CHAPTER 15

Connecting LANs, Backbone Networks, and Virtual LANs

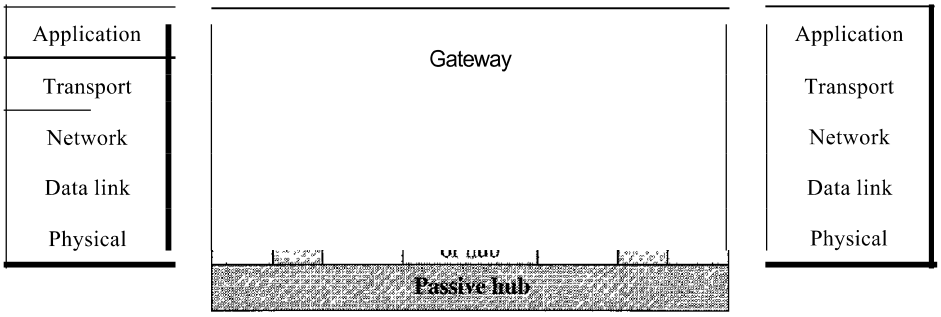
LANs do not normally operate **in** isolation. They are connected to one another or to the Internet. To connect LANs, or segments ofLANs, we use connecting devices. Connecting devices can operate **in** different layers of the Internet model. **In** this chapter, we discuss only those that operate **in** the physical and data link layers; we discuss those that operate in the first three layers in Chapter 19.

After discussing some connecting devices, we show how they are used to create backbone networks. Finally, we discuss virtual local area networks (VLANs).

15.1 CONNECTING DEVICES

In this section, we divide **connecting devices** into five different categories based on the layer **in** which they operate **in** a network, as shown **in** Figure 15.1.

Figure 15.1 *Five categories ofconnecting devices*



The five categories contain devices which can be defined as

1. Those which operate below the physical layer such as a passive hub.
2. Those which operate at the physical layer (a repeater or an active hub).
3. Those which operate at the physical and data link layers (a bridge or a two-layer switch).

4. Those which operate at the physical, data link, and network layers (a router or a three-layer switch).
5. Those which can operate at all five layers (a gateway).

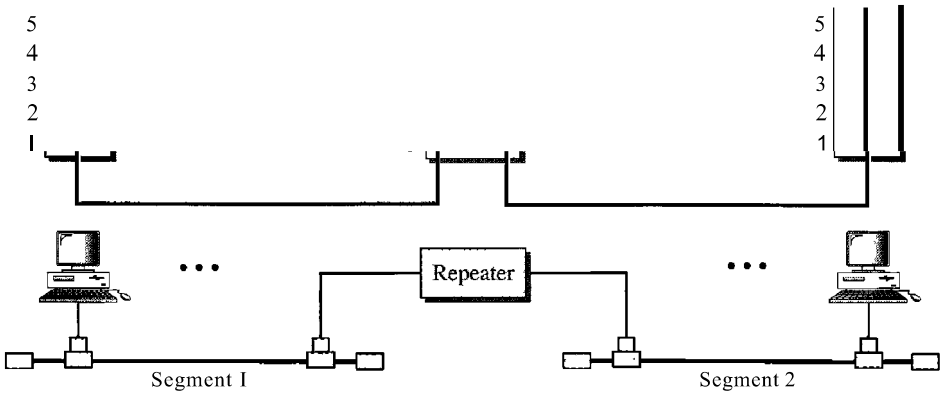
Passive Hubs

A passive hub is just a connector. It connects the wires coming from different branches. In a star-topology Ethernet LAN, a passive hub is just a point where the signals coming from different stations collide; the hub is the collision point. This type of a hub is part of the media; its location in the Internet model is below the physical layer.

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 the original bit pattern. The repeater then sends the refreshed signal. A repeater can extend the physical length of a LAN, as shown in Figure 15.2.

Figure 15.2 *A repeater connecting two segments of a LAN*



A repeater does not actually connect two LANs; it connects two segments of the same LAN. The segments connected are still part of one single LAN. A repeater is not a device that can connect two LANs of different protocols.

A repeater connects segments of a LAN.

A repeater can overcome the 10Base5 Ethernet length restriction. In this standard, the length of the cable is limited to 500 m. To extend this length, we divide the cable into segments and install repeaters between segments. Note that the whole network is still considered one LAN, but the portions of the network separated by repeaters are called segments. The repeater acts as a two-port node, but operates only in the physical layer. When it receives a frame from any of the ports, it regenerates and forwards it to the other port.

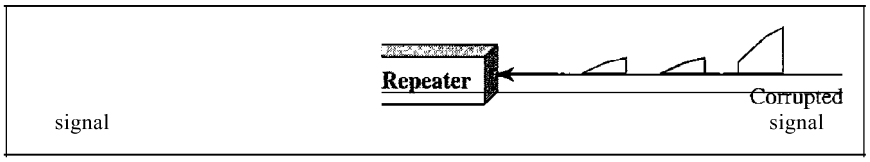
A repeater forwards every frame; it has no filtering capability.

It is tempting to compare a repeater to an amplifier, but the comparison is inaccurate. An amplifier cannot discriminate between the intended signal and noise; it amplifies equally everything fed into it. A repeater does not amplify the signal; it regenerates the signal. When it receives a weakened or corrupted signal, it creates a copy, bit for bit, at the original strength.

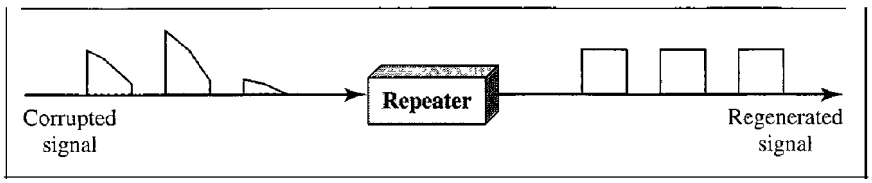
A repeater is a regenerator, not an amplifier.

The location of a repeater on a link is vital. A repeater must be placed so that a signal reaches it before any noise changes the meaning of any of its bits. A little noise can alter the precision of a bit's voltage without destroying its identity (see Figure 15.3). If the corrupted bit travels much farther, however, accumulated noise can change its meaning completely. At that point, the original voltage is not recoverable, and the error needs to be corrected. A repeater placed on the line before the legibility of the signal becomes lost can still read the signal well enough to determine the intended voltages and replicate them in their original form.

Figure 15.3 *Function of a repeater*



a. Right-to-teft transmission.



b. Left-to·right transmission.

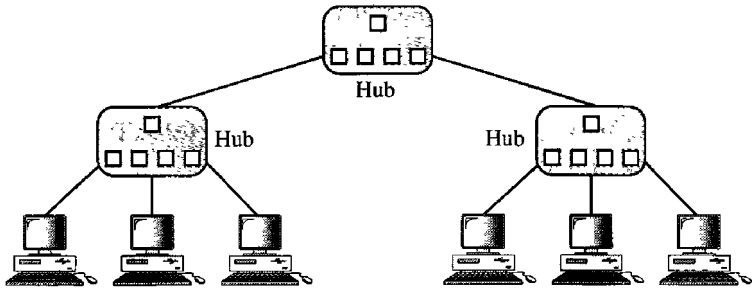
Active Hubs

An active hub is actually a multipart repeater. It is normally used to create connections between stations in a physical star topology. We have seen examples of hubs in some Ethernet implementations (lOBase-T, for example). However, hubs can also be used to create multiple levels of hierarchy, as shown in Figure 15.4. The hierarchical use of hubs removes the length limitation of 10Base-T (100 m).

Bridges

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

Figure 15.4 *A hierarchy of hubs*



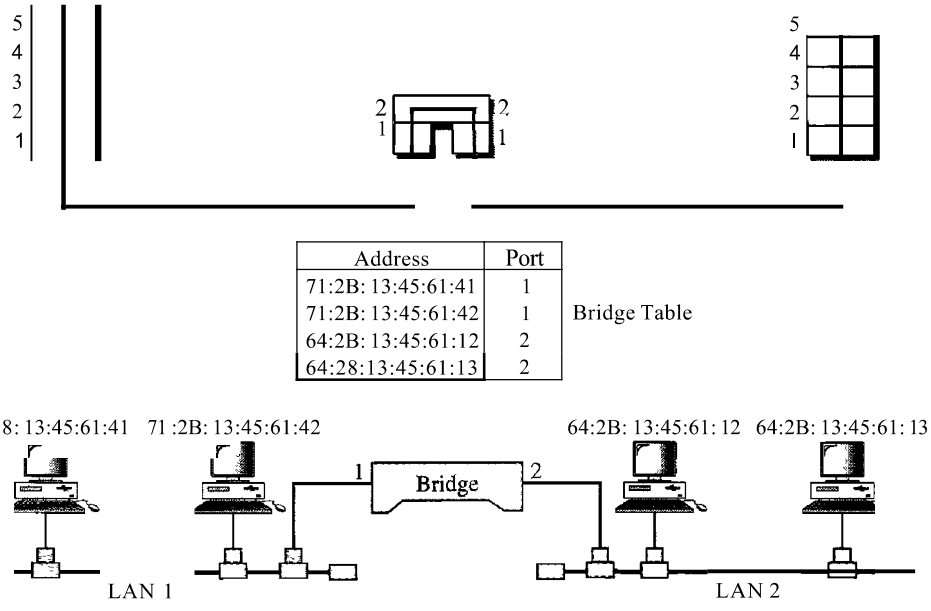
Filtering

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 decide if the frame should be forwarded or dropped. If the frame is to be forwarded, the decision must specify the port. A bridge has a table that maps addresses to ports.

A bridge has a table nsed in filtering decisions.

Let us give an example. In Figure 15.5, two LANs are connected by a bridge. If a frame destined for station 712B13456142 arrives at port 1, the bridge consults its table to find the departing port. According to its table, frames for 712B 13456142 leave through port 1; therefore, there is no need for forwarding, and the frame is dropped. On the other hand, if a frame for 712B13456141 arrives at port 2, the departing port is port 1

Figure 15.5 *A bridge connecting two LANs*



and the frame is forwarded. In the first case, LAN 2 remains free of traffic; in the second case, both LANs have traffic. In our example, we show a two-port bridge; in reality a bridge usually has more ports.

Note also that a bridge does not change the physical addresses contained in the frame.

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 802.1 d 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.

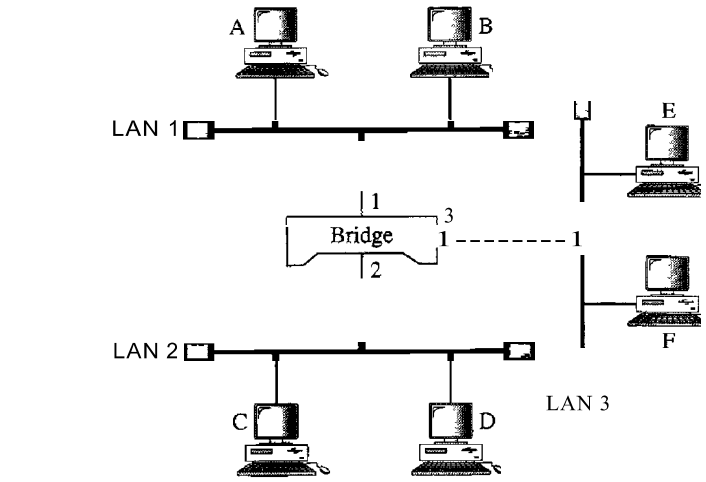
Forwarding A transparent bridge must correctly forward the frames, as discussed in the previous section.

Learning The earliest bridges had forwarding tables that were static. The systems 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 by using Figure 15.6.

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 located on the LAN 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 E sends a frame to station A, the bridge has an entry for A, so it forwards the frame only to port 1. There is no flooding. In addition, it uses the source address of the frame, E, to add a second entry to the table.
3. When station B sends a frame to C, the bridge has no entry for C, so once again it floods the network and adds one more entry to the table.
4. The process of learning continues as the bridge forwards frames.

Figure 15.6 *A learning bridge and the process of learning*



Address	Port

a. Original

Address	Port
A	1

b. After A sends a frame to D

Address	Port
A	1
E	3

c. After E sends a frame to A

Address	Port
A	1
E	3
B	1

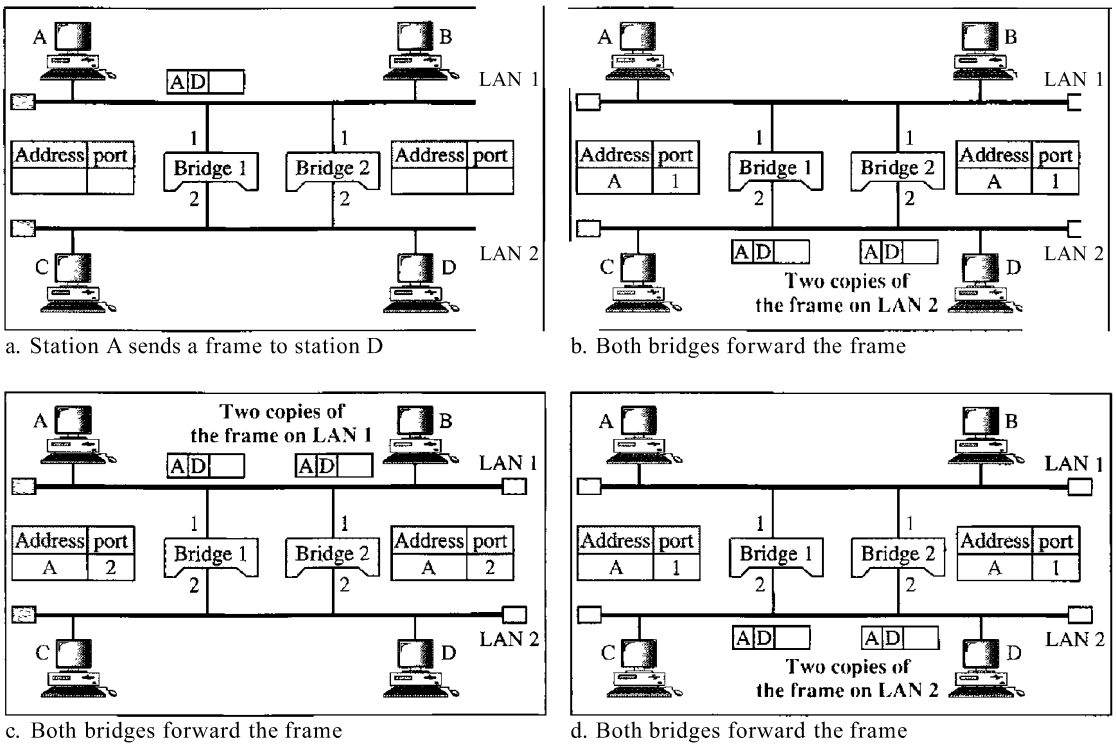
d. After B sends a frame to C

Loop Problem Transparent bridges work fine as long as there are no redundant bridges in the system. Systems administrators, however, like to have redundant bridges (more than one bridge between a pair of LANs) to make the system more reliable. If a bridge fails, another bridge takes over until the failed one is repaired or replaced. Redundancy can create loops in the system, which is very undesirable. Figure 15.7 shows a very simple example of a loop created in a system with two LANs connected by two bridges.

1. Station A sends a frame to station D. The tables of both bridges are empty. Both forward the frame and update their tables based on the source address A.
2. Now there are two copies of the frame on LAN 2. The copy sent out by bridge 1 is received by bridge 2, which does not have any information about the destination address D; it floods the bridge. The copy sent out by bridge 2 is received by bridge 1 and is sent out for lack of information about D. Note that each frame is handled separately because bridges, as two nodes on a network sharing the medium, use an access method such as CSMA/CD. The tables of both bridges are updated, but still there is no information for destination D.
3. Now there are two copies of the frame on LAN 1. Step 2 is repeated, and both copies flood the network.
4. The process continues on and on. Note that bridges are also repeaters and regenerate frames. So in each iteration, there are newly generated fresh copies of the frames.

To solve the looping problem, the IEEE specification requires that bridges use the spanning tree algorithm to create a loopless topology.

Figure 15.7 *Loop problem in a learning bridge*



a. Station A sends a frame to station D

b. Both bridges forward the frame

c. Both bridges forward the frame

d. Both bridges forward the frame

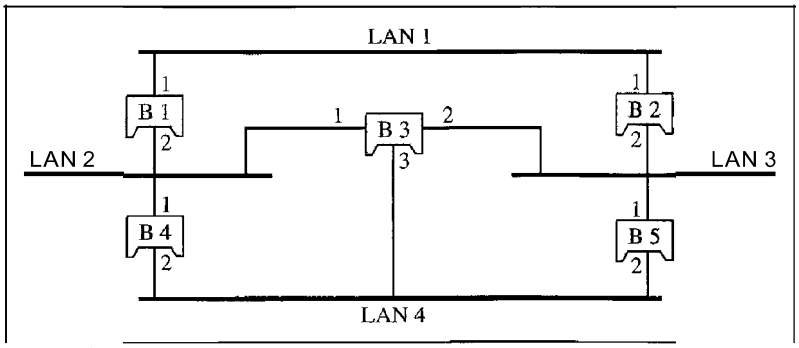
Spanning Tree

In graph theory, a **spanning tree** is a graph in which there is no loop. In a bridged LAN, this means creating a topology in which each LAN can be reached from any other LAN through one path only (no loop). We cannot change the physical topology of the system because of physical connections between cables and bridges, but we can create a logical topology that overlays the physical one. Figure 15.8 shows a system with four LANs and five bridges. We have shown the physical system and its representation in graph theory. Although some textbooks represent the LANs as nodes and the bridges as the connecting arcs, we have shown both LANs and bridges as nodes. The connecting arcs show the connection of a LAN to a bridge and vice versa. To find the spanning tree, we need to assign a cost (metric) to each arc. The interpretation of the cost is left up to the systems administrator. It may be the path with minimum hops (nodes), the path with minimum delay, or the path with maximum bandwidth. If two ports have the same shortest value, the systems administrator just chooses one. We have chosen the minimum hops. However, as we will see in Chapter 22, the hop count is normally 1 from a bridge to the LAN and 0 in the reverse direction.

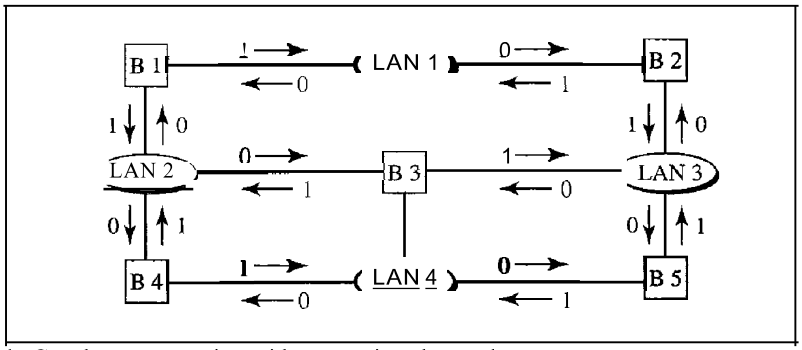
The process to find the spanning tree involves three steps:

1. Every bridge has a built-in ID (normally the serial number, which is unique). Each bridge broadcasts this ID so that all bridges know which one has the smallest ID. The bridge with the smallest ID is selected as the *root* bridge (root of the tree). We assume that bridge B1 has the smallest ID. It is, therefore, selected as the root bridge.

Figure 15.8 *A system of connected LANs and its graph representation*



a. Actual system



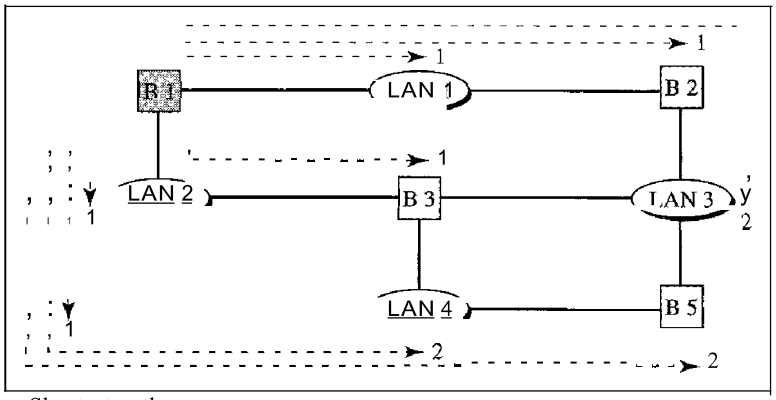
b. Graph representation with cost assigned to each arc

2. The algorithm tries to find the shortest path (a path with the shortest cost) from the root bridge to every other bridge or LAN. The shortest path can be found by examining the total cost from the root bridge to the destination. Figure 15.9 shows the shortest paths.
3. The combination of the shortest paths creates the shortest tree, which is also shown in Figure 15.9.
4. Based on the spanning tree, we mark the ports that are part of the spanning tree, the forwarding ports, which forward a frame that the bridge receives. We also mark those ports that are not part of the spanning tree, the blocking ports, which block the frames received by the bridge. Figure 15.10 shows the physical systems of LANs with forwarding points (solid lines) and blocking ports (broken lines).

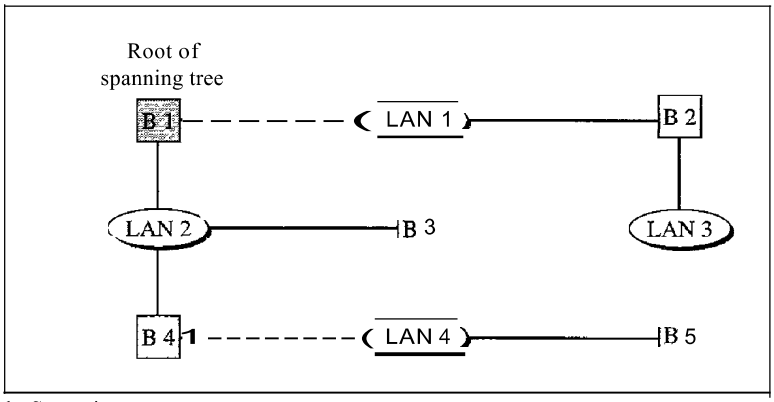
Note that there is only one single path from any LAN to any other LAN in the spanning tree system. This means there is only one single path from one LAN to any other LAN. No loops are created. You can prove to yourself that there is only one path from LAN 1 to LAN 2, LAN 3, or LAN 4. Similarly, there is only one path from LAN 2 to LAN 1, LAN 3, and LAN 4. The same is true for LAN 3 and LAN 4.

Dynamic Algorithm We have described the spanning tree algorithm as though it required manual entries. This is not true. Each bridge is equipped with a software package that carries out this process dynamically. The bridges send special messages to one another, called bridge protocol data units (BPDUs), to update the spanning tree. The spanning tree is updated when there is a change in the system such as a failure of a bridge or an addition or deletion of bridges.

Figure 15.9 *Finding the shortest paths and the spanning tree in a system of bridges*

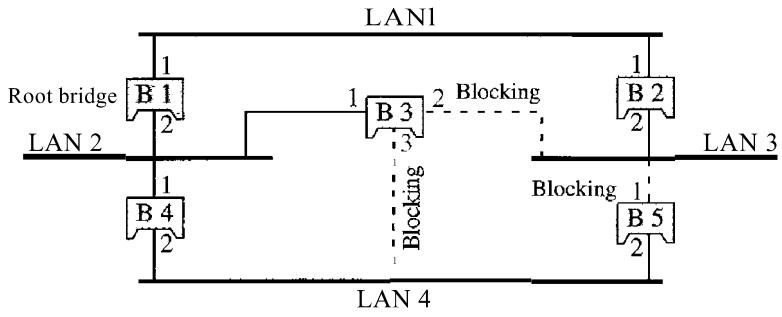


a. Shortest paths



b. Spanning tree

Figure 15.10 *Forwarding and blocking ports after using spanning tree algorithm*



Ports 2 and 3 of bridge B3 are blocking ports (no frame is sent out of these ports).
Port 1 of bridge B5 is also a blocking port (no frame is sent out of this port).

Source Routing Bridges

Another way to prevent loops in a system with redundant bridges is to use source **routing** bridges. A transparent bridge's duties include filtering frames, forwarding, and blocking. In a system that has source routing bridges, these duties are performed by the source station and, to some extent, the destination station.

In source routing, a sending station defines the bridges that the frame must visit. The addresses of these bridges are included in the frame. In other words, the frame contains not only the source and destination addresses, but also the addresses of all bridges to be visited.

The source gets these bridge addresses through the exchange of special frames with the destination prior to sending the data frame.

Source routing bridges were designed by IEEE to be used with Token Ring LANs. These LANs are not very common today.

Bridges Connecting Different LANs

Theoretically a bridge should be able to connect LANs using different protocols at the data link layer, such as an Ethernet LAN to a wireless LAN. However, there are many issues to be considered:

- **Frame format.** Each LAN type has its own frame format (compare an Ethernet frame with a wireless LAN frame).
- **Maximum data** size. If an incoming frame's size is too large for the destination LAN, the data must be fragmented into several frames. The data then need to be reassembled at the destination. However, no protocol at the data link layer allows the fragmentation and reassembly of frames. We will see in Chapter 19 that this is allowed in the network layer. The bridge must therefore discard any frames too large for its system.
- **Data rate.** Each LAN type has its own data rate. (Compare the 10-Mbps data rate of an Ethernet with the I-Mbps data rate of a wireless LAN.) The bridge must buffer the frame to compensate for this difference.
- **Bit order.** Each LAN type has its own strategy in the sending of bits. Some send the most significant bit in a byte first; others send the least significant bit first.
- **Security.** Some LANs, such as wireless LANs, implement security measures in the data link layer. Other LANs, such as Ethernet, do not. Security often involves encryption (see Chapter 30). When a bridge receives a frame from a wireless LAN, it needs to decrypt the message before forwarding it to an Ethernet LAN.
- **Multimedia support.** Some LANs support multimedia and the quality of services needed for this type of communication; others do not.

Two-Layer Switches

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 **three-layer switch** is used at the network layer; it is a kind of router. The **two-layer switch** performs at the physical and data link layers.

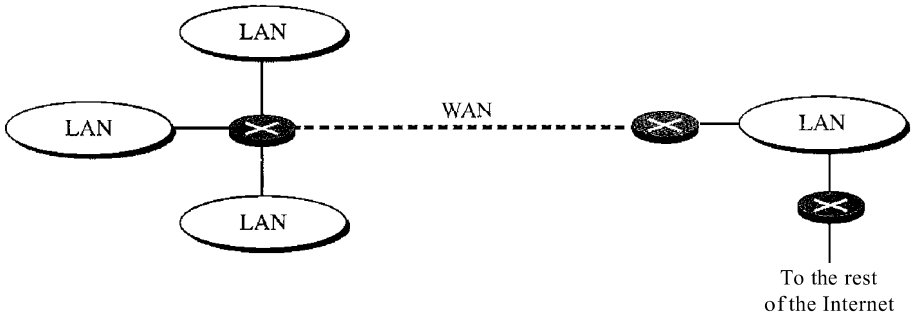
A two-layer switch is a bridge, a bridge with many ports and a design that allows better (faster) performance. A bridge with a few ports can connect a few LANs together. A bridge with many ports may be able to allocate a unique port to each station, with each station on its own independent entity. This means no competing traffic (no collision, as we saw in Ethernet).

A two-layer switch, as a bridge does, makes a filtering decision based on the MAC address of the frame it received. However, a two-layer switch can be more sophisticated. It can have a buffer to hold the frames for processing. It can have a switching factor that forwards the frames faster. Some new two-layer switches, called *cut-through* switches, have been designed to forward the frame as soon as they check the MAC addresses in the header of the frame.

Routers

A router is a three-layer device that routes packets based on their logical addresses (host-to-host addressing). A router normally connects LANs and WANs in the Internet and has a routing table that is used for making decisions about the route. The routing tables are normally dynamic and are updated using routing protocols. We discuss routers and routing in greater detail in Chapters 19 and 21. Figure 15.11 shows a part of the Internet that uses routers to connect LANs and WANs.

Figure 15.11 *Routers connecting independent LANs and WANs*



Three-Layer Switches

A three-layer switch is a router, but a faster and more sophisticated. The switching fabric in a three-layer switch allows faster table lookup and forwarding. In this book, we use the terms *router* and *three-layer switch* interchangeably.

Gateway

Although some textbooks use the terms *gateway* and *router* interchangeably, most of the literature distinguishes between the two. A gateway is normally a computer that operates in all five layers of the Internet or seven layers of OSI model. A gateway takes an application message, reads it, and interprets it. This means that it can be used as a connecting device between two internetworks that use different models. For example, a network designed to use the OSI model can be connected to another network using the Internet model. The gateway connecting the two systems can take a frame as it arrives from the first system, move it up to the OSI application layer, and remove the message.

Gateways can provide security. In Chapter 32, we learn that the gateway is used to filter unwanted application-layer messages.

