

discuss congestion control in a general setting and then discuss how TCP performs congestion control. But before getting into all this good stuff, let's first look at transport-layer multiplexing and demultiplexing.

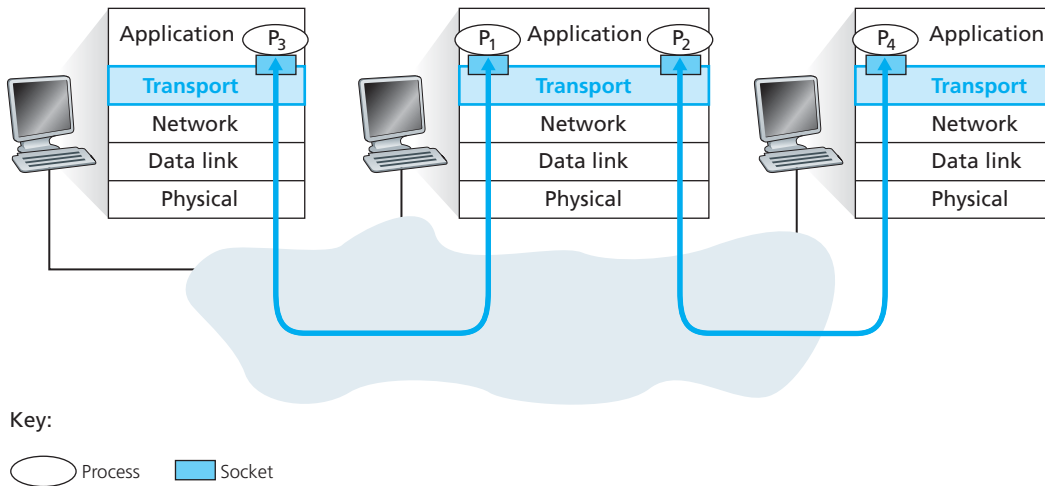
## 3.2 Multiplexing and Demultiplexing

In this section, we discuss transport-layer multiplexing and demultiplexing, that is, extending the host-to-host delivery service provided by the network layer to a process-to-process delivery service for applications running on the hosts. In order to keep the discussion concrete, we'll discuss this basic transport-layer service in the context of the Internet. We emphasize, however, that a multiplexing/demultiplexing service is needed for all computer networks.

At the destination host, the transport layer receives segments from the network layer just below. The transport layer has the responsibility of delivering the data in these segments to the appropriate application process running in the host. Let's take a look at an example. Suppose you are sitting in front of your computer, and you are downloading Web pages while running one FTP session and two Telnet sessions. You therefore have four network application processes running—two Telnet processes, one FTP process, and one HTTP process. When the transport layer in your computer receives data from the network layer below, it needs to direct the received data to one of these four processes. Let's now examine how this is done.

First recall from Section 2.7 that a process (as part of a network application) can have one or more **sockets**, doors through which data passes from the network to the process and through which data passes from the process to the network. Thus, as shown in Figure 3.2, the transport layer in the receiving host does not actually deliver data directly to a process, but instead to an intermediary socket. Because at any given time there can be more than one socket in the receiving host, each socket has a unique identifier. The format of the identifier depends on whether the socket is a UDP or a TCP socket, as we'll discuss shortly.

Now let's consider how a receiving host directs an incoming transport-layer segment to the appropriate socket. Each transport-layer segment has a set of fields in the segment for this purpose. At the receiving end, the transport layer examines these fields to identify the receiving socket and then directs the segment to that socket. This job of delivering the data in a transport-layer segment to the correct socket is called **demultiplexing**. The job of gathering data chunks at the source host from different sockets, encapsulating each data chunk with header information (that will later be used in demultiplexing) to create segments, and passing the segments to the network layer is called **multiplexing**. Note that the transport layer in the middle host in Figure 3.2 must demultiplex segments arriving from the network layer below to either process  $P_1$  or  $P_2$  above; this is done by directing the arriving segment's data to the corresponding process's socket. The transport layer in the middle host must also

