UNIT-IV

TRANSPORT LAYER

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.

3.1. The Transport Service:

transport layer provides different kinds of service to the application layer

3.1.1. Services Provided to the Upper Layers

The ultimate goal of the transport layer is to provide efficient, reliable, and cost-effective service to its users, normally processes in the application layer. To achieve this goal, the transport layer makes use of the services provided by the network layer. The hardware and/or software within the transport layer that does the work is called the transport entity. The transport entity can be located in the operating system kernel, in a separate user process, in a library package bound into network applications, or conceivably on the network interface card.

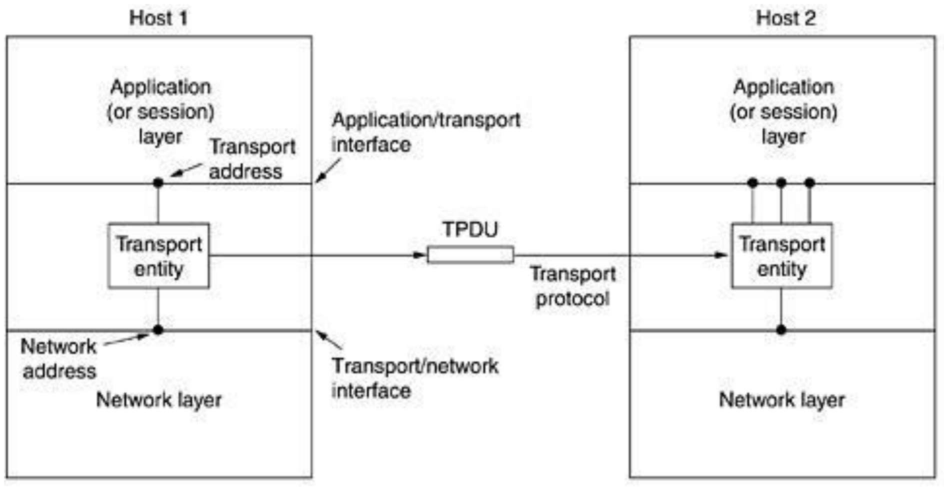


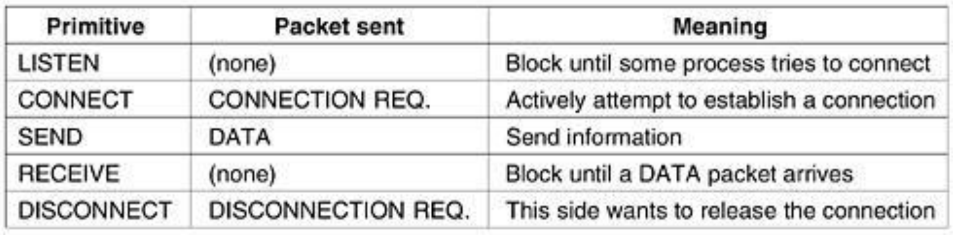
Figure: The network, transport, and application layers

There are two types of network service, **connection-oriented and connectionless**, there are also two types of transport service. The **connection-oriented transport service** is similar to the connection-oriented network service in many ways. In both cases, connections have **three phases: establishment, data transfer, and release**. Addressing and flow control are also similar in both layers. Furthermore, the connectionless transport service is also very similar to the connectionless network service.

3.1.2 Transport Service Primitives:

The transport service is similar to the network service, but there are also some important differences. The main difference is that the network service is intended to model the service offered by real networks, warts and all.

To allow users to access the transport service, the transport layer must provide some operations to application programs, that is, a transport service interface. Each transport service has its own interface. The (connection-oriented) transport service, in contrast, is reliable. Of course, real networks are not error-free, but thatis precisely the purpose of the transport layer—to provide a reliable service on top of an unreliable network.



Primitive	Packet sent	Meaning
LISTEN	(none)	Block until some process tries to connect
CONNECT	CONNECTION REQ.	Actively attempt to establish a connection
SEND	DATA	Send information
RECEIVE	(none)	Block until a DATA packet arrives
DISCONNECT	DISCONNECTION REQ.	This side wants to release the connection

Figure: The primitives for a simple transport service.

TPDU (Transport Protocol Data Unit) for messages sent from transport entity to transport entity. TPDUs (exchanged by the transport layer) are contained in packets (exchanged by the network layer). In turn, packets are contained in frames (exchanged by the data link layer). When a frame arrives, the data link layer processes the frame header and passes the contents of the frame payload field up to the network entity. The network entity processes the packet header and passes the contents of the packet payload up to the transport entity. This nesting is illustrated in Figure.

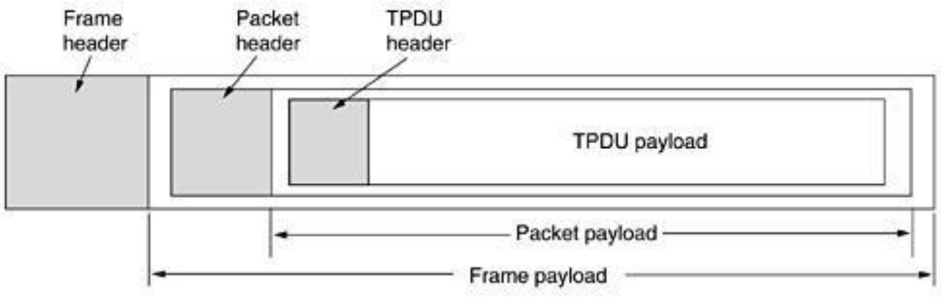
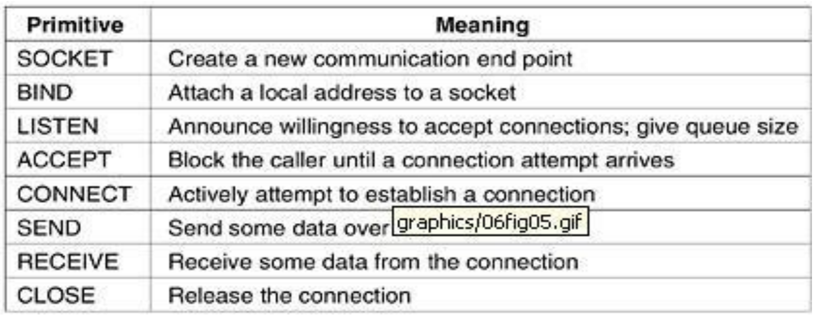


Figure: Nesting of TPDUs, packets, and frames

The client's CONNECT call causes a CONNECTION REQUEST TPDU to be sent to the server. When it arrives, the transport entity checks to see that the server is blocked on a LISTEN (i.e., is interested in handling requests). It then unblocks the server and sends a CONNECTION ACCEPTED TPDU back to the client. When this TPDU arrives, the client is unblocked and the connection is established. Data can now be exchanged using the SEND and RECEIVE primitives. In the simplest form, either party can do a (blocking) RECEIVE to wait for the other party to do a SEND.When the TPDU arrives, the receiver is unblocked.

3.1.3 Berkeley Sockets:

Another set of transport primitives, the socket primitives used in Berkeley UNIX for TCP. These primitives are widely used for Internet programming.



Primitive	Meaning
SOCKET	Create a new communication end point
BIND	Attach a local address to a socket
LISTEN	Announce willingness to accept connections; give queue size
ACCEPT	Block the caller until a connection attempt arrives
CONNECT	Actively attempt to establish a connection
SEND	Send some data over graphics/06fig05.gif
RECEIVE	Receive some data from the connection
CLOSE	Release the connection

Figure:. The socket primitives for TCP.

The first four primitives in the list are executed in that order by servers.

The SOCKET primitive creates a new end point and allocates table space for it within the transport entity. The parameters of the call specify the addressing format to be used, the type of service desired (e.g., reliable byte stream), and the protocol.

A successful SOCKET call returns an ordinary file descriptor for use in succeeding calls, the same way an OPEN call does.

Newly-created sockets do not have network addresses. These are assigned using the BIND primitive.

Next comes the LISTEN call, which allocates space to queue incoming calls for the case that several clients try to connect at the same time. In contrast to LISTEN in our first example, in the socket model LISTEN is not a blocking call.

To block waiting for an incoming connection, the server executes an ACCEPT primitive.

3.2 ELEMENTS OF TRANSPORT PROTOCOLS:

The transport service is implemented by a **transport protocol** used between the two transport entities.

