PART

5

# *Transport Layer*

## Objectives

The transport layer is responsible for process-to-process delivery of the entire message. A process is an application program running on a host. Whereas the network layer oversees source-to-destination delivery of individual packets, it does not recognize any relationship between those packets. It treats each one independently, as though each piece belonged to a separate message, whether or not it does. The transport layer, on the other hand, ensures that the whole message arrives intact and in order, overseeing both error control and flow control at the source-to-destination level.

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**The transport layer is responsible for the delivery of a message from one process to another.**

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Computers often run several programs at the same time. For this reason, source-to-destination delivery means delivery not only from one computer to the next but also from a specific process on one computer to a specific process on the other. The transport layer header must therefore include a type of address called a *service-point address* in the OSI model and port number or port addresses in the Internet and TCP/IP protocol suite.

A transport layer protocol can be either connectionless or connection-oriented. A connectionless transport layer treats each segment as an independent packet and delivers it to the transport layer at the destination machine. A connection-oriented transport layer makes a connection with the transport layer at the destination machine first before delivering the packets. After all the data is transferred, the connection is terminated.

In the transport layer, a message is normally divided into transmittable segments. A connectionless protocol, such as UDP, treats each segment separately. A connection-oriented protocol, such as TCP and SCTP, creates a relationship between the segments using sequence numbers.

Like the data link layer, the transport layer may be responsible for flow and error control. However, flow and error control at this layer is performed end to end rather than across a single link. We will see that one of the protocols discussed in this part of

the book, UDP, is not involved in flow or error control. On the other hand, the other two protocols, TCP and SCTP, use sliding windows for flow control and an acknowledgment system for error control.

**Part 5 of the book is devoted to the transport layer and the services provided by this layer.**

## Chapters

This part consists of two chapters: Chapters 23 and 24.

### *Chapter 23*

Chapter 23 discusses three transport layer protocols in the Internet: UDP, TCP, and SCTP. The first, User Datagram Protocol (UDP), is a connectionless, unreliable protocol that is used for its efficiency. The second, Transmission Control Protocol (TCP), is a connection-oriented, reliable protocol that is a good choice for data transfer. The third, Stream Control Transport Protocol (SCTP) is a new transport-layer protocol designed for multimedia applications.

### *Chapter 24*

Chapter 24 discuss two related topics: congestion control and quality of service. Although these two issues can be related to any layer, we discuss them here with some references to other layers.

# CHAPTER 23

# *Process-to-Process Delivery: UDP, TCP, and SCTP*

We begin this chapter by giving the rationale for the existence of the **transport layer**—the need for process-to-process delivery. We discuss the issues arising from this type of delivery, and we discuss methods to handle them.

The Internet model has three protocols at the transport layer: UDP, TCP, and SCTP. First we discuss UDP, which is the simplest of the three. We see how we can use this very simple transport layer protocol that lacks some of the features of the other two.

We then discuss TCP, a complex transport layer protocol. We see how our previously presented concepts are applied to TCP. We postpone the discussion of congestion control and quality of service in TCP until Chapter 24 because these two topics apply to the data link layer and network layer as well.

We finally discuss SCTP, the new transport layer protocol that is designed for multihomed, multistream applications such as multimedia.

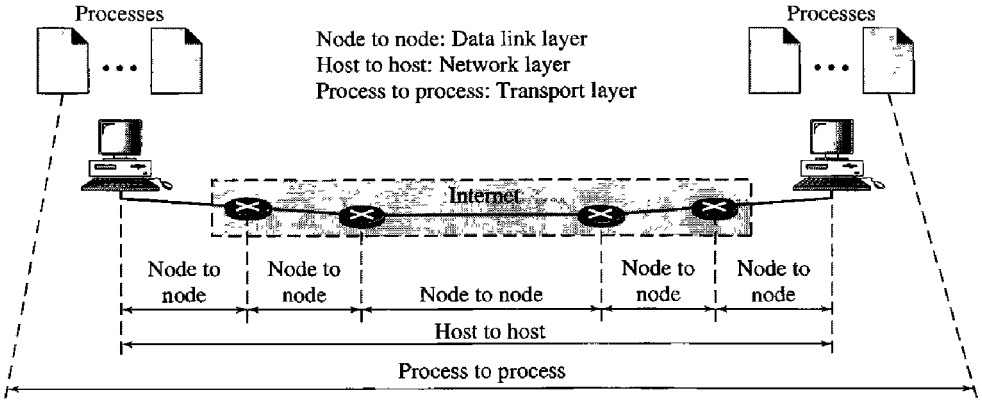
## 23.1 PROCESS-TO-PROCESS DELIVERY

The data link layer is responsible for delivery of frames between two neighboring nodes over a link. This is called *node-to-node delivery.* The network layer is responsible for delivery of datagrams between two hosts. This is called *host-to-host delivery.* Communication on the Internet is not defined as the exchange of data between two nodes or between two hosts. Real communication takes place between two processes (application programs). We need **process-to-process delivery.** However, at any moment, several processes may be running on the source host and several on the destination host. To complete the delivery, we need a mechanism to deliver data from one of these processes running on the source host to the corresponding process running on the destination host.

The transport layer is responsible for process-to-process delivery—the delivery of a packet, part of a message, from one process to another. Two processes communicate in a client/server relationship, as we will see later. Figure 23.1 shows these three types of deliveries and their domains.

**The transport layer is responsible for process-to-process delivery.**

**Figure 23.1** *Types of data deliveries*




## Client/Server Paradigm

Although there are several ways to achieve process-to-process communication, the most common one is through the **client/server paradigm.** A process on the local host, called a **client,** needs services from a process usually on the remote host, called a **server.**

Both processes (client and server) have the same name. For example, to get the day and time from a remote machine, we need a Daytime client process running on the local host and a Daytime server process running on a remote machine.

Operating systems today support both multiuser and multiprogramming environments. A remote computer can run several server programs at the same time, just as local computers can run one or more client programs at the same time. For communication, we must define the following:

1. Local host
2. Local process
3. Remote host
4. Remote process

### *Addressing*

Whenever we need to deliver something to one specific destination among many, we need an address. At the data link layer, we need a MAC address to choose one node among several nodes if the connection is not point-to-point. A frame in the data link layer needs a destination MAC address for delivery and a source address for the next node's reply.

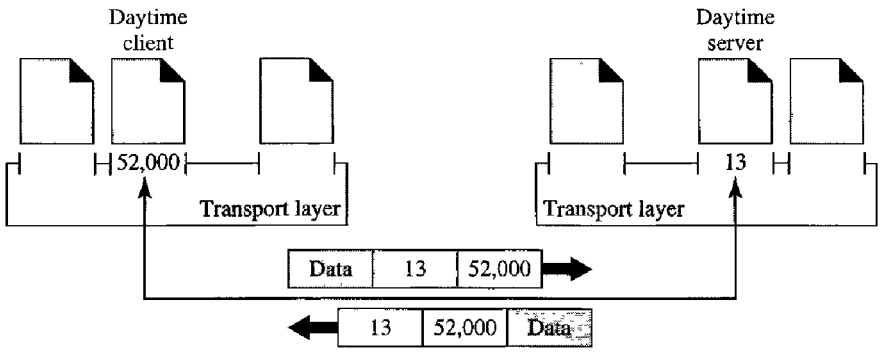
At the network layer, we need an IP address to choose one host among millions. A datagram in the network layer needs a destination IP address for delivery and a source IP address for the destination's reply.

At the transport layer, we need a transport layer address, called a **port number,** to choose among multiple processes running on the destination host. The destination port number is needed for delivery; the source port number is needed for the reply.

In the Internet model, the port numbers are 16-bit integers between 0 and 65,535. The client program defines itself with a port number, chosen randomly by the transport layer software running on the client host. This is the **ephemeral port number.**

The server process must also define itself with a port number. This port number, however, cannot be chosen randomly. If the computer at the server site runs a server process and assigns a random number as the port number, the process at the client site that wants to access that server and use its services will not know the port number. Of course, one solution would be to send a special packet and request the port number of a specific server, but this requires more overhead. The Internet has decided to use universal port numbers for servers; these are called **well-known port numbers.** There are some exceptions to this rule; for example, there are clients that are assigned well-known port numbers. Every client process knows the well-known port number of the corresponding server process. For example, while the Daytime client process, discussed above, can use an ephemeral (temporary) port number 52,000 to identify itself, the Daytime server process must use the well-known (permanent) port number 13. Figure 23.2 shows this concept.

**Figure 23.2** *Port numbers*




It should be clear by now that the IP addresses and port numbers play different roles in selecting the final destination of data. The destination IP address defines the host among the different hosts in the world. After the host has been selected, the port number defines one of the processes on this particular host (see Figure 23.3).

### *IANA Ranges*

The IANA (Internet Assigned Number Authority) has divided the port numbers into three ranges: well known, registered, and dynamic (or private), as shown in Figure 23.4.

- **Well-known ports.** The ports ranging from 0 to 1023 are assigned and controlled by IANA. These are the well-known ports.
- **Registered ports.** The ports ranging from 1024 to 49,151 are not assigned or controlled by IANA. They can only be registered with IANA to prevent duplication.
- **Dynamic ports.** The ports ranging from 49,152 to 65,535 are neither controlled nor registered. They can be used by any process. These are the ephemeral ports.

**Figure 23.3** *IP addresses versus port numbers*

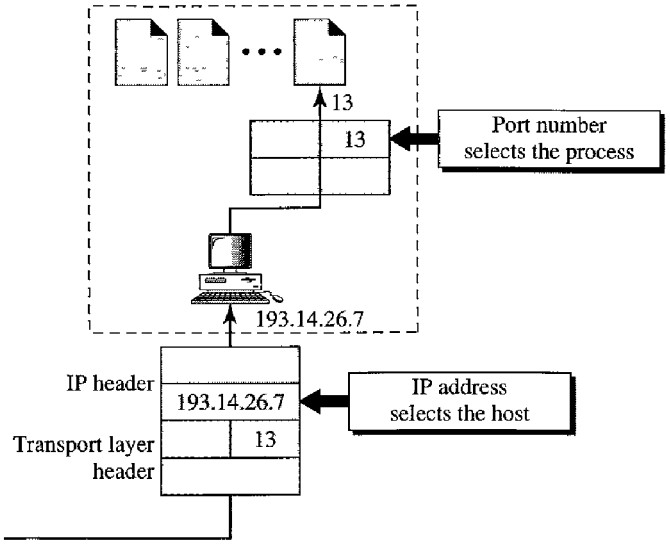
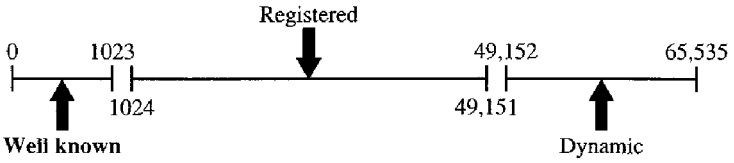



**Figure 23.4** *IANA ranges*

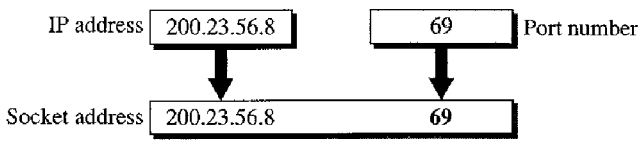



### *Socket Addresses*

Process-to-process delivery needs two identifiers, IP address and the port number, at each end to make a connection. The combination of an IP address and a port number is called a **socket address.** The client socket address defines the client process uniquely just as the server socket address defines the server process uniquely (see Figure 23.5).

A transport layer protocol needs a pair of socket addresses: the client socket address and the server socket address. These four pieces of information are part of the IP header and the transport layer protocol header. The IP header contains the IP addresses; the UDP or TCP header contains the port numbers.

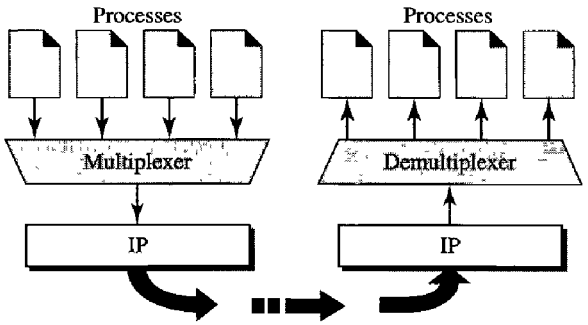
**Figure 23.5** *Socket address*



## Multiplexing and Demultiplexing

The addressing mechanism allows multiplexing and demultiplexing by the transport layer, as shown in Figure 23.6.

**Figure 23.6** *Multiplexing and demultiplexing*




### *Multiplexing*

At the sender site, there may be several processes that need to send packets. However, there is only one transport layer protocol at any time. This is a many-to-one relationship and requires multiplexing. The protocol accepts messages from different processes, differentiated by their assigned port numbers. After adding the header, the transport layer passes the packet to the network layer.

### *Demultiplexing*

At the receiver site, the relationship is one-to-many and requires demultiplexing. The transport layer receives datagrams from the network layer. After error checking and dropping of the header, the transport layer delivers each message to the appropriate process based on the port number.

## Connectionless Versus Connection-Oriented Service

A transport layer protocol can either be connectionless or connection-oriented.

### *Connectionless Service*

In a **connectionless service,** the packets are sent from one party to another with no need for connection establishment or connection release. The packets are not numbered; they may be delayed or lost or may arrive out of sequence. There is no acknowledgment either. We will see shortly that one of the transport layer protocols in the Internet model, UDP, is connectionless.

### *Connection-Oriented Service*

In a **connection-oriented service,** a connection is first established between the sender and the receiver. Data are transferred. At the end, the connection is released. We will see shortly that TCP and SCTP are connection-oriented protocols.

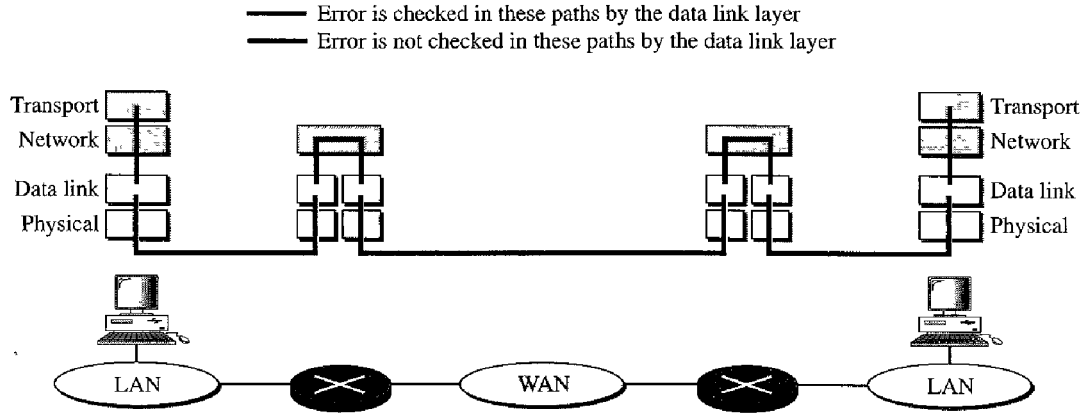
## Reliable Versus Unreliable

The transport layer service can be reliable or unreliable. If the application layer program needs reliability, we use a reliable transport layer protocol by implementing flow and error control at the transport layer. This means a slower and more complex service. On the other hand, if the application program does not need reliability because it uses its own flow and error control mechanism or it needs fast service or the nature of the service does not demand flow and error control (real-time applications), then an unreliable protocol can be used.

In the Internet, there are three common different transport layer protocols, as we have already mentioned. UDP is connectionless and unreliable; TCP and SCTP are connection-oriented and reliable. These three can respond to the demands of the application layer programs.

One question often comes to the mind. If the data link layer is reliable and has flow and error control, do we need this at the transport layer, too? The answer is yes. Reliability at the data link layer is between two nodes; we need reliability between two ends. Because the network layer in the Internet is unreliable (best-effort delivery), we need to implement reliability at the transport layer. To understand that error control at the data link layer does not guarantee error control at the transport layer, let us look at Figure 23.7.

**Figure 23.7** *Error control*

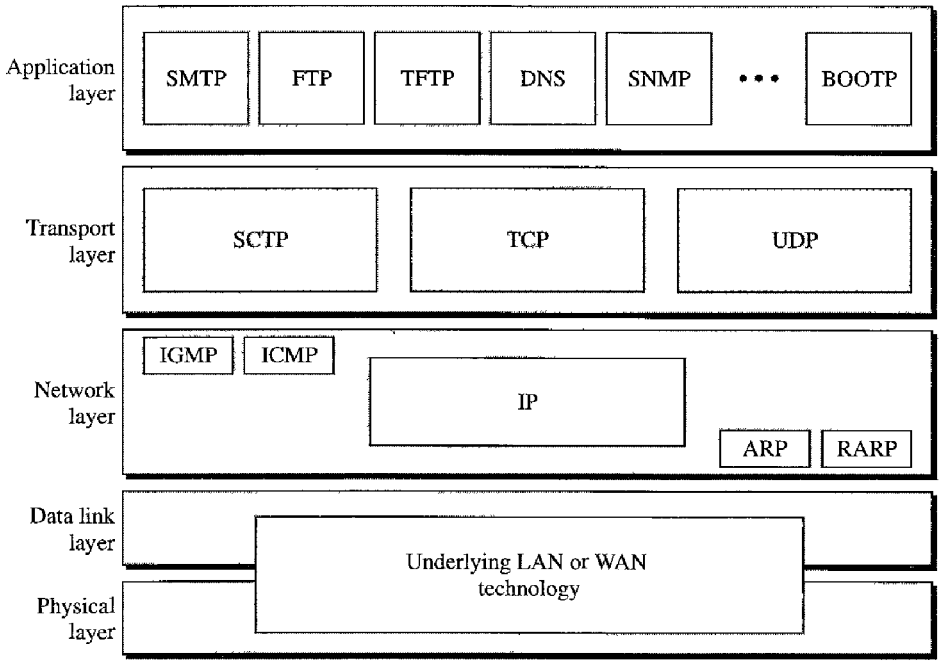



As we will see, flow and error control in TCP is implemented by the sliding window protocol, as discussed in Chapter 11. The window, however, is character-oriented, instead of frame-oriented.

## Three Protocols

The original TCP/IP protocol suite specifies two protocols for the transport layer: UDP and TCP. We first focus on UDP, the simpler of the two, before discussing TCP. A new transport layer protocol, SCTP, has been designed, which we also discuss in this chapter. Figure 23.8 shows the position of these protocols in the TCP/IP protocol suite.

**Figure 23.8** *Position of UDP, TCP, and SCTP in TCP/IP suite*




## 23.2 USER DATAGRAM PROTOCOL (UDP)

The **User Datagram Protocol (UDP)** is called a **connectionless, unreliable transport protocol.** It does not add anything to the services of IP except to provide process-to-process communication instead of host-to-host communication. Also, it performs very limited error checking.

If UDP is so powerless, why would a process want to use it? With the disadvantages come some advantages. UDP is a very simple protocol using a minimum of overhead. If a process wants to send a small message and does not care much about reliability, it can use UDP. Sending a small message by using UDP takes much less interaction between the sender and receiver than using TCP or SCTP.

### Well-Known Ports for UDP

Table 23.1 shows some well-known port numbers used by UDP. Some port numbers can be used by both UDP and TCP. We discuss them when we talk about TCP later in the chapter.

**Table 23.1** *Well-known ports used with UDP*

| *Port* | *Protocol* | *Description* |
|---|---|---|
| 7 | Echo | Echoes a received datagram back to the sender |
| 9 | Discard | Discards any datagram that is received |
| 11 | Users | Active users |

**Table 23.1** *Well-known ports used with UDP (continued)*

| *Port* | *Protocol* | *Description* |
|---|---|---|
| 13 | Daytime | Returns the date and the time |
| 17 | Quote | Returns a quote of the day |
| 19 | Chargen | Returns a string of characters |
| 53 | Nameserver | Domain Name Service |
| 67 | BOOTPs | Server port to download bootstrap information |
| 68 | BOOTPc | Client port to download bootstrap information |
| 69 | TFTP | Trivial File Transfer Protocol |
| 111 | RPC | Remote Procedure Call |
| 123 | NTP | Network Time Protocol |
| 161 | SNMP | Simple Network Management Protocol |
| 162 | SNMP | Simple Network Management Protocol (trap) |

***Example 23.1***

In UNIX, the well-known ports are stored in a file called /etc/services. Each line in this file gives the name of the server and the well-known port number. We can use the *grep* utility to extract the line corresponding to the desired application. The following shows the port for FTP. Note that FTP can use port 21 with either UDP or TCP.

```
$ grep   ftp   /etc/services
ftp        21/tcp
ftp        21/udp
```

SNMP uses two port numbers (161 and 162), each for a different purpose, as we will see in Chapter 28.

```
$ grep    snmp /etc/services
snmp         161/tcp        #Simple Net  Mgmt Proto
snmp         161/udp        #Simple Net  Mgmt Proto
snmptrap     162/udp        #Traps for SNMP
```

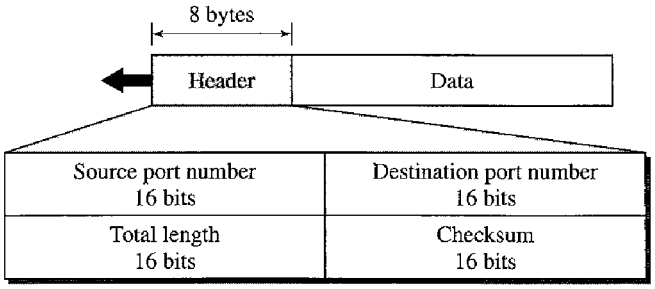
## User Datagram

UDP packets, called **user datagrams,** have a fixed-size header of 8 bytes. Figure 23.9 shows the format of a user datagram.

The fields are as follows:

- **Source port number.** This is the port number used by the process running on the source host. It is 16 bits long, which means that the port number can range from 0 to 65,535. If the source host is the client (a client sending a request), the port number, in most cases, is an ephemeral port number requested by the process and chosen by the UDP software running on the source host. If the source host is the server (a server sending a response), the port number, in most cases, is a well-known port number.

**Figure 23.9** *User datagram format*




- **Destination port number.** This is the port number used by the process running on the destination host. It is also 16 bits long. If the destination host is the server (a client sending a request), the port number, in most cases, is a well-known port number. If the destination host is the client (a server sending a response), the port number, in most cases, is an ephemeral port number. In this case, the server copies the ephemeral port number it has received in the request packet.
- **Length.** This is a 16-bit field that defines the total length of the user datagram, header plus data. The 16 bits can define a total length of 0 to 65,535 bytes. However, the total length needs to be much less because a UDP user datagram is stored in an IP datagram with a total length of 65,535 bytes.

  The length field in a UDP user datagram is actually not necessary. A user datagram is encapsulated in an IP datagram. There is a field in the IP datagram that defines the total length. There is another field in the IP datagram that defines the length of the header. So if we subtract the value of the second field from the first, we can deduce the length of a UDP datagram that is encapsulated in an IP datagram.

**UDP length = IP length – IP header's length**

  However, the designers of the UDP protocol felt that it was more efficient for the destination UDP to calculate the length of the data from the information provided in the UDP user datagram rather than ask the IP software to supply this information. We should remember that when the IP software delivers the UDP user datagram to the UDP layer, it has already dropped the IP header.
- **Checksum.** This field is used to detect errors over the entire user datagram (header plus data). The checksum is discussed next.

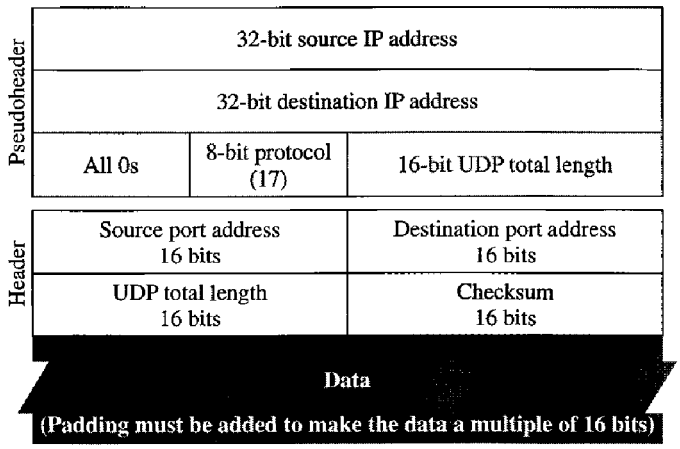
## Checksum

We have already talked about the concept of the checksum and the way it is calculated in Chapter 10. We have also shown how to calculate the checksum for the IP and ICMP packet. We now show how this is done for UDP.

The UDP checksum calculation is different from the one for IP and ICMP. Here the checksum includes three sections: a pseudoheader, the UDP header, and the data coming from the application layer.

The **pseudoheader** is the part of the header of the IP packet in which the user datagram is to be encapsulated with some fields filled with 0s (see Figure 23.10).

**Figure 23.10** *Pseudoheader for checksum calculation*




If the checksum does not include the pseudoheader, a user datagram may arrive safe and sound. However, if the IP header is corrupted, it may be delivered to the wrong host.

The protocol field is added to ensure that the packet belongs to UDP, and not to other transport-layer protocols. We will see later that if a process can use either UDP or TCP, the destination port number can be the same. The value of the protocol field for UDP is 17. If this value is changed during transmission, the checksum calculation at the receiver will detect it and UDP drops the packet. It is not delivered to the wrong protocol.

Note the similarities between the pseudoheader fields and the last 12 bytes of the IP header.

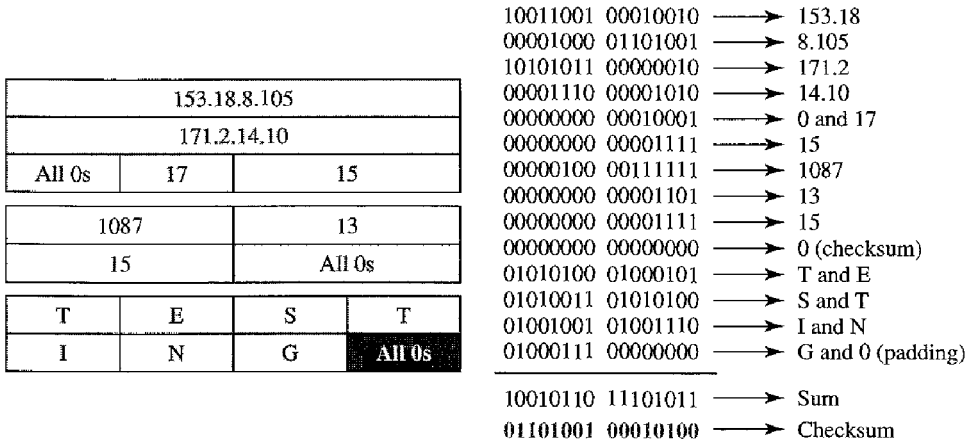
### *Example 23.2*

Figure 23.11 shows the checksum calculation for a very small user datagram with only 7 bytes of data. Because the number of bytes of data is odd, padding is added for checksum calculation. The pseudoheader as well as the padding will be dropped when the user datagram is delivered to IP.

### *Optional Use of the Checksum*

The calculation of the checksum and its inclusion in a user datagram are optional. If the checksum is not calculated, the field is filled with 1s. Note that a calculated checksum can never be all 1s because this implies that the sum is all 0s, which is impossible because it requires that the value of fields to be 0s.

**Figure 23.11** *Checksum calculation of a simple UDP user datagram*




## UDP Operation

UDP uses concepts common to the transport layer. These concepts will be discussed here briefly, and then expanded in the next section on the TCP protocol.

### *Connectionless Services*

As mentioned previously, UDP provides a connectionless service. This means that each user datagram sent by UDP is an independent datagram. There is no relationship between the different user datagrams even if they are coming from the same source process and going to the same destination program. The user datagrams are not numbered. Also, there is no connection establishment and no connection termination, as is the case for TCP. This means that each user datagram can travel on a different path.

One of the ramifications of being connectionless is that the process that uses UDP cannot send a stream of data to UDP and expect UDP to chop them into different related user datagrams. Instead each request must be small enough to fit into one user datagram. Only those processes sending short messages should use UDP.