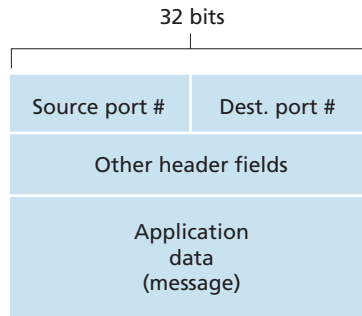


**Figure 3.2** ♦ Transport-layer multiplexing and demultiplexing

gather outgoing data from these sockets, form transport-layer segments, and pass these segments down to the network layer. Although we have introduced multiplexing and demultiplexing in the context of the Internet transport protocols, it's important to realize that they are concerns whenever a single protocol at one layer (at the transport layer or elsewhere) is used by multiple protocols at the next higher layer.

To illustrate the demultiplexing job, recall the household analogy in the previous section. Each of the kids is identified by his or her name. When Bill receives a batch of mail from the mail carrier, he performs a demultiplexing operation by observing to whom the letters are addressed and then hand delivering the mail to his brothers and sisters. Ann performs a multiplexing operation when she collects letters from her brothers and sisters and gives the collected mail to the mail person.

Now that we understand the roles of transport-layer multiplexing and demultiplexing, let us examine how it is actually done in a host. From the discussion above, we know that transport-layer multiplexing requires (1) that sockets have unique identifiers, and (2) that each segment have special fields that indicate the socket to which the segment is to be delivered. These special fields, illustrated in Figure 3.3, are the **source port number field** and the **destination port number field**. (The UDP and TCP segments have other fields as well, as discussed in the subsequent sections of this chapter.) Each port number is a 16-bit number, ranging from 0 to 65535. The port numbers ranging from 0 to 1023 are called **well-known port numbers** and are restricted, which means that they are reserved for use by well-known application protocols such as HTTP (which uses port number 80) and FTP (which uses port number 21). The list of well-known port numbers is given in RFC 1700 and is updated at <http://www.iana.org> [RFC 3232]. When we develop a new



**Figure 3.3** ♦ Source and destination port-number fields in a transport-layer segment

application (such as the simple application developed in Section 2.7), we must assign the application a port number.

It should now be clear how the transport layer *could* implement the demultiplexing service: Each socket in the host could be assigned a port number, and when a segment arrives at the host, the transport layer examines the destination port number in the segment and directs the segment to the corresponding socket. The segment’s data then passes through the socket into the attached process. As we’ll see, this is basically how UDP does it. However, we’ll also see that multiplexing/demultiplexing in TCP is yet more subtle.

### Connectionless Multiplexing and Demultiplexing

Recall from Section 2.7.1 that the Python program running in a host can create a UDP socket with the line

```
clientSocket = socket(socket.AF_INET, socket.SOCK_DGRAM)
```

When a UDP socket is created in this manner, the transport layer automatically assigns a port number to the socket. In particular, the transport layer assigns a port number in the range 1024 to 65535 that is currently not being used by any other UDP port in the host. Alternatively, we can add a line into our Python program after we create the socket to associate a specific port number (say, 19157) to this UDP socket via the socket `bind()` method:

```
clientSocket.bind(('', 19157))
```

If the application developer writing the code were implementing the server side of a “well-known protocol,” then the developer would have to assign the corresponding

well-known port number. Typically, the client side of the application lets the transport layer automatically (and transparently) assign the port number, whereas the server side of the application assigns a specific port number.