3.2.1. Addressing:

When an application (e.g., a user) process wishes to set up a connection to a remote application process, it must specify which one to connect to. In the Internet, these endpoints are called **ports**. We will use the generic term **TSAP** (**Transport Service Access Point**) to mean a specific endpoint in the transport layer. The analogous endpoints in the network layer (i.e., network layer addresses) are not-surprisingly called **NSAPs** (**Network Service Access Points**). IP addresses are examples of NSAPs. Figure illustrates the relationship between the NSAPs, the TSAPs, and a transport connection. Application processes, both clients and servers, can attach themselves to a local TSAP to establish a connection to a remote TSAP.

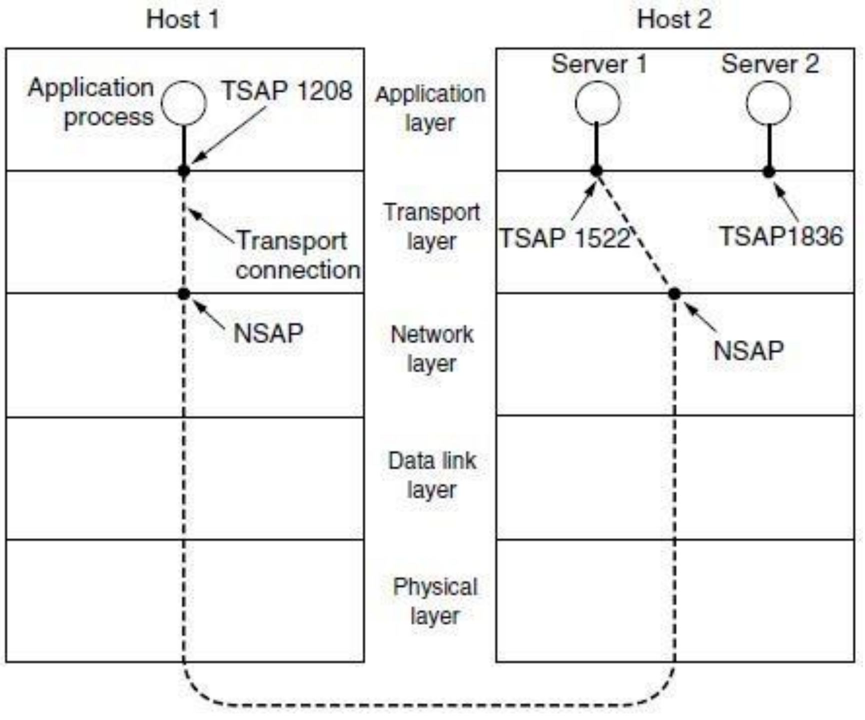


Figure . TSAPs, NSAPs, and transport connections.

A possible scenario for a transport connection is as follows:

1. A mail server process attaches itself to TSAP 1522 on host 2 to wait for an incoming call. How a process attaches itself to a TSAP is outside the networking model and depends entirely on the local operating system. A call such as our LISTEN might be used, for example.

2. An application process on host 1 wants to send an email message, so it attaches itself to TSAP 1208 and issues a CONNECT request. The request specifies TSAP 1208 on host 1 as the source and TSAP 1522 on host 2 as the destination. This action ultimately results in a transport connection being established between the application process and the server.

3. The application process sends over the mail message.

4. The mail server responds to say that it will deliver the message.

5. The transport connection is released.

3.2.2 Connection Establishment:

Establishing a connection sounds easy, but it is actually surprisingly tricky. At first glance, it would seem sufficient for one transport entity to just send a CONNECTION REQUEST segment to the destination and wait for a CONNECTION ACCEPTED reply.

The problem occurs when the network can lose, delay, corrupt, and duplicate packets. This behavior causes serious complications.

Imagine a network that is so congested that acknowledgements hardly ever get back in time and each packet times out and is retransmitted two or three times.

Suppose that the network uses datagrams inside and that every packet follows a different route. Some of the packets might get stuck in a traffic jam inside the network and take a long time to arrive.

Another possibility is to give each connection a unique identifier (i.e., a sequence number incremented for each connection established) chosen by the initiating party and put in each segment, including the one requesting the connection.

Packet lifetime can be restricted to a known maximum using one (or more) of the following techniques:

1. Restricted network design.

2. Putting a hop counter in each packet.

3. Timestamping each packet.

Three-way handshake:

Tomlinson (1975) introduced the **three-way handshake**. This establishment protocol involves one peer checking with the other that the connection request is indeed current. The normal setup procedure when host 1 initiates is shown in Fig. (a). Host 1 chooses a sequence number, *x*, and sends a CONNECTION REQUEST segment containing it to host 2. Host 2 replies with an ACK segment acknowledging *x* and announcing its own initial sequence number, *y*. Finally, host 1 acknowledges host 2's choice of an initial sequence number in the first data segment that it sends.

Now let us see how the three-way handshake works in the presence of delayed duplicate control segments. In Fig.(b), the first segment is a delayed duplicate CONNECTION REQUEST from an old connection. This segment arrives at host 2 without host 1's knowledge. Host 2 reacts to this segment by sending host 1 an ACK segment, in effect asking for verification that host 1 was indeed trying to set up a new connection. When host 1 rejects host 2's attempt to establish a connection, host 2 realizes that it was tricked by a delayed duplicate and abandons the connection. In this way, a delayed duplicate does no damage.

The worst case is when both a delayed CONNECTION REQUEST and an ACK are floating around in the subnet. This case is shown in Fig. (c). As in the previous example, host 2 gets a delayed CONNECTION REQUEST and replies to it. At this point, it is crucial to realize that host 2 has proposed using *y* as the initial sequence number for host 2 to host 1 traffic, knowing full well that no segments containing sequence number *y* or acknowledgements to *y* are still in existence.

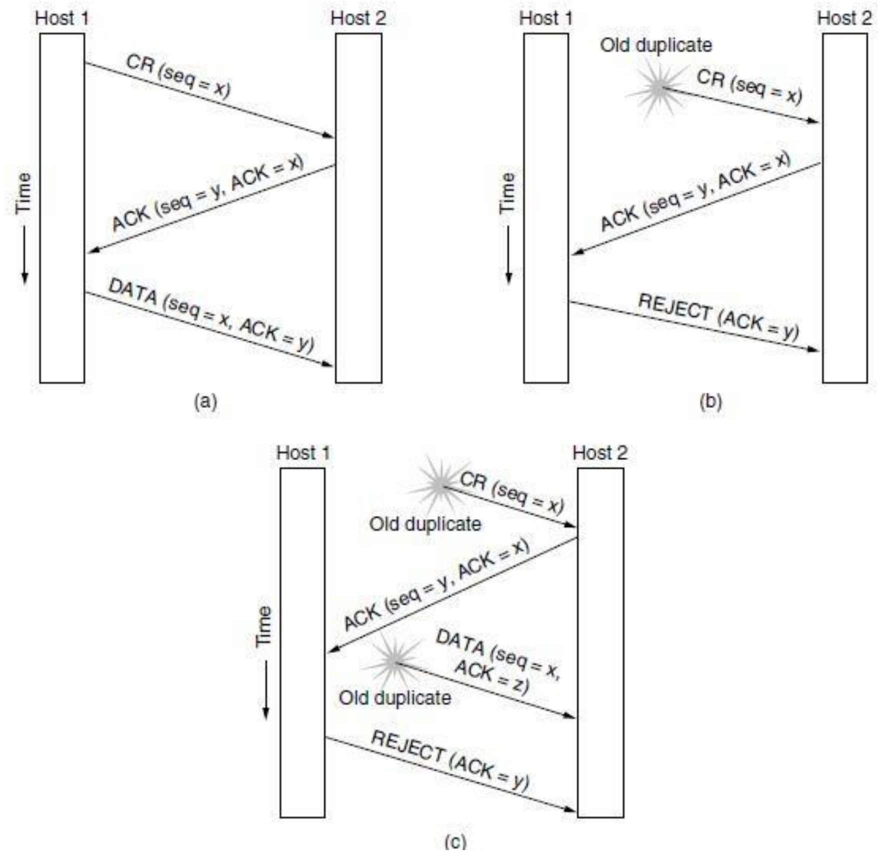


Figure. Three protocol scenarios for establishing a connection using a three-way handshake. CR denotes CONNECTION REQUEST. (a) Normal operation. (b) Old duplicate CONNECTION REQUEST appearing out of nowhere. (c) Duplicate CONNECTION REQUEST and duplicate ACK

3.2.3 Connection Release:

Releasing a connection is easier than establishing one. There are two styles of terminating a connection: asymmetric release and symmetric release. Asymmetric release is the way the telephone system works: when one party hangs up, the connection is broken. Symmetric release treats the connection as two separate unidirectional connections and requires each one to be released separately. Asymmetric release is abrupt and may result in data loss. Consider the scenario of Fig. After the connection is established, host 1 sends a segment that arrives properly at host 2. Then host 1 sends another segment. Unfortunately, host 2 issues a DISCONNECT before the second segment arrives. The result is that the connection is released and data are lost.

