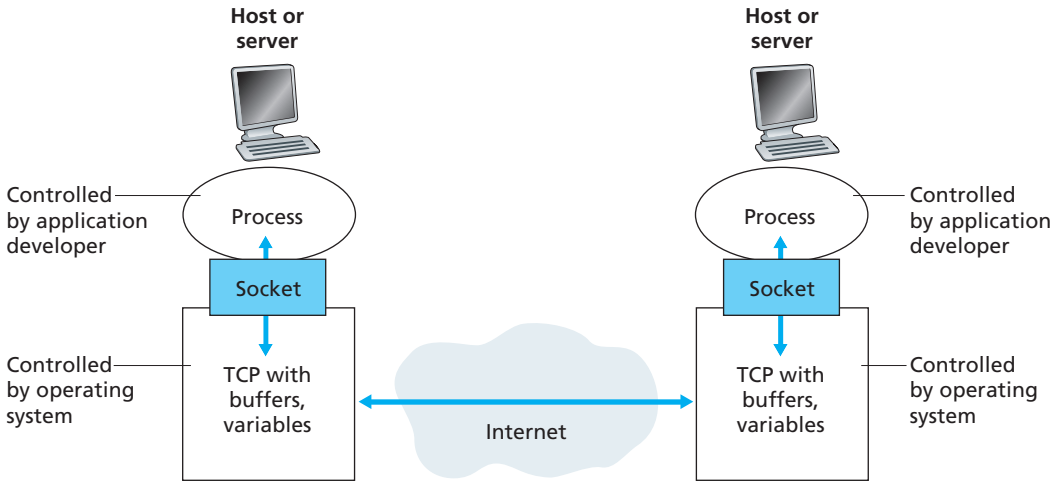

Figure 2.3 ♦ Application processes, sockets, and underlying transport protocol

To identify the receiving process, two pieces of information need to be specified: (1) the address of the host and (2) an identifier that specifies the receiving process in the destination host.

In the Internet, the host is identified by its **IP address**. We'll discuss IP addresses in great detail in Chapter 4. For now, all we need to know is that an IP address is a 32-bit quantity that we can think of as uniquely identifying the host. In addition to knowing the address of the host to which a message is destined, the sending process must also identify the receiving process (more specifically, the receiving socket) running in the host. This information is needed because in general a host could be running many network applications. A destination **port number** serves this purpose. Popular applications have been assigned specific port numbers. For example, a Web server is identified by port number 80. A mail server process (using the SMTP protocol) is identified by port number 25. A list of well-known port numbers for all Internet standard protocols can be found at www.iana.org. We'll examine port numbers in detail in Chapter 3.

2.1.3 Transport Services Available to Applications

Recall that a socket is the interface between the application process and the transport-layer protocol. The application at the sending side pushes messages through the socket. At the other side of the socket, the transport-layer protocol has the responsibility of getting the messages to the socket of the receiving process.

Many networks, including the Internet, provide more than one transport-layer protocol. When you develop an application, you must choose one of the available

transport-layer protocols. How do you make this choice? Most likely, you would study the services provided by the available transport-layer protocols, and then pick the protocol with the services that best match your application's needs. The situation is similar to choosing either train or airplane transport for travel between two cities. You have to choose one or the other, and each transportation mode offers different services. (For example, the train offers downtown pickup and drop-off, whereas the plane offers shorter travel time.)

What are the services that a transport-layer protocol can offer to applications invoking it? We can broadly classify the possible services along four dimensions: reliable data transfer, throughput, timing, and security.

Reliable Data Transfer

As discussed in Chapter 1, packets can get lost within a computer network. For example, a packet can overflow a buffer in a router, or can be discarded by a host or router after having some of its bits corrupted. For many applications—such as electronic mail, file transfer, remote host access, Web document transfers, and financial applications—data loss can have devastating consequences (in the latter case, for either the bank or the customer!). Thus, to support these applications, something has to be done to guarantee that the data sent by one end of the application is delivered correctly and completely to the other end of the application. If a protocol provides such a guaranteed data delivery service, it is said to provide **reliable data transfer**. One important service that a transport-layer protocol can potentially provide to an application is process-to-process reliable data transfer. When a transport protocol provides this service, the sending process can just pass its data into the socket and know with complete confidence that the data will arrive without errors at the receiving process.

When a transport-layer protocol doesn't provide reliable data transfer, some of the data sent by the sending process may never arrive at the receiving process. This may be acceptable for **loss-tolerant applications**, most notably multimedia applications such as conversational audio/video that can tolerate some amount of data loss. In these multimedia applications, lost data might result in a small glitch in the audio/video—not a crucial impairment.