15.2 BACKBONE NETWORKS

Some connecting devices discussed in this chapter can be used to connect LANs in a backbone network. A backbone network allows several LANs to be connected. In a backbone network, no station is directly connected to the backbone; the stations are part of a LAN, and the backbone connects the LANs. The backbone is itself a LAN that uses a LAN protocol such as Ethernet; each connection to the backbone is itself another LAN.

Although many different architectures can be used for a backbone, we discuss only the two most common: the bus and the star.

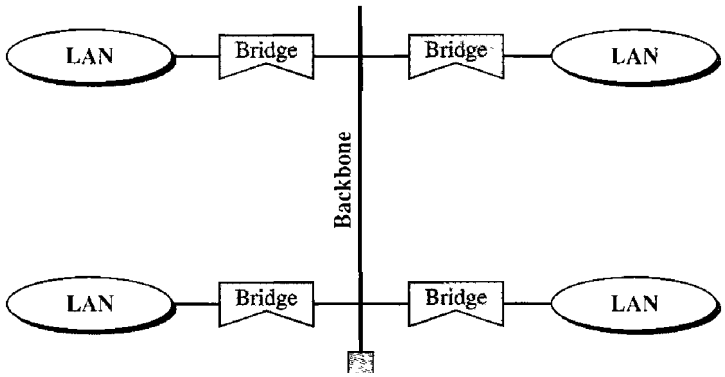
Bus Backbone

In a bus backbone, the topology of the backbone is a bus. The backbone itself can use one of the protocols that support a bus topology such as 10Base5 or 10Base2.

In a bus backbone, the topology of the backbone is a bus.

Bus backbones are normally used as a distribution backbone to connect different buildings in an organization. Each building can comprise either a single LAN or another backbone (normally a star backbone). A good example of a bus backbone is one that connects single- or multiple-floor buildings on a campus. Each single-floor building usually has a single LAN. Each multiple-floor building has a backbone (usually a star) that connects each LAN on a floor. A bus backbone can interconnect these LANs and backbones. Figure 15.12 shows an example of a bridge-based backbone with four LANs.

Figure 15.12 *Bus backbone*



In Figure 15.12, if a station in a LAN needs to send a frame to another station in the same LAN, the corresponding bridge blocks the frame; the frame never reaches the backbone. However, if a station needs to send a frame to a station in another LAN, the bridge passes the frame to the backbone, which is received by the appropriate bridge and is delivered to the destination LAN. Each bridge connected to the backbone has a table that shows the stations on the LAN side of the bridge. The blocking or delivery of a frame is based on the contents of this table.

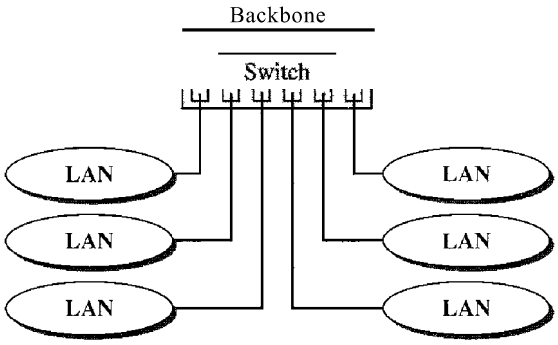
Star Backbone

In a star backbone, sometimes called a collapsed or switched backbone, the topology of the backbone is a star. In this configuration, the backbone is just one switch (that is why it is called, erroneously, a collapsed backbone) that connects the LANs.

In a star backbone, the topology of the backbone is a star; the backbone is just one switch.

Figure 15.13 shows a star backbone. Note that, in this configuration, the switch does the job of the backbone and at the same time connects the LANs.

Figure 15.13 *Star backbone*



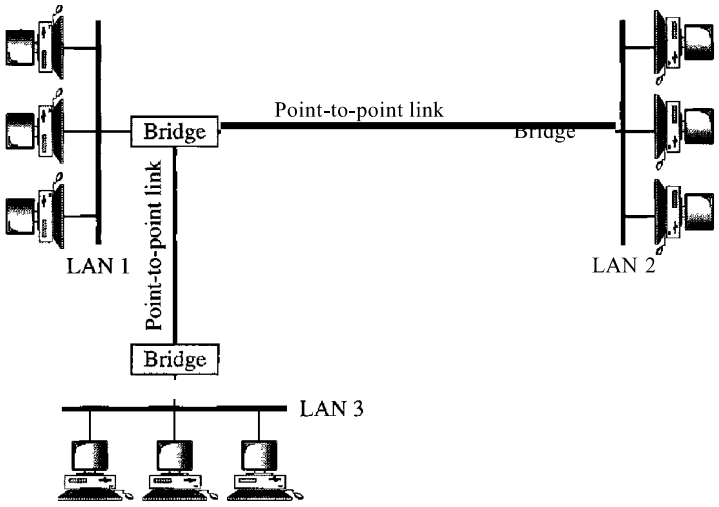
Star backbones are mostly used as a distribution backbone inside a building. In a multifloor building, we usually find one LAN that serves each particular floor. A star backbone connects these LANs. The backbone network, which is just a switch, can be installed in the basement or the first floor, and separate cables can run from the switch to each LAN. If the individual LANs have a physical star topology, either the hubs (or switches) can be installed in a closet on the corresponding floor, or all can be installed close to the switch. We often find a rack or chassis in the basement where the backbone switch and all hubs or switches are installed.

Connecting Remote LANs

Another common application for a backbone network is to connect remote LANs. This type of backbone network is useful when a company has several offices with LANs and needs to connect them. The connection can be done through bridges,

sometimes called remote bridges. The bridges act as connecting devices connecting LANs and point-to-point networks, such as leased telephone lines or ADSL lines. The point-to-point network in this case is considered a LAN without stations. The point-to-point link can use a protocol such as PPP. Figure 15.14 shows a backbone connecting remote LANs.

Figure 15.14 *Connecting remote IANs with bridges*



A point-to-point link acts as a LAN in a remote backbone connected by remote bridges.

15.3 VIRTUAL LANs

A station is considered part of a LAN if it physically belongs to that LAN. The criterion of membership is geographic. What happens if we need a virtual connection between two stations belonging to two different physical LANs? We can roughly define a virtual local area network (VLAN) as a local area network configured by software, not by physical wiring.

Let us use an example to elaborate on this definition. Figure 15.15 shows a switched LAN in an engineering firm in which 10 stations are grouped into three LANs that are connected by a switch. The first four engineers work together as the first group, the next three engineers work together as the second group, and the last three engineers work together as the third group. The LAN is configured to allow this arrangement.

But what would happen if the administrators needed to move two engineers from the first group to the third group, to speed up the project being done by the third group? The LAN configuration would need to be changed. The network technician must rewire. The problem is repeated if, in another week, the two engineers move back to their previous group. In a switched LAN, changes in the work group mean physical changes in the network configuration.

Figure 15.15 *A switch connecting three LANs*

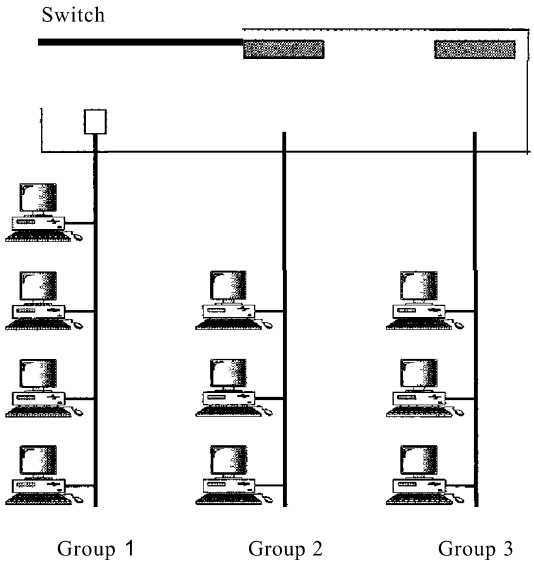
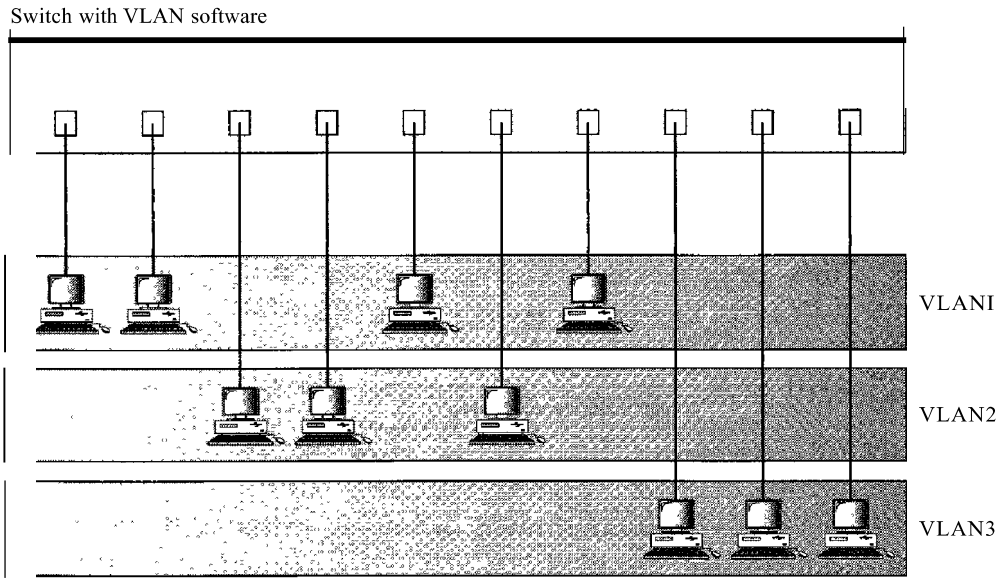


Figure 15.16 shows the same switched LAN divided into VLANs. The whole idea of VLAN technology is to divide a LAN into logical, instead of physical, segments. A LAN can be divided into several logical LANs called VLANs. Each VLAN is a work group in the organization. If a person moves from one group to another, there is no need to change the physical configuration. The group membership in VLANs is defined by software, not hardware. Any station can be logically moved to another VLAN. All members belonging to a VLAN can receive broadcast messages sent to that particular VLAN.

Figure 15.16 *A switch using VLAN software*

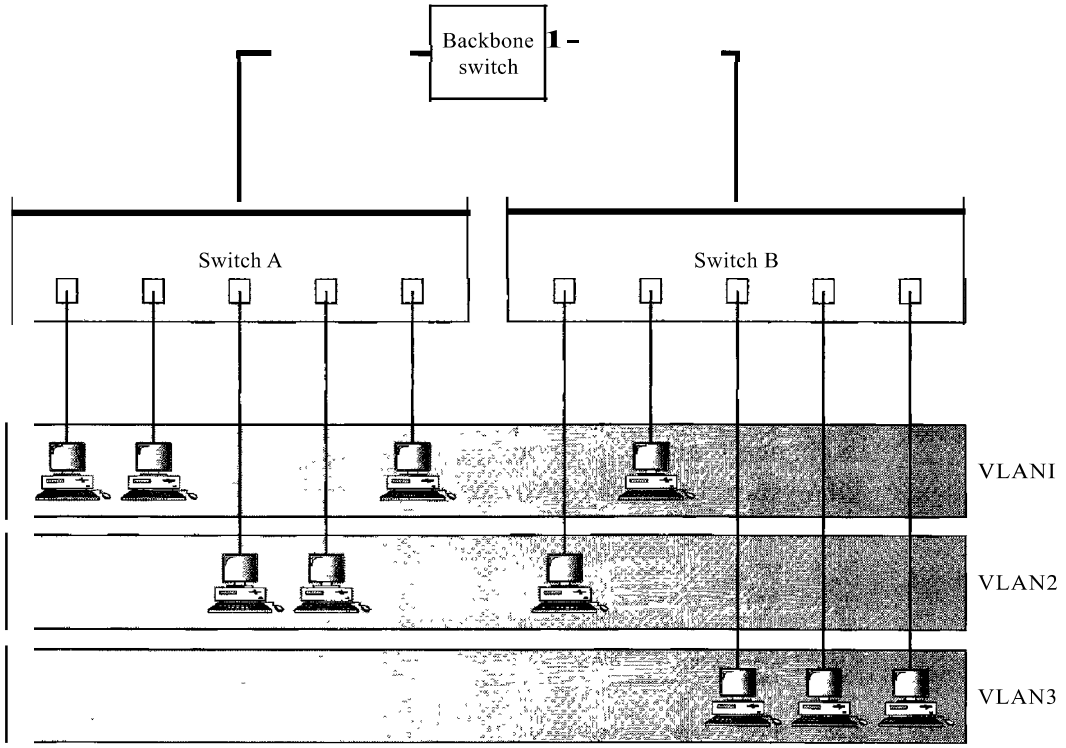


This means if a station moves from VLAN 1 to VLAN 2, it receives broadcast messages sent to VLAN 2, but no longer receives broadcast messages sent to VLAN 1.

It is obvious that the problem in our previous example can easily be solved by using VLANs. Moving engineers from one group to another through software is easier than changing the configuration of the physical network.

VLAN technology even allows the grouping of stations connected to different switches in a VLAN. Figure 15.17 shows a backbone local area network with two switches and three VLANs. Stations from switches A and B belong to each VLAN.

Figure 15.17 *Two switches in a backbone using VLAN software*



This is a good configuration for a company with two separate buildings. Each building can have its own switched LAN connected by a backbone. People in the first building and people in the second building can be in the same work group even though they are connected to different physical LANs.

From these three examples, we can define a VLAN characteristic:

VLANs create broadcast domains.

VLANs group stations belonging to one or more physical LANs into broadcast domains. The stations in a VLAN communicate with one another as though they belonged to a physical segment.

Membership

What characteristic can be used to group stations in a VLAN? Vendors use different characteristics such as port numbers, MAC addresses, IP addresses, IP multicast addresses, or a combination of two or more of these.

Port Numbers

Some VLAN vendors use switch port numbers as a membership characteristic. For example, the administrator can define that stations connecting to ports 1, 2, 3, and 7 belong to VLAN 1; stations connecting to ports 4, 10, and 12 belong to VLAN 2; and so on.

MAC Addresses

Some VLAN vendors use the 48-bit MAC address as a membership characteristic. For example, the administrator can stipulate that stations having MAC addresses E21342A12334 and F2A123BCD341 belong to VLAN 1.

IP Addresses

Some VLAN vendors use the 32-bit IP address (see Chapter 19) as a membership characteristic. For example, the administrator can stipulate that stations having IP addresses 181.34.23.67, 181.34.23.72, 181.34.23.98, and 181.34.23.112 belong to VLAN 1.

Multicast IP Addresses

Some VLAN vendors use the multicast IP address (see Chapter 19) as a membership characteristic. Multicasting at the IP layer is now translated to multicasting at the data link layer.

Combination

Recently, the software available from some vendors allows all these characteristics to be combined. The administrator can choose one or more characteristics when installing the software. In addition, the software can be reconfigured to change the settings.

Configuration

How are the stations grouped into different VLANs? Stations are configured in one of three ways: manual, semiautomatic, and automatic.

Manual Configuration

In a manual configuration, the network administrator uses the VLAN software to manually assign the stations into different VLANs at setup. Later migration from one VLAN to another is also done manually. Note that this is not a physical configuration; it is a logical configuration. The term *manually* here means that the administrator types the port numbers, the IP addresses, or other characteristics, using the VLAN software.

Automatic Configuration

In an automatic configuration, the stations are automatically connected or disconnected from a VLAN using criteria defined by the administrator. For example, the administrator can define the project number as the criterion for being a member of a group. When a user changes the project, he or she automatically migrates to a new VLAN.

Semiautomatic Configuration

A semiautomatic configuration is somewhere between a manual configuration and an automatic configuration. Usually, the initializing is done manually, with migrations done automatically.

Communication Between Switches

In a multiswitched backbone, each switch must know not only which station belongs to which VLAN, but also the membership of stations connected to other switches. For example, in Figure 15.17, switch A must know the membership status of stations connected to switch B, and switch B must know the same about switch A. Three methods have been devised for this purpose: table maintenance, frame tagging, and time-division multiplexing.

Table Maintenance

In this method, when a station sends a broadcast frame to its group members, the switch creates an entry in a table and records station membership. The switches send their tables to one another periodically for updating.

Frame Tagging

In this method, when a frame is traveling between switches, an extra header is added to the MAC frame to define the destination VLAN. The frame tag is used by the receiving switches to determine the VLANs to be receiving the broadcast message.

Time-Division Multiplexing (TDM)

In this method, the connection (trunk) between switches is divided into timeshared channels (see TDM in Chapter 6). For example, if the total number of VLANs in a backbone is five, each trunk is divided into five channels. The traffic destined for VLAN 1 travels in channell, the traffic destined for VLAN 2 travels in channel 2, and so on. The receiving switch determines the destination VLAN by checking the channel from which the frame arrived.

IEEE Standard

In 1996, the IEEE 802.1 subcommittee passed a standard called 802.1 Q that defines the format for frame tagging. The standard also defines the format to be used in multiswitched backbones and enables the use of multivendor equipment in VLANs. IEEE 802.1 Q has opened the way for further standardization in other issues related to VLANs. Most vendors have already accepted the standard.

Advantages

There are several advantages to using VLANs.

Cost and Time Reduction

VLANs can reduce the migration cost of stations going from one group to another. Physical reconfiguration takes time and is costly. Instead of physically moving one station to another segment or even to another switch, it is much easier and quicker to move it by using software.

Creating Virtual Work Groups

VLANs can be used to create virtual work groups. For example, in a campus environment, professors working on the same project can send broadcast messages to one another without the necessity of belonging to the same department. This can reduce traffic if the multicasting capability of IP was previously used.

Security

VLANs provide an extra measure of security. People belonging to the same group can send broadcast messages with the guaranteed assurance that users in other groups will not receive these messages.

15.4 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books and sites. The items in brackets [...] refer to the reference list at the end of the text.

Books

A book devoted to connecting devices is [PerOO]. Connecting devices and VLANs are discussed in Section 4.7 of [Tan03]. Switches, bridges, and hubs are discussed in [Sta03] and [Sta04].

Site

- IEEE 802 LAN/MAN Standards Committee

15.5 KEY TERMS

amplifier
blocking port
bridge
bus backbone
connecting device
filtering
forwarding port
hub
remote bridge
repeater
router
segment

source routing bridge
spanning tree
star backbone
three-layer switch
transparent bridge
two-layer switch
virtual local area network (VLAN)

15.6 SUMMARY

- A repeater is a connecting device that operates in the physical layer of the Internet model. A repeater regenerates a signal, connects segments of a LAN, and has no filtering capability.
- A bridge is a connecting device that operates in the physical and data link layers of the Internet model.
- A transparent bridge can forward and filter frames and automatically build its forwarding table.
- A bridge can use the spanning tree algorithm to create a loopless topology.
- A backbone LAN allows several LANs to be connected.
- A backbone is usually a bus or a star.
- A virtual local area network (VLAN) is configured by software, not by physical wiring.
- Membership in a VLAN can be based on port numbers, MAC addresses, IP addresses, IP multicast addresses, or a combination of these features.
- VLANs are cost- and time-efficient, can reduce network traffic, and provide an extra measure of security.

15.7 PRACTICE SET

Review Questions

1. How is a repeater different from an amplifier?
2. What do we mean when we say that a bridge can filter traffic? Why is filtering important?
3. What is a transparent bridge?
4. How does a repeater extend the length of a LAN?
5. How is a hub related to a repeater?
6. What is the difference between a forwarding port and a blocking port?
7. What is the difference between a bus backbone and a star backbone?
8. How does a VLAN save a company time and money?
9. How does a VLAN provide extra security for a network?
10. How does a VLAN reduce network traffic?
11. What is the basis for membership in a VLAN?

Exercises

12. Complete the table in Figure 15.6 after each station has sent a packet to another station.
13. Find the spanning tree for the system in Figure 15.7.
14. Find the spanning tree for the system in Figure 15.8 if bridge B5 is removed.
15. Find the spanning tree for the system in Figure 15.8 if bridge B2 is removed.
16. Find the spanning tree for the system in Figure 15.8 if B5 is selected as the root bridge.
17. In Figure 15.6, we are using a bridge. Can we replace the bridge with a router? Explain the consequences.
18. A bridge uses a filtering table; a router uses a routing table. Can you explain the difference?
19. Create a system of three LANs with four bridges. The bridges (B1 to B4) connect the LANs as follows:
    a. B1 connects LAN 1 and LAN 2.
    b. B2 connects LAN 1 and LAN 3.
    c. B3 connects LAN 2 and LAN 3.
    d. B4 connects LAN 1, LAN 2, and LAN 3.

    Choose B1as the root bridge. Show the forwarding and blocking ports, after applying the spanning tree procedure.
20. Which one has more overhead, a bridge or a router? Explain your answer.
21. Which one has more overhead, a repeater or a bridge? Explain your answer.
22. Which one has more overhead, a router or a gateway? Explain your answer.

CHAPTER 16

Wireless WANs: Cellular Telephone and Satellite Networks

We discussed wireless LANs in Chapter 14. Wireless technology is also used in cellular telephony and satellite networks. We discuss the former in this chapter as well as examples of channelization access methods (see Chapter 12). We also briefly discuss satellite networks, a technology that eventually will be linked to cellular telephony to access the Internet directly.

16.1 CELLULAR TELEPHONY

Cellular telephony is designed to provide communications between two moving units, called mobile stations (MSs), or between one mobile unit and one stationary unit, often called a land unit. A service provider must be able to locate and track a caller, assign a channel to the call, and transfer the channel from base station to base station as the caller moves out of range.

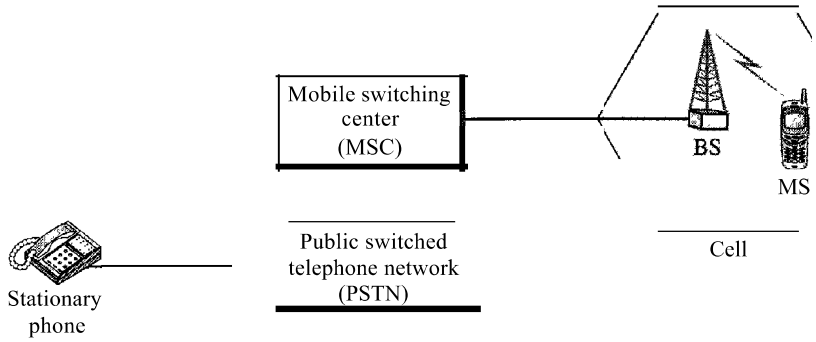
To make this tracking possible, each cellular service area is divided into small regions called cells. Each cell contains an antenna and is controlled by a solar or AC powered network station, called the base station (BS). Each base station, in tum, is controlled by a switching office, called a mobile switching center (MSC). The MSC coordinates communication between all the base stations and the telephone central office. It is a computerized center that is responsible for connecting calls, recording call information, and billing (see Figure 16.1).

Cell size is not fixed and can be increased or decreased depending on the population of the area. The typical radius of a cell is 1 to 12 mi. High-density areas require more, geographically smaller cells to meet traffic demands than do low-density areas. Once determined, cell size is optimized to prevent the interference of adjacent cell signals. The transmission power of each cell is kept low to prevent its signal from interfering with those of other cells.

Frequency-Reuse Principle

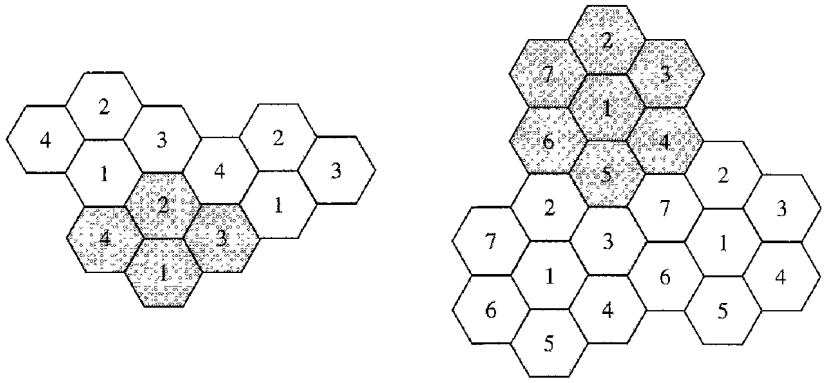
In general, neighboring cells cannot use the same set of frequencies for communication because it may create interference for the users located near the cell boundaries. However, the set of frequencies available is limited, and frequencies need to be reused. A

Figure 16.1 *Cellular system*



frequency reuse pattern is a configuration of N cells, N being the **reuse factor,** in which each cell uses a unique set of frequencies. When the pattern is repeated, the frequencies can be reused. There are several different patterns. Figure 16.2 shows two of them.

Figure 16.2 *Frequency reuse patterns*



The numbers in the cells define the pattern. The cells with the same number in a pattern can use the same set of frequencies. We call these cells the *reusing cells.* As Figure 16.2 shows, in a pattern with reuse factor 4, only one cell separates the cells using the same set of frequencies. In the pattern with reuse factor 7, two cells separate the reusing cells.

Transmitting

To place a call from a mobile station, the caller enters a code of 7 or 10 digits (a phone number) and presses the send button. The mobile station then scans the band, seeking a setup channel with a strong signal, and sends the data (phone number) to the closest base station using that channel. The base station relays the data to the MSC. The MSC

sends the data on to the telephone central office. If the called party is available, a connection is made and the result is relayed back to the MSC. At this point, the MSC assigns an unused voice channel to the call, and a connection is established. The mobile station automatically adjusts its tuning to the new channel, and communication can begin.

Receiving

When a mobile phone is called, the telephone central office sends the number to the MSC. The MSC searches for the location of the mobile station by sending query signals to each cell in a process called *paging*. Once the mobile station is found, the MSC transmits a ringing signal and, when the mobile station answers, assigns a voice channel to the call, allowing voice communication to begin.

Handoff

It may happen that, during a conversation, the mobile station moves from one cell to another. When it does, the signal may become weak. To solve this problem, the MSC monitors the level of the signal every few seconds. If the strength of the signal diminishes, the MSC seeks a new cell that can better accommodate the communication. The MSC then changes the channel carrying the call (hands the signal off from the old channel to a new one).

Hard Handoff Early systems used a hard handoff. In a hard handoff, a mobile station only communicates with one base station. When the MS moves from one cell to another, communication must first be broken with the previous base station before communication can be established with the new one. This may create a rough transition.

Soft Handoff New systems use a soft handoff. In this case, a mobile station can communicate with two base stations at the same time. This means that, during handoff, a mobile station may continue with the new base station before breaking off from the old one.

Roaming

One feature of cellular telephony is called roaming. Roaming means, in principle, that a user can have access to communication or can be reached where there is coverage. A service provider usually has limited coverage. Neighboring service providers can provide extended coverage through a roaming contract. The situation is similar to snail mail between countries. The charge for delivery of a letter between two countries can be divided upon agreement by the two countries.

First Generation

Cellular telephony is now in its second generation with the third on the horizon. The first generation was designed for voice communication using analog signals. We discuss one first-generation mobile system used in North America, AMPS.

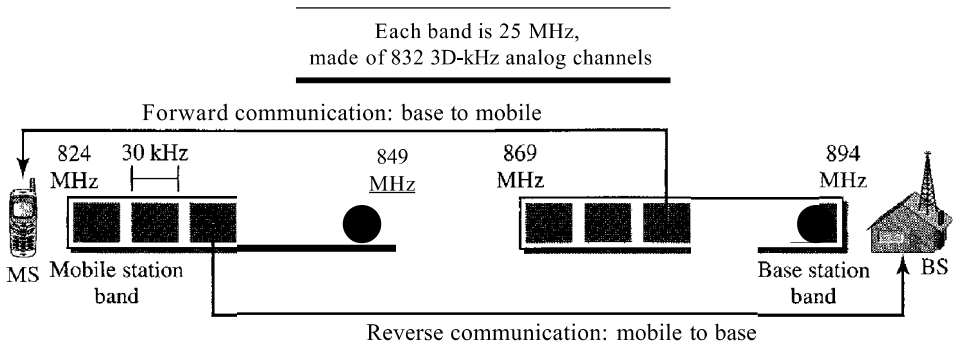
AMPS

Advanced **Mobile Phone** System (AMPS) is one of the leading analog cellular systems in North America. It uses FDMA (see Chapter 12) to separate channels in a link.

AMPS is an analog cellular phone system using FDMA.

Bands AMPS operates in the ISM 800-MHz band. The system uses two separate analog channels, one for forward (base station to mobile station) communication and one for reverse (mobile station to base station) communication. The band between 824 and 849 MHz carries reverse communication; the band between 869 and 894 MHz carries forward communication (see Figure 16.3).

Figure 16.3 *Cellular bands for AMPS*



Each band is divided into 832 channels. However, two providers can share an area, which means 416 channels in each cell for each provider. Out of these 416, 21 channels are used for control, which leaves 395 channels. AMPS has a frequency reuse factor of 7; this means only one-seventh of these 395 traffic channels are actually available in a cell.