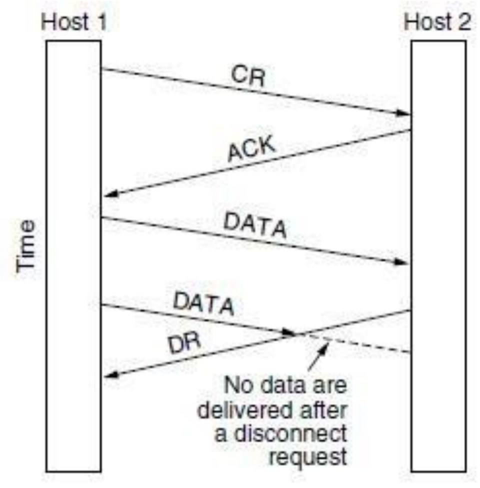


Figure. Abrupt disconnection with loss of data.

Clearly, a more sophisticated release protocol is needed to avoid data loss. One way is to use symmetric release, in which each direction is released independently of the other one. Here, a host can continue to receive data even after it has sent a DISCONNECT segment. Symmetric release does the job when each process has afixed amount of data to send and clearly knows when it has sent it.

Two-army problem: Imagine that a white army is encamped in a valley, as shown in Fig. On both of the surrounding hillsides are blue armies. The white army is larger than either of the blue armies alone, but togetherthe blue armies are larger than the white army. If either blue army attacks by itself, it will be defeated, but if the two blue armies attack simultaneously, they will be victorious.

The blue armies want to synchronize their attacks. However, their only communication medium is to send messengers on foot down into the valley, where they might be captured and the message lost

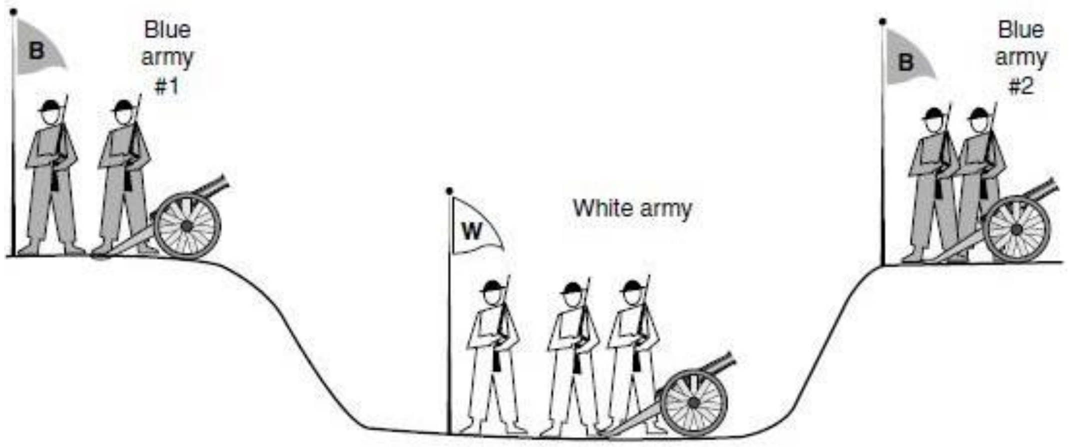


Figure. The two-army problem.

Suppose that the commander of blue army #1 sends a message reading: ‘‘I propose we attack at dawn on March 29. How about it?’’ Now suppose that the message arrives, the commander of blue army #2 agrees, and his reply gets safely back to blue army #1. Will the attack happen? Probably not, because commander #2 does not know if his reply got through. If it did not, blue army #1 will not attack, so it would be foolish for him to charge into battle.

Now let us improve the protocol by making it a three-way handshake. The initiator of the original proposal must acknowledge the response. Assuming no messages are lost, blue army #2 will get the acknowledgement, but the commander of blue army #1 will now hesitate. After all, he does not know if his acknowledgement got through, and if it did not, he knows that blue army #2 will not attack. We could now make a four-way handshake protocol, but that does not help either.

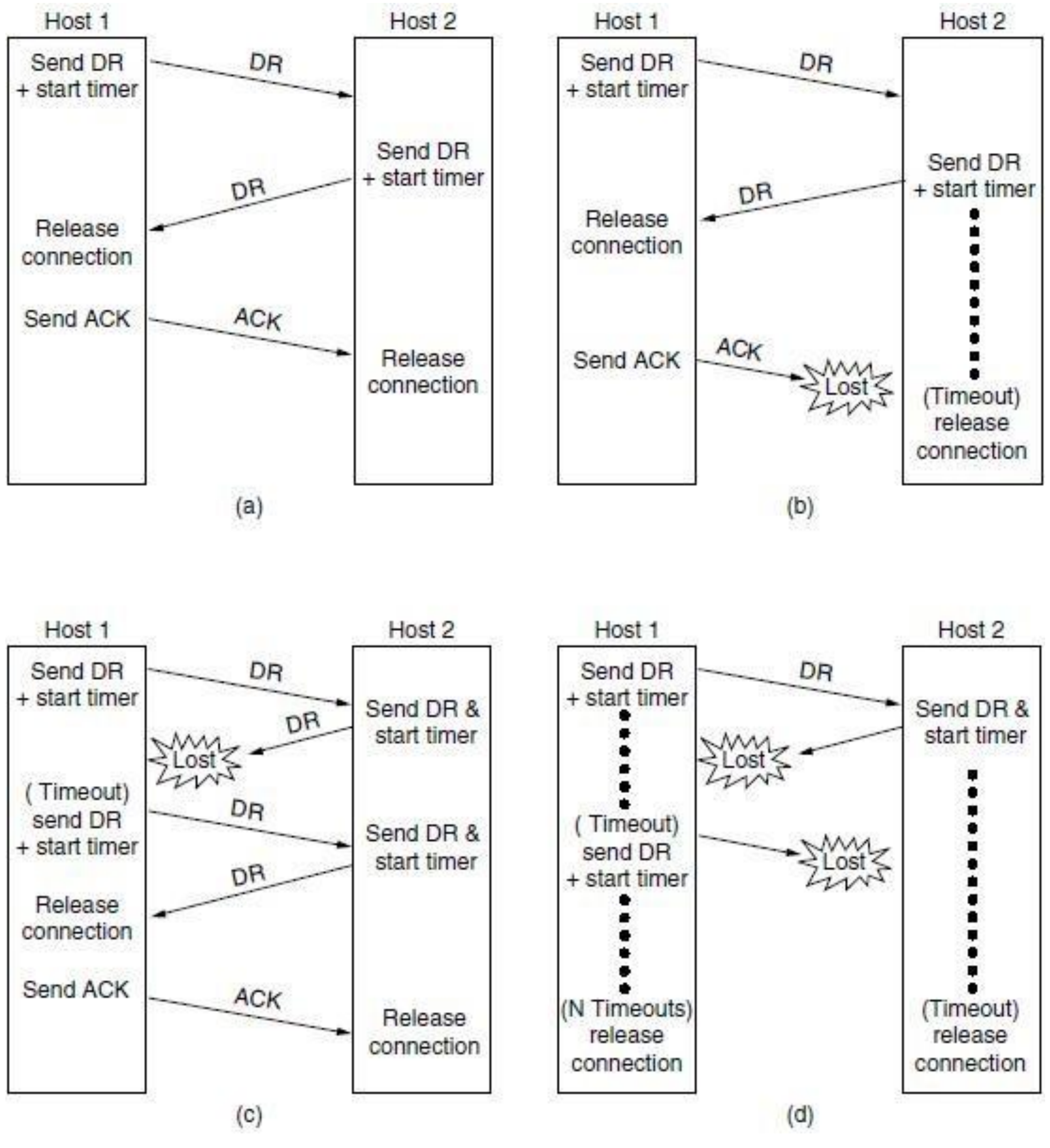


Figure . Four protocol scenarios for releasing a connection. (a) Normal case of three-way handshake. (b) Final ACK lost. (c) Response lost. (d) Response lost and subsequent DRs lost.