### *Flow and Error Control*

UDP is a very simple, unreliable transport protocol. There is no flow control and hence no window mechanism. The receiver may overflow with incoming messages.

There is no error control mechanism in UDP except for the checksum. This means that the sender does not know if a message has been lost or duplicated. When the receiver detects an error through the checksum, the user datagram is silently discarded.

The lack of **flow control** and **error control** means that the process using UDP should provide these mechanisms.

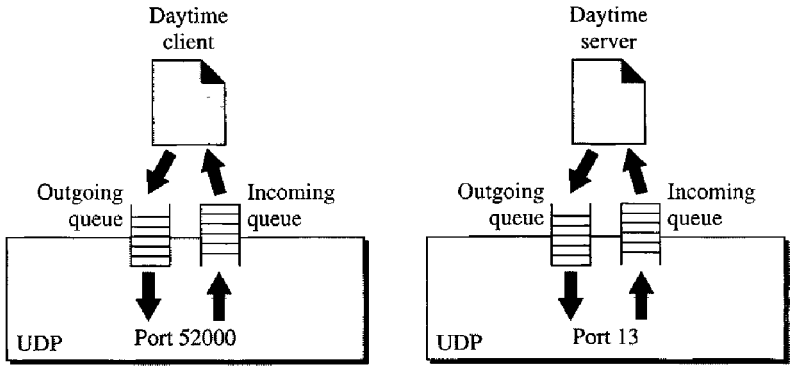
### *Encapsulation and Decapsulation*

To send a message from one process to another, the UDP protocol encapsulates and decapsulates messages in an IP datagram.

### *Queuing*

We have talked about ports without discussing the actual implementation of them. In UDP, queues are associated with ports (see Figure 23.12).

**Figure 23.12** *Queues in UDP*




At the client site, when a process starts, it requests a port number from the operating system. Some implementations create both an incoming and an outgoing queue associated with each process. Other implementations create only an incoming queue associated with each process.

Note that even if a process wants to communicate with multiple processes, it obtains only one port number and eventually one outgoing and one incoming **queue.** The queues opened by the client are, in most cases, identified by ephemeral port numbers. The queues function as long as the process is running. When the process terminates, the queues are destroyed.

The client process can send messages to the outgoing queue by using the source port number specified in the request. UDP removes the messages one by one and, after adding the UDP header, delivers them to IP. An outgoing queue can overflow. If this happens, the operating system can ask the client process to wait before sending any more messages.

When a message arrives for a client, UDP checks to see if an incoming queue has been created for the port number specified in the destination port number field of the user datagram. If there is such a queue, UDP sends the received user datagram to the end of the queue. If there is no such queue, UDP discards the user datagram and asks the ICMP protocol to send a *port unreachable* message to the server. All the incoming messages for one particular client program, whether coming from the same or a different server, are sent to the same queue. An incoming queue can overflow. If this happens, UDP drops the user datagram and asks for a port unreachable message to be sent to the server.

At the server site, the mechanism of creating queues is different. In its simplest form, a server asks for incoming and outgoing queues, using its well-known port, when it starts running. The queues remain open as long as the server is running.

When a message arrives for a server, UDP checks to see if an incoming queue has been created for the port number specified in the destination port number field of the user

datagram. If there is such a queue, UDP sends the received user datagram to the end of the queue. If there is no such queue, UDP discards the user datagram and asks the ICMP protocol to send a port unreachable message to the client. All the incoming messages for one particular server, whether coming from the same or a different client, are sent to the same queue. An incoming queue can overflow. If this happens, UDP drops the user datagram and asks for a port unreachable message to be sent to the client.

When a server wants to respond to a client, it sends messages to the outgoing queue, using the source port number specified in the request. UDP removes the messages one by one and, after adding the UDP header, delivers them to IP. An outgoing queue can overflow. If this happens, the operating system asks the server to wait before sending any more messages.

### Use of UDP

The following lists some uses of the UDP protocol:

- UDP is suitable for a process that requires simple request-response communication with little concern for flow and error control. It is not usually used for a process such as FTP that needs to send bulk data (see Chapter 26).
- UDP is suitable for a process with internal flow and error control mechanisms. For example, the Trivial File Transfer Protocol (TFTP) process includes flow and error control. It can easily use UDP.
- UDP is a suitable transport protocol for multicasting. Multicasting capability is embedded in the UDP software but not in the TCP software.
- UDP is used for management processes such as SNMP (see Chapter 28).
- UDP is used for some route updating protocols such as Routing Information Protocol (RIP) (see Chapter 22).

## 23.3 TCP

The second transport layer protocol we discuss in this chapter is called **Transmission Control Protocol (TCP).** TCP, like UDP, is a process-to-process (program-to-program) protocol. TCP, therefore, like UDP, uses port numbers. Unlike UDP, TCP is a connection-oriented protocol; it creates a virtual connection between two TCPs to send data. In addition, TCP uses flow and error control mechanisms at the transport level.

In brief, TCP is called a *connection-oriented, reliable* transport protocol. It adds connection-oriented and reliability features to the services of IP.

### TCP Services

Before we discuss TCP in detail, let us explain the services offered by TCP to the processes at the application layer.

#### *Process-to-Process Communication*

Like UDP, TCP provides process-to-process communication using port numbers. Table 23.2 lists some well-known port numbers used by TCP.

**Table 23.2** *Well-known ports used by TCP*

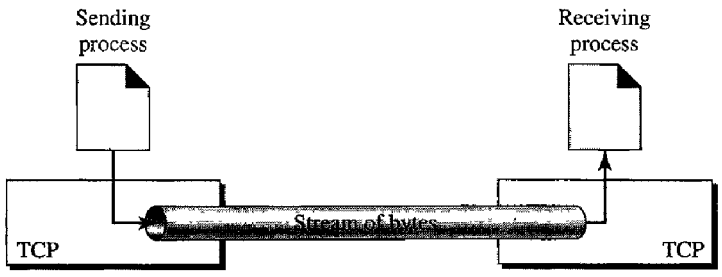
| *Port* | *Protocol* | *Description* |
|---|---|---|
| 7 | Echo | Echoes a received datagram back to the sender |
| 9 | Discard | Discards any datagram that is received |
| 11 | Users | Active users |
| 13 | Daytime | Returns the date and the time |
| 17 | Quote | Returns a quote of the day |
| 19 | Chargen | Returns a string of characters |
| 20 | FTP, Data | File Transfer Protocol (data connection) |
| 21 | FTP, Control | File Transfer Protocol (control connection) |
| 23 | TELNET | Terminal Network |
| 25 | SMTP | Simple Mail Transfer Protocol |
| 53 | DNS | Domain Name Server |
| 67 | BOOTP | Bootstrap Protocol |
| 79 | Finger | Finger |
| 80 | HTTP | Hypertext Transfer Protocol |
| 111 | RPC | Remote Procedure Call |

### *Stream Delivery Service*

TCP, unlike UDP, is a stream-oriented protocol. In UDP, a process (an application program) sends messages, with predefined boundaries, to UDP for delivery. UDP adds its own header to each of these messages and delivers them to IP for transmission. Each message from the process is called a user datagram and becomes, eventually, one IP datagram. Neither IP nor UDP recognizes any relationship between the datagrams.

TCP, on the other hand, allows the sending process to deliver data as a stream of bytes and allows the receiving process to obtain data as a stream of bytes. TCP creates an environment in which the two processes seem to be connected by an imaginary "tube" that carries their data across the Internet. This imaginary environment is depicted in Figure 23.13. The sending process produces (writes to) the stream of bytes, and the receiving process consumes (reads from) them.

**Figure 23.13** *Stream delivery*



**Sending and Receiving Buffers** Because the sending and the receiving processes may not write or read data at the same speed, TCP needs buffers for storage. There are two buffers, the sending buffer and the receiving buffer, one for each direction. (We will see later that these buffers are also necessary for flow and error control mechanisms used by TCP.) One way to implement a buffer is to use a circular array of 1-byte locations as shown in Figure 23.14. For simplicity, we have shown two buffers of 20 bytes each; normally the buffers are hundreds or thousands of bytes, depending on the implementation. We also show the buffers as the same size, which is not always the case.

**Figure 23.14** *Sending and receiving buffers*

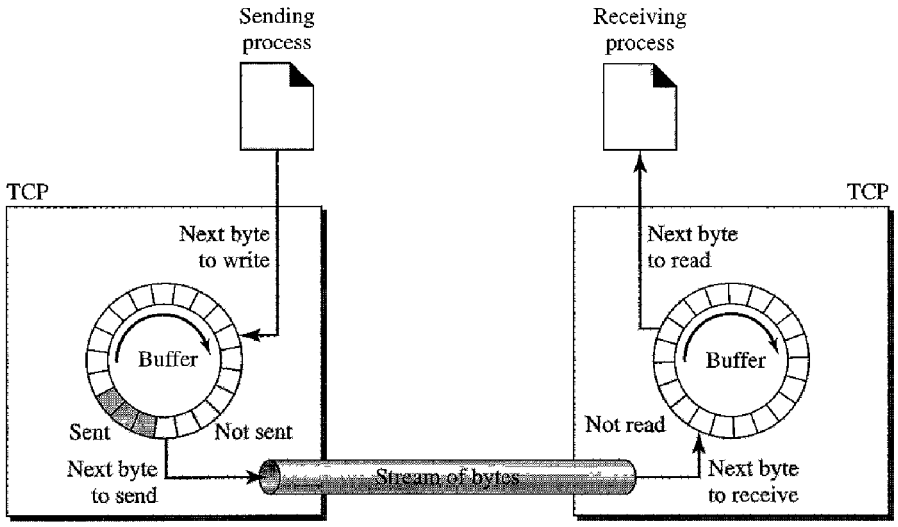


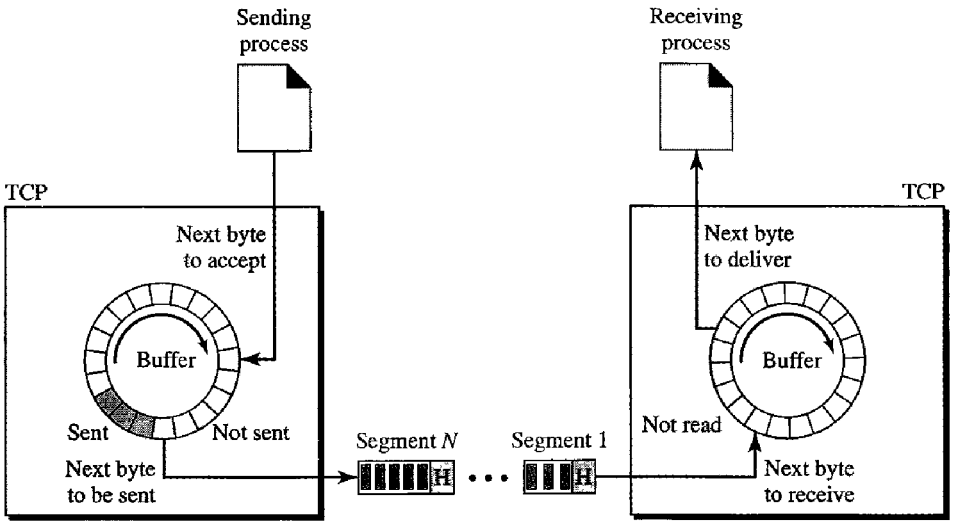

Figure 23.14 shows the movement of the data in one direction. At the sending site, the buffer has three types of chambers. The white section contains empty chambers that can be filled by the sending process (producer). The gray area holds bytes that have been sent but not yet acknowledged. TCP keeps these bytes in the buffer until it receives an acknowledgment. The colored area contains bytes to be sent by the sending TCP. However, as we will see later in this chapter, TCP may be able to send only part of this colored section. This could be due to the slowness of the receiving process or perhaps to congestion in the network. Also note that after the bytes in the gray chambers are acknowledged, the chambers are recycled and available for use by the sending process. This is why we show a circular buffer.

The operation of the buffer at the receiver site is simpler. The circular buffer is divided into two areas (shown as white and colored). The white area contains empty chambers to be filled by bytes received from the network. The colored sections contain received bytes that can be read by the receiving process. When a byte is read by the receiving process, the chamber is recycled and added to the pool of empty chambers.

**Segments** Although buffering handles the disparity between the speed of the producing and consuming processes, we need one more step before we can send data. The IP layer, as a service provider for TCP, needs to send data in packets, not as a stream of bytes. At

the transport layer, TCP groups a number of bytes together into a packet called a **segment.** TCP adds a header to each segment (for control purposes) and delivers the segment to the IP layer for transmission. The segments are encapsulated in IP datagrams and transmitted. This entire operation is transparent to the receiving process. Later we will see that segments may be received out of order, lost, or corrupted and resent. All these are handled by TCP with the receiving process unaware of any activities. Figure 23.15 shows how segments are created from the bytes in the buffers.

**Figure 23.15** *TCP segments*




Note that the segments are not necessarily the same size. In Figure 23.15, for simplicity, we show one segment carrying 3 bytes and the other carrying 5 bytes. In reality, segments carry hundreds, if not thousands, of bytes.

### *Full-Duplex Communication*

TCP offers **full-duplex service,** in which data can flow in both directions at the same time. Each TCP then has a sending and receiving buffer, and segments move in both directions.

### *Connection-Oriented Service*

TCP, unlike UDP, is a connection-oriented protocol. When a process at site A wants to send and receive data from another process at site B, the following occurs:

1. The two TCPs establish a connection between them.
2. Data are exchanged in both directions.
3. The connection is terminated.

Note that this is a virtual connection, not a physical connection. The TCP segment is encapsulated in an IP datagram and can be sent out of order, or lost, or corrupted, and then resent. Each may use a different path to reach the destination. There is no physical connection. TCP creates a stream-oriented environment in which it accepts the responsibility of

delivering the bytes in order to the other site. The situation is similar to creating a bridge that spans multiple islands and passing all the bytes from one island to another in one single connection. We will discuss this feature later in the chapter.

### *Reliable Service*

TCP is a reliable transport protocol. It uses an acknowledgment mechanism to check the safe and sound arrival of data. We will discuss this feature further in the section on error control.

## TCP Features

To provide the services mentioned in the previous section, TCP has several features that are briefly summarized in this section and discussed later in detail.

### *Numbering System*

Although the TCP software keeps track of the segments being transmitted or received, there is no field for a segment number value in the segment header. Instead, there are two fields called the **sequence number** and the **acknowledgment number.** These two fields refer to the byte number and not the segment number.

**Byte Number** TCP numbers all data bytes that are transmitted in a connection. Numbering is independent in each direction. When TCP receives bytes of data from a process, it stores them in the sending buffer and numbers them. The numbering does not necessarily start from 0. Instead, TCP generates a random number between 0 and $2^{32} - 1$ for the number of the first byte. For example, if the random number happens to be 1057 and the total data to be sent are 6000 bytes, the bytes are numbered from 1057 to 7056. We will see that byte numbering is used for flow and error control.

---

**The bytes of data being transferred in each connection are numbered by TCP. The numbering starts with a randomly generated number.**

---

**Sequence Number** After the bytes have been numbered, TCP assigns a sequence number to each segment that is being sent. The sequence number for each segment is the number of the first byte carried in that segment.

### *Example 23.3*

Suppose a TCP connection is transferring a file of 5000 bytes. The first byte is numbered 10,001. What are the sequence numbers for each segment if data are sent in five segments, each carrying 1000 bytes?

### Solution

The following shows the sequence number for each segment:

**Segment 1 ➡ Sequence Number: 10,001 (range: 10,001 to 11,000)**
**Segment 2 ➡ Sequence Number: 11,001 (range: 11,001 to 12,000)**
**Segment 3 ➡ Sequence Number: 12,001 (range: 12,001 to 13,000)**
**Segment 4 ➡ Sequence Number: 13,001 (range: 13,001 to 14,000)**
**Segment 5 ➡ Sequence Number: 14,001 (range: 14,001 to 15,000)**

**The value in the sequence number field of a segment defines the number of the first data byte contained in that segment.**

When a segment carries a combination of data and control information (piggybacking), it uses a sequence number. If a segment does not carry user data, it does not logically define a sequence number. The field is there, but the value is not valid. However, some segments, when carrying only control information, need a sequence number to allow an acknowledgment from the receiver. These segments are used for connection establishment, termination, or abortion. Each of these segments consumes one sequence number as though it carried 1 byte, but there are no actual data. If the randomly generated sequence number is $x$, the first data byte is numbered $x + 1$. The byte $x$ is considered a phony byte that is used for a control segment to open a connection, as we will see shortly.

**Acknowledgment Number** As we discussed previously, communication in TCP is full duplex; when a connection is established, both parties can send and receive data at the same time. Each party numbers the bytes, usually with a different starting byte number. The sequence number in each direction shows the number of the first byte carried by the segment. Each party also uses an acknowledgment number to confirm the bytes it has received. However, the acknowledgment number defines the number of the next byte that the party expects to receive. In addition, the acknowledgment number is cumulative, which means that the party takes the number of the last byte that it has received, safe and sound, adds 1 to it, and announces this sum as the acknowledgment number. The term *cumulative* here means that if a party uses 5643 as an acknowledgment number, it has received all bytes from the beginning up to 5642. Note that this does not mean that the party has received 5642 bytes because the first byte number does not have to start from 0.

**The value of the acknowledgment field in a segment defines the number of the next byte a party expects to receive. The acknowledgment number is cumulative.**

### *Flow Control*

TCP, unlike UDP, provides *flow control*. The receiver of the data controls the amount of data that are to be sent by the sender. This is done to prevent the receiver from being overwhelmed with data. The numbering system allows TCP to use a byte-oriented flow control.

### *Error Control*

To provide reliable service, TCP implements an error control mechanism. Although error control considers a segment as the unit of data for error detection (loss or corrupted segments), error control is byte-oriented, as we will see later.

### *Congestion Control*

TCP, unlike UDP, takes into account congestion in the network. The amount of data sent by a sender is not only controlled by the receiver (flow control), but is also determined by the level of congestion in the network.

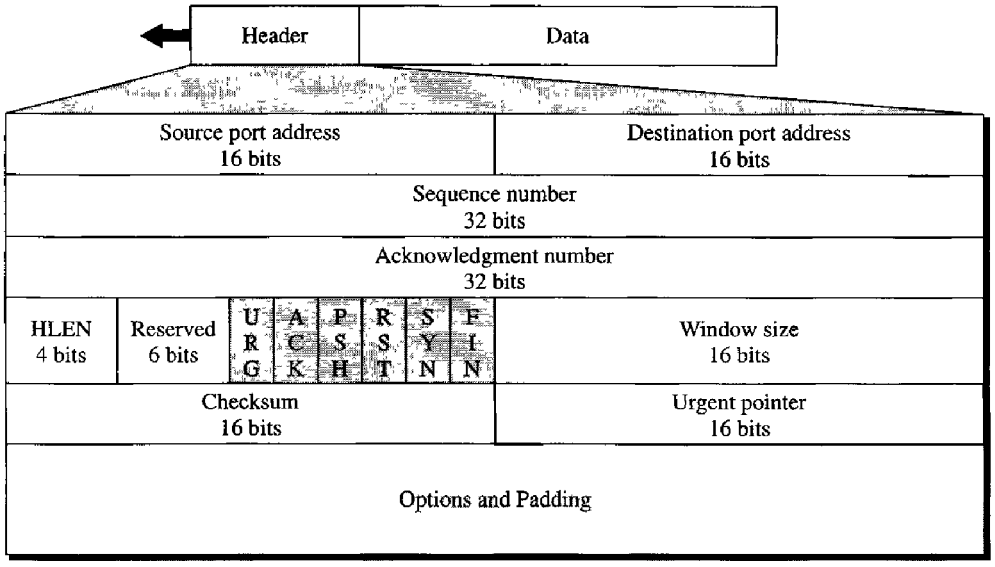
## Segment

Before we discuss TCP in greater detail, let us discuss the TCP packets themselves. A packet in TCP is called a **segment.**

### *Format*

The format of a segment is shown in Figure 23.16.

**Figure 23.16** *TCP segment format*



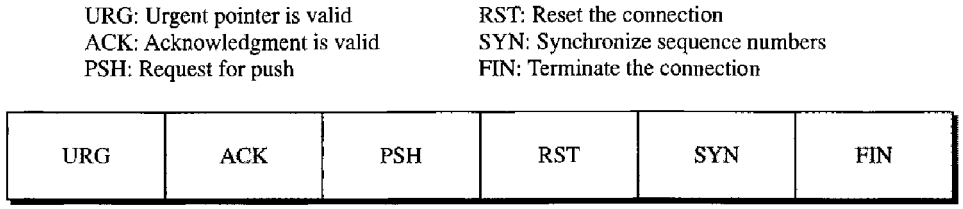

The segment consists of a 20- to 60-byte header, followed by data from the application program. The header is 20 bytes if there are no options and up to 60 bytes if it contains options. We will discuss some of the header fields in this section. The meaning and purpose of these will become clearer as we proceed through the chapter.

- **Source port address.** This is a 16-bit field that defines the port number of the application program in the host that is sending the segment. This serves the same purpose as the source port address in the UDP header.
- **Destination port address.** This is a 16-bit field that defines the port number of the application program in the host that is receiving the segment. This serves the same purpose as the destination port address in the UDP header.
- **Sequence number.** This 32-bit field defines the number assigned to the first byte of data contained in this segment. As we said before, TCP is a stream transport protocol. To ensure connectivity, each byte to be transmitted is numbered. The sequence number tells the destination which byte in this sequence comprises the first byte in the segment. During connection establishment, each party uses a random number generator to create an **initial sequence number (ISN),** which is usually different in each direction.
- **Acknowledgment number.** This 32-bit field defines the byte number that the receiver of the segment is expecting to receive from the other party. If the receiver

of the segment has successfully received byte number $x$ from the other party, it defines $x + 1$ as the acknowledgment number. Acknowledgment and data can be piggybacked together.

- **Header length.** This 4-bit field indicates the number of 4-byte words in the TCP header. The length of the header can be between 20 and 60 bytes. Therefore, the value of this field can be between 5 ($5 \times 4 = 20$) and 15 ($15 \times 4 = 60$).
- **Reserved.** This is a 6-bit field reserved for future use.
- **Control.** This field defines 6 different control bits or flags as shown in Figure 23.17. One or more of these bits can be set at a time.

**Figure 23.17** *Control field*




These bits enable flow control, connection establishment and termination, connection abortion, and the mode of data transfer in TCP. A brief description of each bit is shown in Table 23.3. We will discuss them further when we study the detailed operation of TCP later in the chapter.

**Table 23.3** *Description of flags in the control field*

| *Flag* | *Description* |
|---|---|
| URG | The value of the urgent pointer field is valid. |
| ACK | The value of the acknowledgment field is valid. |
| PSH | Push the data. |
| RST | Reset the connection. |
| SYN | Synchronize sequence numbers during connection. |
| FIN | Terminate the connection. |

- **Window size.** This field defines the size of the window, in bytes, that the other party must maintain. Note that the length of this field is 16 bits, which means that the maximum size of the window is 65,535 bytes. This value is normally referred to as the receiving window (rwnd) and is determined by the receiver. The sender must obey the dictation of the receiver in this case.
- **Checksum.** This 16-bit field contains the checksum. The calculation of the checksum for TCP follows the same procedure as the one described for UDP. However, the inclusion of the checksum in the UDP datagram is optional, whereas the inclusion of the checksum for TCP is mandatory. The same pseudoheader, serving the same

purpose, is added to the segment. For the TCP pseudoheader, the value for the protocol field is 6.

- **Urgent pointer.** This 16-bit field, which is valid only if the urgent flag is set, is used when the segment contains urgent data. It defines the number that must be added to the sequence number to obtain the number of the last urgent byte in the data section of the segment. This will be discussed later in this chapter.
- **Options.** There can be up to 40 bytes of optional information in the TCP header. We will not discuss these options here; please refer to the reference list for more information.

## A TCP Connection

TCP is connection-oriented. A connection-oriented transport protocol establishes a virtual path between the source and destination. All the segments belonging to a message are then sent over this virtual path. Using a single virtual pathway for the entire message facilitates the acknowledgment process as well as retransmission of damaged or lost frames. You may wonder how TCP, which uses the services of IP, a connectionless protocol, can be connection-oriented. The point is that a TCP connection is virtual, not physical. TCP operates at a higher level. TCP uses the services of IP to deliver individual segments to the receiver, but it controls the connection itself. If a segment is lost or corrupted, it is retransmitted. Unlike TCP, IP is unaware of this retransmission. If a segment arrives out of order, TCP holds it until the missing segments arrive; IP is unaware of this reordering.

In TCP, connection-oriented transmission requires three phases: connection establishment, data transfer, and connection termination.

### *Connection Establishment*

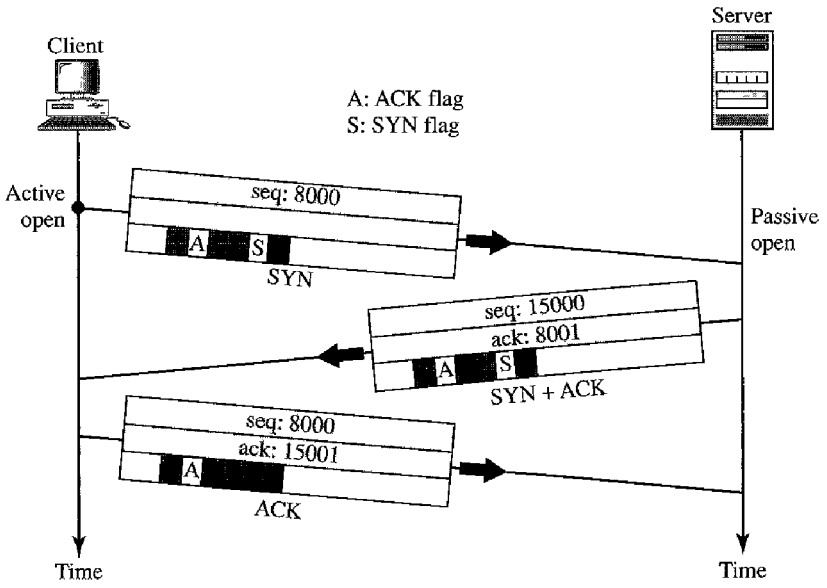
TCP transmits data in full-duplex mode. When two TCPs in two machines are connected, they are able to send segments to each other simultaneously. This implies that each party must initialize communication and get approval from the other party before any data are transferred.

**Three-Way Handshaking** The connection establishment in TCP is called **three-way handshaking.** In our example, an application program, called the client, wants to make a connection with another application program, called the server, using TCP as the transport layer protocol.

The process starts with the server. The server program tells its TCP that it is ready to accept a connection. This is called a request for a *passive open*. Although the server TCP is ready to accept any connection from any machine in the world, it cannot make the connection itself.

The client program issues a request for an *active open*. A client that wishes to connect to an open server tells its TCP that it needs to be connected to that particular server. TCP can now start the three-way handshaking process as shown in Figure 23.18. To show the process, we use two time lines: one at each site. Each segment has values for all its header fields and perhaps for some of its option fields, too. However, we show only the few fields necessary to understand each phase. We show the sequence number,

**Figure 23.18** *Connection establishment using three-way handshaking*




the acknowledgment number, the control flags (only those that are set), and the window size, if not empty. The three steps in this phase are as follows.

1. The client sends the first segment, a SYN segment, in which only the SYN flag is set. This segment is for synchronization of sequence numbers. It consumes one sequence number. When the data transfer starts, the sequence number is incremented by 1. We can say that the SYN segment carries no real data, but we can think of it as containing 1 imaginary byte.

**A SYN segment cannot carry data, but it consumes one sequence number.**

2. The server sends the second segment, a SYN + ACK segment, with 2 flag bits set: SYN and ACK. This segment has a dual purpose. It is a SYN segment for communication in the other direction and serves as the acknowledgment for the SYN segment. It consumes one sequence number.

**A SYN + ACK segment cannot carry data, but does consume one sequence number.**

3. The client sends the third segment. This is just an ACK segment. It acknowledges the receipt of the second segment with the ACK flag and acknowledgment number field. Note that the sequence number in this segment is the same as the one in the SYN segment; the ACK segment does not consume any sequence numbers.

