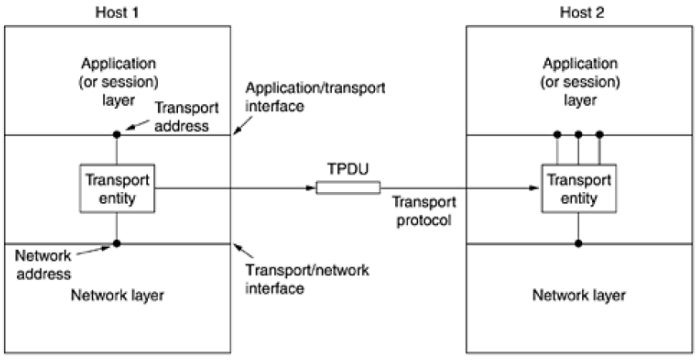
Transport Layer

Unit 5

The transport layer is not just another layer. It is the heart of the whole protocol hierarchy. Its task is to provide reliable, cost-effective data transport from the source machine to the destination machine, independently of the physical network or networks currently in use. Without the transport layer, the whole concept of layered protocols would make little sense. In this chapter we will study the transport layer in detail, including its services, design, protocols, and performance.



Connection Management Modeling

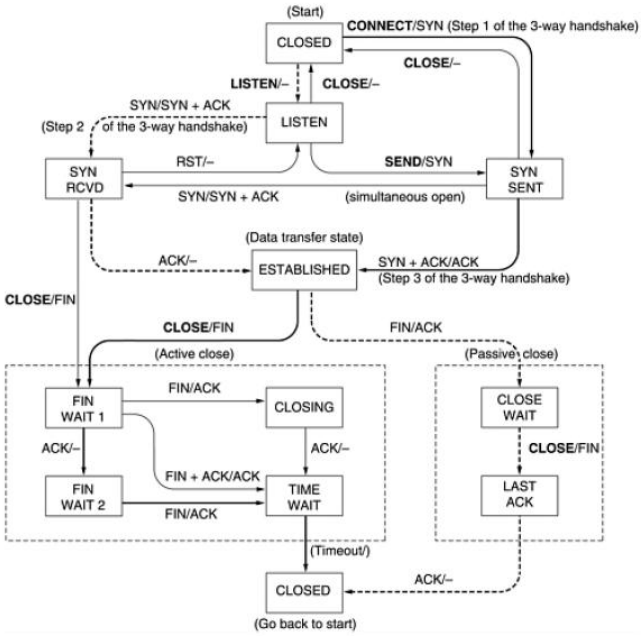
The steps required to establish and release connections can be represented in a finite state machine with the 11 states listed in Fig. 6-32. In each state, certain events are legal. When a legal event happens, some action may be taken. If some other event happens, an error is reported.

State	Description
CLOSED	No connection is active or pending
LISTEN	The server is waiting for an incoming call
SYN RCVD	A connection request has arrived; wait for ACK
SYN SENT	The application has started to open a connection
ESTABLISHED	The normal data transfer state
FIN WAIT 1	The application has said it is finished
FIN WAIT 2	The other side has agreed to release
TIMED WAIT	Wait for all packets to die off
CLOSING	Both sides have tried to close simultaneously
CLOSE WAIT	The other side has initiated a release
LAST ACK	Wait for all packets to die off

Each connection starts in the *CLOSED* state. It leaves that state when it does either a passive open (LISTEN), or an active open (CONNECT). If the other side does the opposite one, a connection is established and the state becomes *ESTABLISHED*. Connection release can be initiated by either side. When it is complete, the state returns to *CLOSED*.

The finite state machine itself is shown in Fig. 6-33. The common case of a client actively connecting to a passive server is shown with heavy lines—solid for the client, dotted for the server. The lightface lines are unusual event sequences. Each line in Fig. 6-33 is marked by an *event/action* pair. The event can either be a user-initiated system call (CONNECT, LISTEN, SEND, or CLOSE), a segment arrival (*SYN*, *FIN*, *ACK*, or *RST*), or, in one case, a

timeout of twice the maximum packet lifetime. The action is the sending of a control segment (*SYN*, *FIN*, or *RST*) or nothing, indicated by —. Comments are shown in parentheses.



Introduction to TCP

TCP (**Transmission Control Protocol**) was specifically designed to provide a reliable end-to-end byte stream over an unreliable internetwork. An internetwork differs from a single network because different parts may have wildly different topologies, bandwidths, delays, packet sizes, and other parameters. TCP was designed to dynamically adapt to properties of the internetwork and to be robust in the face of many kinds of failures.

TCP was formally defined in RFC 793. As time went on, various errors and inconsistencies were detected, and the requirements were changed in some areas. These clarifications and some bug fixes are detailed in RFC 1122. Extensions are given in RFC 1323.