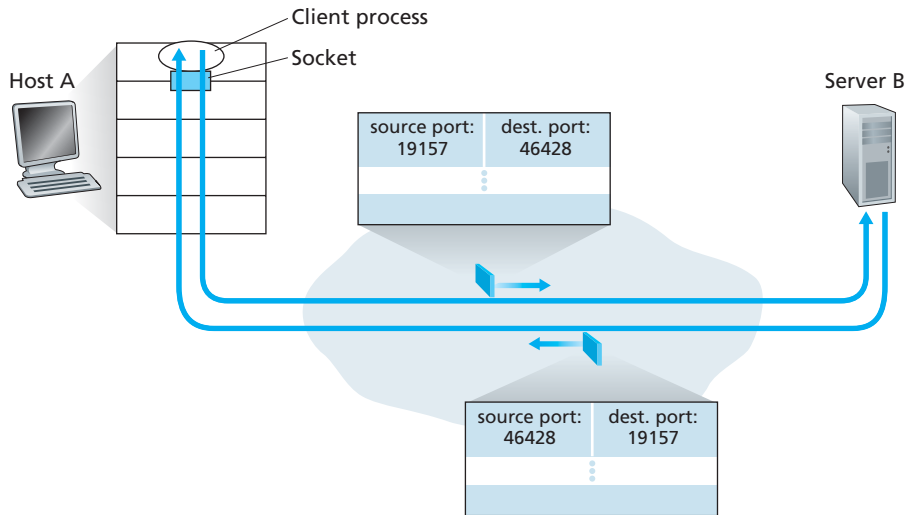
With port numbers assigned to UDP sockets, we can now precisely describe UDP multiplexing/demultiplexing. Suppose a process in Host A, with UDP port 19157, wants to send a chunk of application data to a process with UDP port 46428 in Host B. The transport layer in Host A creates a transport-layer segment that includes the application data, the source port number (19157), the destination port number (46428), and two other values (which will be discussed later, but are unimportant for the current discussion). The transport layer then passes the resulting segment to the network layer. The network layer encapsulates the segment in an IP datagram and makes a best-effort attempt to deliver the segment to the receiving host. If the segment arrives at the receiving Host B, the transport layer at the receiving host examines the destination port number in the segment (46428) and delivers the segment to its socket identified by port 46428. Note that Host B could be running multiple processes, each with its own UDP socket and associated port number. As UDP segments arrive from the network, Host B directs (demultiplexes) each segment to the appropriate socket by examining the segment's destination port number.

It is important to note that a UDP socket is fully identified by a two-tuple consisting of a destination IP address and a destination port number. As a consequence, if two UDP segments have different source IP addresses and/or source port numbers, but have the same *destination* IP address and *destination* port number, then the two segments will be directed to the same destination process via the same destination socket.

You may be wondering now, what is the purpose of the source port number? As shown in Figure 3.4, in the A-to-B segment the source port number serves as part of a “return address”—when B wants to send a segment back to A, the destination port in the B-to-A segment will take its value from the source port value of the A-to-B segment. (The complete return address is A's IP address and the source port number.) As an example, recall the UDP server program studied in Section 2.7. In `UDPServer.py`, the server uses the `recvfrom()` method to extract the client-side (source) port number from the segment it receives from the client; it then sends a new segment to the client, with the extracted source port number serving as the destination port number in this new segment.

### Connection-Oriented Multiplexing and Demultiplexing

In order to understand TCP demultiplexing, we have to take a close look at TCP sockets and TCP connection establishment. One subtle difference between a TCP socket and a UDP socket is that a TCP socket is identified by a four-tuple: (source IP address, source port number, destination IP address, destination port number). Thus, when a TCP segment arrives from the network to a host, the host uses all four values to direct (demultiplex) the segment to the appropriate socket. In particular, and in contrast with UDP, two arriving TCP segments with different source IP



**Figure 3.4** ♦ The inversion of source and destination port numbers

addresses or source port numbers will (with the exception of a TCP segment carrying the original connection-establishment request) be directed to two different sockets. To gain further insight, let's reconsider the TCP client-server programming example in Section 2.7.2:

- The TCP server application has a “welcoming socket,” that waits for connection-establishment requests from TCP clients (see Figure 2.29) on port number 12000.
- The TCP client creates a socket and sends a connection establishment request segment with the lines:

```
clientSocket = socket(AF_INET, SOCK_STREAM)
clientSocket.connect((serverName, 12000))
```

- A connection-establishment request is nothing more than a TCP segment with destination port number 12000 and a special connection-establishment bit set in the TCP header (discussed in Section 3.5). The segment also includes a source port number that was chosen by the client.
- When the host operating system of the computer running the server process receives the incoming connection-request segment with destination port 12000, it locates the server process that is waiting to accept a connection on port number 12000. The server process then creates a new socket:

```
connectionSocket, addr = serverSocket.accept()
```

- Also, the transport layer at the server notes the following four values in the connection-request segment: (1) the source port number in the segment, (2) the IP address of the source host, (3) the destination port number in the segment, and (4) its own IP address. The newly created connection socket is identified by these four values; all subsequently arriving segments whose source port, source IP address, destination port, and destination IP address match these four values will be demultiplexed to this socket. With the TCP connection now in place, the client and server can now send data to each other.

The server host may support many simultaneous TCP connection sockets, with each socket attached to a process, and with each socket identified by its own four-tuple. When a TCP segment arrives at the host, all four fields (source IP address, source port, destination IP address, destination port) are used to direct (demultiplex) the segment to the appropriate socket.