**An ACK segment, if carrying no data, consumes no sequence number.**

**Simultaneous Open** A rare situation, called a **simultaneous open,** may occur when both processes issue an active open. In this case, both TCPs transmit a SYN + ACK segment to each other, and one single connection is established between them.

**SYN Flooding Attack** The connection establishment procedure in TCP is susceptible to a serious security problem called the **SYN flooding attack.** This happens when a malicious attacker sends a large number of SYN segments to a server, pretending that each of them is coming from a different client by faking the source IP addresses in the datagrams. The server, assuming that the clients are issuing an active open, allocates the necessary resources, such as creating communication tables and setting timers. The TCP server then sends the SYN + ACK segments to the fake clients, which are lost. During this time, however, a lot of resources are occupied without being used. If, during this short time, the number of SYN segments is large, the server eventually runs out of resources and may crash. This SYN flooding attack belongs to a type of security attack known as a **denial-of-service attack,** in which an attacker monopolizes a system with so many service requests that the system collapses and denies service to every request.

Some implementations of TCP have strategies to alleviate the effects of a SYN attack. Some have imposed a limit on connection requests during a specified period of time. Others filter out datagrams coming from unwanted source addresses. One recent strategy is to postpone resource allocation until the entire connection is set up, using what is called a **cookie.** SCTP, the new transport layer protocol that we discuss in the next section, uses this strategy.

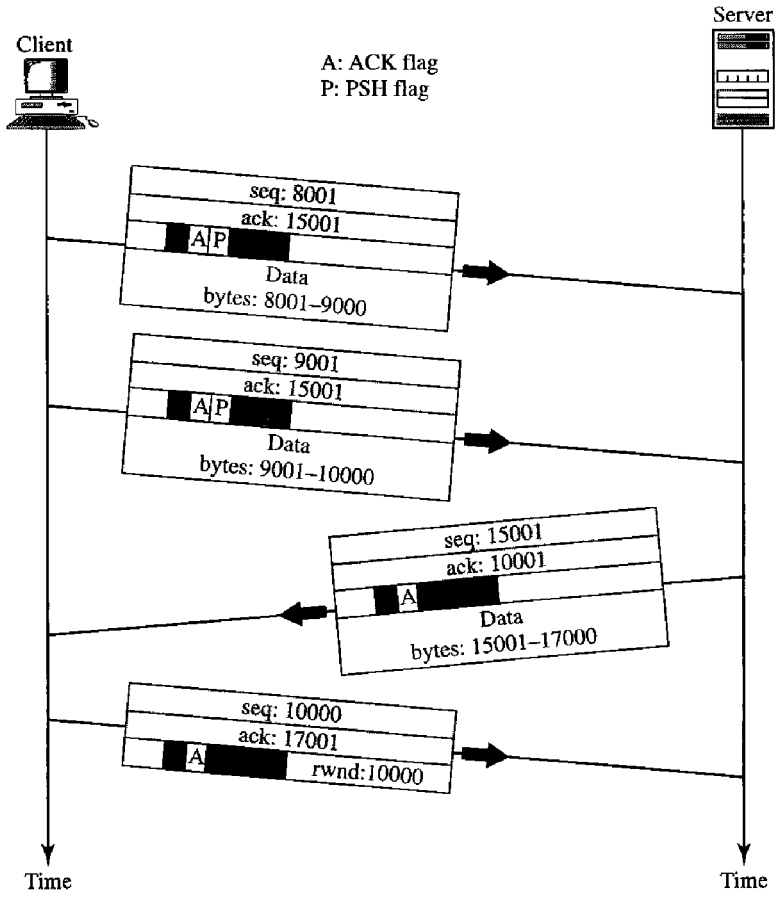
### *Data Transfer*

After connection is established, bidirectional **data transfer** can take place. The client and server can both send data and acknowledgments. We will study the rules of acknowledgment later in the chapter; for the moment, it is enough to know that data traveling in the same direction as an acknowledgment are carried on the same segment. The acknowledgment is piggybacked with the data. Figure 23.19 shows an example. In this example, after connection is established (not shown in the figure), the client sends 2000 bytes of data in two segments. The server then sends 2000 bytes in one segment. The client sends one more segment. The first three segments carry both data and acknowledgment, but the last segment carries only an acknowledgment because there are no more data to be sent. Note the values of the sequence and acknowledgment numbers. The data segments sent by the client have the PSH (push) flag set so that the server TCP knows to deliver data to the server process as soon as they are received. We discuss the use of this flag in greater detail later. The segment from the server, on the other hand, does not set the push flag. Most TCP implementations have the option to set or not set this flag.

**Pushing Data** We saw that the sending TCP uses a buffer to store the stream of data coming from the sending application program. The sending TCP can select the segment size. The receiving TCP also buffers the data when they arrive and delivers them to the application program when the application program is ready or when it is convenient for the receiving TCP. This type of flexibility increases the efficiency of TCP.

However, on occasion the application program has no need for this flexibility. For example, consider an application program that communicates interactively with another

**Figure 23.19** *Data transfer*




application program on the other end. The application program on one site wants to send a keystroke to the application at the other site and receive an immediate response. Delayed transmission and delayed delivery of data may not be acceptable by the application program.

TCP can handle such a situation. The application program at the sending site can request a *push* operation. This means that the sending TCP must not wait for the window to be filled. It must create a segment and send it immediately. The sending TCP must also set the push bit (PSH) to let the receiving TCP know that the segment includes data that must be delivered to the receiving application program as soon as possible and not to wait for more data to come.

Although the push operation can be requested by the application program, most current implementations ignore such requests. TCP can choose whether or not to use this feature.

**Urgent Data** TCP is a stream-oriented protocol. This means that the data are presented from the application program to TCP as a stream of bytes. Each byte of data has a position in the stream. However, on occasion an application program needs to send *urgent* bytes. This means that the sending application program wants a piece of data to be read out of order by the receiving application program. As an example, suppose that the sending

application program is sending data to be processed by the receiving application program. When the result of processing comes back, the sending application program finds that everything is wrong. It wants to abort the process, but it has already sent a huge amount of data. If it issues an abort command (control + C), these two characters will be stored at the end of the receiving TCP buffer. It will be delivered to the receiving application program after all the data have been processed.

The solution is to send a segment with the URG bit set. The sending application program tells the sending TCP that the piece of data is urgent. The sending TCP creates a segment and inserts the urgent data at the beginning of the segment. The rest of the segment can contain normal data from the buffer. The urgent pointer field in the header defines the end of the urgent data and the start of normal data.

When the receiving TCP receives a segment with the URG bit set, it extracts the urgent data from the segment, using the value of the urgent pointer, and delivers them, out of order, to the receiving application program.

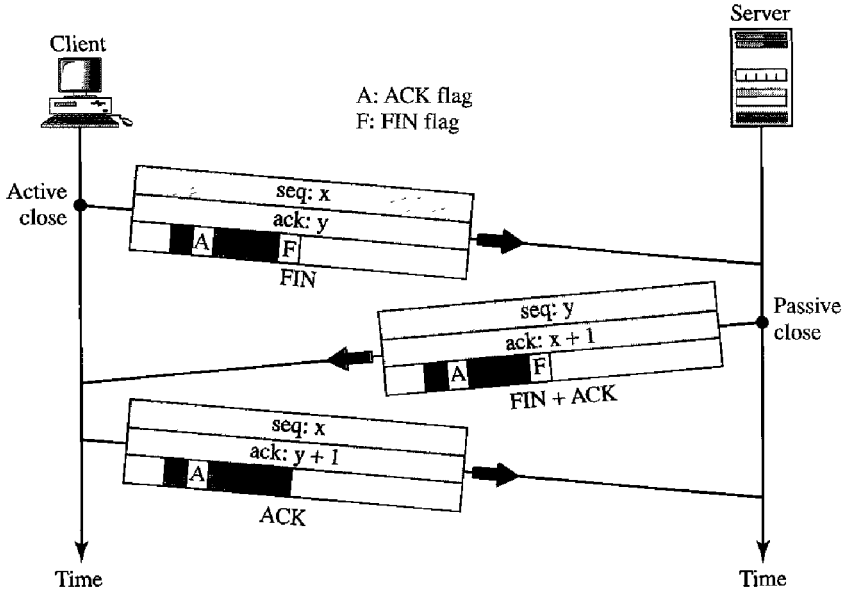
### *Connection Termination*

Any of the two parties involved in exchanging data (client or server) can close the connection, although it is usually initiated by the client. Most implementations today allow two options for connection termination: three-way handshaking and four-way handshaking with a half-close option.

**Three-Way Handshaking** Most implementations today allow *three-way handshaking* for connection termination as shown in Figure 23.20.

1. In a normal situation, the client TCP, after receiving a close command from the client process, sends the first segment, a FIN segment in which the FIN flag is set. Note that a FIN segment can include the last chunk of data sent by the client, or it

**Figure 23.20** *Connection termination using three-way handshaking*



can be just a control segment as shown in Figure 23.20. If it is only a control segment, it consumes only one sequence number.

**The FIN segment consumes one sequence number if it does not carry data.**

2. The server TCP, after receiving the FIN segment, informs its process of the situation and sends the second segment, a FIN + ACK segment, to confirm the receipt of the FIN segment from the client and at the same time to announce the closing of the connection in the other direction. This segment can also contain the last chunk of data from the server. If it does not carry data, it consumes only one sequence number.

**The FIN + ACK segment consumes one sequence number if it does not carry data.**

3. The client TCP sends the last segment, an ACK segment, to confirm the receipt of the FIN segment from the TCP server. This segment contains the acknowledgment number, which is 1 plus the sequence number received in the FIN segment from the server. This segment cannot carry data and consumes no sequence numbers.

**Half-Close** In TCP, one end can stop sending data while still receiving data. This is called a **half-close.** Although either end can issue a half-close, it is normally initiated by the client. It can occur when the server needs all the data before processing can begin. A good example is sorting. When the client sends data to the server to be sorted, the server needs to receive all the data before sorting can start. This means the client, after sending all the data, can close the connection in the outbound direction. However, the inbound direction must remain open to receive the sorted data. The server, after receiving the data, still needs time for sorting; its outbound direction must remain open.

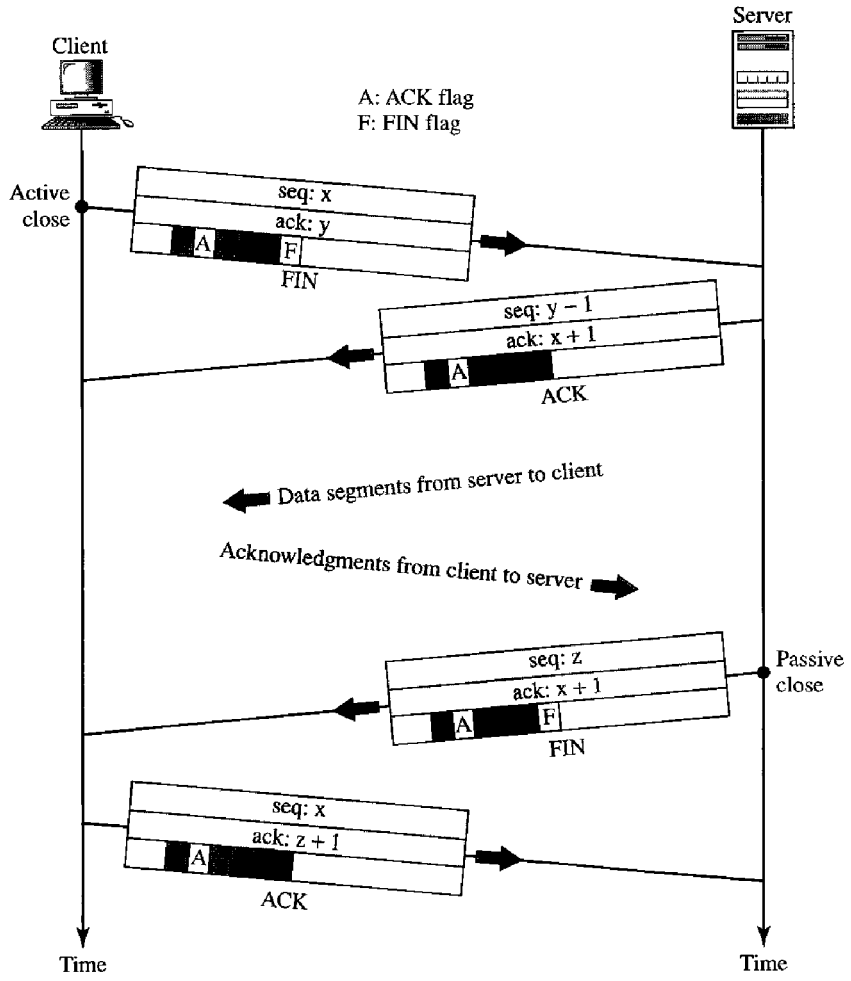
Figure 23.21 shows an example of a half-close. The client half-closes the connection by sending a FIN segment. The server accepts the half-close by sending the ACK segment. The data transfer from the client to the server stops. The server, however, can still send data. When the server has sent all the processed data, it sends a FIN segment, which is acknowledged by an ACK from the client.

After half-closing of the connection, data can travel from the server to the client and acknowledgments can travel from the client to the server. The client cannot send any more data to the server. Note the sequence numbers we have used. The second segment (ACK) consumes no sequence number. Although the client has received sequence number $y - 1$ and is expecting $y$, the server sequence number is still $y - 1$. When the connection finally closes, the sequence number of the last ACK segment is still $x$, because no sequence numbers are consumed during data transfer in that direction.

## Flow Control

TCP uses a sliding window, as discussed in Chapter 11, to handle flow control. The sliding window protocol used by TCP, however, is something between the Go-Back-$N$ and Selective Repeat sliding window. The sliding window protocol in TCP looks like

**Figure 23.21** *Half-close*



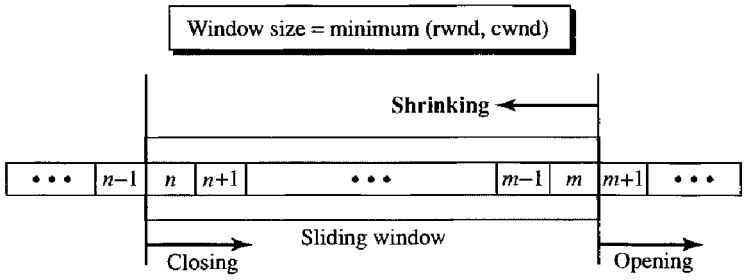

the Go-Back-*N* protocol because it does not use NAKs; it looks like Selective Repeat because the receiver holds the out-of-order segments until the missing ones arrive. There are two big differences between this sliding window and the one we used at the data link layer. First, the sliding window of TCP is byte-oriented; the one we discussed in the data link layer is frame-oriented. Second, the TCP's sliding window is of variable size; the one we discussed in the data link layer was of fixed size.

Figure 23.22 shows the sliding window in TCP. The window spans a portion of the buffer containing bytes received from the process. The bytes inside the window are the bytes that can be in transit; they can be sent without worrying about acknowledgment. The imaginary window has two walls: one left and one right.

The window is *opened, closed,* or *shrunk*. These three activities, as we will see, are in the control of the receiver (and depend on congestion in the network), not the sender. The sender must obey the commands of the receiver in this matter.

Opening a window means moving the right wall to the right. This allows more new bytes in the buffer that are eligible for sending. Closing the window means moving the left wall to the right. This means that some bytes have been acknowledged and the sender

**Figure 23.22** *Sliding window*




need not worry about them anymore. Shrinking the window means moving the right wall to the left. This is strongly discouraged and not allowed in some implementations because it means revoking the eligibility of some bytes for sending. This is a problem if the sender has already sent these bytes. Note that the left wall cannot move to the left because this would revoke some of the previously sent acknowledgments.

**A sliding window is used to make transmission more efficient as well as to control the flow of data so that the destination does not become overwhelmed with data. TCP sliding windows are byte-oriented.**

The size of the window at one end is determined by the lesser of two values: *receiver window* (*rwnd*) or *congestion window* (*cwnd*). The *receiver window* is the value advertised by the opposite end in a segment containing acknowledgment. It is the number of bytes the other end can accept before its buffer overflows and data are discarded. The congestion window is a value determined by the network to avoid congestion. We will discuss congestion later in the chapter.

### *Example 23.4*

What is the value of the receiver window (*rwnd*) for host A if the receiver, host B, has a buffer size of 5000 bytes and 1000 bytes of received and unprocessed data?

### Solution

The value of *rwnd* = 5000 − 1000 = 4000. Host B can receive only 4000 bytes of data before overflowing its buffer. Host B advertises this value in its next segment to A.

### *Example 23.5*

What is the size of the window for host A if the value of *rwnd* is 3000 bytes and the value of *cwnd* is 3500 bytes?

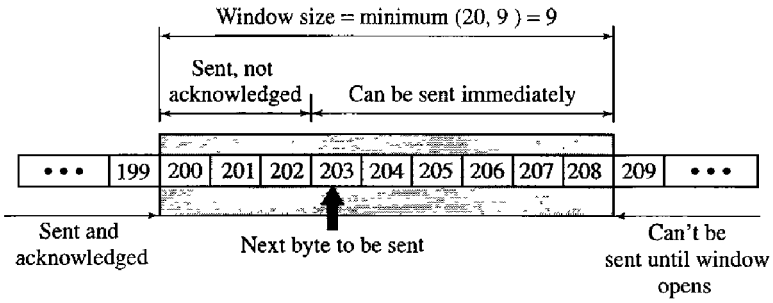
### Solution

The size of the window is the smaller of *rwnd* and *cwnd*, which is 3000 bytes.

### *Example 23.6*

Figure 23.23 shows an unrealistic example of a sliding window. The sender has sent bytes up to 202. We assume that cwnd is 20 (in reality this value is thousands of bytes). The receiver has sent

**Figure 23.23** *Example 23.6*




an acknowledgment number of 200 with an *rwnd* of 9 bytes (in reality this value is thousands of bytes). The size of the sender window is the minimum of *rwnd* and *cwnd,* or 9 bytes. Bytes 200 to 202 are sent, but not acknowledged. Bytes 203 to 208 can be sent without worrying about acknowledgment. Bytes 209 and above cannot be sent.

**Some points about TCP sliding windows:**

- The size of the window is the lesser of *rwnd* and *cwnd.*
- The source does not have to send a full window's worth of data.
- The window can be opened or closed by the receiver, but should not be shrunk.
- The destination can send an acknowledgment at any time as long as it does not result in a shrinking window.
- The receiver can temporarily shut down the window; the sender, however, can always send a segment of 1 byte after the window is shut down.

## Error Control

TCP is a reliable transport layer protocol. This means that an application program that delivers a stream of data to TCP relies on TCP to deliver the entire stream to the application program on the other end in order, without error, and without any part lost or duplicated.

TCP provides reliability using error control. Error control includes mechanisms for detecting corrupted segments, lost segments, out-of-order segments, and duplicated segments. Error control also includes a mechanism for correcting errors after they are detected. Error detection and correction in TCP is achieved through the use of three simple tools: checksum, acknowledgment, and time-out.

### *Checksum*

Each segment includes a checksum field which is used to check for a corrupted segment. If the segment is corrupted, it is discarded by the destination TCP and is considered as lost. TCP uses a 16-bit checksum that is mandatory in every segment. We will see, in Chapter 24, that the 16-bit checksum is considered inadequate for the new transport

layer, SCTP. However, it cannot be changed for TCP because this would involve reconfiguration of the entire header format.

### *Acknowledgment*

TCP uses acknowledgments to confirm the receipt of data segments. Control segments that carry no data but consume a sequence number are also acknowledged. ACK segments are never acknowledged.

---

**ACK segments do not consume sequence numbers and are not acknowledged.**

---

### *Retransmission*

The heart of the error control mechanism is the retransmission of segments. When a segment is corrupted, lost, or delayed, it is retransmitted. In modern implementations, a segment is retransmitted on two occasions: when a **retransmission timer** expires or when the sender receives three duplicate ACKs.

---

**In modern implementations, a retransmission occurs if the retransmission timer expires or three duplicate ACK segments have arrived.**

---

Note that no retransmission occurs for segments that do not consume sequence numbers. In particular, there is no transmission for an ACK segment.

---

**No retransmission timer is set for an ACK segment.**

---

**Retransmission After RTO** A recent implementation of TCP maintains one **retransmission time-out (RTO)** timer for all outstanding (sent, but not acknowledged) segments. When the timer matures, the earliest outstanding segment is retransmitted even though lack of a received ACK can be due to a delayed segment, a delayed ACK, or a lost acknowledgment. Note that no time-out timer is set for a segment that carries only an acknowledgment, which means that no such segment is resent. The value of RTO is dynamic in TCP and is updated based on the **round-trip time (RTT)** of segments. An RTT is the time needed for a segment to reach a destination and for an acknowledgment to be received. It uses a back-off strategy similar to one discussed in Chapter 12.

**Retransmission After Three Duplicate ACK Segments** The previous rule about retransmission of a segment is sufficient if the value of RTO is not very large. Sometimes, however, one segment is lost and the receiver receives so many out-of-order segments that they cannot be saved (limited buffer size). To alleviate this situation, most implementations today follow the three-duplicate-ACKs rule and retransmit the missing segment immediately. This feature is referred to as **fast retransmission,** which we will see in an example shortly.

### *Out-of-Order Segments*

When a segment is delayed, lost, or discarded, the segments following that segment arrive out of order. Originally, TCP was designed to discard all out-of-order segments, resulting

in the retransmission of the missing segment and the following segments. Most implementations today do not discard the out-of-order segments. They store them temporarily and flag them as out-of-order segments until the missing segment arrives. Note, however, that the out-of-order segments are not delivered to the process. TCP guarantees that data are delivered to the process in order.

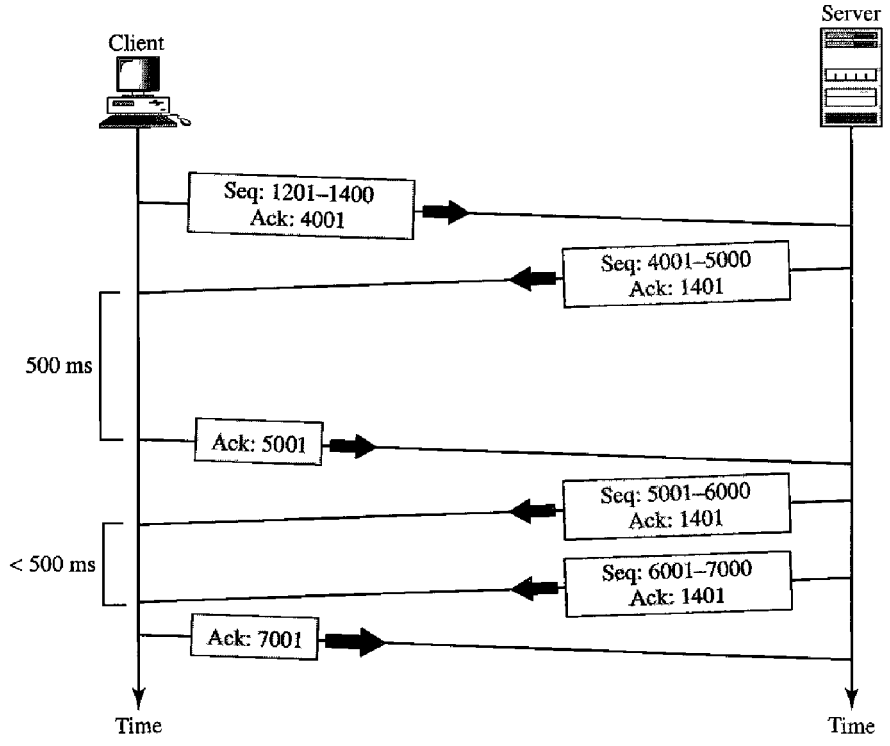
**Data may arrive out of order and be temporarily stored by the receiving TCP, but TCP guarantees that no out-of-order segment is delivered to the process.**

### *Some Scenarios*

In this section we give some examples of scenarios that occur during the operation of TCP. In these scenarios, we show a segment by a rectangle. If the segment carries data, we show the range of byte numbers and the value of the acknowledgment field. If it carries only an acknowledgment, we show only the acknowledgment number in a smaller box.

**Normal Operation** The first scenario shows bidirectional data transfer between two systems, as in Figure 23.24. The client TCP sends one segment; the server TCP sends three. The figure shows which rule applies to each acknowledgment. There are data to be sent, so the segment displays the next byte expected. When the client receives the first segment from the server, it does not have any more data to send; it sends only an

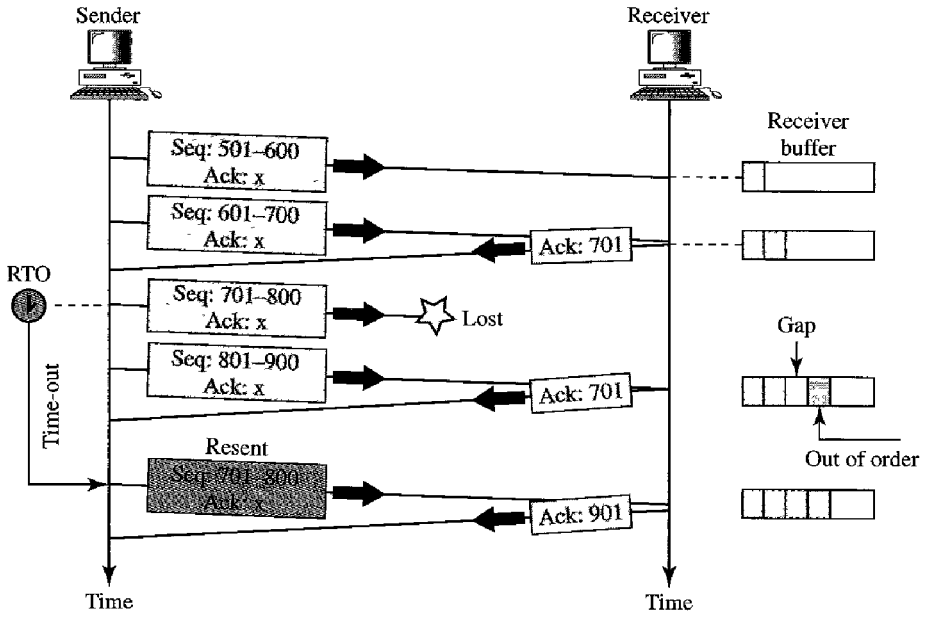
**Figure 23.24** *Normal operation*



ACK segment. However, the acknowledgment needs to be delayed for 500 ms to see if any more segments arrive. When the timer matures, it triggers an acknowledgment. This is so because the client has no knowledge if other segments are coming; it cannot delay the acknowledgment forever. When the next segment arrives, another acknowledgment timer is set. However, before it matures, the third segment arrives. The arrival of the third segment triggers another acknowledgment.

**Lost Segment** In this scenario, we show what happens when a segment is lost or corrupted. A lost segment and a corrupted segment are treated the same way by the receiver. A lost segment is discarded somewhere in the network; a corrupted segment is discarded by the receiver itself. Both are considered lost. Figure 23.25 shows a situation in which a segment is lost and discarded by some router in the network, perhaps due to congestion.