Each machine supporting TCP has a TCP transport entity, either a library procedure, a user process, or part of the kernel. In all cases, it manages TCP streams and interfaces to the IP layer. A TCP entity accepts user data streams from local processes, breaks them up into pieces not exceeding 64 KB (in practice, often 1460 data bytes in order to fit in a single Ethernet frame with the IP and TCP headers), and sends each piece as a separate IP datagram. When datagrams containing TCP data arrive at a machine, they are given to the TCP entity, which reconstructs the original byte streams. For simplicity, we will sometimes use just "TCP" to mean the TCP transport entity (a piece of software) or the TCP protocol (a set of rules). From the context it will be clear which is meant. For example, in "The user gives TCP the data," the TCP transport entity is clearly intended.

The IP layer gives no guarantee that datagrams will be delivered properly, so it is up to TCP to time out and retransmit them as need be. Datagrams that do arrive may well do so in the wrong order; it is also up to TCP to reassemble them into messages in the proper sequence. In short, TCP must furnish the reliability that most users want and that IP does not provide.

The TCP Service Model

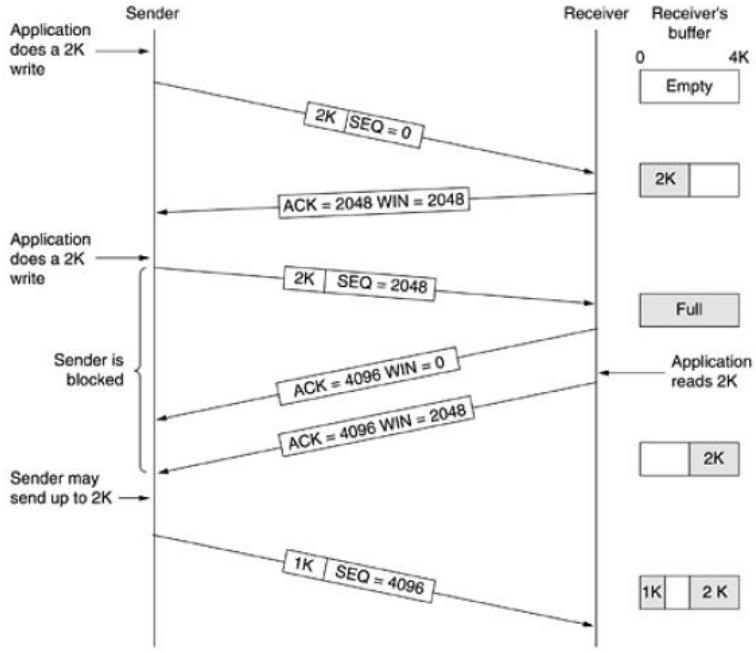
TCP service is obtained by both the sender and receiver creating end points, called sockets, as discussed in Sec. 6.1.3. Each socket has a socket number (address) consisting of the IP address of the host and a 16-bit number local to that host, called a **port**. A port is the TCP name for a TSAP. For TCP service to be obtained, a connection must be explicitly established between a socket on the sending machine and a socket on the receiving machine. The socket calls are listed in Fig. 6-5.

A socket may be used for multiple connections at the same time. In other words, two or more connections may terminate at the same socket. Connections are identified by the socket identifiers at both ends, that is, (*socket1*, *socket2*). No virtual circuit numbers or other identifiers are used.

Port numbers below 1024 are called **well-known ports** and are reserved for standard services. For example, any process wishing to establish a connection to a host to transfer a file using FTP can connect to the destination host's port 21 to contact its FTP daemon.

Window Management

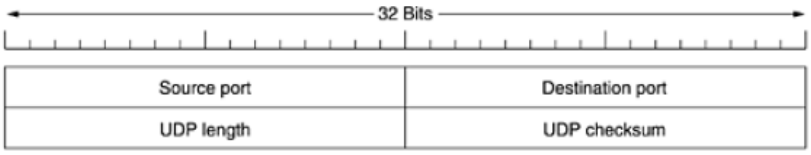
Window management in TCP is not directly tied to acknowledgements as it is in most data link protocols. For example, suppose the receiver has a 4096-byte buffer, as shown in Fig. 6-34. If the sender transmits a 2048-byte segment that is correctly received, the receiver will acknowledge the segment. However, since it now has only 2048 bytes of buffer space (until the application removes some data from the buffer), it will advertise a window of 2048 starting at the next byte expected.



Introduction to UDP

The Internet protocol suite supports a connectionless transport protocol, **UDP** (**User Datagram Protocol**). UDP provides a way for applications to send encapsulated IP datagrams and send them without having to establish a connection. UDP is described in RFC 768.