Transmission AMPS uses FM and FSK for modulation. Figure 16.4 shows the transmission in the reverse direction. Voice channels are modulated using FM, and control channels use FSK to create 30-kHz analog signals. AMPS uses FDMA to divide each 25-MHz band into 3D-kHz channels.

Second Generation

To provide higher-quality (less noise-prone) mobile voice communications, the second generation of the cellular phone network was developed. While the first generation was designed for analog voice communication, the second generation was mainly designed for digitized voice. Three major systems evolved in the second generation, as shown in Figure 16.5. We will discuss each system separately.

Figure 16.4 *AMPS reverse communication band*

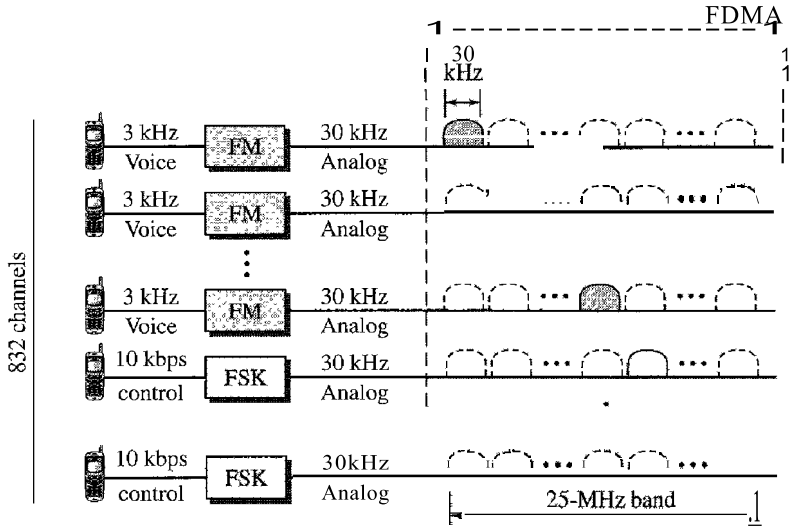
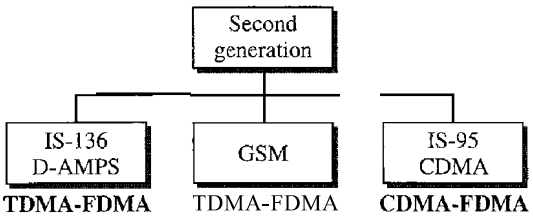


Figure 16.5 *Second-generation cellular phone systems*



D-AMPS

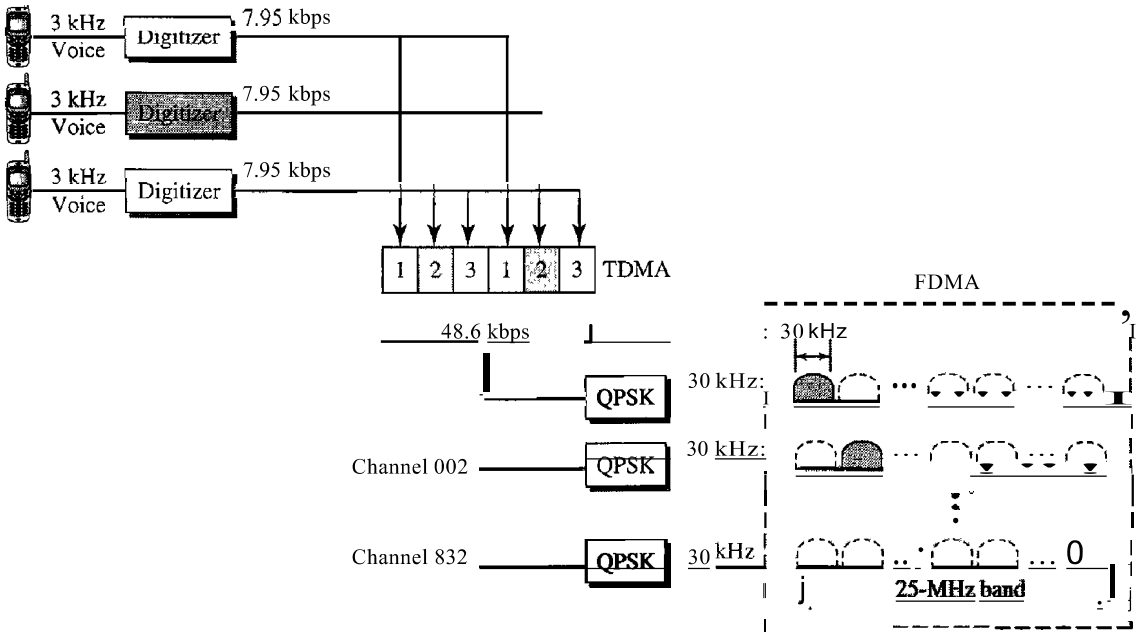
The product of the evolution of the analog AMPS into a digital system is digital AMPS (D-AMPS). D-AMPS was designed to be backward-compatible with AMPS. This means that in a cell, one telephone can use AMPS and another D-AMPS. D-AMPS was first defined by IS-54 (Interim Standard 54) and later revised by IS-136.

Band D-AMPS uses the same bands and channels as AMPS.

Transmission Each voice channel is digitized using a very complex PCM and compression technique. A voice channel is digitized to 7.95 kbps. Three 7.95-kbps digital voice channels are combined using TDMA. The result is 48.6 kbps of digital data; much of this is overhead. As Figure 16.6 shows, the system sends 25 frames per second, with 1944 bits per frame. Each frame lasts 40 ms (1/25) and is divided into six slots shared by three digital channels; each channel is allotted two slots.

Each slot holds 324 bits. However, only 159 bits comes from the digitized voice; 64 bits are for control and 101 bits are for error correction. In other words, each channel drops 159 bits of data into each of the two channels assigned to it. The system adds 64 control bits and 101 error-correcting bits.

Figure 16.6 *D-AMPS*



The resulting 48.6 kbps of digital data modulates a carrier using QPSK; the result is a 3D-kHz analog signal. Finally, the 3D-kHz analog signals share a 25-MHz band (FDMA). D-AMPS has a frequency reuse factor of 7.

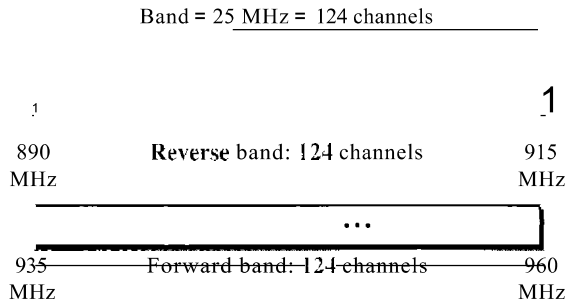
D-AMPS, or 18-136, is a digital cellular phone system using TDMA and FDMA.

GSM

The Global System for Mobile Communication (GSM) is a European standard that was developed to provide a common second-generation technology for all Europe. The aim was to replace a number of incompatible first-generation technologies.

Bands GSM uses two bands for duplex communication. Each band is 25 MHz in width, shifted toward 900 MHz, as shown in Figure 16.7. Each band is divided into 124 channels of 200 kHz separated by guard bands.

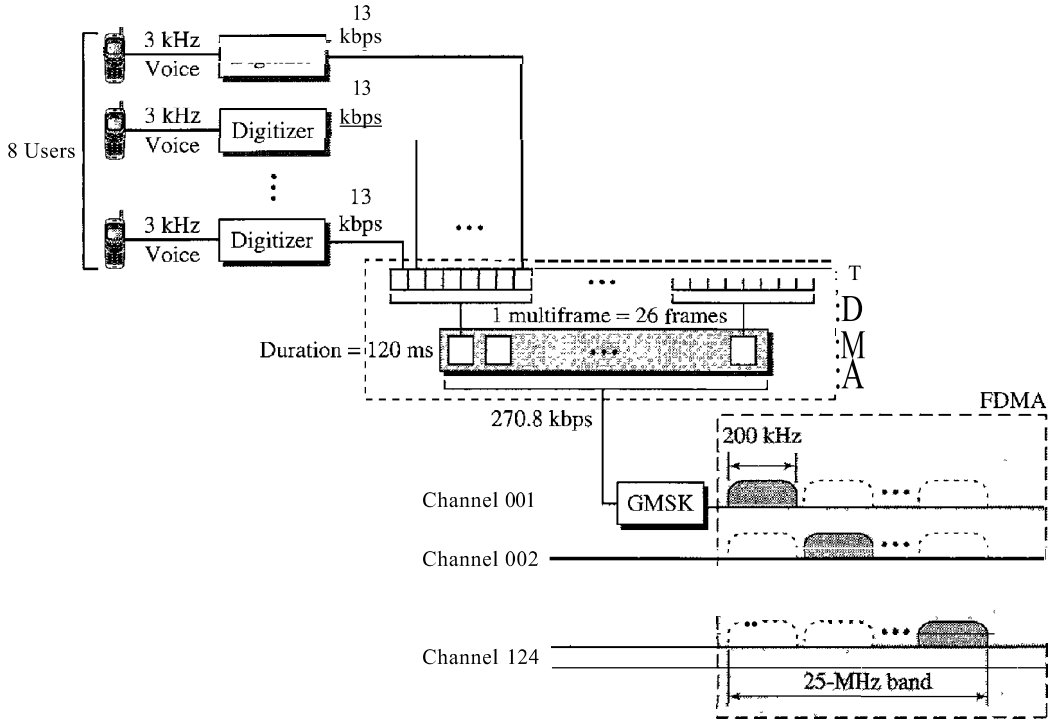
Figure 16.7 *GSM bands*



Transmission Figure 16.8 shows a GSM system. Each voice channel is digitized and compressed to a 13-kbps digital signal. Each slot carries 156.25 bits (see Figure 16.9). Eight slots share a frame (TDMA). Twenty-six frames also share a multiframe (TDMA). We can calculate the bit rate of each channel as follows:

$$\text{Channel data rate} = (11120 \text{ IDS}) \times 26 \times 8 \times 156.25 = 270.8 \text{ kbps}$$

Figure 16.8 *GSM*

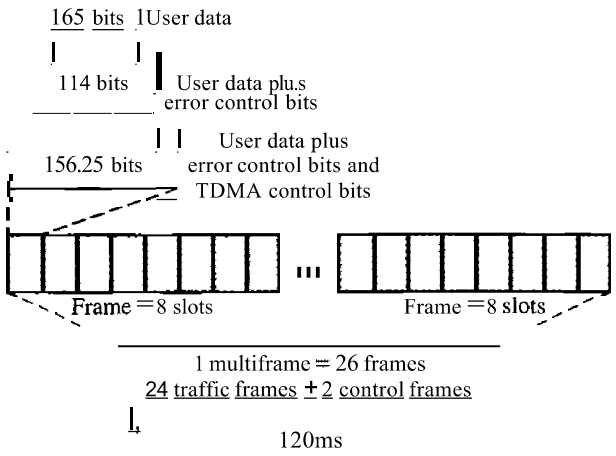


Each 270.8-kbps digital channel modulates a carrier using GMSK (a form of FSK used mainly in European systems); the result is a 200-kHz analog signal. Finally 124 analog channels of 200 kHz are combined using FDMA. The result is a 25-MHz band. Figure 16.9 shows the user data and overhead in a multiframe.

The reader may have noticed the large amount of overhead in TDMA. The user data are only 65 bits per slot. The system adds extra bits for error correction to make it 114 bits per slot. To this, control bits are added to bring it up to 156.25 bits per slot. Eight slots are encapsulated in a frame. Twenty-four traffic frames and two additional control frames make a multiframe. A multiframe has a duration of 120 ms. However, the architecture does define superframes and hyperframes that do not add any overhead; we will not discuss them here.

Reuse Factor Because of the complex error correction mechanism, GSM allows a reuse factor as low as 3.

Figure 16.9 *Multiframe components*



GSM is a digital cellular phone system using TDMA and FDMA.

IS-95

One of the dominant second-generation standards in North America is Interim Standard 95 (IS-95). It is based on CDMA and DSSS.

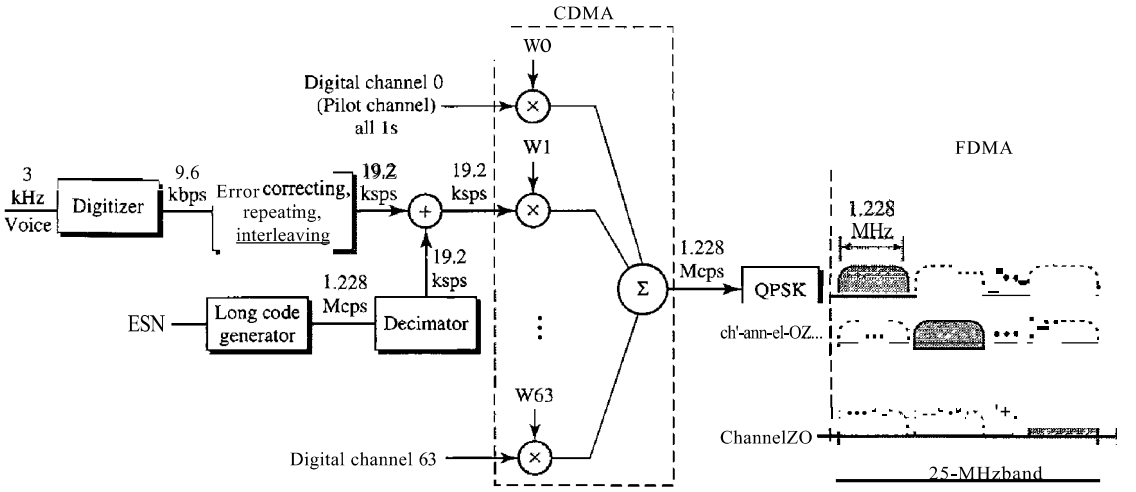
Bands and Channels IS-95 uses two bands for duplex communication. The bands can be the traditional ISM 800-MHz band or the ISM 1900-MHz band. Each band is divided into 20 channels of 1.228 MHz separated by guard bands. Each service provider is allotted 10 channels. IS-95 can be used in parallel with AMPS. Each IS-95 channel is equivalent to 41 AMPS channels (41 x 30 kHz = 1.23 MHz).

Synchronization All base channels need to be synchronized to use CDMA. To provide synchronization, bases use the services of GPS (Global Positioning System), a satellite system that we discuss in the next section.

Forward Transmission IS-95 has two different transmission techniques: one for use in the forward (base to mobile) direction and another for use in the reverse (mobile to base) direction. In the forward direction, communications between the base and all mobiles are synchronized; the base sends synchronized data to all mobiles. Figure 16.10 shows a simplified diagram for the forward direction.

Each voice channel is digitized, producing data at a basic rate of 9.6 kbps. After adding error-correcting and repeating bits, and interleaving, the result is a signal of 19.2 ksps (kilosignals per second). This output is now scrambled using a 19.2-ksps signal. The scrambling signal is produced from a long code generator that uses the electronic serial number (ESN) of the mobile station and generates 2^{42} pseudorandom chips, each chip having 42 bits. Note that the chips are generated pseudorandomly, not randomly, because the pattern repeats itself. The output of the long code generator is fed to a decimator, which chooses 1 bit out of 64 bits. The output of the decimator is used for scrambling. The scrambling is used to create privacy; the ESN is unique for each station.

Figure 16.10 *IS-95 forward transmission*



The result of the scrambler is combined using CDMA. For each traffic channel, one Walsh 64 x 64 row chip is selected. The result is a signal of 1.228 Mcps (megachips per second).

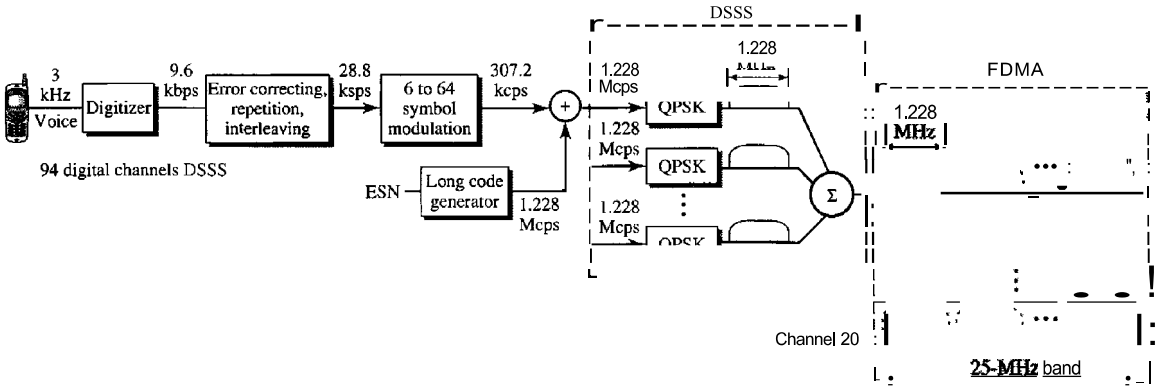
$$19.2 \text{ ksps} \times 64 \text{ cps} = 1.228 \text{ Mcps}$$

The signal is fed into a QPSK modulator to produce a signal of 1.228 MHz. The resulting bandwidth is shifted appropriately, using FDMA. An analog channel creates 64 digital channels, of which 55 channels are traffic channels (carrying digitized voice). Nine channels are used for control and synchronization:

- Channel 0 is a pilot channel. This channel sends a continuous stream of 1s to mobile stations. The stream provides bit synchronization, serves as a phase reference for demodulation, and allows the mobile station to compare the signal strength of neighboring bases for handoff decisions.
- Channel 32 gives information about the system to the mobile station.
- Channels 1 to 7 are used for paging, to send messages to one or more mobile stations.
- Channels 8 to 31 and 33 to 63 are traffic channels carrying digitized voice from the base station to the corresponding mobile station.

Reverse **Transmission** The use of CDMA in the forward direction is possible because the pilot channel sends a continuous sequence of Is to synchronize transmission. The synchronization is not used in the reverse direction because we need an entity to do that, which is not feasible. Instead of CDMA, the reverse channels use DSSS (direct sequence spread spectrum), which we discussed in Chapter 8. Figure 16.11 shows a simplified diagram for reverse transmission.

Figure 16.11 *IS-95 reverse transmission*



Each voice channel is digitized, producing data at a rate of 9.6 kbps. However, after adding error-correcting and repeating bits, plus interleaving, the result is a signal of 28.8 ksps. The output is now passed through a 6/64 symbol modulator. The symbols are divided into six-symbol chunks, and each chunk is interpreted as a binary number (from 0 to 63). The binary number is used as the index to a 64 x 64 Walsh matrix for selection of a row of chips. Note that this procedure is not CDMA; each bit is not multiplied by the chips in a row. Each six-symbol chunk is replaced by a 64-chip code. This is done to provide a kind of orthogonality; it differentiates the streams of chips from the different mobile stations. The result creates a signal of 307.2 kcps or (28.8/6) x 64.

Spreading is the next step; each chip is spread into 4. Again the ESN of the mobile station creates a long code of 42 bits at a rate of 1.228 Mcps, which is 4 times 307.2. After spreading, each signal is modulated using QPSK, which is slightly different from the one used in the forward direction; we do not go into details here. Note that there is no multiple-access mechanism here; all reverse channels send their analog signal into the air, but the correct chips will be received by the base station due to spreading.

Although we can create $2^{42} - 1$ digital channels in the reverse direction (because of the long code generator), normally 94 channels are used; 62 are traffic channels, and 32 are channels used to gain access to the base station.

IS-95 is a digital cellular phone system using CDMA/DSSS and FDMA.

Two Data Rate Sets IS-95 defines two data rate sets, with four different rates in each set. The first set defines 9600, 4800, 2400, and 1200 bps. If, for example, the selected rate is 1200 bps, each bit is repeated 8 times to provide a rate of 9600 bps. The second set defines 14,400, 7200, 3600, and 1800 bps. This is possible by reducing the number of bits used for error correction. The bit rates in a set are related to the activity of the channel. If the channel is silent, only 1200 bits can be transferred, which improves the spreading by repeating each bit 8 times.

Frequency-Reuse Factor In an IS-95 system, the frequency-reuse factor is normally 1 because the interference from neighboring cells cannot affect CDMA or DSSS transmission.

Soft Handoff Every base station continuously broadcasts signals using its pilot channel. This means a mobile station can detect the pilot signal from its cell and neighboring cells. This enables a mobile station to do a soft handoff in contrast to a hard handoff.

pes

Before we leave the discussion of second-generation cellular telephones, let us explain a term generally heard in relation to this generation: PCS. Personal communications system (peS) does not refer to a single technology such as GSM, 18-136, or 18-95. It is a generic name for a commercial system that offers several kinds of communication services. Common features of these systems can be summarized:

1. They may use any second-generation technology (GSM, IS-136, or IS-95).
2. They use the 1900-MHz band, which means that a mobile station needs more power because higher frequencies have a shorter range than lower ones. However, since a station's power is limited by the FCC, the base station and the mobile station need to be close to each other (smaller cells).
3. They offer communication services such as short message service (SMS) and limited Internet access.

Third Generation

The third generation of cellular telephony refers to a combination of technologies that provide a variety of services. Ideally, when it matures, the third generation can provide both digital data and voice communication. Using a small portable device, a person should be able to talk to anyone else in the world with a voice quality similar to that of the existing fixed telephone network. A person can download and watch a movie, can download and listen to music, can surf the Internet or play games, can have a video conference, and can do much more. One of the interesting characteristics of a third-generation system is that the portable device is always connected; you do not need to dial a number to connect to the Internet.

The third-generation concept started in 1992, when ITU issued a blueprint called the Internet Mobile Communication 2000 (IMT-2000). The blueprint defines some criteria for third-generation technology as outlined below:

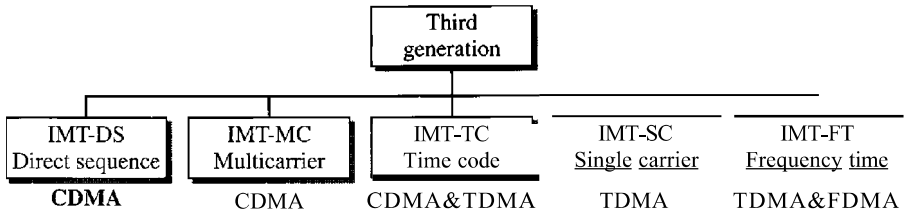
- Voice quality comparable to that of the existing public telephone network.
- Data rate of 144 kbps for access in a moving vehicle (car), 384 kbps for access as the user walks (pedestrians), and 2 Mbps for the stationary user (office or home).
- Support for packet-switched and circuit-switched data services.
- A band of 2 GHz.
- Bandwidths of 2 MHz.
- Interface to the Internet.

The main goal of third-generation cellular telephony is to provide universal personal communication.

IMT-2000 Radio Interface

Figure 16.12 shows the radio interfaces (wireless standards) adopted by 1MT-2000. All five are developed from second-generation technologies. The first two evolve from COMA technology. The third evolves from a combination of COMA and TOMA. The fourth evolves from TOMA, and the last evolves from both FOMA and TOMA.

Figure 16.12 *IMT-2000 radio interfaces*



IMT-DS This approach uses a version of COMA called wideband COMA or W-COMA. W-COMA uses a 5-MHz bandwidth. It was developed in Europe, and it is compatible with the COMA used in IS-95.

IMT-MC This approach was developed in North America and is known as COMA 2000. It is an evolution of COMA technology used in IS-95 channels. It combines the new wideband (I5-MHz) spread spectrum with the narrowband (l.25-MHz) COMA of IS-95. It is backward-compatible with IS-95. It allows communication on multiple 1.25-MHz channels (I, 3, 6, 9, 12 times), up to 15 MHz. The use of the wider channels allows it to reach the 2-Mbps data rate defined for the third generation.

IMT-TC This standard uses a combination of W-COMA and TDMA. The standard tries to reach the IMT-2000 goals by adding TOMA multiplexing to W-COMA.

IMT-SC This standard only uses TOMA.

IMT-FT This standard uses a combination of FDMA and TOMA.

16.2 SATELLITE NETWORKS

A satellite network is a combination of nodes, some of which are satellites, that provides communication from one point on the Earth to another. A node in the network can be a satellite, an Earth station, or an end-user terminal or telephone. Although a natural satellite, such as the Moon, can be used as a relaying node in the network, the use of artificial satellites is preferred because we can install electronic equipment on the satellite to regenerate the signal that has lost its energy during travel. Another restriction on using natural satellites is their distances from the Earth, which create a long delay in communication.

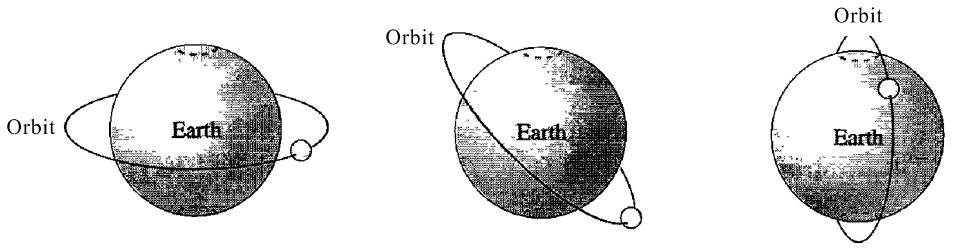
Satellite networks are like cellular networks in that they divide the planet into cells. Satellites can provide transmission capability to and from any location on Earth, no matter how remote. This advantage makes high-quality communication available to

undeveloped parts of the world without requiring a huge investment in ground-based infrastructure.

Orbits

An artificial satellite needs to have an **orbit,** the path in which it travels around the Earth. The orbit can be equatorial, inclined, or polar, as shown in Figure 16.13.

Figure 16.13 *Satellite orbits*



a. Equatorial-orbit satellite b. Inclined-orbit satellite c. Polar-orbit satellite

The period of a satellite, the time required for a satellite to make a complete trip around the Earth, is determined by Kepler's law, which defines the period as a function of the distance of the satellite from the center of the Earth.

Example 16.1

What is the period of the Moon, according to Kepler's law?

$$\text{Period} = C \times \text{distance}^{1.5}$$

Here C is a constant approximately equal to 1/100. The period is in seconds and the distance in kilometers.

Solution

The Moon is located approximately 384,000 km above the Earth. The radius of the Earth is 6378 km. Applying the formula, we get

$$\text{Period} = \frac{1}{100}(384{,}000 + 6378)^{1.5} = 2{,}439{,}090 \text{ s} = 1 \text{ month}$$

Example 16.2

According to Kepler's law, what is the period of a satellite that is located at an orbit approximately 35,786 km above the Earth?

Solution

Applying the formula, we get

$$\text{Period} = \frac{1}{100}(35{,}786 + 6378)^{1.5} = 86{,}579 \text{ s} = 24 \text{ h}$$

This means that a satellite located at 35,786 km has a period of 24 h, which is the same as the rotation period of the Earth. A satellite like this is said to be *stationary* to the Earth. The orbit, as we will see, is called a geosynchronous orbit.

Footprint

Satellites process microwaves with bidirectional antennas (line-of-sight). Therefore, the signal from a satellite is normally aimed at a specific area called the footprint. The signal power at the center of the footprint is maximum. The power decreases as we move out from the footprint center. The boundary of the footprint is the location where the power level is at a predefined threshold.

Three Categories of Satellites

Based on the location of the orbit, satellites can be divided into three categOlies: geostationary Earth orbit (GEO), low-Earth-orbit (LEO), and middle-Earth-orbit (MEO). Figure 16.14 shows the taxonomy.

Figure 16.14 *Satellite categories*

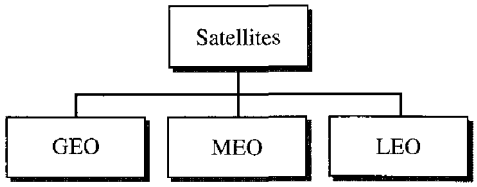
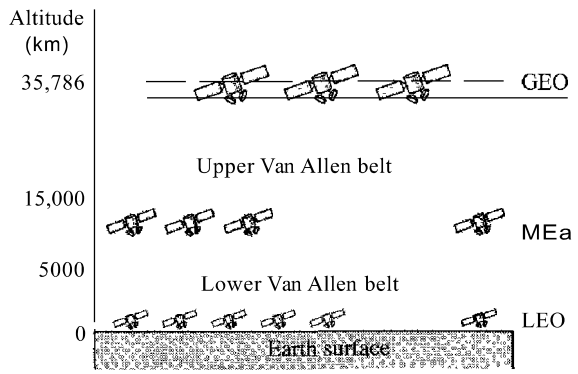


Figure 16.15 shows the satellite altitudes with respect to the surface of the Earth. There is only one orbit, at an altitude of 35,786 kIn for the OEO satellite. MEO satellites are located at altitudes between 5000 and 15,000 kIn. LEO satellites are normally below an altitude of 2000 km.