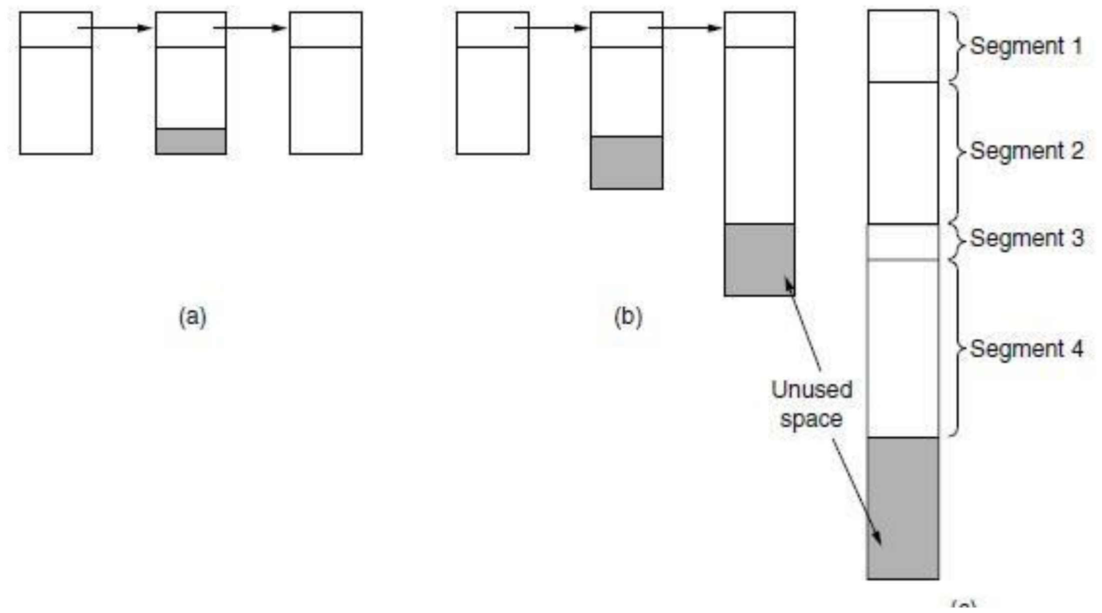
3.2.4 Error Control and Flow Control:

Error control is ensuring that the data is delivered with the desired level of reliability, usually that all of the data is delivered without any errors. Flow control is keeping a fast transmitter from overrunning a slow receiver.

1. A frame carries an error-detecting code (e.g., a CRC or checksum) that is used to check if the information was correctly received.
2. A frame carries a sequence number to identify itself and is retransmitted by the sender until it receives an

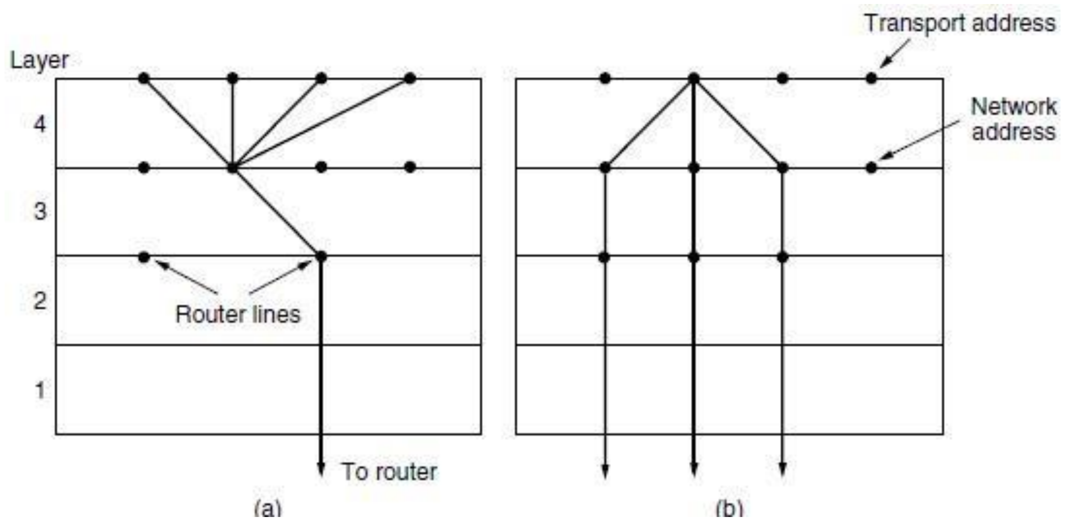
acknowledgement of successful receipt from the receiver. This is called **ARQ** (**Automatic Repeat reQuest**).

3. There is a maximum number of frames that the sender will allow to be outstanding at any time, pausing if the receiver is not acknowledging frames quickly enough. If this maximum is one packet the protocol is called **stop-and-wait**. Larger windows enable pipelining and improve performance on long, fast links.
4. The **sliding window** protocol combines these features and is also used to support bidirectional data transfer.



3.2.5 Multiplexing

Multiplexing, or sharing several conversations over connections, virtual circuits, and physical links plays a role in several layers of the network architecture. In the transport layer, the need for multiplexing can arise in a number of ways. When a segment comes in, some way is needed to tell which process to give it to. This situation, called **multiplexing**, is shown in Fig.(a). In this figure, four distinct transport connections all use the same network connection (e.g., IP address) to the remote host.



3.2.6 Crash Recovery

If hosts and routers are subject to crashes or connections are long-lived (e.g., large software or media downloads), recovery from these crashes becomes an issue. If the transport entity is entirely within the hosts, recovery from network and router crashes is straightforward. The transport entities expect lost segments
all the time and know how to cope with them by using retransmissions.

A more troublesome problem is how to recover from host crashes. In particular, it may be desirable for clients to be able to continue working when servers crash and quickly reboot.
In an attempt to recover its previous status, the server might send a broadcast segment to all other hosts, announcing that it has just crashed and requesting that its clients inform it of the status of all open connections.Each client can be in one of two states: one segment outstanding, *S1*, or no segments outstanding, *S0*. Based on only this state information, the client must decide whether to retransmit the most recent segment.

3.3 CONGESTION CONTROL:

If the transport entities on many machines send too many packets into the network too quickly, the network will become congested, with performance degraded as packets are delayed and lost. Controlling congestion to avoid this problem is the combined responsibility of the network and transport layers. Congestion occurs at routers, so it is detected at the network layer. However, congestion is ultimately caused by traffic sent into the network by the transport layer. The only effective way to control congestion is for the transport protocolsto send packets into the network more slowly.

3.3.1 Desirable Bandwidth Allocation:

The goal is more than to simply avoid congestion. It is to find a good allocation of bandwidth to thetransport entities that are using the network. A good allocation will deliver good performance because it uses allthe available bandwidth but avoids congestion, it will be fair across competing transport entities, and it will quickly track changes in traffic demands. **Efficiency and Power** An efficient allocation of bandwidth across transport entities will use all of the network capacity that is available.

However, it is not quite right to think that if there is a 100-Mbps link, five transport entities should get 20 Mbps each. They should usually get less than 20 Mbps for good performance. The reason is that the traffic is often bursty.

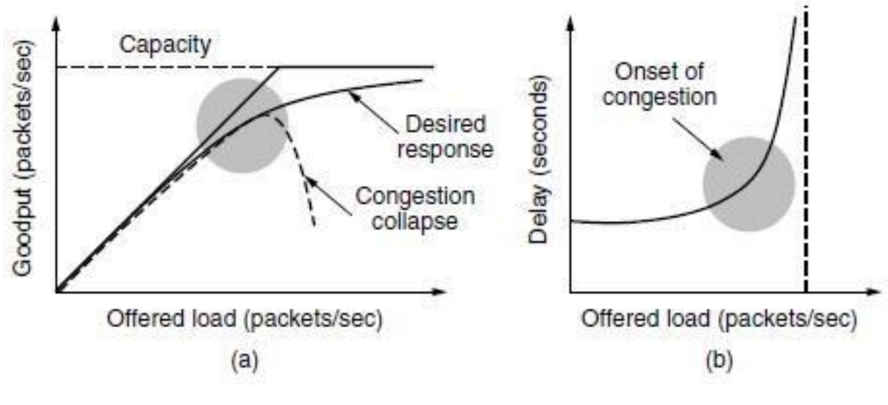


Figure . (a) Goodput and (b) delay as a function of offered load.