**Figure 23.25** *Lost segment*



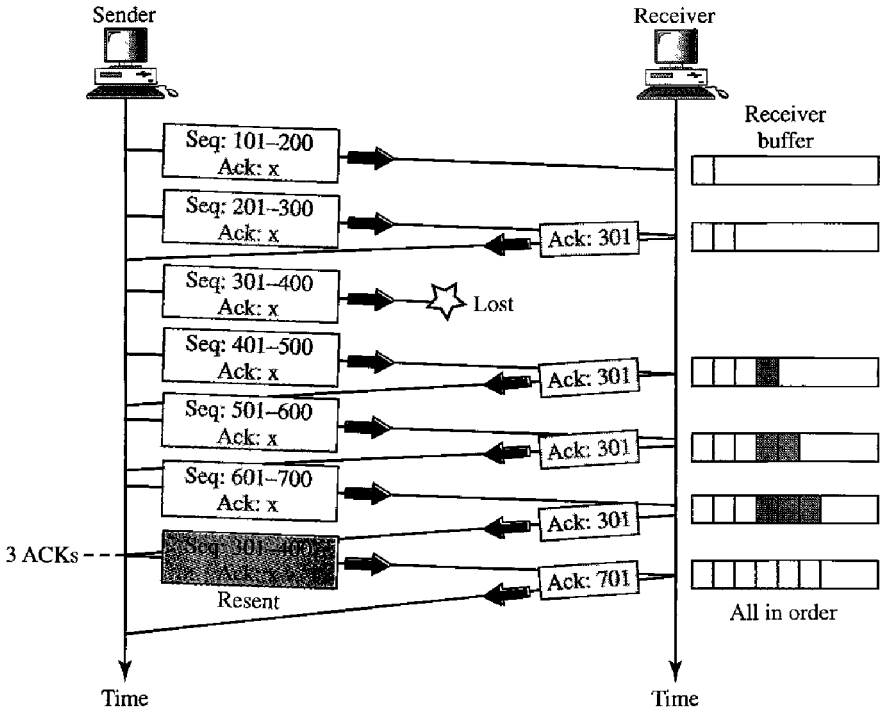

We are assuming that data transfer is unidirectional: one site is sending, the other is receiving. In our scenario, the sender sends segments 1 and 2, which are acknowledged immediately by an ACK. Segment 3, however, is lost. The receiver receives segment 4, which is out of order. The receiver stores the data in the segment in its buffer but leaves a gap to indicate that there is no continuity in the data. The receiver immediately sends an acknowledgment to the sender, displaying the next byte it expects. Note that the receiver stores bytes 801 to 900, but never delivers these bytes to the application until the gap is filled.

**The receiver TCP delivers only ordered data to the process.**

We have shown the timer for the earliest outstanding segment. The timer for this definitely runs out because the receiver never sends an acknowledgment for lost or out-of-order segments. When the timer matures, the sending TCP resends segment 3, which arrives this time and is acknowledged properly. Note that the value in the second and third acknowledgments differs according to the corresponding rule.

**Fast Retransmission** In this scenario, we want to show the idea of fast retransmission. Our scenario is the same as the second except that the RTO has a higher value (see Figure 23.26).

**Figure 23.26** *Fast retransmission*




When the receiver receives the fourth, fifth, and sixth segments, it triggers an acknowledgment. The sender receives four acknowledgments with the same value (three duplicates). Although the timer for segment 3 has not matured yet, the fast transmission requires that segment 3, the segment that is expected by all these acknowledgments, be resent immediately.

Note that only one segment is retransmitted although four segments are not acknowledged. When the sender receives the retransmitted ACK, it knows that the four segments are safe and sound because acknowledgment is cumulative.

## Congestion Control

We discuss congestion control of TCP in Chapter 24.

## 23.4 SCTP

Stream Control Transmission Protocol (SCTP) is a new reliable, message-oriented transport layer protocol. SCTP, however, is mostly designed for Internet applications that have recently been introduced. These new applications, such as IUA (ISDN over IP), M2UA and M3UA (telephony signaling), H.248 (media gateway control), H.323 (IP telephony), and SIP (IP telephony), need a more sophisticated service than TCP can provide. SCTP provides this enhanced performance and reliability. We briefly compare UDP, TCP, and SCTP:

- UDP is a **message-oriented** protocol. A process delivers a message to UDP, which is encapsulated in a user datagram and sent over the network. UDP *conserves the message boundaries;* each message is independent of any other message. This is a desirable feature when we are dealing with applications such as IP telephony and transmission of real-time data, as we will see later in the text. However, UDP is unreliable; the sender cannot know the destiny of messages sent. A message can be lost, duplicated, or received out of order. UDP also lacks some other features, such as congestion control and flow control, needed for a friendly transport layer protocol.
- TCP is a **byte-oriented** protocol. It receives a message or messages from a process, stores them as a stream of bytes, and sends them in segments. There is no preservation of the message boundaries. However, TCP is a reliable protocol. The duplicate segments are detected, the lost segments are resent, and the bytes are delivered to the end process in order. TCP also has congestion control and flow control mechanisms.
- SCTP combines the best features of UDP and TCP. SCTP is a reliable message-oriented protocol. It preserves the message boundaries and at the same time detects lost data, duplicate data, and out-of-order data. It also has congestion control and flow control mechanisms. Later we will see that SCTP has other innovative features unavailable in UDP and TCP.

**SCTP is a *message-oriented, reliable* protocol that combines the best features of UDP and TCP.**

### SCTP Services

Before we discuss the operation of SCTP, let us explain the services offered by SCTP to the application layer processes.

#### *Process-to-Process Communication*

SCTP uses all well-known ports in the TCP space. Table 23.4 lists some extra port numbers used by SCTP.

#### *Multiple Streams*

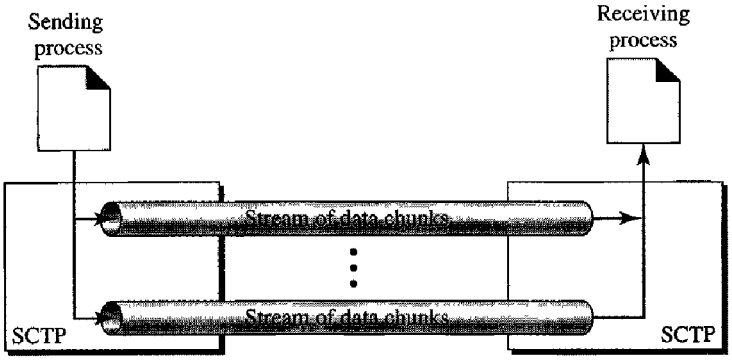
We learned in the previous section that TCP is a stream-oriented protocol. Each connection between a TCP client and a TCP server involves one single stream. The problem

**Table 23.4** *Some SCTP applications*

| *Protocol* | *Port Number* | *Description* |
|---|---|---|
| IUA | 9990 | ISDN over IP |
| M2UA | 2904 | SS7 telephony signaling |
| M3UA | 2905 | SS7 telephony signaling |
| H.248 | 2945 | Media gateway control |
| H.323 | 1718, 1719, 1720, 11720 | IP telephony |
| SIP | 5060 | IP telephony |

with this approach is that a loss at any point in the stream blocks the delivery of the rest of the data. This can be acceptable when we are transferring text; it is not when we are sending real-time data such as audio or video. SCTP allows **multistream service** in each connection, which is called **association** in SCTP terminology. If one of the streams is blocked, the other streams can still deliver their data.The idea is similar to multiple lanes on a highway. Each lane can be used for a different type of traffic. For example, one lane can be used for regular traffic, another for car pools. If the traffic is blocked for regular vehicles, car pool vehicles can still reach their destinations. Figure 23.27 shows the idea of multiple-stream delivery.

**Figure 23.27** *Multiple-stream concept*



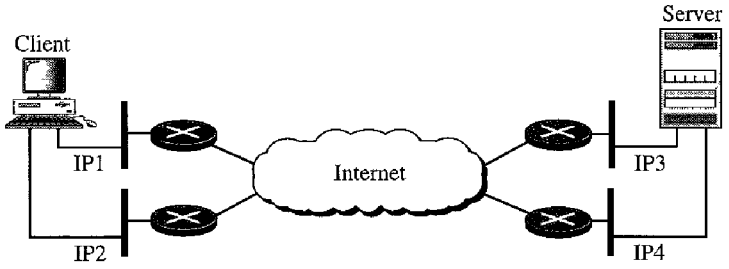

**An association in SCTP can involve multiple streams.**

### *Multihoming*

A TCP connection involves one source and one destination IP address. This means that even if the sender or receiver is a multihomed host (connected to more than one physical address with multiple IP addresses), only one of these IP addresses per end can be utilized during the connection. An SCTP association, on the other hand, supports **multihoming service.** The sending and receiving host can define multiple IP addresses in each end for an association. In this fault-tolerant approach, when one path fails, another interface can be used for data delivery without interruption. This fault-tolerant

feature is very helpful when we are sending and receiving a real-time payload such as Internet telephony. Figure 23.28 shows the idea of multihoming.

**Figure 23.28** *Multihoming concept*




In Figure 23.28, the client is connected to two local networks with two IP addresses. The server is also connected to two networks with two IP addresses. The client and the server can make an association, using four different pairs of IP addresses. However, note that in the current implementations of SCTP, only one pair of IP addresses can be chosen for normal communication; the alternative is used if the main choice fails. In other words, at present, SCTP does not allow load sharing between different paths.

**SCTP association allows multiple IP addresses for each end.**

### *Full-Duplex Communication*

Like TCP, SCTP offers full-duplex service, in which data can flow in both directions at the same time. Each SCTP then has a sending and receiving buffer, and packets are sent in both directions.

### *Connection-Oriented Service*

Like TCP, SCTP is a connection-oriented protocol. However, in SCTP, a connection is called an association. When a process at site A wants to send and receive data from another process at site B, the following occurs:

1. The two SCTPs establish an association between each other.
2. Data are exchanged in both directions.
3. The association is terminated.

### *Reliable Service*

SCTP, like TCP, is a reliable transport protocol. It uses an acknowledgment mechanism to check the safe and sound arrival of data. We will discuss this feature further in the section on error control.

## SCTP Features

Let us first discuss the general features of SCTP and then compare them with those of TCP.

### *Transmission Sequence Number*

The unit of data in TCP is a byte. Data transfer in TCP is controlled by numbering bytes by using a sequence number. On the other hand, the unit of data in SCTP is a **DATA chunk** which may or may not have a one-to-one relationship with the message coming from the process because of fragmentation (discussed later). Data transfer in SCTP is controlled by numbering the data chunks. SCTP uses a **transmission sequence number (TSN)** to number the data chunks. In other words, the TSN in SCTP plays the analogous role to the sequence number in TCP. TSNs are 32 bits long and randomly initialized between 0 and $2^{32} - 1$. Each data chunk must carry the corresponding TSN in its header.

---

**In SCTP, a data chunk is numbered using a TSN.**

---

### *Stream Identifier*

In TCP, there is only one stream in each connection. In SCTP, there may be several streams in each association. Each stream in SCTP needs to be identified by using a **stream identifier (SI).** Each data chunk must carry the SI in its header so that when it arrives at the destination, it can be properly placed in its stream. The SI is a 16-bit number starting from 0.

---

**To distinguish between different streams, SCTP uses an SI.**

---

### *Stream Sequence Number*

When a data chunk arrives at the destination SCTP, it is delivered to the appropriate stream and in the proper order. This means that, in addition to an SI, SCTP defines each data chunk in each stream with a **stream sequence number (SSN).**

---

**To distinguish between different data chunks belonging to the same stream, SCTP uses SSNs.**

---

### *Packets*

In TCP, a segment carries data and control information. Data are carried as a collection of bytes; control information is defined by six control flags in the header. The design of SCTP is totally different: data are carried as data chunks, control information is carried as control chunks. Several control chunks and data chunks can be packed together in a packet. A packet in SCTP plays the same role as a segment in TCP. Figure 23.29 compares a segment in TCP and a packet in SCTP. Let us briefly list the differences between an SCTP packet and a TCP segment:

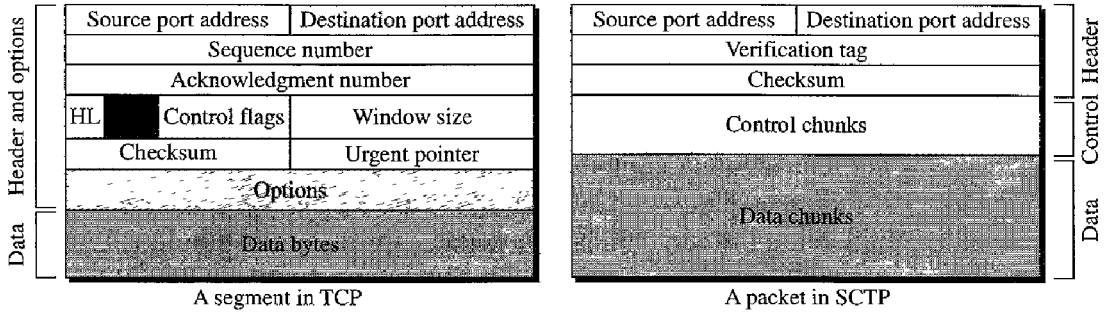
---

**TCP has segments; SCTP has packets.**

---

1. The control information in TCP is part of the header; the control information in SCTP is included in the control chunks. There are several types of control chunks; each is used for a different purpose.

**Figure 23.29** *Comparison between a TCP segment and an SCTP packet*




2. The data in a TCP segment treated as one entity; an SCTP packet can carry several data chunks; each can belong to a different stream.
3. The options section, which can be part of a TCP segment, does not exist in an SCTP packet. Options in SCTP are handled by defining new chunk types.
4. The mandatory part of the TCP header is 20 bytes, while the general header in SCTP is only 12 bytes. The SCTP header is shorter due to the following:
   a. An SCTP sequence number (TSN) belongs to each data chunk and hence is located in the chunk's header.
   b. The acknowledgment number and window size are part of each control chunk.
   c. There is no need for a header length field (shown as HL in the TCP segment) because there are no options to make the length of the header variable; the SCTP header length is fixed (12 bytes).
   d. There is no need for an urgent pointer in SCTP.
5. The checksum in TCP is 16 bits; in SCTP, it is 32 bits.
6. The **verification** tag in SCTP is an association identifier, which does not exist in TCP. In TCP, the combination of IP and port addresses defines a connection; in SCTP we may have multihoming using different IP addresses. A unique verification tag is needed to define each association.
7. TCP includes one sequence number in the header, which defines the number of the first byte in the data section. An SCTP packet can include several different data chunks. TSNs, SIs, and SSNs define each data chunk.
8. Some segments in TCP that carry control information (such as SYN and FIN) need to consume one sequence number; control chunks in SCTP never use a TSN, SI, or SSN. These three identifiers belong only to data chunks, not to the whole packet.

**In SCTP, control information and data information are carried in separate chunks.**

In SCTP, we have data chunks, streams, and packets. An association may send many packets, a packet may contain several chunks, and chunks may belong to different streams. To make the definitions of these terms clear, let us suppose that process A needs to send 11 messages to process B in three streams. The first four messages are in

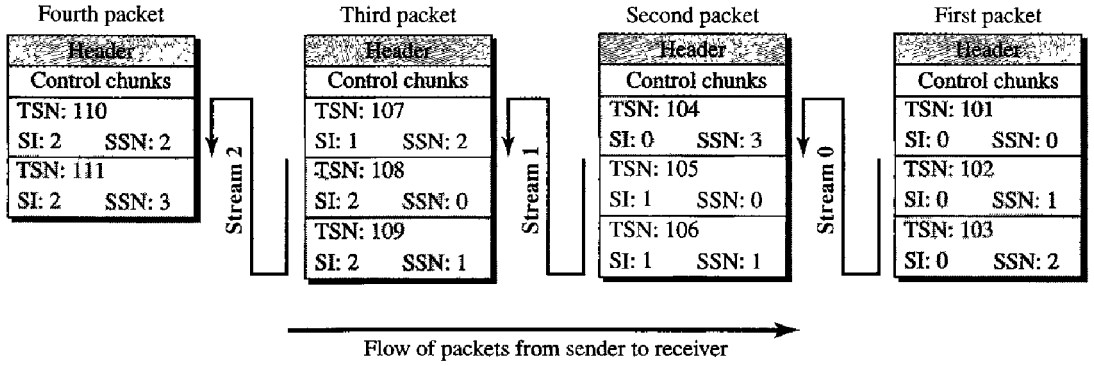
the first stream, the second three messages are in the second stream, and the last four messages are in the third stream.

Although a message, if long, can be carried by several data chunks, we assume that each message fits into one data chunk. Therefore, we have 11 data chunks in three streams.

The application process delivers 11 messages to SCTP, where each message is earmarked for the appropriate stream. Although the process could deliver one message from the first stream and then another from the second, we assume that it delivers all messages belonging to the first stream first, all messages belonging to the second stream next, and finally, all messages belonging to the last stream.

We also assume that the network allows only three data chunks per packet, which means that we need four packets as shown in Figure 23.30. Data chunks in stream 0 are carried in the first packet and part of the second packet; those in stream 1 are carried in the second and third packets; those in stream 2 are carried in the third and fourth packets.

**Figure 23.30** *Packet, data chunks, and streams*




Note that each data chunk needs three identifiers: TSN, SI, and SSN. TSN is a cumulative number and is used, as we will see later, for flow control and error control. SI defines the stream to which the chunk belongs. SSN defines the chunk's order in a particular stream. In our example, SSN starts from 0 for each stream.

**Data chunks are identified by three items: TSN, SI, and SSN. TSN is a cumulative number identifying the association; SI defines the stream; SSN defines the chunk in a stream.**

### *Acknowledgment Number*

TCP acknowledgment numbers are byte-oriented and refer to the sequence numbers. SCTP acknowledgment numbers are chunk-oriented. They refer to the TSN. A second difference between TCP and SCTP acknowledgments is the control information. Recall that this information is part of the segment header in TCP. To acknowledge segments that carry only control information, TCP uses a sequence number and acknowledgment number (for example, a SYN segment needs to be acknowledged by an ACK segment). In SCTP, however, the control information is carried by control chunks, which do not

need a TSN. These control chunks are acknowledged by another control chunk of the appropriate type (some need no acknowledgment). For example, an INIT control chunk is acknowledged by an INIT ACK chunk. There is no need for a sequence number or an acknowledgment number.

**In SCTP, acknowledgment numbers are used to acknowledge only data chunks; control chunks are acknowledged by other control chunks if necessary.**

### *Flow Control*

Like TCP, SCTP implements flow control to avoid overwhelming the receiver. We will discuss SCTP flow control later in the chapter.

### *Error Control*

Like TCP, SCTP implements error control to provide reliability. TSN numbers and acknowledgment numbers are used for error control. We will discuss error control later in the chapter.

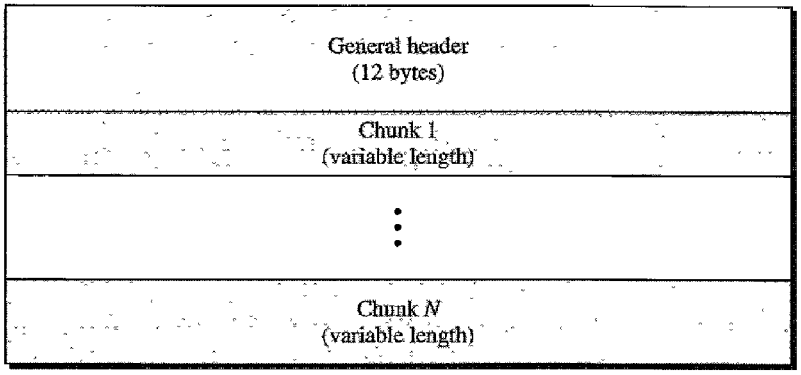
### *Congestion Control*

Like TCP, SCTP implements congestion control to determine how many data chunks can be injected into the network. We will discuss congestion control in Chapter 24.

## Packet Format

In this section, we show the format of a packet and different types of chunks. Most of the information presented in this section will become clear later; this section can be skipped in the first reading or used only as a reference. An SCTP packet has a mandatory general header and a set of blocks called chunks. There are two types of chunks: control chunks and data chunks. A control chunk controls and maintains the association; a data chunk carries user data. In a packet, the control chunks come before the data chunks. Figure 23.31 shows the general format of an SCTP packet.

**Figure 23.31** *SCTP packet format*

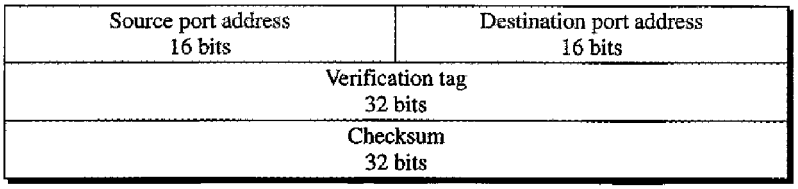


**In an SCTP packet, control chunks come before data chunks.**

### *General Header*

The **general header** (packet header) defines the endpoints of each association to which the packet belongs, guarantees that the packet belongs to a particular association, and preserves the integrity of the contents of the packet including the header itself. The format of the general header is shown in Figure 23.32.

**Figure 23.32** *General header*




There are four fields in the general header:

- **Source port address.** This is a 16-bit field that defines the port number of the process sending the packet.
- **Destination port address.** This is a 16-bit field that defines the port number of the process receiving the packet.
- **Verification tag.** This is a number that matches a packet to an association. This prevents a packet from a previous association from being mistaken as a packet in this association. It serves as an identifier for the association; it is repeated in every packet during the association. There is a separate verification used for each direction in the association.
- **Checksum.** This 32-bit field contains a CRC-32 checksum. Note that the size of the checksum is increased from 16 (in UDP, TCP, and IP) to 32 bits to allow the use of the CRC-32 checksum.

### *Chunks*

Control information or user data are carried in chunks. The detailed format of each chunk is beyond the scope of this book. See [For06] for details. The first three fields are common to all chunks; the information field depends on the type of chunk. The important point to remember is that SCTP requires the information section to be a multiple of 4 bytes; if not, padding bytes (eight 0s) are added at the end of the section. See Table 23.5 for a list of chunks and their descriptions.

## An SCTP Association

SCTP, like TCP, is a connection-oriented protocol. However, a connection in SCTP is called an *association* to emphasize multihoming.

**Table 23.5** *Chunks*

| *Type* | *Chunk* | *Description* |
|---|---|---|
| **0** | **DATA** | User data |
| **1** | **INIT** | Sets up an association |
| **2** | **INIT ACK** | Acknowledges INIT chunk |
| **3** | **SACK** | Selective acknowledgment |
| **4** | **HEARTBEAT** | Probes the peer for liveliness |
| **5** | **HEARTBEAT ACK** | Acknowledges HEARTBEAT chunk |
| **6** | **ABORT** | Aborts an association |
| **7** | **SHUTDOWN** | Terminates an association |
| **8** | **SHUTDOWN ACK** | Acknowledges SHUTDOWN chunk |
| **9** | **ERROR** | Reports errors without shutting down |
| **10** | **COOKIE ECHO** | Third packet in association establishment |
| **11** | **COOKIE ACK** | Acknowledges COOKIE ECHO chunk |
| **14** | **SHUTDOWN COMPLETE** | Third packet in association termination |
| **192** | **FORWARD TSN** | For adjusting cumulative TSN |

**A connection in SCTP is called an association.**

### *Association Establishment*

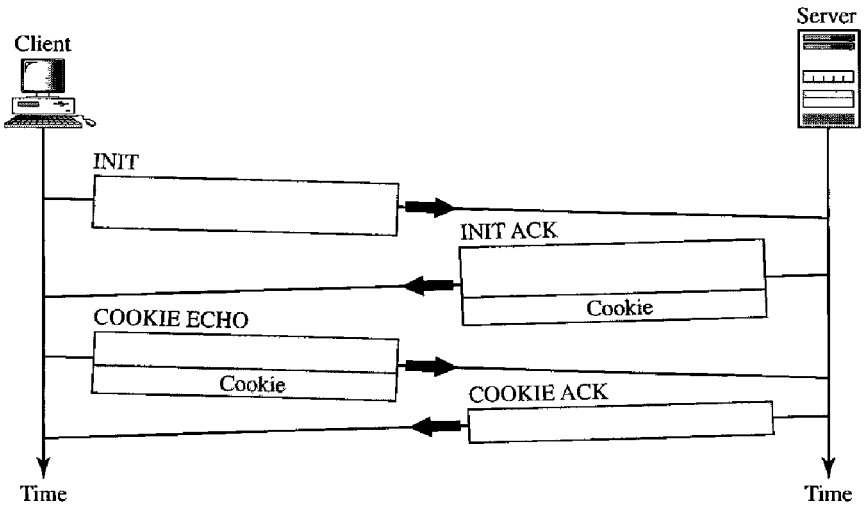
**Association establishment** in SCTP requires a **four-way handshake.** In this procedure, a process, normally a client, wants to establish an association with another process, normally a server, using SCTP as the transport layer protocol. Similar to TCP, the SCTP server needs to be prepared to receive any association (passive open). Association establishment, however, is initiated by the client (active open). SCTP association establishment is shown in Figure 23.33. The steps, in a normal situation, are as follows:

1. The client sends the first packet, which contains an **INIT chunk.**
2. The server sends the second packet, which contains an **INIT ACK chunk.**
3. The client sends the third packet, which includes a **COOKIE ECHO chunk.** This is a very simple chunk that echoes, without change, the cookie sent by the server. SCTP allows the inclusion of data chunks in this packet.
4. The server sends the fourth packet, which includes the **COOKIE ACK chunk** that acknowledges the receipt of the COOKIE ECHO chunk. SCTP allows the inclusion of data chunks with this packet.

**No other chunk is allowed in a packet carrying an INIT or INIT ACK chunk. A COOKIE ECHO or a COOKIE ACK chunk can carry data chunks.**

**Cookie** We discussed a SYN flooding attack in the previous section. With TCP, a malicious attacker can flood a TCP server with a huge number of phony SYN segments using different forged IP addresses. Each time the server receives a SYN segment, it

**Figure 23.33** *Four-way handshaking*




sets up a state table and allocates other resources while waiting for the next segment to arrive. After a while, however, the server may collapse due to the exhaustion of resources.

The designers of SCTP have a strategy to prevent this type of attack. The strategy is to postpone the allocation of resources until the reception of the third packet, when the IP address of the sender is verified. The information received in the first packet must somehow be saved until the third packet arrives. But if the server saved the information, that would require the allocation of resources (memory); this is the dilemma. The solution is to pack the information and send it back to the client. This is called generating a **cookie.** The cookie is sent with the second packet to the address received in the first packet. There are two potential situations.

1. If the sender of the first packet is an attacker, the server never receives the third packet; the cookie is lost and no resources are allocated. The only effort for the server is "baking" the cookie.
2. If the sender of the first packet is an honest client that needs to make a connection, it receives the second packet, with the cookie. It sends a packet (third in the series) with the cookie, with no changes. The server receives the third packet and knows that it has come from an honest client because the cookie that the sender has sent is there. The server can now allocate resources.

The above strategy works if no entity can "eat" a cookie "baked" by the server. To guarantee this, the server creates a digest (see Chapter 30) from the information, using its own secret key. The information and the digest together make the cookie, which is sent to the client in the second packet. When the cookie is returned in the third packet, the server calculates the digest from the information. If the digest matches the one that is sent, the cookie has not been changed by any other entity.

### *Data Transfer*

The whole purpose of an association is to transfer data between two ends. After the association is established, bidirectional data transfer can take place. The client and the server can both send data. Like TCP, SCTP supports piggybacking.

There is a major difference, however, between data transfer in TCP and SCTP. TCP receives messages from a process as a stream of bytes without recognizing any boundary between them. The process may insert some boundaries for its peer use, but TCP treats that mark as part of the text. In other words, TCP takes each message and appends it to its buffer. A segment can carry parts of two different messages. The only ordering system imposed by TCP is the byte numbers.