Figure 16.15 *Satellite orbit altitudes*



One reason for having different orbits is due to the existence of two Van Allen belts. A Van Allen belt is a layer that contains charged particles. A satellite orbiting in one of these two belts would be totally destroyed by the energetic charged particles. The MEO orbits are located between these two belts.

Frequency Bands for Satellite Communication

The frequencies reserved for satellite microwave communication are in the gigahertz (OHz) range. Each satellite sends and receives over two different bands. Transmission from the Earth to the satellite is called the uplink. Transmission from the satellite to the Earth is called the downlink. Table 16.1 gives the band names and frequencies for each range.

Table 16.1 *Satellite frequency bands*

Band	*Downlink, GHz*	*Uplink, GHz*	*Bandwidth, MHz*
L	1.5	1.6	15
S	1.9	2.2	70
C	4.0	6.0	500
Ku	11.0	14.0	500
Ka	20.0	30.0	3500

GEO Satellites

Line-of-sight propagation requires that the sending and receiving antennas be locked onto each other's location at all times (one antenna must have the other in sight). For this reason, a satellite that moves faster or slower than the Earth's rotation is useful only for short periods. To ensure constant communication, the satellite must move at the same speed as the Earth so that it seems to remain fixed above a certain spot. Such satellites are called *geostationary.*

Because orbital speed is based on the distance from the planet, only one orbit can be geostationary. This orbit occurs at the equatorial plane and is approximately 22,000 mi from the surface of the Earth.

But one geostationary satellite cannot cover the whole Earth. One satellite in orbit has line-of-sight contact with a vast number of stations, but the curvature of the Earth still keeps much of the planet out of sight. It takes a minimum of three satellites equidistant from each other in geostationary Earth orbit (OEO) to provide full global transmission. Figure 16.16 shows three satellites, each 120° from another in geosynchronous orbit around the equator. The view is from the North Pole.

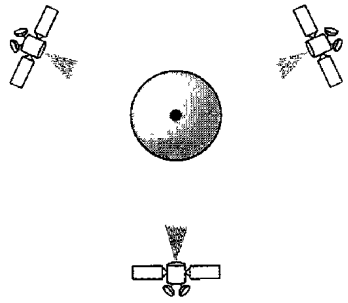
MEO Satellites

Medium-Earth-orbit (MEO) satellites are positioned between the two Van Allen belts. A satellite at this orbit takes approximately 6-8 hours to circle the Earth.

Global Positioning System

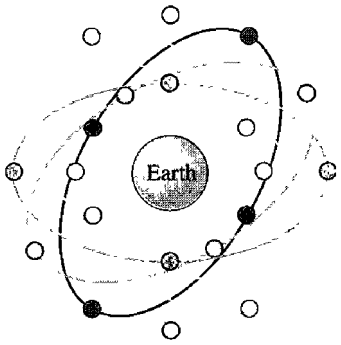
One example of a MEO satellite system is the Global Positioning System (GPS), constracted and operated by the US Department of Defense, orbiting at an altitude about

Figure 16.16 *Satellites in geostationary orbit*



18,000 km (11,000 mi) above the Earth. The system consists of 24 satellites and is used for land, sea, and air navigation to provide time and locations for vehicles and ships. GPS uses 24 satellites in six orbits, as shown in Figure 16.17. The orbits and the locations of the satellites in each orbit are designed in such a way that, at any time, four satellites are visible from any point on Earth. A GPS receiver has an almanac that tells the current position of each satellite.

Figure 16.17 *Orbits for global positioning system (GPS) satellites*

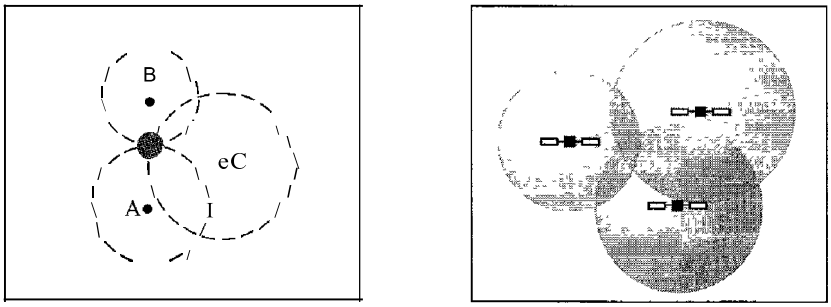


Trilateration GPS is based on a principle called trilateration.† On a plane, if we know our distance from three points, we know exactly where we are. Let us say that we are 10 miles away from point A, 12 miles away from point B, and 15 miles away from point C. If we draw three circles with the centers at A, B, and C, we must be somewhere on circle A, somewhere on circle B, and somewhere on circle C. These three circles meet at one single point (if our distances are correct), our position. Figure 16.18a shows the concept.

In three-dimensional space, the situation is different. Three spheres meet in two points as shown in Figure 16.18b. We need at least four spheres to find our exact position in space (longitude, latitude, and altitude). However, if we have additional facts about our location (for example, we know that we are not inside the ocean or somewhere in

†The terms *trilateration* and *triangulation* are normally used interchangeably. We use the word *trilateration,* which means using three distances, instead of triangulation, which may mean using three angles.

Figure 16.18 *Trilateration on a plane*



a. Two-dimensional trilateration

b. Three-dimensional trilateration

space), three spheres are enough, because one of the two points, where the spheres meet, is so improbable that the other can be selected without a doubt.

Measuring the Distance The trilateration principle can find our location on the earth if we know our distance from three satellites and know the position of each satellite. The position of each satellite can be calculated by a GPS receiver (using the predetermined path of the satellites). The GPS receiver, then, needs to find its distance from at least three GPS satellites (center of the spheres). Measuring the distance is done using a principle called one-way ranging. For the moment, let us assume that all GPS satellites and the receiver on the Earth are synchronized. Each of 24 satellites synchronously transmits a complex signal each having a unique pattern. The computer on the receiver measures the delay between the signals from the satellites and its copy of signals to determine the distances to the satellites.

Synchronization The previous discussion was based on the assumption that the satellites' clock are synchronized with each other and with the receiver's clock. Satellites use atomic clock that are precise and can function synchronously with each other. The receiver's clock however, is a normal quartz clock (an atomic clock costs more that $50,000), and there is no way to synchronize it with the satellite clocks. There is an unknown offset between the satellite clocks and the receiver clock that introduces a corresponding offset in the distance calculation. Because of this offset, the measured distance is called a *pseudorange.*

GPS uses an elegant solution to the clock offset problem, by recognizing that the offset's value is the same for all satellite being used. The calculation of position becomes finding four unknowns: the x_r, YP zp coordinates of the receiver, and common clock offset *dt.* For finding these four unknown values, we need at least four equations. This means that we need to measure pesudoranges from four satellite instead of three. If we call the four measured pseudoranges PRI, PR2, PR3 and PR4 and the coordinates of each satellite Xi, yj, and Zj (for i =1 to 4), we can find the four previously mentioned unknown values using the following four equations (the four unknown values are shown in color).

$$PR_I = [(Xl - x_r)2 + (YI - Yr)2 + (zl - zr)zJIIZ + c\,X\,dt$$
$$PR_Z = [(xz - x_r)2 + (yz - Yr)z + (zz - zr)2JI/Z + ex\,dt$$
$$PR_3 = [(x3 - xr)z + (Y3 - Yr)z + (z3 - zr)ZJ1/2 + ex\,dt$$
$$PR_4 = [(x4 - x_r)^2 + (Y4 - Yr)z + (z4 - zr)ZJI/Z + ex\,dt$$

The coordinates used in the above formulas are in an Earth-Centered Earth-Fixed (ECEF) reference frame, which means that the origin of the coordinate space is at the center of the Earth and the coordinate space rotate with the Earth. This implies that the ECEF coordinates of a fixed point on the surface of the earth do not change.

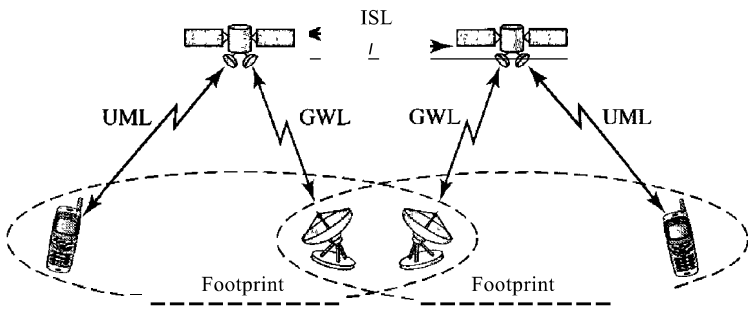
Application GPS is used by military forces. For example, thousands of portable GPS receivers were used during the Persian Gulf war by foot soldiers, vehicles, and helicopters. Another use of GPS is in navigation. The driver of a car can find the location of the car. The driver can then consult a database in the memory of the automobile to be directed to the destination. **In** other words, GPS gives the location of the car, and the database uses this information to find a path to the destination. A very interesting application is clock synchronization. As we mentioned previously, the IS-95 cellular telephone system uses GPS to create time synchronization between the base stations.

LEO Satellites

Low-Earth-orbit (LEO) satellites have polar orbits. The altitude is between 500 and 2000 km, with a rotation period of 90 to 120 min. The satellite has a speed of 20,000 to 25,000 km/h. An LEO system usually has a cellular type of access, similar to the cellular telephone system. The footprint normally has a diameter of 8000 km. Because LEO satellites are close to Earth, the round-trip time propagation delay is normally less than 20 ms, which is acceptable for audio communication.

An LEO system is made of a constellation of satellites that work together as a network; each satellite acts as a switch. Satellites that are close to each other are connected through intersatellite links (ISLs). A mobile system communicates with the satellite through a user mobile link (UML). A satellite can also communicate with an Earth station (gateway) through a gateway link (GWL). Figure 16.19 shows a typical LEO satellite network.

Figure 16.19 *LEO satellite system*



LEO satellites can be divided into three categories: little LEOs, big LEOs, and broadband LEOs. The little LEOs operate under 1 GHz. They are mostly used for low-data-rate messaging. The big LEOs operate between 1 and 3 GHz. Globalstar and Iridium systems are examples of big LEOs. The broadband LEOs provide communication similar to fiber-optic networks. The first broadband LEO system was Teledesic.

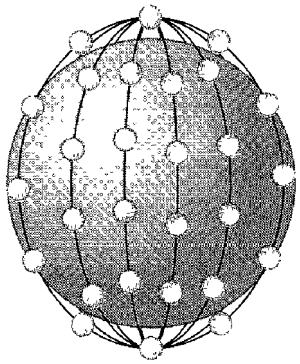
Iridium System

The concept of the Iridium system, a 77-satellite network, was started by Motorola in 1990. The project took eight years to materialize. During this period, the number of satellites was reduced. Finally, in 1998, the service was started with 66 satellites. The original name, Iridium, came from the name of the 77th chemical element; a mOre appropriate name is Dysprosium (the name of element 66).

Iridium has gone through rough times. The system was halted in 1999 due to financial problems; it was sold and restarted in 2001 under new ownership.

The system has 66 satellites divided into six orbits, with 11 satellites in each orbit. The orbits are at an altitude of 750 km. The satellites in each orbit are separated from one another by approximately 32° of latitude. Figure 16.20 shows a schematic diagram of the constellation.

Figure 16.20 *Iridium constellation*



The Iridium system has 66 satellites in six LEO orbits, each at an altitude of 750 kIn.

Since each satellite has 48 spot beams, the system can have up to 3168 beams. However, some of the beams are turned off as the satellite approaches the pole. The number of active spot beams at any moment is approximately 2000. Each spot beam covers a cell on Earth, which means that Earth is divided into approximately 2000 (overlapping) cells.

In the Iridium system, communication between two users takes place through satellites. When a user calls another user, the call can go through several satellites before reaching the destination. This means that relaying is done in space and each satellite needs to be sophisticated enough to do relaying. This strategy eliminates the need for many terrestrial stations.

The whole purpose of Iridium is to provide direct worldwide communication using handheld terminals (same concept as cellular telephony). The system can be used for voice, data, paging, fax, and even navigation. The system can provide connectivity between users at locations where other types of communication are not possible. The system provides 2.4- to 4.8-kbps voice and data transmission between portable telephones. Transmission occurs in the 1.616- to 1.6126-GHz frequency band. Intersatellite communication occurs in the 23.18- to 23.38-GHz frequency band.

Iridium is designed to provide direct worldwide voice and data communication using handheld terminals, a service similar to cellular telephony but on a global scale.

Globalstar

Globalstar is another LEO satellite system. The system uses 48 satellites in six polar orbits with each orbit hosting eight satellites. The orbits are located at an altitude of almost 1400 lan.

The Globalstar system is similar to the Iridium system; the main difference is the relaying mechanism. Communication between two distant users in the Iridium system requires relaying between several satellites; Globalstar communication requires both satellites and Earth stations, which means that ground stations can create more powerful signals.

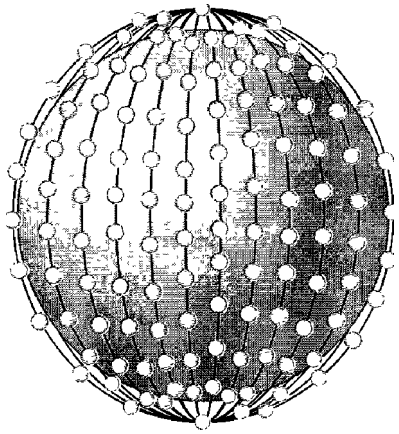
Teledesic

Teledesic is a system of satellites that provides fiber-optic-like (broadband channels, low error rate, and low delay) communication. Its main purpose is to provide broadband Internet access for users allover the world. It is sometimes called "Internet in the sky."

The project was started in 1990 by Craig McCaw and Bill Gates; later, other investors joined the consortium. The project is scheduled to be fully functional in the near future.

Constellation Teledesic provides 288 satellites in 12 polar orbits with each orbit hosting 24 satellites. The orbits are at an altitude of 1350 lan, as shown in Figure 16.21.

Figure 16.21 *Teledesic*



Teledesic has 288 satellites in 12 LEO orbits, each at an altitude of 1350 kln.

Communication The system provides three types of communication. Intersatellite communication allows eight neighboring satellites to communicate with one another. Communication is also possible between a' satellite and an Earth gateway station. Users can communicate directly with the network using terminals. Earth is divided into tens of thousands of cells. Each cell is assigned a time slot, and the satellite focuses its beam to the cell

at the corresponding time slot. The tenninal can send data during its time slot. A tenninal receives all packets intended for the cell, but selects only those intended for its address.

Bands Transmission occurs in the Ka bands.

Data Rate The data rate is up to 155 Mbps for the uplink and up to 1.2 Gbps for the downlink.

16.3 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [...] refer to the reference list at the end of the text.

Books

Wireless WANs are completely covered in [Sta02], [Jam03], [AZ03], and [Sch03]. Communication satellites are discussed in Section 2.4 of [Tan03] and Section 8.5 of [CouOl]. Mobile telephone system is discussed in Section 2.6 of [Tan03] and Section 8.8 of [CouOl].

16.4 KEY TERMS

Advanced Mobile Phone System (AMPS)
cellular telephony
digital AMPS (D-AMPS)
downlink
footprint
geostationary Earth orbit (GEO)
Global Positioning System (GPS)
Global System for Mobile Communication (GSM)
Globalstar
handoff
Interim Standard 95 (IS-95)
Internet Mobile Communication 2000 (IMT-2000)
Iridium
low-Earth-orbit (LEO)
medium-Earth-orbit (MEO)
mobile switching center (MSC)
orbit
personal communications system (PCS)
reuse factor
roamIng
satellite network
Teledesic
triangulation
trilateration
uplink

16.5 SUMMARY

- Cellular telephony provides communication between two devices. One or both may be mobile.
- A cellular service area is divided into cells.
- Advanced Mobile Phone System (AMPS) is a first-generation cellular phone system.

- Digital AMPS CD-AMPS) is a second-generation cellular phone system that is a digital version of AMPS.
- Global System for Mobile Communication CGSM) is a second-generation cellular phone system used in Europe.
- Interim Standard 95 CIS-95) is a second-generation cellular phone system based on CDMA and DSSS.
- The third-generation cellular phone system will provide universal personal communication.
- A satellite network uses satellites to provide communication between any points on Earth.
- A geostationary Earth orbit CGEO) is at the equatorial plane and revolves in phase with Earth.
- Global Positioning System CGPS) satellites are medium-Earth-orbit (MEO) satellites that provide time and location information for vehicles and ships.
- Iridium satellites are low-Earth-orbit (LEO) satellites that provide direct universal voice and data communications for handheld terminals.
- Teledesic satellites are low-Earth-orbit satellites that will provide universal broadband Internet access.

16.6 PRACTICE SET

Review Questions

1. What is the relationship between a base station and a mobile switching center?
2. What are the functions of a mobile switching center?
3. Which is better, a low reuse factor or a high reuse factor? Explain your answer.
4. What is the difference between a hard handoff and a soft handoff?
5. What is AMPS?
6. What is the relationship between D-AMPS and AMPS?
7. What is GSM?
8. What is the function of the CDMA in IS-95?
9. What are the three types of orbits?
10. Which type of orbit does a GEO satellite have? Explain your answer.
11. What is a footprint?
12. What is the relationship between the Van Allen belts and satellites?
13. Compare an uplink with a downlink.
14. What is the purpose of GPS?
15. What is the main difference between Iridium and Globalstar?

Exercises

16. Draw a cell pattern with a frequency-reuse factor of 3.
17. What is the maximum number of callers in each cell in AMPS?

18. What is the maximum number of simultaneous calls in each cell in an 15-136 (D-AMPS) system, assuming no analog control channels?
19. What is the maximum number of simultaneous calls in each cell in a GSM assuming no analog control channels?
20. What is the maximum number of caners in each cell in an IS-95 system?
21. Find the efficiency of AMPS in terms of simultaneous calls per megahertz of bandwidth. In other words, find the number of calls that can be used in I-MHz bandwidth allocation.
22. Repeat Exercise 21 for D-AMPS.
23. Repeat Exercise 21 for GSM.
24. Repeat Exercise 21 for IS-95.
25. Guess the relationship between a 3-kHz voice channel and a 30-kHz modulated channel in a system using AMPS.
26. How many slots are sent each second in a channel using D-AMPS? How many slots are sent by each user in 1 s?
27. Use Kepler's formula to check the accuracy of a given period and altitude for a GPS satellite.
28. Use Kepler's formula to check the accuracy of a given period and altitude for an Iridium satellite.
29. Use Kepler's formula to check the accuracy of a given period and altitude for a Globalstar satellite.

CHAPTER 17

SONETISDH

In this chapter, we introduce a wide area network (WAN), SONET, that is used as a transport network to carry loads from other WANs. We first discuss SONET as a protocol, and we then show how SONET networks can be constructed from the standards defined in the protocol.

The high bandwidths of fiber-optic cable are suitable for today's high-data-rate technologies (such as video conferencing) and for carrying large numbers of lower-rate technologies at the same time. For this reason, the importance of fiber optics grows in conjunction with the development of technologies requiring high data rates or wide bandwidths for transmission. With their prominence came a need for standardization. The United States (ANSI) and Europe (ITU-T) have responded by defining standards that, though independent, are fundamentally similar and ultimately compatible. The ANSI standard is called the Synchronous Optical Network (SONET). The ITU-T standard is called the Synchronous Digital Hierarchy (SOH).

SONET was developed by ANSI; SDH was developed by lTV·T.

SONET/SDH is a synchronous network using synchronous TDM multiplexing. All clocks in the system are locked to a master clock.

17.1 ARCHITECTURE

Let us first introduce the architecture of a SONET system: signals, devices, and connections.

Signals

SONET defines a hierarchy of electrical signaling levels called synchronous transport signals (STSs). Each STS level (STS-l to STS-192) supports a certain data rate, specified in megabits per second (see Table 17.1). The corresponding optical signals are called optical carriers (OCs). SDH specifies a similar system called a synchronous transport module (STM). STM is intended to be compatible with existing European

hierarchies, such as E lines, and with STS levels. To this end, the lowest STM level, STM-l, is defined as 155.520 Mbps, which is exactly equal to STS-3.

Table 17.1 *SONETISDH rates*

STS	OC	*Rate (Mbps)*	*STM*
STS-l	OC-l	51.840	
STS-3	OC-3	155.520	STM-l
STS-9	OC-9	466.560	STM-3
STS-12	OC-12	622.080	STM-4
STS-18	OC-lS	933.120	STM-6
STS-24	OC-24	1244.160	STM-8
STS-36	OC-36	1866.230	STM-12
STS-48	OC-48	2488.320	STM-16
STS-96	OC-96	4976.640	STM-32
STS-I92	OC-I92	9953.280	STM-64

A glance through Table 17.1 reveals some interesting points. First, the lowest level in this hierarchy has a data rate of 51.840 Mbps, which is greater than that of the DS-3 service (44.736 Mbps). In fact, the STS-l is designed to accommodate data rates equivalent to those of the DS-3. The difference in capacity is provided to handle the overhead needs of the optical system.

Second, the STS-3 rate is exactly three times the STS-l rate; and the STS-9 rate is exactly one-half the STS-18 rate. These relationships mean that 18 STS-l channels can be multiplexed into one STS-18, six STS-3 channels can be multiplexed into one STS-18, and so on.

SONET Devices

Figure 17.1 shows a simple link using SONET devices. SONET transmission relies on three basic devices: STS multiplexers/demultiplexers, regenerators, add/drop multiplexers and terminals.

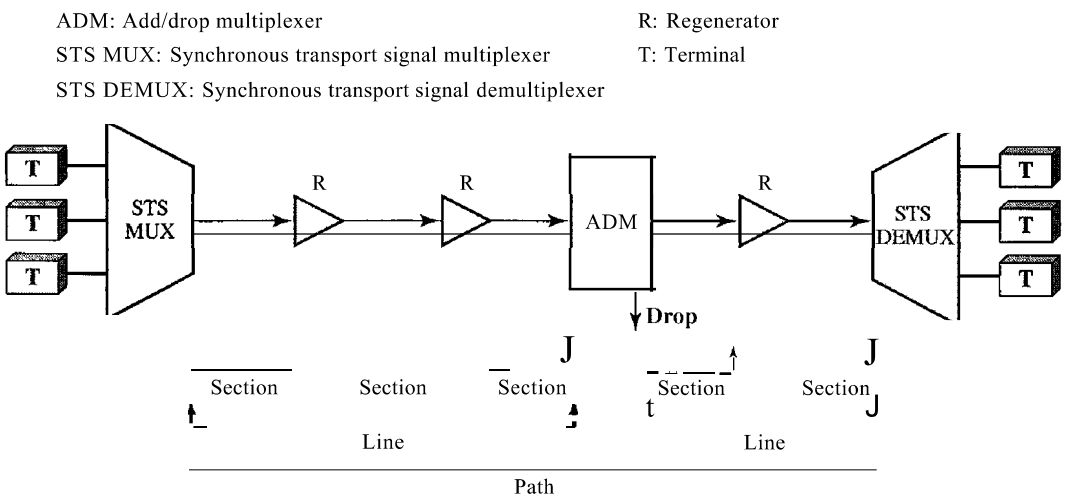
STS Multiplexer/Demultiplexer

STS multiplexers/demultiplexers mark the beginning points and endpoints of a SONET link. They provide the interface between an electrical tributary network and the optical network. An STS multiplexer multiplexes signals from multiple electrical sources and creates the corresponding OC signal. An STS demultiplexer demultiplexes an optical OC signal into corresponding electric signals.

Regenerator

Regenerators extend the length of the links. A regenerator is a repeater (see Chapter 15) that takes a received optical signal *(OC-n),* demodulates it into the corresponding electric signal *(STS-n),* regenerates the electric signal, and finally modulates the electric

Figure 17.1 *A simple network using SONET equipment*



signal into its correspondent *OC-n* signal. A SONET regenerator replaces some of the existing overhead information (header information) with new information.

Add/drop Multiplexer

Add/drop multiplexers allow insertion and extraction of signals. An **add/drop multiplexer (ADM)** can add STSs coming from different sources into a given path or can remove a desired signal from a path and redirect it without demultiplexing the entire signal. Instead of relying on timing and bit positions, add/drop multiplexers use header information such as addresses and pointers (described later in this section) to identify individual streams.

In the simple configuration shown by Figure 17.1, a number of incoming electronic signals are fed into an STS multiplexer, where they are combined into a single optical signal. The optical signal is transmitted to a regenerator, where it is recreated without the noise it has picked up in transit. The regenerated signals from a number of sources are then fed into an add/drop multiplexer. The add/drop multiplexer reorganizes these signals, if necessary, and sends them out as directed by information in the data frames. These remultiplexed signals are sent to another regenerator and from there to the receiving STS demultiplexer, where they are returned to a format usable by the receiving links.

Terminals

A **terminal** is a device that uses the services of a SONET network. For example, in the Internet, a terminal can be a router that needs to send packets to another router at the other side of a SONET network.

Connections

The devices defined in the previous section are connected using *sections, lines,* and *paths.*

Sections

A section is the optical link connecting two neighbor devices: multiplexer to multiplexer, multiplexer to regenerator, or regenerator to regenerator.

Lines

A line is the portion of the network between two multiplexers: STS multiplexer to add/drop multiplexer, two add/drop multiplexers, or two STS multiplexers.

Paths

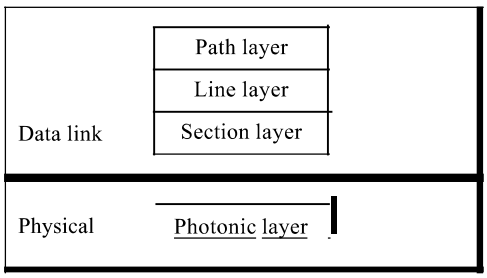
A path is the end-to-end portion of the network between two STS multiplexers. In a simple SONET of two STS multiplexers linked directly to each other, the section, line, and path are the same.

17.2 SONET LAYERS

The SONET standard includes four functional layers: the photonic, the section, the line, and the path layer. They correspond to both the physical and the data link layers (see Figure 17.2). The headers added to the frame at the various layers are discussed later in this chapter.

SONET defines four layers: path, line, section, and photonic.

Figure 17.2 *SONET layers compared with OSI or the Internet layers*



Path Layer

The path layer is responsible for the movement of a signal from its optical source to its optical destination. At the optical source, the signal is changed from an electronic form into an optical form, multiplexed with other signals, and encapsulated in a frame. At the optical destination, the received frame is demultiplexed, and the individual optical signals are changed back into their electronic forms. Path layer overhead is added at this layer. STS multiplexers provide path layer functions.

Line Layer

The **line layer** is responsible for the movement of a signal across a physical line. Line layer overhead is added to the frame at this layer. STS multiplexers and add/drop multiplexers provide line layer functions.

Section Layer

The **section layer** is responsible for the movement of a signal across a physical section. It handles framing, scrambling, and error control. Section layer overhead is added to the frame at this layer.

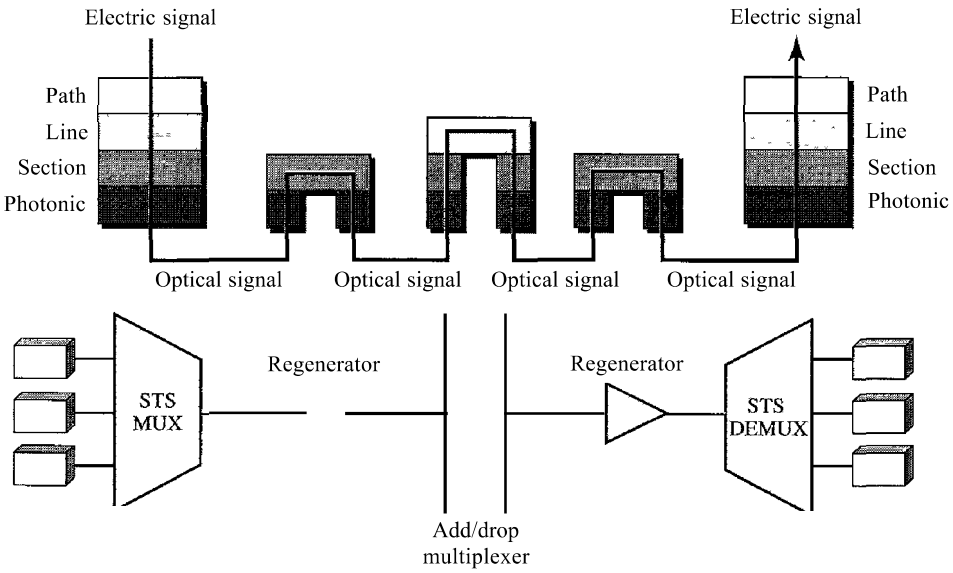
Photonic Layer

The **photonic layer** corresponds to the physical layer of the OSI model. It includes physical specifications for the optical fiber channel, the sensitivity of the receiver, multiplexing functions, and so on. SONET uses NRZ encoding with the presence of light representing 1 and the absence of light representing 0.