As the load increases in Fig. (a) goodput initially increases at the same rate, but as the load approaches the capacity, goodput rises more gradually. This falloff is because bursts of traffic can occasionally mount up and cause some losses at buffers inside the network. For both goodput and delay, performance begins to degrade at the onset of congestion.

$$power = \frac{load}{delay}$$

Power will initially rise with offered load, as delay remains small and roughly constant, but will reach a maximum and fall as delay grows rapidly. The load with the highest power represents an efficient load for the transport entity to place on the network.

3.3.2 Regulating the Sending Rate:

The sending rate may be limited by two factors. The first is flow control, in the case that there is insufficient buffering at the receiver. The second is congestion, in the case that there is insufficient capacity in the network. In Fig. This problem illustrated hydraulically. In Fig.(a), we see a thick pipe leading to a small-capacity receiver. This is a flow-control limited situation. As long as the sender does not send more water than the bucket can contain, no water will be lost. In Fig.(b), the limiting factor is not the bucket capacity, but the internal carrying capacity of the network. If too much water comes in too fast, it will back up and some will be lost (in this case, by overflowing the funnel).

These cases may appear similar to the sender, as transmitting too fast causes packets to be lost. However, they have different causes and call for different solutions. We have already talked about a flow-control solution with a variable-sized window.

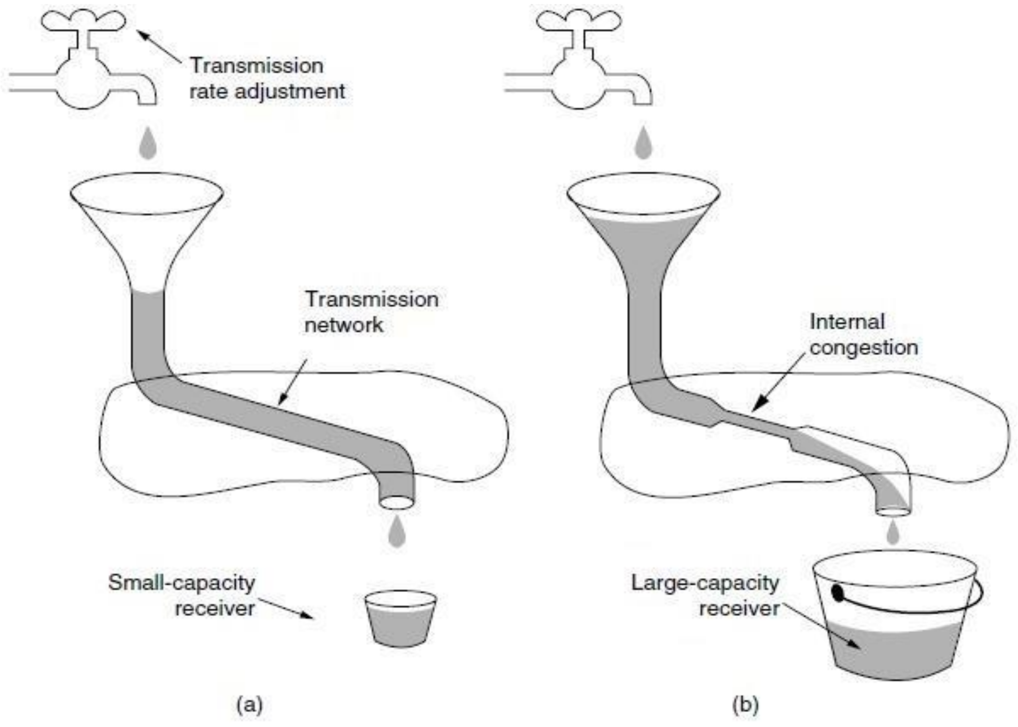


Figure . (a) A fast network feeding a low-capacity receiver. (b) A slow network feeding a high-capacity receiver.

3.4 THE INTERNET TRANSPORT PROTOCOLS: UDP

The Internet has two main protocols in the transport layer, a connectionless protocol and a connection- oriented one. The protocols complement each other. The connectionless protocol is UDP. It does almost nothing beyond sending packets between applications, letting applications build their own protocols on top as
needed. The connection-oriented protocol is TCP.

Since UDP is a transport layer protocol that typically runs in the operating system and protocols that use UDP typically run in user space, these uses might be considered applications.

3.4.1 Introduction to UDP

The Internet protocol suite supports a connectionless transport protocol called **UDP** (**User Datagram Protocol**). UDP provides a way for applications to send encapsulated IP datagrams without having to establish a connection.

UDP transmits **segments** consisting of an 8-byte header followed by the payload. The header is shown in Fig. 6-27. The two **ports** serve to identify the endpoints within the source and destination machines. When a UDP packet arrives, its payload is handed to the process attached to the destination port.

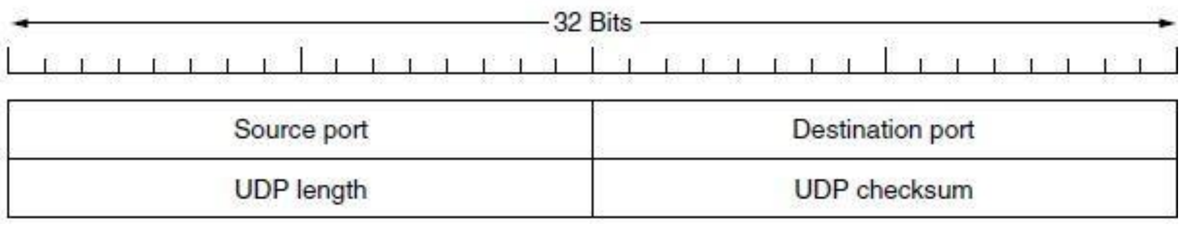


Figure . The UDP header.

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 minimum length is 8 bytes, to cover the header. The maximum length is 65,515 bytes, which is lower than the largest number that will fit in 16 bits because of the size limit on IP packets.

An optional *Checksum* is also provided for extra reliability. It checksums the header, the data, and a conceptual IP pseudoheader. When performing this computation, the *Checksum* field is set to zero and the data field is padded out with an additional zero byte if its length is an odd number.

The pseudoheader for the case of IPv4 is shown in Fig. It contains the 32-bit IPv4 addresses of the source and destination machines, the protocol number for UDP (17), and the byte count for the UDP segment (including the header).

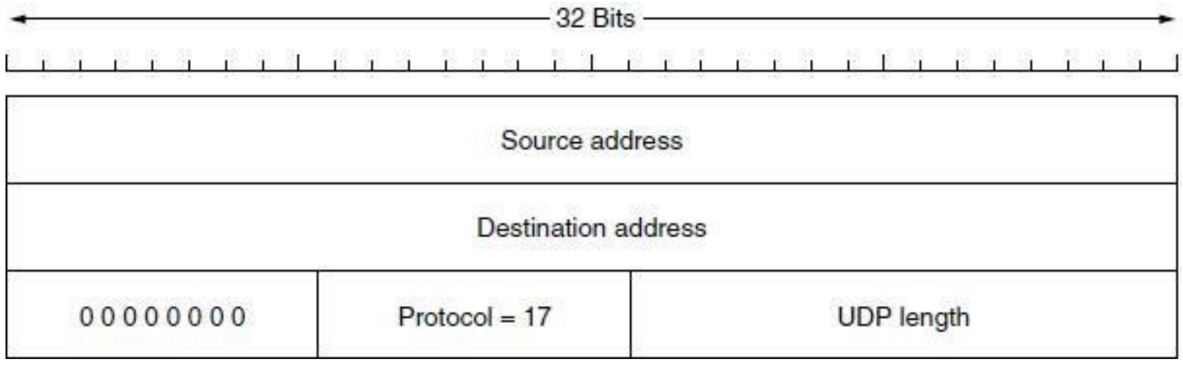


Figure. The IPv4 pseudoheader included in the UDP checksum.