Throughput

In Chapter 1 we introduced the concept of available throughput, which, in the context of a communication session between two processes along a network path, is the rate at which the sending process can deliver bits to the receiving process. Because other sessions will be sharing the bandwidth along the network path, and because these other sessions will be coming and going, the available throughput can fluctuate with time. These observations lead to another natural service that a transport-layer protocol could provide, namely, guaranteed available throughput at

some specified rate. With such a service, the application could request a guaranteed throughput of *r* bits/sec, and the transport protocol would then ensure that the available throughput is always at least *r* bits/sec. Such a guaranteed throughput service would appeal to many applications. For example, if an Internet telephony application encodes voice at 32 kbps, it needs to send data into the network and have data delivered to the receiving application at this rate. If the transport protocol cannot provide this throughput, the application would need to encode at a lower rate (and receive enough throughput to sustain this lower coding rate) or may have to give up, since receiving, say, half of the needed throughput is of little or no use to this Internet telephony application. Applications that have throughput requirements are said to be **bandwidth-sensitive applications**. Many current multimedia applications are bandwidth sensitive, although some multimedia applications may use adaptive coding techniques to encode digitized voice or video at a rate that matches the currently available throughput.

While bandwidth-sensitive applications have specific throughput requirements, **elastic applications** can make use of as much, or as little, throughput as happens to be available. Electronic mail, file transfer, and Web transfers are all elastic applications. Of course, the more throughput, the better. There'san adage that says that one cannot be too rich, too thin, or have too much throughput!

Timing

A transport-layer protocol can also provide timing guarantees. As with throughput guarantees, timing guarantees can come in many shapes and forms. An example guarantee might be that every bit that the sender pumps into the socket arrives at the receiver's socket no more than 100 msec later. Such a service would be appealing to interactive real-time applications, such as Internet telephony, virtual environments, teleconferencing, and multiplayer games, all of which require tight timing constraints on data delivery in order to be effective. (See Chapter 9, [Gauthier 1999; Ramjee 1994].) Long delays in Internet telephony, for example, tend to result in unnatural pauses in the conversation; in a multiplayer game or virtual interactive environment, a long delay between taking an action and seeing the response from the environment (for example, from another player at the end of an end-to-end connection) makes the application feel less realistic. For non-real-time applications, lower delay is always preferable to higher delay, but no tight constraint is placed on the end-to-end delays.

Security

Finally, a transport protocol can provide an application with one or more security services. For example, in the sending host, a transport protocol can encrypt all data transmitted by the sending process, and in the receiving host, the transport-layer protocol can decrypt the data before delivering the data to the receiving process. Such a service would provide confidentiality between the two processes, even if the data is

somehow observed between sending and receiving processes. A transport protocol can also provide other security services in addition to confidentiality, including data integrity and end-point authentication, topics that we'll cover in detail in Chapter 8.

2.1.4 Transport Services Provided by the Internet

Up until this point, we have been considering transport services that a computer network *could* provide in general. Let's now get more specific and examine the type of transport services provided by the Internet. The Internet (and, more generally, TCP/IP networks) makes two transport protocols available to applications, UDP and TCP. When you (as an application developer) create a new network application for the Internet, one of the first decisions you have to make is whether to use UDP or TCP. Each of these protocols offers a different set of services to the invoking applications. Figure 2.4 shows the service requirements for some selected applications.

TCP Services

The TCP service model includes a connection-oriented service and a reliable data transfer service. When an application invokes TCP as its transport protocol, the application receives both of these services from TCP.

- *Connection-oriented service.* TCP has the client and server exchange transport-layer control information with each other *before* the application-level messages begin to flow. This so-called handshaking procedure alerts the client and server, allowing them to prepare for an onslaught of packets. After the handshaking phase, a **TCP connection** is said to exist between the sockets

Application	Data Loss	Throughput	Time-Sensitive
File transfer/download	No loss	Elastic	No
E-mail	No loss	Elastic	No
Web documents	No loss	Elastic (few kbps)	No
Internet telephony/ Video conferencing	Loss-tolerant	Audio: few kbps–1Mbps Video: 10 kbps–5 Mbps	Yes: 100s of msec
Streaming stored audio/video	Loss-tolerant	Same as above	Yes: few seconds
Interactive games	Loss-tolerant	Few kbps–10 kbps	Yes: 100s of msec
Smartphone messaging	No loss	Elastic	Yes and no

Figure 2.4 ♦ Requirements of selected network applications

of the two processes. The connection is a full-duplex connection in that the two processes can send messages to each other over the connection at the same time. When the application finishes sending messages, it must tear down the connection. In Chapter 3 we'll discuss connection-oriented service in detail and examine how it is implemented.

- *Reliable data transfer service.* The communicating processes can rely on TCP to deliver all data sent without error and in the proper order. When one side of the application passes a stream of bytes into a socket, it can count on TCP to deliver the same stream of bytes to the receiving socket, with no missing or duplicate bytes.