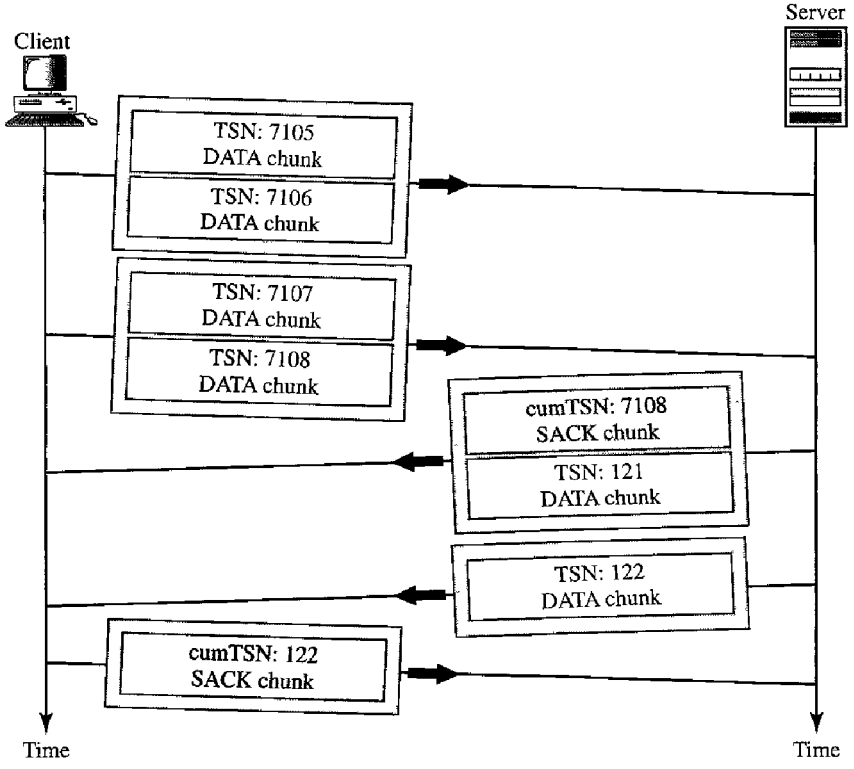
SCTP, on the other hand, recognizes and maintains boundaries. Each message coming from the process is treated as one unit and inserted into a **DATA chunk** unless it is fragmented (discussed later). In this sense, SCTP is like UDP, with one big advantage: data chunks are related to each other.

A message received from a process becomes a DATA chunk, or chunks if fragmented, by adding a DATA chunk header to the message. Each DATA chunk formed by a message or a fragment of a message has one TSN. We need to remember that only DATA chunks use TSNs and only DATA chunks are acknowledged by SACK chunks.

**In SCTP, only DATA chunks consume TSNs;**
**DATA chunks are the only chunks that are acknowledged.**

Let us show a simple scenario in Figure 23.34. In this figure a client sends four DATA chunks and receives two DATA chunks from the server. Later, we will discuss

**Figure 23.34** *Simple data transfer*



the use of flow and error control in SCTP. For the moment, we assume that everything goes well in this scenario.

1. The client sends the first packet carrying two DATA chunks with TSNs 7105 and 7106.
2. The client sends the second packet carrying two DATA chunks with TSNs 7107 and 7108.
3. The third packet is from the server. It contains the SACK chunk needed to acknowledge the receipt of DATA chunks from the client. Contrary to TCP, SCTP acknowledges the last in-order TSN received, not the next expected. The third packet also includes the first DATA chunk from the server with TSN 121.
4. After a while, the server sends another packet carrying the last DATA chunk with TSN 122, but it does not include a SACK chunk in the packet because the last DATA chunk received from the client was already acknowledged.
5. Finally, the client sends a packet that contains a SACK chunk acknowledging the receipt of the last two DATA chunks from the server.

---

**The acknowledgment in SCTP defines the cumulative TSN, the TSN of the last data chunk received in order.**

---

**Multihoming Data Transfer** We discussed the multihoming capability of SCTP, a feature that distinguishes SCTP from UDP and TCP. Multihoming allows both ends to define multiple IP addresses for communication. However, only one of these addresses can be defined as the **primary address;** the rest are alternative addresses. The primary address is defined during association establishment. The interesting point is that the primary address of an end is determined by the other end. In other words, a source defines the primary address for a destination.

**Multistream Delivery** One interesting feature of SCTP is the distinction between data transfer and data delivery. SCTP uses TSN numbers to handle data transfer, movement of data chunks between the source and destination. The delivery of the data chunks is controlled by SIs and SSNs. SCTP can support multiple streams, which means that the sender process can define different streams and a message can belong to one of these streams. Each stream is assigned a stream identifier (SI) which uniquely defines that stream.

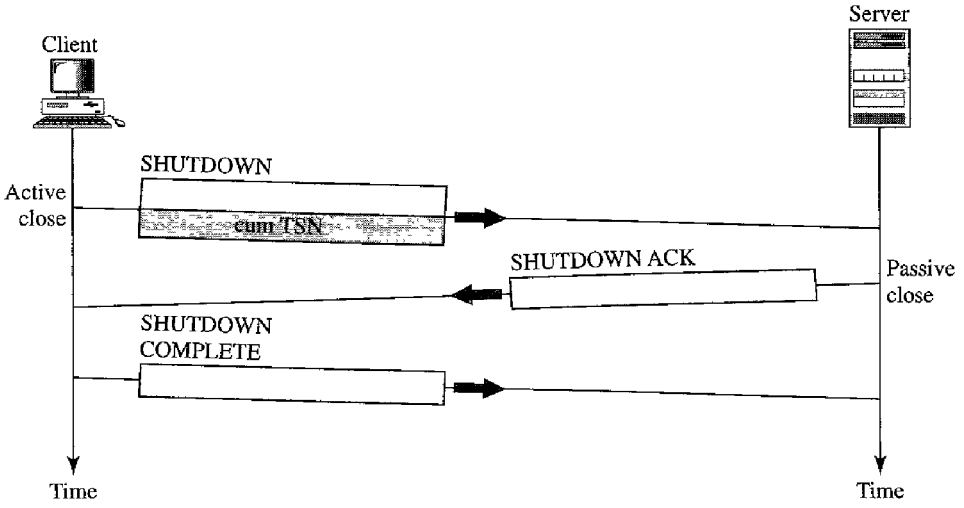
**Fragmentation** Another issue in data transfer is **fragmentation.** Although SCTP shares this term with IP, fragmentation in IP and in SCTP belongs to different levels: the former at the network layer, the latter at the transport layer.

SCTP preserves the boundaries of the message from process to process when creating a DATA chunk from a message if the size of the message (when encapsulated in an IP datagram) does not exceed the MTU of the path. The size of an IP datagram carrying a message can be determined by adding the size of the message, in bytes, to the four overheads: data chunk header, necessary SACK chunks, SCTP general header, and IP header. If the total size exceeds the MTU, the message needs to be fragmented.

### *Association Termination*

In SCTP, like TCP, either of the two parties involved in exchanging data (client or server) can close the connection. However, unlike TCP, SCTP does not allow a half-close situation. If one end closes the association, the other end must stop sending new data. If any data are left over in the queue of the recipient of the termination request, they are sent and the association is closed. **Association termination** uses three packets, as shown in Figure 23.35. Note that although the figure shows the case in which termination is initiated by the client, it can also be initiated by the server. Note that there can be several scenarios of association termination. We leave this discussion to the references mentioned at the end of the chapter.

**Figure 23.35** *Association termination*




## Flow Control

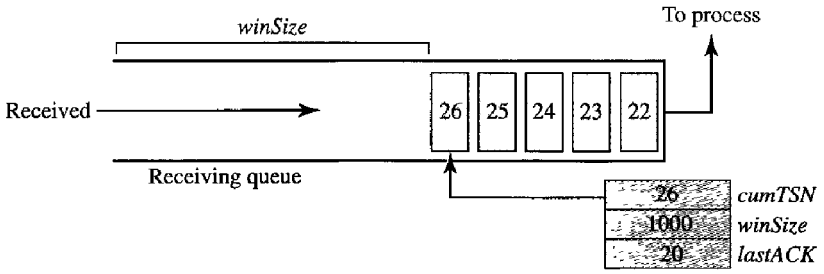
Flow control in SCTP is similar to that in TCP. In TCP, we need to deal with only one unit of data, the byte. In SCTP, we need to handle two units of data, the byte and the chunk. The values of *rwnd* and *cwnd* are expressed in bytes; the values of TSN and acknowledgments are expressed in chunks. To show the concept, we make some unrealistic assumptions. We assume that there is never congestion in the network and that the network is error-free. In other words, we assume that cwnd is infinite and no packet is lost or delayed or arrives out of order. We also assume that data transfer is unidirectional. We correct our unrealistic assumptions in later sections. Current SCTP implementations still use a byte-oriented window for flow control. We, however, show the buffer in terms of chunks to make the concept easier to understand.

### *Receiver Site*

The receiver has one buffer (queue) and three variables. The queue holds the received data chunks that have not yet been read by the process. The first variable holds the last TSN received, *cumTSN*. The second variable holds the available buffer size, *winsize*.

The third variable holds the last accumulative acknowledgment, *lastACK*. Figure 23.36 shows the queue and variables at the receiver site.

**Figure 23.36** *Flow control, receiver site*

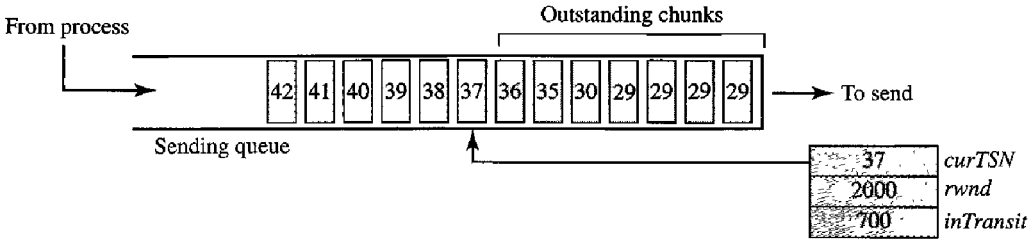



1. When the site receives a data chunk, it stores it at the end of the buffer (queue) and subtracts the size of the chunk from *winSize*. The TSN number of the chunk is stored in the *cumTSN* variable.
2. When the process reads a chunk, it removes it from the queue and adds the size of the removed chunk to *winSize* (recycling).
3. When the receiver decides to send a SACK, it checks the value of *lastAck;* if it is less than *cumTSN,* it sends a SACK with a cumulative TSN number equal to the *cumTSN*. It also includes the value of *winSize* as the advertised window size.

### *Sender Site*

The sender has one buffer (queue) and three variables: *curTSN, rwnd,* and *inTransit,* as shown in Figure 23.37. We assume each chunk is 100 bytes long.

**Figure 23.37** *Flow control, sender site*



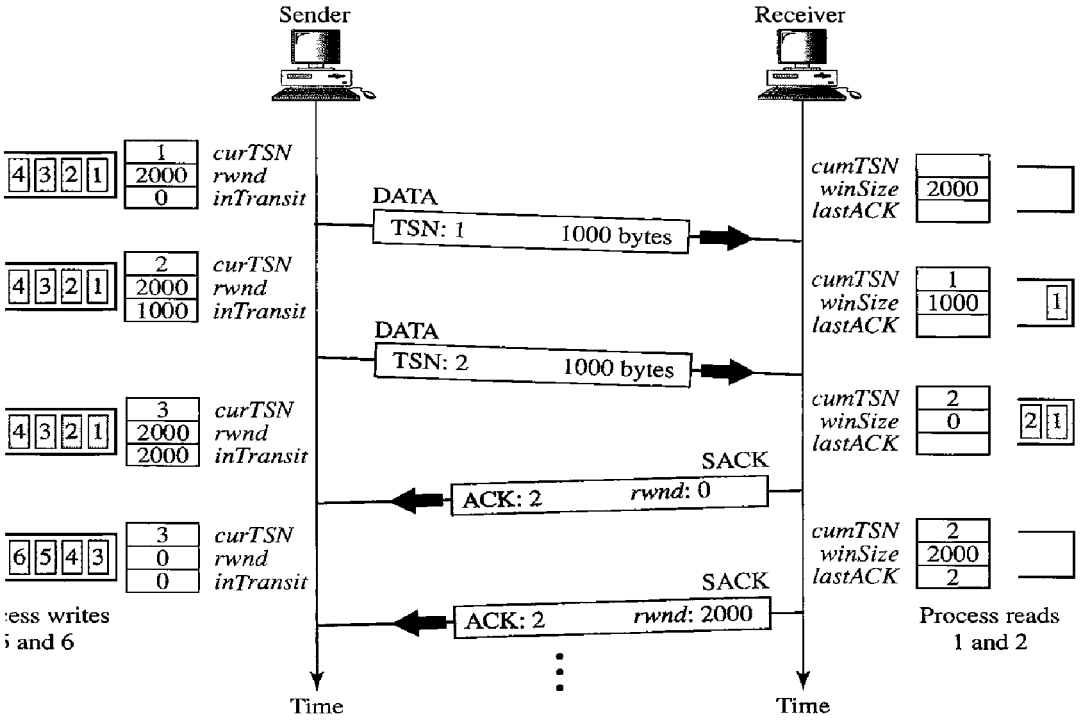

The buffer holds the chunks produced by the process that either have been sent or are ready to be sent. The first variable, *curTSN,* refers to the next chunk to be sent. All chunks in the queue with a TSN less than this value have been sent, but not acknowledged; they are outstanding. The second variable, *rwnd,* holds the last value advertised by the receiver (in bytes). The third variable, *inTransit,* holds the number of bytes in transit, bytes sent but not yet acknowledged. The following is the procedure used by the sender.

1. A chunk pointed to by *curTSN* can be sent if the size of the data is less than or equal to the quantity *rwnd – inTransit*. After sending the chunk, the value of *curTSN* is incremented by 1 and now points to the next chunk to be sent. The value of *inTransit* is incremented by the size of the data in the transmitted chunk.
2. When a SACK is received, the chunks with a TSN less than or equal to the cumulative TSN in the SACK are removed from the queue and discarded. The sender does not have to worry about them any more. The value of *inTransit* is reduced by the total size of the discarded chunks. The value of *rwnd* is updated with the value of the advertised window in the SACK.

### *A Scenario*

Let us give a simple scenario as shown in Figure 23.38. At the start the value of *rwnd* at the sender site and the value of *winSize* at the receiver site are 2000 (advertised during association establishment). Originally, there are four messages in the sender queue. The sender sends one data chunk and adds the number of bytes (1000) to the *inTransit* variable. After awhile, the sender checks the difference between the *rwnd* and *inTransit,* which is 1000 bytes, so it can send another data chunk. Now the difference between the two variables is 0 and no more data chunks can be sent. After awhile, a SACK arrives that acknowledges data chunks 1 and 2. The two chunks are removed from the queue. The value of *inTransit* is now 0. The SACK, however, advertised a receiver window of value 0, which makes the sender update *rwnd* to 0. Now the sender is blocked; it cannot send any data chunks (with one exception explained later).

**Figure 23.38** *Flow control scenario*



At the receiver site, the queue is empty at the beginning. After the first data chunk is received, there is one message in the queue and the value of *cumTSN* is 1. The value of *winSize* is reduced to 1000 because the first message occupies 1000 bytes. After the second data chunk is received, the value of window size is 0 and *cumTSN* is 2. Now, as we will see, the receiver is required to send a SACK with cumulative TSN of 2. After the first SACK was sent, the process reads the two messages, which means that there is now room in the queue; the receiver advertises the situation with a SACK to allow the sender to send more data chunks. The remaining events are not shown in the figure.

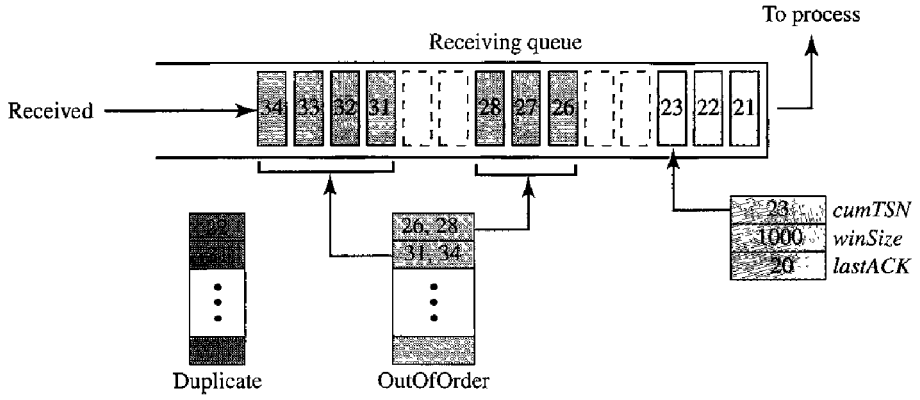
## Error Control

SCTP, like TCP, is a reliable transport layer protocol. It uses a SACK chunk to report the state of the receiver buffer to the sender. Each implementation uses a different set of entities and timers for the receiver and sender sites. We use a very simple design to convey the concept to the reader.

### *Receiver Site*

In our design, the receiver stores all chunks that have arrived in its queue including the out-of-order ones. However, it leaves spaces for any missing chunks. It discards duplicate messages, but keeps track of them for reports to the sender. Figure 23.39 shows a typical design for the receiver site and the state of the receiving queue at a particular point in time.

**Figure 23.39** *Error control, receiver site*



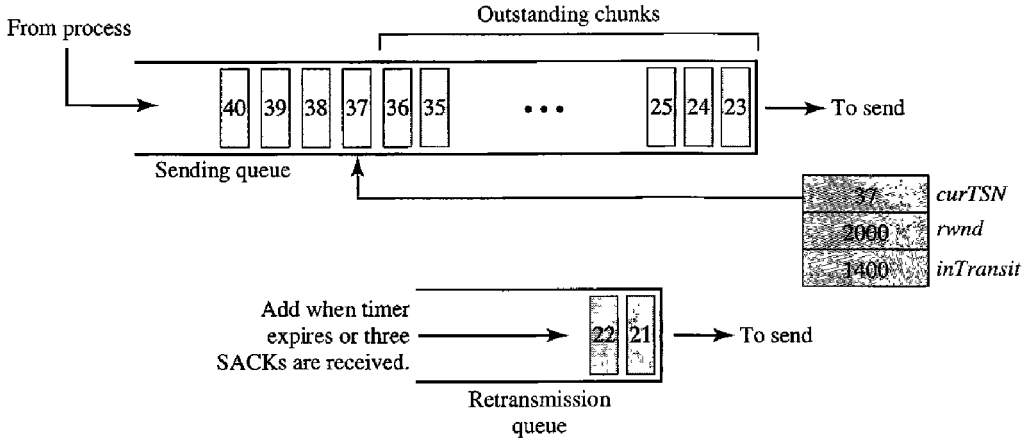

The last acknowledgment sent was for data chunk 20. The available window size is 1000 bytes. Chunks 21 to 23 have been received in order. The first out-of-order block contains chunks 26 to 28. The second out-of-order block contains chunks 31 to 34. A variable holds the value of *cumTSN*. An array of variables keeps track of the beginning and the end of each block that is out of order. An array of variables holds the duplicate chunks received. Note that there is no need for storing duplicate chunks in the queue; they will be discarded. The figure also shows the SACK chunk that will be sent to

report the state of the receiver to the sender. The TSN numbers for out-of-order chunks are relative (offsets) to the cumulative TSN.

### *Sender Site*

At the sender site, our design demands two buffers (queues): a sending queue and a retransmission queue. We also use the three variables *rwnd, inTransit,* and *curTSN* as described in the previous section. Figure 23.40 shows a typical design.

**Figure 23.40** *Error control, sender site*




The sending queue holds chunks 23 to 40. The chunks 23 to 36 have already been sent, but not acknowledged; they are outstanding chunks. The *curTSN* points to the next chunk to be sent (37). We assume that each chunk is 100 bytes, which means that 1400 bytes of data (chunks 23 to 36) is in transit. The sender at this moment has a retransmission queue. When a packet is sent, a retransmission timer starts for that packet (all data chunks in that packet). Some implementations use one single timer for the entire association, but we continue with our tradition of one timer for each packet for simplification. When the retransmission timer for a packet expires, or four duplicate SACKs arrive that declare a packet as missing (fast retransmission was discussed in Chapter 12), the chunks in that packet are moved to the retransmission queue to be resent. These chunks are considered lost, rather than outstanding. The chunks in the retransmission queue have priority. In other words, the next time the sender sends a chunk, it would be chunk 21 from the retransmission queue.

### *Sending Data Chunks*

An end can send a data packet whenever there are data chunks in the sending queue with a TSN greater than or equal to *curTSN* or if there are data chunks in the retransmission queue. The retransmission queue has priority. However, the total size of the data chunk or chunks included in the packet must not exceed *rwnd – inTransit,* and the total size of the frame must not exceed the MTU size as we discussed in previous sections.

**Retransmission** To control a lost or discarded chunk, SCTP, like TCP, employs two strategies: using retransmission timers and receiving four SACKs with the same missing chunks.

*Generating SACK Chunks*

Another issue in error control is the generation of SACK chunks. The rules for generating SCTP SACK chunks are similar to the rules used for acknowledgment with the TCP ACK flag.

### Congestion Control

SCTP, like TCP, is a transport layer protocol with packets subject to congestion in the network. The SCTP designers have used the same strategies we will describe for congestion control in Chapter 24 for TCP. SCTP has slow start (exponential increase), congestion avoidance (additive increase), and congestion detection (multiplicative decrease) phases. Like TCP, SCTP also uses fast retransmission and fast recovery.

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## 23.5 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

UDP is discussed in Chapter 11 of [For06], Chapter 11 of [Ste94], and Chapter 12 of [Com00]. TCP is discussed in Chapter 12 of [For06], Chapters 17 to 24 of [Ste94], and Chapter 13 of [Com00]. SCTP is discussed in Chapter 13 of [For06] and [SX02]. Both UDP and TCP are discussed in Chapter 6 of [Tan03].

### Sites

- www.ietf.org/rfc.html Information about RFCs

### RFCs

A discussion of UDP can be found in RFC 768.
A discussion of TCP can be found in the following RFCs:

675, 700, 721, 761, 793, 879, 896, 1078, 1106, 1110, 1144, 1145, 1146, 1263, 1323, 1337, 1379, 1644, 1693, 1901, 1905, 2001, 2018, 2488, 2580

A discussion of SCTP can be found in the following RFCs:

2960, 3257, 3284, 3285, 3286, 3309, 3436, 3554, 3708, 3758

## 23.6 KEY TERMS

acknowledgment number
association
association establishment
association termination
byte-oriented
chunk
client
client/server paradigm
connection abortion
connection-oriented service
connectionless service
connectionless, unreliable transport protocol
cookie
COOKIE ACK chunk
COOKIE ECHO chunk
cumulative TSN
DATA chunk
data transfer
denial-of-service attack
dynamic port
ephemeral port number
error control
fast retransmission
flow control
four-way handshaking
fragmentation
full-duplex service
general header
half-close
inbound stream
INIT ACK chunk
INIT chunk
initial sequence number (ISN)
message-oriented
multihoming service
multistream service
port number
primary address
process-to-process delivery
pseudoheader
queue
registered port
retransmission time-out (RTO)
retransmission timer
round-trip time (RTT)
SACK chunk
segment
sequence number
server
simultaneous close
simultaneous open
socket address
stream identifier (SI)
stream sequence number (SSN)
SYN flooding attack
three-way handshaking
Transmission Control Protocol (TCP)
transmission sequence number (TSN)
transport layer
user datagram
User Datagram Protocol (UDP)
verification tag
well-known port number

## 23.7 SUMMARY

- In the client/server paradigm, an application program on the local host, called the client, needs services from an application program on the remote host, called a server.

- Each application program has a port number that distinguishes it from other programs running at the same time on the same machine.
- The client program is assigned a random port number called an ephemeral port number; the server program is assigned a universal port number called a well-known port number.
- The ICANN has specified ranges for the different types of port numbers.
- The combination of the IP address and the port number, called the socket address, defines a process and a host.
- UDP is a connectionless, unreliable transport layer protocol with no embedded flow or error control mechanism except the checksum for error detection.
- The UDP packet is called a user datagram. A user datagram is encapsulated in the data field of an IP datagram.
- Transmission Control Protocol (TCP) is one of the transport layer protocols in the TCP/IP protocol suite.
- TCP provides process-to-process, full-duplex, and connection-oriented service.
- The unit of data transfer between two devices using TCP software is called a segment; it has 20 to 60 bytes of header, followed by data from the application program.
- A TCP connection normally consists of three phases: connection establishment, data transfer, and connection termination.
- Connection establishment requires three-way handshaking; connection termination requires three- or four-way handshaking.
- TCP uses flow control, implemented as a sliding window mechanism, to avoid overwhelming a receiver with data.
- The TCP window size is determined by the receiver-advertised window size (*rwnd*) or the congestion window size (*cwnd*), whichever is smaller. The window can be opened or closed by the receiver, but should not be shrunk.
- The bytes of data being transferred in each connection are numbered by TCP. The numbering starts with a randomly generated number.
- TCP uses error control to provide a reliable service. Error control is handled by the checksum, acknowledgment, and time-out. Corrupted and lost segments are retransmitted, and duplicate segments are discarded. Data may arrive out of order and are temporarily stored by the receiving TCP, but TCP guarantees that no out-of-order segment is delivered to the process.
- In modern implementations, a retransmission occurs if the retransmission timer expires or three duplicate ACK segments have arrived.
- SCTP is a message-oriented, reliable protocol that combines the good features of UDP and TCP.
- SCTP provides additional services not provided by UDP or TCP, such as multiple-stream and multihoming services.
- SCTP is a connection-oriented protocol. An SCTP connection is called an association.
- SCTP uses the term *packet* to define a transportation unit.
- In SCTP, control information and data information are carried in separate chunks.

- An SCTP packet can contain control chunks and data chunks with control chunks coming before data chunks.
- In SCTP, each data chunk is numbered using a transmission sequence number (TSN).
- To distinguish between different streams, SCTP uses the sequence identifier (SI).
- To distinguish between different data chunks belonging to the same stream, SCTP uses the stream sequence number (SSN).
- Data chunks are identified by three identifiers: TSN, SI, and SSN. TSN is a cumulative number recognized by the whole association; SSN starts from 0 in each stream.
- SCTP acknowledgment numbers are used only to acknowledge data chunks; control chunks are acknowledged, if needed, by another control chunk.
- An SCTP association is normally established using four packets (four-way handshaking). An association is normally terminated using three packets (three-way handshaking).
- An SCTP association uses a cookie to prevent blind flooding attacks and a verification tag to avoid insertion attacks.
- SCTP provides flow control, error control, and congestion control.
- The SCTP acknowledgment SACK reports the cumulative TSN, the TSN of the last data chunk received in order, and selective TSNs that have been received.

## 23.8 PRACTICE SET

### Review Questions

1. In cases where reliability is not of primary importance, UDP would make a good transport protocol. Give examples of specific cases.
2. Are both UDP and IP unreliable to the same degree? Why or why not?
3. Do port addresses need to be unique? Why or why not? Why are port addresses shorter than IP addresses?
4. What is the dictionary definition of the word *ephemeral?* How does it apply to the concept of the ephemeral port number?
5. What is the minimum size of a UDP datagram?
6. What is the maximum size of a UDP datagram?
7. What is the minimum size of the process data that can be encapsulated in a UDP datagram?
8. What is the maximum size of the process data that can be encapsulated in a UDP datagram?
9. Compare the TCP header and the UDP header. List the fields in the TCP header that are missing from UDP header. Give the reason for their absence.
10. UDP is a message-oriented protocol. TCP is a byte-oriented protocol. If an application needs to protect the boundaries of its message, which protocol should be used, UDP or TCP?

11. What can you say about the TCP segment in which the value of the control field is one of the following?
    a. 000000
    b. 000001
    c. 010001
12. What is the maximum size of the TCP header? What is the minimum size of the TCP header?

## Exercises

13. Show the entries for the header of a UDP user datagram that carries a message from a TFTP client to a TFTP server. Fill the checksum field with 0s. Choose an appropriate ephemeral port number and the correct well-known port number. The length of data is 40 bytes. Show the UDP packet, using the format in Figure 23.9.
14. An SNMP client residing on a host with IP address 122.45.12.7 sends a message to an SNMP server residing on a host with IP address 200.112.45.90. What is the pair of sockets used in this communication?
15. A TFTP server residing on a host with IP address 130.45.12.7 sends a message to a TFTP client residing on a host with IP address 14.90.90.33. What is the pair of sockets used in this communication?
16. A client has a packet of 68,000 bytes. Show how this packet can be transferred by using only one UDP user datagram.
17. A client uses UDP to send data to a server. The data are 16 bytes. Calculate the efficiency of this transmission at the UDP level (ratio of useful bytes to total bytes).
18. Redo Exercise 17, calculating the efficiency of transmission at the IP level. Assume no options for the IP header.
19. Redo Exercise 18, calculating the efficiency of transmission at the data link layer. Assume no options for the IP header and use Ethernet at the data link layer.
20. The following is a dump of a UDP header in hexadecimal format.