3.4.2 Remote Procedure Call:

In a certain sense, sending a message to a remote host and getting a reply back is a lot like making a function call in a programming language. In both cases, you start with one or more parameters and you get back a result. This observation has led people to try to arrange request-reply interactions on networks to be cast in the form of procedure calls. Such an arrangement makes network applications much easier to program and more familiar to deal with.

The key work in this area was done by Birrell and Nelson (1984). In a nutshell, what Birrell and Nelson suggested was allowing programs to call procedures located on remote hosts. When a process on machine 1 calls a procedure on machine 2, the calling process on 1 is suspended and execution of the called procedure
takes place on 2. Information can be transported from the caller to the called in the parameters and can come back in the procedure result. No message passing is visible to the application programmer. This technique is known as **RPC** (**Remote Procedure Call**) and has become the basis for many networking applications.

Traditionally, the calling procedure is known as the client and the called procedure is known as the server, and we will use those names here too.

3.5 THE INTERNET TRANSPORT PROTOCOLS: TCP

TCP and is the main workhorse of the Internet.

3.5.1 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 partsmay 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.

3.5.2 The TCP Service Model

TCP service is obtained by both the sender and the 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 one machine and a socket on another machine. Port numbers below 1024 are reserved for standard services that can usually only be started by privileged users (e.g., root in UNIX systems). They are called **well-known ports.**

The list of well-known ports is given at *www.iana.org*. Over 700 have been assigned. A few of the better-known ones are listed in Fig.

Port	Protocol	Use
20, 21	FTP	File transfer
22	SSH	Remote login, replacement for Telnet
25	SMTP	Email
80	HTTP	World Wide Web
110	POP-3	Remote email access
143	IMAP	Remote email access
443	HTTPS	Secure Web (HTTP over SSL/TLS)
543	RTSP	Media player control
631	IPP	Printer sharing

Figure . Some assigned ports.

Other ports from 1024 through 49151 can be registered with IANA for use by unprivileged users, but applications can and do choose their own ports. For example, the BitTorrent peer-to-peer file-sharing application

3.5.3 The TCP Protocol:

A key feature of TCP, and one that dominates the protocol design, is that every byte on a TCP connection has its own 32-bit sequence number. The sending and receiving TCP entities exchange data in the form of segments. A **TCP segment** consists of a fixed 20-byte header (plus an optional part) followed by zero or more data bytes.

3.5.4 The TCP Segment Header

Figure shows the layout of a TCP segment. Every segment begins with a fixed-format, 20-byte header.The fixed header may be followed by header options. After the options, if any, up to 65,535 − 20 − 20 65,495 data bytes may follow, where the first 20 refer to the IP header and the second to the TCP header. Segments without any data are legal and are commonly used for acknowledgements and control messages.

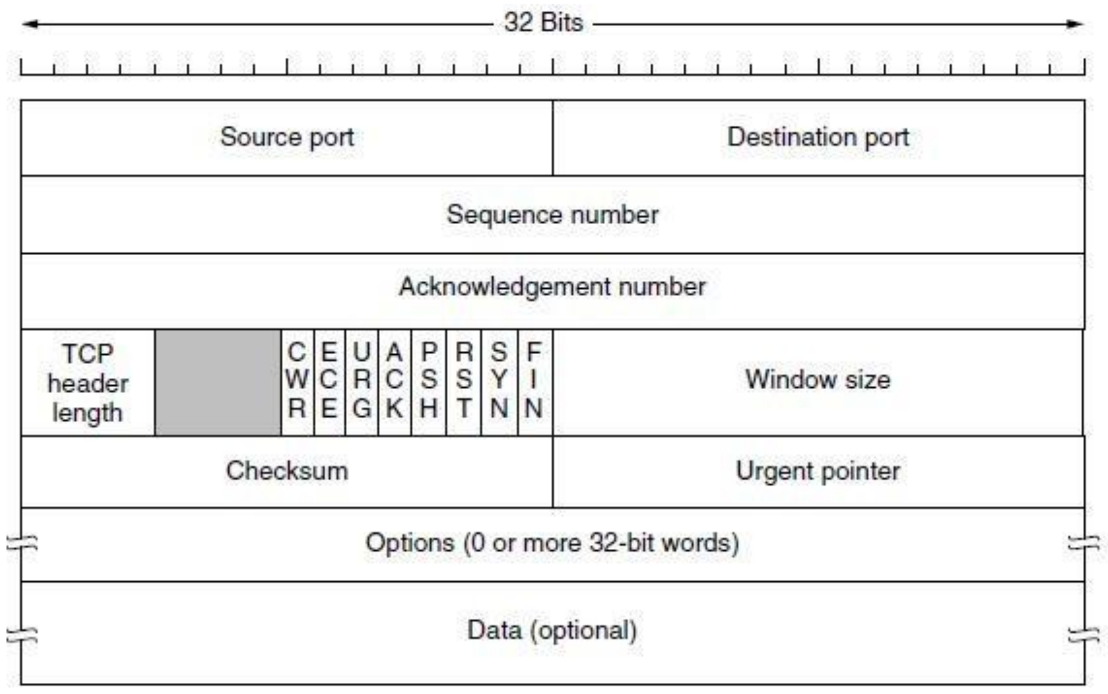


Figure. The TCP header.

The *Source port* and *Destination port* fields identify the local end points of the connection. A TCP port plus its host's IP address forms a 48-bit unique end point. The source and destination end points together identify the connection. This connection identifier is called a **5 tuple** because it consists of five pieces of information: the protocol (TCP), source IP and source port, and destination IP and destination port.

The *Sequence number* and *Acknowledgement number* fields perform their usual functions.

The *TCP header length* tells how many 32-bit words are contained in the TCP header. This information is needed because the *Options* field is of variable length.

Next comes a 4-bit field that is not used.

Now come eight 1-bit flags. *CWR* and *ECE* are used to signal congestion when ECN (Explicit Congestion Notification) is used, as specified in RFC 3168. *ECE* is set to signal an *ECN-Echo* to a TCP sender to tell it to slow down when the TCP receiver gets a congestion indication from the network. *CWR* is set to signal *Congestion Window Reduced* from the TCP sender to the TCP receiver so that it knows the sender has slowed down and can stop sending the *ECN-Echo*.

URG is set to 1 if the *Urgent pointer* is in use. The *Urgent pointer* is used to indicate a byte offset from the current sequence number at which urgent data are to be found.

The *ACK* bit is set to 1 to indicate that the *Acknowledgement number* is valid. This is the case for nearly all packets. If *ACK* is 0, the segment does not contain an acknowledgement, so the *Acknowledgement number* field is ignored.

The *PSH* bit indicates PUSHed data. The receiver is hereby kindly requested to deliver the data to the application upon arrival and not buffer it until a full buffer has been received (which it might otherwise do for efficiency). The *RST* bit is used to abruptly reset a connection that has become confused due to a host crash or some other reason.

The *SYN* bit is used to establish connections. The connection request has *SYN* 1 and *ACK* 0 to indicate that the piggyback acknowledgement field is not in use. The connection reply does bear an acknowledgement, however, so it has *SYN* 1 and *ACK* 1. In essence, the *SYN* bit is used to denote both CONNECTION REQUEST and CONNECTION ACCEPTED, with the *ACK* bit used to distinguish between those two possibilities.

The *FIN* bit is used to release a connection. It specifies that the sender has no more data to *transmit*. However, after closing a connection, the closing process may continue to *receive* data indefinitely. Both *SYN* and *FIN* segments have sequence numbers and are thus guaranteed to be processed in the correct order.

Flow control in TCP is handled using a variable-sized sliding window. The *Window size* field tells how many bytes may be sent starting at the byte acknowledged. A *Window size* field of 0 is legal and says that the bytes up to and including *Acknowledgement number* − 1 have been received.

A *Checksum* is also provided for extra reliability. It checksums the header, the data, and a conceptual pseudoheader in exactly the same way as UDP, except that the pseudoheader has the protocol number for TCP (6) and the checksum is mandatory.

The *Options* field provides a way to add extra facilities not covered by the regular header.

3.5.5 TCP Connection Establishment:

Connections are established in TCP by means of the three-way handshake protocol. To establish a connection, one side, say, the server, passively waits for an incoming connection by executing the LISTEN and ACCEPT primitives in that order, either specifying a specific source or nobody in particular.

The other side, say, the client, executes a CONNECT primitive, specifying the IP address and port to which it wants to connect, the maximum TCP segment size it is willing to accept, and optionally some user data (e.g., a password). The CONNECT primitive sends a TCP segment with the *SYN* bit on and *ACK* bit off and waits for a response.