Device-Layer Relationships

Figure 17.3 shows the relationship between the devices used in SONET transmission and the four layers of the standard. As you can see, an STS multiplexer is a four-layer device. An add/drop multiplexer is a three-layer device. A regenerator is a two-layer device.

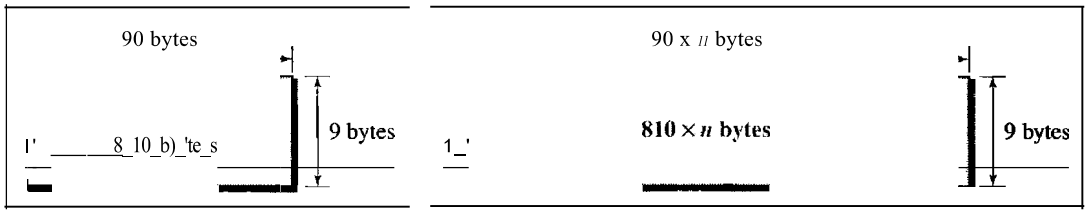
Figure 17.3 *Device-layer relationship in SONET*



17.3 SONET FRAMES

Each synchronous transfer signal *STS-n* is composed of 8000 frames. Each frame is a two-dimensional matrix of bytes with 9 rows by 90 x *n* columns. For example, STS-1 frame is 9 rows by 90 columns (810 bytes), and an STS-3 is 9 rows by 270 columns (2430 bytes). Figure 17.4 shows the general format of an STS-1 and an *STS-n*.

Figure 17.4 *An STS-1 and an STS-nframe*

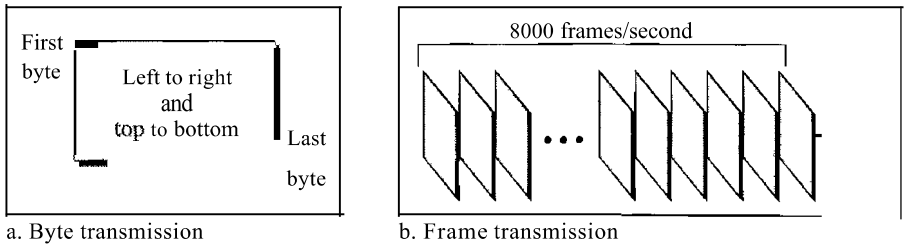


a. STS-1 frame b. STS-n frame

Frame, Byte, and Bit Transmission

One of the interesting points about SONET is that each *STS-n* signal is transmitted at a fixed rate of 8000 frames per second. This is the rate at which voice is digitized (see Chapter 4). For each frame the bytes are transmitted from the left to the right, top to the bottom. For each byte, the bits are transmitted from the most significant to the least significant (left to right). Figure 17.5 shows the order of frame and byte transmission.

Figure 17.5 *STS-1 frames in transition*



a. Byte transmission b. Frame transmission

A SONET *STS-n* signal is transmitted at 8000 frames per second.

If we sample a voice signal and use 8 bits (1 byte) for each sample, we can say that each byte in a SONET frame can carry information from a digitized voice channel. In other words, an STS-1 signal can carry 774 voice channels simultaneously (810 minus required bytes for overhead).

Each byte in a SONET frame can carry a digitized voice channel.

Example 17.1

Find the data rate of an STS-1 signal.

Solution

STS-1, like other STS signals, sends 8000 frames per second. Each STS-1 frame is made of 9 by (1 X 90) bytes. Each byte is made of 8 bits. The data rate is

$$\text{STS-1 data rate} = 8000 \times 9 \times (1 \times 90) \times 8 = 51.840 \text{ Mbps}$$

Example 17.2

Find the data rate of an STS-3 signal.

Solution

STS-3, like other STS signals, sends 8000 frames per second. Each STS-3 frame is made of 9 by (3 X 90) bytes. Each byte is made of 8 bits. The data rate is

$$\text{STS-3 data rate} = 8000 \times 9 \times (3 \times 90) \times 8 = 155.52 \text{ Mbps}$$

Note that in SONET, there is an exact relationship between the data rates of different STS signals. We could have found the data rate of STS-3 by using the data rate of STS-1 (multiply the latter by 3).

In SONET, the data rate of an *STS-n* signal is *n* times the data rate of an STS-1 signals.

Example 17.3

What is the duration of an STS-1 frame? STS-3 frame? *STS-n* frame?

Solution

In SONET, 8000 frames are sent per second. This means that the duration of an STS-1, STS-3, or *STS-n* frame is the same and equal to 1/8000 s, or 125 μs.

In SONET, the duration of any frame is 125 μs.

STS-1 Frame Format

The basic format of an STS-1 frame is shown in Figure 17.6. As we said before, a SONET frame is a matrix of 9 rows of 90 bytes (octets) each, for a total of 810 bytes.

The first three columns of the frame are used for section and line overhead. The upper three rows of the first three columns are used for section overhead (SOH). The lower six are line overhead (LOH). The rest of the frame is called the synchronous payload envelope (SPE). It contains user data and path overhead (POH) needed at the user data level. We will discuss the format of the SPE shortly.

Section Overhead

The section overhead consists of nine octets. The labels, functions, and organization of these octets are shown in Figure 17.7.

Figure 17.6 *STS-1 frame overheads*

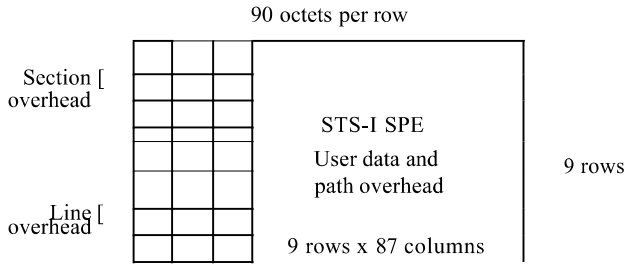
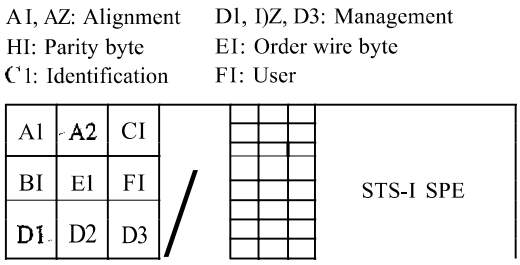


Figure 17.7 *STS-1 frame: section overhead*



- Alignment bytes (AI and A2). Bytes A1 and A2 are used for framing and synchronization and are called alignment bytes. These bytes alert a receiver that a frame is arriving and give the receiver a predetermined bit pattern on which to synchronize. The bit patterns for these two bytes in hexadecimal are OxF628. The bytes serve as a flag.
- Section parity byte (BI). Byte B1 is for bit interleaved parity (BIP-8). Its value is calculated over all bytes of the previous frame. In other words, the ith bit of this byte is the parity bit calculated over all ith bits of the previous *STS-n* frame. The value of this byte is filled only for the first STS-l in an *STS-n* frame. In other words, although an *STS-n* frame has *n* B1 bytes, as we will see later, only the first byte has this value; the rest are filled with Os.
- Identification byte (e1). Byte C1 carries the identity of the STS-l frame. This byte is necessary when multiple STS-ls are multiplexed to create a higher-rate STS (STS-3, STS-9, STS-12, etc.). Information in this byte allows the various signals to be recognized easily upon demultiplexing. For example, in an STS-3 signal, the value of the C1 byte is 1 for the first STS-l; it is 2 for the second; and it is 3 for the third.
- Management bytes (DI, D2, and D3). Bytes D1, D2, and D3 together form a 192-kbps channel (3 x 8000 x 8) called the data communication channel. This channel is required for operation, administration, and maintenance (OA&M) signaling.
- Order wire byte (EI). Byte E1 is the order wire byte. Order wire bytes in consecutive frames form a channel of 64 kbps (8000 frames per second times 8 bits per

frame). This channel is used for communication between regenerators, or between terminals and regenerators.

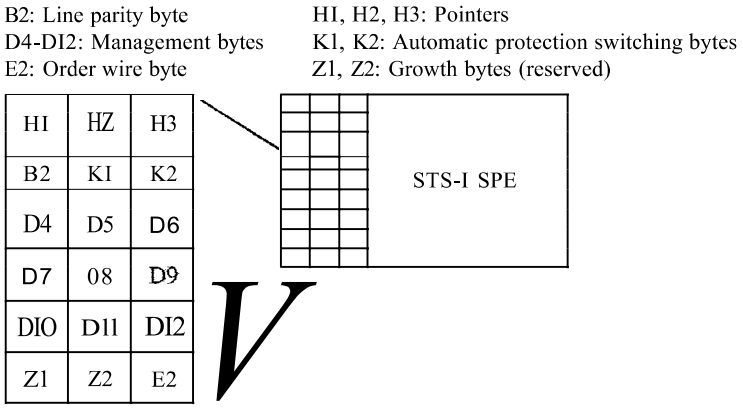
- User's byte (F1). The FI bytes in consecutive frames form a 64-kbps channel that is reserved for user needs at the section level.

Section overhead is recalculated for each SONET device (regenerators and multiplexers).

Line Overhead

Line overhead consists of 18 bytes. The labels, functions, and arrangement of these bytes are shown in Figure 17.8.

Figure 17.8 *STS-l frame: line overhead*



- Line parity byte (B2). Byte B2 is for bit interleaved parity. It is for error checking of the frame over a line (between two multiplexers). In an *STS-n* frame, B2 is calculated for all bytes in the previous STS-I frame and inserted at the B2 byte for that frame. In other words, in a STS-3 frame, there are three B2 bytes, each calculated for one STS-I frame. Contrast this byte with BI in the section overhead.
- Data communication channel bytes (D4 to D12). The line overhead D bytes (D4 to D12) in consecutive frames form a 576-kbps channel that provides the same service as the DI-D3 bytes (OA&M), but at the line rather than the section level (between multiplexers).
- Order wire byte (E2). The E2 bytes in consecutive frames form a 64-kbps channel that provides the same functions as the EI order wire byte, but at the line level.
- Pointer bytes (HI, 82, and 83). Bytes HI, H2, and H3 are pointers. The first two bytes are used to show the offset of the SPE in the frame; the third is used for justification. We show the use of these bytes later.
- Automatic protection switching bytes (K1 and K2). The KI and K2 bytes in consecutive frames form a I28-kbps channel used for automatic detection of problems in

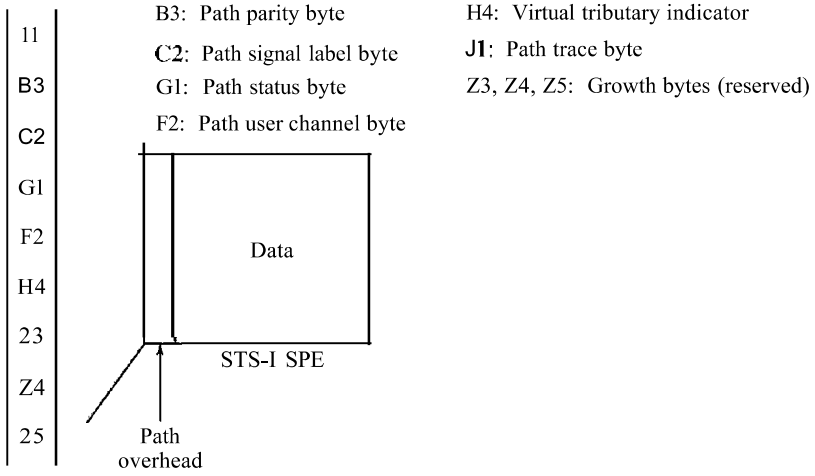
line-terminating equipment. We discuss automatic protection switching (APS) later in the chapter.

- Growth bytes (Z1 and Z2). The Z1 and Z2 bytes are reserved for future use.

Synchronous Payload Envelope

The synchronous payload envelope (SPE) contains the user data and the overhead related to the user data (path overhead). One SPE does not necessarily fit it into one STS-1 frame; it may be split between two frames, as we will see shortly. This means that the path overhead, the leftmost column of an SPE, does not necessarily align with the section or line overhead. The path overhead must be added first to the user data to create an SPE, and then an SPE can be inserted into one or two frames. Path overhead consists of 9 bytes. The labels, functions, and arrangement of these bytes are shown in Figure 17.9.

Figure 17.9 *STS-1 frame: path overhead*



- Path parity byte (B3). Byte B3 is for bit interleaved parity, like bytes B1 and B2, but calculated over SPE bits. It is actually calculated over the previous SPE in the stream.
- Path signal label byte (C2). Byte C2 is the path identification byte. It is used to identify different protocols used at higher levels (such as IP or ATM) whose data are being carried in the SPE.
- Path user channel byte (F2). The F2 bytes in consecutive frames, like the F1 bytes, form a 64-kbps channel that is reserved for user needs, but at the path level.
- Path status byte (G1). Byte G1 is sent by the receiver to communicate its status to the sender. It is sent on the reverse channel when the communication is duplex. We will see its use in the linear or ring networks later in the chapter.
- Multiframe indicator (H4). Byte H4 is the multiframe indicator. It indicates payloads that cannot fit into a single frame. For example, virtual tributaries can be

combined to form a frame that is larger than an SPE frame and need to be divided into different frames. Virtual tributaries are discussed in the next section.

- Path trace byte (JI). The 11 bytes in consecutive frames form a 64-kbps channel used for tracking the path. The 11 byte sends a continuous 64-byte string to verify the connection. The choice of the string is left to the application program. The receiver compares each pattern with the previous one to ensure nothing is wrong with the communication at the path layer.
- Growth bytes (Z3, Z4, and Z5). Bytes Z3, Z4, and Z5 are reserved for future use.

Path overhead is only calculated for end-to-end (at STS multiplexers).

Overhead Summary

Table 17.2 compares and summarizes the overheads used in a section, line, and path.

Table 17.2 *SONETISDH rates*

Byte Function	*Section*	*Line*	*Path*
Alignment	A1,A2		
Parity	B1	B2	B3
Identifier	CI		C2
OA&M	DI-D3	D4-DI2	
Order wire	EI		
User	FI		F2
Status			G1
Pointers		HI-H3	H4
Trace			11
Failure tolerance		KI,K2	
Growth (reserved for future)		ZI, Z2	Z3-Z5

Example 17.4

What is the user data rate of an STS-1 frame (without considering the overheads)?

Solution

The user data part in an STS-I frame is made of 9 rows and 86 columns. So we have

$$\text{STS-1 user data rate} = 8000 \times 9 \times (1 \times 86) \times 8 = 49.536 \text{ Mbps}$$

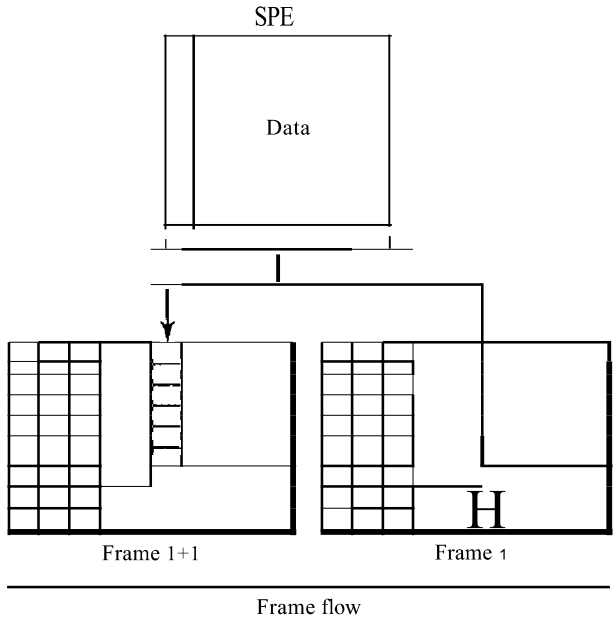
Encapsulation

The previous discussion reveals that an SPE needs to be encapsulated in an STS-I frame. Encapsulation may create two problems that are handled elegantly by SONET using pointers (HI to H3). We discuss the use of these bytes in this section.

Offsetting

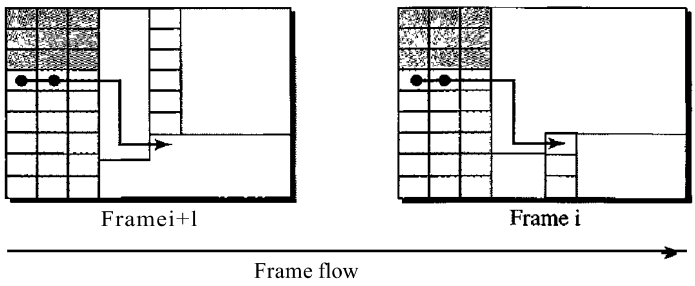
SONET allows one SPE to span two frames, part of the SPE is in the first frame and part is in the second. This may happen when one SPE that is to be encapsulated is not aligned time-wise with the passing synchronized frames. Figure 17.10 shows this situation. SPE bytes are divided between the two frames. The first set of bytes is encapsulated in the first frame; the second set is encapsulated in the second frame. The figure also shows the path overhead, which is aligned with the section/line overhead of any frame. The question is, How does the SONET multiplexer know where the SPE starts or ends in the frame? The solution is the use of pointers HI and H2 to define the beginning of the SPE; the end can be found because each SPE has a fixed number of bytes. SONET allows the offsetting of an SPE with respect to an STS-1 frame.

Figure 17.10 *Offsetting of SPE related to frame boundary*



To find the beginning of each SPE in a frame, we need two pointers HI and H2 in the line overhead. Note that these pointers are located in the line overhead because the encapsulation occurs at a multiplexer. Figure 17.11 shows how these 2 bytes point to

Figure 17.11 *The use of Hi and H2 pointers to show the start of an SPE in a frame*



the beginning of the SPEs. Note that we need 2 bytes to define the position of a byte in a frame; a frame has 810 bytes, which cannot be defined using 1 byte.

Example 17.5

What are the values of HI and H2 if an SPE starts at byte number 650?

Solution

The number 650 can be expressed in four hexadecimal digits as Ox028A. This means the value of HI is Ox02 and the value of H2 is Ox8A.

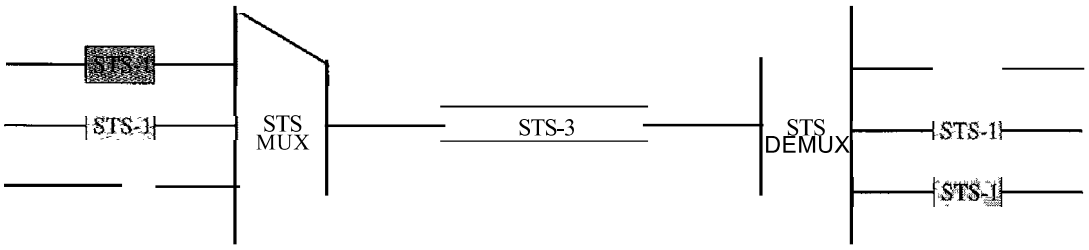
Justification

Now suppose the transmission rate of the payload is just slightly different from the transmission rate of SONET. First, assume that the rate of the payload is higher. This means that occasionally there is 1 extra byte that cannot fit in the frame. In this case, SONET allows this extra byte to be inserted in the H3 byte. Now, assume that the rate of the payload is lower. This means that occasionally 1 byte needs to be left empty in the frame. SONET allows this byte to be the byte after the H3 byte.

17.4 STS MULTIPLEXING

In SONET, frames of lower rate can be synchronously time-division multiplexed into a higher-rate frame. For example, three STS-1 signals (channels) can be combined into one STS-3 signal (channel), four STS-3s can be multiplexed into one STS-12, and so on, as shown in Figure 17.12.

Figure 17.12 *STS multiplexing/demultiplexing*



Multiplexing is synchronous TDM, and all clocks in the network are locked to a master clock to achieve synchronization.

In SONET, all clocks in the network are locked to a master clock.

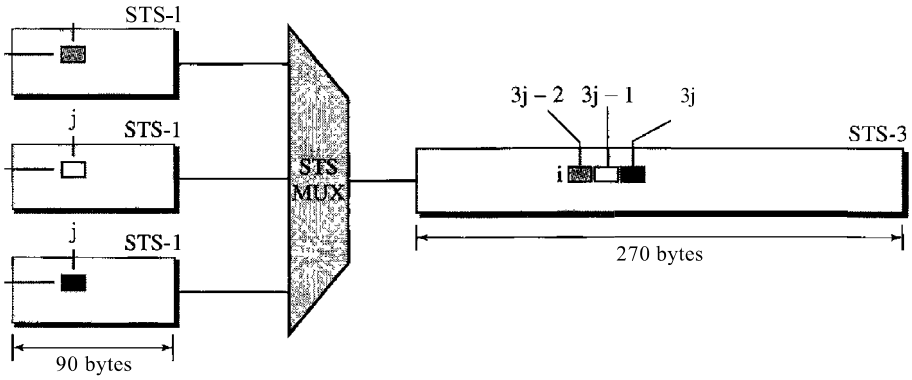
We need to mention that multiplexing can also take place at the higher data rates. For example, four STS-3 signals can be multiplexed into an STS-12 signal. However, the STS-3 signals need to first be demultiplexed into 12 STS-1 signals, and then these

twelve signals need to be multiplexed into an STS-I2 signal. The reason for this extra work will be clear after our discussion on byte interleaving.

Byte Interleaving

Synchronous TDM multiplexing in SONET is achieved by using byte interleaving. For example, when three STS-I signals are multliplexed into one STS-3 signal, each set of 3 bytes in the STS-3 signal is associated with 1 byte from each STS-I signal. Figure 17.13 shows the interleaving.

Figure 17.13 *Byte interleaving*



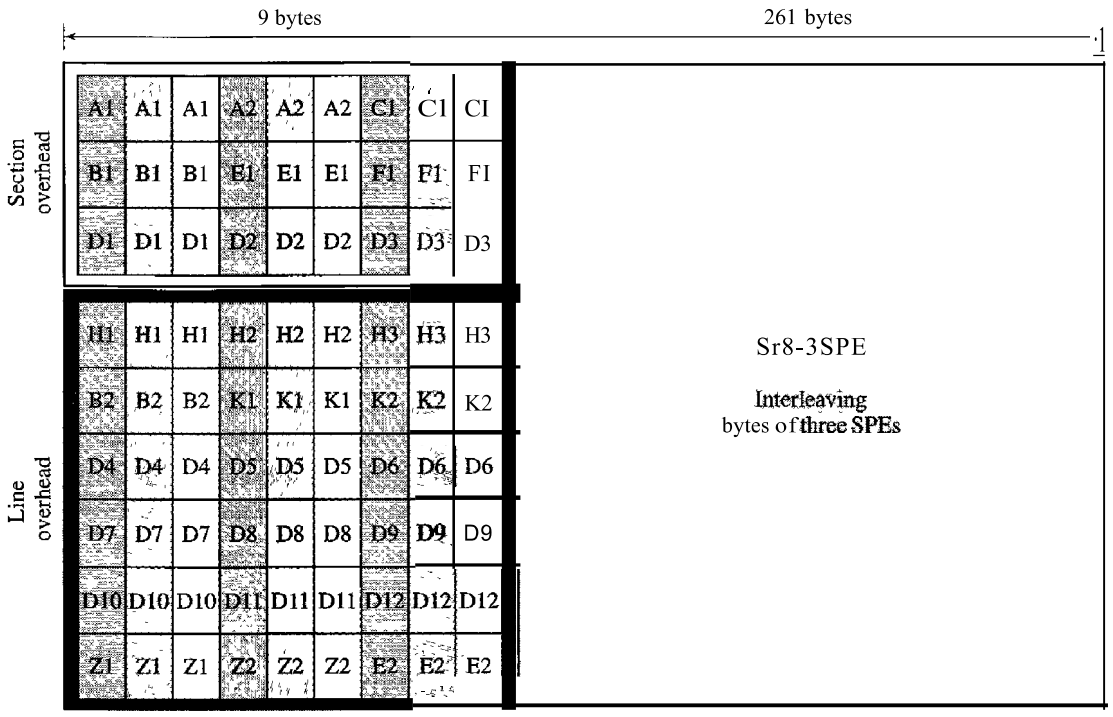
Note that a byte in an STS-I frame keeps its row position, but it is moved into a different column. The reason is that while all signal frames have the same number of rows (9), the number of columns changes. The number of columns in an *STS-n* signal frame is *n* times the number of columns in an STS-I frame. One *STS-n* row, therefore, can accommodate all *n* rows in the STS-I frames.

Byte interleaving also preserves the corresponding section and line overhead as shown in Figure 17.14. As the figure shows, the section overheads from three STS-1 frames are interleaved together to create a section overhead for an STS-1 frame. The same is true for the line overheads. Each channel, however, keeps the corresponding bytes that are used to control that channel. In other words, the sections and lines keep their own control bytes for each multiplexed channel. This interesting feature will allow the use of add/drop multiplexers, as discussed shortly. As the figure shows, there are three A1 bytes, one belonging to each of the three multiplexed signals. There are also three A2 bytes, three B 1 bytes, and so on.

Demultiplexing here is easier than in the statistical TDM we discussed in Chapter 6 because the demultiplexer, with no regard to the function of the bytes, removes the first A 1 and assigns it to the first STS-I, removes the second AI, and assigns it to second STS-1, and removes the third A1 and assigns it to the third STS-I. In other words, the demultiplexer deals only with the position of the byte, not its function.

What we said about the section and line overheads does not exactly apply to the path overhead. This is because the path overhead is part of the SPE that may have splitted into two STS-I frames. The byte interleaving, however, is the same for the data section of SPEs.

Figure 17.14 *An STS-3 frame*



The byte interleaving process makes the multiplexing at higher data rates a little bit more complex. How can we multiplex four STS-3 signals into one STS-12 signal? This can be done in two steps: First, the STS-3 signals must be demultiplexed to create 12 STS-1 signals. The 12 STS-1 signals are then multiplexed to create an STS-12 signal.

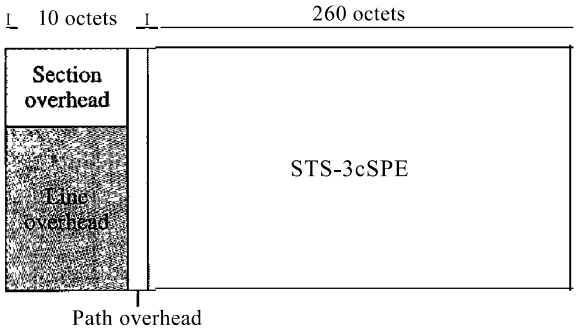
Concatenated Signal

In normal operation of the SONET, an *STS-n* signal is made of *n* multiplexed STS-1 signals. Sometimes, we have a signal with a data rate higher than what an STS-1 can carry. In this case, SONET allows us to create an *STS-n* signal which is not considered as *n* STS-1 signals; it is one STS-*n* signal (channel) that cannot be demultiplexed into *n* STS-1 signals. To specify that the signal cannot be demultiplexed, the suffix c (for concatenated) is added to the name of the signal. For example, STS-3c is a signal that cannot be demultiplexed into three STS-1 signals. However, we need to know that the whole payload in an STS-3c signal is one SPE, which means that we have only one column (9 bytes) of path overhead. The used data in this case occupy 260 columns, as shown in Figure 17.15.

Concatenated Signals Carrying ATM Cells

We will discuss ATM and ATM cells in Chapter 18. An ATM network is a cell network in which each cell has a fixed size of 53 bytes. The SPE of an STS-3c signal can be a carrier of ATM cells. The SPE of an STS-3c can carry 9 x 260 = 2340 bytes, which can accommodate approximately 44 ATM cells, each of 53 bytes.

Figure 17.15 *A concatenated STS-3c signal*

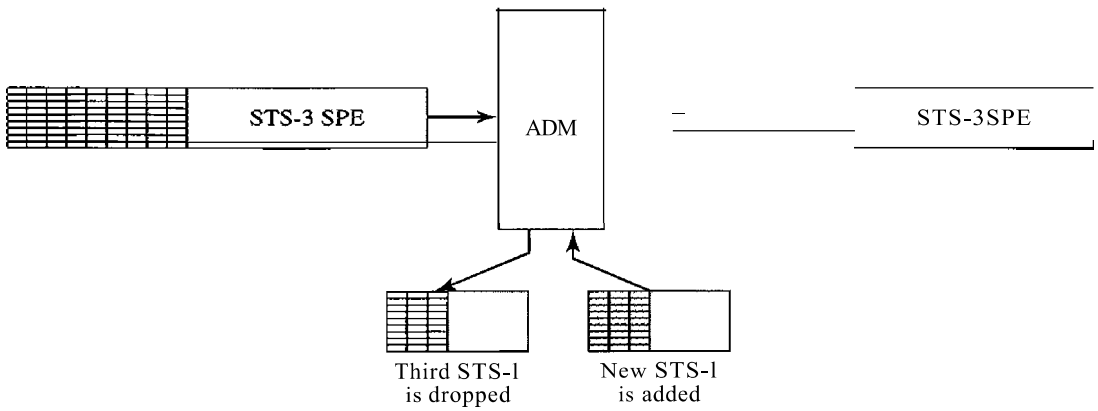


An STS-3c signal can carry 44 ATM cells as its SPE.