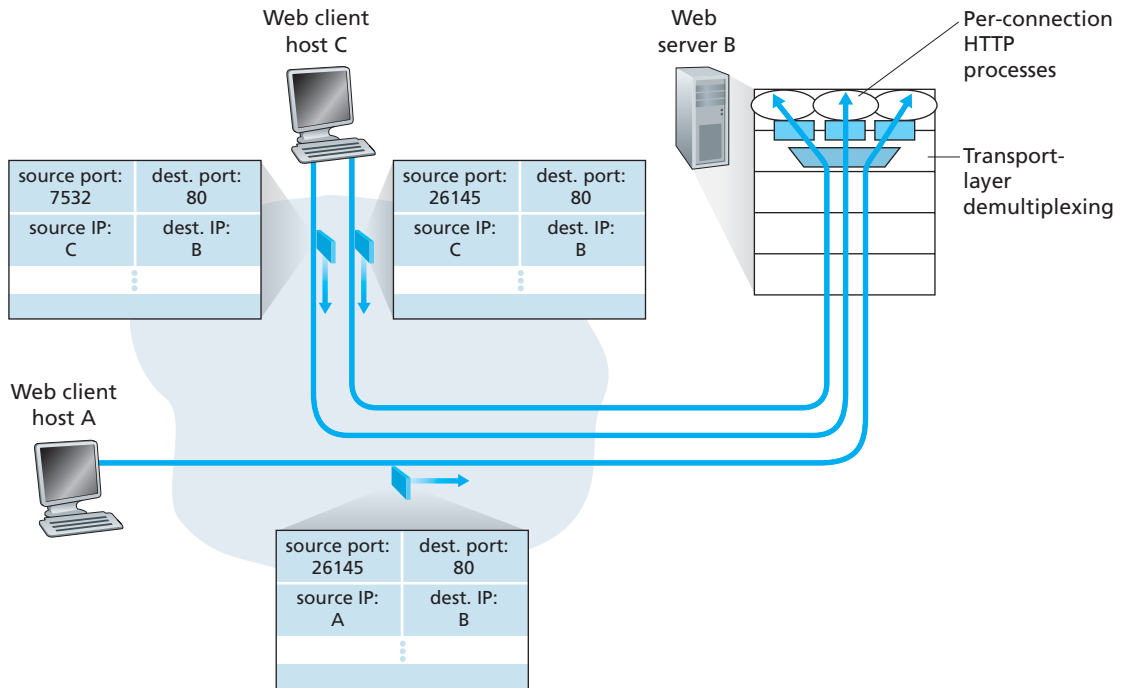


## FOCUS ON SECURITY

### PORT SCANNING

We've seen that a server process waits patiently on an open port for contact by a remote client. Some ports are reserved for well-known applications (e.g., Web, FTP, DNS, and SMTP servers); other ports are used by convention by popular applications (e.g., the Microsoft 2000 SQL server listens for requests on UDP port 1434). Thus, if we determine that a port is open on a host, we may be able to map that port to a specific application running on the host. This is very useful for system administrators, who are often interested in knowing which network applications are running on the hosts in their networks. But attackers, in order to "case the joint," also want to know which ports are open on target hosts. If a host is found to be running an application with a known security flaw (e.g., a SQL server listening on port 1434 was subject to a buffer overflow, allowing a remote user to execute arbitrary code on the vulnerable host, a flaw exploited by the Slammer worm [CERT 2003-04]), then that host is ripe for attack.

Determining which applications are listening on which ports is a relatively easy task. Indeed there are a number of public domain programs, called port scanners, that do just that. Perhaps the most widely used of these is *nmap*, freely available at <http://nmap.org> and included in most Linux distributions. For TCP, *nmap* sequentially scans ports, looking for ports that are accepting TCP connections. For UDP, *nmap* again sequentially scans ports, looking for UDP ports that respond to transmitted UDP segments. In both cases, *nmap* returns a list of open, closed, or unreachable ports. A host running *nmap* can attempt to scan any target host *anywhere* in the Internet. We'll revisit *nmap* in Section 3.5.6, when we discuss TCP connection management.



**Figure 3.5** ♦ Two clients, using the same destination port number (80) to communicate with the same Web server application

The situation is illustrated in Figure 3.5, in which Host C initiates two HTTP sessions to server B, and Host A initiates one HTTP session to B. Hosts A and C and server B each have their own unique IP address—A, C, and B, respectively. Host C assigns two different source port numbers (26145 and 7532) to its two HTTP connections. Because Host A is choosing source port numbers independently of C, it might also assign a source port of 26145 to its HTTP connection. But this is not a problem—server B will still be able to correctly demultiplex the two connections having the same source port number, since the two connections have different source IP addresses.

## Web Servers and TCP

Before closing this discussion, it's instructive to say a few additional words about Web servers and how they use port numbers. Consider a host running a Web server, such as an Apache Web server, on port 80. When clients (for example, browsers) send segments to the server, *all* segments will have destination port 80. In particular, both the initial connection-establishment segments and the segments carrying HTTP request messages will have destination port 80. As we have just described,

the server distinguishes the segments from the different clients using source IP addresses and source port numbers.

Figure 3.5 shows a Web server that spawns a new process for each connection. As shown in Figure 3.5, each of these processes has its own connection socket through which HTTP requests arrive and HTTP responses are sent. We mention, however, that there is not always a one-to-one correspondence between connection sockets and processes. In fact, today's high-performing Web servers often use only one process, and create a new thread with a new connection socket for each new client connection. (A thread can be viewed as a lightweight subprocess.) If you did the first programming assignment in Chapter 2, you built a Web server that does just this. For such a server, at any given time there may be many connection sockets (with different identifiers) attached to the same process.

If the client and server are using persistent HTTP, then throughout the duration of the persistent connection the client and server exchange HTTP messages via the same server socket. However, if the client and server use non-persistent HTTP, then a new TCP connection is created and closed for every request/response, and hence a new socket is created and later closed for every request/response. This frequent creating and closing of sockets can severely impact the performance of a busy Web server (although a number of operating system tricks can be used to mitigate the problem). Readers interested in the operating system issues surrounding persistent and non-persistent HTTP are encouraged to see [Nielsen 1997; Nahum 2002].