When this segment arrives at the destination, the TCP entity there checks to see if there is a process that has done a LISTEN on the port given in the *Destination port* field. If not, it sends a reply with the *RST* bit on to reject the connection. If some process is listening to the port, that process is given the incoming TCP segment. It can either accept or reject the connection. If it accepts, an acknowledgement segment is sent back. The sequenceof TCP segments sent in the normal case is shown in Fig.(a). Note that a *SYN* segment consumes 1 byte of sequence space so that it can be acknowledged unambiguously.

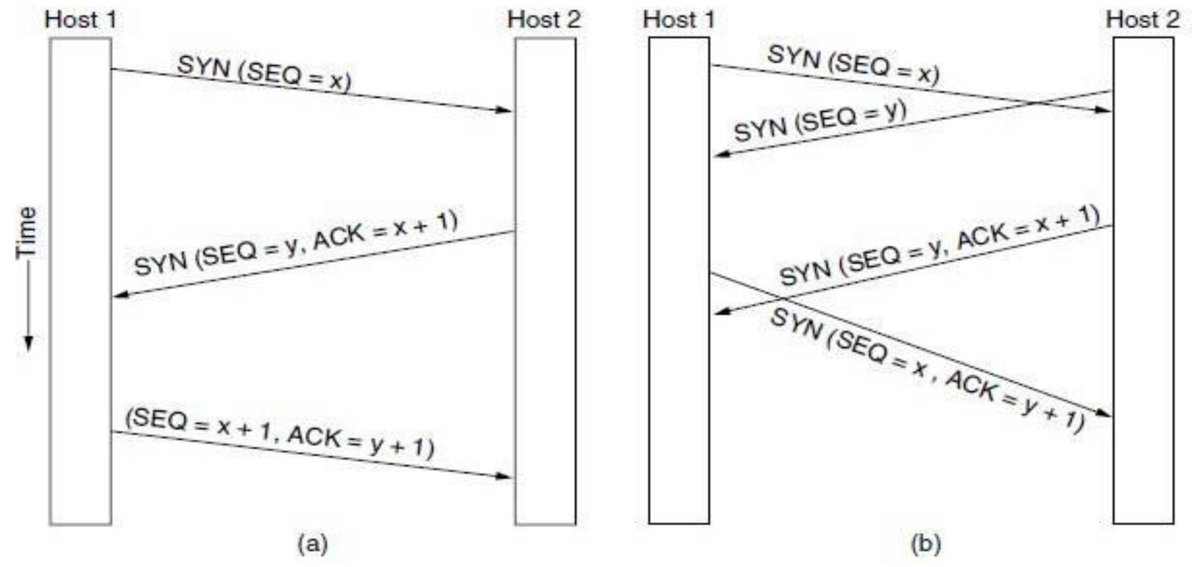

Host 1
Host 2
SYN (SEQ = x)
SYN (SEQ = y, ACK = x + 1)
(SEQ = x + 1, ACK = y + 1)
Time
(a)
Host 1
Host 2
SYN (SEQ = x)
SYN (SEQ = y)
SYN (SEQ = y, ACK = x + 1)
SYN (SEQ = x, ACK = y + 1)
(b)

Figure. (a) TCP connection establishment in the normal case. (b) Simultaneous connection establishment on both sides.

In the event that two hosts simultaneously attempt to establish a connection between the same two sockets, the sequence of events is as illustrated in Fig.(b). The result of these events is that just one connection.

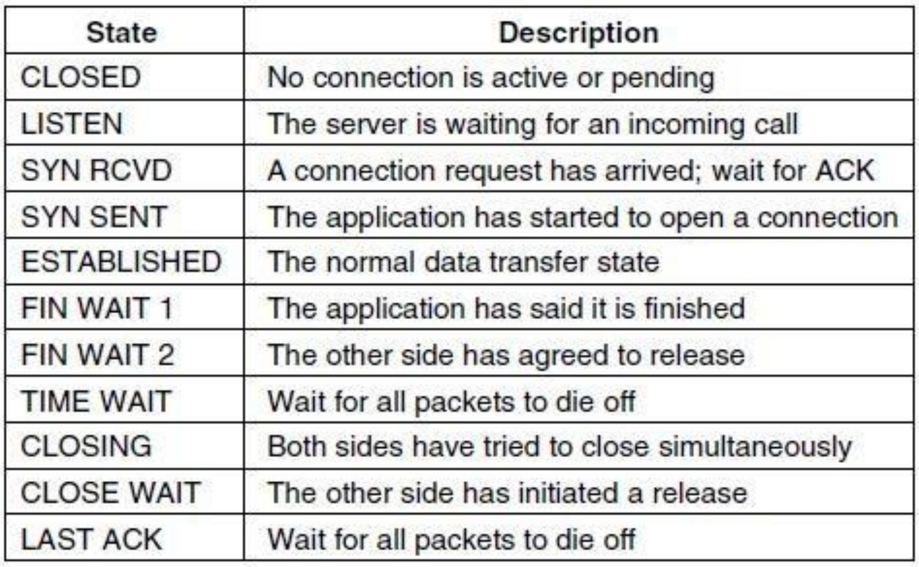
3.5.6 TCP Connection Release:

Although TCP connections are full duplex, to understand how connections are released it is best to think of them as a pair of simplex connections. Each simplex connection is released independently of its sibling. To release a connection, either party can send a TCP segment with the *FIN* bit set, which means that it has no more data to transmit. When the *FIN* is acknowledged, that direction is shut down for new data. Data may continue to flow indefinitely in the other direction, however.

When both directions have been shut down, the connection is released. Normally, four TCP segments are needed to release a connection: one *FIN* and one *ACK* for each direction. However, it is possible for the first *ACK* and the second *FIN* to be contained in the same segment, reducing the total count to three. Just as with telephone calls in which both people say goodbye and hang up the phone simultaneously, both ends of a TCP connection may send *FIN* segments at the same time. These are each acknowledged in the usual way, and the connection is shut down. There is, in fact, no essential difference between the two hosts releasing sequentially or simultaneously.

3.5.7 TCP 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 below table. 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
TIME 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

Table. The states used in the TCP connection management finite state machine.

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. The common case of a client actively connecting to a passiveserver is shown with heavy lines—solid for the client, dotted for the server. The lightface lines are unusual event sequences.