AddlDrop Multiplexer

Multiplexing of several STS-1 signals into an *STS-n* signal is done at the STS multiplexer (at the path layer). Demultiplexing of an *STS-n* signal into STS-1 components is done at the STS demultiplexer. In between, however, SONET uses add/drop multiplexers that can replace a signal with another one. We need to know that this is not demultiplexing/multiplexing in the conventional sense. An add/drop multiplexer operates at the line layer. An add/drop multiplexer does not create section, line, or path overhead. It almost acts as a switch; it removes one STS-1 signal and adds another one. The type of signal at the input and output of an add/drop multiplexer is the same (both *STS-3* or both STS-12, for example). The add/drop multiplexer (ADM) only removes the corresponding bytes and replaces them with the new bytes (including the bytes in the section and line overhead). Figure 17.16 shows the operation of an ADM.

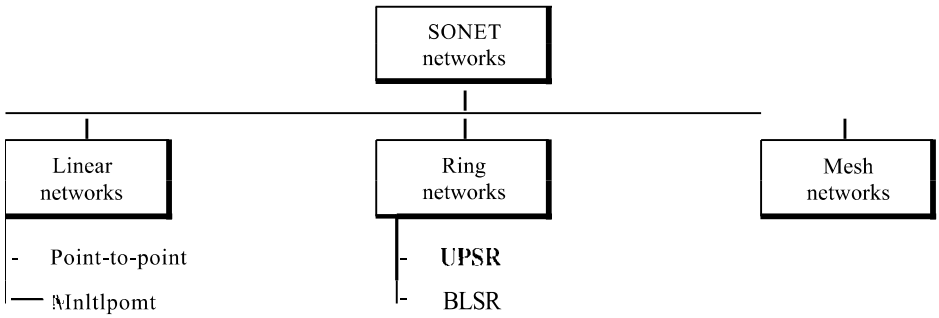
Figure 17.16 *Dropping and adding STS-l frames in an add/drop multiplexer*



17.5 SONET NETWORKS

Using SONET equipment, we can create a SONET network that can be used as a high-speed backbone carrying loads from other networks such as ATM (Chapter 18) or IF (Chapter 20). We can roughly divide SONET networks into three categories: linear, ring, and mesh networks, as shown in Figure 17.17.

Figure 17.17 *Taxonomy of SONET networks*



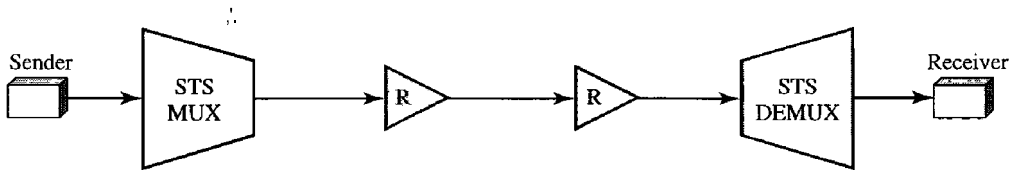
Linear Networks

A linear SONET network can be point-to-point or multipoint.

Point-to-Point Network

A point-to-point network is normally made of an STS multiplexer, an STS demultiplexer, and zero or more regenerators with no add/drop multiplexers, as shown in Figure 17.18. The signal flow can be unidirectional or bidirectional, although Figure 17.18 shows only unidirectional for simplicity.

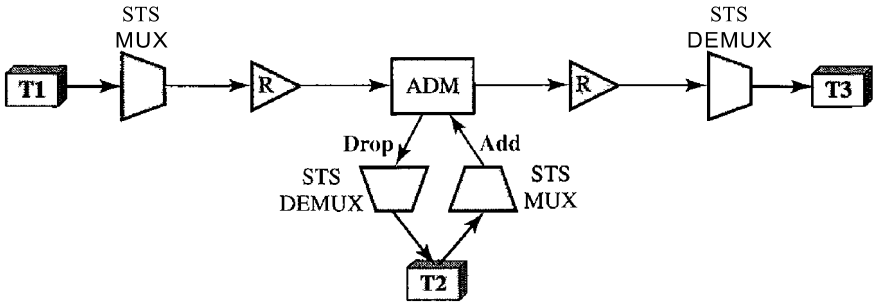
Figure 17.18 *A point-to-point SONET network*



Multipoint Network

A multipoint network uses ADMs to allow the communications between several terminals. An ADM removes the signal belonging to the terminal connected to it and adds the signal transmitted from another terminal. Each terminal can send data to one or more downstream terminals. Figure 17.19 shows a unidirectional scheme in which each terminal can send data only to the downstream terminals, but the a multipoint network can be bidirectional, too.

Figure 17.19 *A multipoint SONET network*

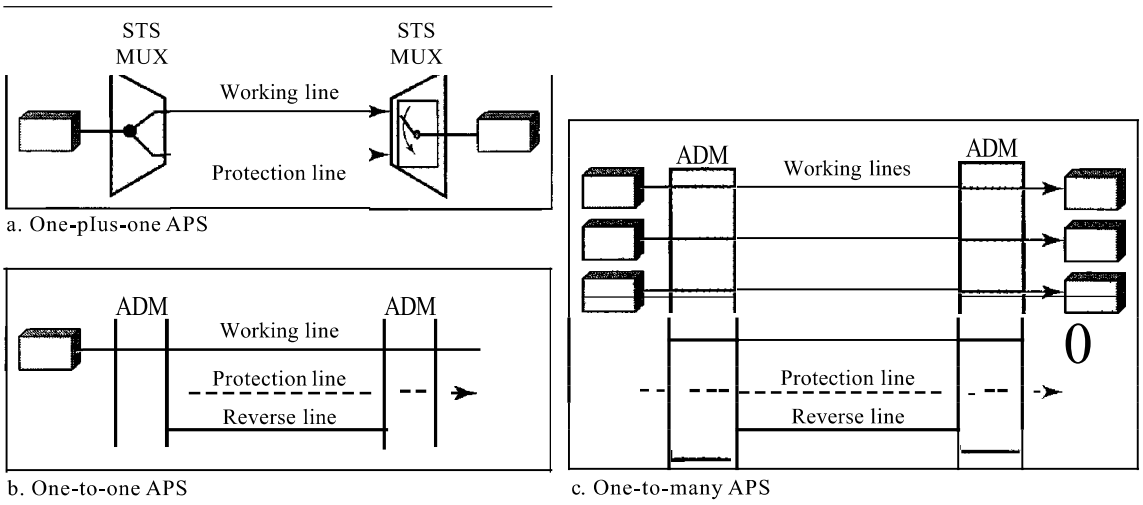


In Figure 17.19, T1 can send data to T2 and T3 simultaneously. T2, however, can send data only to T3. The figure shows a very simple configuration; in normal situations, we have moreADMs and more terminals.

Automatic Protection Switching

To create protection against failure in linear networks, SONET defines automatic protection switching (APS). APS in linear networks is defined at the line layer, which means the protection is between two ADMs or a pair of STS multiplexer/demultiplexers. The idea is to provide redundancy; a redundant line (fiber) can be used in case of failure in the main one. The main line is referred to as the work line and the redundant line as the protection line. Three schemes are common for protection in linear channels: one-plus-one, one-to-one, and one-to-many. Figure 17.20 shows all three schemes.

Figure 17.20 *Automatic protection switching in linear networks*



One-Plus-One APS In this scheme, there are normally two lines: one working line and one protection line. Both lines are active all the time. The sending multiplexer

sends the same data on both lines; the receiver multiplexer monitors the line and chooses the one with the better quality. If one of the lines fails, it loses its signal, and, of course, the other line is selected at the receiver. Although, the failure recovery for this scheme is instantaneous, the scheme is inefficient because two times the bandwidth is required. Note that one-plus-one switching is done at the path layer.

One-to-One APS In this scheme, which looks like the one-plus-one scheme, there is also one working line and one protection line. However, the data are normally sent on the working line until it fails. At this time, the receiver, using the reverse channel, informs the sender to use the protection line instead. Obviously, the failure recovery is slower than that of the one-plus-scheme, but this scheme is more efficient because the protection line can be used for data transfer when it is not used to replace the working line. Note that the one-to-one switching is done at the line layer.

One-to-Many APS This scheme is similar to the one-to-one scheme except that there is only one protection line for many working lines. When a failure occurs in one of the working lines, the protection line takes control until the failed line is repaired. It is not as secure as the one-to-one scheme because if more than one working line fails at the same time, the protection line can replace only one of them. Note that one-to-many APS is done at the line layer.

Ring Networks

ADMs make it possible to have SONET ring networks. SONET rings can be used in either a unidirectional or a bidirectional configuration. In each case, we can add extra rings to make the network self-healing, capable of self-recovery from line failure.

Unidirectional Path Switching Ring

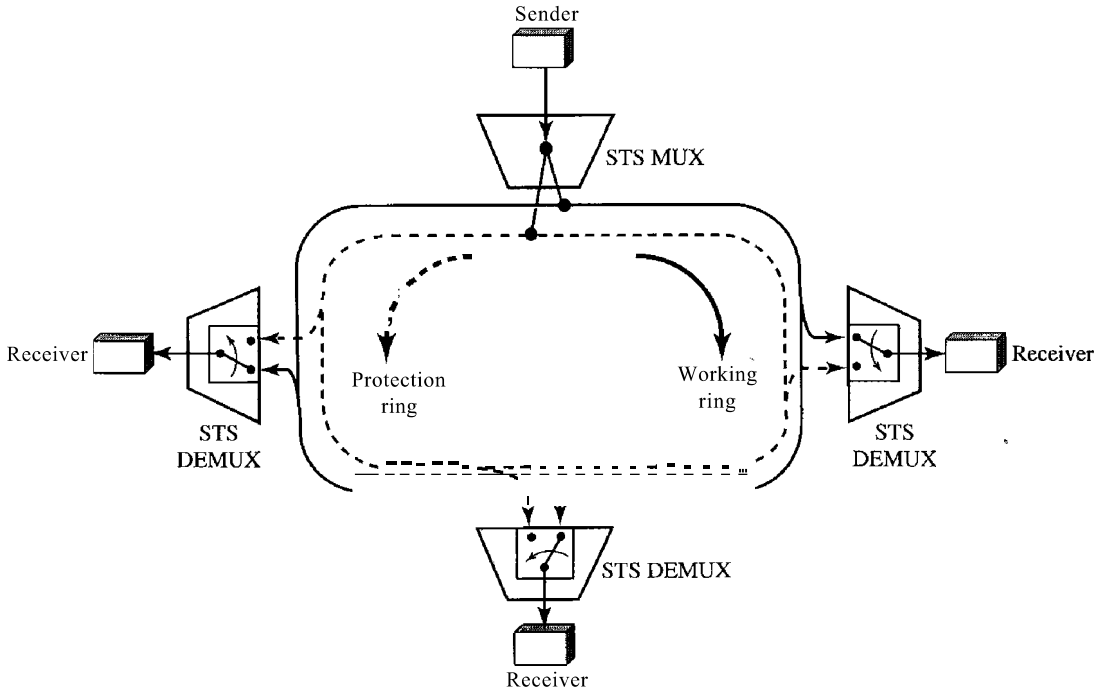
A unidirectional path switching ring (UPSR) is a unidirectional network with two rings: one ring used as the working ring and the other as the protection ring. The idea is similar to the one-plus-one APS scheme we discussed in a linear network. The same signal flows through both rings, one clockwise and the other counterclockwise. It is called UPSR because monitoring is done at the path layer. A node receives two copies of the electrical signals at the path layer, compares them, and chooses the one with the better quality. If part of a ring between two ADMs fails, the other ring still can guarantee the continuation of data flow. UPSR, like the one-plus-one scheme, has fast failure recovery, but it is not efficient because we need to have two rings that do the job of one. Half of the bandwidth is wasted. Figure 17.21 shows a UPSR network.

Although we have chosen one sender and three receivers in the figure, there can be many other configurations. The sender uses a two-way connection to send data to both rings simultaneously; the receiver uses selecting switches to select the ring with better signal quality. We have used one STS multiplexer and three STS demultiplexers to emphasize that nodes operate on the path layer.

Bidirectional Line Switching Ring

Another alternative in a SONET ring network is bidirectional line switching ring (BLSR). In this case, communication is bidirectional, which means that we need

Figure 17.21 *A unidirectional path switching ring*



two rings for working lines. We also need two rings for protection lines. This means BLSR uses four rings. The operation, however, is similar to the one-to-one APS scheme. If a working ring in one direction between two nodes fails, the receiving node can use the reverse ring to inform the upstream node in the failed direction to use the protection ring. The network can recover in several different failure situations that we do not discuss here. Note that the discovery of a failure in BLSR is at the line layer, not the path layer. The ADMs find the failure and inform the adjacent nodes to use the protection rings. Figure 17.22 shows a BLSR ring.

Combination of Rings

SONET networks today use a combination of interconnected rings to create services in a wide area. For example, a SONET network may have a regional ring, several local rings, and many site rings to give services to a wide area. These rings can be UPSR, BLSR, or a combination of both. Figure 17.23 shows the idea of such a wide-area ring network.

Mesh Networks

One problem with ring networks is the lack of scalability. When the traffic in a ring increases, we need to upgrade not only the lines, but also the ADMs. In this situation, a mesh network with switches probably give better performance. A switch in a network mesh is called a cross-connect. A cross-connect, like other switches we have seen, has input and output ports. In an input port, the switch takes an *OC-n* signal, changes it to an STS-n signal, demultiplexes it into the corresponding STS-1 signals, and sends each

Figure 17.22 *A bidirectional line switching ring*

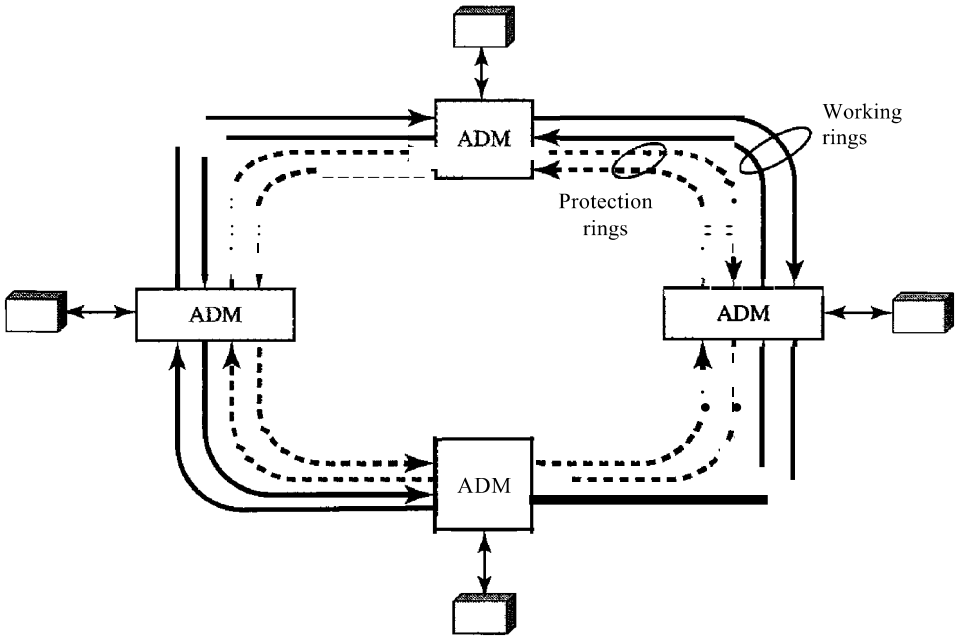
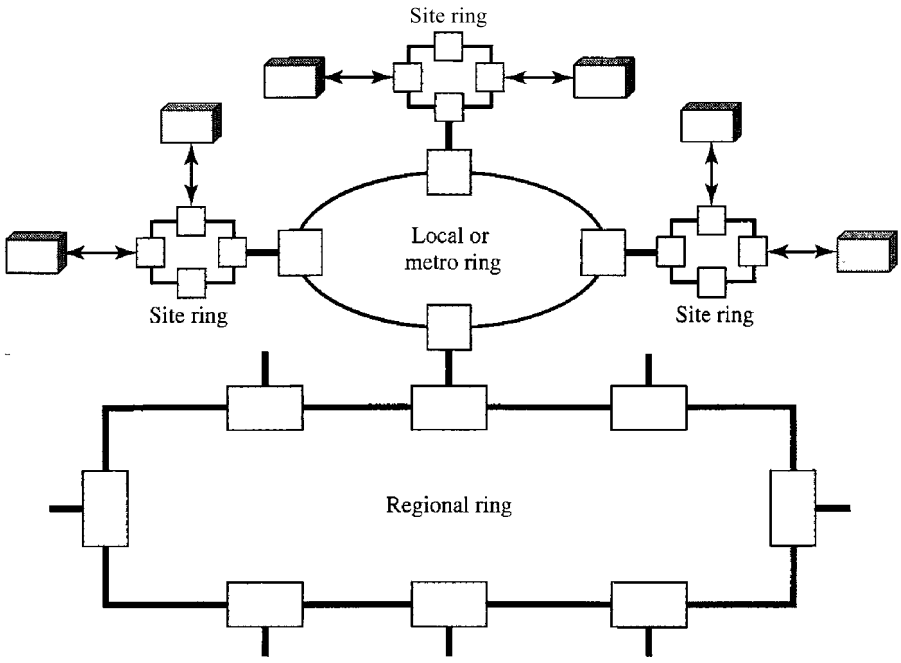
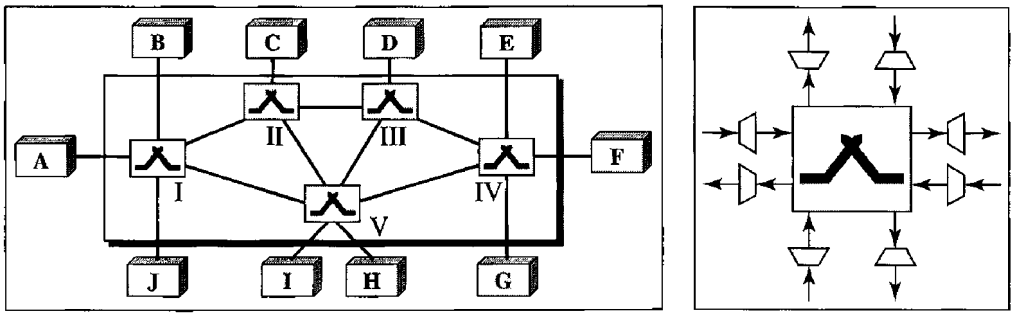


Figure 17.23 *A combination of rings in a SONET network*



STS-1 signal to the appropriate output port. An output port takes STS-1 signals coming from different input ports, multiplexes them into an *STS-n* signal, and makes an *OC-n* signal for transmission. Figure 17.24 shows a mesh SONET network, and the structure of a switch.

Figure 17.24 *A mesh SONET network*

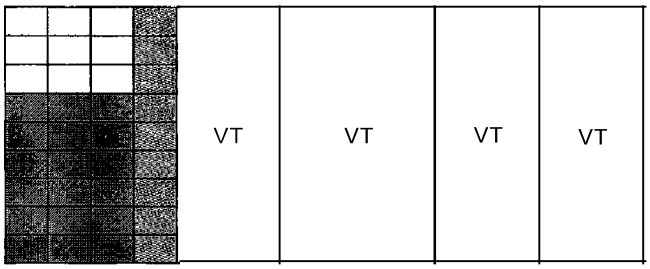


a. SONET mesh network b. Cross-connect switch

17.6 VIRTUAL TRIBUTARIES

SONET is designed to carry broadband payloads. Current digital hierarchy data rates (DS-I to DS-3), however, are lower than STS-l. To make SONET backward-compatible with the current hierarchy, its frame design includes a system of virtual tributaries (VTs) (see Figure 17.25). A virtual tributary is a partial payload that can be inserted into an STS-l and combined with other partial payloads to fill out the frame. Instead of using all 86 payload columns of an STS-l frame for data from one source, we can subdivide the SPE and call each component a VT.

Figure 17.25 *Virtual tributaries*

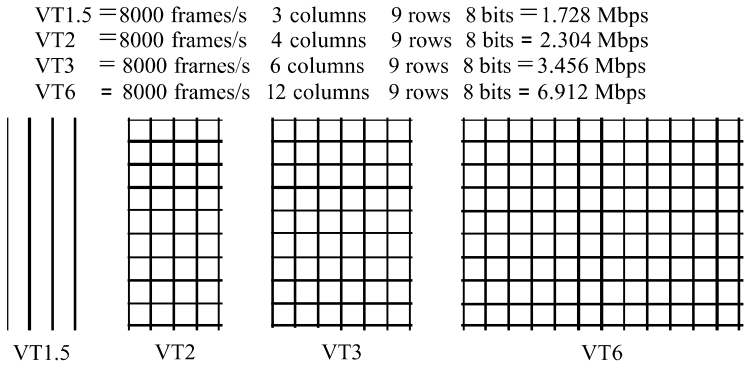


Types of VTs

Four types of VTs have been defined to accommodate existing digital hierarchies (see Figure 17.26). Notice that the number of columns allowed for each type of VT can be determined by doubling the type identification number (VT1.5 gets three columns, VT2 gets four columns, etc.).

- VT1.5 accommodates the U.S. DS-I service (1.544 Mbps).
- VT2 accommodates the European CEPT-l service (2.048 Mbps).
- VT3 accommodates the DS-IC service (fractional DS-l, 3.152 Mbps).
- VT6 accommodates the DS-2 service (6.312 Mbps).

Figure 17.26 *Virtual tributary types*



When two or more tributaries are inserted into a single STS-1 frame. they are interleaved column by column. SONET provides mechanisms for identifying each VT and separating them without demultiplexing the entire stream. Discussion of these mechanisms and the control issues behind them is beyond the scope of this book.

17.7 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [...] refer to the reference list at the end of the text.

Books

SONET is discussed in Section 2.5 of [Tan03], Section 15.2 of [Kes97], Sections 4.2 and 4.3 of [GW04], Section 8.2 of [Sta04], and Section 5.2 of [WVOOl

17.8 KEY TERMS

add/drop multiplexer (ADM)
automatic protection switching (APS)
bidirectional line switching ring (BLSR)
byte interleaving
line
line layer
line overhead (LOH)
optical carrier (Oe)
path
path layer
path overhead (POH)
photonic layer
regenerator
section
section layer
section overhead (SOH)
STS demultiplexer
STS multiplexer
Synchronous Digital Hierarchy (SDH)
Synchronous Optical Network (SONET)

synchronous payload envelope (SPE)
synchronous transport module (STM)
synchronous transport signal (STS)
terminal
unidirectional path switching ring (UPSR)
virtual tributary (VT)

17.9 SUMMARY

- Synchronous Optical Network (SONET) is a standard developed by ANSI for fiber-optic networks: Synchronous Digital Hierarchy (SDH) is a similar standard developed by ITU-T.
- SONET has defined a hierarchy of signals called synchronous transport signals (STSs). SDH has defined a similar hierarchy of signals called synchronous transfer modules (STMs).
- An *OC-n* signal is the optical modulation of an *STS-n* (or *STM-n)* signal.
- SONET defines four layers: path, line, section, and photonic.
- SONET is a synchronous TDM system in which all clocks are locked to a master clock.
- A SONET system can use the following equipment:
  1. STS multiplexers
  2. STS demultiplexers
  3. Regenerators
  4. Add/drop multiplexers
  5. Terminals
- SONET sends 8000 frames per second; each frame lasts 125 μs.
- An STS-I frame is made of 9 rows and 90 columns; an *STS-n* frame is made of 9 rows and *n* x 90 columns.
- STSs can be multiplexed to get a new STS with a higher data rate.
- SONET network topologies can be linear, ring, or mesh.
- A linear SONET network can be either point-to-point or multipoint.
- A ring SONET network can be unidirectional or bidirectional.
- To make SONET backward-compatible with the current hierarchy, its frame design includes a system of virtual tributaries (VTs).

17.10 PRACTICE SET

Review Questions

1. What is the relationship between SONET and SDH?
2. What is the relationship between STS and STM?
3. How is an STS multiplexer different from an add/drop multiplexer since both can add signals together?

4. What is the relationship between STS signals and OC signals?
5. What is the purpose of the pointer in the line overhead?
6. Why is SONET called a synchronous network?
7. What is the function of a SONET regenerator?
8. What are the four SONET layers?
9. Discuss the functions of each SONET layer.
10. What is a virtual tributary?

Exercises

11. What are the user data rates of STS-3, STS-9, and STS-12?
12. Show how STS-9's can be multiplexed to create an STS-36. Is there any extra overhead involved in this type of multiplexing?
13. A stream of data is being carried by STS-1 frames. If the data rate of the stream is 49.540 Mbps, how many STS-1 frames per second must let their H3 bytes carry data?
14. A stream of data is being carried by STS-1 frames. If the data rate of the stream is 49.530 Mbps, how many frames per second should leave one empty byte after the H3 byte?
15. Table 17.2 shows that the overhead bytes can be categorized as A, B, C, D, E, F, G, H, J, K, and Z bytes.
    a. Why are there no A bytes in the LOH or POH?
    b. Why are there no C bytes in the LOH?
    c. Why are there no D bytes in the POH?
    d. Why are there no E bytes in the LOH or POR?
    e. Why are there no F bytes in the LOH or POH?
    f. Why are there no G bytes in the SOH or LOH?
    g. Why are there no H bytes in the SOH?
    h. Why are there no J bytes in the SOH or LOH?
    i. Why are there no K bytes in the SOH or POH?
    j. Why are there no Z bytes in the SOH?
16. Why are B bytes present in all three headers?

CHAPTER 18

Virtual-Circuit Networks: Frame Relay and ATM

In Chapter 8, we discussed switching techniques. We said that there are three types of switching: circuit switching, packet switching, and message switching. We also mentioned that packet switching can use two approaches: the virtual-circuit approach and the datagram approach.

In this chapter, we show how the virtual-circuit approach can be used in wide-area networks. Two common WAN technologies use virtual-circuit switching. Frame Relay is a relatively high-speed protocol that can provide some services not available in other WAN technologies such as DSL, cable TV, and T lines. ATM, as a high-speed protocol, can be the superhighway of communication when it deploys physical layer carriers such as SONET.

We first discuss Frame Relay. We then discuss ATM in greater detaiL Finally, we show how ATM technology, which was originally designed as a WAN technology, can also be used in LAN technology, ATM LANs.

18.1 FRAME RELAY

Frame Relay is a virtual-circuit wide-area network that was designed in response to demands for a new type of WAN in the late 1980s and early 1990s.

1. Prior to Frame Relay, some organizations were using a virtual-circuit switching network called X.25 that performed switching at the network layer. For example, the Internet, which needs wide-area networks to carry its packets from one place to another, used X.25. And X.25 is still being used by the Internet, but it is being replaced by other WANs. However, X.25 has several drawbacks:
   a. X.25 has a low 64-kbps data rate. By the 1990s, there was a need for higher-data-rate WANs.
   b. X.25 has extensive flow and error control at both the data link layer and the network layer. This was so because X.25 was designed in the 1970s, when the available transmission media were more prone to errors. Flow and error control at both layers create a large overhead and slow down transmissions. X.25 requires acknowledgments for both data link layer frames and network layer packets that are sent between nodes and between source and destination.

c. Originally X.25 was designed for private lise, not for the Internet. X.25 has its own network layer. This means that the user's data are encapsulated in the network layer packets of X.25. The Internet, however, has its own network layer, which means if the Internet wants to use X.25, the Internet must deliver its network layer packet, called a datagram, to X.25 for encapsulation in the X.25 packet. This doubles the overhead.

2. Disappointed with X.25, some organizations started their own private WAN by leasing T-l or T-3 lines from public service providers. This approach also has some drawbacks.
   a. If an organization has n branches spread over an area, it needs $n(n - 1)/2$ T-I or T-3 lines. The organization pays for all these lines although it may use the lines only 10 percent of the time. This can be very costly:
   b. The services provided by T-I and T-3 lines assume that the user has fixed-rate data all the time. For example, a T-l 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 that need to send **bursty data.** For example, a user may want to send data at 6 Mbps for 2 s, 0 Mbps (nothing) for 7 s, and 3.44 Mbps for 1 s for a total of 15.44 Mbits during a period of 10 s. Although the average data rate is still 1.544 Mbps, the T-I line cannot accept this type of demand because it is designed for fixed-rate data, not bursty data. Bursty data require what is called **bandwidth on demand.** The user needs different bandwidth allocations at different times.