    06 32 00 0D 00 1C E2 17

    a. What is the source port number?
    b. What is the destination port number?
    c. What is the total length of the user datagram?
    d. What is the length of the data?
    e. Is the packet directed from a client to a server or vice versa?
    f. What is the client process?
21. An IP datagram is carrying a TCP segment destined for address 130.14.16.17/16. The destination port address is corrupted, and it arrives at destination 130.14.16.19/16. How does the receiving TCP react to this error?
22. In TCP, if the value of HLEN is 0111, how many bytes of option are included in the segment?

23. Show the entries for the header of a TCP segment that carries a message from an FTP client to an FTP server. Fill the checksum field with 0s. Choose an appropriate ephemeral port number and the correct well-known port number. The length of the data is 40 bytes.
24. The following is a dump of a TCP header in hexadecimal format.

    05320017 00000001 00000000 500207FF 00000000

    a. What is the source port number?
    b. What is the destination port number?
    c. What the sequence number?
    d. What is the acknowledgment number?
    e. What is the length of the header?
    f. What is the type of the segment?
    g. What is the window size?
25. To make the initial sequence number a random number, most systems start the counter at 1 during bootstrap and increment the counter by 64,000 every 0.5 s. How long does it take for the counter to wrap around?
26. In a connection, the value of *cwnd* is 3000 and the value of *rwnd* is 5000. The host has sent 2000 bytes which has not been acknowledged. How many more bytes can be sent?
27. TCP opens a connection using an initial sequence number (ISN) of 14,534. The other party opens the connection with an ISN of 21,732. Show the three TCP segments during the connection establishment.
28. A client uses TCP to send data to a server. The data are 16 bytes. Calculate the efficiency of this transmission at the TCP level (ratio of useful bytes to total bytes). Calculate the efficiency of transmission at the IP level. Assume no options for the IP header. Calculate the efficiency of transmission at the data link layer. Assume no options for the IP header and use Ethernet at the data link layer.
29. TCP is sending data at 1 Mbyte/s. If the sequence number starts with 7000, how long does it take before the sequence number goes back to zero?
30. A TCP connection is using a window size of 10,000 bytes, and the previous acknowledgment number was 22,001. It receives a segment with acknowledgment number 24,001 and window size advertisement of 12,000. Draw a diagram to show the situation of the window before and after.
31. A window holds bytes 2001 to 5000. The next byte to be sent is 3001. Draw a figure to show the situation of the window after the following two events.
    a. An ACK segment with the acknowledgment number 2500 and window size advertisement 4000 is received.
    b. A segment carrying 1000 bytes is sent.
32. In SCTP, the value of the cumulative TSN in a SACK is 23. The value of the previous cumulative TSN in the SACK was 29. What is the problem?
33. In SCTP, the state of a receiver is as follows:
    a. The receiving queue has chunks 1 to 8, 11 to 14, and 16 to 20.
    b. There are 1800 bytes of space in the queue.

c. The value of *lastAck* is 4.

d. No duplicate chunk has been received.

e. The value of *cumTSN* is 5.

Show the contents of the receiving queue and the variables.

34. In SCTP, the state of a sender is as follows:

a. The sending queue has chunks 18 to 23.

b. The value of *cumTSN* is 20.

c. The value of the window size is 2000 bytes.

d. The value of *inTransit* is 200.

If each data chunk contains 100 bytes of data, how many DATA chunks can be sent now? What is the next DATA chunk to be sent?

## Research Activities

35. Find more information about ICANN. What was it called before its name was changed?
36. TCP uses a transition state diagram to handle sending and receiving segments. Find out about this diagram and how it handles flow and control.
37. SCTP uses a transition state diagram to handle sending and receiving segments. Find out about this diagram and how it handles flow and control.
38. What is the half-open case in TCP?
39. What is the half-duplex close case in TCP?
40. The *tcpdump* command in UNIX or LINUX can be used to print the headers of packets of a network interface. Use *tcpdump* to see the segments sent and received.
41. In SCTP, find out what happens if a SACK chunk is delayed or lost.
42. Find the name and functions of timers used in TCP.
43. Find the name and functions of timers used in SCTP.
44. Find out more about ECN in SCTP. Find the format of these two chunks.
45. Some application programs, such as FTP, need more than one connection when using TCP. Find how the multistream service of SCTP can help these applications establish only one association with several streams.

# CHAPTER 24

# *Congestion Control and Quality of Service*

Congestion control and quality of service are two issues so closely bound together that improving one means improving the other and ignoring one usually means ignoring the other. Most techniques to prevent or eliminate congestion also improve the quality of service in a network.

We have postponed the discussion of these issues until now because these are issues related not to one layer, but to three: the data link layer, the network layer, and the transport layer. We waited until now so that we can discuss these issues once instead of repeating the subject three times. Throughout the chapter, we give examples of congestion control and quality of service at different layers.

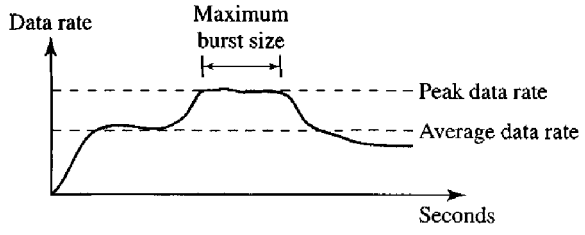
## 24.1 DATA TRAFFIC

The main focus of congestion control and quality of service is data traffic. In congestion control we try to avoid traffic congestion. In quality of service, we try to create an appropriate environment for the traffic. So, before talking about congestion control and quality of service, we discuss the data traffic itself.

### Traffic Descriptor

Traffic descriptors are qualitative values that represent a data flow. Figure 24.1 shows a traffic flow with some of these values.

**Figure 24.1** *Traffic descriptors*



### *Average Data Rate*

The **average data rate** is the number of bits sent during a period of time, divided by the number of seconds in that period. We use the following equation:

$$\text{Average data rate} = \frac{\text{amount of data}}{\text{time}}$$

The average data rate is a very useful characteristic of traffic because it indicates the average bandwidth needed by the traffic.

### *Peak Data Rate*

The **peak data rate** defines the maximum data rate of the traffic. In Figure 24.1 it is the maximum $y$ axis value. The peak data rate is a very important measurement because it indicates the peak bandwidth that the network needs for traffic to pass through without changing its data flow.

### *Maximum Burst Size*

Although the peak data rate is a critical value for the network, it can usually be ignored if the duration of the peak value is very short. For example, if data are flowing steadily at the rate of 1 Mbps with a sudden peak data rate of 2 Mbps for just 1 ms, the network probably can handle the situation. However, if the peak data rate lasts 60 ms, there may be a problem for the network. The **maximum burst size** normally refers to the maximum length of time the traffic is generated at the peak rate.

### *Effective Bandwidth*

The **effective bandwidth** is the bandwidth that the network needs to allocate for the flow of traffic. The effective bandwidth is a function of three values: average data rate, peak data rate, and maximum burst size. The calculation of this value is very complex.

## Traffic Profiles

For our purposes, a data flow can have one of the following traffic profiles: constant bit rate, variable bit rate, or bursty as shown in Figure 24.2.

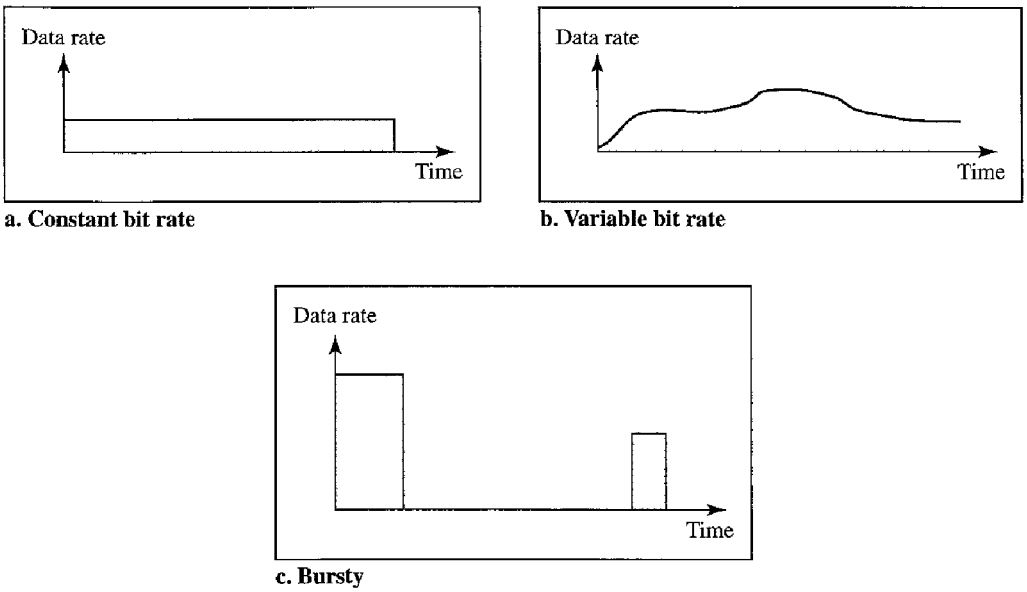
### *Constant Bit Rate*

A **constant-bit-rate (CBR),** or a fixed-rate, traffic model has a data rate that does not change. In this type of flow, the average data rate and the peak data rate are the same. The maximum burst size is not applicable. This type of traffic is very easy for a network to handle since it is predictable. The network knows in advance how much bandwidth to allocate for this type of flow.

### *Variable Bit Rate*

In the **variable-bit-rate (VBR)** category, the rate of the data flow changes in time, with the changes smooth instead of sudden and sharp. In this type of flow, the average data

**Figure 24.2** *Three traffic profiles*




rate and the peak data rate are different. The maximum burst size is usually a small value. This type of traffic is more difficult to handle than constant-bit-rate traffic, but it normally does not need to be reshaped, as we will see later.

### *Bursty*

In the **bursty data** category, the data rate changes suddenly in a very short time. It may jump from zero, for example, to 1 Mbps in a few microseconds and vice versa. It may also remain at this value for a while. The average bit rate and the peak bit rate are very different values in this type of flow. The maximum burst size is significant. This is the most difficult type of traffic for a network to handle because the profile is very unpredictable. To handle this type of traffic, the network normally needs to reshape it, using reshaping techniques, as we will see shortly. Bursty traffic is one of the main causes of congestion in a network.

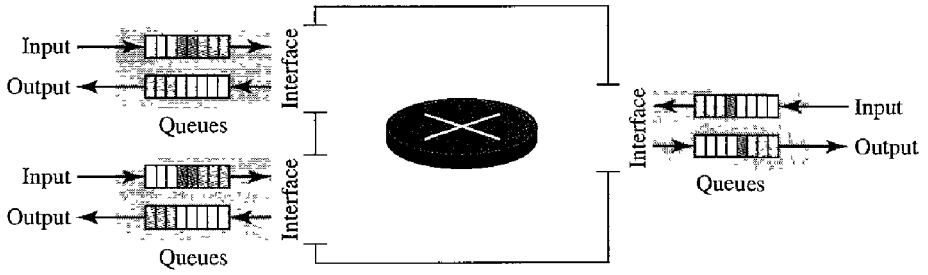
## 24.2 CONGESTION

An important issue in a packet-switched network is **congestion.** Congestion in a network may occur if the **load** on the network—the number of packets sent to the network—is greater than the *capacity* of the network—the number of packets a network can handle. **Congestion control** refers to the mechanisms and techniques to control the congestion and keep the load below the capacity.

We may ask why there is congestion on a network. Congestion happens in any system that involves waiting. For example, congestion happens on a freeway because any abnormality in the flow, such as an accident during rush hour, creates blockage.

Congestion in a network or internetwork occurs because routers and switches have queues—buffers that hold the packets before and after processing. A router, for example, has an input queue and an output queue for each interface. When a packet arrives at the incoming interface, it undergoes three steps before departing, as shown in Figure 24.3.

**Figure 24.3** *Queues in a router*



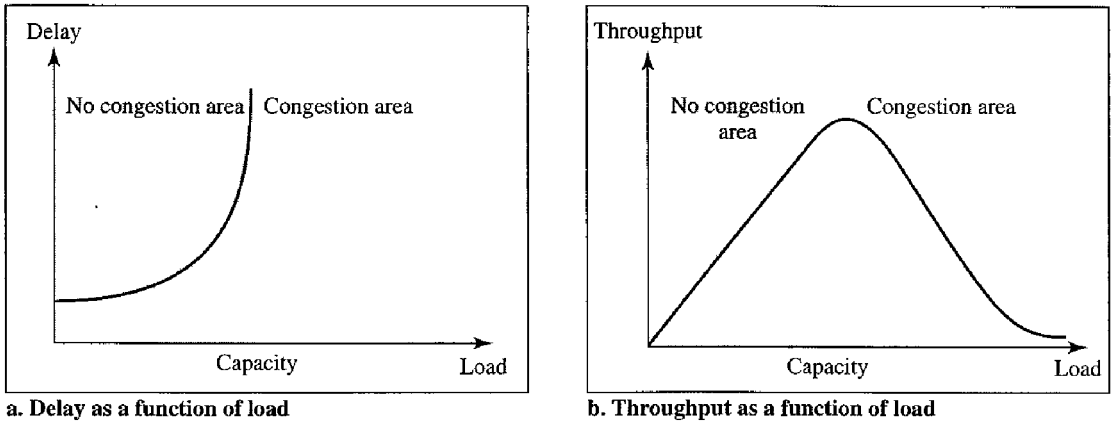

1. The packet is put at the end of the input queue while waiting to be checked.
2. The processing module of the router removes the packet from the input queue once it reaches the front of the queue and uses its routing table and the destination address to find the route.
3. The packet is put in the appropriate output queue and waits its turn to be sent.

We need to be aware of two issues. First, if the rate of packet arrival is higher than the packet processing rate, the input queues become longer and longer. Second, if the packet departure rate is less than the packet processing rate, the output queues become longer and longer.

## Network Performance

Congestion control involves two factors that measure the performance of a network: *delay* and *throughput*. Figure 24.4 shows these two performance measures as function of load.

**Figure 24.4** *Packet delay and throughput as functions of load*




**a. Delay as a function of load**

**b. Throughput as a function of load**

### *Delay Versus Load*

Note that when the load is much less than the capacity of the network, the **delay** is at a minimum. This minimum delay is composed of propagation delay and processing delay, both of which are negligible. However, when the load reaches the network capacity, the delay increases sharply because we now need to add the waiting time in the queues (for all routers in the path) to the total delay. Note that the delay becomes infinite when the load is greater than the capacity. If this is not obvious, consider the size of the queues when almost no packet reaches the destination, or reaches the destination with infinite delay; the queues become longer and longer. Delay has a negative effect on the load and consequently the congestion. When a packet is delayed, the source, not receiving the acknowledgment, retransmits the packet, which makes the delay, and the congestion, worse.

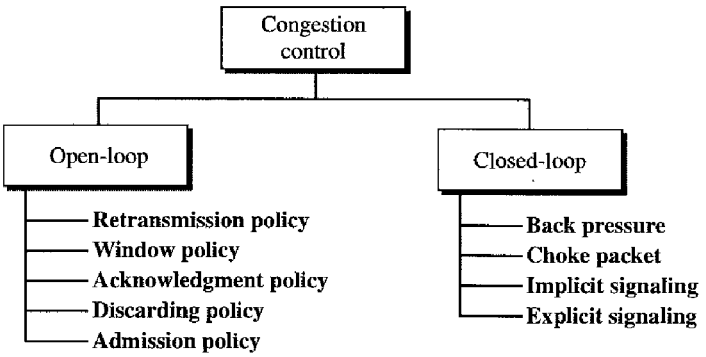
### *Throughput Versus Load*

We defined throughput in Chapter 3 as the number of bits passing through a point in a second. We can extend that definition from bits to packets and from a point to a network. We can define **throughput** in a network as the number of packets passing through the network in a unit of time. Notice that when the load is below the capacity of the network, the throughput increases proportionally with the *load*. We expect the throughput to remain constant after the load reaches the capacity, but instead the throughput declines sharply. The reason is the discarding of packets by the routers. When the load exceeds the capacity, the queues become full and the routers have to discard some packets. Discarding packets does not reduce the number of packets in the network because the sources retransmit the packets, using time-out mechanisms, when the packets do not reach the destinations.

## 24.3 CONGESTION CONTROL

Congestion control refers to techniques and mechanisms that can either prevent congestion, before it happens, or remove congestion, after it has happened. In general, we can divide congestion control mechanisms into two broad categories: open-loop congestion control (prevention) and closed-loop congestion control (removal) as shown in Figure 24.5.

**Figure 24.5** *Congestion control categories*



## Open-Loop Congestion Control

In **open-loop congestion control,** policies are applied to prevent congestion before it happens. In these mechanisms, congestion control is handled by either the source or the destination. We give a brief list of policies that can prevent congestion.

### *Retransmission Policy*

Retransmission is sometimes unavoidable. If the sender feels that a sent packet is lost or corrupted, the packet needs to be retransmitted. Retransmission in general may increase congestion in the network. However, a good retransmission policy can prevent congestion. The retransmission policy and the retransmission timers must be designed to optimize efficiency and at the same time prevent congestion. For example, the retransmission policy used by TCP (explained later) is designed to prevent or alleviate congestion.

### *Window Policy*

The type of window at the sender may also affect congestion. The Selective Repeat window is better than the Go-Back-*N* window for congestion control. In the Go-Back-*N* window, when the timer for a packet times out, several packets may be resent, although some may have arrived safe and sound at the receiver. This duplication may make the congestion worse. The Selective Repeat window, on the other hand, tries to send the specific packets that have been lost or corrupted.

### *Acknowledgment Policy*

The acknowledgment policy imposed by the receiver may also affect congestion. If the receiver does not acknowledge every packet it receives, it may slow down the sender and help prevent congestion. Several approaches are used in this case. A receiver may send an acknowledgment only if it has a packet to be sent or a special timer expires. A receiver may decide to acknowledge only *N* packets at a time. We need to know that the acknowledgments are also part of the load in a network. Sending fewer acknowledgments means imposing less load on the network.

### *Discarding Policy*

A good discarding policy by the routers may prevent congestion and at the same time may not harm the integrity of the transmission. For example, in audio transmission, if the policy is to discard less sensitive packets when congestion is likely to happen, the quality of sound is still preserved and congestion is prevented or alleviated.

### *Admission Policy*

An admission policy, which is a quality-of-service mechanism, can also prevent congestion in virtual-circuit networks. Switches in a flow first check the resource requirement of a flow before admitting it to the network. A router can deny establishing a virtual-circuit connection if there is congestion in the network or if there is a possibility of future congestion.

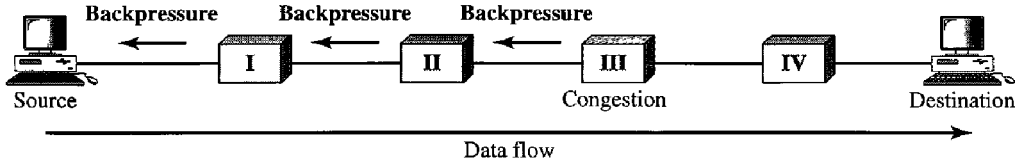
## Closed-Loop Congestion Control

**Closed-loop congestion control** mechanisms try to alleviate congestion after it happens. Several mechanisms have been used by different protocols. We describe a few of them here.

### *Backpressure*

The technique of *backpressure* refers to a congestion control mechanism in which a congested node stops receiving data from the immediate upstream node or nodes. This may cause the upstream node or nodes to become congested, and they, in turn, reject data from their upstream nodes or nodes. And so on. Backpressure is a node-to-node congestion control that starts with a node and propagates, in the opposite direction of data flow, to the source. The backpressure technique can be applied only to virtual circuit networks, in which each node knows the upstream node from which a flow of data is coming. Figure 24.6 shows the idea of backpressure.

**Figure 24.6** *Backpressure method for alleviating congestion*




Node III in the figure has more input data than it can handle. It drops some packets in its input buffer and informs node II to slow down. Node II, in turn, may be congested because it is slowing down the output flow of data. If node II is congested, it informs node I to slow down, which in turn may create congestion. If so, node I informs the source of data to slow down. This, in time, alleviates the congestion. Note that the *pressure* on node III is moved backward to the source to remove the congestion.

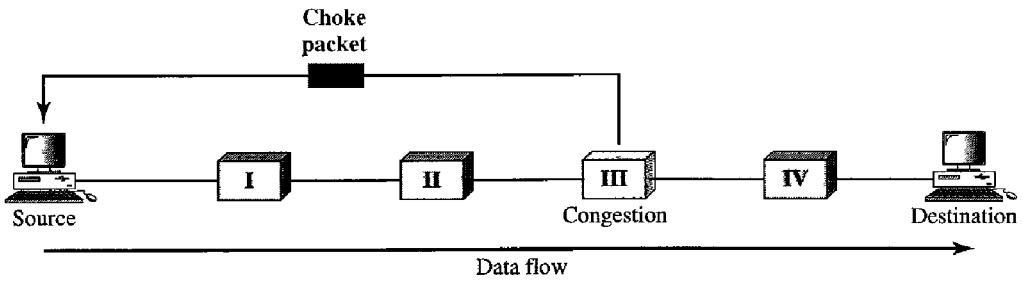
None of the virtual-circuit networks we studied in this book use backpressure. It was, however, implemented in the first virtual-circuit network, X.25. The technique cannot be implemented in a datagram network because in this type of network, a node (router) does not have the slightest knowledge of the upstream router.

### *Choke Packet*

A **choke packet** is a packet sent by a node to the source to inform it of congestion. Note the difference between the backpressure and choke packet methods. In backpressure, the warning is from one node to its upstream node, although the warning may eventually reach the source station. In the choke packet method, the warning is from the router, which has encountered congestion, to the source station directly. The intermediate nodes through which the packet has traveled are not warned. We have seen an example of this type of control in ICMP. When a router in the Internet is overwhelmed with IP datagrams, it may discard some of them; but it informs the source

host, using a source quench ICMP message. The warning message goes directly to the source station; the intermediate routers, and does not take any action. Figure 24.7 shows the idea of a choke packet.

**Figure 24.7** *Choke packet*




### *Implicit Signaling*

In implicit signaling, there is no communication between the congested node or nodes and the source. The source guesses that there is a congestion somewhere in the network from other symptoms. For example, when a source sends several packets and there is no acknowledgment for a while, one assumption is that the network is congested. The delay in receiving an acknowledgment is interpreted as congestion in the network; the source should slow down. We will see this type of signaling when we discuss TCP congestion control later in the chapter.

### *Explicit Signaling*

The node that experiences congestion can explicitly send a signal to the source or destination. The explicit signaling method, however, is different from the choke packet method. In the choke packet method, a separate packet is used for this purpose; in the explicit signaling method, the signal is included in the packets that carry data. Explicit signaling, as we will see in Frame Relay congestion control, can occur in either the forward or the backward direction.

**Backward Signaling** A bit can be set in a packet moving in the direction opposite to the congestion. This bit can warn the source that there is congestion and that it needs to slow down to avoid the discarding of packets.

**Forward Signaling** A bit can be set in a packet moving in the direction of the congestion. This bit can warn the destination that there is congestion. The receiver in this case can use policies, such as slowing down the acknowledgments, to alleviate the congestion.

## 24.4 TWO EXAMPLES

To better understand the concept of congestion control, let us give two examples: one in TCP and the other in Frame Relay.

## Congestion Control in TCP

We discussed TCP in Chapter 23. We now show how TCP uses congestion control to avoid congestion or alleviate congestion in the network.

### *Congestion Window*

In Chapter 23, we talked about flow control and tried to discuss solutions when the receiver is overwhelmed with data. We said that the sender window size is determined by the available buffer space in the receiver (*rwnd*). In other words, we assumed that it is only the receiver that can dictate to the sender the size of the sender's window. We totally ignored another entity here—the network. If the network cannot deliver the data as fast as they are created by the sender, it must tell the sender to slow down. In other words, in addition to the receiver, the network is a second entity that determines the size of the sender's window.

Today, the sender's window size is determined not only by the receiver but also by congestion in the network.

The sender has two pieces of information: the receiver-advertised window size and the congestion window size. The actual size of the window is the minimum of these two.

**Actual window size = minimum (rwnd, cwnd)**

We show shortly how the size of the congestion window (*cwnd*) is determined.

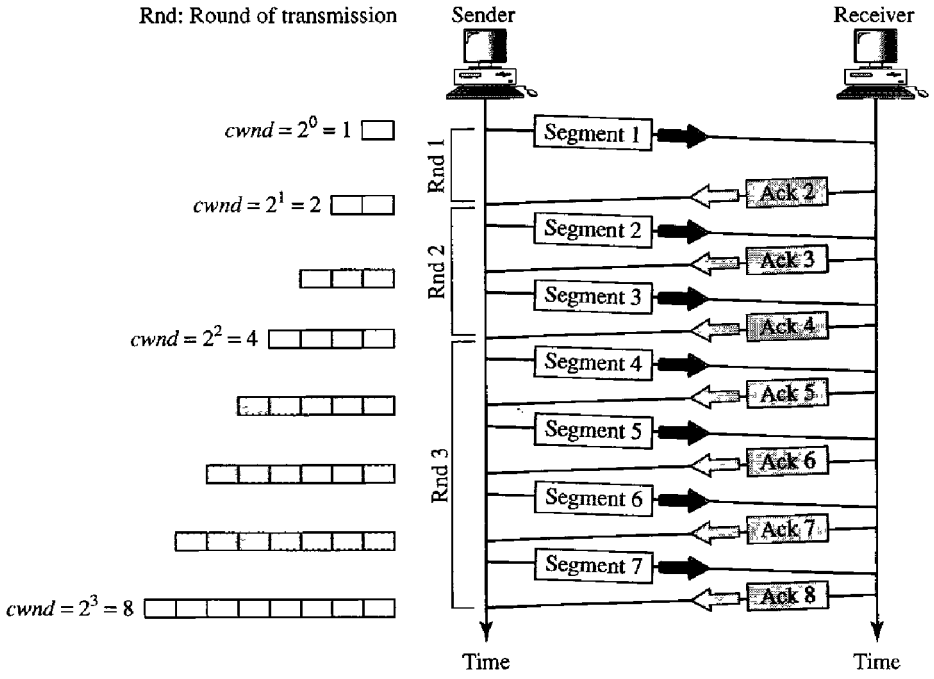
### *Congestion Policy*

TCP's general policy for handling congestion is based on three phases: slow start, congestion avoidance, and congestion detection. In the slow-start phase, the sender starts with a very slow rate of transmission, but increases the rate rapidly to reach a threshold. When the threshold is reached, the data rate is reduced to avoid congestion. Finally if congestion is detected, the sender goes back to the slow-start or congestion avoidance phase based on how the congestion is detected.

**Slow Start: Exponential Increase** One of the algorithms used in TCP congestion control is called **slow start.** This algorithm is based on the idea that the size of the congestion window (*cwnd*) starts with one maximum segment size (MSS). The MSS is determined during connection establishment by using an option of the same name. The size of the window increases one MSS each time an acknowledgment is received. As the name implies, the window starts slowly, but grows exponentially. To show the idea, let us look at Figure 24.8. Note that we have used three simplifications to make the discussion more understandable. We have used segment numbers instead of byte numbers (as though each segment contains only 1 byte). We have assumed that *rwnd* is much higher than *cwnd,* so that the sender window size always equals *cwnd.* We have assumed that each segment is acknowledged individually.

The sender starts with *cwnd* = 1 MSS. This means that the sender can send only one segment. After receipt of the acknowledgment for segment 1, the size of the congestion window is increased by 1, which means that *cwnd* is now 2. Now two more segments can be sent. When each acknowledgment is received, the size of the window is increased by 1 MSS. When all seven segments are acknowledged, *cwnd* = 8.

**Figure 24.8** *Slow start, exponential increase*




If we look at the size of *cwnd* in terms of rounds (acknowledgment of the whole window of segments), we find that the rate is exponential as shown below:

| | | |
|---|---|---|
| **Start** | ➡ | $cwnd = 1$ |
| **After round 1** | ➡ | $cwnd = 2^1 = 2$ |
| **After round 2** | ➡ | $cwnd = 2^2 = 4$ |
| **After round 3** | ➡ | $cwnd = 2^3 = 8$ |

We need to mention that if there is delayed ACKs, the increase in the size of the window is less than power of 2.