Now that we've discussed transport-layer multiplexing and demultiplexing, let's move on and discuss one of the Internet's transport protocols, UDP. In the next section we'll see that UDP adds little more to the network-layer protocol than a multiplexing/demultiplexing service.

### 3.3 Connectionless Transport: UDP

In this section, we'll take a close look at UDP, how it works, and what it does. We encourage you to refer back to Section 2.1, which includes an overview of the UDP service model, and to Section 2.7.1, which discusses socket programming using UDP.

To motivate our discussion about UDP, suppose you were interested in designing a no-frills, bare-bones transport protocol. How might you go about doing this? You might first consider using a vacuous transport protocol. In particular, on the sending side, you might consider taking the messages from the application process and passing them directly to the network layer; and on the receiving side, you might consider taking the messages arriving from the network layer and passing them directly to the application process. But as we learned in the previous section, we have to do a little more than nothing! At the very least, the transport layer has to



provide a multiplexing/demultiplexing service in order to pass data between the network layer and the correct application-level process.

UDP, defined in [RFC 768], does just about as little as a transport protocol can do. Aside from the multiplexing/demultiplexing function and some light error checking, it adds nothing to IP. In fact, if the application developer chooses UDP instead of TCP, then the application is almost directly talking with IP. UDP takes messages from the application process, attaches source and destination port number fields for the multiplexing/demultiplexing service, adds two other small fields, and passes the resulting segment to the network layer. The network layer encapsulates the transport-layer segment into an IP datagram and then makes a best-effort attempt to deliver the segment to the receiving host. If the segment arrives at the receiving host, UDP uses the destination port number to deliver the segment's data to the correct application process. Note that with UDP there is no handshaking between sending and receiving transport-layer entities before sending a segment. For this reason, UDP is said to be *connectionless*.

DNS is an example of an application-layer protocol that typically uses UDP. When the DNS application in a host wants to make a query, it constructs a DNS query message and passes the message to UDP. Without performing any handshaking with the UDP entity running on the destination end system, the host-side UDP adds header fields to the message and passes the resulting segment to the network layer. The network layer encapsulates the UDP segment into a datagram and sends the datagram to a name server. The DNS application at the querying host then waits for a reply to its query. If it doesn't receive a reply (possibly because the underlying network lost the query or the reply), either it tries sending the query to another name server, or it informs the invoking application that it can't get a reply.

Now you might be wondering why an application developer would ever choose to build an application over UDP rather than over TCP. Isn't TCP always preferable, since TCP provides a reliable data transfer service, while UDP does not? The answer is no, as many applications are better suited for UDP for the following reasons:

- *Finer application-level control over what data is sent, and when.* Under UDP, as soon as an application process passes data to UDP, UDP will package the data inside a UDP segment and immediately pass the segment to the network layer. TCP, on the other hand, has a congestion-control mechanism that throttles the transport-layer TCP sender when one or more links between the source and destination hosts become excessively congested. TCP will also continue to resend a segment until the receipt of the segment has been acknowledged by the destination, regardless of how long reliable delivery takes. Since real-time applications often require a minimum sending rate, do not want to overly delay segment transmission, and can tolerate some data loss, TCP's service model is not particularly well matched to these applications' needs. As discussed below, these applications can use UDP and implement, as part of the application, any additional functionality that is needed beyond UDP's no-frills segment-delivery service.

- *No connection establishment.* As we'll discuss later, TCP uses a three-way handshake before it starts to transfer data. UDP just blasts away without any formal preliminaries. Thus UDP does not introduce any delay to establish a connection. This is probably the principal reason why DNS runs over UDP rather than TCP—DNS would be much slower if it ran over TCP. HTTP uses TCP rather than UDP, since reliability is critical for Web pages with text. But, as we briefly discussed in Section 2.2, the TCP connection-establishment delay in HTTP is an important contributor to the delays associated with downloading Web documents.
- *No connection state.* TCP maintains connection state in the end systems. This connection state includes receive and send buffers, congestion-control parameters, and sequence and acknowledgment number parameters. We will see in Section 3.5 that this state information is needed to implement TCP's reliable data transfer service and to provide congestion control. UDP, on the other hand, does not maintain connection state and does not track any of these parameters. For this reason, a server devoted to a particular application can typically support many more active clients when the application runs over UDP rather than TCP.
- *Small packet header overhead.* The TCP segment has 20 bytes of header overhead in every segment, whereas UDP has only 8 bytes of overhead.