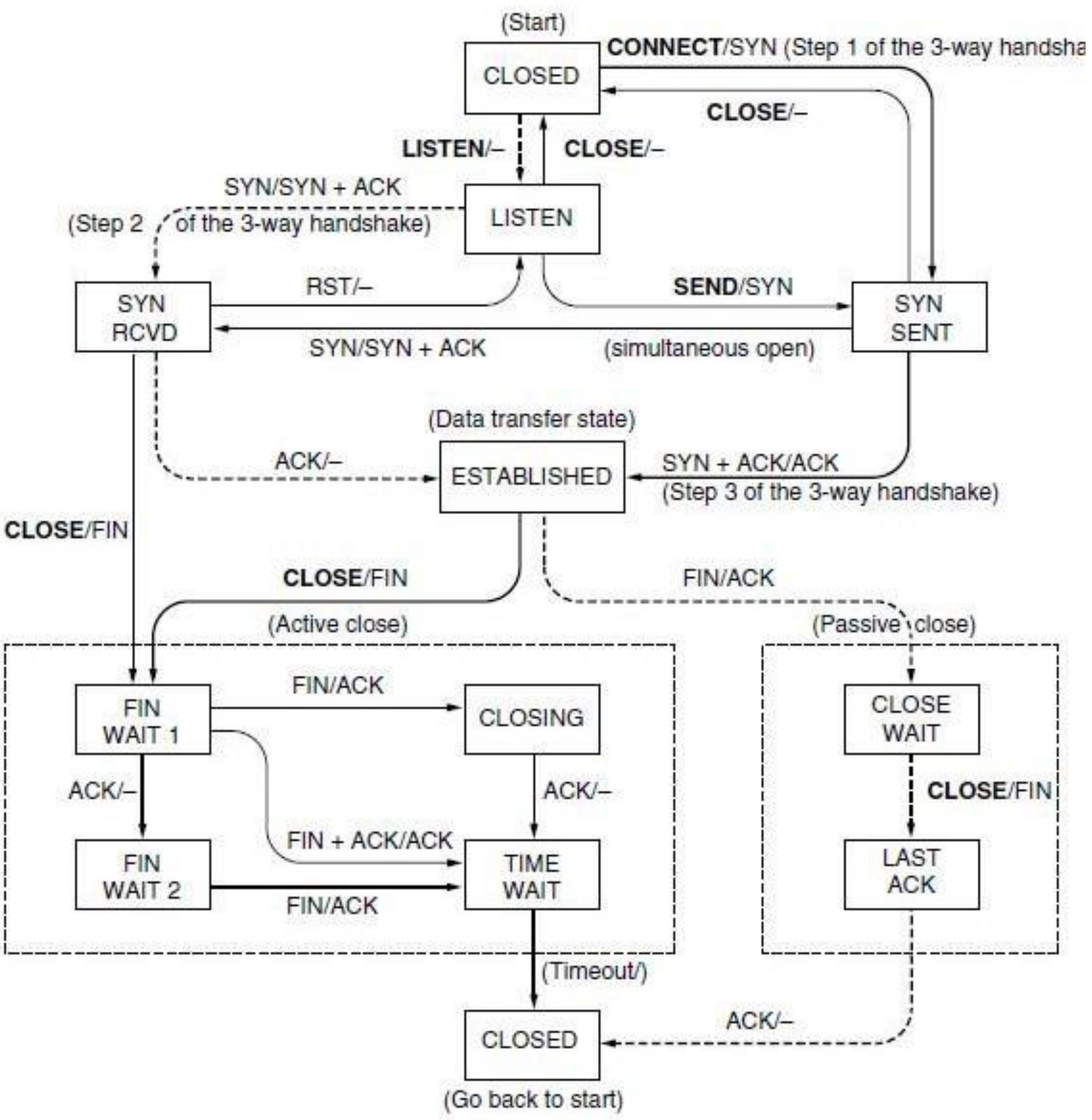


Figure. TCP connection management finite state machine. The heavy solid line is the normal path for a client. The heavy dashed line is the normal path for a server. The light lines are unusual events. Each transition is labeled with the event causing it and the action resulting from it, separated by a slash.

3.5.8 TCP Sliding Window:

Window management in TCP decouples the issues of acknowledgement of the correct receipt of segments and receiver buffer allocation. For example, suppose the receiver has a 4096-byte buffer, as shown in Fig. 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.

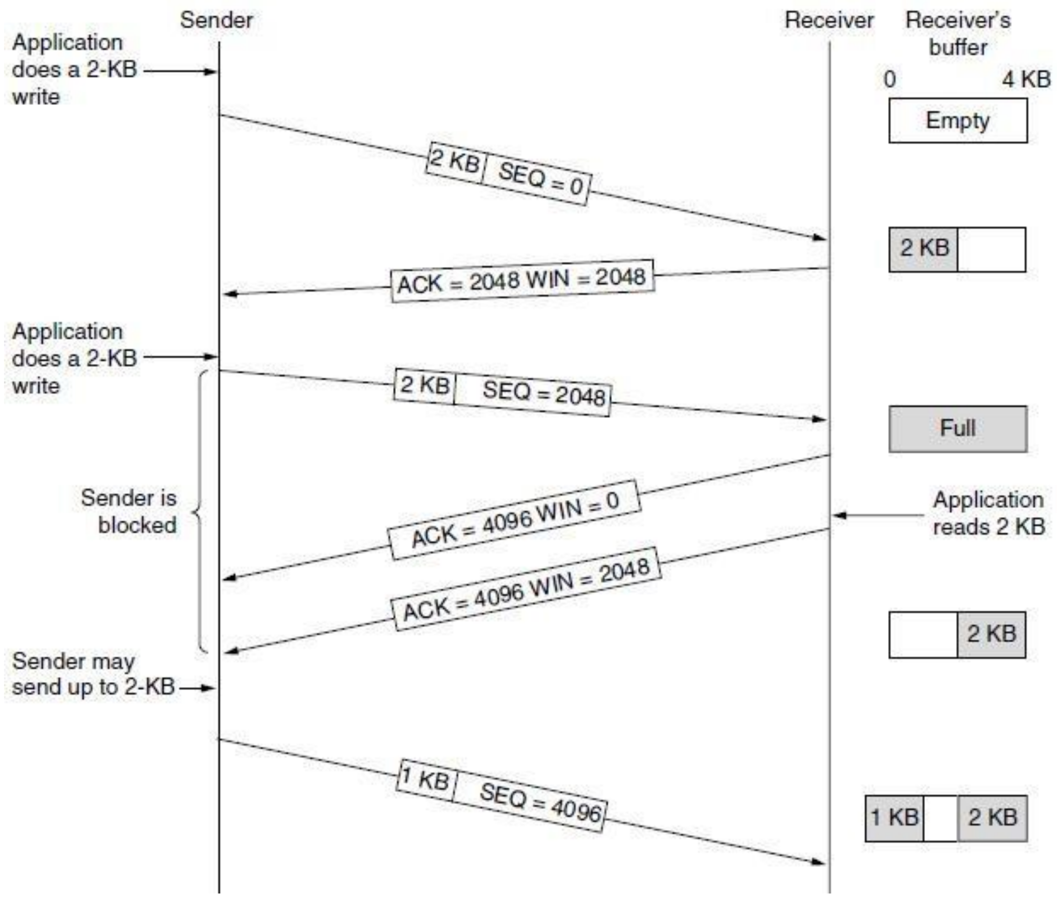


Figure. Window management in TCP

Now the sender transmits another 2048 bytes, which are acknowledged, but the advertised window is of size 0. The sender must stop until the application. Process on the receiving host has removed some data from the buffer, at which time TCP can advertise a larger window and more data can be sent. When the window is 0, the sender may not normally send segments, with two exceptions. First, urgent data may be sent, for example, to allow the user to kill the process running on the remote machine.

3.5.10 TCP Congestion Control

One of the key functions of TCP is congestion control. When the load offered to any network is more than it can handle, congestion builds up. The Internet is no exception. The network layer detects congestion when queues grow large at routers and tries to manage it, if only by dropping packets. It is up to the transport layer to receive congestion feedback from the network layer and slow down the rate of traffic that it is sending into the network. In the Internet, TCP plays the main role in controlling congestion, as well as the main role in reliable transport. That is why it is such a special protocol.

3.6 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.

3.6.1 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 willsuffer.

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 host 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. As an example, if a segment 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 segment is broadcast to 1000 machines: each one might send back an error message. The resulting **broadcast storm** could cripple the network.

Another tuning issue is setting timeouts. When a segment is sent, a timer is set to guard against loss of the segment. If the timeout is set too short, unnecessary retransmissions will occur, clogging the wires. If the timeoutis set too long, unnecessary delays will occur after a segment is lost.

Another performance problem that occurs with real-time applications like audio and video **is jitter.**

3.6.2 Network Performance Measurement:

When a network performs poorly, its users often complain to the folks running it, demanding improvements. To improve the performance, the operators must first determine exactly what is going on. To find out what is really happening, the operators must make measurements.

Measurements can be made in different ways and at many locations (both in the protocol stack and physically). The most basic kind of measurement is to start a timer when beginning some activity and see how long that activity takes.

Measuring network performance and parameters has many potential pitfalls.

1. Make Sure That the Sample Size Is Large Enough

Do not measure the time to send one segment, but repeat the measurement, say, one million times and take the average.

2. Make Sure That the Samples Are Representative

Ideally, the whole sequence of one million measurements should be repeated at different times of the day and the week to see the effect of different network conditions on the measured quantity.

3. Caching Can Wreak Havoc with Measurements

Repeating a measurement many times will return an unexpectedly fast answer if the protocols use caching mechanisms. For instance, fetching a Web page or looking up a DNS name (to find the IP address) may involve a network exchange the first time, and then return the answer from a local cache without sending any packets over the network.

4. Be Careful When Using a Coarse-Grained Clock

Computer clocks function by incrementing some counter at regular intervals. For example, a millisecond timer adds 1 to a counter every 1 msec. Using such a timer to measure an event that takes less than 1 msec is possible but requires some care. Some computers have more accurate clocks, of course, but there are always shorter events to measure too.

5. Be Careful about Extrapolating the Results

Suppose that you make measurements with simulated network loads running from 0 (idle) to 0.4 (40% of capacity).