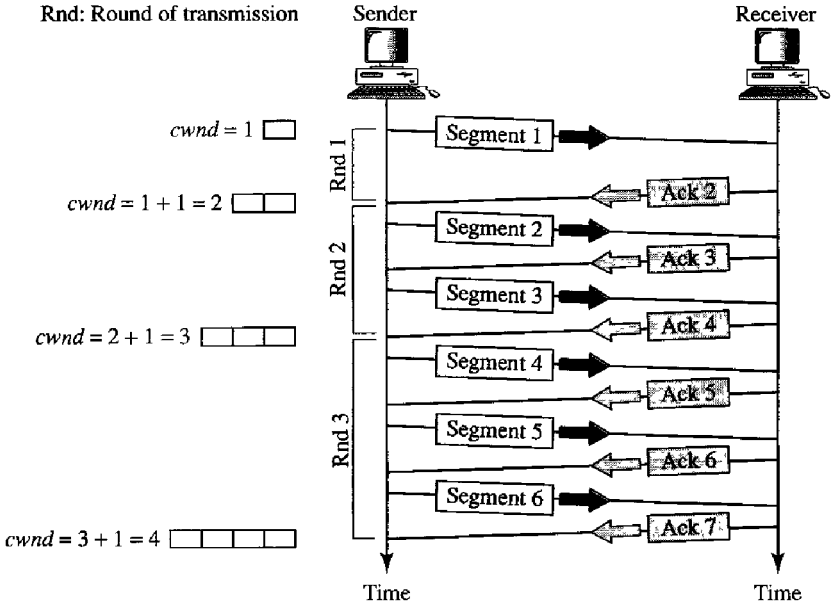
Slow start cannot continue indefinitely. There must be a threshold to stop this phase. The sender keeps track of a variable named *ssthresh* (slow-start threshold). When the size of window in bytes reaches this threshold, slow start stops and the next phase starts. In most implementations the value of *ssthresh* is 65,535 bytes.

**In the slow-start algorithm, the size of the congestion window increases exponentially until it reaches a threshold.**

**Congestion Avoidance: Additive Increase** If we start with the slow-start algorithm, the size of the congestion window increases exponentially. To avoid congestion before it happens, one must slow down this exponential growth. TCP defines another algorithm called **congestion avoidance,** which undergoes an **additive increase** instead of an exponential one. When the size of the congestion window reaches the slow-start threshold, the slow-start phase stops and the additive phase begins. In this algorithm, each time the whole window of segments is acknowledged (one round), the size of the

congestion window is increased by 1. To show the idea, we apply this algorithm to the same scenario as slow start, although we will see that the congestion avoidance algorithm usually starts when the size of the window is much greater than 1. Figure 24.9 shows the idea.

**Figure 24.9** *Congestion avoidance, additive increase*




In this case, after the sender has received acknowledgments for a complete window size of segments, the size of the window is increased by one segment.

If we look at the size of *cwnd* in terms of rounds, we find that the rate is additive as shown below:

| | | |
|---|---|---|
| **Start** | ➡ | ***cwnd* = 1** |
| **After round 1** | ➡ | ***cwnd* = 1 + 1 = 2** |
| **After round 2** | ➡ | ***cwnd* = 2 + 1 = 3** |
| **After round 3** | ➡ | ***cwnd* = 3 + 1 = 4** |

**In the congestion avoidance algorithm, the size of the congestion window increases additively until congestion is detected.**

**Congestion Detection: Multiplicative Decrease** If congestion occurs, the congestion window size must be decreased. The only way the sender can guess that congestion has occurred is by the need to retransmit a segment. However, retransmission can occur in one of two cases: when a timer times out or when three ACKs are received. In both cases, the size of the threshold is dropped to one-half, a **multiplicative decrease.** Most TCP implementations have two reactions:

1. If a time-out occurs, there is a stronger possibility of congestion; a segment has probably been dropped in the network, and there is no news about the sent segments.

In this case TCP reacts strongly:

a. It sets the value of the threshold to one-half of the current window size.
b. It sets *cwnd* to the size of one segment.
c. It starts the slow-start phase again.

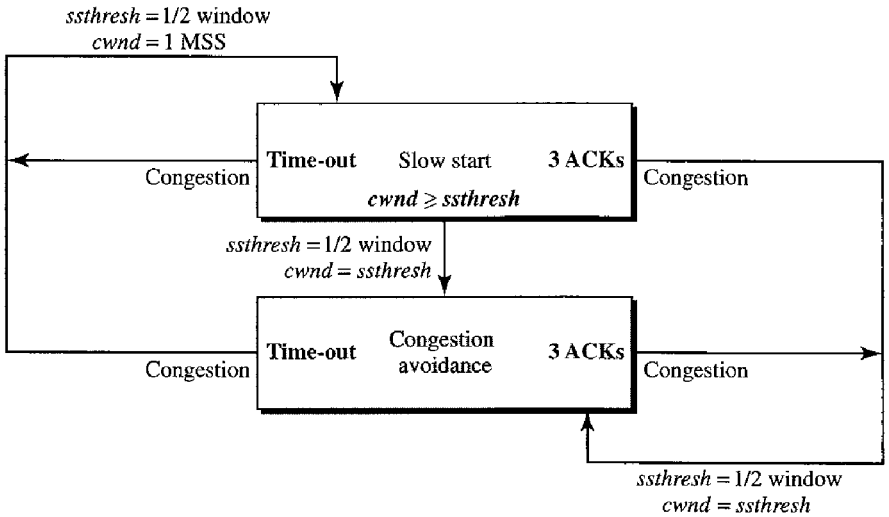
2. If three ACKs are received, there is a weaker possibility of congestion; a segment may have been dropped, but some segments after that may have arrived safely since three ACKs are received. This is called fast transmission and fast recovery. In this case, TCP has a weaker reaction:
   a. It sets the value of the threshold to one-half of the current window size.
   b. It sets *cwnd* to the value of the threshold (some implementations add three segment sizes to the threshold).
   c. It starts the congestion avoidance phase.

**An implementations reacts to congestion detection in one of the following ways:**

- If detection is by time-out, a new *slow-start* phase starts.
- If detection is by three ACKs, a new *congestion avoidance* phase starts.

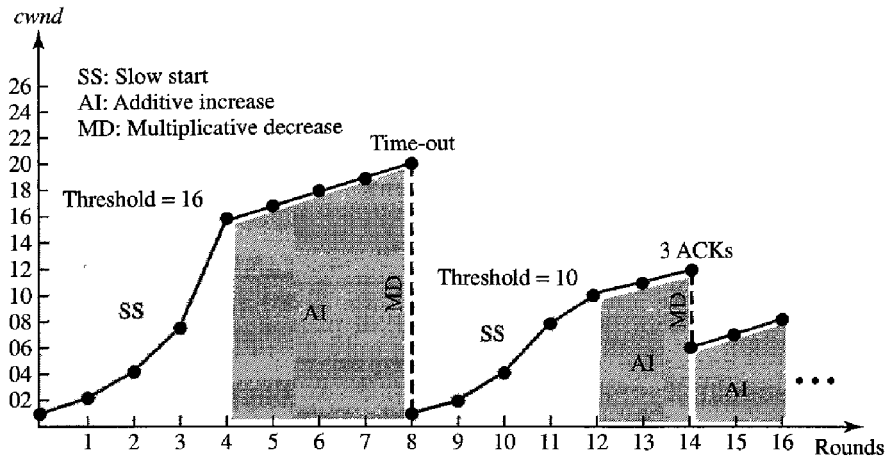
**Summary** In Figure 24.10, we summarize the congestion policy of TCP and the relationships between the three phases.

**Figure 24.10** *TCP congestion policy summary*




We give an example in Figure 24.11. We assume that the maximum window size is 32 segments. The threshold is set to 16 segments (one-half of the maximum window size). In the *slow-start* phase the window size starts from 1 and grows exponentially until it reaches the threshold. After it reaches the threshold, the *congestion avoidance (additive increase)* procedure allows the window size to increase linearly until a time-out occurs or the maximum window size is reached. In Figure 24.11, the time-out occurs when the window size is 20. At this moment, the *multiplicative decrease* procedure takes

**Figure 24.11** *Congestion example*




over and reduces the threshold to one-half of the previous window size. The previous window size was 20 when the time-out happened so the new threshold is now 10.

TCP moves to slow start again and starts with a window size of 1, and TCP moves to additive increase when the new threshold is reached. When the window size is 12, a three-ACKs event happens. The multiplicative decrease procedure takes over again. The threshold is set to 6 and TCP goes to the additive increase phase this time. It remains in this phase until another time-out or another three ACKs happen.

## Congestion Control in Frame Relay

Congestion in a Frame Relay network decreases throughput and increases delay. A high throughput and low delay are the main goals of the Frame Relay protocol. Frame Relay does not have flow control. In addition, Frame Relay allows the user to transmit bursty data. This means that a Frame Relay network has the potential to be really congested with traffic, thus requiring congestion control.

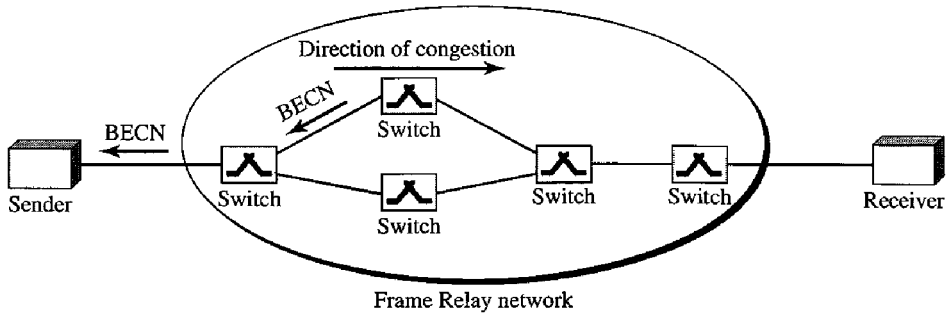
### *Congestion Avoidance*

For congestion avoidance, the Frame Relay protocol uses 2 bits in the frame to explicitly warn the source and the destination of the presence of congestion.

**BECN** The **backward explicit congestion notification (BECN)** bit warns the sender of congestion in the network. One might ask how this is accomplished since the frames are traveling away from the sender. In fact, there are two methods: The switch can use response frames from the receiver (full-duplex mode), or else the switch can use a predefined connection (DLCI = 1023) to send special frames for this specific purpose. The sender can respond to this warning by simply reducing the data rate. Figure 24.12 shows the use of BECN.

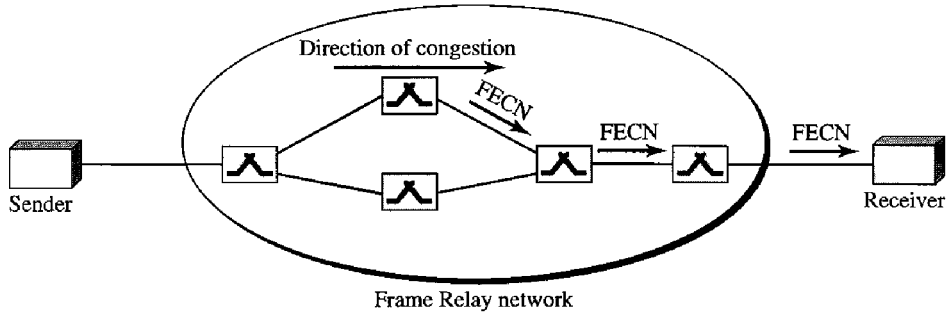
**FECN** The **forward explicit congestion notification (FECN)** bit is used to warn the receiver of congestion in the network. It might appear that the receiver cannot do anything to relieve the congestion. However, the Frame Relay protocol assumes that the sender and receiver are communicating with each other and are using some type of flow control at a higher level. For example, if there is an acknowledgment mechanism at this

**Figure 24.12** *BECN*



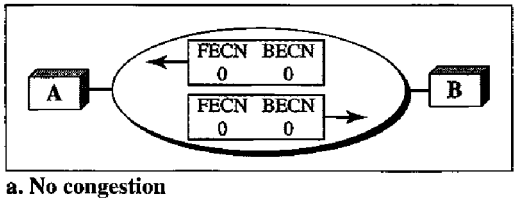

higher level, the receiver can delay the acknowledgment, thus forcing the sender to slow down. Figure 24.13 shows the use of FECN.

**Figure 24.13** *FECN*

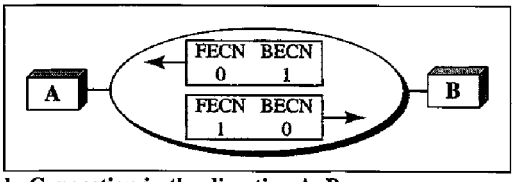



When two endpoints are communicating using a Frame Relay network, four situations may occur with regard to congestion. Figure 24.14 shows these four situations and the values of FECN and BECN.

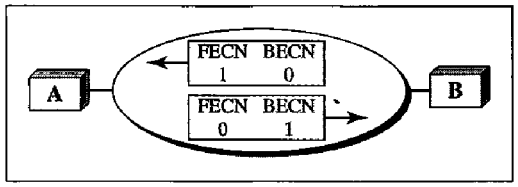
**Figure 24.14** *Four cases of congestion*



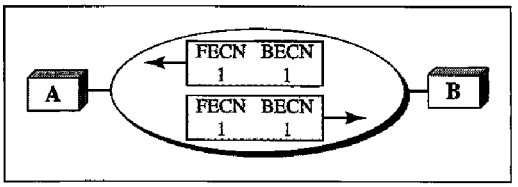

**a. No congestion**




**b. Congestion in the direction A–B**




**c. Congestion in the direction B–A**




**d. Congestion in both directions**

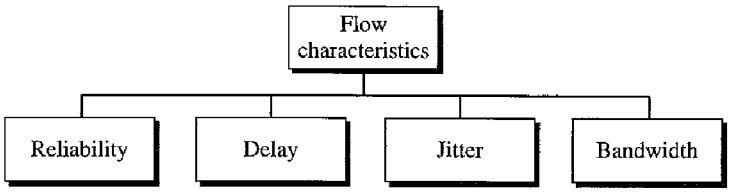
## 24.5 QUALITY OF SERVICE

**Quality of service (QoS)** is an internetworking issue that has been discussed more than defined. We can informally define quality of service as something a flow seeks to attain.

### Flow Characteristics

Traditionally, four types of characteristics are attributed to a flow: reliability, delay, jitter, and bandwidth, as shown in Figure 24.15.

**Figure 24.15** *Flow characteristics*




#### *Reliability*

**Reliability** is a characteristic that a flow needs. Lack of reliability means losing a packet or acknowledgment, which entails retransmission. However, the sensitivity of application programs to reliability is not the same. For example, it is more important that electronic mail, file transfer, and Internet access have reliable transmissions than telephony or audio conferencing.

#### *Delay*

Source-to-destination **delay** is another flow characteristic. Again applications can tolerate delay in different degrees. In this case, telephony, audio conferencing, video conferencing, and remote log-in need minimum delay, while delay in file transfer or e-mail is less important.

#### *Jitter*

**Jitter** is the variation in delay for packets belonging to the same flow. For example, if four packets depart at times 0, 1, 2, 3 and arrive at 20, 21, 22, 23, all have the same delay, 20 units of time. On the other hand, if the above four packets arrive at 21, 23, 21, and 28, they will have different delays: 21, 22, 19, and 24.

For applications such as audio and video, the first case is completely acceptable; the second case is not. For these applications, it does not matter if the packets arrive with a short or long delay as long as the delay is the same for all packets. For this application, the second case is not acceptable.

Jitter is defined as the variation in the packet delay. High jitter means the difference between delays is large; low jitter means the variation is small.

In Chapter 29, we show how multimedia communication deals with jitter. If the jitter is high, some action is needed in order to use the received data.

### *Bandwidth*

Different applications need different bandwidths. In video conferencing we need to send millions of bits per second to refresh a color screen while the total number of bits in an e-mail may not reach even a million.

## Flow Classes

Based on the flow characteristics, we can classify flows into groups, with each group having similar levels of characteristics. This categorization is not formal or universal; some protocols such as ATM have defined classes, as we will see later.

# 24.6 TECHNIQUES TO IMPROVE QoS

In Section 24.5 we tried to define QoS in terms of its characteristics. In this section, we discuss some techniques that can be used to improve the quality of service. We briefly discuss four common methods: scheduling, traffic shaping, admission control, and resource reservation.

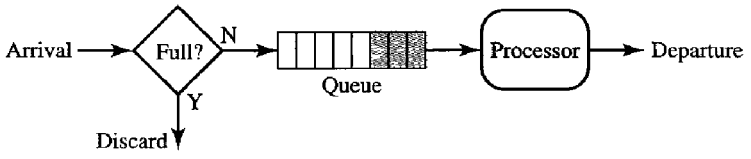
## Scheduling

Packets from different flows arrive at a switch or router for processing. A good scheduling technique treats the different flows in a fair and appropriate manner. Several scheduling techniques are designed to improve the quality of service. We discuss three of them here: FIFO queuing, priority queuing, and weighted fair queuing.

### *FIFO Queuing*

In **first-in, first-out (FIFO) queuing,** packets wait in a buffer (queue) until the node (router or switch) is ready to process them. If the average arrival rate is higher than the average processing rate, the queue will fill up and new packets will be discarded. A FIFO queue is familiar to those who have had to wait for a bus at a bus stop. Figure 24.16 shows a conceptual view of a FIFO queue.

**Figure 24.16** *FIFO queue*

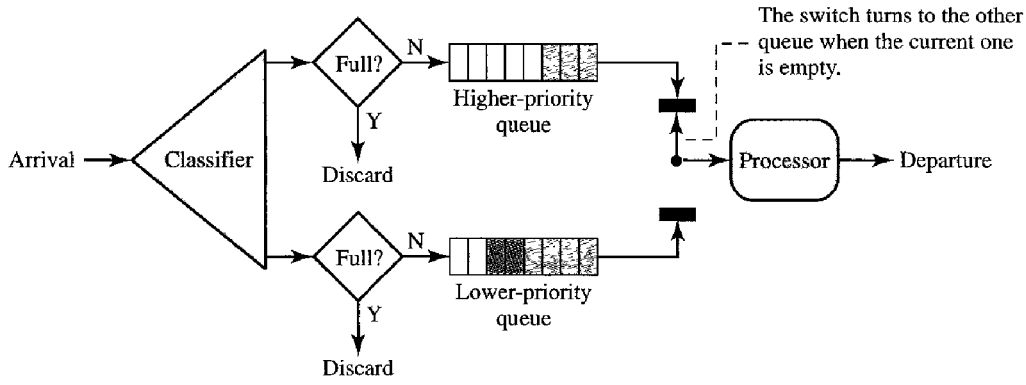



### *Priority Queuing*

In **priority queuing,** packets are first assigned to a priority class. Each priority class has its own queue. The packets in the highest-priority queue are processed first. Packets in the lowest-priority queue are processed last. Note that the system does not stop serving

a queue until it is empty. Figure 24.17 shows priority queuing with two priority levels (for simplicity).

**Figure 24.17** *Priority queuing*




A priority queue can provide better QoS than the FIFO queue because higher-priority traffic, such as multimedia, can reach the destination with less delay. However, there is a potential drawback. If there is a continuous flow in a high-priority queue, the packets in the lower-priority queues will never have a chance to be processed. This is a condition called *starvation*.

### *Weighted Fair Queuing*

A better scheduling method is **weighted fair queuing.** In this technique, the packets are still assigned to different classes and admitted to different queues. The queues, however, are weighted based on the priority of the queues; higher priority means a higher weight. The system processes packets in each queue in a round-robin fashion with the number of packets selected from each queue based on the corresponding weight. For example, if the weights are 3, 2, and 1, three packets are processed from the first queue, two from the second queue, and one from the third queue. If the system does not impose priority on the classes, all weights can be equal. In this way, we have fair queuing with priority. Figure 24.18 shows the technique with three classes.

## Traffic Shaping

**Traffic shaping** is a mechanism to control the amount and the rate of the traffic sent to the network. Two techniques can shape traffic: leaky bucket and token bucket.

### *Leaky Bucket*

If a bucket has a small hole at the bottom, the water leaks from the bucket at a constant rate as long as there is water in the bucket. The rate at which the water leaks does not depend on the rate at which the water is input to the bucket unless the bucket is empty. The input rate can vary, but the output rate remains constant. Similarly, in networking, a technique called **leaky bucket** can smooth out bursty traffic. Bursty chunks are stored in the bucket and sent out at an average rate. Figure 24.19 shows a leaky bucket and its effects.

**Figure 24.18** *Weighted fair queuing*

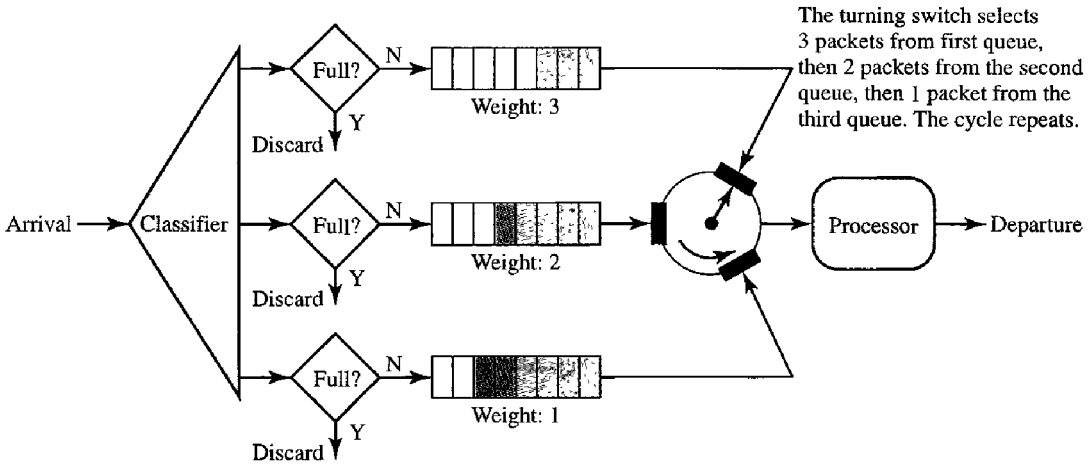
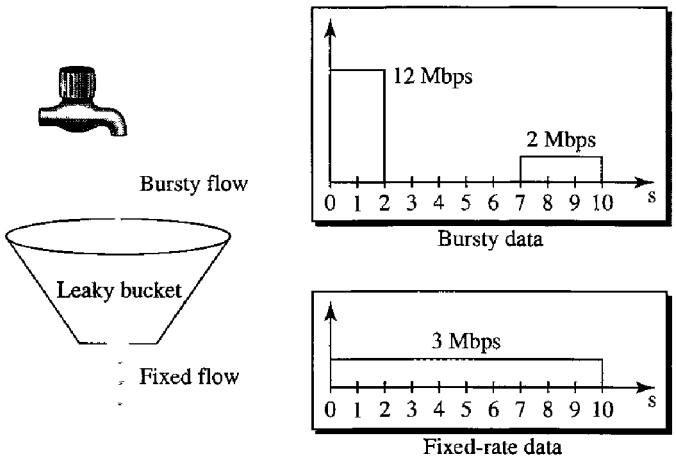


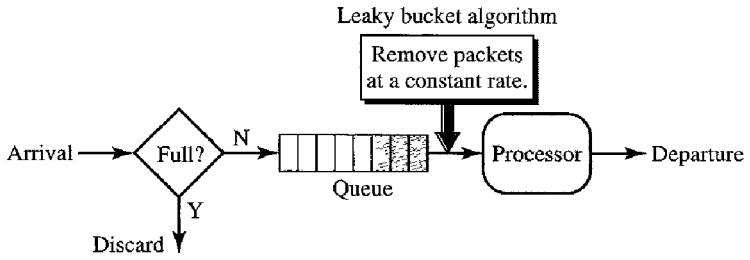

**Figure 24.19** *Leaky bucket*




In the figure, we assume that the network has committed a bandwidth of 3 Mbps for a host. The use of the leaky bucket shapes the input traffic to make it conform to this commitment. In Figure 24.19 the host sends a burst of data at a rate of 12 Mbps for 2 s, for a total of 24 Mbits of data. The host is silent for 5 s and then sends data at a rate of 2 Mbps for 3 s, for a total of 6 Mbits of data. In all, the host has sent 30 Mbits of data in 10 s. The leaky bucket smooths the traffic by sending out data at a rate of 3 Mbps during the same 10 s. Without the leaky bucket, the beginning burst may have hurt the network by consuming more bandwidth than is set aside for this host. We can also see that the leaky bucket may prevent congestion. As an analogy, consider the freeway during rush hour (bursty traffic). If, instead, commuters could stagger their working hours, congestion on our freeways could be avoided.

A simple leaky bucket implementation is shown in Figure 24.20. A FIFO queue holds the packets. If the traffic consists of fixed-size packets (e.g., cells in ATM

**Figure 24.20** *Leaky bucket implementation*




networks), the process removes a fixed number of packets from the queue at each tick of the clock. If the traffic consists of variable-length packets, the fixed output rate must be based on the number of bytes or bits.

The following is an algorithm for variable-length packets:

1. Initialize a counter to $n$ at the tick of the clock.
2. If $n$ is greater than the size of the packet, send the packet and decrement the counter by the packet size. Repeat this step until $n$ is smaller than the packet size.
3. Reset the counter and go to step 1.

**A leaky bucket algorithm shapes bursty traffic into fixed-rate traffic by averaging the data rate. It may drop the packets if the bucket is full.**

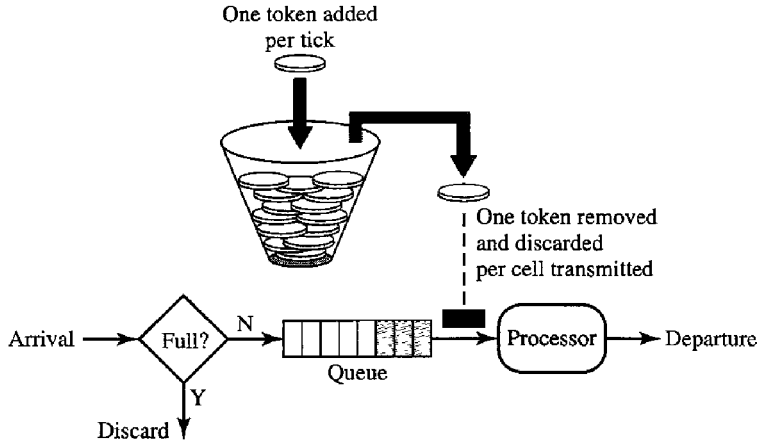
### *Token Bucket*

The leaky bucket is very restrictive. It does not credit an idle host. For example, if a host is not sending for a while, its bucket becomes empty. Now if the host has bursty data, the leaky bucket allows only an average rate. The time when the host was idle is not taken into account. On the other hand, the **token bucket** algorithm allows idle hosts to accumulate credit for the future in the form of tokens. For each tick of the clock, the system sends $n$ tokens to the bucket. The system removes one token for every cell (or byte) of data sent. For example, if $n$ is 100 and the host is idle for 100 ticks, the bucket collects 10,000 tokens. Now the host can consume all these tokens in one tick with 10,000 cells, or the host takes 1000 ticks with 10 cells per tick. In other words, the host can send bursty data as long as the bucket is not empty. Figure 24.21 shows the idea.

The token bucket can easily be implemented with a counter. The token is initialized to zero. Each time a token is added, the counter is incremented by 1. Each time a unit of data is sent, the counter is decremented by 1. When the counter is zero, the host cannot send data.

**The token bucket allows bursty traffic at a regulated maximum rate.**

**Figure 24.21** *Token bucket*




### *Combining Token Bucket and Leaky Bucket*

The two techniques can be combined to credit an idle host and at the same time regulate the traffic. The leaky bucket is applied after the token bucket; the rate of the leaky bucket needs to be higher than the rate of tokens dropped in the bucket.

## Resource Reservation

A flow of data needs resources such as a buffer, bandwidth, CPU time, and so on. The quality of service is improved if these resources are reserved beforehand. We discuss in this section one QoS model called Integrated Services, which depends heavily on resource reservation to improve the quality of service.