UDP transmits **segments** consisting of an 8-byte header followed by the payload. The header is shown in Fig. 6-23. The two ports serve to identify the end points within the source and destination machines. When a UDP packet arrives, its payload is handed to the process attached to the destination port. This attachment occurs when BIND primitive or something similar is used, as we saw in Fig. 6-6 for TCP (the binding process is the same for UDP). In fact, the main value of having UDP over just using raw IP is the addition of the source and destination ports. Without the port fields, the transport layer would not know what to do with the packet. With them, it delivers segments correctly.



The source port is primarily needed when a reply must be sent back to the source. By copying the *source port* field from the incoming segment into the *destination port* field of the outgoing segment, the process sending the reply can specify which process on the sending machine is to get it.

The *UDP length* field includes the 8-byte header and the data. The *UDP checksum* is optional and stored as 0 if not computed (a true computed 0 is stored as all 1s). Turning it off is foolish unless the quality of the data does not matter (e.g., digitized speech).

It is probably worth mentioning explicitly some of the things that UDP does *not* do. It does not do flow control, error control, or retransmission upon receipt of a bad segment. All of that is up to the user processes. What it does do is provide an interface to the IP protocol with the added feature of demultiplexing multiple processes using the ports. That is all it does. For applications that need to have precise control over the packet flow, error control, or timing, UDP provides just what the doctor ordered.

One area where UDP is especially useful is in client-server situations. Often, the client sends a short request to the server and expects a short reply back. If either the request or reply is lost, the client can just time out and try again. Not only is the code simple, but fewer messages are required (one in each direction) than with a protocol requiring an initial setup.

Performance Issues

Performance issues are very important in computer networks. When hundreds or thousands of computers are interconnected, complex interactions, with unforeseen consequences, are common. Frequently, this complexity leads to poor performance and no one knows why. In the following sections, we will examine many issues related to network performance to see what kinds of problems exist and what can be done about them.

Unfortunately, understanding network performance is more an art than a science. There is little underlying theory that is actually of any use in practice. The best we can do is give rules of thumb gained from hard experience and present examples taken from the real world. We have intentionally delayed this discussion until we studied the transport layer in TCP in order to be able to use TCP as an example in various places.

The transport layer is not the only place performance issues arise. We saw some of them in the network layer in the previous chapter. Nevertheless, the network layer tends to be largely concerned with routing and congestion control. The broader, system-oriented issues tend to be transport related, so this chapter is an appropriate place to examine them.

In the next five sections, we will look at five aspects of network performance:

1. Performance problems.
2. Measuring network performance.
3. System design for better performance.
4. Fast TPDU processing.
5. Protocols for future high-performance networks.

As an aside, we need a generic name for the units exchanged by transport entities. The TCP term, segment, is confusing at best and is never used outside the TCP world in this context. The ATM terms (CS-PDU, SAR-PDU, and CPCS-PDU) are specific to ATM. Packets clearly refer to the network layer, and messages belong to the application layer. For lack of a standard term, we will go back to calling the units exchanged by transport entities TPDUs. When we mean both TPDU and packet together, we will use packet as the collective term, as in "The CPU must be fast enough to process incoming packets in real time." By this we mean both the network layer packet and the TPDU encapsulated in it.

1 Performance Problems in Computer Networks

Some performance problems, such as congestion, are caused by temporary resource overloads. If more traffic suddenly arrives at a router than the router can handle, congestion will build up and performance will suffer. We studied congestion in detail in the previous chapter.

Performance also degrades when there is a structural resource imbalance. For example, if a gigabit communication line is attached to a low-end PC, the poor CPU will not be able to process the incoming packets fast enough and some will be lost. These packets will eventually be retransmitted, adding delay, wasting bandwidth, and generally reducing performance.

Overloads can also be synchronously triggered. For example, if a TPDU contains a bad parameter (e.g., the port for which it is destined), in many cases the receiver will thoughtfully send back an error notification. Now consider what could happen if a bad TPDU is broadcast to 10,000 machines: each one might send back an error message. The resulting **broadcast storm** could cripple the network. UDP suffered from this problem until the protocol was changed to cause hosts to refrain from responding to errors in UDP TPDUs sent to broadcast addresses.

A second example of synchronous overload is what happens after an electrical power failure. When the power comes back on, all the machines simultaneously jump to their ROMs to start rebooting. A typical reboot sequence might require first going to some (DHCP) server to learn one's true identity, and then to some file server to get a copy of the operating system. If hundreds of machines all do this at once, the server will probably collapse under the load.

Even in the absence of synchronous overloads and the presence of sufficient resources, poor performance can occur due to lack of system tuning. For example, if a machine has plenty of CPU power and memory but not enough of the memory has been allocated for buffer space, overruns will occur and TPDUs will be lost. Similarly, if the scheduling algorithm does not give a high enough priority to processing incoming TPDUs, some of them may be lost.