TCP also includes a congestion-control mechanism, a service for the general welfare of the Internet rather than for the direct benefit of the communicating processes. The TCP congestion-control mechanism throttles a sending process (client or server) when the network is congested between sender and receiver. As we will see

FOCUS ON SECURITY

SECURING TCP

Neither TCP nor UDP provides any encryption—the data that the sending process passes into its socket is the same data that travels over the network to the destination process. So, for example, if the sending process sends a password in cleartext (i.e., unencrypted) into its socket, the cleartext password will travel over all the links between sender and receiver, potentially getting sniffed and discovered at any of the intervening links. Because privacy and other security issues have become critical for many applications, the Internet community has developed an enhancement for TCP, called **Secure Sockets Layer (SSL)**. TCP-enhanced-with-SSL not only does everything that traditional TCP does but also provides critical process-to-process security services, including encryption, data integrity, and end-point authentication. We emphasize that SSL is not a third Internet transport protocol, on the same level as TCP and UDP, but instead is an enhancement of TCP, with the enhancements being implemented in the application layer. In particular, if an application wants to use the services of SSL, it needs to include SSL code (existing, highly optimized libraries and classes) in both the client and server sides of the application. SSL has its own socket API that is similar to the traditional TCP socket API. When an application uses SSL, the sending process passes cleartext data to the SSL socket; SSL in the sending host then encrypts the data and passes the encrypted data to the TCP socket. The encrypted data travels over the Internet to the TCP socket in the receiving process. The receiving socket passes the encrypted data to SSL, which decrypts the data. Finally, SSL passes the cleartext data through its SSL socket to the receiving process. We'll cover SSL in some detail in Chapter 8.

in Chapter 3, TCP congestion control also attempts to limit each TCP connection to its fair share of network bandwidth.

UDP Services

UDP is a no-frills, lightweight transport protocol, providing minimal services. UDP is connectionless, so there is no handshaking before the two processes start to communicate. UDP provides an unreliable data transfer service—that is, when a process sends a message into a UDP socket, UDP provides *no* guarantee that the message will ever reach the receiving process. Furthermore, messages that do arrive at the receiving process may arrive out of order.

UDP does not include a congestion-control mechanism, so the sending side of UDP can pump data into the layer below (the network layer) at any rate it pleases. (Note, however, that the actual end-to-end throughput may be less than this rate due to the limited transmission capacity of intervening links or due to congestion).

Services Not Provided by Internet Transport Protocols

We have organized transport protocol services along four dimensions: reliable data transfer, throughput, timing, and security. Which of these services are provided by TCP and UDP? We have already noted that TCP provides reliable end-to-end data transfer. And we also know that TCP can be easily enhanced at the application layer with SSL to provide security services. But in our brief description of TCP and UDP, conspicuously missing was any mention of throughput or timing guarantees—services *not* provided by today's Internet transport protocols. Does this mean that time-sensitive applications such as Internet telephony cannot run in today's Internet? The answer is clearly no—the Internet has been hosting time-sensitive applications for many years. These applications often work fairly well because they have been designed to cope, to the greatest extent possible, with this lack of guarantee. We'll investigate several of these design tricks in Chapter 9. Nevertheless, clever design has its limitations when delay is excessive, or the end-to-end throughput is limited. In summary, today's Internet can often provide satisfactory service to time-sensitive applications, but it cannot provide any timing or throughput guarantees.

Figure 2.5 indicates the transport protocols used by some popular Internet applications. We see that e-mail, remote terminal access, the Web, and file transfer all use TCP. These applications have chosen TCP primarily because TCP provides reliable data transfer, guaranteeing that all data will eventually get to its destination. Because Internet telephony applications (such as Skype) can often tolerate some loss but require a minimal rate to be effective, developers of Internet telephony applications usually prefer to run their applications over UDP, thereby circumventing TCP's congestion control mechanism and packet overheads. But because many firewalls are configured to block (most types of) UDP traffic, Internet telephony applications often are designed to use TCP as a backup if UDP communication fails.

Application	Application-Layer Protocol	Underlying Transport Protocol
Electronic mail	SMTP [RFC 5321]	TCP
Remote terminal access	Telnet [RFC 854]	TCP
Web	HTTP [RFC 2616]	TCP
File transfer	FTP [RFC 959]	TCP
Streaming multimedia	HTTP (e.g., YouTube)	TCP
Internet telephony	SIP [RFC 3261], RTP [RFC 3550], or proprietary (e.g., Skype)	UDP or TCP

Figure 2.5 ♦ Popular Internet applications, their application-layer protocols, and their underlying transport protocols

2.1.5 Application-Layer Protocols

We have just learned that network processes communicate with each other by sending messages into sockets. But how are these messages structured? What are the meanings of the various fields in the messages? When do the processes send the messages? These questions bring us into the realm of application-layer protocols. An **application-layer protocol** defines how an application's processes, running on different end systems, pass messages to each other. In particular, an application-layer protocol defines:

- The types of messages exchanged, for example, request messages and response messages
- The syntax of the various message types, such as the fields in the message and how the fields are delineated
- The semantics of the fields, that is, the meaning of the information in the fields
- Rules for determining when and how a process sends messages and responds to messages

Some application-layer protocols are specified in RFCs and are therefore in the public domain. For example, the Web's application-layer protocol, HTTP (the HyperText Transfer Protocol [RFC 2616]), is available as an RFC. If a browser developer follows the rules of the HTTP RFC, the browser will be able to retrieve Web pages from any Web server that has also followed the rules of the HTTP RFC. Many other application-layer protocols are proprietary and intentionally not available in the public domain. For example, Skype uses proprietary application-layer protocols.