## Admission Control

Admission control refers to the mechanism used by a router, or a switch, to accept or reject a flow based on predefined parameters called flow specifications. Before a router accepts a flow for processing, it checks the flow specifications to see if its capacity (in terms of bandwidth, buffer size, CPU speed, etc.) and its previous commitments to other flows can handle the new flow.

# 24.7 INTEGRATED SERVICES

Based on the topics in Sections 24.5 and 24.6, two models have been designed to provide quality of service in the Internet: Integrated Services and Differentiated Services. Both models emphasize the use of quality of service at the network layer (IP), although the model can also be used in other layers such as the data link. We discuss Integrated Services in this section and Differentiated Service in Section 24.8.

As we learned in Chapter 20, IP was originally designed for *best-effort* delivery. This means that every user receives the same level of services. This type of delivery

does not guarantee the minimum of a service, such as bandwidth, to applications such as real-time audio and video. If such an application accidentally gets extra bandwidth, it may be detrimental to other applications, resulting in congestion.

**Integrated Services,** sometimes called **IntServ,** is a *flow-based* QoS model, which means that a user needs to create a flow, a kind of virtual circuit, from the source to the destination and inform all routers of the resource requirement.

---

**Integrated Services is a *flow-based* QoS model designed for IP.**

---

## Signaling

The reader may remember that IP is a connectionless, datagram, packet-switching protocol. How can we implement a flow-based model over a connectionless protocol? The solution is a signaling protocol to run over IP that provides the signaling mechanism for making a reservation. This protocol is called **Resource Reservation Protocol (RSVP)** and will be discussed shortly.

## Flow Specification

When a source makes a reservation, it needs to define a flow specification. A flow specification has two parts: Rspec (resource specification) and Tspec (traffic specification). Rspec defines the resource that the flow needs to reserve (buffer, bandwidth, etc.). Tspec defines the traffic characterization of the flow.

## Admission

After a router receives the flow specification from an application, it decides to admit or deny the service. The decision is based on the previous commitments of the router and the current availability of the resource.

## Service Classes

Two classes of services have been defined for Integrated Services: guaranteed service and controlled-load service.

### *Guaranteed Service Class*

This type of service is designed for real-time traffic that needs a guaranteed minimum end-to-end delay. The end-to-end delay is the sum of the delays in the routers, the propagation delay in the media, and the setup mechanism. Only the first, the sum of the delays in the routers, can be guaranteed by the router. This type of service guarantees that the packets will arrive within a certain delivery time and are not discarded if flow traffic stays within the boundary of Tspec. We can say that guaranteed services are quantitative services, in which the amount of end-to-end delay and the data rate must be defined by the application.

### *Controlled-Load Service Class*

This type of service is designed for applications that can accept some delays, but are sensitive to an overloaded network and to the danger of losing packets. Good examples of these types of applications are file transfer, e-mail, and Internet access. The controlled-load service is a qualitative type of service in that the application requests the possibility of low-loss or no-loss packets.

## RSVP

In the Integrated Services model, an application program needs resource reservation. As we learned in the discussion of the IntServ model, the resource reservation is for a *flow.* This means that if we want to use IntServ at the IP level, we need to create a flow, a kind of virtual-circuit network, out of the IP, which was originally designed as a datagram packet-switched network. A virtual-circuit network needs a signaling system to set up the virtual circuit before data traffic can start. The Resource Reservation Protocol (RSVP) is a signaling protocol to help IP create a flow and consequently make a resource reservation. Before discussing RSVP, we need to mention that it is an independent protocol separate from the Integrated Services model. It may be used in other models in the future.

### *Multicast Trees*

RSVP is different from some other signaling systems we have seen before in that it is a signaling system designed for multicasting. However, RSVP can be also used for unicasting because unicasting is just a special case of multicasting with only one member in the multicast group. The reason for this design is to enable RSVP to provide resource reservations for all kinds of traffic including multimedia which often uses multicasting.

### *Receiver-Based Reservation*

In RSVP, the receivers, not the sender, make the reservation. This strategy matches the other multicasting protocols. For example, in multicast routing protocols, the receivers, not the sender, make a decision to join or leave a multicast group.

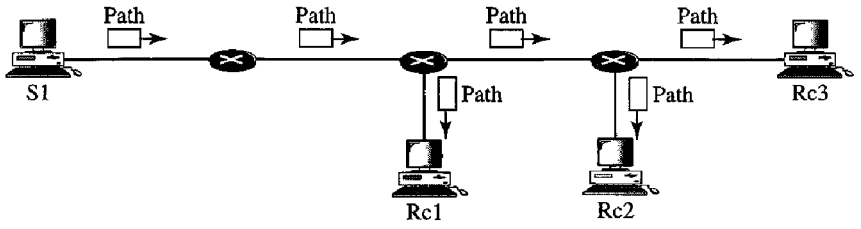
### *RSVP Messages*

RSVP has several types of messages. However, for our purposes, we discuss only two of them: **Path** and **Resv.**

**Path Messages** Recall that the receivers in a flow make the reservation in RSVP. However, the receivers do not know the path traveled by packets before the reservation is made. The path is needed for the reservation. To solve the problem, RSVP uses *Path* messages. A Path message travels from the sender and reaches all receivers in the multicast path. On the way, a Path message stores the necessary information for the receivers. A Path message is sent in a multicast environment; a new message is created when the path diverges. Figure 24.22 shows path messages.

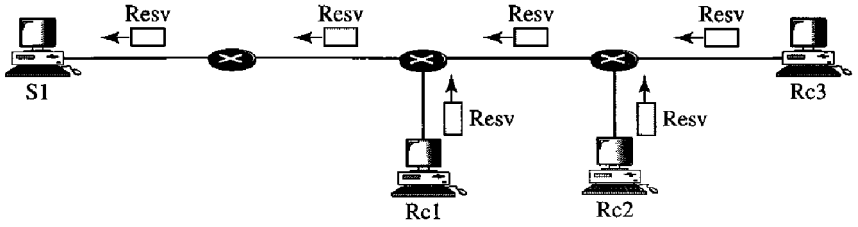
**Resv Messages** After a receiver has received a Path message, it sends a *Resv* message. The Resv message travels toward the sender (upstream) and makes a resource reservation on the routers that support RSVP. If a router does not support RSVP on the path, it routes

**Figure 24.22** *Path messages*




the packet based on the best-effort delivery methods we discussed before. Figure 24.23 shows the Resv messages.

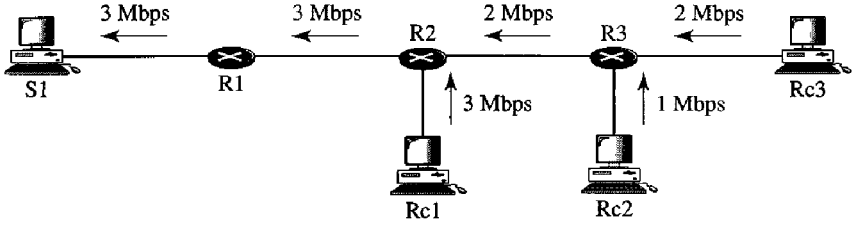
**Figure 24.23** *Resv messages*




### *Reservation Merging*

In RSVP, the resources are not reserved for each receiver in a flow; the reservation is merged. In Figure 24.24, Rc3 requests a 2-Mbps bandwidth while Rc2 requests a 1-Mbps bandwidth. Router R3, which needs to make a bandwidth reservation, merges the two requests. The reservation is made for 2 Mbps, the larger of the two, because a 2-Mbps input reservation can handle both requests. The same situation is true for R2. The reader may ask why Rc2 and Rc3, both belonging to one single flow, request different amounts of bandwidth. The answer is that, in a multimedia environment, different receivers may handle different grades of quality. For example, Rc2 may be able to receive video only at 1 Mbps (lower quality), while Rc3 may be able to receive video at 2 Mbps (higher quality).

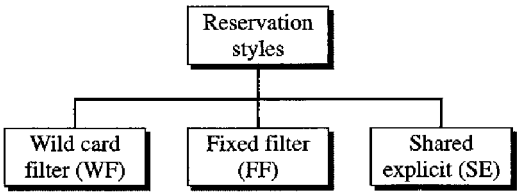
**Figure 24.24** *Reservation merging*



### *Reservation Styles*

When there is more than one flow, the router needs to make a reservation to accommodate all of them. RSVP defines three types of reservation styles, as shown in Figure 24.25.

**Figure 24.25** *Reservation styles*




**Wild Card Filter Style** In this style, the router creates a single reservation for all senders. The reservation is based on the largest request. This type of style is used when the flows from different senders do not occur at the same time.

**Fixed Filter Style** In this style, the router creates a distinct reservation for each flow. This means that if there are $n$ flows, $n$ different reservations are made. This type of style is used when there is a high probability that flows from different senders will occur at the same time.

**Shared Explicit Style** In this style, the router creates a single reservation which can be shared by a set of flows.

### *Soft State*

The reservation information (state) stored in every node for a flow needs to be refreshed periodically. This is referred to as a *soft state* as compared to the *hard state* used in other virtual-circuit protocols such as ATM or Frame Relay, where the information about the flow is maintained until it is erased. The default interval for refreshing is currently 30 s.

## Problems with Integrated Services

There are at least two problems with Integrated Services that may prevent its full implementation in the Internet: scalability and service-type limitation.

### *Scalability*

The Integrated Services model requires that each router keep information for each flow. As the Internet is growing every day, this is a serious problem.

### *Service-Type Limitation*

The Integrated Services model provides only two types of services, guaranteed and control-load. Those opposing this model argue that applications may need more than these two types of services.

## 24.8 DIFFERENTIATED SERVICES

**Differentiated Services (DS** or **Diffserv)** was introduced by the IETF (Internet Engineering Task Force) to handle the shortcomings of Integrated Services. Two fundamental changes were made:

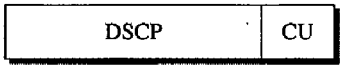
1. The main processing was moved from the core of the network to the edge of the network. This solves the scalability problem. The routers do not have to store information about flows. The applications, or hosts, define the type of service they need each time they send a packet.
2. The per-flow service is changed to per-class service. The router routes the packet based on the class of service defined in the packet, not the flow. This solves the service-type limitation problem. We can define different types of classes based on the needs of applications.

**Differentiated Services is a class-based QoS model designed for IP.**

### DS Field

In Diffserv, each packet contains a field called the DS field. The value of this field is set at the boundary of the network by the host or the first router designated as the boundary router. IETF proposes to replace the existing TOS (type of service) field in IPv4 or the class field in IPv6 by the DS field, as shown in Figure 24.26.

**Figure 24.26** *DS field*




The DS field contains two subfields: DSCP and CU. The DSCP (Differentiated Services Code Point) is a 6-bit subfield that defines the **per-hop behavior (PHB).** The 2-bit CU (currently unused) subfield is not currently used.

The Diffserv capable node (router) uses the DSCP 6 bits as an index to a table defining the packet-handling mechanism for the current packet being processed.

#### *Per-Hop Behavior*

The Diffserv model defines per-hop behaviors (PHBs) for each node that receives a packet. So far three PHBs are defined: DE PHB, EF PHB, and AF PHB.

**DE PHB** The DE PHB (default PHB) is the same as best-effort delivery, which is compatible with TOS.

**EF PHB** The EF PHB (expedited forwarding PHB) provides the following services:

- Low loss

- Low latency
- Ensured bandwidth

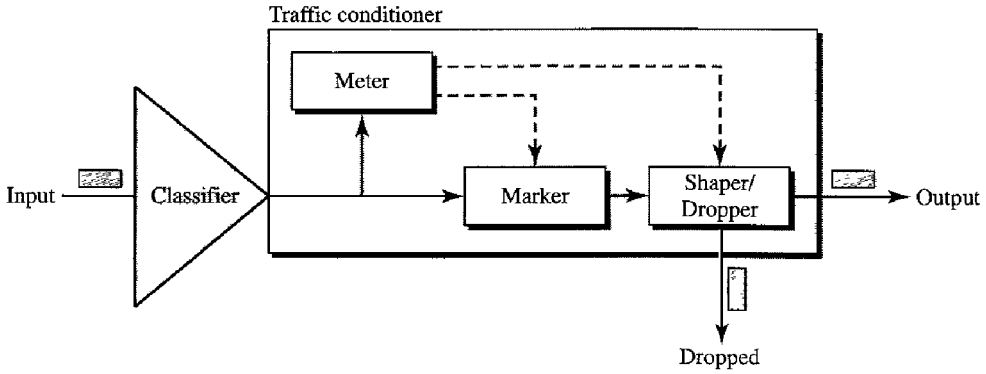
This is the same as having a virtual connection between the source and destination.

**AF PHB** The AF PHB (assured forwarding PHB) delivers the packet with a high assurance as long as the class traffic does not exceed the traffic profile of the node. The users of the network need to be aware that some packets may be discarded.

### *Traffic Conditioner*

To implement Diffserv, the DS node uses traffic conditioners such as meters, markers, shapers, and droppers, as shown in Figure 24.27.

**Figure 24.27** *Traffic conditioner*




**Meters** The meter checks to see if the incoming flow matches the negotiated traffic profile. The meter also sends this result to other components. The meter can use several tools such as a token bucket to check the profile.

**Marker** A marker can remark a packet that is using best-effort delivery (DSCP: 000000) or down-mark a packet based on information received from the meter. Down-marking (lowering the class of the flow) occurs if the flow does not match the profile. A marker does not up-mark (promote the class) a packet.

**Shaper** A shaper uses the information received from the meter to reshape the traffic if it is not compliant with the negotiated profile.

**Dropper** A dropper, which works as a shaper with no buffer, discards packets if the flow severely violates the negotiated profile.

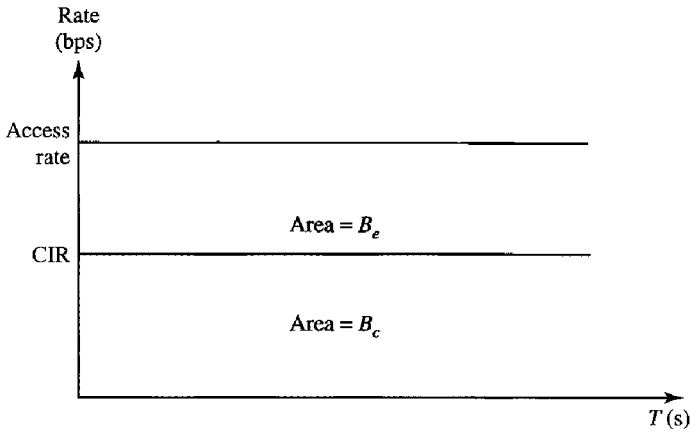
## 24.9 QoS IN SWITCHED NETWORKS

We discussed the proposed models for QoS in the IP protocols. Let us now discuss QoS as used in two switched networks: Frame Relay and ATM. These two networks are virtual-circuit networks that need a signaling protocol such as RSVP.

## QoS in Frame Relay

Four different attributes to control traffic have been devised in Frame Relay: access rate, committed burst size $B_c$, committed information rate (CIR), and excess burst size $B_e$. These are set during the negotiation between the user and the network. For PVC connections, they are negotiated once; for SVC connections, they are negotiated for each connection during connection setup. Figure 24.28 shows the relationships between these four measurements.

**Figure 24.28** *Relationship between traffic control attributes*




### *Access Rate*

For every connection, an **access rate** (in bits per second) is defined. The access rate actually depends on the bandwidth of the channel connecting the user to the network. The user can never exceed this rate. For example, if the user is connected to a Frame Relay network by a T-1 line, the access rate is 1.544 Mbps and can never be exceeded.

### *Committed Burst Size*

For every connection, Frame Relay defines a **committed burst size $B_c$.** This is the maximum number of bits in a predefined time that the network is committed to transfer without discarding any frame or setting the DE bit. For example, if a $B_c$ of 400 kbits for a period of 4 s is granted, the user can send up to 400 kbits during a 4-s interval without worrying about any frame loss. Note that this is not a rate defined for each second. It is a cumulative measurement. The user can send 300 kbits during the first second, no data during the second and the third seconds, and finally 100 kbits during the fourth second.

### *Committed Information Rate*

The **committed information rate (CIR)** is similar in concept to committed burst size except that it defines an average rate in bits per second. If the user follows this rate continuously, the network is committed to deliver the frames. However, because it is an average measurement, a user may send data at a higher rate than the CIR at times or at

a lower rate other times. As long as the average for the predefined period is met, the frames will be delivered.

The cumulative number of bits sent during the predefined period cannot exceed $B_c$. Note that the CIR is not an independent measurement; it can be calculated by using the following formula:

$$\text{CIR} = \frac{B_c}{T} \text{ bps}$$

For example, if the $B_c$ is 5 kbits in a period of 5 s, the CIR is 5000/5, or 1 kbps.

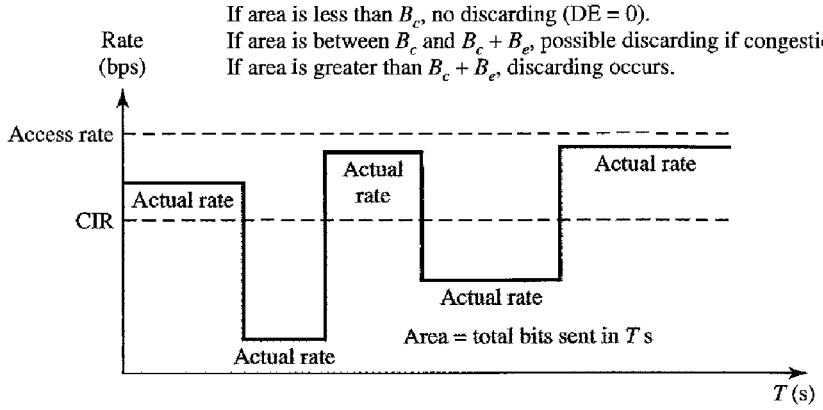
### *Excess Burst Size*

For every connection, Frame Relay defines an **excess burst size $B_e$.** This is the maximum number of bits in excess of $B_c$ that a user can send during a predefined time. The network is committed to transfer these bits if there is no congestion. Note that there is less commitment here than in the case of $B_c$. The network is committing itself conditionally.

### *User Rate*

Figure 24.29 shows how a user can send bursty data. If the user never exceeds $B_c$, the network is committed to transmit the frames without discarding any. If the user exceeds $B_c$ by less than $B_e$ (that is, the total number of bits is less than $B_c + B_e$), the network is committed to transfer all the frames if there is no congestion. If there is congestion, some frames will be discarded. The first switch that receives the frames from the user has a counter and sets the DE bit for the frames that exceed $B_c$. The rest of the switches will discard these frames if there is congestion. Note that a user who needs to send data faster may exceed the $B_c$ level. As long as the level is not above $B_c + B_e$, there is a chance that the frames will reach the destination without being discarded. Remember, however, that the moment the user exceeds the $B_c + B_e$ level, all the frames sent after that are discarded by the first switch.

**Figure 24.29** *User rate in relation to $B_c$ and $B_c + B_e$*



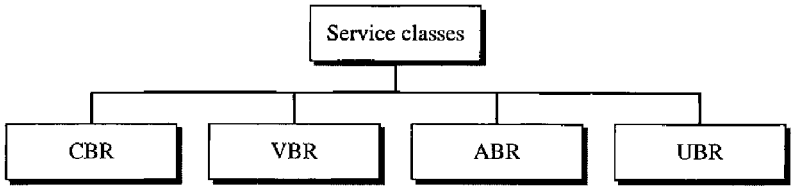
## QoS in ATM

The QoS in ATM is based on the class, user-related attributes, and network-related attributes.

### *Classes*

The ATM Forum defines four service classes: CBR, VBR, ABR, and UBR (see Figure 24.30).

**Figure 24.30** *Service classes*




**CBR** The **constant-bit-rate (CBR)** class is designed for customers who need real-time audio or video services. The service is similar to that provided by a dedicated line such as a T line.

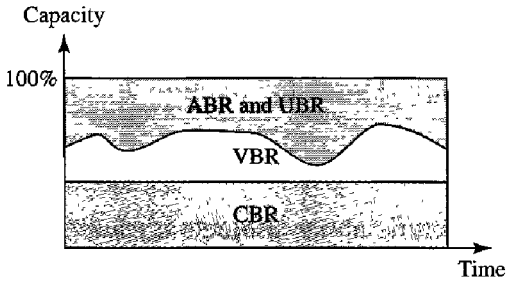
**VBR** The **variable-bit-rate (VBR)** class is divided into two subclasses: real-time (VBR-RT) and non-real-time (VBR-NRT). VBR-RT is designed for those users who need real-time services (such as voice and video transmission) and use compression techniques to create a variable bit rate. VBR-NRT is designed for those users who do not need real-time services but use compression techniques to create a variable bit rate.

**ABR** The **available-bit-rate (ABR)** class delivers cells at a minimum rate. If more network capacity is available, this minimum rate can be exceeded. ABR is particularly suitable for applications that are bursty.

**UBR** The **unspecified-bit-rate (UBR)** class is a best-effort delivery service that does not guarantee anything.

Figure 24.31 shows the relationship of different classes to the total capacity of the network.

**Figure 24.31** *Relationship of service classes to the total capacity of the network*



### *User-Related Attributes*

ATM defines two sets of attributes. User-related attributes are those attributes that define how fast the user wants to send data. These are negotiated at the time of contract between a user and a network. The following are some user-related attributes:

**SCR** The *sustained cell rate* (SCR) is the average cell rate over a long time interval. The actual cell rate may be lower or higher than this value, but the average should be equal to or less than the SCR.

**PCR** The *peak cell rate* (PCR) defines the sender's maximum cell rate. The user's cell rate can sometimes reach this peak, as long as the SCR is maintained.

**MCR** The *minimum cell rate* (MCR) defines the minimum cell rate acceptable to the sender. For example, if the MCR is 50,000, the network must guarantee that the sender can send at least 50,000 cells per second.

**CVDT** The *cell variation delay tolerance* (CVDT) is a measure of the variation in cell transmission times. For example, if the CVDT is 5 ns, this means that the difference between the minimum and the maximum delays in delivering the cells should not exceed 5 ns.

### *Network-Related Attributes*

The network-related attributes are those that define characteristics of the network. The following are some network-related attributes:

**CLR** The *cell loss ratio* (CLR) defines the fraction of cells lost (or delivered so late that they are considered lost) during transmission. For example, if the sender sends 100 cells and one of them is lost, the CLR is

$$\text{CLR} = \frac{1}{100} = 10^{-2}$$

**CTD** The *cell transfer delay* (CTD) is the average time needed for a cell to travel from source to destination. The maximum CTD and the minimum CTD are also considered attributes.

**CDV** The *cell delay variation* (CDV) is the difference between the CTD maximum and the CTD minimum.

**CER** The *cell error ratio* (CER) defines the fraction of the cells delivered in error.

## 24.10 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

Congestion control and QoS are discussed in Sections 5.3 and 5.5 of [Tan03] and in Chapter 3 of [Sta98]. Easy reading about QoS can be found in [FH98]. The full discussion of QoS is covered in [Bla00].

## 24.11 KEY TERMS

access rate
additive increase
available bit rate (ABR)
average data rate
backward explicit congestion notification (BECN)
bursty data
choke packet
closed-loop congestion control
committed burst size $B_c$
committed information rate (CIR)
congestion
congestion avoidance
congestion control
constant bit rate (CBR)
delay
Differentiated Services (DS or Diffserv)
effective bandwidth
excess burst size $B_e$
first-in, first-out (FIFO) queuing
forward explicit congestion notification (FECN)
Integrated Services (IntServ)
jitter
leaky bucket
load
maximum burst size
multiplicative decrease
open-loop congestion control
Path message
peak data rate
per-hop behavior (PHB)
priority queuing
quality of service (QoS)
reliability
Resource Reservation Protocol (RSVP)
Resv message
slow start
throughput
token bucket
traffic shaping
unspecified bit rate (UBR)
variable bit rate (VBR)
weighted fair queuing

## 24.12 SUMMARY

- The average data rate, peak data rate, maximum burst size, and effective band width are qualitative values that describe a data flow.
- A data flow can have a constant bit rate, a variable bit rate, or traffic that is bursty.
- Congestion control refers to the mechanisms and techniques to control congestion and keep the load below capacity.
- Delay and throughput measure the performance of a network.
- Open-loop congestion control prevents congestion; closed-loop congestion control removes congestion.

- TCP avoids congestion through the use of two strategies: the combination of slow start and additive increase, and multiplicative decrease.
- Frame Relay avoids congestion through the use of two strategies: backward explicit congestion notification (BECN) and forward explicit congestion notification (FECN).
- A flow can be characterized by its reliability, delay, jitter, and bandwidth.
- Scheduling, traffic shaping, resource reservation, and admission control are techniques to improve quality of service (QoS).
- FIFO queuing, priority queuing, and weighted fair queuing are scheduling techniques.
- Leaky bucket and token bucket are traffic shaping techniques.
- Integrated Services is a flow-based QoS model designed for IP.
- The Resource Reservation Protocol (RSVP) is a signaling protocol that helps IP create a flow and makes a resource reservation.
- Differential Services is a class-based QoS model designed for IP.
- Access rate, committed burst size, committed information rate, and excess burst size are attributes to control traffic in Frame Relay.
- Quality of service in ATM is based on service classes, user-related attributes, and network-related attributes.

## 24.13 PRACTICE SET

### Review Questions

1. How are congestion control and quality of service related?
2. What is a traffic descriptor?
3. What is the relationship between the average data rate and the peak data rate?
4. What is the definition of bursty data?
5. What is the difference between open-loop congestion control and closed-loop congestion control?
6. Name the policies that can prevent congestion.
7. Name the mechanisms that can alleviate congestion.
8. What determines the sender window size in TCP?
9. How does Frame Relay control congestion?
10. What attributes can be used to describe a flow of data?
11. What are four general techniques to improve quality of service?
12. What is traffic shaping? Name two methods to shape traffic.
13. What is the major difference between Integrated Services and Differentiated Services?
14. How is Resource Reservation Protocol related to Integrated Services?
15. What attributes are used for traffic control in Frame Relay?
16. In regard to quality of service, how do user-related attributes differ from network-related attributes in ATM?

## Exercises

17. The address field of a Frame Relay frame is 1011000000010111. Is there any congestion in the forward direction? Is there any congestion in the backward direction?
18. A frame goes from A to B. There is congestion in both directions. Is the FECN bit set? Is the BECN bit set?
19. In a leaky bucket used to control liquid flow, how many gallons of liquid are left in the bucket if the output rate is 5 gal/min, there is an input burst of 100 gal/min for 12 s, and there is no input for 48 s?
20. An output interface in a switch is designed using the leaky bucket algorithm to send 8000 bytes/s (tick). If the following frames are received in sequence, show the frames that are sent during each second.

    Frames 1, 2, 3, 4: 4000 bytes each

    Frames 5, 6, 7: 3200 bytes each

    Frames 8, 9: 400 bytes each

    Frames 10, 11, 12: 2000 bytes each
21. A user is connected to a Frame Relay network through a T-1 line. The granted CIR is 1 Mbps with a $B_c$ of 5 million bits/5 s and $B_e$ of 1 million bits/5 s.
    a. What is the access rate?
    b. Can the user send data at 1.6 Mbps?
    c. Can the user send data at 1 Mbps all the time? Is it guaranteed that frames are never discarded in this case?
    d. Can the user send data at 1.2 Mbps all the time? Is it guaranteed that frames are never discarded in this case? If the answer is no, is it guaranteed that frames are discarded only if there is congestion?
    e. Repeat the question in part (d) for a constant rate of 1.4 Mbps.
    f. What is the maximum data rate the user can use all the time without worrying about the frames being discarded?
    g. If the user wants to take a risk, what is the maximum data rate that can be used with no chance of discarding if there is no congestion?
22. In Exercise 21 the user sends data at 1.4 Mbps for 2 s and nothing for the next 3 s. Is there a danger of discarded data if there is no congestion? Is there a danger of discarded data if there is congestion?
23. In ATM, if each cell takes 10 μs to reach the destination, what is the CTD?
24. An ATM network has lost 5 cells out of 10,000 and 2 are in error. What is the CLR? What is the CER?