Figure 3.6 lists popular Internet applications and the transport protocols that they use. As we expect, e-mail, remote terminal access, the Web, and file transfer run over TCP—all these applications need the reliable data transfer service of TCP. Nevertheless, many important applications run over UDP rather than TCP. UDP is used for RIP routing table updates (see Section 4.6.1). Since RIP updates are sent periodically (typically every five minutes), lost updates will be replaced by more recent updates, thus making the lost, out-of-date update useless. UDP is also used to carry network management (SNMP; see Chapter 9) data. UDP is preferred to TCP in this case, since network management applications must often run when the network is in a stressed state—precisely when reliable, congestion-controlled data transfer is difficult to achieve. Also, as we mentioned earlier, DNS runs over UDP, thereby avoiding TCP's connection-establishment delays.

As shown in Figure 3.6, both UDP and TCP are used today with multimedia applications, such as Internet phone, real-time video conferencing, and streaming of stored audio and video. We'll take a close look at these applications in Chapter 7. We just mention now that all of these applications can tolerate a small amount of packet loss, so that reliable data transfer is not absolutely critical for the application's success. Furthermore, real-time applications, like Internet phone and video conferencing, react very poorly to TCP's congestion control. For these reasons, developers of multimedia applications may choose to run their applications over UDP instead of TCP. However, TCP is increasingly being used for streaming media transport. For example, [Sripanidkulchai 2004] found that nearly 75% of on-demand and live streaming used TCP. When packet loss rates are low, and with some organizations

Application	Application-Layer Protocol	Underlying Transport Protocol
Electronic mail	SMTP	TCP
Remote terminal access	Telnet	TCP
Web	HTTP	TCP
File transfer	FTP	TCP
Remote file server	NFS	Typically UDP
Streaming multimedia	typically proprietary	UDP or TCP
Internet telephony	typically proprietary	UDP or TCP
Network management	SNMP	Typically UDP
Routing protocol	RIP	Typically UDP
Name translation	DNS	Typically UDP

**Figure 3.6** ♦ Popular Internet applications and their underlying transport protocols

blocking UDP traffic for security reasons (see Chapter 8), TCP becomes an increasingly attractive protocol for streaming media transport.

Although commonly done today, running multimedia applications over UDP is controversial. As we mentioned above, UDP has no congestion control. But congestion control is needed to prevent the network from entering a congested state in which very little useful work is done. If everyone were to start streaming high-bit-rate video without using any congestion control, there would be so much packet overflow at routers that very few UDP packets would successfully traverse the source-to-destination path. Moreover, the high loss rates induced by the uncontrolled UDP senders would cause the TCP senders (which, as we'll see, *do* decrease their sending rates in the face of congestion) to dramatically decrease their rates. Thus, the lack of congestion control in UDP can result in high loss rates between a UDP sender and receiver, and the crowding out of TCP sessions—a potentially serious problem [Floyd 1999]. Many researchers have proposed new mechanisms to force all sources, including UDP sources, to perform adaptive congestion control [Mahdavi 1997; Floyd 2000; Kohler 2006: RFC 4340].

Before discussing the UDP segment structure, we mention that it *is* possible for an application to have reliable data transfer when using UDP. This can be done if reliability is built into the application itself (for example, by adding acknowledgment and retransmission mechanisms, such as those we'll study in the next section). But this is a nontrivial task that would keep an application developer busy debugging for