple access (CSMA)
carrier sense multiple access with collision avoidance (CSMA/CA)
carrier sense multiple access with collision detection (CSMA/CD)
cell
community antenna TV (CATV)
connecting device
digital subscriber line (DSL)
digital subscriber line access multiplexer (DSLAM)
distributed interframe space (DIFS)
downloading
downstream data band
ESS-transition mobility
Ethernet
extended service set (ESS)
Fast Ethernet
fiber node
filtering
frame bursting
Frame Relay
Gigabit Ethernet
handshaking period
head end
hexadecimal notation
high bit rate digital subscriber line (HDSL)
hub
hybrid fiber-coaxial (HFC) network
IEEE 802.11
jamming signal
Link Control Protocol (LCP)
logical link control (LLC)
media access control (MAC)
network allocation vector (NAV)
Network Control Protocol (NCP)
network interface card (NIC)
no-transition mobility
optical carrier (OC)
piconet
point coordination function (PCF)
Point-to-Point Protocol (PPP)
PPP over Ethernet (PPPoE)
primary
Project 802
repeater
router
scatternet
secondaries
short interframe space (SIFS)
simple and efficient adaptation layer (SEAL)
Standard Ethernet
symmetric digital subscriber line (SDSL)
Synchronous Optical Network (SONET)
synchronous transport signal (STS)
T lines
T-1 line
T-3 line
Ten-Gigabit Ethernet
three-layer switch
transmission path (TP)
transparent bridge
two-layer switch
uploading

upstream data band
V.90
V.92
very high bit rate digital subscriber line (VDSL)
video band
virtual circuit (VC)
virtual circuit identifier (VCI)
virtual path (VP)
virtual path identifier (VPI)
wireless LAN
X.25

3.8 SUMMARY

- A local area network (LAN) is a computer network that is designed for a limited geographic area. The LAN market has seen several technologies such as Ethernet, token ring, token bus, FDDI, and ATM LAN. Some of these technologies survived for a while, but Ethernet is by far the dominant technology. Ethernet has gone through a long evolution. The most dominant versions of Ethernet today are Gigabit and Ten-Gigabit Ethernet.
- One of the dominant standards for wireless LAN is the one defined under IEEE 802.11 standard and sometimes called wireless Ethernet. Another popular technology is Bluetooth, which is a wireless LAN technology designed to connect devices of different functions such as telephones, notebooks, computers (desktop and laptop), cameras, printers, coffee makers, and so on.
- A point-to-point WAN technology provides a direct connection to the Internet using regular telephone lines and traditional modems, DSL lines, cable modems, T-lines, or SONET networks. The Point-to-Point Protocol (PPP) was designed for users who need a reliable point-to-point connection to the Internet. PPP operates at the physical and data link layers of the OSI model.
- A switched WAN technology provides a backbone connection in the Internet. Asynchronous Transfer Mode (ATM) is the cell relay protocol designed to support the transmission of data, voice, and video through high data-rate transmission media such as fiber-optic cable.
- Connecting devices can connect segments of a network together; they can also connect networks together to create an internet. There are three types of connecting devices: repeaters (hubs), bridges (two-layer switches), and routers (three-layer switches). Repeaters regenerate a signal at the physical layer. A hub is a multiport repeater. Bridges have access to station addresses and can forward or filter a packet in a network. They operate at the physical and data link layers. A two-layer switch is a sophisticated bridge. Routers determine the path a packet should take. They operate at the physical, data link, and network layers. A three-layer switch is a sophisticated router.

3.9 PRACTICE SET

Exercises

1. Imagine the length of a 10Base5 cable is 2500 meters. If the speed of propagation in a thick coaxial cable is 200,000,000 meters/second, how long does it take for a

bit to travel from the beginning to the end of the network? Ignore any propagation delay in the equipment.

2. Using the data in Exercise 2, find the maximum time it takes to sense a collision. The worst case occurs when data are sent from one end of the cable and the collision happens at the other end. Remember that the signal needs to make a round trip.
3. The data rate of 10Base5 is 10 Mbps. How long does it take to create the smallest frame? Show your calculation.
4. Using the data in Exercises 3 and 4, find the minimum size of an Ethernet frame for collision detection to work properly.
5. An Ethernet MAC sublayer receives 42 bytes of data from the LLC sublayer. How many bytes of padding must be added to the data?
6. An Ethernet MAC sublayer receives 1510 bytes of data from the LLC layer. Can the data be encapsulated in one frame? If not, how many frames need to be sent? What is the size of the data in each frame?
7. Compare and contrast CSMA/CD with CSMA/CA.
8. Use Table 3.10 to compare and contrast the fields in IEEE 802.3 and 802.11.

Table 3.10 *Exercise 8*

Fields	*IEEE 802.3 Field Size*	*IEEE 802.11 Field Size*
Destination address		
Source address		
Address 1		
Address 2		
Address 3		
Address 4		
FC		
D/ID		
SC		
PDU length		
Data and padding		
Frame body		
FCS (CRC)		

Research Activities

9. Traditional Ethernet uses a version of the CSMA/CD access method. It is called CSMA/CD with 1-persistent. Find some information about this method.
10. DSL uses a modulation technique called DMT. Find some information about this modulation technique and how it can be used in DSL.
11. PPP goes through different phases, which can be shown in a transition state diagram. Find the transition diagram for a PPP connection.
12. Find the format of an LCP packet (encapsulated in a PPP frame). Include all fields, their codes, and their purposes.

13. Find the format of an NCP packet (encapsulated in a PPP frame). Include all fields, their codes, and their purposes.
14. Find the format of an ICP packet (encapsulated in a PPP frame). Include all fields, their codes, and their purposes.
15. PPP uses two authentication protocols, PAP and CHAP. Find some information about these two protocols and how they are used in PPP.
16. Find how an IP packet can be encapsulated in ATM cells using AAL5 layer.
17. To prevent loops in a network using transparent bridges, one uses the spanning tree algorithm. Find some information about this algorithm and how it can prevent loops.