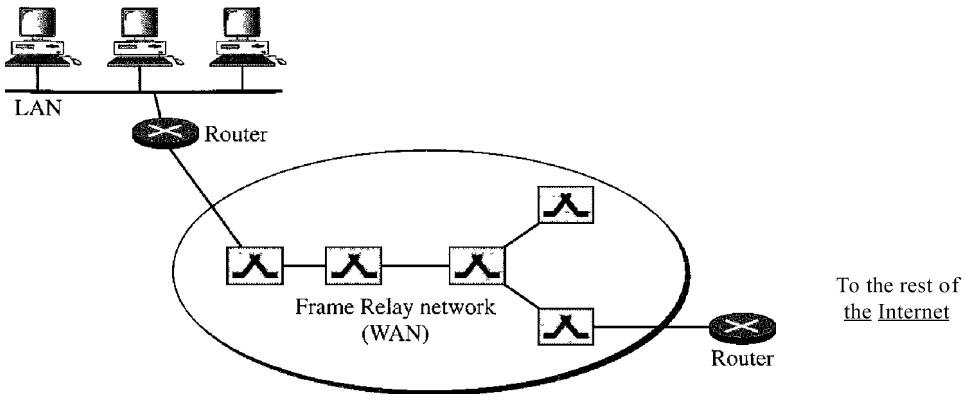
In response to the above drawbacks, Frame Relay was designed. Frame Relay is a wide-area network with the following features:

1. Frame Relay operates at a higher speed (1.544 Mbps and recently 44.376 Mbps). This means that it can easily be used instead of a mesh of T-I or T-3 lines.
2. Frame Relay operates in just the physical and data link layers. This means it can easily be used as a backbone network to provide services to protocols that already have a network layer protocol, such as the Internet.
3. Frame Relay allows bursty data.
4. Frame Relay allows a frame size of 9000 bytes, which can accommodate all local-area network frame sizes.
5. Frame Relay is less expensive than other traditional WANs.
6. Frame Relay has error detection at the data link layer only. There is no flow control or error control. There is not even a retransmission policy if a frame is damaged; it is silently dropped. Frame Relay was designed in this way to provide fast transmission capability for more reliable media and for those protocols that have flow and error control at the higher layers.

Architecture

Frame Relay provides permanent virtual circuits and switched virtual circuits. Figure 18.1 shows an example of a Frame Relay network connected to the Internet. The routers are used, as we will see in Chapter 22, to connect LANs and WANs in the Internet. In the figure, the Frame Relay WAN is used as one link in the global Internet.

Figure 18.1 *Frame Relay network*



Virtual Circuits

Frame Relay is a virtual circuit network. A virtual circuit in Frame Relay is identified by a number called a data link connection identifier (DLCI).

VCIs in Frame Relay are called DLCIs.

Permanent Versus Switched Virtual Circuits

A source and a destination may choose to have a permanent virtual circuit (PVC). In this case, the connection setup is simple. The corresponding table entry is recorded for all switches by the administrator (remotely and electronically, of course). An outgoing DLCI is given to the source, and an incoming DLCI is given to the destination.

PVC connections have two drawbacks. First, they are costly because two parties pay for the connection all the time even when it is not in use. Second, a connection is created from one source to one single destination. If a source needs connections with several destinations, it needs a PVC for each connection. An alternate approach is the switched virtual circuit (SVC). The SVC creates a temporary, short connection that exists only when data are being transferred between source and destination. An SVC requires establishing and terminating phases as discussed in Chapter 8.

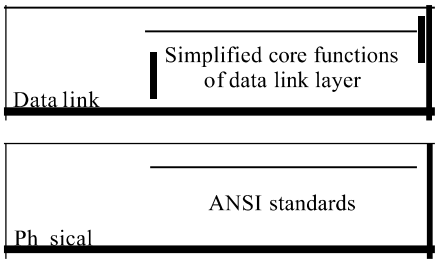
Switches

Each switch in a Frame Relay network has a table to route frames. The table matches an incoming port-DLCI combination with an outgoing port-DLCI combination as we described for general virtual-circuit networks in Chapter 8. The only difference is that VCIs are replaced by DLCIs.

Frame Relay Layers

Figure 18.2 shows the Frame Relay layers. Frame Relay has only physical and data link layers.

Figure 18.2 *Frame Relay layers*



Frame Relay operates only at the physical and data link layers.

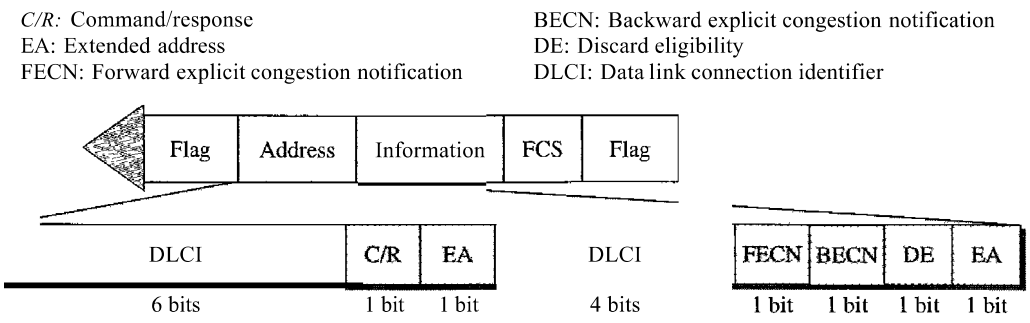
Physical Layer

No specific protocol is defined for the physical layer in Frame Relay. Instead, it is left to the implementer to use whatever is available. Frame Relay supports any of the protocols recognized by ANSI.

Data Link Layer

At the data link layer, Frame Relay uses a simple protocol that does not support flow or error control. It only has an error detection mechanism. Figure 18.3 shows the format of a Frame Relay frame. The address field defines the DLCI as well as some bits used to control congestion.

Figure 18.3 *Frame Relay frame*



The descriptions of the fields are as follows:

- Address (DLCI) field. The first 6 bits of the first byte makes up the first part of the DLCI. The second part of the DLCI uses the first 4 bits of the second byte. These bits are part of the lO-bit data link connection identifier defined by the standard. We will discuss extended addressing at the end of this section.

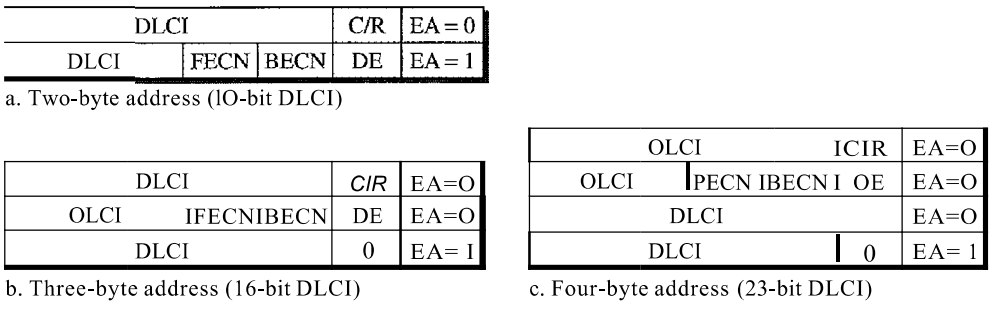
- Command/response (CIR). The command/response (C/R) bit is provided to allow upper layers to identify a frame as either a command or a response. It is not used by the Frame Relay protocol.
- Extended address (EA). The extended address (EA) bit indicates whether the current byte is the final byte of the address. An EA of 0 means that another address byte is to follow (extended addressing is discussed later). An EA of 1 means that the current byte is the final one.
- Forward explicit congestion notification (FECN). The forward explicit congestion notification (FECN) bit can be set by any switch to indicate that traffic is congested. This bit informs the destination that congestion has occurred. In this way, the destination knows that it should expect delay or a loss of packets. We will discuss the use of this bit when we discuss congestion control in Chapter 24.
- Backward explicit congestion notification (BECN). The backward explicit congestion notification (BECN) bit is set (in frames that travel in the other direction) to indicate a congestion problem in the network. This bit informs the sender that congestion has occurred. In this way, the source knows it needs to slow down to prevent the loss of packets. We will discuss the use of this bit when we discuss congestion control in Chapter 24.
- Discard eligibility (DE). The discard eligibility (DE) bit indicates the priority level of the frame. In emergency situations, switches may have to discard frames to relieve bottlenecks and keep the network from collapsing due to overload. When set (DE 1), this bit tells the network to discard this frame if there is congestion. This bit can be set either by the sender of the frames (user) or by any switch in the network.

Frame Relay does not provide flow or error control;
they must be provided by the upper-layer protocols.

Extended Address

To increase the range of DLCIs, the Frame Relay address has been extended from the original 2-byte address to 3- or 4-byte addresses. Figure 18.4 shows the different addresses. Note that the EA field defines the number of bytes; it is 1 in the last byte of the address, and it is a in the other bytes. Note that in the 3- and 4-byte formats, the bit before the last bit is set to 0.

Figure 18.4 *Three address formats*



a. Two-byte address (lO-bit DLCI)

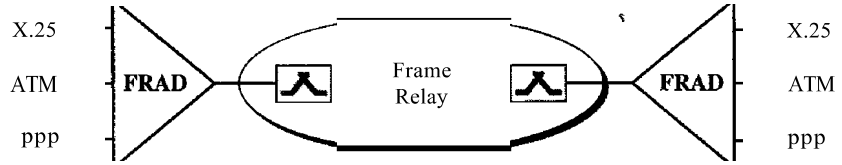
b. Three-byte address (16-bit DLCI)

c. Four-byte address (23-bit DLCI)

FRADs

To handle frames arriving from other protocols, Frame Relay uses a device called a Frame Relay assembler/disassembler (FRAD). A FRAD assembles and disassembles frames coming from other protocols to allow them to be carried by Frame Relay frames. A FRAD can be implemented as a separate device or as part of a switch. Figure 18.5 shows two FRADs connected to a Frame Relay network.

Figure 18.5 *FRAD*



VOFR

Frame Relay networks offer an option called Voice Over Frame Relay (VOFR) that sends voice through the network. Voice is digitized using PCM and then compressed. The result is sent as data frames over the network. This feature allows the inexpensive sending of voice over long distances. However, note that the quality of voice is not as good as voice over a circuit-switched network such as the telephone network. Also, the varying delay mentioned earlier sometimes corrupts real-time voice.

LMI

Frame Relay was originally designed to provide PVC connections. There was not, therefore, a provision for controlling or managing interfaces. Local Management Information (LMI) is a protocol added recently to the Frame Relay protocol to provide more management features. In particular, LMI can provide

- A keep-alive mechanism to check if data are flowing.
- A multicast mechanism to allow a local end system to send frames to more than one remote end system.
- A mechanism to allow an end system to check the status of a switch (e.g., to see if the switch is congested).

Congestion Control and Quality of Service

One of the nice features of Frame Relay is that it provides congestion control and quality of service (QoS). We have not discussed these features yet. In Chapter 24, we introduce these two important aspects of networking and discuss how they are implemented in Frame Relay and some other networks.

18.2 ATM

Asynchronous Transfer Mode (ATM) is the cell relay protocol designed by the ATM Forum and adopted by the ITU-T. The combination of ATM and SONET will allow high-speed interconnection of all the world's networks. In fact, ATM can be thought of as the "highway" of the information superhighway.

Design Goals

Among the challenges faced by the designers of ATM, six stand out.

1. Foremost is the need for a transmission system to optimize the use of high-data-rate transmission media, in particular optical fiber. In addition to offering large bandwidths, newer transmission media and equipment are dramatically less susceptible to noise degradation. A technology is needed to take advantage of both factors and thereby maximize data rates.
2. The system must interface with existing systems and provide wide-area interconnectivity between them without lowering their effectiveness or requiring their replacement.
3. The design must 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.
4. 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).
5. The new system must be connection-oriented to ensure accurate and predictable delivery.
6. 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).

Problems

Before we discuss the solutions to these design requirements, it is useful to examine some of the problems associated with existing systems.

Frame Networks

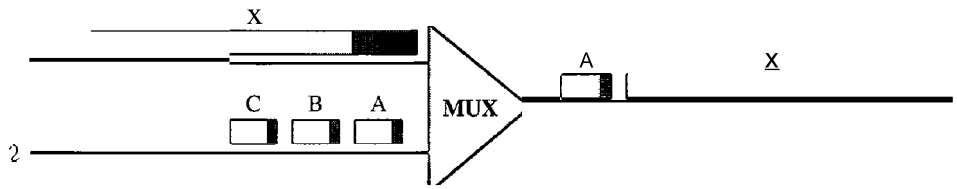
Before ATM, data communications at the data link layer had been based on frame switching and frame networks. Different protocols use frames of varying size and intricacy. As networks become more complex, the information that must be carried in the header becomes more extensive. The result is larger and larger headers relative to the size of the data unit. In response, some protocols have enlarged the size of the data unit to make header use more efficient (sending more data with the same size header). Unfortunately, large data fields create waste. If there is not much information to transmit, much of the field goes unused. To improve utilization, some protocols provide variable frame sizes to users.

Mixed Network Traffic

As you can imagine, the variety of frame sizes makes traffic unpredictable. Switches, multiplexers, and routers must incorporate elaborate software systems to manage the various sizes of frames. A great deal of header information must be read, and each bit counted and evaluated to ensure the integrity of every frame. Internetworking among the different frame networks is slow and expensive at best, and impossible at worst.

Another problem is that of providing consistent data rate delivery when frame sizes are unpredictable and can vary so dramatically. To get the most out of broadband technology, traffic must be time-division multiplexed onto shared paths. Imagine the results of multiplexing frames from two networks with different requirements (and frame designs) onto one link (see Figure 18.6). What happens when line 1 uses large frames (usually data frames) while line 2 uses very small frames (the norm for audio and video information)?

Figure 18.6 *Multiplexing using different frame sizes*



If line 1's gigantic frame X arrives at the multiplexer even a moment earlier than line 2's frames, the multiplexer puts frame X onto the new path first. After all, even if line 2's frames have priority, the multiplexer has no way of knowing to wait for them and so processes the frame that has arrived. Frame A must therefore wait for the entire X bit stream to move into place before it can follow. The sheer size of X creates an unfair delay for frame A. The same imbalance can affect all the frames from line 2.

Because audio and video frames ordinarily are small, mixing them with conventional data traffic often creates unacceptable delays of this type and makes shared frame links unusable for audio and video information. Traffic must travel over different paths, in much the same way that automobile and train traffic does. But to fully utilize broad bandwidth links, we need to be able to send all kinds of traffic over the same links.

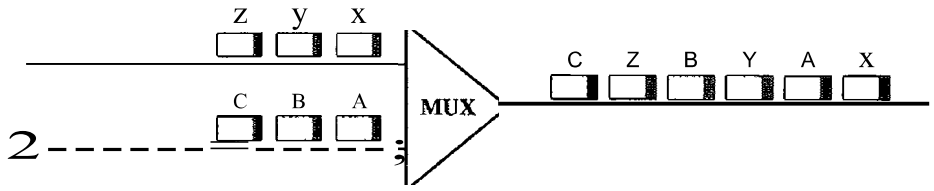
Cell Networks

Many of the problems associated with frame internetworking are solved by adopting a concept called cell networking. A cell is a small data unit of fixed size. In a **cell** network, which uses the **cell** as the basic unit of data exchange, all data are loaded into identical cells that can be transmitted with complete predictability and uniformity. As frames of different sizes and formats reach the cell network from a tributary network, they are split into multiple small data units of equal length and are loaded into cells. The cells are then multiplexed with other cells and routed through the cell network. Because each cell is the same size and all are small, the problems associated with multiplexing different-sized frames 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.

Figure 18.7 shows the multiplexer from Figure 18.6 with the two lines sending cells instead of frames. Frame X has been segmented into three cells: X, Y, and Z. Only the first cell from line 1 gets put on the link before the first cell from line 2. The cells from the two lines are interleaved so that none suffers a long delay.

Figure 18.7 *Multiplexing using cells*



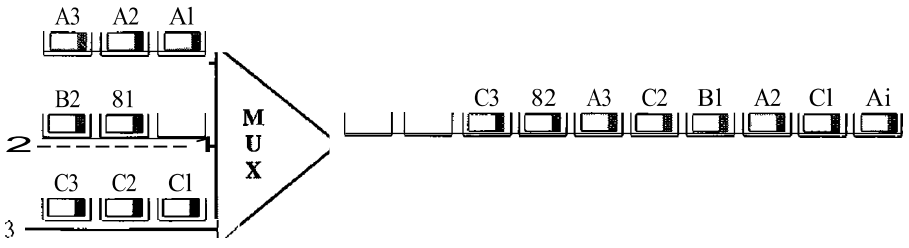
A second point in this same scenario is that the high speed of the links coupled with the small size of the cells means that, despite interleaving, cells from each line arrive at their respective destinations in an approximation of a continuous stream (much as a movie appears to your brain to be continuous action when in fact it is really a series of separate, still photographs). In this way, a cell network can handle real-time transmissions, such as a phone call, without the parties being aware of the segmentation or multiplexing at all.

Asynchronous TDM

ATM uses asynchronous time-division multiplexing-that is why it is called Asynchronous Transfer Mode-to multiplex cells corning from different channels. It uses fixed-size slots (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 18.8 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.

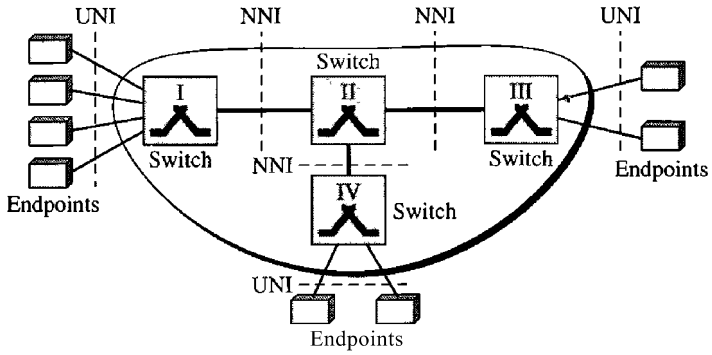
Figure 18.8 *ATM multiplexing*



Architecture

ATM is a cell-switched network. The user access devices, called the endpoints, are connected through a user-to-network interface (UNI) to the switches inside the network. The switches are connected through network-to-network interfaces (NNIs). Figure 18.9 shows an example of an ATM network.

Figure 18.9 *Architecture of an ATM network*



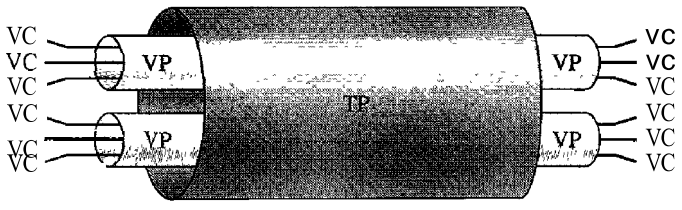
Virtual Connection

Connection between two endpoints is accomplished through transmission paths (TPs), virtual paths (YPs), and virtual circuits (YCs). A transmission path (TP) is the physical connection (wire, cable, satellite, and so on) between an endpoint 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 connect the two cities.

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.

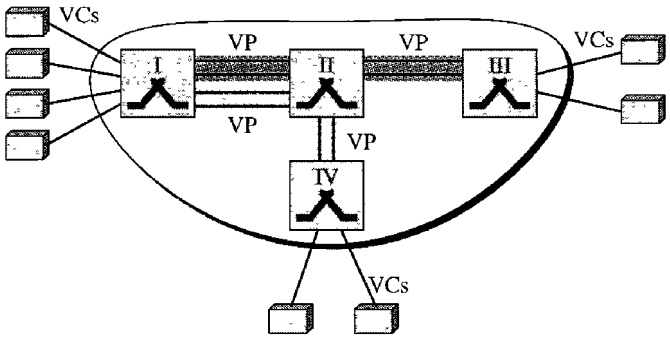
Cell networks are based on virtual circuits (VCs). All cells belonging to a single message follow the same virtual circuit and remain in their original order until they reach their destination. Think of a virtual circuit as the lanes of a highway (virtual path). Figure 18.10 shows the relationship between 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.

Figure 18.10 *Tp, VPs, and VCs*



To better understand the concept of VPs and VCs, look at Figure 18.11. In this figure, eight endpoints are communicating using four VCs. However, the first two VCs seem to share the same virtual path from switch I to switch III, so it is reasonable to bundle these two VCs together to form one VP. On the other hand, it is clear that the other two VCs share the same path from switch I to switch IV, so it is also reasonable to combine them to form one VP.

Figure 18.11 *Example of VPs and VCs*



Identifiers In a virtual circuit network, to route data from one endpoint 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 (Vel). The VPI defines the specific VP, and the Vel defines a particular VC inside the VP. The VPI is the same for all virtual connections that are bundled (logically) into one VP.

Note that a virtual connection is defined by a pair of numbers: the VPI and the vel.

Figure 18.12 shows the VPIs and VCIs for a transmission path. The rationale for dividing an identifier into two parts will become clear when we discuss routing in an ATM network.

The lengths of the VPIs for UNIs and NNIs are different. In a UNI, the VPI is 8 bits, whereas in an NNI, the VPI is 12 bits. The length of the VCI is the same in both interfaces (16 bits). We therefore can say that a virtual connection is identified by 24 bits in a UNI and by 28 bits in an NNI (see Figure 18.13).

The whole idea behind dividing a virtual circuit identifier into two parts is to allow hierarchical routing. Most of the switches in a typical ATM network are routed using VPIs. The switches at the boundaries of the network, those that interact directly with the endpoint devices, use both VPIs and VCIs.

Cells

The basic data unit in an ATM network is called a cell. A cell is only 53 bytes long with 5 bytes allocated to the header and 48 bytes carrying the payload (user data may be less

Figure 18.12 *Connection identifiers*

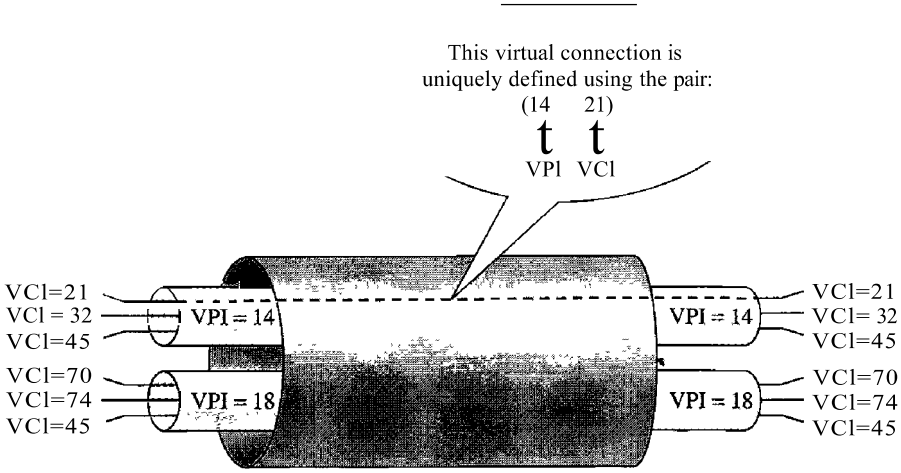
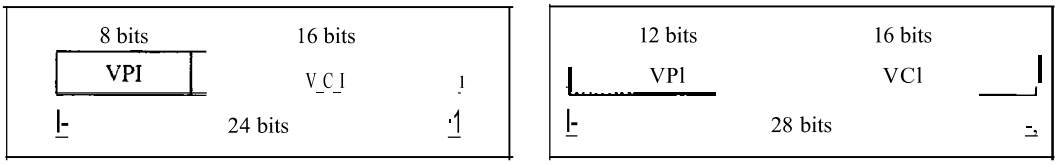


Figure 18.13 *Virtual connection identifiers in UNIs and NNIs*

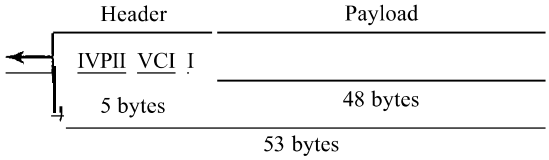


a. VPI and VCI in a UNI

b. VPI and VCI in an NNI

than 48 bytes). We will study in detail the fields of a cell, but for the moment it suffices to say that most of the header is occupied by the VPI and VCI that define the virtual connection through which a cell should travel from an endpoint to a switch or from a switch to another switch. Figure 18.14 shows the cell structure.

Figure 18.14 *An ATM cell*



Connection Establishment and Release

Like Frame Relay, ATM uses two types of connections: PVC and SVc.

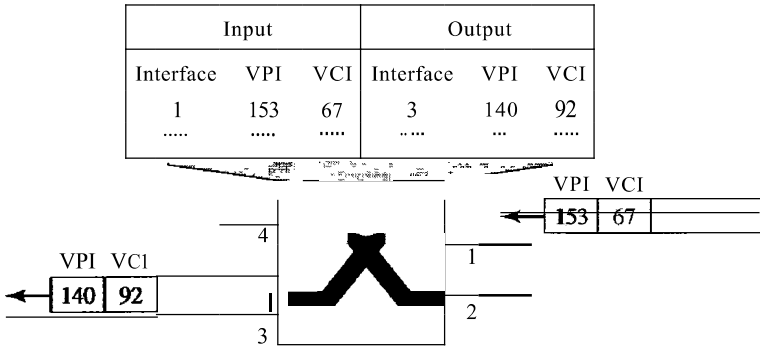
PVC A permanent virtual-circuit connection is established between two endpoints by the network provider. The VPls and vcrs are defined for the permanent connections, and the values are entered for the tables of each switch.

SVC In a switched virtual-circuit connection, each time an endpoint wants to make a connection with another endpoint, a new virtual circuit must be established. ATM cannot do the job by itself, but needs the network layer addresses and the services of another protocol (such as IP). The signaling mechanism of this other protocol makes a connection request by using the network layer addresses of the two endpoints. The actual mechanism depends on the network layer protocol.

Switching

ATM uses switches to route the cell from a source endpoint to the destination endpoint. A switch routes the cell using both the VPIs and the VCIs. The routing requires the whole identifier. Figure 18.15 shows how a VPC switch routes the cell. A cell with a VPI of 153 and VCI of 67 arrives at switch interface (port) 1. The switch checks its switching table, which stores six pieces of information per row: arrival interface number, incoming VPI, incoming VCI, corresponding outgoing interface number, the new VPI, and the new VCL The switch finds the entry with the interface 1, VPI 153, and VCI 67 and discovers that the combination corresponds to output interface 3, VPI 140, and VCI 92. It changes the VPI and VCI in the header to 140 and 92, respectively, and sends the cell out through interface 3.

Figure 18.15 *Routing with a switch*



Switching Fabric

The switching technology has created many interesting features to increase the speed of switches to handle data. We discussed switching fabrics in Chapter 8.

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 (see Figure 18.16).

The endpoints use all three layers while the switches use only the two bottom layers (see Figure 18.17).

Physical Layer

Like Ethernet and wireless LANs, ATM cells can be carried by any physical layer carrier.

Figure 18.16 *ATM layers*

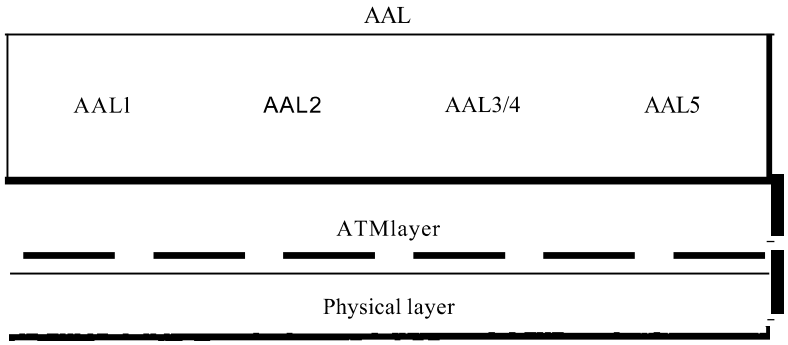
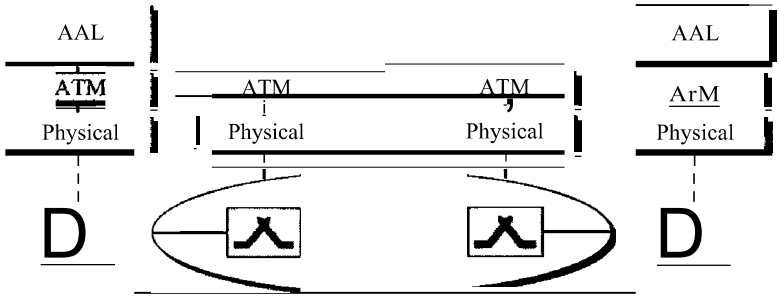


Figure 18.17 *ATM layers in endpoint devices and switches*



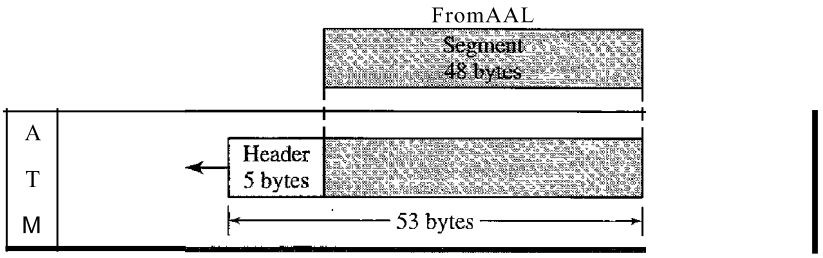
SONET The original design of ATM was based on *SONET* (see Chapter 17) as the physical layer carrier. SONET is preferred for two reasons. First, the high data rate of SONET's carrier reflects the design and philosophy of ATM. Second, in using SONET, the boundaries of cells can be clearly defined. As we saw in Chapter 17, SONET specifies the use of a pointer to define the beginning of a payload. If the beginning of the first ATM cell is defined, the rest of the cells in the same payload can easily be identified because there are no gaps between cells. Just count 53 bytes ahead to find the next cell.

Other Physical Technologies ATM does not limit the physical layer to SONET. Other technologies, even wireless, may be used. However, the problem of cell boundaries must be solved. One solution is for the receiver to guess the end of the cell and apply the CRC to the 5-byte header. If there is no error, the end of the cell1s found, with a high probability, correctly. Count 52 bytes back to find the beginning of the cell.

ATM Layer

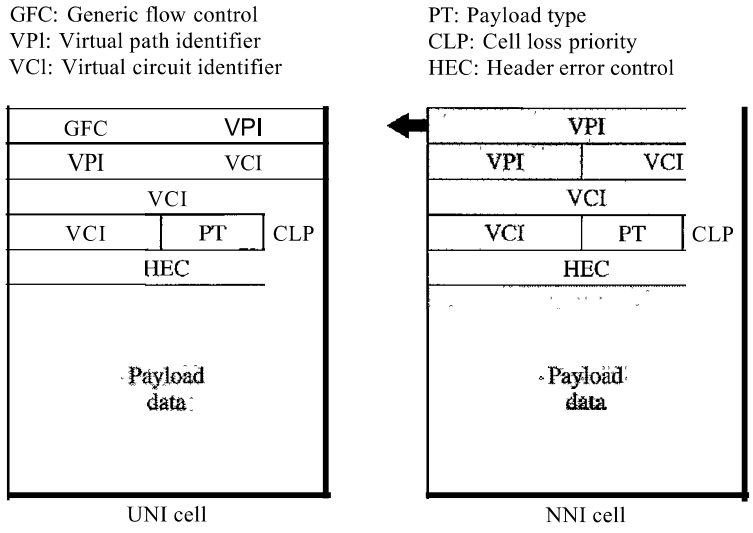
The ATM layer provides routing, traffic management, switching, and multiplexing services. It processes outgoing traffic by accepting 48-byte segments from the AAL sublayers and transforming them into 53-byte cells by the addition of a 5-byte header (see Figure 18.18).

Figure 18.18 *ATM layer*



Header Format ATM uses two fonnats for this header, one for user-to-network interface (UNI) cells and another for network-to-network interface (NNI) cells. Figure 18.19 shows these headers in the byte-by-byte format preferred by the ITU-T (each row represents a byte).

Figure 18.19 *ATM headers*



- Generic flow control (GFC). The 4-bit GFC field provides flow control at the UNI level. The ITU-T has determined that this level of flow control is not necessary at the NNI level. In the NNI header, therefore, these bits are added to the VPI. The longer VPI allows more virtual paths to be defined at the NNI level. The format for this additional VPI has not yet been determined.
- Virtual path identifier (VPI). The VPI is an 8-bit field in a UNI cell and a 12-bit field in an NNI cell (see above).
- Virtual circuit identifier (VCI). The VCI is a 16-bit field in both frames.
- Payload type (PT). In the 3-bit PT field, the first bit defines the payload as user data or managerial information. The interpretation of the last 2 bits depends on the first bit.

- Cell loss priority (CLP). The I-bit CLP field is provided for congestion control. A cell with its CLP bit set to I must be retained as long as there are cells with a CLP of 0. We discuss congestion control and quality of service in an ATM network in Chapter 24.
- Header error correction (HEC). The HEC is a code computed for the first 4 bytes of the header. It is a CRC with the divisor $x^8 + x^2 + x + 1$ that is used to correct single-bit errors and a large class of multiple-bit errors.

Application Adaptation Layer

The application adaptation layer (AAL) was designed to enable two ATM concepts. First, ATM must accept any type of payload, both data frames and streams of bits. A data frame can come from an upper-layer protocol that creates a clearly defined frame to be sent to a carrier network such as ATM. A good exampre is the Internet. ATM must also carry multimedia payload. It can accept continuous bit streams and break them into chunks to be encapsulated into a cell at the ATM layer. AAL uses two sublayers to accomplish these tasks.

Whether the data are a data frame or a stream of bits, the payload must be segmented into 48-byte segments to be carried by a cell. At the destination, these segments need to be reassembled to recreate the original payload. The AAL defines a sublayer, called a segmentation and reassembly (SAR) sublayer, to do so. Segmentation is at the source; reassembly, at the destination.

Before data are segmented by SAR, they must be prepared to guarantee the integrity of the data. This is done by a sublayer called the convergence sublayer (CS).

ATM defines four versions of the AAL: AAL1, AAL2, *AAL3/4,* and AAL5. Although we discuss all these versions, we need to inform the reader that the common versions today are AAL I and AAL5. The first is used in streaming audio and video communication; the second, in data communications.

AALI AALI supports applications that transfer information at constant bit rates, such as video and voice. It allows ATM to connect existing digital telephone networks such as voice channels and T lines. Figure 18.20 shows how a bit stream of data is chopped into 47-byte chunks and encapsulated in cells.

The CS sublayer divides the bit stream into 47-byte segments and passes them to the SAR sublayer below. Note that the CS sublayer does not add a header.

The SAR sublayer adds 1 byte of header and passes the 48-byte segment to the ATM layer. The header has two fields:

- Sequence number (SN). This 4-bit field defines a sequence number to order the bits. The first bit is sometimes used for timing, which leaves 3 bits for sequencing (modulo 8).
- Sequence number protection (SNP). The second 4-bit field protects the first field. The first 3 bits automatically correct the SN field. The last bit is a parity bit that detects error over all 8 bits.

AAL2 Originally AAL2 was intended to support a variable-data-rate bit stream, but it has been redesigned. It is now used for low-bit-rate traffic and short-frame traffic such as audio (compressed or uncompressed), video, or fax. A good example of AAL2 use is in mobile telephony. AAL2 allows the multiplexing of short frames into one cell.

Figure 18.20 *AAL1*

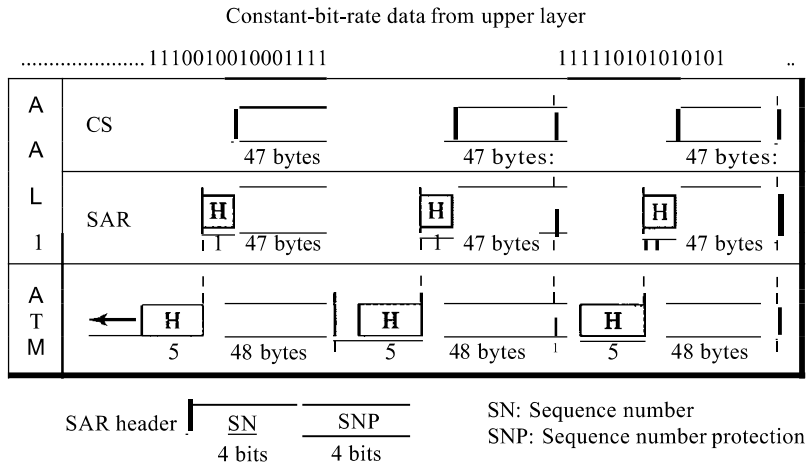
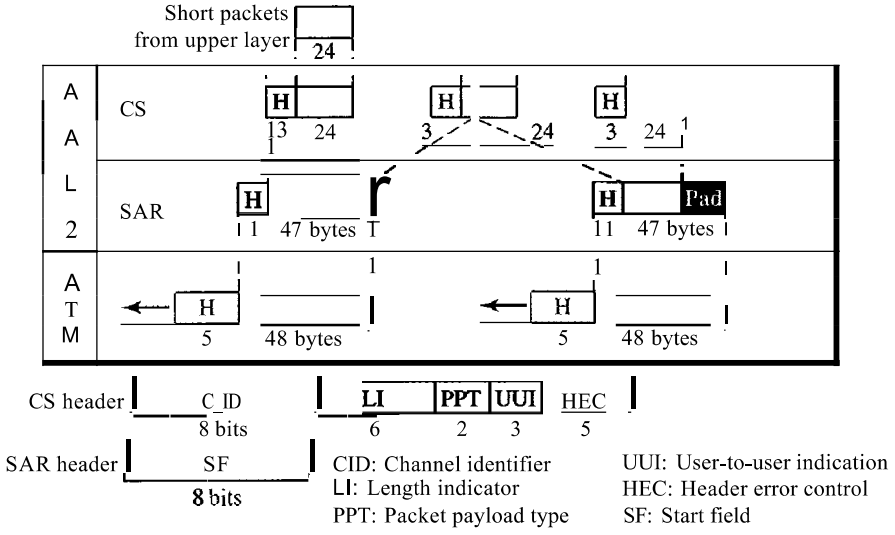


Figure 18.21 shows the process of encapsulating a short frame from the same source (the same user of a mobile phone) or from several sources (several users of mobile telephones) into one cell.

Figure 18.21 *AAL2*



The CS layer overhead consists of five fields:

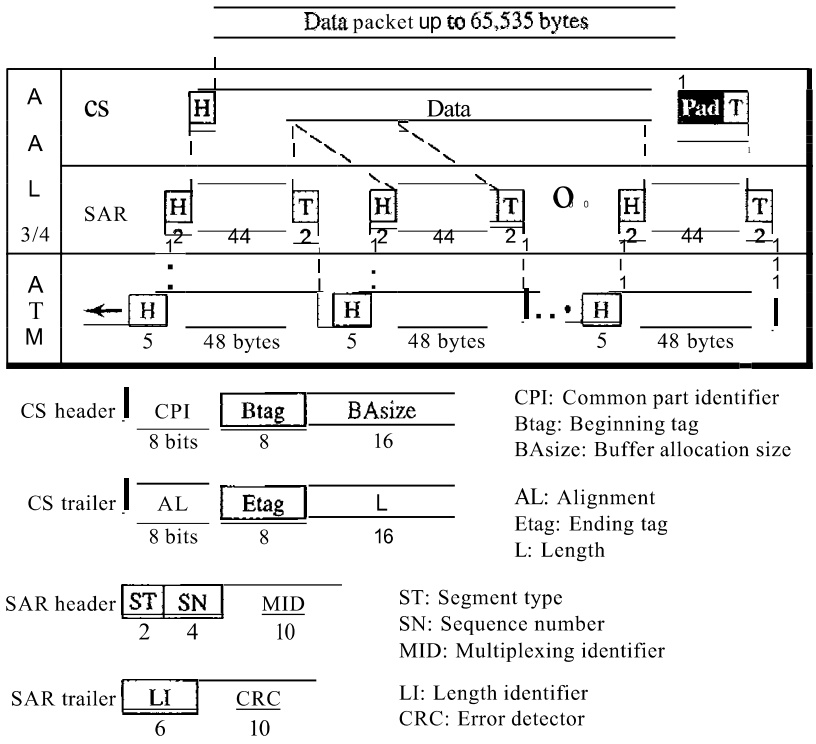
- Channel identifier (CID). The 8-bit CID field defines the channel (user) of the short packet.
- Length indicator (LI). The 6-bit LI field indicates how much of the final packet is data.
- Packet payload type (PPT). The PPT field defines the type of packet.

- User-to-user indicator (UUI). The UUI field can be used by end-to-end users.
- Header error control (HEC). The last 5 bits is used to correct errors in the header.

The only overhead at the SAR layer is the start field (SF) that defines the offset from the beginning of the packet.

AAL3/4 Initially, AAL3 was intended to support connection-oriented data services and AAL4 to support connectionless services. As they evolved, however, it became evident that the fundamental issues of the two protocols were the same. They have therefore been combined into a single format called AAL3/4. Figure 18.22 shows the AAL3/4 sublayer.

Figure 18.22 *AAL3/4*



The CS layer header and trailer consist of six fields:

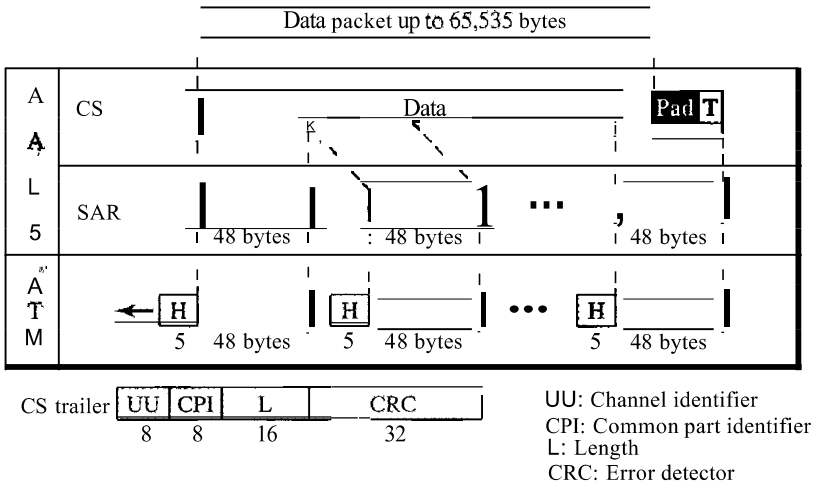
- Common part identifier (CPI). The CPI defines how the subsequent fields are to be interpreted. The value at present is 0.
- Begin tag (Btag). The value of this field is repeated in each ceU to identify all the cells belonging to the same packet. The value is the same as the Etag (see below).
- Buffer allocation size (BAsize). The 2-byte BA field tells the receiver what size buffer is needed for the coming data.
- Alignment (AL). The I-byte AL field is included to make the rest of the trailer 4 bytes long.
- Ending tag (Etag). The I-byte ET field serves as an ending flag. Its value is the same as that of the beginning tag.
- Length (L). The 2-byte L field indicates the length of the data unit.

The SAR header and trailer consist of five fields:

- Segment type (ST). The 2-bit ST identifier specifies the position of the segment in the message: beginning (00), middle (01), or end (10). A single-segment message has an ST of 11.
- Sequence number (SN). This field is the same as defined previously.
- Multiplexing identifier (MID). The 10-bit MID field identifies cells coming from different data flows and multiplexed on the same virtual connection.
- Length indicator (LI). This field defines how much of the packet is data, not padding.
- CRC. The last 10 bits of the trailer is a CRC for the entire data unit.

AALS AAL3/4 provides comprehensive sequencing and error control mechanisms that are not necessary for every application. For these applications, the designers of ATM have provided a fifth AAL sublayer, called the simple and efficient adaptation layer (SEAL). AALS 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. Figure 18.23 shows the AAL5 sublayer.

Figure 18.23 *AAL5*



The four trailer fields in the CS layer are

- User-to-user (UU). This field is used by end users, as described previously.
- Common part identifier (CPI). This field is the same as defined previously.
- Length (L). The 2-byte L field indicates the length of the original data.
- CRC. The last 4 bytes is for error control on the entire data unit.

Congestion Control and Quality of Service

ATM has a very developed congestion control and quality of service that we discuss in Chapter 24.

18.3 ATM LANs

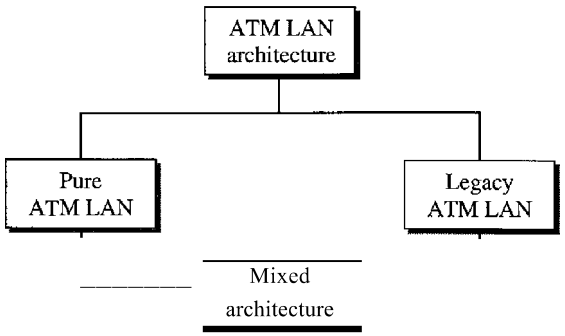
ATM is mainly a wide-area network (WAN ATM); however, the technology can be adapted to local-area networks (ATM LANs). The high data rate of the technology (155 and 622 Mbps) has attracted the attention of designers who are looking for greater and greater speeds in LANs. In addition, ATM technology has several advantages that make it an ideal LAN:

- ATM technology supports different types of connections between two end users. It supports permanent and temporary connections.
- ATM technology supports multimedia communication with a variety of bandwidths for different applications. It can guarantee a bandwidth of several megabits per second for real-time video. It can also provide support for text transfer during off-peak hours.
- An ATM LAN can be easily adapted for expansion in an organization.

ATM LAN Architecture

Today, we have two ways to incorporate ATM technology in a LAN architecture: creating a pure ATM LAN or making a legacy ATM LAN. Figure 18.24 shows the taxonomy.

Figure 18.24 *ATM LANs*



Pure ATM Architecture

In a pure ATM LAN, an ATM switch is used to connect the stations in a LAN, in exactly the same way stations are connected to an Ethernet switch. Figure 18.25 shows the situation.

In this way, stations can exchange data at one of two standard rates of ATM technology (155 and 652 Mbps). However, the station uses a virtual path identifier (VPI) and a virtual circuit identifier (Vel), instead of a source and destination address.

This approach has a major drawback. The system needs to be built from the ground up; existing LANs cannot be upgraded into pure ATM LANs.

Legacy LAN Architecture

A second approach is to use ATM technology as a backbone to connect traditional LANs. Figure 18.26 shows this architecture, a legacy ATM LAN.

Figure 18.25 *Pure ATM LAN*

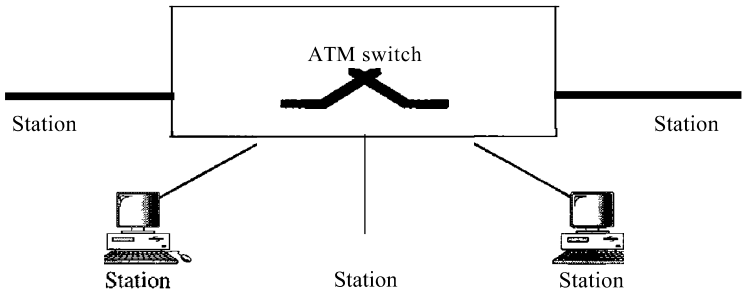
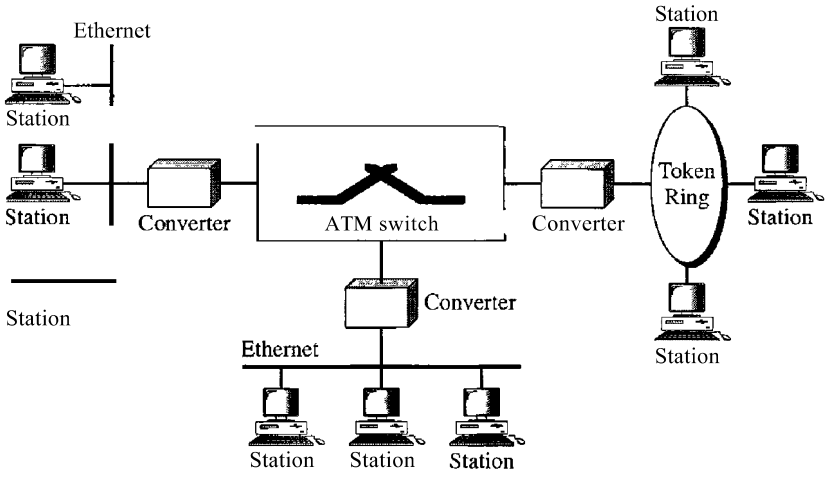


Figure 18.26 *Legacy ATM LAN*



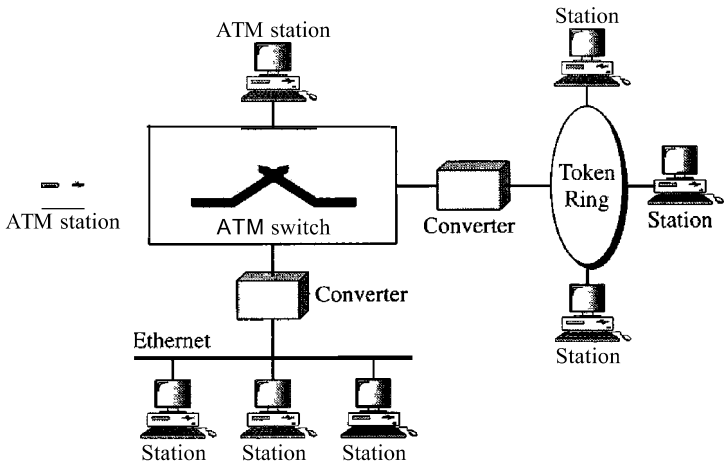
In this way, stations on the same LAN can exchange data at the rate and format of traditional LANs (Ethernet, Token Ring, etc.). But when two stations on two different LANs need to exchange data, they can go through a converting device that changes the frame format. The advantage here is that output from several LANs can be multiplexed together to create a high-data-rate input to the ATM switch. There are several issues that must be resolved first.

Mixed Architecture

Probably the best solution is to mix the two previous architectures. This means keeping the existing LANs and, at the same time, allowing new stations to be directly connected to an ATM switch. The mixed architecture LAN allows the gradual migration of legacy LANs onto ATM LANs by adding more and more directly connected stations to the switch. Figure 18.27 shows this architecture.

Again, the stations in one specific LAN can exchange data using the format and data rate of that particular LAN. The stations directly connected to the ATM switch can use an ATM frame to exchange data. However, the problem is, How can a station in a

Figure 18.27 *Mixed architecture ATM IAN*



traditional LAN communicate with a station directly connected to the ATM switch or vice versa? We see how the problem is resolved now.

LAN Emulation (LANE)

At the surface level, the use of ATM technology in LANs seems like a good idea. However, many issues need to be resolved, as summarized below:

- Connectionless versus connection-oriented. Traditional LANs, such as Ethernet, are *connectionless protocols.* A station sends data packets to another station whenever the packets are ready. There is no *connection establishment* or *connection termination* phase. On the other hand, ATM is a *connection-oriented protocol;* a station that wishes to send cells to another station must first establish a connection and, after all the cells are sent, terminate the connection.
- Physical addresses versus virtual-circuit identifiers. Closely related to the first issue is the difference in addressing. A connectionless protocol, such as Ethernet, defines the route of a packet through *source* and *destination addresses.* However, a connection-oriented protocol, such as ATM, defines the route of a cell through virtual connection identifiers (VPIs and VCIs).
- Multicasting and broadcasting delivery. Traditional LANs, such as Ethernet, can both *multicast* and *broadcast* packets; a station can send packets to a group of stations or to all stations. There is no easy way to multicast or broadcast on an ATM network although point-to-multipoint connections are available.
- Interoperability. In a mixed architecture, a station connected to a legacy LAN must be able to communicate with a station directly connected to an ATM switch.

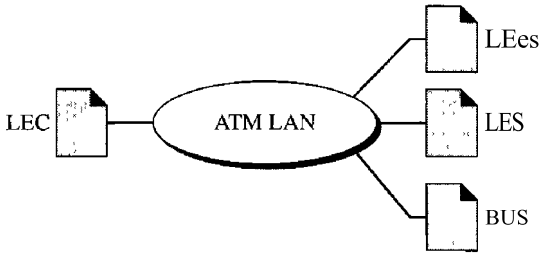
An approach called local-area network emulation (LANE) solves the above-mentioned problems and allows stations in a mixed architecture to communicate with one another. The approach uses emulation. Stations can use a connectionless service that emulates a connection-oriented service. Stations use the source and destination addresses for initial

connection and then use VPI and VCI addressing. The approach allows stations to use unicast, multicast, and broadcast addresses. Finally, the approach converts frames using a legacy format to ATM cells before they are sent through the switch.

Client/Server Model

LANE is designed as a client/server model to handle the four previously discussed problems. The protocol uses one type of client and three types of servers, as shown in Figure 18.28.

Figure 18.28 *Client and servers in a LANE*



LAN Emulation Client

All ATM stations have LAN emulation client (LEC) software installed on top of the three ATM protocols. The upper-layer protocols are unaware of the existence of the ATM technology. These protocols send their requests to LEC for a LAN service such as connectionless delivery using MAC unicast, multicast, or broadcast addresses. The LEC, however, just interprets the request and passes the result on to the servers.

LAN Emulation Configuration Server

The LAN emulation configuration server (LECS) is used for the initial connection between the client and LANE. This server is always waiting to receive the initial contact. It has a well-known ATM address that is known to every client in the system.

LAN Emulation Server

LAN emulation server (LES) software is installed on the LES. When a station receives a frame to be sent to another station using a physical address, LEC sends a special frame to the LES. The server creates a virtual circuit between the source and the destination station. The source station can now use this virtual circuit (and the corresponding identifier) to send the frame or frames to the destination.

Broadcast/Unknown Server

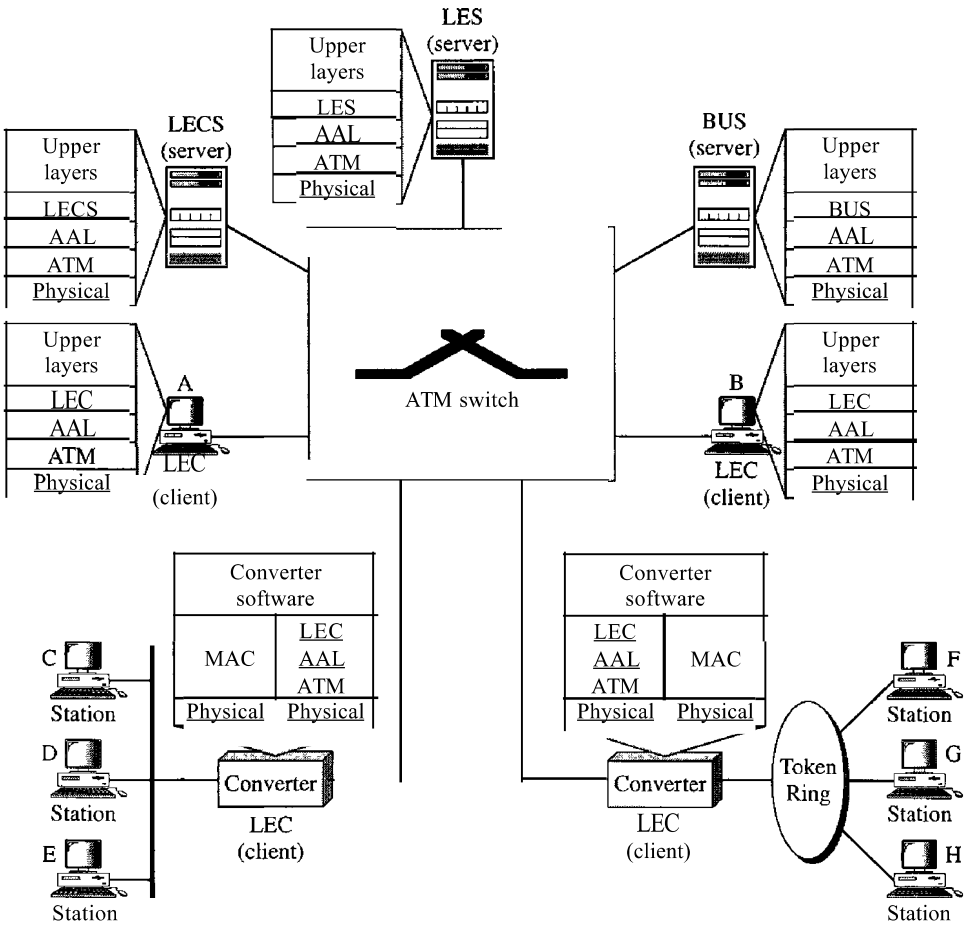
Multicasting and broadcasting require the use of another server called the broadcast! unknown server (BUS). If a station needs to send a frame to a group of stations or to every station, the frame first goes to the BUS; this server has permanent virtual connections to every station. The server creates copies of the received frame and sends a copy to a group of stations or to all stations, simulating a multicasting or broadcasting process.

The server can also deliver a unicast frame by sending the frame to every station. In this case the destination address is unknown. This is sometimes more efficient than getting the connection identifier from the LES.

Mixed Architecture with Client/Server

Figure 18.29 shows clients and servers in a mixed architecture ATM LAN. In the figure, three types of servers are connected to the ATM switch (they can actually be part of the switch). Also we show two types of clients. Stations A and B, designed to send and receive LANE communication, are directly connected to the ATM switch. Stations C, D, E, F, G, and H in traditional legacy LANs are connected to the switch via a converter. These converters act as LEC clients and communicate on behalf of their connected stations.

Figure 18.29 *A mixed architecture ATM LAN using LANE*



18.4 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [...] refer to the reference list at the end of the text.